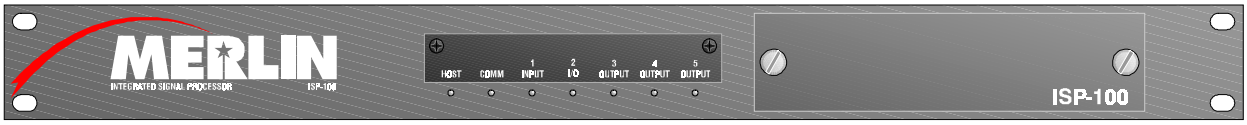
MERLIN™
INTEGRATED SIGNAL PROCESSING

ISP-100™



INTEGRATED SIGNAL PROCESSOR

User's Manual

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Technical questions should be directed to:

Customer Service Department
Merlin/Telex
9600 Aldrich Avenue South
Minneapolis, MN 55420 U.S.A.
Telephone: (612) 884-4051
Fax: (612) 884-0043

RETURN SHIPPING INSTRUCTIONS

Procedure for Returns

If a repair is necessary, contact the dealer where this unit was purchased.

If repair through the dealer is not possible, obtain a RETURN AUTHORIZATION from:

Customer Service Department
Telex Communications, Inc.
Telephone: 1-800-828-6107 or (612) 884-4051 extension 425 or 420
Fax: 1-800-323-0498 or (612) 884-0043

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Be prepared to provide the company name, address, phone number, a person to contact regarding the repair, the type and quantity of equipment, a description of the problem and the serial number(s).

Shipping to Manufacturer for Repair or Adjustment

All shipments of Merlin products should be made via United Parcel Service or the best available shipper prepaid. The equipment should be shipped in the original packing carton; if that is not available, use any suitable container that is rigid and of adequate size. If a substitute container is used, the equipment should be wrapped in paper and surrounded with at least four inches of excelsior or similar shock-absorbing material. All shipments must be sent to the following address and must include the Return Authorization.

Factory Service department
Telex Communications, Inc.
West 1st Street
Blue Earth, MN 56013 U.S.A.

Upon completion of any repair the equipment will be returned via United Parcel Service or specified shipper collect.

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INTRODUCTION

An Overview of the ISP-100

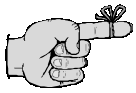
Thank you for choosing the ISP-100™ for your signal processing needs. The ISP-100 is the first in a line of high quality signal processing products from TELEX Communications, Inc.'s Merlin division. The ISP-100 represents a continued commitment to understanding and meeting our customers needs through the development of innovative and intuitive solutions.

The ISP-100 is designed with flexibility in mind, which results in a powerful tool that allows you to adapt to ever-changing markets and applications. The utmost in quality is also a prime directive in the design and manufacture of this product.

A series of predefined signal path topologies called QuickMAPs™ are offered which enable the designer to quickly define the system's processing structure. This approach allows for a continuation of new processing solutions and/or variations that provide "market specific" templates to help minimize your design time and increase your profit.

This single rack space unit can replace a multitude of traditional analog components. Time and money savings are realized in reduced labor costs for wire harnessing, rack size, assembly, and minimized failure due to reduced interconnection. As a result of this consolidation of processing, overall system performance and audio integrity is greatly increased.

An ergonomically designed software interface called VUE-IT™ provides graphic control panels with the conventional look of signal processors, as well as an advanced and easy to use filter tool panel which graphically displays the configuration of your filter block settings.



The four in eight out configuration is only possible when the second card slot contains a digital I/O card.

One of the most impressive and unique features of the ISP-100 is the configurable input/output (I/O). This feature allows the designer or installer to select between analog input or output modules, and/or a digital input/output combination module. These modules are two channel units and can be intermixed between analog and digital, thus providing a variety of I/O combinations. The ISP-100 supports a maximum of four inputs and eight outputs. The inputs and outputs can be either analog, digital, or a combination of both.

Because of our commitment to providing the utmost in audio quality the dynamic range of the ISP-100 rivals anything currently on the market.

The ISP-100 has a typical noise floor of –110 dBu and a typical THD+N of <0.004%. Propagation time is minimized in the ISP-100. No more than two milliseconds of delay is introduced from any analog input to analog output with full processing and delay set to zero.

The ISP-100 is flexible. A General Purpose Interface (GPI) allows multiple system configurations to be selected without the need of a PC to control the unit. This is accomplished through user-supplied contact

closures. This interface allows users to change system settings directly and/or scroll through various settings.

About this Manual

This manual is covers the installation and operation of the ISP-100's hardware and software.

