INSTALLATION & OPERATION MANUAL
FOR THE

IVIE *1026*

DSP AUTOMIXER

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Introduction

The **1026** Integrated Automixer/Digital Signal Processor is an EtherNet® controlled, ten input by two output automixer in combination with a two input by six output DSP signal processor. The automixer is rich in functions, with automatic level control on each input, remote level control capability and sixteen programmable presets. The DSP inputs and outputs are user configurable with combinations of parametric EQ, delay, comp/limiting, and crossover functions being assignable to inputs and/or outputs. Inputs may be assigned to any or all outputs as a pre-EQ input, post-EQ input or as a post-EQ, mixed input. These assignment options can be very useful when employing a dual channel analyzer to compare pre and post-EQ signals. A total of 50 parametric filters are available to assign in any combination to DSP inputs and/or outputs.

The **1026** Integrated Automixer is an EtherNet® based product and <u>cannot</u> be controlled by the same ANSW+ software that controls Ivie matrix mixers, 1/3 octave equalizers etc. The **1026** has its own control software which is discussed in detail in **Section II** of this manual titled "**1026 Software and Programming**."

System and Software Requirements

The **1026** Integrated Automixer comes with its own software which is not compatible with other Ivie Audio Net® products. The **1026** software is a Windows® application requiring a Windows® based computer with an EtherNet® port - the faster the computer, the better. The software control screen for the **1026** will operate in a "off-line" mode without a **1026** actually being attached (this is not true of other AudioNet+ products such as the 884+). If a **1026** is attached, the software will query the unit and display its actual settings. Unlike most other AudioNet+ products that need to be attached for programming to occur, <u>the **1026** allows completely off-line programming for later upload into a unit</u>. A hard disk drive is required for overall operation of the system. A mouse is required to use this program.

This Manual

This manual is divided into two sections. **Section I: Hardware Overview**, describes the mixer and DSP signal processor hardware: input/output connections, physical connector assignments, etc. All 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, connection of inputs and outputs, remote controls and other physical functions.

Section II: 1026 Software and Programming, describes the programming of the mixers using the Ivie's Windows® based **1026** software. Computer system requirements are presented in detail, the software is explained, and programming examples are clearly presented.

Both the hardware connections and software functions of the **1026** are intuitive and user friendly. Everything necessary for a successful installation is provided within the two sections of this manual.

Section I: Hardware Overview

Power Supply

The *1026* uses the HUP45-32 universal power supply. It is UL, CSA and CE approved, and will oper-
ate at line voltages from 100 VAC to 240 VAC, 47 Hz to 63 Hz. It comes with a United States stan-
dard, 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

A reasonable question is, "What happens to the programming on the *1026* if power fails?" The answer
is that the memory is nonvolatile, so it "remembers" where it was when power went off, and will return
to that same state when power is restored.

Getting Started

Default Settings: The standard *1026* is preset at the factory so that all ten inputs are assigned to all out-
puts via both buses. All inputs are software set for microphone input levels. The Phantom Power is
turned off. The preamp gain is set to 50 dB which means that total gain through the mixer is approxi-
mately 50 dB. This is about 70% of the mixer's available gain.

Important Note: Avoid surprises (and loud noises)!! Before activating the mixer in an audio chain,
be sure that the software settings for input level, DSP functions, phantom power, etc., are set to your
preferences and needs.

1026 units 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 Output" to the "DSP Input."
A white line is drawn between these two panel jacks, indicating the standard 78C-6 connection. There
will be no audio output if this cable is either missing or improperly installed. See Figure 1, below.
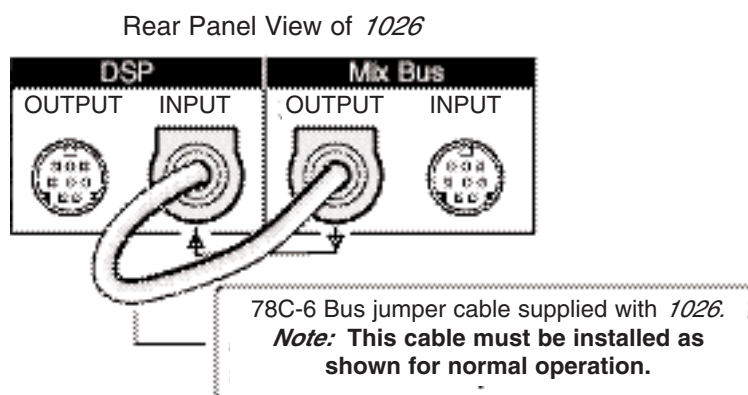

Figure 1

Successful Power Up

The HUP45-32 universal power supply is used with the *1026* and provides a broad mains voltage operating range: 100 to 240 VAC, 47 - 63 Hz. Successful powering of the *1026* can be verified by the lighting of the power indicator LED. When first powered on, the *1026* initiates a self test which checks a number of their operating parameters to verify proper performance.

Channel Inputs

All *1026* channel inputs 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. Switching from mic to line level input is a software function. The input preamp gain trim has a range of from -20 dB to +50 dB. With the input gain trim turned up to + 50 dB, (dynamic mic position), the input terminals are connected directly to the preamp. Hotter microphones will require preamp gain trim settings of less than +50 dB. A line level input will require a channel input gain trim setting of approximately 0 to +10 dB (40 to 50 dB less gain than a mic level input), since line levels are typically hotter than mic levels by about 35 to 40 dB. The preamp gain stage of the *1026* 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 *1026* and the input connector.

Connecting a Dynamic (Moving Coil) Microphone

A standard dynamic microphone connects to the *1026* via the Euro connectors located on the rear panel. Pin one of the microphone is connected to the ground terminal on the connector. Pins two (+) and three (-) are connected to their respective terminals.   Phantom Power should be OFF. A typical connection is shown in Figure 2, below:
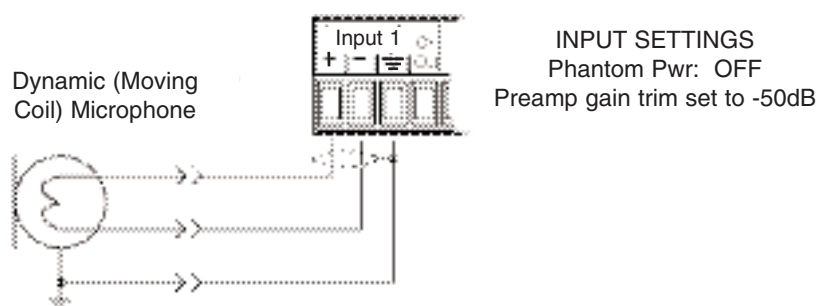

Figure 2

Connecting a Condenser Microphone

Condenser microphones connect to the *1026* via the Euro connectors located on the rear panel. 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.   Phantom Power should be ON. Since condenser mics are usually "hotter" than dynamic mics, the preamp gain setting should be less than for dynamic mics by some 10 dB to 20 dB. A typical connection is shown in Figure 3, following:
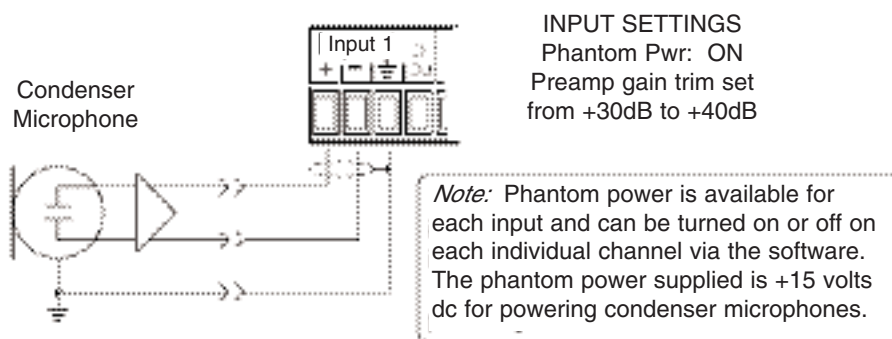
INPUT SETTINGS
Phantom Pwr: ON
Preamp gain trim set
from +30dB to +40dB

Condenser
Microphone

Input 1

Note: Phantom power is available for each input and can be turned on or off on each individual channel via the software. The phantom power supplied is +15 volts dc for powering condenser microphones.

Figure 3

Connecting a Balanced Line Level to a Channel Input

Line level inputs are connected to the *1026* via the Euro connectors 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, it should not be connected at the *1026*. 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 of the channel input. The input preamp gain trim should be set at about 0 to +10 dB, depending on how "hot" the line level input is. The Phantom Power should be turned OFF. Figure 4, below, details these connections:
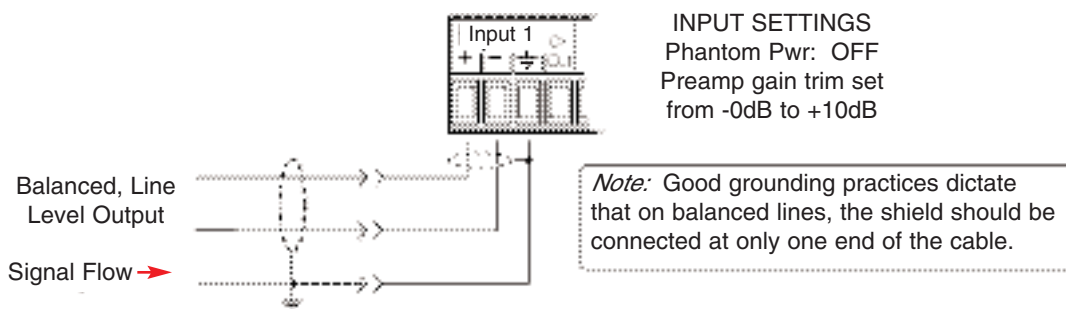
INPUT SETTINGS
Phantom Pwr: OFF
Preamp gain trim set
from -0dB to +10dB

Input 1

Balanced, Line
Level Output

Signal Flow

Note: Good grounding practices dictate that on balanced lines, the shield should be connected at only one end of the cable.

Figure 4

Connecting an Unbalanced Line Level to an Input

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 input gain trim should be set somewhere between 0 and +10 dB. Phantom Power Switch should be set to the OFF position, as shown in Figure 5, following:
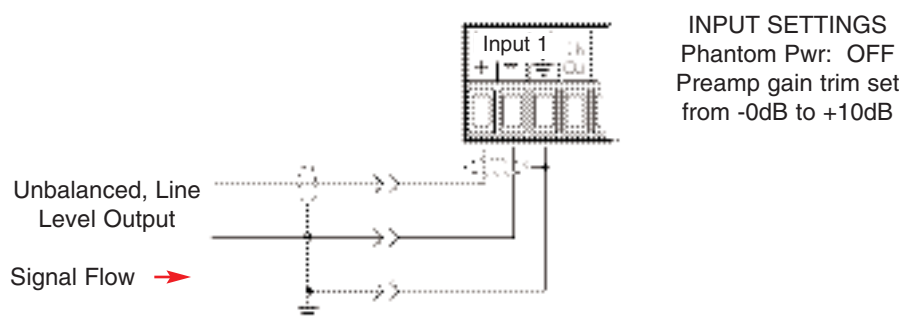

INPUT SETTINGS
Phantom Pwr: OFF
Preamp gain trim set
from -0dB to +10dB

Input 1

Unbalanced, Line
Level Output

Signal Flow

Figure 5

Master Outputs

The *1026* has six master outputs. They are referred to as master outputs 1 through 6. Each master output is electronically balanced. The master outputs are capable of driving a 600 Ω load to a maximum of +18 dBu.

Connecting a Master Output to a Balanced Line Level Input

The output of the *1026* 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 6 on the following page:
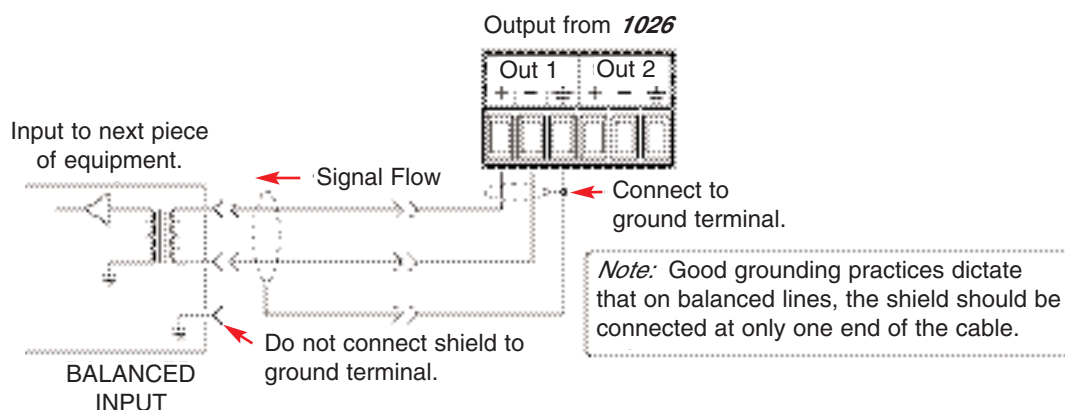
Output from *1026*

Figure 6

Connecting a Master Output to an Unbalanced Line Level Input

The output of the *1026* 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 7, following, shows this connection in detail:
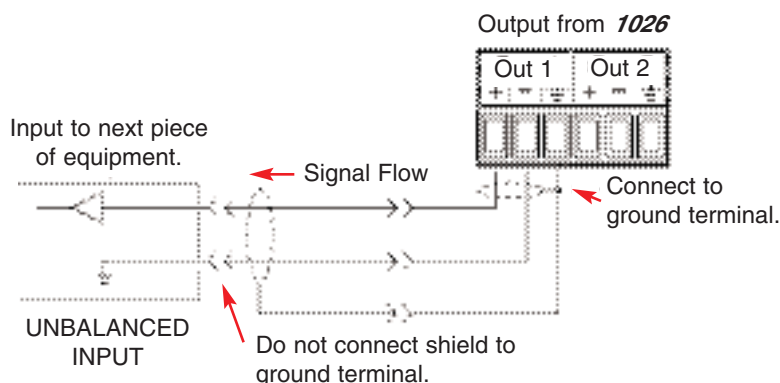
Output from *1026*

Figure 7

Channel Direct Outputs

Each *1026* channel has an unbalanced, Direct Output which accesses analog audio for that individual channel input only (the Direct Output is + and the ground of the associated input is -). If an input

channel is not assigned to one of the two signal buses, the only output from that channel would be found at the Direct Output for that input channel. The Direct Outputs provide a very useful feature. Any input channel, for example, could be used as a microphone-to-line preamplifier. The (unbalanced) line level output would appear at the channel's Direct Output. The outputs can be taken as separate outputs for any number of purposes. The Direct Output is connected through a 604 Ω resistor to the Euro terminal block on the rear panel. 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 6 dB drop in gain when-ever the number of outputs being connected together is doubled.

