5 January 2005 – This contact update page has been added to the Acrobat document you have downloaded. Please disregard any contact information printed within the document.

Our Mailing and Shipping Address:

1514 Ed Bluestein Blvd., Suite 201 (for U.S. Mail)
Austin, TX 78721 U.S.A.
Phone: 512-389-5358
Fax: 512-301-3932
Main Email Address: cvanr@whiteinstruments.com
World Wide Web Site: http://www.whiteinstruments.com/

Note: Repairs and packages should be shipped to Suite 202
UNPACKING

Carefully unpack your unit and inspect it for damage. If it is damaged, notify the carrier and White Instruments immediately. The carton should contain the following items:

1. One DSP series processor.
2. Four rack screws.
3. One user service card.
4. One 3.5" floppy disk (DSP5000IQ/5024IQ units only).

After unpacking, fill in the user service card and mail it to White Instruments.

QUICK START

Follow these instructions to connect your DSP series unit.

1. Using the 4 screws provided, mount the unit in your rack.
2. With the power switch off, plug the unit into an appropriate electrical outlet.
3. Connect the analog input and output XLR cable(s) to the appropriate XLR input(s) and output(s) at the back of the device.
4. Turn the unit on and verify that the power LED is lit and the front panel display shows the model number.
5. Although not absolutely necessary, it is good practice to become familiar with the unit's security features immediately. Securing your unit with your own system password prevents unwanted tampering. The procedure for setting the system password is given in the programming guide.

INTRODUCTION

WHAT IS THE DSP SERIES?

The DSP series units are audio signal processors which input and output analog audio signals, but perform all of their processing functions digitally. Analog input signals are digitized and fed into a digital signal processor. The processor uses software programs to perform all signal processing functions. The results are then converted back to analog signals and sent to the outputs. Consequently, a single DSP hardware design can be programmed and reprogrammed to perform many signal processing functions.

Because it is DSP based, the DSP series units also have other advantages such as inherent filter accuracy, repeatability and the ability to store settings in memory. Non-volatile EEPROM memory, rather than battery powered memory, is used to store settings so that they won't be lost because of inevitable battery failure.
WHAT CAN THE DSP SERIES DO?

The DSP series units can perform many of the audio processing functions that have been traditionally performed by separate devices, including the following:

- Crossovers
- Parametric Equalizers
- Delays
- Distribution Amplifiers
- Limiters
- Mixers
- Custom filters (e.g. highpass, lowpass, shelving)

In terms of filtering power, the DSP series units provide up to 35 second order filters which can be used as crossover, parametric or other filters. The DSP5000 and DSP5000IQ have 1 input and 4 outputs. The DSP5024 and DSP5024IQ have 2 inputs and 4 outputs. Figure 1 shows a block diagram of the audio signal paths and the placement of the different signal processing functions in those paths.

![Figure 1. Block Diagram](image)

The DSP series units can be controlled by various methods. All DSP series units can be controlled from their own front panels via a menu driven LCD display. DSP5000 and DSP5024 units can also be controlled via a serial PA-422 link, which allows these units and any of the White Instruments 4700 series units to be connected together in a control network. DSP5000IQ and DSP5024IQ units can be controlled via a Crown IQ Turbo control interface.

FRONT PANEL FEATURES

The front panels of the DSP5000, DSP5000IQ, DSP5024 and DSP5024IQ are essentially similar to the front panel shown in Figure 2.

MENU KEY
The menu key scrolls forward through the various system menus or exits from a function screen back to the parent menu.

**FUNCTION KEY**
The function key scrolls forward through the function screens under a particular menu.

**ENTER KEY**
The enter key is generally used to activate a parameter change such as a parametric filter or alphanumeric name. This key can also be used to change the active input or output channel when programming input or output channel parameters.

**ARROW KEYS**
The arrow keys are used to change parameters in a function screen or to move between parameter fields in a particular function screen. These keys can also be used to scroll forward and backward through the system menus. The left and right arrow keys, if pressed simultaneously, can be used to quickly clear a system or preset security code to all dashes.

**INPUT LEVEL LEDs**
The input level LEDs are bicolor red/green LEDs that have multiple functions. They monitor the analog input level after the input gain stage and can detect internal clipping in the DSP input filter stage. They may also indicate that a parameter change is pending and requires the enter key to pressed for activation. The LEDs turn green for an input signal level within 10 dB of clipping. They turn red for an input signal level within 1 dB of clipping. The input LED (only the left input LED for the DSP5024/IQ) flashes between off and yellow if there is a parameter change pending.

**OUTPUT CLIP LEDs**
The output clip LEDs are red LEDs that monitor the analog output levels as well as detect internal clipping in the DSP output filter stages. They turn on when the output level is within 1 dB of clipping.

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**Figure 2.**
**DSP5000 series front panel**

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**REAR PANEL FEATURES**
Four different back panels exist for the DSP series. The standard back panel used for the DSP5000 and DSP5024 is shown in Figure 3. Note that there is only one analog input on the
DSP5000. The DSP5000IQ/5024IQ back panel is shown in Figure 4. Note that the DSP5000IQ has only one analog input.

**ANALOG INPUTS and OUTPUTS**

Analog inputs and outputs in Figures 3 and 4 are bulkhead mounted 3-pin XLR sockets.

**PA-422 INPUT and OUTPUT PORTS**

PA-422 input and output serial communication ports in Figure 3 are standard 9-pin sub D jacks with strain reliefs (Refer to the Remote Control section for details).

**CONTACT CLOSURE PORT**

The contact closure remote control switch port in Figure 3 is a standard 9-pin sub D jack with strain reliefs (Refer to the Remote Control section for details).

![Figure 3. DSP5024 Back Panel](image)

**IQ INTERFACE PORT**

The Crown IQ serial communications control port in Figure 4 is a 4-pin removable screw terminal header (Refer to the IQ Control section for details).