The ISP-100 has been designed to be as user-friendly as possible. However, this manual should be read before attempting to install or operate the ISP-100.

Customer Support

Be sure to fill out the customer support registration card included in the software envelope. This will register you in our database of users.

Conventions Used in this Manual

The margins in this manual include space for icons that provide additional information to the user. The icons and their uses are given in Table 1-1.

Calls attention to potential problems.	Calls attention to remind users to perform a task or action.
Calls attention to helpful tips.	Calls attention to things that could cause damage.

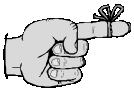
Table 1-1 Information icons.

SETUP & INSTALLATION

Introduction

This section details the setup and installation of the MERLIN ISP-100. Information is provided on the following: front and rear panel features, physical requirements, installation of expansion cards, signal connections for audio, data, and control.

Unpacking



Save the shipping carton in case the ISP-100 needs to be returned for service.

The shipping carton is specially designed to protect the ISP-100 while transporting under normal conditions. It is still possible for damage to occur. Therefore, carefully inspect the outside carton for signs of abuse. If for any reason the ISP-100 should need to be returned, use the shipping carton that it came in. TELEX Communications, Inc. cannot warranty against damage that occurs as a result of improper packaging.

The shipping box should contain the following items:

- ISP-100
- User's Manual
- VUE-IT Software Package
- Spare Fuses (2)
- Label Paper
- IEC Power Cord

Front Panel Features (see Figure 2-1)

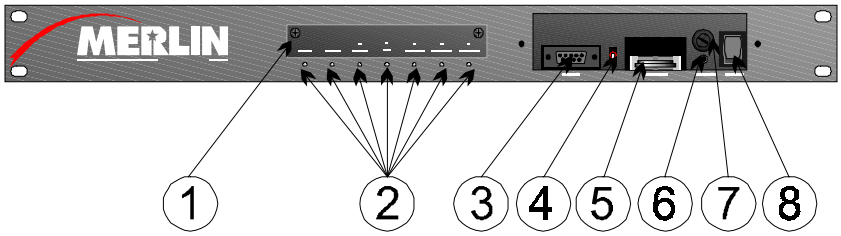


Figure 2-1 ISP-100 front panel features.

1. Label Holder—Provides a space to indicate the assignments of the input and output modules.
2. Status LEDs—Provides power on/off, low battery, module presence, signal clipping, and host communication status.
3. RS-232 Port—Links the PC to the unit via a DB-9 female connector.
4. RS-232 Selector—Selects between the front panel DB-9 port and the rear panel RJ-45 port. Placing the switch in the down position selects the DB-9 port located on the front of the unit. Placing the switch in the up position selects the RJ-45 port located on the rear of the unit.
5. Battery—Provides memory backup power. The expected life is approximately four years.
6. Default Switch—Used to place unit in OS download request mode.
7. Fuse—Protects the power supply.
8. Power—Turns the unit on and off. Placing the switch in the down position turns the unit on. Placing the switch in the up position turns the unit off.

Rear Panel Features (see Figure 2-2)

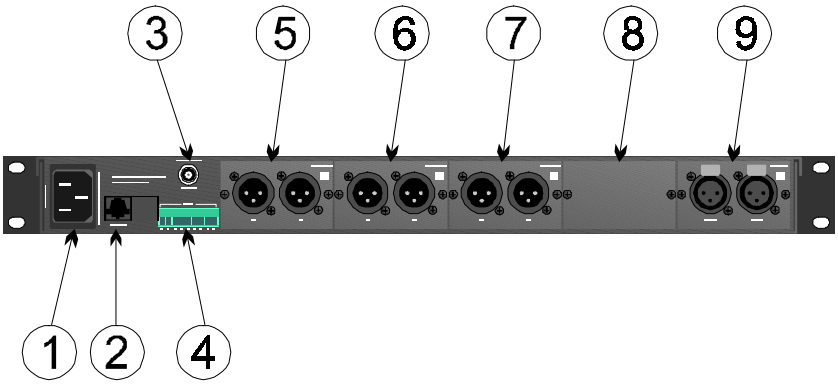


Figure 2-2 ISP-100 rear panel features.