Level Controls

There are sixteen level controls on the *1026*, ten for the channel inputs and six 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

"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 "slider pots" of the *1026* software.

The Level control works as an in-line attenuator and can attenuate signal from 0 dB to -99 dB. It is very helpful to think of "turning down" the volume as opposed to "turning up" the gain. The gain for the channel is set at input preamplifier stage. The Level control acts as an attenuator to the audio sig-nal through that channel.

Range Control

As previously noted, the Level control has an adjustment range of 100 dB. The Range control (accessed via a pop-up window in the *1026* software) 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. If, for example, the volume control range were limited to 20 dB, the sound sys-tem could never be turned all the way off. There would always be some sound from the system when somebody talked into or tapped on a microphone. If there were always some sound, operators would not mistakenly believe a system turned down below the audible level had died, or was not functioning. They would, instead, begin to look for a volume control to turn the system up.

The Range control limits the range of the volume control. Instead of the full 100 dB 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 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 (Preamp Gain Trim)

Trim is a very powerful, software based feature of the *1026*. 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 Trim is also used to allow for microphone versus line level input signals, as previously discussed. Keep in mind that the majority of gain in the channel is set in the input preamplifier stage. The Trim adjustment has a range of from -20 dB to +60 dB, which allows not only for microphone or line level input configurations, but also can be used to permanently limit the maximum gain available to a channel. Figure 8, following, shows the relationship in the Level, Range and Gain Trim adjustments:
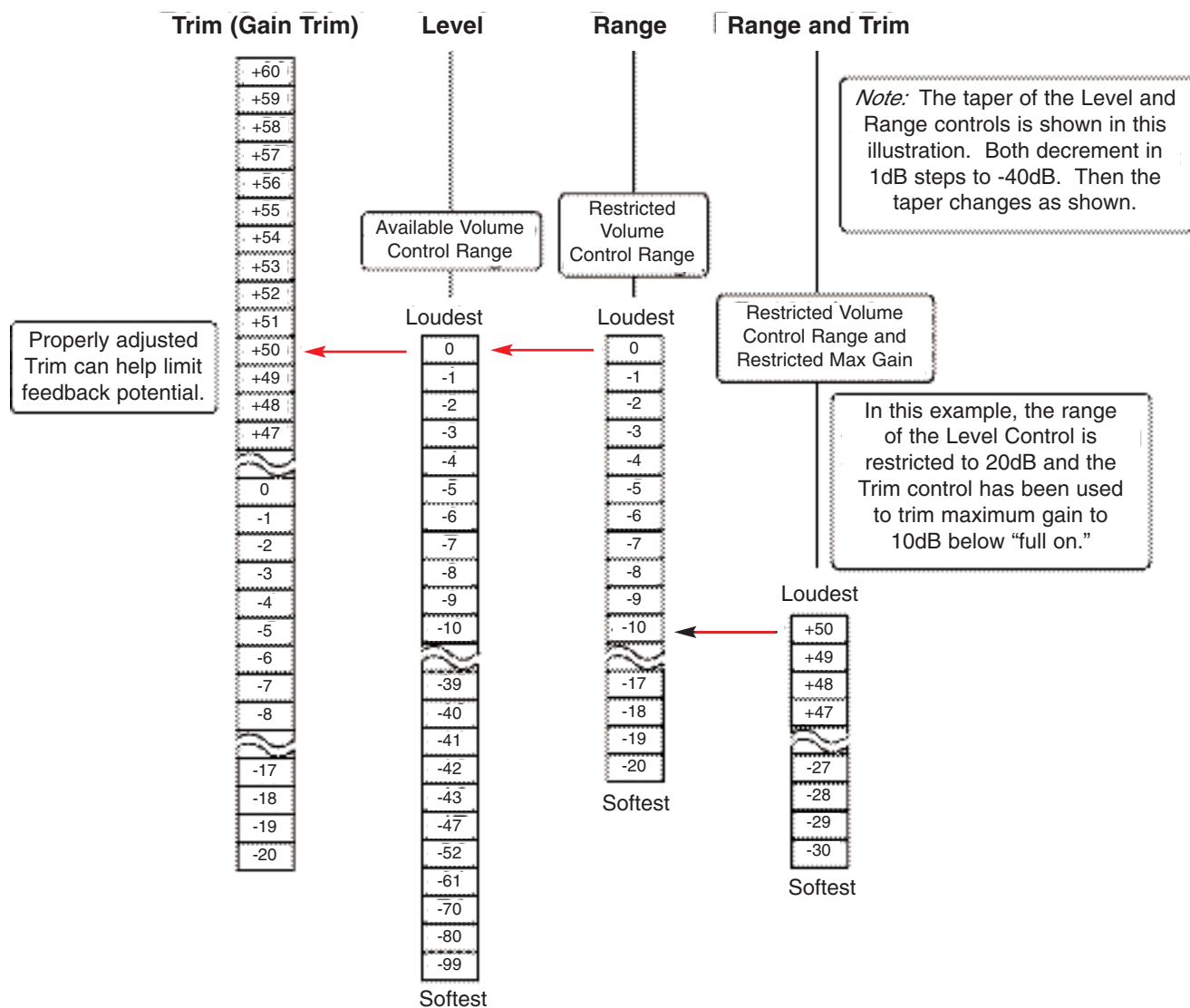
Trim (Gain Trim)	Level	Range	Range and Trim
+60			*Note:* The taper of the Level and Range controls is shown in this illustration. Both decrement in 1dB steps to -40dB. Then the taper changes as shown.
+59			
+58			
+57			
+56			
+55	Available Volume Control Range	Restricted Volume Control Range	
+54			
+53			
+52			
+51	Loudest	Loudest	Restricted Volume Control Range and Restricted Max Gain
+50	0	0	
+49	-1	-1	In this example, the range of the Level Control is restricted to 20dB and the Trim control has been used to trim maximum gain to 10dB below "full on."
+48	-2	-2	
+47	-3	-3	
	-4	-4	
0	-5	-5	
-1	-6	-6	
-2	-7	-7	
-3	-8	-8	
-4	-9	-9	Loudest
-5	-10	-10	+50
-6			+49
-7	-39	-17	+48
-8	-40	-18	+47
	-41	-19	
-17	-42	-20	-27
-18	-43	Softest	-28
-19	-47		-29
-20	-52		-30
	-61		Softest
	-70		
	-80		
	-99		
	Softest		

Properly adjusted Trim can help limit feedback potential.

Figure 8

Note: The normal full range of the VCA is 100 dB (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 Gain Trim control can be used to limit the maximum gain available. If the maximum gain available is limited by the Gain Trim control, the full 0 to -99 dB

of attenuation is still available below the maximum level that has been set by the Gain Trim control, unless, of course, that range of attenuation has been limited by using the Range control.

Remote Volume Controls

Please note: Before any remote control can become active, the *1026* must be programmed to accept input from a remote control.

The ten input channel volume controls of the *1026* and the six master output volume controls may all be controlled remotely. 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 *1026*.) All power required by the remotes is supplied by the *1026*. The RMT (remote) voltage supplied by the *1026* 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, except LED's.

The control range of all remotes follows the programming of the *1026* for channel and master volume controls. The programming can be used to limit the range of the remote control (Range Function) and the maximum level (Trim Function) of the remote volume controls. A limited range plus "OFF" can also be programmed.

Using a Potentiometer Remote

Please note: Before any remote control can become active, the *1026* 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 *1026* and other 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.

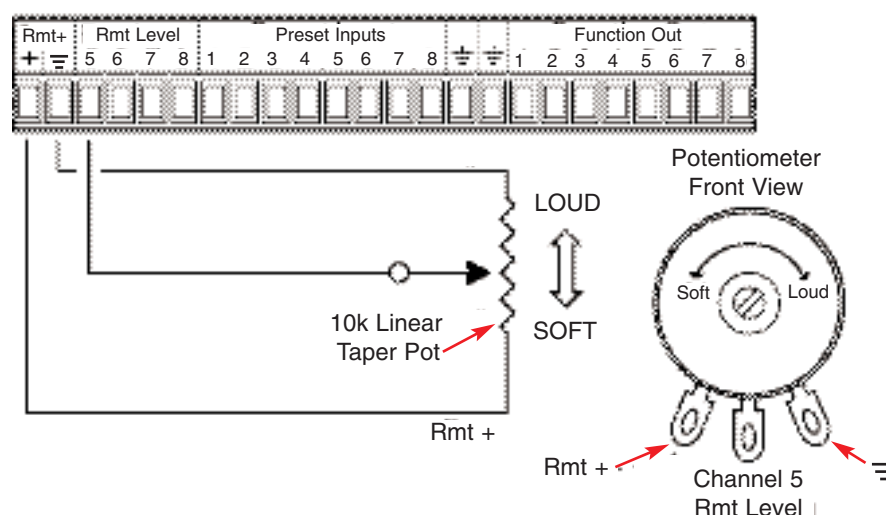A standard 10k Ω, linear taper, potentiometer is connected as illustrated in Figure 9, following.

Figure 9

Note: In Figure 9, notice that the potentiometer is connected across the RMT voltage and ground. The

wiper of the pot 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, except LEDs.

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 full 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 *1026* 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 *1026* 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 the *1026*. The RMA-10L 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-6L may be used, which also comes with a matching wall plate and connector assembly.

Using Up/Down Push Button Remotes

Please note: Before any remote control can become active, the *1026* must be programmed to accept input from that remote control.

Ivie manufactures two types of push button remote level controls that can be used with the *1026*. 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 *1026* and are therefore connected the same way that potentiometer remotes would be connected. In order for an RM-1 or RM-2 to function, the *1026* must have its software "Remote Pot" control enabled. The RM-1 and RM-2 have on-board microprocessors, so they must be fed power from the *1026* 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 LEDs 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, the *1026* must have its software "Remote Button" control enabled. The *1026* provides the necessary logic for RB-1's to track each other in room combining applications.

Making Custom Up/Down Buttons

As illustrated in Figure 10, 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 *1026*. The RMT voltage supplied by the *1026* 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, except LEDs.

Connection of Push Button Remote Volume Controls

UP/DOWN Pushbutton
(Shown in parallel with a second remote control)
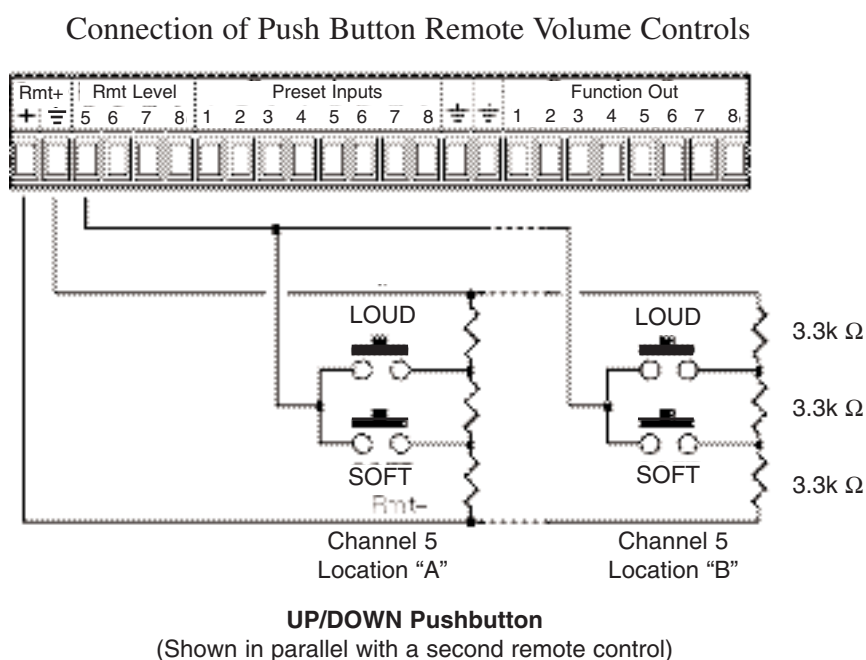
Figure 10

Hardware Activation of Presets

Presets can be activated remotely on the *1026* via a switch connected to the appropriate terminals on the rear panel. Presets can also be activated via the *1026* software. There are sixteen presets associated with the mixer section of the *1026* which are not mutually exclusive. In other words, more than one of

these presets can be active at a time. There are also sixteen presets associated with the digital signal processing section of the *1026* which are mutually exclusive, that is, only one of these presets can be active at once.

While there are sixteen presets for the mixer section and sixteen presets for the DSP section of the *1026* (all of which can all be activated or deactivated via the *1026* software), there are only eight hardware connections on the rear panel of the *1026* for activating presets. This means that there are only eight preset combinations which can be activated by an external contact closure.