**IQ ADDRESS SWITCH**

The IQ device address select switch in Figure 4, numbered 1-8 from left to right, is an 8-lever DIP switch (Refer to the IQ Control section for details).

**DSP LED**

The DSP LED in Figure 4 is a yellow LED that flashes when the device is either sending/receiving data to/from the IQ bus. It can also be turned on or off via an IQ command.

![Figure 4. DSP5024IQ Back Panel](image)

**IQ AUXILIARY INPUT/OUTPUT PORT**

The "AUX" port in Figure 4 is used for binary control of some external function as well as for monitoring the binary state of some external 5V logic variable. The AUX control equivalent
output circuit is a +15VDC source with a 1 kilohm series resistor. The AUX input circuit can monitor ±5 VDC logic levels. This port can be accessed only through the IQ system.

SOUND SYSTEM CONNECTIONS

BALANCED and UNBALANCED SIGNALS

Balanced audio signal connections are used when audio equipment must be operated in high EMI (electromagnetic interference) environments. They can prevent interference signals, such as those from power transformers or motors, from contaminating an audio signal.

Balancing is a function of 3 things: the driving circuit, the connecting cable and the receiving circuit. All 3 must be balanced in order for a system to be balanced. The DSP series units have electronically balanced inputs and outputs. In order to maintain a balanced audio system, the devices before and after the DSP series unit in the signal chain must be balanced. In addition, the cables between the devices must also be properly configured. Figure 5 and Figure 6 illustrate the correct method for making balanced and unbalanced connections. Although these figures show 3-conductor shielded cable, it is common industry practice to use 2-conductor shielded cable and omit the common connection. Generally, both driving and receiving circuits are referenced to chassis ground with the cable shield tied to chassis ground at one end and left unconnected at the other end.

Note: The DSP series units are factory set with circuit common internally tied to chassis ground. If it is desired to isolate earth ground from chassis, the internal jumper labeled 'E1', which is located on the main circuit board near the power supply, must be clipped out.

![Figure 5. Unbalanced Connections](image)

![Figure 6. Balanced Connections](image)

SIGNAL PROCESSING FEATURES

The DSP series units incorporate many audio functions into a compact, one rack space device. Therefore, a brief explanation of each available function, along with some pertinent application examples, is given in the following section to familiarize the user with the signal processing flexibility and power available to them with this series.

The filter capacity of the DSP series units is measured by the number of available second order filters. Each programmed filter uses up to 2 available filters, depending on the type. After reading the following section, and before programming any filters, fill out the Filter Capacity Worksheet, using the signal flow block diagram in Figure 1 for reference. The DSP series units provide 35 filters for programming, which is usually more than sufficient. However, filling out the worksheet may help you to more efficiently allocate your filters.
Crossovers
All of the DSP series units have fully adjustable crossover filters with the following features:

- 2, 3 or 4 way configuration.
- Butterworth, Bessel or Linkwitz-Riley response shape.
- 6, 12, 18 or 24 dB/octave slope.
- 1 Hz adjustment from 20 Hz - 20 kHz.

Each crossover filter uses either 1 or 2 of the 35 available filters in the DSP series units. 6 or 12 dB/octave crossover types use 1 filter and 18 or 24 dB/octave crossover types use 2 filters.

Crossover filters are programmed by edge. Each of the 4 outputs has a low and high edge which correspond to the low and high crossover frequencies for that output. Each edge is independently programmed for the desired response shape, slope and frequency.

Parametric Filters
All of the DSP series units have fully adjustable parametric filters with the following features:

- Variable width, \( Q = 0.2 \) to 100 (\( BW = 4.8 \) to \( 1/70 \) octaves).
- Variable level, \(+12\) to \(-60\) dB; increment = \( 0.1 \) dB .
- 1 Hz adjustment from 20 Hz - 20 kHz.

Each parametric filter is a 2-pole design and uses up 1 filter.

The following table relates the Q-factor of a parametric filter to its approximate bandwidth in octaves:

<table>
<thead>
<tr>
<th>Q-factor</th>
<th>BW (octaves)</th>
<th>Q-factor</th>
<th>BW (octaves)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.2</td>
<td>4.8</td>
<td>4.3</td>
<td>1/3</td>
</tr>
<tr>
<td>0.27</td>
<td>4</td>
<td>8.7</td>
<td>1/6</td>
</tr>
<tr>
<td>0.67</td>
<td>2</td>
<td>14.4</td>
<td>1/10</td>
</tr>
<tr>
<td>1.4</td>
<td>1</td>
<td>28.9</td>
<td>1/20</td>
</tr>
<tr>
<td>2.1</td>
<td>2/3</td>
<td>43.2</td>
<td>1/30</td>
</tr>
<tr>
<td>2.9</td>
<td>1/2</td>
<td>50</td>
<td>1/35</td>
</tr>
<tr>
<td></td>
<td></td>
<td>100</td>
<td>1/70</td>
</tr>
</tbody>
</table>

Table 1. Q-factor vs. Bandwidth

Delay
The DSP series units have fully adjustable delay on each output with the following features:

- Maximum delay = 688 msec. for DSP5000/IQ and DSP5024/IQ in 1in/4out mode.
- Maximum delay = 344 msec. for DSP5024/IQ in 2in/4out mode.
• Delay increment = 20.8 μsec (microseconds).
• Delay displayed in msec, feet or meters.

Delay does not use up any filters.

This large amount of delay, adjustable in very small increments, makes the DSP series units ideal for echo cancellation in large venues and time alignment of speaker cluster drivers.

**HIGHPASS and LOWPASS FILTERS**
The DSP series units have fully adjustable highpass and lowpass filters with the following features:

• Butterworth response shape.
• 6, 12, 18 or 24 dB/octave slope.
• 1 Hz adjustment from 20 Hz - 20 kHz.

Each highpass or lowpass filter uses either 1 or 2 of the 35 available filters in the DSP series units. 6 or 12 dB/octave types use 1 filter and 18 or 24 dB/octave types use 2 filters.