1. AC Power—Connects to mains via an IEC type AC power cord.
2. RS-232 Port—Links the PC to the unit via a RJ-45 connector.
3. External Sync—Precision frequency reference input to the unit.
4. GPI—Provides a General Purpose Interface for the user. The user provides inputs to the system via dry or electronic contact closure inputs. These inputs can be used to change system settings directly and/or scroll through various settings.
5. Output Slot 5—Accepts either an analog output module or digital module.
6. Output Slot 4—Accepts either an analog output module or digital module.
7. Output Slot 3—Accepts either an analog output module or digital module.
8. Input/Output Slot 2—Accepts an analog input module, analog output module, or digital module.
9. Input Slot 1—Accepts either an analog input module or digital module.

Power Requirements

The ISP-100 uses a universal switching power supply that accepts input voltages in the range of 90 VAC to 264 VAC, 50/60 Hz. All countries using 220/240 VAC except Australia use a ½ amp slow-blow fuse. All countries using 100/120 VAC and Australia use a 1 amp slow-blow fuse.

Operating Environment

Internal temperature: 50°F to 95°F (10°C to 35°C)
Humidity: 20% to 80%

Rack Mounting

The ISP-100 may be installed in a standard 19-inch (483-mm) equipment rack with one rack unit (1.75 inches or 44.5 mm) of vertical rack space per unit.

Card Installation Procedure

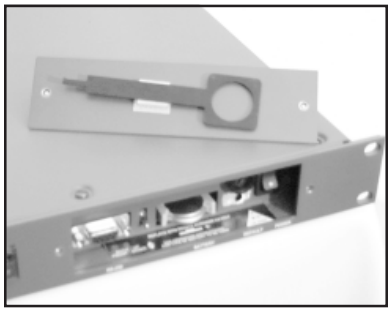


Figure 2-3 Battery/module tool.

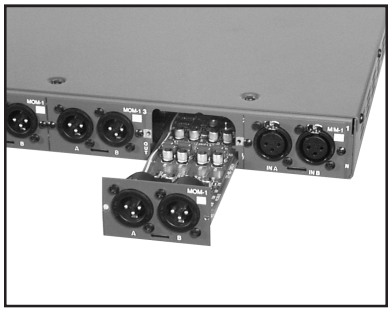


Figure 2-4 Card being inserted.

CAUTION:
Failure to observe anti-static handling procedures could result in damage to equipment.

1. Ground yourself using a wrist strap and anti-static mat.
2. Turn off the ISP-100.
3. Remove the screws securing the module slot cover.
4. If you are not replacing an existing card please proceed to step 6.
5. Use the battery/module removal tool to remove the existing module. The battery/module removal tool is attached to the inside of the front security cover. Place the hook end of the tool into the slot at the bottom of the module and pull. Place the old module in an anti-static bag. See Figure 2-3 for more information.
6. Being careful to avoid scraping the underside of the board, slide the new module into the bottom slots on the card guides located inside the opening. Push the module into the unit until you feel resistance. See Figures 2-4 and 2-5.
7. Firmly push on the module's faceplate to seat the module.
8. Replace the screws that were removed in step 3.
9. Turn on the ISP-100 and verify that the slot indicator for the module is either solid green or blinking green. If the slot indicator is not solid green or blinking green, then there is a problem with the current hardware and/or software. Please refer to the Troubleshooting section to identify the problem.

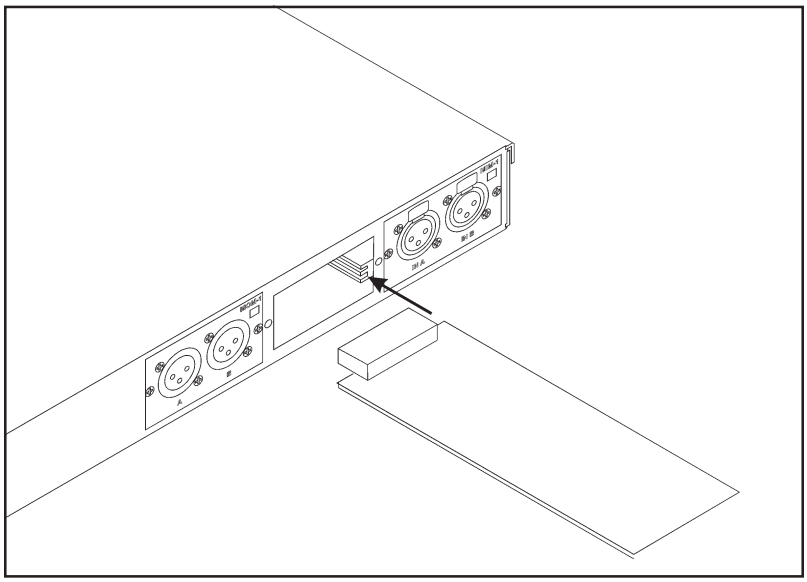


Figure 2-5 Card insert detail.