One contact closure between a Preset Input and ground (using a switch or relay) actually performs three simultaneous functions. A contact closure between Preset Input 1 and ground will activate Preset 1 in the mixer section, Preset 1 in the DSP section, and additionally, will activate Function Out 1. Figure 11, following, details the connections for activating presets.
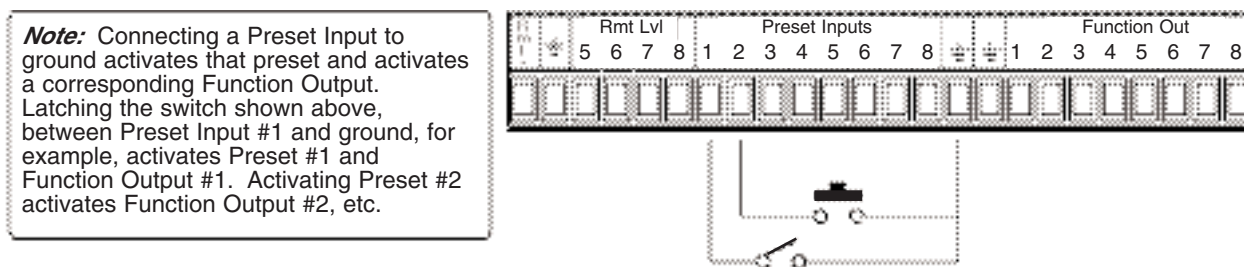
Figure 11

Activating Presets on Two Units at the Same Time

Presets on two different *1026*'s may be activated at the same time. Diodes must be used to isolate one *1026* from the other. A standard 1N914 diode can be used, as shown in Figure 12, following:
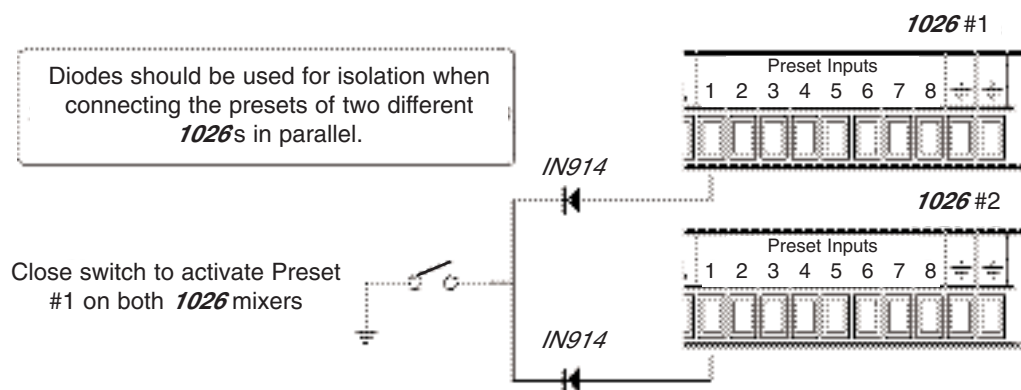
Figure 12

Function Output

Each Function Output on the *1026* 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 as part of a preset group as explained previously under the heading **Hardware Activation of Presets** found on pages 11 and 12. It can be used to light an LED, or with an external (installer provided) power supply, could be used to control an external relay.

On the *1026*, Function Output number 1 is activated when Preset 1 is activated, Function Output 2 is activated when Preset 2 is activated, and so on.

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 13, below, shows an LED connected to the Function Output. The Remote+ terminal on the rear of the *1026* has a built-in power supply voltage for use with LEDs and remote controls. It must not be used for any other purpose than LEDs and remote controls. Figure 13, following, shows the proper connection of an LED used to provide visual indication of Function Out status:
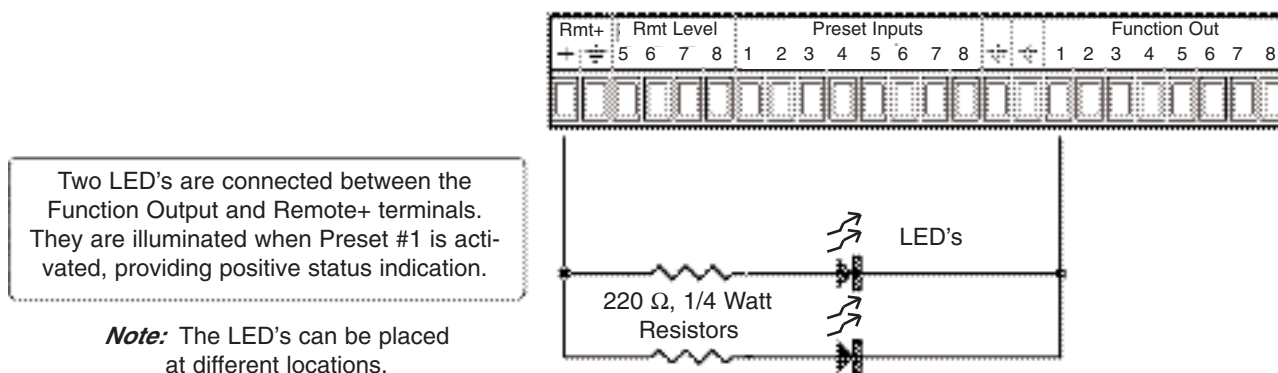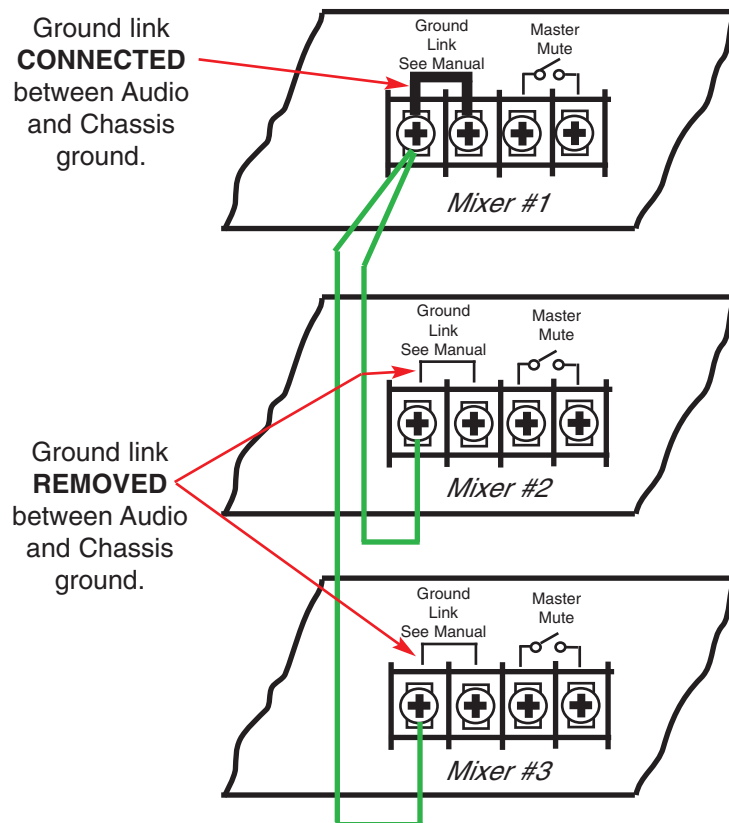

Two LED's are connected between the Function Output and Remote+ terminals. They are illuminated when Preset #1 is activated, providing positive status indication.

Note: The LED's can be placed at different locations.

LED's

220 Ω, 1/4 Watt Resistors

Figure 13

Ground Link Connections

The four position terminal block located on the rear of the *1026* 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 14 below shows this method of grounding. "Star grounding" greatly reduces noise problems due to ground loops between units.

Ground link **CONNECTED** between Audio and Chassis ground.

Ground Link See Manual   Master Mute

Mixer #1

Ground Link See Manual   Master Mute

Mixer #2

Ground link **REMOVED** between Audio and Chassis ground.

Ground Link See Manual   Master Mute

Mixer #3

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 the 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 together.

Figure 14

Please note that the unit 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 unit. This separate wire runs all the way from each unit back to the first *1026*, which is very important. Do not daisy chain from one Audio Gnd terminal to another and then to the first *1026*. When connecting to the first unit, the Chassis Gnd terminal can be used if the Audio Gnd terminal gets too congested.

Section II: *1026* Software and Programming

Getting Started

Software Installation

1026 software is shipped with the *1026*. An install utility 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 *1026* software is for Windows® (98 or above) operating systems. To install the software, insert the *1026* software CD into the computer's CD drive, navigate to the CD and click on the "setup.exe" file.

Alternatively, download the software from the Ivie website to your desktop, then navigate to the downloaded file and double click on it. Ivie recommends that you download the software from our website as you will be assured of having the latest version.

Standard installation of the *1026* software will install an icon on your desktop screen, which can be used to start the software. The *1026* software can be run off-line, with no *1026* connected. This can be very useful to the installer or consultant who wishes to do some setup prior to arriving at the job site. Configurations can be changed, presets programmed, levels set, and so on, with the off-line software. This programming may then be saved to a file and uploaded to the hardware at a later time.

EtherNet® Connections Between a Computer and a *1026*

Wireless EtherNet® Connection to a *1026*

An interesting possibility available with the EtherNet® controlled *1026* is to connect it to the computer using a wireless link. This can be done using a relatively inexpensive wireless EtherNet® hub and can allow the installer or consultant to make level and signal processing adjustments from a laptop or other computer from any location within range of the wireless link. Connection is intuitive and not difficult. Standard EtherNet cables would be used (not crossover cables) and standard wiring for EtherNet® hub usage applies.

Standard EtherNet® Connection to a *1026 (Crossover Cable Required)*

On the rear panel of the *1026* is a standard RJ45 connector. Since the *1026* is a standard EtherNet® device, connection from a computer to the RJ45 EtherNet® port on the back of the *1026* is done in the same way one would connect any EtherNet® device to a computer. A direct, CAT 5 cable connection from the EtherNet® port of a computer to the RJ45 EtherNet® connector on the back of the *1026* is possible. A "crossover" cable is required for this type of connection. If connection is made using a hub or switch between the computer and the *1026* (such as in the wireless connection described above),a crossover cable is not required. Figure 15, on the following page, shows the proper wiring for a crossover cable:
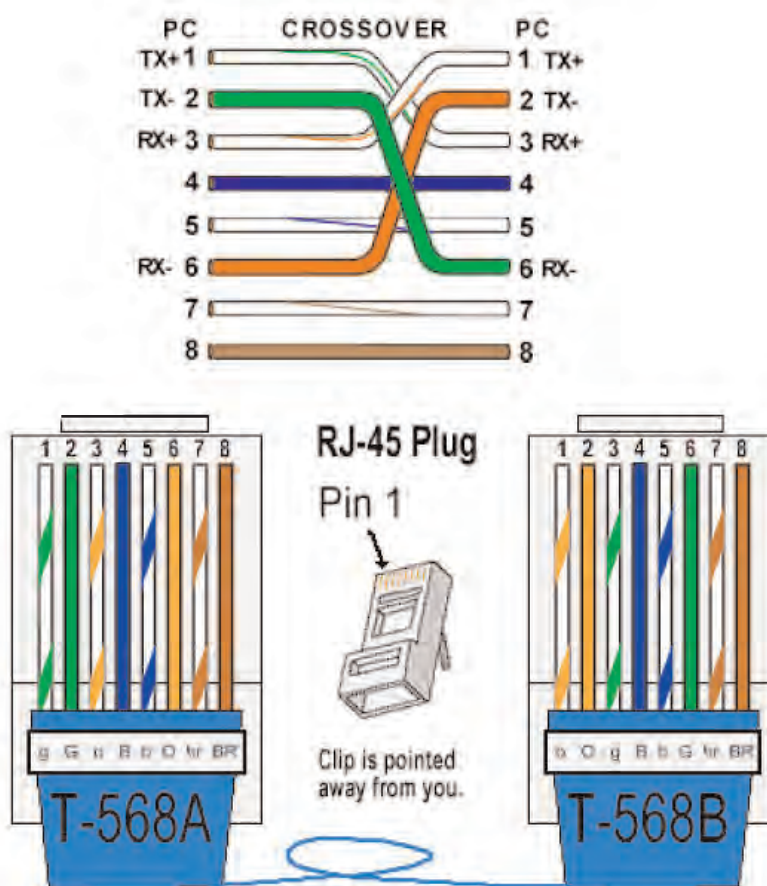
Figure 15

Additional Software: 485MON32

The CD included with the *1026* contains another highly useful program in addition to the Ivie *1026* software (which can also be downloaded from the Ivie website). It is called 485MON32. This is a monitoring program that is used while plugged into the EtherNet® connection of the *1026*. 485MON32 can be a tremendous help in programming external control devices such as touch screens.

An "EtherNet 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 EtherNet® products. Use of the 485MON32 program when programming is very beneficial and highly recommended.

Starting the *1026* Software with a Unit Connected to the Network

Successful connection of a *1026* to the network is indicated by the green LED on the front panel of the

1026. When a computer is also successfully connected to the network as well, starting the *1026* software will bring up the first software screen, the Network Information Screen. The window on the right will show any *1026* units that are found on the network. The window on the left will show any *1026* units on the network that are "IP compatible" with your computer's network setup. If a mixer is identified in the right-hand window, but not in the left-hand window, it is not currently network-address compatible, and therefore, cannot be controlled with your computer's network settings. Because the *1026* is an IP address-able device, it comes with a default IP address and subnet mask programmed into it. The default IP address in the *1026* may not (probably will not) be compatible with the IP setup in your computer.