In addition to their standard uses for bandlimiting, these filters can also be used in conjunction with crossover filters to increase the effective crossover slope.

**SHELVING FILTERS**
The DSP series units have fully adjustable highshelf and lowshelf filters with the following features:

• Variable shelf height, -12 to +12 dB.
• Height increment = 0.1 dB.
• 6 dB/octave slope.
• 1 Hz knee frequency adjustment from 20 Hz - 20 kHz.

Each shelving filter uses 1 of the 35 available filters in the DSP series units.

Many sound systems incorporate CD (constant directivity) horns. A highshelf filter can be used for CD horn correction by setting the knee frequency equal to the CD horn corner frequency and setting the shelf height to ±12dB.

**INPUT and OUTPUT LEVELS**
The DSP series units have fully adjustable input and output levels with the following feature:

• Variable levels, -21 to +27 dBu.

Changing input and output levels does not use up any filters.

The input and output level controls are designed to make it easy to match system input and output levels to the DSP series units. For instance, for a -6 dBu peak input signal, the DSP5000 input
level would be set to -6 dBu. To boost this level so that the peak output level would be 0 dBu, the appropriate output channel level would be set to 0 dBu.

For the suggested procedure in adjusting the input and output level controls for maximum system dynamic range, see the System Setup section.

**OUTPUT LIMITING**
The DSP series units have adjustable output limiting with the following feature:

- Variable level, -21 to +27 dBu.

The DSP series units' hard output limiting is essentially a part of their output level circuit. The output signal will be hard clipped at the programmed output signal level. This is useful as redundant speaker protection over and above other speaker protection methods.

Output limiting does not use up any filters.

**OUTPUT MUTING and INVERSION**
The DSP series units have output muting and inversion functions. These functions are useful for crossover polarity correction, system set-up and troubleshooting.

Output muting and inversion do not use up any filters.

**OUTPUT ASSIGNMENT and MIXING**
The DSP5024/IQ units have adjustable output assignment and mixing with the following feature:

- Output assignable from Left input, Right input or S=L+R inputs.

Output assignments and mixing do not use up any filters.

This feature can be used to generate a summed or "mono" signal from an 2 channel or "stereo" system that can be fed into a recording device, sub-woofer, etc.

**MEMORIES AND PRESETS**
The DSP series units have 3 different types of memories:

- 1 non-volatile scratchpad memory.
- 10 non-volatile memories.
- 10 non-volatile presets.

The scratchpad memory is the active memory. All of the active DSP parameters are viewed and programmed here. This memory is automatically saved when the unit is turned off or loses power and is automatically recalled on power-up.
The DSP series units have 10 non-volatile memories for storing different configurations. Memories are stored by copying the scratchpad memory configuration into the selected memory. Memories are recalled by copying the selected memory into the active scratchpad memory.

The DSP series units also have 10 non-volatile presets. These are generally used for remotely recalling different device configurations using a PA-422 serial interface, a contact closure switch or through a Crown IQ Turbo system. Each preset is assigned to correspond to a memory number. When a preset is remotely recalled, the memory number that is assigned to it is copied into the active scratchpad memory.

System presets and system memories can be assigned alpha-numeric names to aid in identifying them. The following characters are allowed in a name or password:

01234567890ABCDEFGHIJKLMNOPQRSTUVWXYZ/#@&-

REMOTE CONTROL

The DSP5000 and DSP5024 can be remotely controlled via the PA-422 serial interface or the contact closure switch. The DSP5000IQ and DSP5024IQ can be remotely controlled via a Crown IQ interface using Crown Turbo software.

PA-422

PA-422 remote control of the DSP5000/5024 is similar to remote control of the White Instruments 4700 or 4710 equalizers. The DSP5000/5024 and 4700 series units may be networked via their PA-422 ports. A unit, or master, can control from its front panel all units "downstream" from it in the chain. Downstream is defined as in the direction of the PA-422 output connector. Although any unit can be considered as the master to all units downstream from it, in most situations only the first unit in the chain will be used as the master. No computers, external controllers or other devices are necessary. The master can recall presets in all of its slave units simultaneously and can individually address any device to manipulate its settings and/or memories.

Cables

PA-422 interfaces use balanced signal lines, which are inherently tolerant of EMI. Depending on the installation and EMI environment, PA-422 signals can be sent over cables of up to 6500 ft. in length. For cable runs of less than 10 feet, any 9 conductor shielded cable will be sufficient (White Instruments stocks a 2' cable, part # 4704). For runs greater than 10 ft., a shielded cable with at least 4 twisted pairs plus a single conductor will be needed. Shielded 5-pair cable, such as Alpha 5475 or Belden 9505, can be used with one wire left over. Care should be exercised to ensure that the positive and negative lines of each signal are sent over one twisted pair and not intermingled with other signal lines. All 9 conductors should be wired "straight through" (i.e. pin 1 to 1, pin 2 to 2, etc.). The four twisted pairs should be attached to pins 1-2, 4-5, 6-7 and 8-9.
Connectors
9-pin sub D connectors are used to connect the control cables to the device interfaces. Male connectors are used to connect to the output of each interface. Female connectors are used to connect to the input of each interface.

Address Assignment
Before a PA-422 network can be made fully functional, a unique address must be assigned to each unit in the system. Valid addresses range from 1 to 247. The address is assigned with the Set Device Address function under the Utility Menu from the front panel of each unit.

CONTACT CLOSURE
Contact closures allow the user to remotely select any of the user-programmed presets using simple dry switch contact closures. A momentary (>0.25 seconds), or continuous contact between the appropriate pins of the contact closure port is all that is needed to recall one of 10 presets. A high quality single pole, 10 position, shorting or non-shorting switch may be used or the complete White Instruments 4705 Remote Preset Select Switch may be purchased. Digital circuitry can also operate this port as long as the digital circuit's common is connected to pin 9 and the voltage at the other pins never exceeds +5 volts nor goes below zero. Voltages above 3.7 V are read as an open circuit and voltages below 0.8 V are read as a short circuit. Note that the 0.25 second delay built into the system is to prevent recalling unwanted intervening presets if a rotary switch is turned moderately fast.