Cable Diagrams

Analog Audio (MIM-1, MIM-2, MOM-1)

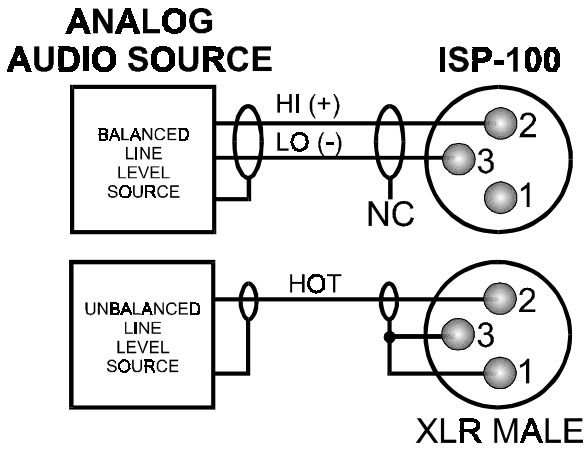
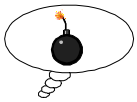


Figure 2-6 Analog inputs.



Using low-cost connectors can save money in the short term, but failures could cost more.

Balanced Input from Balanced Output
Connect the positive (+) side of the line to pin 2 of the male 3-pin XLR connector and the negative (-) side of the line to pin 3 of the connector. In keeping with standard wiring practices, the shield should not be connected at this end; it should only be connected to pin 1 of the 3-pin female XLR connector or ground of the source end. See Figure 2-6 for more information.

Balanced Input from Unbalanced Output
Connect the “hot” wire to pin 2 of the 3-pin male XLR connector and the shield wire to pin 1 of the connector. To avoid a 6 dB drop in level, connect pin 3 to pin 1 of the connector. See Figure 2-6 for more information.

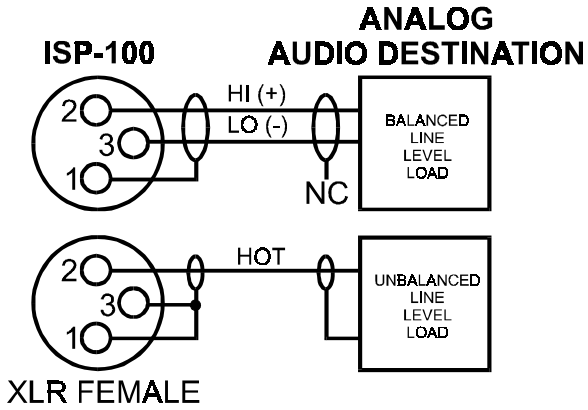


Figure 2-7 Analog outputs.

Confused about what type of cable to use? Try Belden 8451 or 8723. Both are good quality and inexpensive.

Balanced Output to Balanced Input
Connect the positive (+) side of the line to pin 2 of the 3-pin female XLR connector and the negative (-) side of the line to pin 3 of the connector. In keeping with standard wiring practices, the shield should be connected at this end; it should not be connected to pin 1 of the male 3-pin XLR connector or ground of the load end. See Figure 2-7 for more information.

Balanced Output to Unbalanced Input
Connect the “hot” wire to pin 2 of the 3-pin female XLR connector and the shield wire to pin 1 of the connector. To avoid a 6 dB drop in level, connect pin 3 to pin 1 of the connector. See Figure 2-7 for more information.

Digital Audio (MDM-1 Cards)

Use the white block on the I/O module panels to write a reference number or other designator on by using a permanent marker. Then make identical labels for the cables that plug into the module.

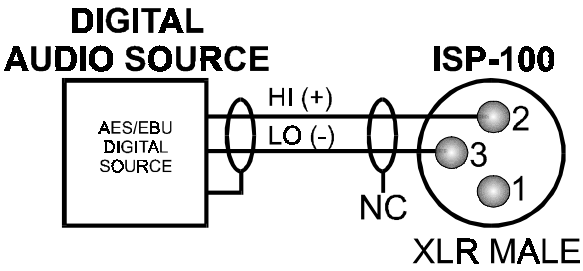
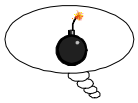


Figure 2-8 AES/EBU inputs.

AES/EBU Input
Connect the positive (+) side of the line to pin 2 of the male 3-pin XLR connector and the negative (-) side of the line to pin 3 of the connector. In keeping with standard wiring practices, the shield should not be connected at this end; it should only be connected to pin 1 of the 3-pin female XLR connector or ground of the source end. The cable used should be 110Ω AES/EBU compliant cable such as MOGAMI 3080. See Figure 2-8 for more information.