To make a compatible connection between your computer's network port and the *1026*, something will have to change. You COULD change the networking setup of your computer to that of the *1026*, but that is inconvenient, at best. The Ivie *1026* software provides a means of changing the IP setup of the *1026* to match that of your computer. With a single software button push, you can change the *1026* networking setup to be compatible with your computer.

Simply click on the "Match Subnet" button in the Network Info screen. The previously incompatible *1026* will be adjusted to operate on your computer's networking settings, and is now identified in the left-hand "Units Controllable From This Computer" window. If, in the future, a different computer needs to be used to adjust the *1026*, this IP adjustment procedure can be done again as the software is started up to change the *1026* to the new computer's network settings.

Programming the *1026*

Presets

A *1026* has sixteen, nonexclusive presets available on the mixer side, and sixteen exclusive presets on the DSP section. On the DSP side, configuration and setup information can be saved to one preset only, or to all sixteen presets. Thus, universal programming can be saved to all presets, while specific information can be saved to one preset only. Additionally, the binary logic preset port configuration feature allows a combination of mixer and DSP presets to be activated simultaneously by a software command, or a contact closure. There are eight contacts on the rear panel to provide the binary logic closures which results in 256 possible preset activation combinations.

Programming Preparation

Wiring a *1026* into a system, as described in detail in Section I of this manual, is simple and straight forward. Standard audio wiring and grounding techniques should be employed. Inputs and outputs are clearly labeled on the rear chassis and the detachable Euroblock connectors are labeled as well. Front panel LEDs clearly indicate signal passing through the unit. Once the *1026* has been wired into the system and the *1026* software has been loaded, programming can begin.

When the *1026* software is activated, it first polls the network to locate any units on the network. After polling is complete, the default screen for the *1026* software comes up. This control screen, as shown in Figure 16 on the following page, depicts the architecture of the *1026* in graphic form. The ten inputs are shown on the left, the Routing Matrix at the top and the six outputs on the right. The various signal processing modules that can be activated or deactivated for the inputs and outputs are depicted by the blocks labeled "DSP" on the two input buses and the six outputs.
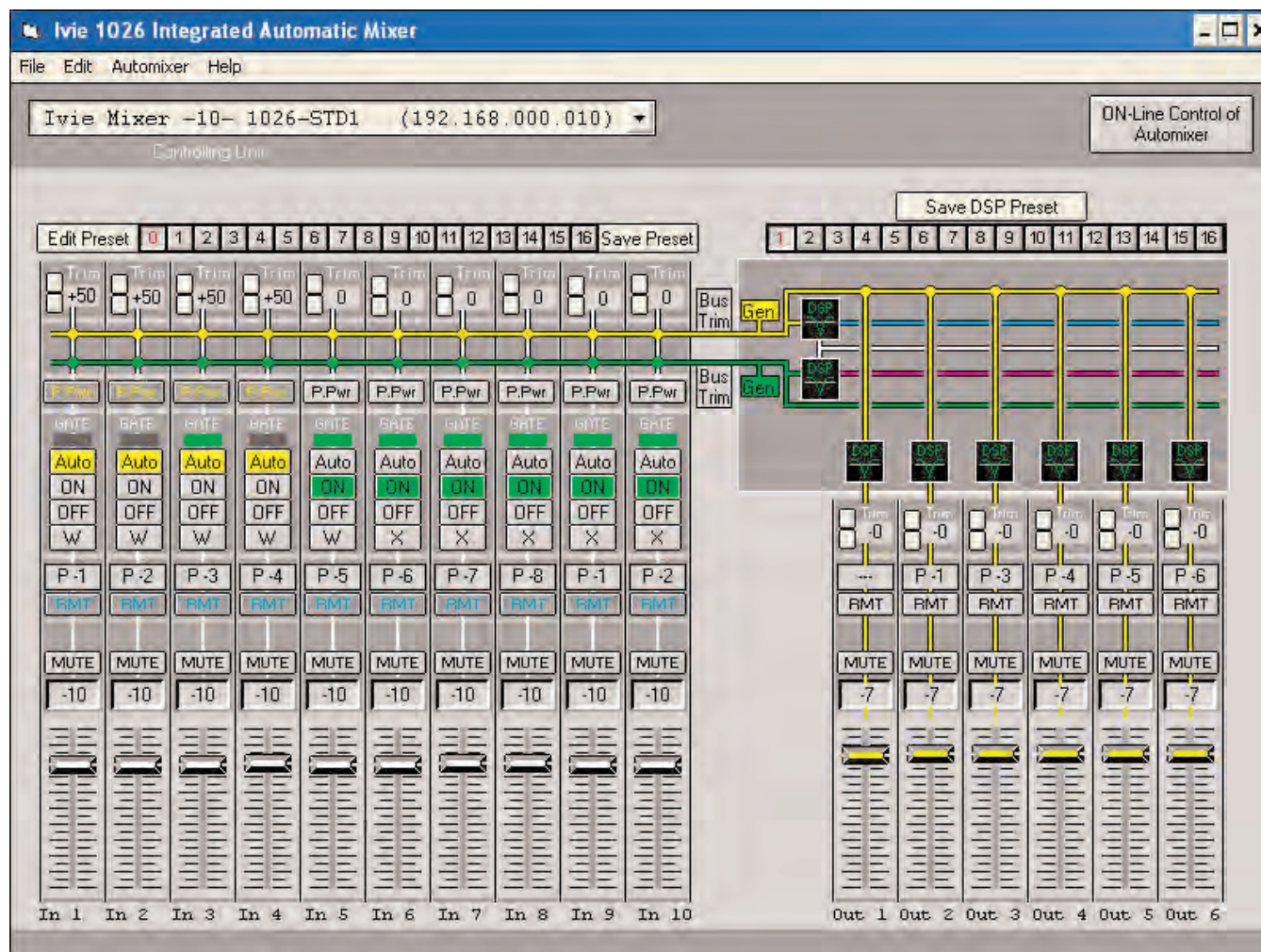
Figure 16

Important Note: Left clicking on the "DSP" block brings up the PEQ Filter Control Pop Up Window. Right clicking on the "DSP" block activates a menu allowing selection of pop up control windows for Delay, or Compressor as well as for Equalization

1026 Main Control Screen

The first step in programming the *1026* is to configure the unit for the signal processing requirements of the application. To configure the unit, the DSP Configure pop-up window (located under the "Edit" pull-down menu of the main control screen) is used as described below. Configuring a *1026* using the DSP Configure pop-up window is one of the first steps in programming the unit.

DSP Configure (Edit Pull-down Menu)

Accessed from the "Edit" pull down menu, the DSP Configure pop-up window allows the creation of one

or more configurations to more nearly match the specific needs of an application. While the architecture of two in by six out is fixed, the signal control modules (PEQ, delay compression/limiting, crossovers etc.) can be activated or deactivated for each input and output, allowing many possible configurations. The DSP Configure pop-up window is shown in Figure 17 below:
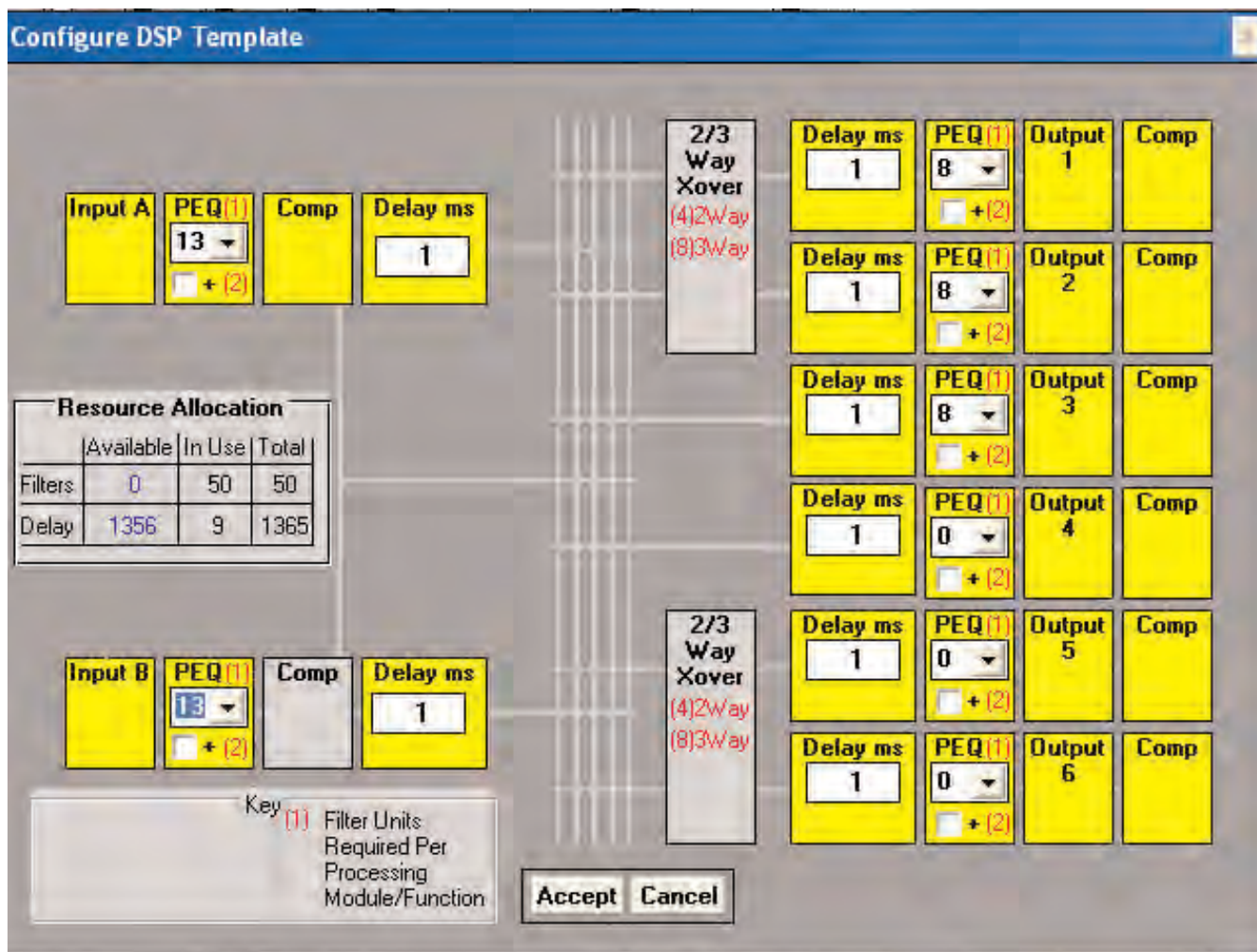
Figure 17

The number of PEQ filters, amount of delay, etc. can also be distributed to inputs and/or outputs as desired. A configuration can be saved to one preset or to all presets. Thus, a desired common configuration can be created and saved to all presets with just one click of the mouse, or a complete reconfiguration of the *1026* with each preset is possible.

Mix Bus Input Path

One of the individual functions in the configuration screen, crossovers, can be turned "on" or "off," depending on whether that function is needed for the intended application. A gray color indicates a disabled crossover and yellow indicates an enabled crossover. Placing the cursor on a crossover and clicking

will alternately enable and disable the crossover. Once a configuration has been selected, that configuration can be saved to one or to all presets.

The next modules shown in the signal processing chain are the PEQ modules. Clicking on the number tab allows the selection of the appropriate number of PEQ filters in the input path for the required application. Each of the filters can be individually defined as parametric, High-cut Low-cut, Hi-shelf, etc. as required. It may be necessary to configure a filter such as a Low-cut filter with a sharper or more complex characteristic. This type of filter is called an "extended" filter, due to its extended capabilities, and it consumes two of the fifty available "filter units," instead of one.

To enable extended filters, click on the "+" box directly beneath the number tab in the filter module block. The small "(2)" indicates that using a more complex filter will allocate two filter units instead of one. Clicking this box makes available sharper and more complex filter types, such as Butterworth, Bessel, and Linkwitz-Riley, on either end of the spectrum.

Resource Allocation Table

Before making changes to the PEQ modules, it is necessary to consider the Resource Allocation Table, found centered on the left side of the DSP Configuration screen (see Figure 17, on the preceding page). The first line of the table indicates allocation of the PEQ filters in the *1026*. The total number of filter units available is fifty. Both PEQ's and Crossovers draw filter units from this bank of fifty filter units. These filter units can be allocated throughout the inputs and outputs on the DSP Configuration screen as they are needed. The total of fifty filter units cannot be exceeded.

After the PEQ function module, the next module in the input path is a Compressor/Limiter. A Delay Module follows the Compressor/Limiter. It is possible to select the maximum amount of delay allocated to each of the Delay Modules. The second line in the Resource Allocation Table is very helpful in this process. Up to 1,365 milliseconds of delay is available for allocation.

Going from left to right on the DSP Configure screen (Figure 17), following the input Delay Modules, is the Routing Matrix. The Routing Matrix is discussed in detail in the section of the manual titled "Signal Routing" (see page 21). Located between the Routing Matrix and the Output Paths, are the Crossover Modules. Clicking on one of the Crossover Modules brings up one of three states: Inactive, 2-Way or 3-Way. Select the appropriate state for the application by successive clicking. *Note:* The slopes of the crossovers generally require use of steeper filters, thus a greater number of filter units will be employed in the Resource Allocation Table when crossovers are selected.

Output Paths

Each of the six outputs of the *1026* has an output path with Delay, PEQ, and Compressor/Limiter Modules. Once the *1026* has been configured, it is time to return to the main *1026* control screen to assign the signal routing and complete the setup of the unit. This is accomplished by pressing the "Accept" button.