<table>
<thead>
<tr>
<th>Preset Number</th>
<th>Pins Tied to Pin 9</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>3</td>
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<td>4</td>
<td>4</td>
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<td>6</td>
<td>6</td>
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<tr>
<td>7</td>
<td>7</td>
</tr>
<tr>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>9</td>
<td>7,8</td>
</tr>
<tr>
<td>10</td>
<td>6,8</td>
</tr>
</tbody>
</table>

Table 2. Preset Select Switch Pin Assignments

Cables
A shielded 9-conductor cable, similar to Alpha 1219 or Belden 9539, up to approximately 500 ft. long should be used between the input of the device Remote Preset port and its remote switch. If remote access to fewer than 8 presets is desired, fewer conductors can be used. Table 2. shows the pin connection required to recall each preset. The rule is: the number of conductors needed to access up to 8 presets is equal to the number of presets plus 1. For accessing 8, 9 or 10 presets, only 9 conductors are needed. The shield drain wire should be soldered to the metal connector body at the equalizer end of the cable and left unconnected at the switch end. Figure 7 shows the
recommended wiring diagram for a 10 contact switch.

![Remote Preset Switch Schematic](image)

**CROWN IQ**

DSP5000IQ and DSP5024IQ units can be remotely controlled via a Crown IQ Turbo Control system. This system allows computer control of all of the DSP series unit's functions as well as monitoring capabilities of input and output signal levels. Note: Turbo v.1.3 beta 27 or higher is required.

**Hardware**

The IQ communication network is a 20 milliamp current loop using inexpensive 2-conductor twisted pair cable. In order to connect the DSP5000IQ/5024IQ into the IQ control loop, perform the following steps:

1. Using the 2-conductor cable, connect the + and - terminals of the IQ "IN" port on the DSP5000IQ to the respective + and - terminals of the IQ "OUT" port of the IQ device immediately preceding it in the network.
2. Using the 2-conductor cable, connect the + and - terminals of the IQ "OUT" port on the DSP5000IQ to the respective + and - terminals of the IQ "IN" port of the IQ device immediately following it in the network.
3. Set the IQ Address DIP switch on the back of the DSP5000IQ to the desired binary coded address. Note: that the switch is labeled 1-8 from left to right so that the lever binary weights are, from left to right, 1, 2, 4, 8, 16, 32, 64 and 128.
4. Power-up the unit. Note: The unit power must be cycled off and on for an address change to take effect.

**Software**

Your DSP5000IQ/5024IQ was shipped with a floppy disk that contains 2 files, an "*.OIF" file and a "*.GDMD" file. Both of these files should be copied into the controlling computer's Crown Turbo working directory. Note: Crown Turbo v.1.3 beta 27 or higher is required.

**SECURITY FEATURES**

The DSP series units have 2 user selectable security codes, a 2 character preset access code and a 4 character system access code. These give the user the ability to allow free, limited or no access to the unit. In all, the DSP series units offer 4 levels of security as described below.
1. Free Access

- System access code = —— (all dashes).
- Preset access code = —— (all dashes).

Free access to all system levels.

2. System Secured, Presets Unsecured

- System access code = 4 character password.
- Preset access code = —— (all dashes).

Free access to recall presets. Restricted access to change system settings.

Note: The unit automatically sets the Preset Access Code to be the first 2 characters in the System Access Code. Therefore, after setting the System Access Code, the Preset Access Code will have to be reset to all dashes.

3. System Secured, Presets Secured

- System access code = 4 character password.
- Preset access code = 2 character password.

Restricted access to recall presets. Restricted access to change system settings.

Note: The unit initially sets the Preset Access Code to be the first 2 characters in the System Access Code. The user may change the Preset Access Code, if desired.

4. Security Disabled

- Security permanently disabled (under Utility Menu).

Free access to all system levels. All security disabled IRREVERSIBLY.

NETWORK SECURITY

For a PA-422 networked system, the entire system, or any portion of it, can be secured with password protection. A password entered from the front panel of any unit in the network will secure that unit and all units downstream from it. Thus, to secure an entire network, the password should be entered from the front panel of the first unit in the network. This password is then passed to each unit following it in the network.

With knowledge of the appropriate password, any unit in the chain can be controlled from its own front panel or from any unit preceding it. If different sections of the network need to be secured with different passwords, this can be done by entering the first password (if any) at the start of the chain and then entering the other passwords at appropriate points along the chain, keeping in
mind that each time a password is entered in a unit, all units following that unit will receive the
new password. In addition, accessing any unit in the chain automatically allows access to all units
after it in the chain, regardless of their passwords.

SYSTEM SETUP

The following is a recommended procedure for setting up your DSP series unit.

1. Complete the Filter Capacity Worksheet.

2. Program the DSP series unit's crossovers. If the unit is a DSP5024/5024IQ to be
   operated in 2in/4out mode, set the output assignments.

3. Set the DSP series unit's input and output levels.
   a. Turn off all power amplifiers driven by the DSP5000.
   b. Send program material at the maximum anticipated signal level into the DSP5000.
   c. Set the input level such that the Input Level LED is green most of the time with
      occasional flashes of red.
   d. Set the DSP5000 output levels to -21 dB.
   e. While still passing program material through the DSP5000, turn on all power
      amplifiers and adjust their gains to maximum.
   f. Set the DSP5000 output level so that the signal level at the speakers is at the
      maximum speaker rating.
   g. Reduce the power amplifier gain to achieve the maximum desired sound level out of
      the speakers.

4. Program the rest of the DSP series unit's parameters such as filters and delay.

PROGRAMMING GUIDE

The DSP series units' front panel controls are menu driven. That is, all functions are programmed
by making selections from lists of available options. This allows the user to concentrate on
programming the device rather than having to remember mnemonic commands or valid
parameters.