SPDIF Input



Long unbalanced cable runs (analog or digital) are not recommended.

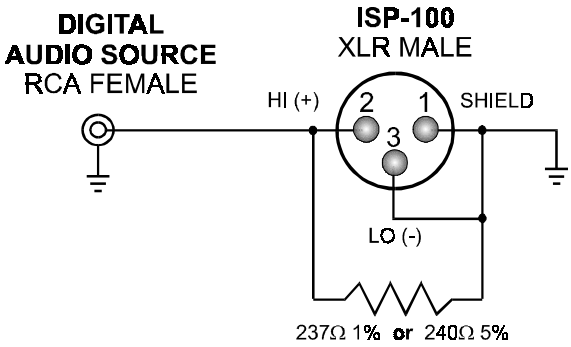


Figure 2-9 SPDIF inputs.

Connect the positive (+) side of the line to pin 2 of the male 3-pin XLR connector and the shield of the cable to pins 1 and 3. Connect a ¼ watt 237Ω 1% resistor between pins 1 and 2 of the 3-pin male XLR connector. Connect the positive (+) side of the line to the center pin of the RCA connector on the opposite end of the cable. Connect the shield to the shield of the RCA connector on the opposite end of the cable. The cable used should be RG-59 or other similar 75Ω cable. See Figure 2-9 or the application note MDM-1 SPDIF COMPATABILITY P/N 42-02-053086 for more information.

AES/EBU Output

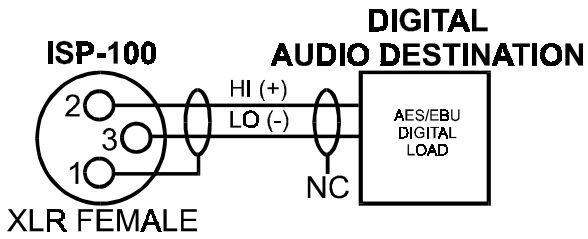


Figure 2-10 AES/EBU output.

Connect the positive (+) side of the line to pin 2 of the 3-pin female XLR connector and the negative (-) side of the line to pin 3 of the connector. In keeping with standard wiring practices, the shield should be connected at this end; it should not be connected to pin 1 of the male 3-pin XLR connector or ground of the receiving end. The cable used should be 110Ω AES/EBU compliant cable such as MOGAMI 3080. See Figure 2-10 for more information.

SPDIF Output

The ISP-100 is not capable of providing SPDIF outputs.

Serial Communications

Serial communications with the ISP-100 can be accomplished either through the DB-9 connector located under the front panel security cover or through the RJ-45 connector located on the rear panel. The computer connected to the ISP-100 must have a free COM port (1-4). Communications with the ISP-100 will not work properly if the COM port is shared with another device such as a TEF™ analyzer. It is important to set the serial port selection switch located under the front panel security cover to the serial port used. The cable used should be of low capacitance and suited to data communications.

When tuning the system with a computer controled analyzer use a separate computer for the analyzer. Using two PCs will save time.

Front Panel RS-232

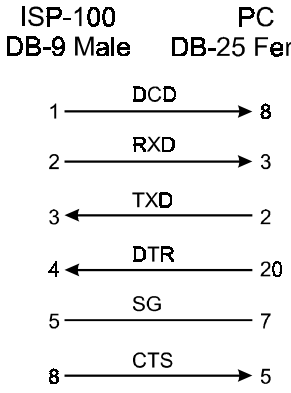


Figure 2-11 DB-9 to DB-25.

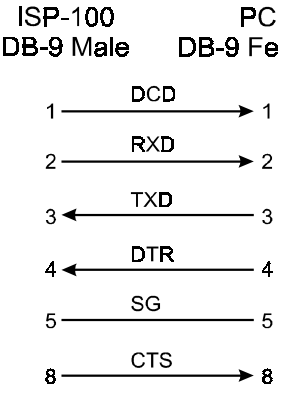
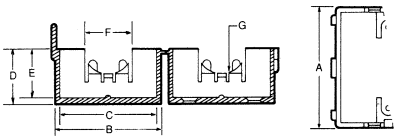


Figure 2-12 DB-9 to DB-9.