Signal Routing

The *1026* control screen shows the two inputs of the unit feeding several audio buses. There is a pre-EQ and post-EQ feed from each input, as well as a mixed feed from both inputs. Each of these feeds can be routed to any output. To complete the routing of an input to an output, place the cursor on the Routing Matrix graphically illustrated (with vertical lines) between the inputs and outputs. Select an output for connection by placing the cursor on the Routing Matrix directly in line with the red line which graphically illustrates the feed to the output. Click the mouse. A connection will be made and will appear on the screen. Successive clicks of the mouse will toggle through each of the possible input feeds to the selected output. The signal path illustrated will change color each time a change is made to help clarify the selection.

An input to output connection that has been made is illustrated in Figure 18 below:

Cursor Placement on the Routing Matrix for Making Input to Output Connections
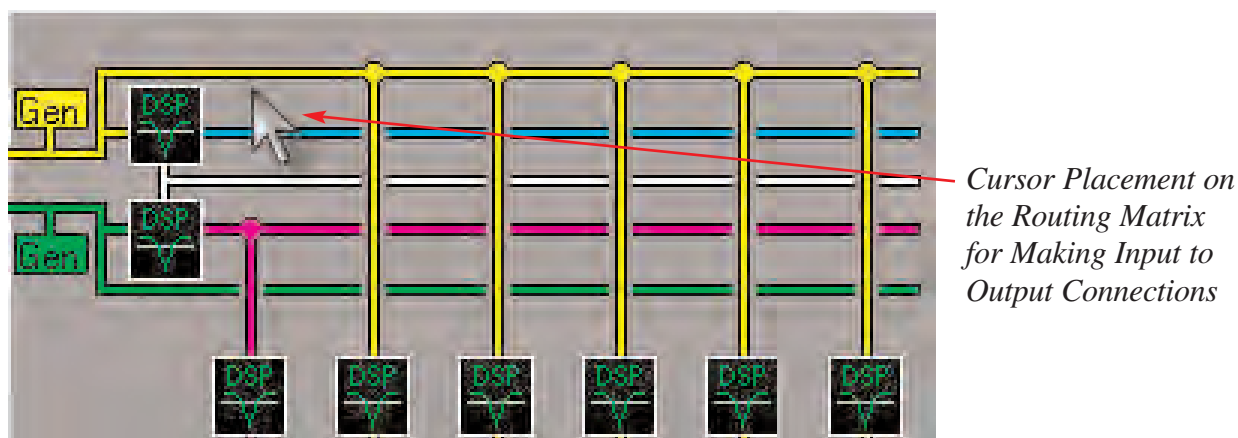
Figure 18

As shown, the post-EQ feed from input B has been routed to output number 1. Notice the position of the cursor on the Routing Matrix directly across from Output 1. The cursor must be in this position to make connections to Output 1. The signal can be routed to the remaining outputs in the same way. Once the signal routing has been completed, the unit is ready for the next programming step, setting up the individual signal processing modules.

Note: The "View Labels" button has been pressed, in Figure 18, above which causes the labels for the inputs and outputs to be displayed. Factory default names on the labels are Inputs A and B and Outputs 1 through 6. Placing the cursor on the "View Labels" button and "right clicking" will activate a pop-up window which allows the inputs and outputs to be custom named.

Setting Bus Levels

The *1026* software makes setting the levels simple. As with any multi-stage audio device, it is important to optimize levels. In the case of the *1026*, one would want to maximize the levels from the mixing sec-

tion as they enter the DSP processing section. Too little level has a negative effect on signal to noise, while too much level leads to distortion.

To optimize levels, Ivie provides a "Bus Trim" control for each of the two audio busses. To adjust bus trim, place the cursor on the box labeled "Bus Trim" and click, as shown in Figure 19 below:

Proper Cursor Placement for Opening the "Bus Trim" Pop-up Control Window
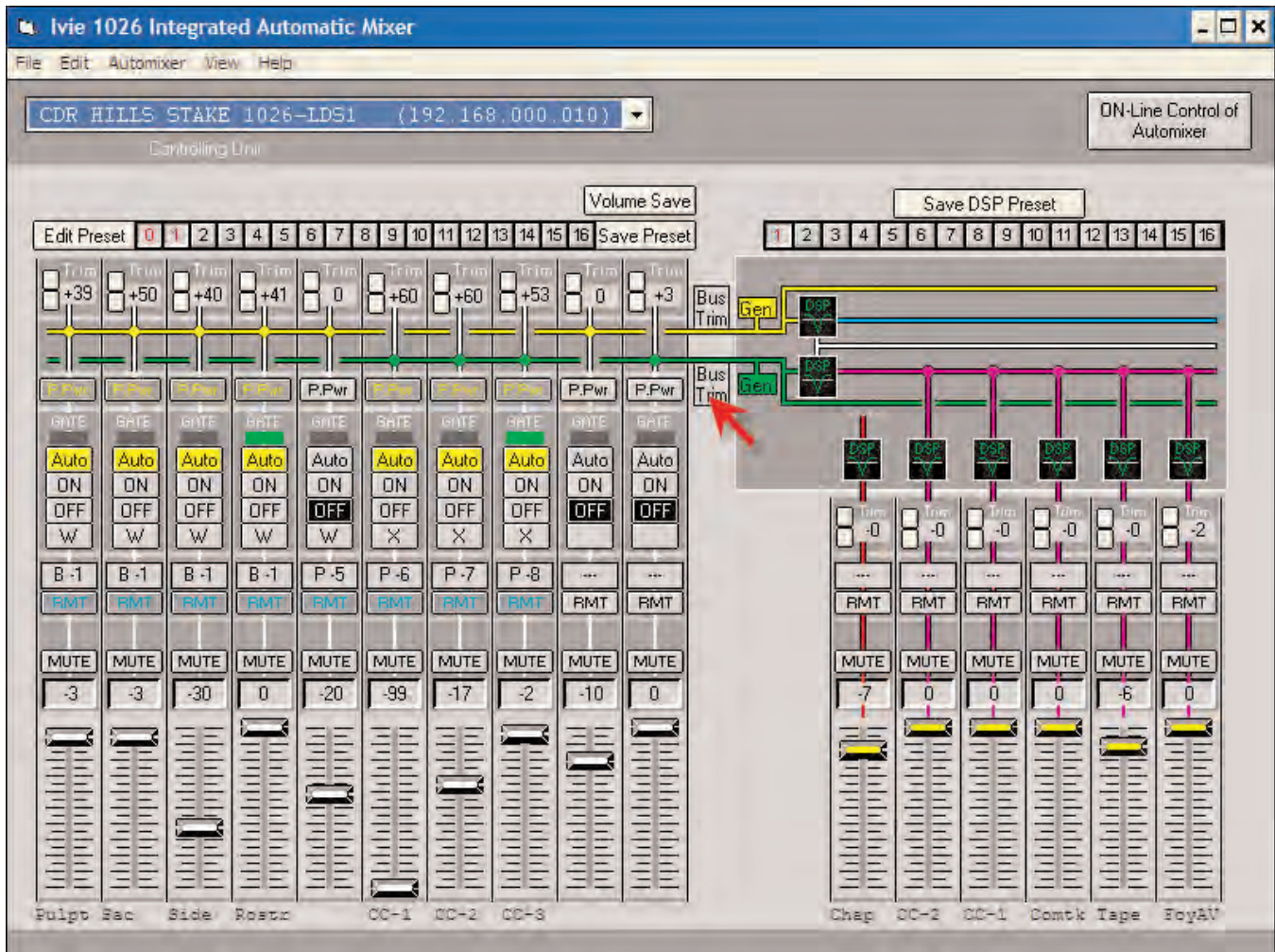

Figure 19

This will select a pop-up window titled "Bus Trim." As shown in Figure 19 above, there is a separate Bus Trim pop-up window for each of the outputs, output "A" (yellow) and output "B" (green). The cursor in the illustration above has been placed on the "B" output Bus Trim box. The Bus Trim "A" pop-up window is shown in Figure 20 on the following page. It is identified as the "A" Bus Trim by reference to the bus color (yellow) used by the software to visually identify mix bus "A".

The "Input Trim" Pop-up Control Window

The color identifies the Mix Bus
Trim Control Selected:
 Yellow = Mix Bus "A."
Green = Mix Bus "B."
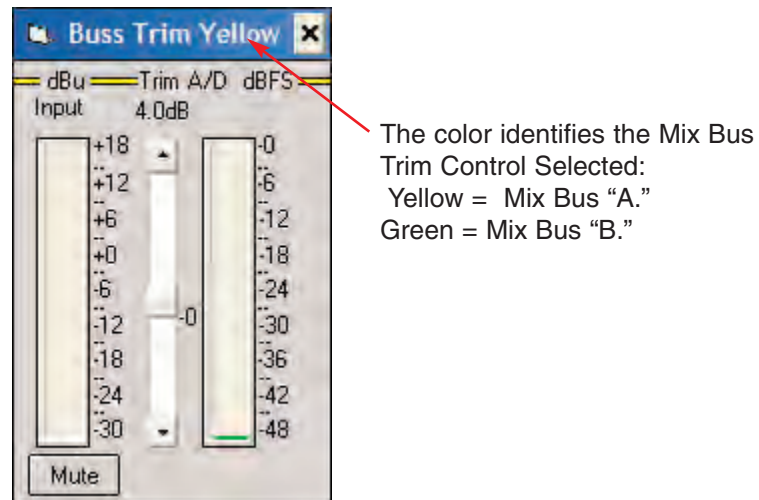
Figure 20

The proper setting of the signal level coming into a *1026* and signal level exiting the unit is crucial to the performance of the unit. As with all DSP based products, the *1026* is far less forgiving of being driven into clipping than an analog product would be. The level meter on the left shows the analog signal level coming from the mixer section of the *1026*. For the best performance, this level should be as close to 0 dBu as possible. To accomplish this, the individual input signal levels may be increased or decreased before arriving at the DSP section of the *1026*. The input level in Figure 20 above, is at about -6, which is still not bad. Once set to as close to 0 dBu as possible, the input level will need to be trimmed for optimum processing by the DSP. The Trim slider adjusts level prior to the *1026*'s A/D converter.

The level control slider between the two meters is used to trim the input. The meter on the right shows the trimmed signal arriving at the A/D converter in dB full scale (dBFS). Best performance occurs when this level is about -18dB to -12dB (as the example in Figure 20 above shows). This provides good headroom before the meter on the right will top out at 0dB. When the dBFS level exceeds 0dB, audible distortion may occur. It is very important not to overdrive the inputs.

The output levels can be trimmed in a like manner. Placing the cursor on the section in an output labeled "Level" and double clicking will cause a pop-up window labeled "Output Trim." The output may be trimmed in the same manner as discussed previously for an input. Again, an output level of about -18dB full scale will provide optimum performance. Clipping an output will cause the same unhappy results as clipping an input. Once levels have been properly set, the rest of the signal processing modules of the *1026* can be properly set. The first signal processing module in the input path that can be adjusted is the parametric equalizer module. Final output level adjustments should be made after PEQ adjustments are completed.

The PEQ Modules

The PEQ (Parametric Equalizer) Modules on the input and outputs provide tremendous flexibility in creating broad and narrow-band shaping and filtering. Each PEQ Module provides a user-selected num-

ber of individual filters which can be selected to operate as Low-shelf, Low-pass, High-shelf, High-pass, tunable parametric, or Constant-Directivity Horn Correction.

High and Low-pass filters are used to band-limit the system's frequency range. The *1026* High-pass and Low-pass filters are fixed, 6dB/Oct. filters. If sharper filter skirts are needed, the *1026* allows the selection of "Extended" filters (see the section titled "Extended Filters" on page 25). Shelving filters are also available to provide additional flexibility in shaping response curves.

Constant-Directivity Horn Correction filtering provides the standard high frequency equalization often used with C-D horns. The "knee" or starting point of the filter can be selected to customize for the individual horn and driver combination.

Parametric EQ filters can be used for a wide variety of purposes. In general, there are three main purposes for parametric filters: A) Used in broad-band (low-Q) configuration, they can shape the general frequency response to correct imperfections in the sound system transducers. This shaping is akin to customized tone controls. B) As narrow-band (high-Q) filters, PEQ's can be used to "tune out" feedback frequencies. Tuned properly, surprisingly little attenuation is required to remove the main feedback frequencies in a room. (Sometimes, broad-band and narrow-band filters can be constructed to work in concert to reduce feedback). C) Finally, narrow-band PEQ's are extremely useful in attenuating or eliminating room-ring modes. The knowledgeable use of narrow-band PEQ filters by an experienced professional in the removal of room-ring modes often has a very dramatic effect on the intelligibility of the overall system in a room.