DSP series units' functions are logically grouped under descriptive headings called menus. Figure
8 shows a block diagram of the DSP series menus and functions.

The most practical way of learning to operate the DSP series units are by programming them. The
number of features available and the menu driven user interface preclude the generation of a short,
but comprehensive, programming guide. Instead, some examples are given to allow the user to
become familiar with the operation of the DSP series units. Afterwards, no references, other than
possibly Figure 8, should be required for programming.
**HOW DO I PROGRAM A SYSTEM ACCESS CODE?**

To program the security code "ABCD", for example, follow these steps.

1. Press the MENU key until the screen appears.

2. Press the FUNCTION key until the screen appears.

3. Use the arrow keys to enter the code "ABCD". The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the character. Hint: Press the LEFT and RIGHT keys simultaneously to clear the code (all dashes).

4. Press the ENTER key to store the code. The unit displays the screen.

**HOW DO I PROGRAM A PRESET ACCESS CODE?**

If you have already programmed a security code, say "ABCD" and would like to program the preset access code "YZ", for example, follow these steps. Note: When programming the system access code, its first 2 characters are automatically copied into the preset access code.

1. Press the MENU key until the screen appears.

2. Press the FUNCTION key until the screen appears.

3. Use the arrow keys to enter the code "YZ". The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the character. Hint: Press the LEFT and RIGHT keys simultaneously to clear the code (all dashes).

4. Press the ENTER key to store the code. The unit displays the screen.

**HOW DO I ENTER A LOCKED UNIT?**

If you have programmed a security access code, say "ABCD", and a preset access code, say "YZ", and you want to enter the locked unit, follow these steps.

1. From the "main boot-up screen" press the MENU key, the screen appears.

2. If you want to recall presets only, then enter the preset access code "YZ" using the arrow keys and press the ENTER key. The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the character. The screen appears. Use the arrow keys to select the desired preset and press the ENTER key to activate it.
3. If you want to access the system, press the FUNCTION key, the ENTER SYSTEM ACCESS CODE: CCD screen appears. Use the arrow keys to enter the system access code "ABCD" and press the ENTER key. The Crossover Adjust Menu screen appears.

**HOW DO I PROGRAM A CROSSED FILTER?**

To program, for example, a Bessel, 18dB/octave, low crossover edge (i.e. highpass filter cutoff frequency) at 1kHz on output channel B, follow these steps.

1. Press the MENU key until the Crossover Adjust Menu screen appears.

2. Press the FUNCTION key until the LOWER screen appears.

3. Press the UP arrow key until the CROSSOVER EDGE L-LO screen appears.

4. Press the FUNCTION key, the 8-LO RESPONSE FLAT screen appears.

5. Press the UP arrow key until the CROSSOVER EDGE 8-LG screen appears. Note: at this time, the input level LED nearest to the LCD display is flashing yellow to indicate that a filter change is pending.

6. Press the FUNCTION key until the LOWER screen appears.

7. Use the arrow keys to set the frequency to 1000Hz. The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the digit.

8. Press the FUNCTION key until the CROSSOVER EDGE L-LO, ROLLOFF 75 dB/octave screen appears.

9. Use the arrow keys to set the rolloff to 18 dB/Octave.

10. Press the ENTER key once, the CHANGES MADE. ENTER TO ACCEPT screen appears.

11. Press the ENTER key again, the PARAMETERS UPDATED screen appears and the flashing yellow input level LED turns off. After a short delay, the Crossover Adjust Menu screen appears. If you don't want to save the changes, press the MENU key instead of the ENTER key and the changes will be discarded.

**HOW DO I PROGRAM A PARAMETRIC FILTER?**

To program, for example, filter number 1 as a parametric filter at 81.7 Hz with a level of +3 dB and a width of 2/3 octave on output channel C, follow these steps.
1. Press the MENU key until the screen appears.

2. Press the FUNCTION key twice or until the screen appears.

3. Use the UP or DOWN arrow key to select channel C, the screen appears.
   Note: at this time, the input level LED nearest to the LCD display is flashing yellow to indicate that a filter change is pending.

4. Press the FUNCTION key to move the cursor into the frequency field.

5. Use the arrow keys to set the frequency to 817 Hz. The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the digit.

6. Press the FUNCTION key until the cursor appears under the level field.

7. Use the arrow keys to set the level to +3.0.

8. Press the FUNCTION key to move the cursor into the Q field.

9. Use the arrow keys to set the Q to 2.1 (see Table 1).

10. Press the ENTER key once, the screen appears.

11. Press the ENTER key again, the screen appears and the flashing yellow input level LED turns off. After a short delay, the screen appears. If you don’t want to save the changes, press the MENU key instead of the ENTER key and the changes will be discarded.

**HOW DO I PROGRAM A HIGHPASS OR LOWPASS FILTER?**

To program, for example, filter number 5 of a DSP5024/IQ as a 24dB/octave, highpass filter at 125 Hz on input channel R, follow these steps. Note: The single input channel on DSP5000/IQ units and on DSP5024/IQ units in 1in/4out or mono mode is labeled "ALL."

1. Press the MENU key until the screen appears.

2. Press the FUNCTION key once, the screen appears.

3. Use the UP or DOWN arrow keys to select filter number 5.
4. Use the UP or DOWN arrow key to select channel R, the screen appears. Note: at this time, the input level LED nearest to the LCD display is flashing yellow to indicate that a filter change is pending.

5. Press the FUNCTION key to move the cursor into the frequency field.

6. Use the arrow keys to set the frequency to 125 Hz. The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the digit.

7. Press the FUNCTION key to move the cursor under the filter field.

8. Use the UP arrow key to select a highpass filter.

9. Press the FUNCTION key to move to cursor to the slope field.

10. Use the UP arrow key to set the slope to 24dB/octave.

11. Press the ENTER key once, the screen appears.

12. Press the ENTER key again, the screen appears and the flashing yellow input level LED turns off. After a short delay, the screen appears. If you don't want to save the changes, press the MENU key instead of the ENTER key and the changes will be discarded.