The serial port selector switch should be set in the down position. If the PC's COM port is a DB-25 connector, wire the cable according to Figure 2-11. If the PC's COM port is a DB-9 connector, wire the cable according to Figure 2-12. A snap-on

Steward
Part No. 28B2025-0A0

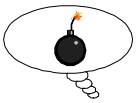


Data

Units	A DIM	B DIM	C DIM	D DIM	E DIM	F DIM	G DIM	Z @ 25 MHz (ohms)	Z @ 100 MHz (ohms)
in	1.280	0.705	0.610	0.362	0.303	0.276	0.138	168	298
mm	31.75	17.91	15.50	9.19	7.70	7.01	3.51		

Steward
East 36th Street
Chattanooga, TN 37401
USA
Tel (615) 867-4100 Fax (615) 867-4102

Figure 2-13 RFI choke data.



Not using an RFI choke while using the front panel serial port could cause increased interference with other devices and void the FCC compliance.

RFI choke such as Steward PN: 28B2025-0A0 should be used at the ISP-100 end of the cable. See Figures 2-13 and 2-16, and Table 2-1 for more information.

Rear Panel RS-232

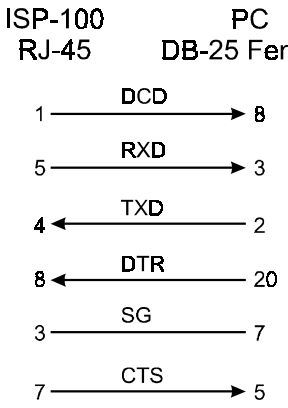


Figure 2-14 RJ-45 to DB-25.

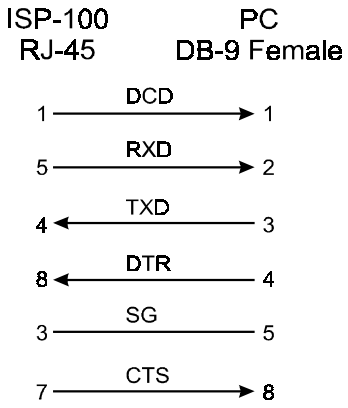
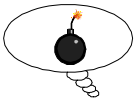


Figure 2-15 RJ-45 to DB-9.

The serial port selector switch should be set in the up position. If the PC's COM port is a DB-25 connector, wire the cable according to Figure 2-14. If the PC's COM port is a DB-9 connector, wire the cable according to Figure 2-15. No RFI choke is needed. See Figure 2-16 and Table 2-1 for more information.



RS-232 cable runs over 50 feet could cause unreliable communications. It is better to use RS-485 converters for connections over 50 feet. Be sure to read the application note on this topic before buying converters.

Connections Over 50 Feet

The EIA specifications for RS-232 communications allow for connections up to 50 feet. For longer connections, the RS-232 signals must be converted to a standard, such as RS-485, that is tolerant of long distances. The converter on the PC end of the connection can be PC powered, but the converter at the ISP-100 end of the connection must have its own power supply. Before buying converters please contact TELEX Communications, Inc. for an application note on this topic.

RJ-45 Pin #	DB-9 Pin #	DB-25 Pin #	Signal Name	Direction
7	8	5	Clear To Send (CTS)	To PC
1	1	8	Data Carrier Detect (DCD)	To PC
*	6	6	Data Set Ready (DSR)	To PC
8	4	20	Data Terminal Ready (DTR)	From PC
5	2	3	Receive Data (RXD)	To PC
2	7	4	Request To Send (RTS)	From PC
*	9	22	Ring Indicator (RI)	To PC
3 & 6	5	7	Signal Ground (SG)	N/A
4	3	2	Transmit Data (TXD)	From PC

* Not Implemented

Table 2-1 RS-232 signal pinouts for RJ-45, DB-9, and DB-25 connectors.

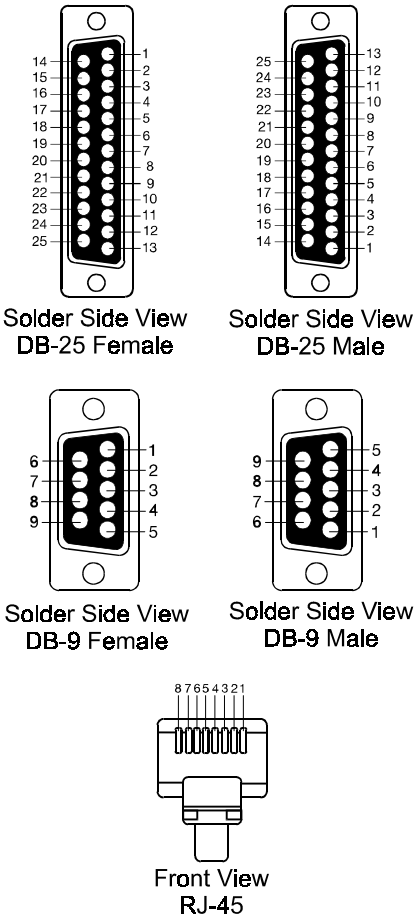


Figure 2-16 Connector pinouts.