Manipulating the Filters

Double clicking on any PEQ module (input or output) will open the PEQ Filter Window. The PEQ window provides a graphic interface which gives quick, easy control over any of the definitions or functions available for PEQ filters. The PEQ filter pop-up window is shown in Figure 21 below:
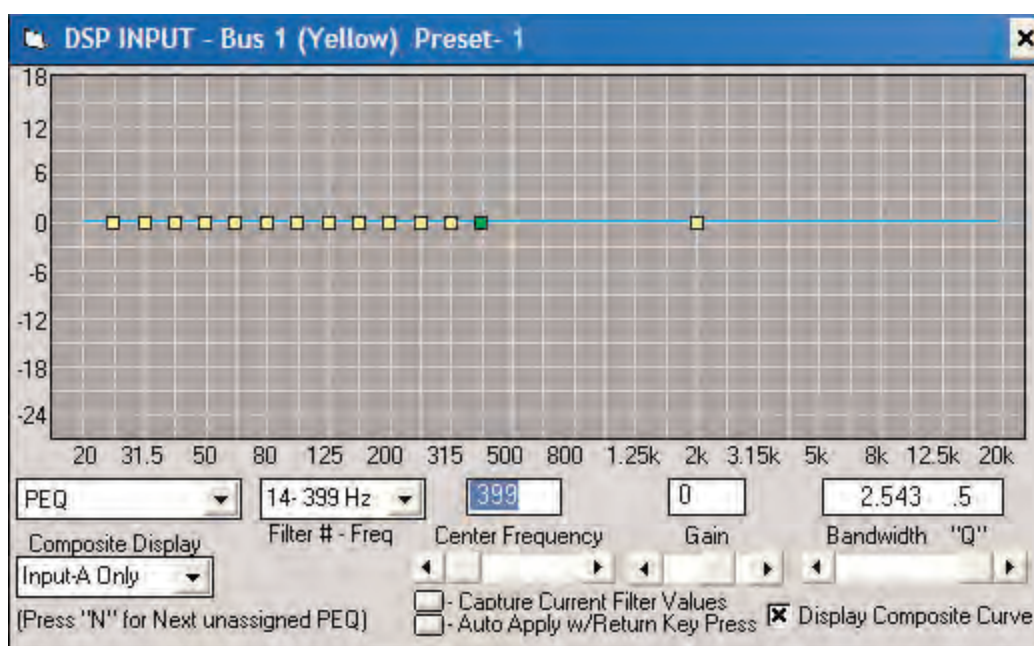

Figure 21

When opening a PEQ window that has not been previously adjusted, a horizontal row of small boxes in the graph area of the display will be found. Each of these boxes represents a user-definable filter. The position of the box on the graph determines that filter's frequency (x-direction) and level (y-direction). The number of filters in the graph is determined by the number of filters allocated in the DSP Configure Screen. The "selected" filter is shown in green and the "deselected" filters are shown in yellow. The data windows below the selected filter provide information about it. Additionally, slider controls located just below the data windows allow adjustment of the filter parameters reported by the data windows. On the far left, a data window identifies the filter type. Associated with this data window is a pull-down menu which allows another filter type to be chosen by the standard "click and drag" method of selection. The next data window to the right indicates the center frequency of the filter. The third data window indicates the level of the center frequency of the filter, and the fourth data window indicates the "Q," or bandwidth of the filter.

To select and manipulate a filter, place the cursor in the center of a filter box. This selects the filter (its color will change to green). To manipulate the filter, "click and drag" as desired. The data windows will continue to provide detailed information about the selected filter and the filter will be shown graphically on the screen. To select a different filter, place the cursor on the box representing it. *Note:* Key strokes for adjusting a selected filter: up/down arrow keys change gain in .25dB steps. Left/right arrow keys change frequency in one Hertz steps. Holding down the Shift key while using the arrow keys changes gain in .75dB steps and frequency in ten Hertz steps.

The "check" boxes at the bottom of the PEQ control window provide very useful functions. The one on the far right allows the composite curve to be displayed, or not displayed. The check box in the center, causes the "Q" of the last filter adjusted to be saved and automatically applied to the next filter selected for adjustment. When this option is active, as soon as a new filter is selected, the frequency data window will become highlighted as well as the new filter. Typing in a new frequency will move the filter to this new center frequency and apply the same "Q" as the last filter adjusted. As the note to the left of the check box suggests, typing "N" followed by the frequency will automatically select the next "unused" filter and move it to the chosen frequency with the same "Q" as the last filter adjusted. This feature can save a lot of time, especially when setting "notch" filters.

Extended Filters

Extended filters may be selected when a sharper skirt is required for a High-pass or a Low-pass filter. Selection of extended filters is accomplished using the DSP Configure pop-up control screen located under the Edit pull-down menu. The PEQ Module on the DSP Configure pop-up control screen provides a box labeled "+" which, when checked, activates the extended filter capability. When active, the extended filter feature adds extended filter capability to the "left-most" and "right-most" filters of the PEQ. Figure 22, on the following page, shows the action of an extended, High-pass filter:
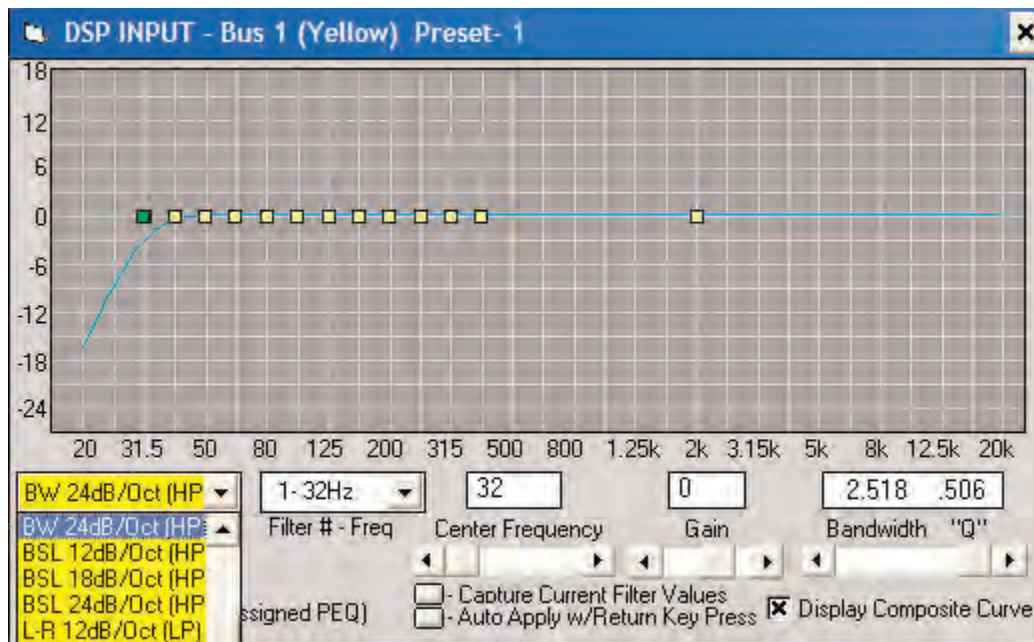
The Extended High-pass Filter

Figure 22

As can be seen, the extended filter that has been selected in Figure 22 above is a Butterworth, 24dB/Octave, High-pass filter. When an extended filter is selected by the cursor, the window identifying the filter type turns from white to yellow. The yellow color is a reminder that this filter has extended capability. When the pull-down menu of this window is opened, the entire menu will be displayed in yellow instead of white, even though the first selections of the menu are not extended filters. The extended filters are the last ones listed on the menu and they appear as menu items only when a filter with extended capability has been selected.

PEQ Screen Graph Color Codes

To aid in programming and to help unclutter the filter screen, color coding is used. The filter "composite" (the resultant action of all filters) is always shown in blue. When manipulating a filter, only two curves will be displayed on screen - the composite in blue and the filter currently being manipulated in red. When no filters are being manipulated (the mouse button is released), all the filters will be shown in yellow, as well as the composite in blue. When several filters are active, the advantage of displaying only the filter being manipulated and the composite is readily apparent.

The Compressor/Limiter Modules

Compression and Limiting can be added to any *1026* input or output by activating the "Comp" modules while in the DSP Configure screen. To set up the Compressor/Limiter, double click on the Compressor module in the Main Control Screen. The Compressor/Limiter pop-up window will appear as

demonstrated in Figure 22 below:

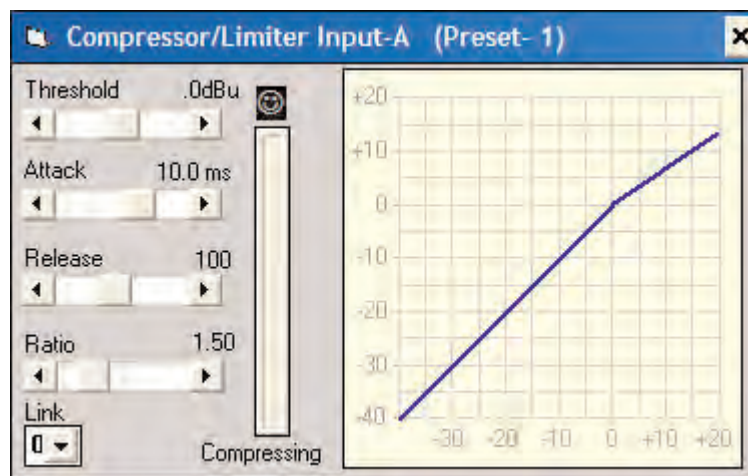*The Compressor/Limiter Pop-up Control Window*

Figure 22

Compressor attack and release times are fully programmable, as well as compressor threshold and ratio. The *1026* software allows these parameters to be set using either slider controls, or by clicking on the compression graphic to set the threshold at the spot of the click. After the threshold has been set, clicking at the top end of the compressor curve and then dragging and releasing sets the ratio. The slider controls will move to reflect the changes made by clicking and dragging.

The Delay Modules

Delay can be added to any *1026* input or output by activating the Delay modules while in the DSP Configure screen. To set Delay, double click on the Delay module in the Main Control Screen. The Delay pop-up window will appear as demonstrated in Figure 23 below:

The Delay Pop-up Control Window
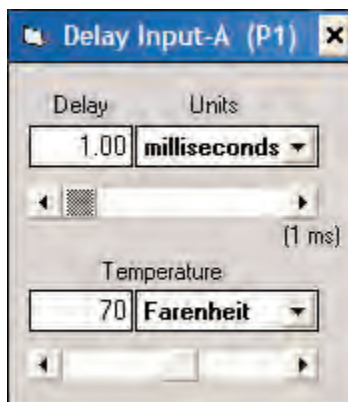
Figure 23

Up to 1365 milliseconds of delay is available for allocation. Delay can be increased or decreased in one millisecond increments, and/or two microsecond increments. Using the *1026* software in either the *1026* Main Control Screen, or the Delay pop-up control window, clicking on the up/down arrows adds delay in one millisecond increments. Holding down the "Shift" key while clicking on the up/down arrows adds delay in two microsecond increments. *Note: 1026* propagation delay, a constant, is one millisecond.

The slider at the bottom of the Delay pop-up control window allows setting the temperature at the expected level during sound system operation. Temperature may be set in Fahrenheit or Celsius by clicking on the Temperature pull-down menu.

The Crossover Modules

Crossovers (either two-way or three-way) can be added to the *1026* outputs by activating the Crossover modules while in the DSP Configure screen, as shown in Figure 24 below:
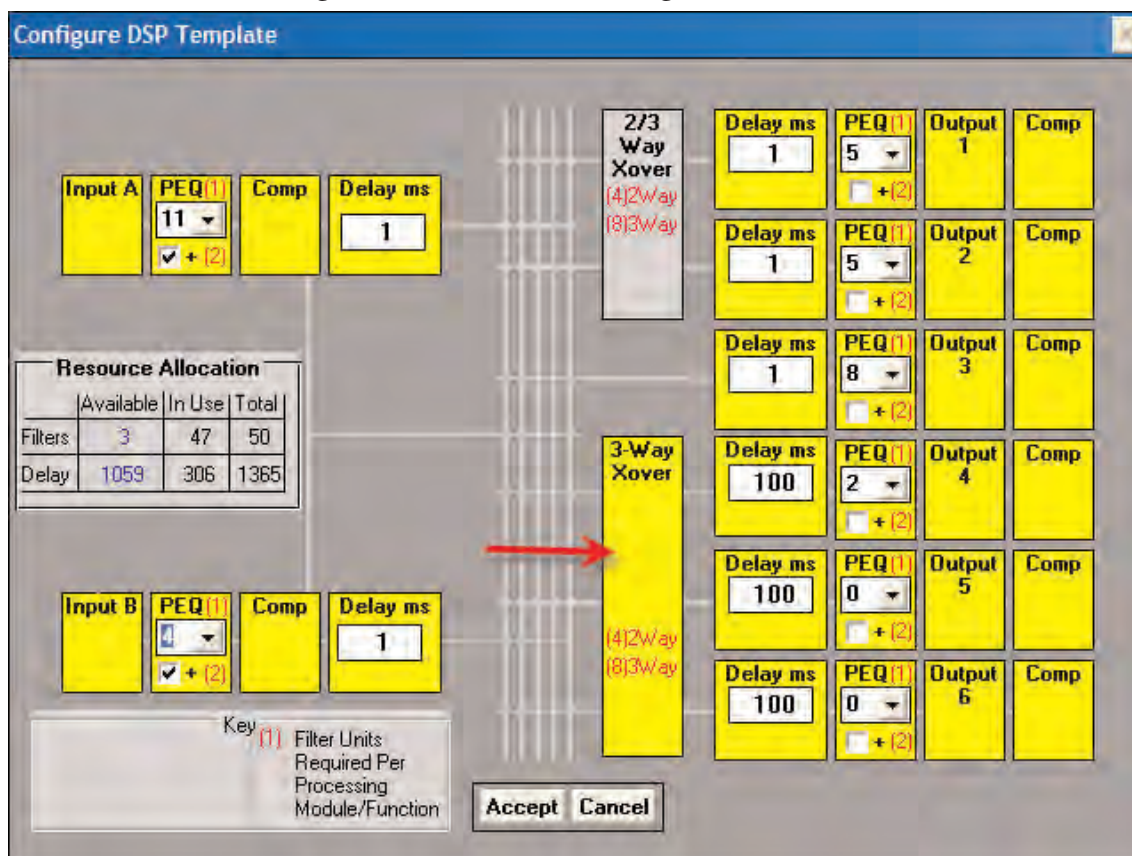
Figure 24

There are a variety of crossover filters available for selection (Butterworth, Linkwitz-Riley and Bessel) with filter skirts ranging from 12dB per octave to 24dB per octave. To set up a crossover, double click on the Crossover module in the Main Control Screen, as demonstrated in Figure 25 on the following page:

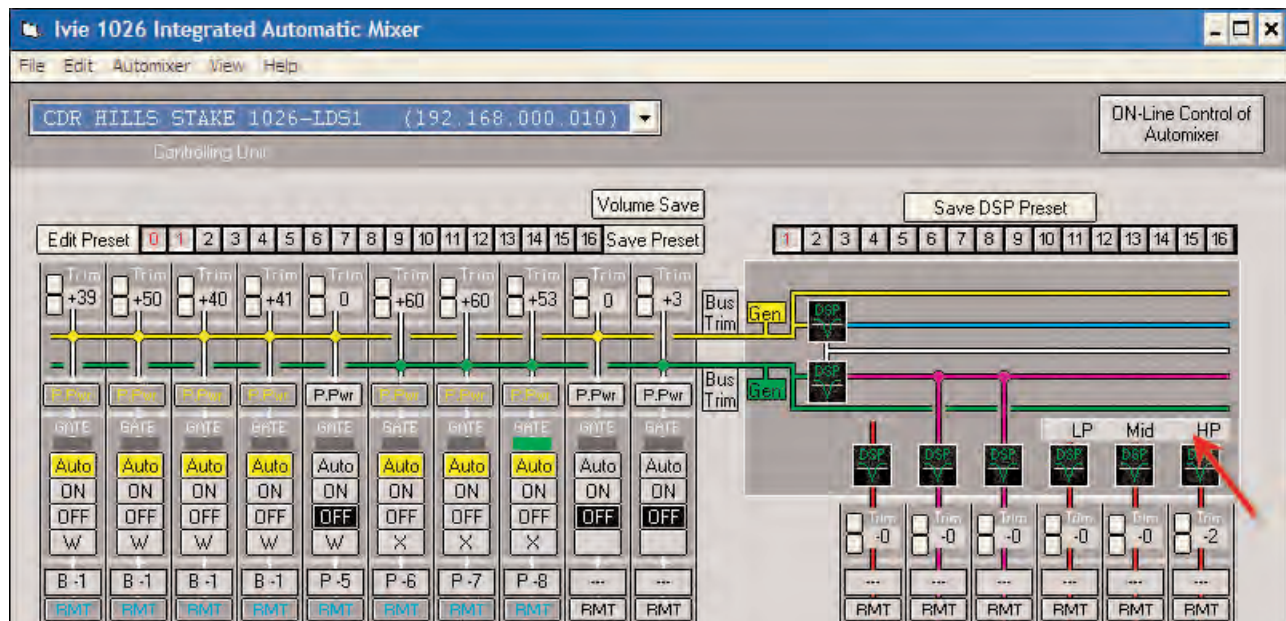*Cursor Placement to Activate the Crossover Pop-up Window*

Figure 25

Double clicking, as shown in Figure 25 above, will activate the crossover pop-up window as shown in Figure 26 below:

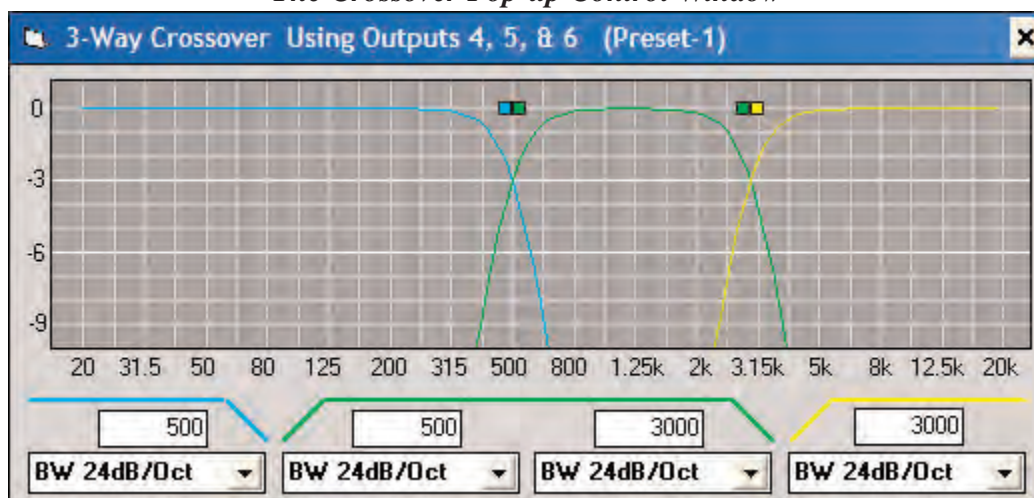*The Crossover Pop-up Control Window*

Figure 26

Placing the cursor on the box representing the crossover filter will allow changing the filter frequency by clicking and dragging. Filter type desired can be selected by using the pull-down menus associated with each individual filter. The data boxes above the pull-down menus show the frequency of the filters at their 3dB down points. Crossover frequencies can also be directly entered by "clicking" on a crossover data box which then allows typing the desired frequency. The "Enter" key completes this operation.

1026 Internal Sine Wave and Pink Noise Generators

In addition to the signal processing capability of the *1026*, there is a built-in sine wave and pink noise generator on each of the two input channels. These pink noise generators are independent of each other and are not coherent. When using the sine wave generator, frequency is adjustable in one Hertz increments, and signal level for both sine waves and pink noise is adjustable.

To set up the sine wave and pink noise generator, a pop-up control window is provided, as with other functions of the *1026*. To access the pink noise pop-up control window, place the cursor next to the word "Gen," located at the left-center of the Main Control screen. Double clicking will open the pop-up control window. Cursor placement for this operation is shown in Figure 27 below:

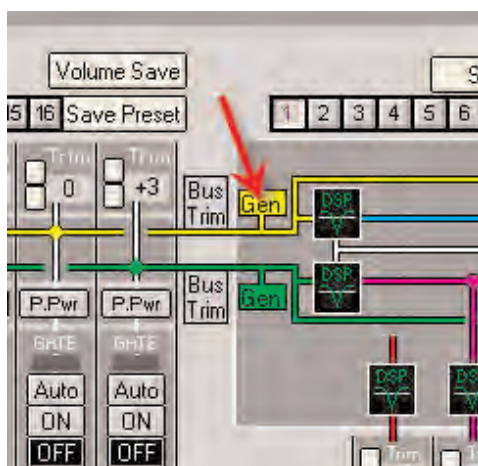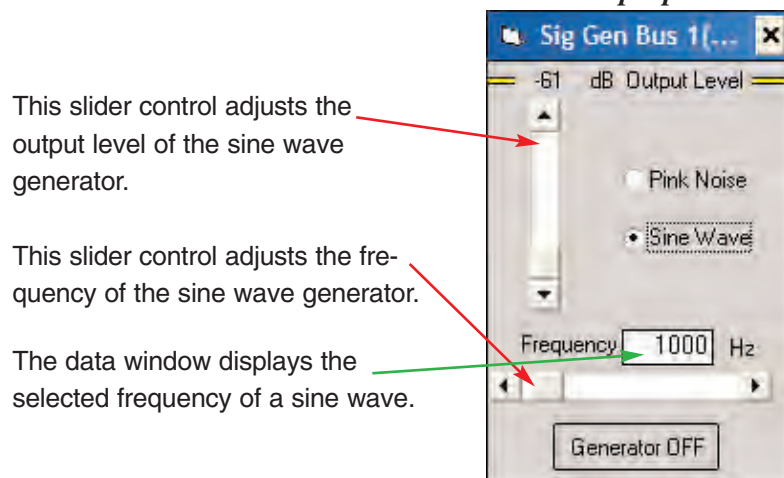*Proper Cursor Placement to Access the Pink Noise Pop-up Control Window*

Figure 27

The placement of the cursor in Figure 27 above allows selection of the pink noise pop-up control window for input Channel A. Double clicking will open this control window, as shown in Figure 28 below:

The Pink Noise Pop-up Control Window

This slider control adjusts the output level of the sine wave generator.

This slider control adjusts the frequency of the sine wave generator.

The data window displays the selected frequency of a sine wave.

Note: Pressing the *1026* "Mute" button mutes the channel input, not the pink noise generator.

Figure 28

Software Pull-Down Menus for the 1026

Many of the functions of the *1026* software have been concentrated in pull-down menus. Many of these menus have pop-up screens associated with them. These pop-up screens are quite self explanatory and intuitive in their operation, following standard Windows® conventions.

File Pull-Down Menu

The File pull-down menu contains five menu items. These include: "Load settings FROM disk", "Save settings TO disk", "Force Sync with Unit", "Poll", 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 *1026* settings (presets, etc.) from disk storage to a *1026*. Once a unit has been fully programmed, it is prudent to download its program-ming to a memory stick or other storage medium. When this is done, it is a simple matter to reprogram the unit from the 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 AudioNet+® units will not accept files for uploading that are not specifically intended for them. A 784+ Matrix Mixer, for example, will not accept *1026* files.

When uploading to a *1026*, files with the extension ".126" must be selected. After .126 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 win-dow. 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 *1026* settings to disk. Once a unit's programming has been saved to disk, it is easy to reprogram the unit from the disk, should this become necessary, using the Load utility. All information about a *1026* is saved in a single file. This includes DSP Configure, names, Preset Port Configure and all preset information.

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 (.126, for example, is the unit type designator for a *1026*). 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 *1026*, for example, must be saved as .126 files because only .126 files can be uploaded into a *1026*. A desirable file name for a *1026* might be BallrmA.126, identifying the unit as the Ballroom A *1026* Digital Signal Processor.   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+ , .80P for the 880+, and .626 for the 626. The "Save File as Type:" pull-down window lists the Ivie AudioNet+® 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.

Force Sync with Unit (File Pull-Down Menu): When the *1026* software initially polls and finds a *1026* on line that it has not previously seen, it will interrogate the unit for its complete configuration and status and display the results on the Main Control screen. While this updating is taking place, the presets will appear to be counting down as data is sent from the *1026* to the computer. This process takes a few seconds to complete. Other AudioNet® products maintain constant communication with the AudioNet® software, even displaying changes on the software control screen when external controlling devices, such as remote level controls, send commands to them. For a *1026* to act in this same manner, the communication would be so intensive as to be impractical.

When the *1026* software is exited, the software sends each *1026* on the network a "date and time" code and stores a matching code and file in the computer. When the software is activated again, it looks for a matching "date and time" code on each of the *1026*s it finds during the polling process. If a *1026* is selected and a matching "date and time" code is noted, the software loads the corresponding file and displays the status of the "recognized" unit as it was the last time the unit and the software communicated. If remote controls, such as touch panels have changed the status of the unit, these changes will not automatically be displayed. The Force Sync with Unit utility forces the unit to communicate with the software and display its current status.

Poll (File Pull-Down Menu): Selecting Poll from the File pull-down menu commands the software to poll the network and list all the AudioNet+® 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 AudioNet® units on the network is still available from the pull-down menu in the "Controlling Unit" window of the *1026* software.

Exit (File Pull-Down Menu): This command exits the *1026* 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

The Edit pull-down menu contains five menu items. These include: "DSP Configure", " Preset Port Configure", "Set Unit and Channel Names", "Lock with Password" and "Set Unit IP Address".

DSP Configure (Edit Pull-Down Menu): The DSP Configure pop-up window allows the user to configure the *1026* for the specific needs of a given application. A complete discussion of the DSP Configure can be found in this manual under the section titled "DSP Configure" on page 18.

Preset Port Configure (Edit Pull-Down Menu): There are sixteen presets available on a *1026*, but only eight sets of contact closure terminals to be used for activating presets. What if more than eight presets requiring contact closure activation are necessary? The Preset Port Configure utility is intended to provide this function. There are 256 different preset port switch patterns available which can be used to activate presets by various contact closure combinations.

The Preset Port Configure pop-up window with the standard factory default configuration is shown in Figure 29 below:

The "Default" Preset Port Configure Pop-up Window
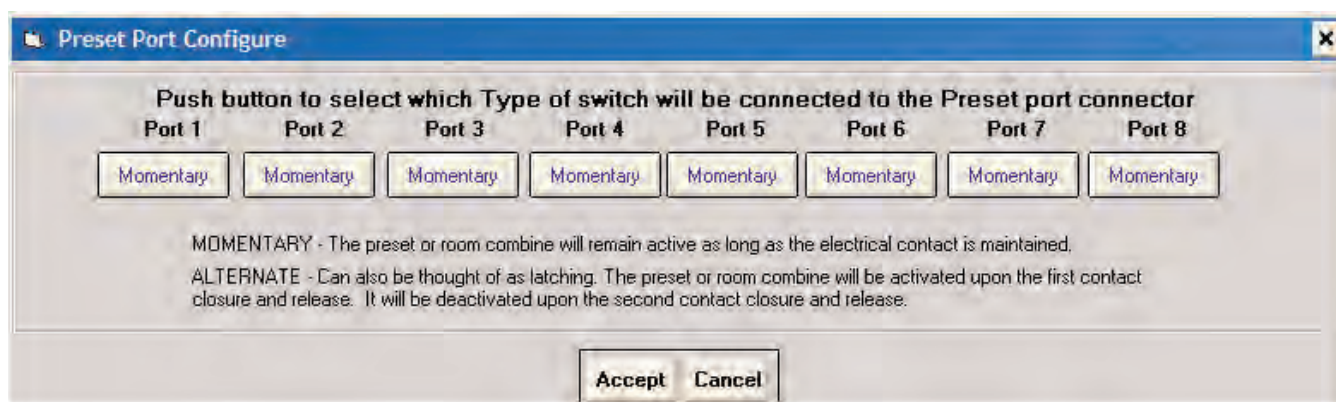
Figure 29

The Preset Port Configuration is set in a default position when a *1026* is shipped from the factory. In the default configuration, Preset 1 will be activated when Port Terminal 1 is grounded, Preset 2 will be activated when Port Terminal 2 is grounded, etc.