**HOW DO I PROGRAM A SHELVING FILTER?**

To program, for example, filter number 5 of a DSP5000/IQ or DSP5024/IQ (in 1in/4out or mono mode) as a +12dB highshelf filter at 125 Hz on input channel ALL, do steps 1-5 of “HOW DO I PROGRAM A HIGHPASS or LOWPASS FILTER?” (select channel ALL in step 3) and then follow these steps.

1. Press the FUNCTION key to move the cursor under the filter field.

2. Use the UP arrow key to select a highshelf filter.

3. Press the FUNCTION key to move to the shelf height field.

4. Use the arrow keys to set the shelf height to +12dB.
5. Press the ENTER key once, the screen appears.

6. Press the ENTER key again, the screen appears and the flashing yellow input level LED turns off. After a short delay, the screen appears. If you don't want to save the changes, press the MENU key instead of the ENTER key and the changes will be discarded.

**HOW DO I SELECT 1 IN/4 OUT (MONO) OPERATION?**

If you have a DSP5024/IQ, the default input mode is 2in/4out or stereo. If you would like to operate your unit in 1in/4out or mono mode to increase the available delay, then follow these steps:

1. Press the MENU key until the screen appears.

2. Press the FUNCTION key until the screen appears.

3. Use the arrow keys to select "MONO". The screen appears. Note: at this time, the input level LED nearest to the LCD display is flashing yellow to indicate that a change is pending.

4. Press the ENTER key once. The screen appears.

5. Press the ENTER key again, the screen momentarily appears and the flashing yellow input level LED turns off. After a short delay, the screen appears. If you don't want to save the changes, press the MENU key instead of the ENTER key and the changes will be discarded.

**HOW DO I ASSIGN OUTPUTS?**

If you have a DSP5024/IQ operating in 2in/4out or "stereo" mode, then each output must be assigned to either the Left input, the Right input or a Sum=L+R inputs. For example, to assign output D to be a sum of L+R inputs, follow these steps:

1. Press the MENU key until the screen appears.

2. Press the FUNCTION key until the screen appears.

3. Use the LEFT or RIGHT arrow keys to move the cursor to the D field.
4. Use the UP or DOWN arrow keys to select "S"

**HOW DO I STORE A MEMORY?**

To store a system setting into memory 3, for example, program the active scratchpad memory as desired and then follow these steps. Note: It is recommended to always store a system setting into memory during and after programming it. This prevents loss of the settings if another memory or preset is inadvertently recalled.

1. Press the MENU key until the **MEMORY** screen appears.

2. Press the FUNCTION key until the **STORE** screen appears.

3. Use the UP arrow key to select memory 3.

4. Press the ENTER key to store the memory (the cursor disappears).

**HOW DO I RECALL A MEMORY?**

To recall a system setting from memory 3, for example, follow these steps.

1. Press the MENU key until the **MEMORY** screen appears.

2. Press the FUNCTION key until the **RECALL** screen appears.

3. Use the UP arrow key to select memory 3.

4. Press the ENTER key to store the memory (the cursor disappears).

**HOW DO I NAME A MEMORY?**

To name memory 3 "BALCONY", for example, follow these steps.

1. Press the MENU key until the **MEMORY** screen appears.

2. Press the FUNCTION key until the **NAME** screen appears.

3. Use the UP arrow key to select memory 3.

4. Press the ENTER key to move to the name field.
5. Use the arrow keys to enter the name \textbf{NAME: 3} \textbf{<BALCONY...>}. The LEFT and RIGHT keys move the cursor and the UP and DOWN keys change the character.

6. Press the ENTER key to store the name (the cursor disappears) \textbf{NAME: 3} \textbf{<BALCONY...>}. 

\textbf{HOW DO I ASSIGN A PRESET TO A MEMORY?}

To assign preset 3 to memory 2, for example, follow these steps.

1. Press the MENU key until the \textbf{MEMORY MENU} screen appears.

2. Press the FUNCTION key until the \textbf{ASSIGN PRESETS (ENTER)} screen appears.

3. Press the ENTER key until the \textbf{PSET: 1 = MEM: 1} \textbf{<-------->} screen appears.

4. Use the UP arrow key to select preset 3 and press the RIGHT arrow key. The cursor will move to the memory number field \textbf{PSET: 3 = MEM: 1} \textbf{<-------->}.

5. Use the UP arrow key to select memory 2 and press the ENTER key to activate the assignment (the cursor disappears) \textbf{PSET: 3 = MEM: 2} \textbf{<-------->}.

6. To continue, press the LEFT arrow key to move the cursor back to the preset number field.

\textbf{HOW DO I SET UP FOR PA-422 COMMUNICATION?}

In order to set up a PA-422 communication network, the following must be done:

A. The units in the network must be connected together via their PA-422 ports as described in the PA-422 section of the main manual.

B. Each unit in the network must have its control port set to PA-422.

C. Each unit in the network must be assigned a distinct address between 1 and 247.

Part B above must be done manually to each unit in the network from its own front panel. Part C, however, can be done manually as in Part B or automatically from the front panel of the master unit. Part C is done manually by doing steps 1-4 below for each unit in the network. Part C is done automatically from the master unit by following all of the steps below.

1. Press the MENU key until the \textbf{UTILITY MENU} screen appears.

2. Press the FUNCTION key until the \textbf{CONTROL PORT: NONE} screen appears.

3. Use the arrow keys to select the PA-422 port \textbf{CONTROL PORT: PA-422}. 

4.
4. Press the FUNCTION key until the [SET LOCAL ADDRESS] screen appears. Use the arrow keys to set the address. For a master unit, as in this example, this address is usually set to 1.