The RS-232 standard is very common to computer controlled devices. Table 2-1 and Figure 2-16 can help you decipher the RS-232 interfaces of many devices.

General Purpose Inputs (GPI)

If a permanent contact closure is used, the ISP-100 will remember the QuickSET selected even if the AC power fails.

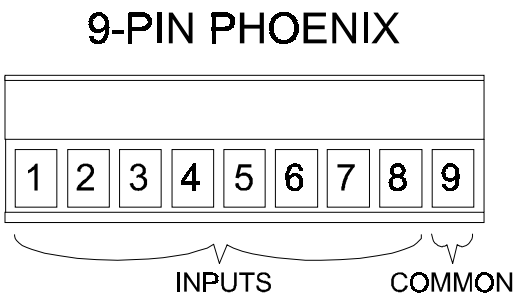


Figure 2-17 GPI connector pinout.

Dry (i.e. switch) or electronic (i.e. open collector) contact closure should be sufficient. The closure can be either momentary (10 ms minimum duration) or continuous. Connect the desired input (1-8) through the contact closure to common (9). See Figure 2-17 for more information.

Master Sync Input

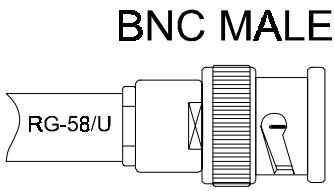


Figure 2-18 BNC cable example.

Synchronizing to an external clock source requires RG-58 or other suitable 50Ω cable terminated with a BNC connector. A precision (±5 ppm) clock source of 12.288 MHz, 6.144MHz, or 3.072 MHz is supported. The Master Clock Input must be set to external from the **Properties for Merlin ISP-100** panel in the VUE-IT software. See Figure 2-18 and page 7-6 of this manual for more information.

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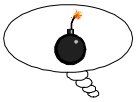
TROUBLESHOOTING

> > > CAUTION < < <

NO USER SERVICEABLE PARTS INSIDE. HAZARDOUS VOLTAGES AND CURRENTS MAY BE ENCOUNTERED WITHIN THE CHASSIS. TO AVOID ELECTRICAL SHOCK DO NOT PERFORM ANY SERVICING OTHER THAN THAT CONTAINED IN THESE OPERATING INSTRUCTIONS.



User modifications void the warranty.

User Modifications

Modifications to Merlin products are not recommended. Such modifications shall be at the sole expense of the person(s) or company responsible, and any damage resulting from said modifications shall not be covered under warranty or otherwise.

In Case of Problems

Check the following items:

- Verify that the unit is properly connected to an AC power source and the source is active.
- Check the fuse.
- Verify that the input connections are properly made.
- Verify that the output connections are properly made.
- Make sure the output relay indicators are green.
- Check the input and output cables for proper wiring and continuity.
- Check the signal source(s).
- Check the connection between the computer and the ISP-100 if a computer is being used.
- Check the GPI connections if the GPI is being used.

WARNING:
Make sure the power is disconnected from the unit before checking the fuse.

Diagnostics

LED IDENTIFIER	CAUSE OR MEANING
Host LED Off	Loss of internal or external power.
Host LED Green	Memory is OK.
Host LED Yellow	Memory was reset.
Host LED Red	An input-only card is in an output only slot or vice versa.
COMM LED Off	No communications activity.
COMM LED Green	Communications activity.
COMM LED Yellow	None.
COMM LED Red	Communications error.
Slot LED (1-5) Off	Slot does not have a card inserted.
Slot LED (1-5) Green	Card Inserted In slot.
Slot LED (1-5) Yellow	None.
Slot LED (1-5) Red	Clipping occurred on one or both inputs and/or outputs on card in slot.
Host LED Blinks Green	Memory backup battery voltage low (replace battery).
Host LED Red & Slot (1-5) LED Blinks Red	An input only card is in an output only slot OR an output only card is in an input only slot.
Slot (1-5) LED Blinks Yellow	The selected QuickMAP expects a card different from the one currently in the slot OR the slot is empty and the selected QuickMAP requires a card in the slot.
Slot (1-5) Blinks Green	A card was identified, but is not needed by the selected QuickMAP.
Rotating Red LED	Factory initialization mode. Initiated by holding in the paper clip switch located under the security panel while turning on the unit.
LEDs Light Yellow When Power Is Applied	Memory Is In the same state as It was when the unIt was turned off.
LEDs Light Red When Power is Applied	Memory is clear. This results from a factory initialization.
COMM, Slot 4, and Slot 5 LEDs Blink Red	An unrecoverable hardware error has occurred. Contact tech support.