Important Note: The Preset Port is not field configurable via the *1026* software. Custom Preset Port configurations are available from the factory. Contact Ivie for further information on custom Preset Port configurations.

Set Unit and Channel Names (Edit Pull-Down Menu): The *1026* software allows each *1026* 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 digital signal processor with the name "Exhibit Hall" is much more meaningful than *1026* at address number 5. To give a *1026* a name, and/or give names to the individual inputs and outputs, select the Set Unit and Channel Names utility.   A box will appear with the current or default unit name and channel names. Type in the desired name and press the OK button. To exit without making any changes, press the cancel button.

Lock with Password (Edit Pull-Down Menu): Password security is available for the *1026*, as well as for other AudioNet® products. The Lock 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. The assigned password is retained in the *1026* unit

itself, not in the *1026* software.

Set Unit IP Address (Edit Pull-Down Menu): When there is more than one *1026* on the network simultaneously, each unit must have its own unique address. *1026*s are shipped from the factory with a default unit address. When the *1026* software starts up, it scans the network for units and lists their addresses. If there is more than one *1026* on the network, they will automatically be assigned different IP addresses. Although this process is automatic, selecting Set Unit IP Address will force the software to reinitiate the scan process, so if units have been added to the network after the software was started, they will be found.

Automixer Pull-Down Menu

The Automixer Pull-Down Menu contains two items: "ALC Levels", and "Setup Options".

ALC Levels (Automixer Pull-Down Menu): It would be beneficial to discuss the ALC of the *1026* in some detail. All *1026* inputs 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 *1026* 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.

Again, 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". The target level is adjustable from -10dBu to +4dBu in 1dB steps. The factory default setting for the target level is -10dBu.

Figure 30, on the following page, graphically presents the 32dB range of the ALC and its function over that range:
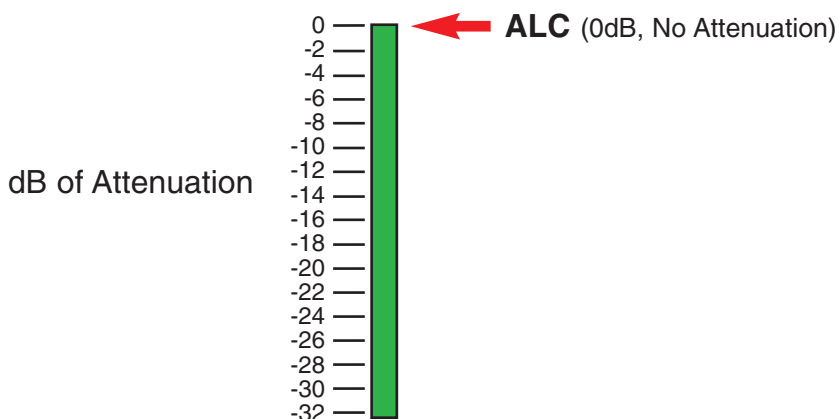
ALC 32dB Operating Range

Figure 30

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 30 above, 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 reduce the speaker's level. Consider Figure 31 below:
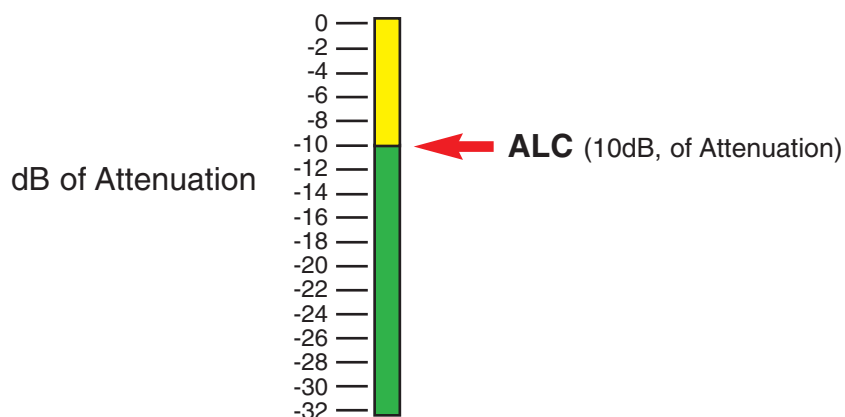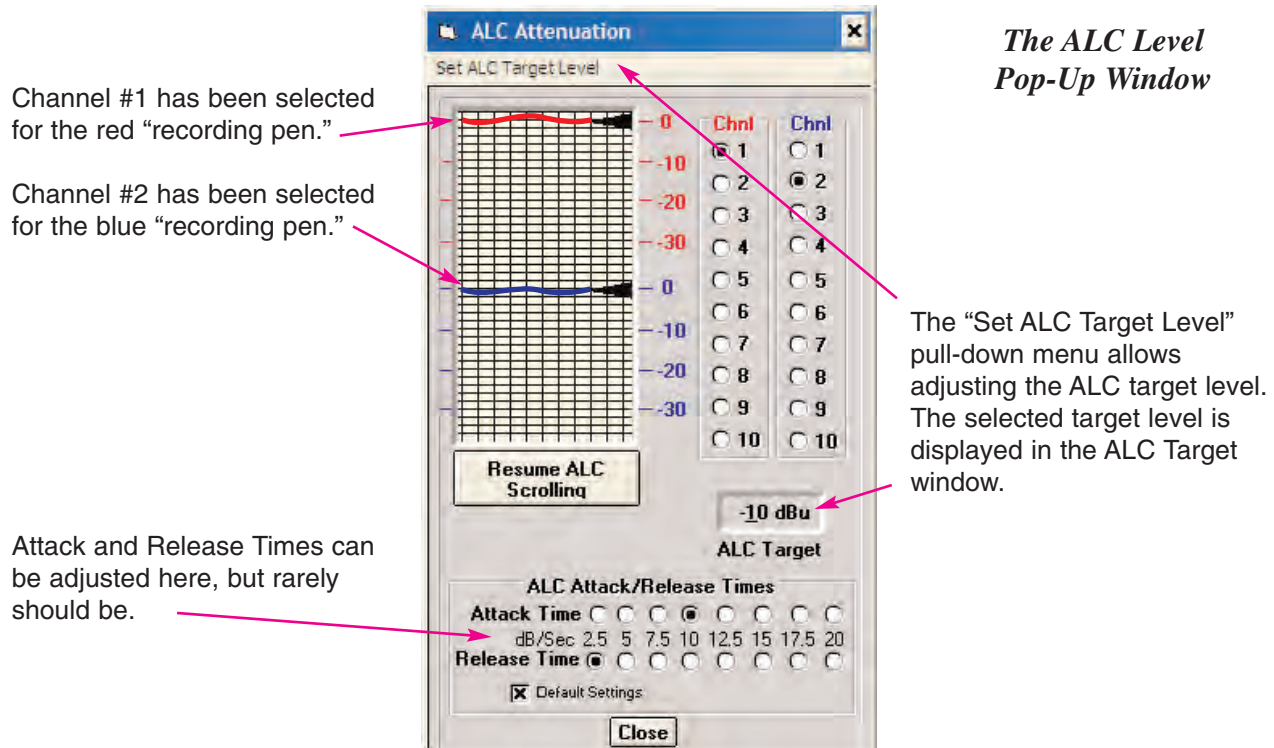

Figure 31

In Figure 31, on the previous page, the ALC (remote control) has reduced the speaker's level 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 Trim setting desired, and gain structures through the rest of the system finalized. To maintain good signal to noise, the input and output levels 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 "Setup/Options" menu item under the Automixer 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 32 below details the the controls and strip chart recorder display of the ALC Level pop-up window.

The ALC Level Pop-Up Window

Channel #1 has been selected for the red "recording pen."

Channel #2 has been selected for the blue "recording pen."

The "Set ALC Target Level" pull-down menu allows adjusting the ALC target level. The selected target level is displayed in the ALC Target window.

Attack and Release Times can be adjusted here, but rarely should be.

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 "Setup/Options" menu item under the Automixer pull-down menu.

3.   Open the ALC Level pop-up window under the Automixer 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. 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 that 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 1026 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.

Setup/Options (Automixer Pull-Down Menu): The "Setup/Options" pop-up window is shown in Figure 33 below:

Figure 33

As Figure 33 above shows, the Setup/Options pop-up window allows the adjustment of a number of automix parameters.

"Last ON", when activated, keeps at least one microphone gated on at all times (the last microphone to gate on). This Ivie feature solves the background noise on and off ("pumping") that can be annoying in an ancillary feed.

"Difference" sets the level (in dB) that must be seen at any input, compared to the average of all inputs, before the input will gate on. The 6dB factory setting works well in most applications.

"Off Attenuation" sets how far the VCA will attenuate the signal when a channel is gated off.

"NOM Attenuation" (Number of Open Microphones Attenuation) sets how far the level (in dB) at the output will be reduced when the number of open microphones is doubled. The level will be automatically reduced by the NOM Attenuation selected to prevent sending the sound system into feedback as successively more microphones are gated on.

"Release" sets the time (in seconds) that a channel will remain gated on without seeing a sufficient gating level input, before it gates off.

"Sensitivity" applies to the situation where only one microphone is plugged in (which is always the case when VOX has been selected). Since there is no average level of other open microphones to compare to in order to effect gating, the Sensitivity setting facilitates gating a microphone on when the signal from the mic exceeds that level set in Sensitivity.

"ALC On/Off" allows turning on, or off the ALC for any input.

Note: VOX operation would be selected when a gated, stand-alone microphone is required. The microphone would not be part of an automix circuit, but would be gated when its input level exceeded the threshold level set for gating (Vox Sensitivity).

Help Pull-Down Menu

At the time of the printing of this *1026* Digital Signal Processor manual, the Help pull-down menu contains only one item titled, "About."

About (Help Pull-Down Menu)
The About menu item from the Help pull-down is informational. It identifies the version of *1026* 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 *1026* 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.

Copy and Paste, Pop-up Menus

The *1026* provides tremendous flexibility in copying and pasting programming information from one unit to another, or from preset to preset. Programming from one preset can be copied and pasted to another preset or to all presets. To facilitate this feature, Copy and Paste, Pop-up Menus are provided. Individual menus are associated with the level controls, the PEQ modules, the Compressor/Limiter modules, the Delay modules, the Crossover modules and the Routing matrix.

With the cursor on any of these modules or controls, a "right click" will open its associated Copy and

Paste, Pop-up Menu. Figure 34 below shows the Copy and Paste pop-up menu associated with the Routing Matrix which has been activated by placing the cursor over the Routing Matrix and "right clicking":

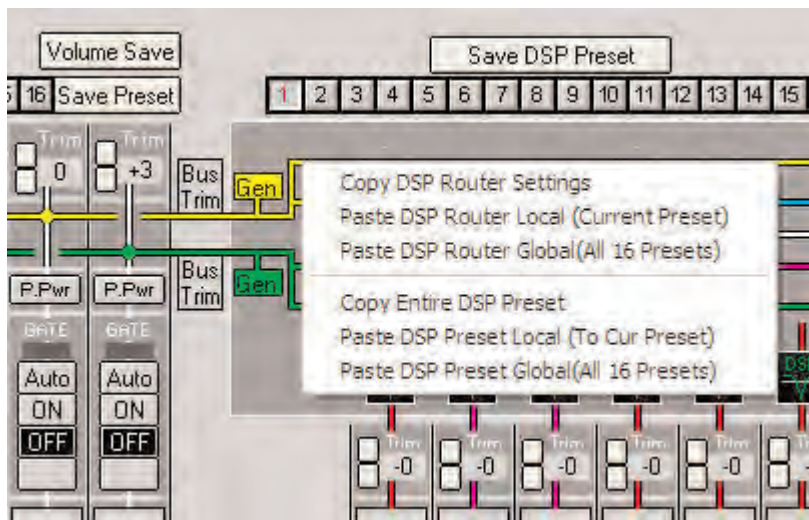*The Routing Matrix Copy and Paste pop-up Window*

Figure 34

The menu items are self-explanatory. The Routing Matrix settings can be copied and pasted to another preset by selecting the appropriate preset and then pasting to it, or the Routing Matrix settings can be pasted to all presets (Global). Notice also that ability to copy the whole preset, not just the Routing Matrix settings, is provided. All the preset programming, after copying, can be pasted to another preset, or to all other presets. This flexibility in copying and pasting information can greatly speed *1026* programming. Similar Copy and Paste pop-up menus for other modules can be accessed by placing the cursor on the module and "right clicking." All of the menus function in a similar manner to the menu in Figure 34 above.

Note: In copying and pasting from one unit to another unit, use the Save and Load utilities found under the File pull-down menu.

Saving Programming Data - The Automatic "Save" Feature

The *1026* has an automatic save feature designed to prevent major programming frustration. When a status change is initiated that could cause programming data loss, the *1026* will automatically save the data before executing the requested change. For example, consider the following scenario: If a programmer has both the *1026* software control screen and a control screen for a different product displayed simultaneously while he is programming a *1026* (the *1026* screen would be active and the other screen would be in the background). If he were to pass the cursor over the other screen, it would activate that screen and send the *1026* screen to the background. At this point, all unsaved *1026* programming data would normally be lost. However, the automatic save feature of the software forces the *1026* to save the programming data before it executes the command to activate the mixer screen and deactivate the *1026* screen.

This same "saving" feature is activated when changing from a *1026* to a different product, from one *1026* to another *1026*, or when exiting the software. Any command which would cause unsaved data to be lost activates the automatic save feature.

Of course, if power is lost in the middle of programming, all unsaved programming will be lost as well. It is not a bad idea to use the "Save" button from time to time, even though the automatic save feature can prevent most losses of programming data.

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