5. Press the MENU key until the [DEVICE SELECT] screen appears. Note: this menu only appears when the PA-422 control port is enabled.

6. Press the FUNCTION key until the [INIT REMOTE ADDRESSES] screen appears.

7. Use the arrow keys to select "YES" and press the FUNCTION key. The [ARE YOU SURE?] screen appears.

8. Use the arrow keys to select "YES" and press the ENTER key, the [REMOTE DEVICES FOUND: X] screen appears. "X" is the number of remote units found and automatically assigned sequential addresses beyond the address selected for the master in step 4 above.

9. Press the FUNCTION key until the [DEVICE SELECT 1] screen appears.

10. Use the arrow keys to select the remote device to be accessed. When a remote device is being accessed, that device's front panel will display "REMOTE ACCESS" and all of the menus and functions seen from the master unit will be those of the remote unit.

**HOW DO I SET UP FOR CONTACT CLOSURE OPERATION?**

Contact closures may be operated "stand-alone" with one unit or in conjunction with PA-422 for a network of devices. For a network:

A. Set up for normal PA-422 operation as described in the previous section.
B. Connect your contact closure switch to the Contact Closure Port of the master unit.
C. Follow the steps below for the master unit.

For stand-alone operation, do only parts B and C.

1. Press the MENU key until the [UTILITY] screen appears.

2. Press the FUNCTION key until the [DISABLED] screen appears.

3. Use the arrow keys to select "PRESETS" [CONTACT CLOSURE PRESETS]
SPECIFICATIONS

Number of Channels:
1in/4out for DSP5000/5000IQ.
2in/4out for DSP5024/5024IQ.

Number of Available 2-Pole Filters:
22 available for programming as crossover, parametric,
shelving or highpass/lowpass filters.

Frequency Response:
20 Hz (-0.5dB) to 20 kHz (-1dB).

Dynamic Range:
107 dB.

Distortion:
Less than 0.05% at 27 dBu into 600Ω.

Input Level Range:
-21 to +27 dBu.

Output Level Range:
-21 to +27 dBu into 600Ω.

Input Impedance:
30 kΩ balanced, 10 kΩ unbalanced.

Input Circuit:
Active servo-balanced. Can operate unbalanced with no
gain change.

Output Impedance:
102Ω balanced, 51Ω unbalanced.

Output Circuit:
Active servo-balanced. Can operate unbalanced with no
gain change.

Controls:
Front panel buttons with backlit LCD.

Crossover Filters:
- 2, 3 or 4 way,
- 6, 12, 18 or 24 dB/octave,
- Bessel, Butterworth, Linkwitz-Riley,
- 20 Hz - 20 kHz, 1 Hz increments.

Parametric Filters:
- Q = 0.2 to 100 (4.3 to 1.70 octave),
- level = -50 to +12 dB, 0.1 dB increments,
- 20 Hz - 20 kHz, 1 Hz increments.

Delay:
1in/4out - 688 msec., 20.8 μsec increments.
2in/4out - 344 msec., 20.8 μsec increments.

Shelving Filters:
- level = -12 to +12 dB, 0.1 dB increments.
- 20 Hz - 20 kHz, 1 Hz increments.

Highpass/Lowpass Filters:
- 6, 12, 18 or 24 dB/octave.
- Butterworth response.
- 20 Hz - 20 kHz, 1 Hz increments.

Output Assignment (DSP5024/5024IQ):
Outputs assignable from Left, Right or
Sum=Left+Right inputs when in 2in/4out mode.

Muting:
Outputs muted on power-up and power-down. Manual
output muting.

Inversion:

Number of Memories:
Ten, non-volatile.

Number of Presets:
Ten, non-volatile.

Remote Control:
- PA-422 for DSP5000/5024.
- Contact closure for DSP5000/5024.
- Crown IQ for DSP5000IQ/5024IQ.

Connector Types:
Input/output - type XLR; PA-422 - 9-pin sub D
connector, IQ - 4 pin removable screw terminal.

Power Requirements:
105-130VAC, 50/60 Hz, 30 Watts.
UL Listed.

Dimensions and Weight:
1.75" (4.5 cm) X 19" (48.3 cm) X 12" (30.5 cm),
rack mount. Weight 8 lbs. (3.6 kg).

Safety Agency Approval:
105-150 VAC version is UL listed.

We reserve the right to improve our products and change features and specifications without notice.
WARRANTY POLICY

Your White Instruments DSP series unit is warranted against defects in manufacturing, workmanship and original components for a period of ONE YEAR from the date of purchase. During this period, White Instruments will repair or replace the unit, at our option, so long as it has not been subjected to abuse. Abuse may be physical and/or electrical in nature. White Instruments will be the sole judge of this criteria.

White Instruments is the only warranty repair facility in the United States. Outside the United States, White Instruments distributors are authorized to make warranty repairs.

HOW TO OBTAIN WARRANTY REPAIRS

Your unit should be securely packed and shipped, prepaid, to White Instruments or one of its authorized offshore distributors. A return authorization is required.

Our U.S. shipping address may be found on the front of this manual. Contact the factory for the name and address of the offshore distributor nearest you. A copy of your sales receipt should be included to establish the warranty date. Without it we will have to rely on the serial number, which indicates when we originally shipped the device to a dealer.

A letter detailing the unit's symptoms, with your name, shipping address and telephone number, must be included. Every effort will be made to complete repairs within 5 working days. Your unit will be returned to you via surface freight, prepaid. If you instruct us to return your unit via air freight, it will be shipped with freight charges collect.

HOW TO OBTAIN OUT-OF-WARRANTY REPAIRS

Your unit should be securely packed and shipped, prepaid, to White Instruments or one of its authorized offshore distributors. A return authorization is required.

Should the required repairs not be covered under warranty, you will be charged for parts, labor and freight. Should you require an estimate prior to repairing the unit, you should notify White Instruments of this when returning the unit. Every effort will be made to complete the repair within 5 working days. The unit will be returned C.O.D. unless otherwise instructed.

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Note, however, this product is covered by White Instruments' standard product warranty.
FILTER CAPACITY WORKSHEET
for DSP5000/5000IQ/5024/5024IQ

Title: ______________________ Date: ______________________

This worksheet is designed to help you determine the maximum number of filters available for a DSP series unit. To use it, fill in the desired configuration and calculate the final capacity index.

Definitions

filter - a frequency selective response entered in either the crossover or equalizer sections. Possible filters are: Butterworth (BW), Bessel (BE), Linkwitz-Riley (LR), Parametric (PM), Highpass (HPF), Lowpass (LPF), Highshelf (HSHLF), and Lowshelf (LSHLF).

degde - a crossover output bandpass response edge, high or low.

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<td># of 18 or 24 dB/oct. BW or BE edges</td>
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<td># of 18 or 24 dB/oct HPFs or LPFs</td>
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TOTAL

If the total is less than 36, the configuration will fit in one DSP series unit.
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Appendix — Compression and Limiting Feature

Version 2.22 and higher firmware supports compression/limiting on each output channel. The limiting and compression functions are related but have different effects on the audio signal.

Limiting prevents analog clipping of the signal by monitoring the output signal and adjusting the output gain. When limiting kicks in, it may be possible to detect slight distortion in the output signal. As such, the suggested use of limiting is to enforce a maximum signal amplitude for the purposes of protecting equipment.

Compression/expansion is also a mechanism for reducing the amplitude of signals, but engages and releases more slowly. As such, it is not used so much for protecting equipment as for limiting the dynamic range of the audio signal in a less noticeable fashion. When the compressor senses a large amplitude signal, it gradually decreases the output gain. When the signal drops back down to a lower lever, it gradually increases the output gain. The levels at which compression turns on, the amount of reduction it employs, and the speed at which it reacts to changes in signal level are all parameters that the user can set.

The expander works exactly the same as the compressor, except that instead of reducing signals that are too high, it squelches signals that are too low. Signals below the expander threshold, such as low-level background noise, are attenuated. This feature reduces random noise between active signal periods.

Operational Description

The DSP502X/Q allows limiting to be employed on each output channel individually. Limiting can be enabled by descending into the COMP-LIMITER menu and setting the appropriate channels to '1' in the COMP-LIM ON/OFF screen. Once limiting has been enabled on any of the channels, the user may choose to employ compression on these same channels by pressing the FUNCTION key and then toggling the COMPRESSION ENABLED field. When compression is enabled it will be automatically on for every channel that has limiting selected. If limiting is disabled on all channels, compression is automatically reset to a DISABLED state.

Limiting, once enabled, will intelligently scale over-driven signals to the maximum level just below analog clipping. There are no parameters which need to be set by the user. Remember that limiting only corrects high-amplitude signals caused by filtering and the output gain; it will not prevent input clipping issues.

Compression and expansion are specified by a number of parameters. Several set a "gain curve" as shown in figure 1. Several more set the attack and release times which govern how quickly compression/expansion engages and releases, respectively. Each parameter can be accessed by pressing the FUNCTION key and scrolling through the various menu screens.
The parameters that control the gain curve are the expander threshold and ratio, and the thresholds and ratios of two compressor regions. The function of these parameters are best understood by illustrating the resulting gain curve, shown in figure 1 below.

Figure 1. Gain curve described by expander/compressor parameters

"Compressor 1" and "compressor 2" are merely names of convenience; either can be set with the lower threshold. In figure 1 and the discussion that follows, we have assumed that compressor 1 has a lower threshold setting than compressor 2, but this is not necessarily the case in general.

The curve in figure 1 should be interpreted as follows: Signals that fall below the expander threshold will be reduced by a factor of -NdB for each 1dB that they are below the threshold. The effect of this is to reduce noise or quiet signals that are below the threshold. Signals that fall above the expander 1 threshold but below the compressor 2 threshold will be reduced by a factor of -C1 dB for each 1dB that they are above the threshold. Likewise, signals that fall above the compressor 2 threshold will be reduced by a factor of -C2 dB for each 1dB that they are above the threshold. Signals which do not fall into any of these regions (between the expander and the compressors) have a 1:1 ratio in dB.

Because compression/expansion reduces the overall level of the signal (it can never boost it), a fixed gain parameter can be used to shift the above gain curve a given amount up or down.
The attack and release times control the speed at which compression occurs. A small attack time means that if a large signal is encountered, the gain is adjusted relatively quickly. Likewise, a small release time means that once the signal returns to normal levels, the gain is quickly returned to its original level. These parameters can be adjusted to make the system responsive to quick transients in signal level, or to follow the average amplitude of the signal over a longer period.

The expander threshold can be set from -108dB to -18dB. The compressor thresholds can range from -90dB to 0dB. The expander supports ratios from 1:60 to 1:1, and the compressors support inf:1 to 1:1. Attack times can be set from 5s to 1ms, and release times can be specified for 10 minutes to .1s.

The menu for COMP LINK GROUPS controls how compression on one channel can affect the other. When two channels are linked, the lowest after-compression gain of the two channels is used for both channels. If signals on multiple channels need to maintain their volume relationship (so as to keep a stereo image centered, for instance), this will ensure proper function even when one channel is compressing. Channels can be assigned to one of two link groups (denoted "1" and "2") or none (denoted by ".-"). Note that a link group can have up to all four channels in it.

**Computational Cost**

There is a computational penalty of four filters per channel of limiting. For example, if three channels have limiting enabled (regardless of compression being enabled/disabled), the maximum number of filters available will be reduced from 35 to 23. If the user attempts to enable limiting on an additional channel when the number of filters already being used prohibits this (based on the highest number filter employed), the request will be denied. Please note that number of filters being used does not determine this limit, but the number of the highest-numbered filter employed. Thus even if EQ filter number 35 is the only filter being used, the system will be seen to operate at maximum capacity and no limiting may be employed.