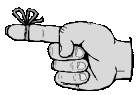
Table 3-1 LED diagnostic messages.

The ISP-100 was designed with built-in diagnostics that activate when power is applied to the unit. During the power-up cycle, the ISP-100 performs several self-checks. The ISP-100 also will report problems, as they happen, while the unit is running. The diagnostic messages are communicated through the front panel LEDs. Refer to Table 3-1 for a list of common LED conditions and corresponding diagnostic messages.

Factory Default Mode

A yellow host LED indicates factory default mode. This mode usually results from either an OS update or memory erasure. When in this mode a default QuickMAP and QuickSET are loaded that prevents the unit from passing damaging signals. The QuickMAP configures the ISP-100 as a straight through processor with high pass filters on the outputs. The corner frequencies of the high pass filters are set to 20 kHz.

To enter factory default mode, hold in the default switch (see Figure 2-1, item 6) while turning on the power. Continue to hold in the switch until the front panel LEDs turn red and begin to rotate.



Note that when the factory default QuickMAP is loaded one or more slot LEDs may blink yellow. This is normal operation and not an error.

Retrieving Version Numbers

Figure 3-1 About VUE-IT dialog

Application, OS, and DSP version numbers can be obtained by clicking on the **Help** menu and then selecting the **About** entry. See Figure 3-1. If an ISP-100 is not online, only the application version number will be reported.

Retrieving Serial Numbers

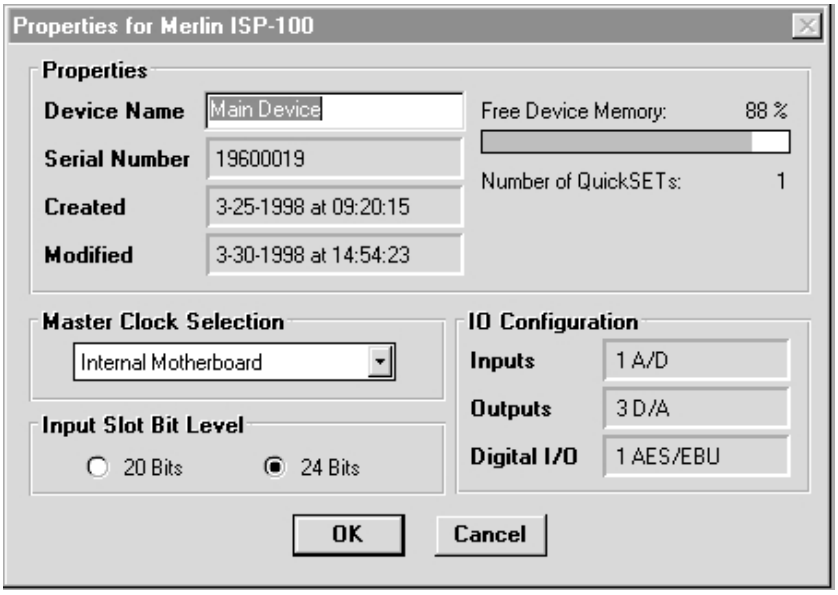


Figure 3-2 Properties dialog.

Clicking on the star icon in the Project Manager and then clicking on the **Properties** button will reveal the ISP-100's serial number. See Figure 3-2.

Changing the Fuse

The universal switching power supply in the ISP-100 adapts to input voltages from 90 to 264 VAC, 50/60 Hz. To reduce risk of fire, replace only with same type fuse. The fuse holder is located under the security cover on the front panel.

WARNING:
Disconnect AC power before changing the fuse. Failure to do so could result in a shock hazard.

The fuse is a standard 20mm X 5mm European type. The values used are as follows:

All countries using 220/240 VAC, except Australia, use a ½ A slow-blow fuse.

WARNING:
Failure to install the proper fuse could result in damage to the unit, property, and loss of life.

Schurter P/N: 0001.2501
Telex P/N: 51-04-052488

All countries using 100/120 VAC and Australia use a 1 A slow-blow fuse.

Schurter P/N: 0001.2504
Telex P/N: 51-04-051487

Changing the Battery

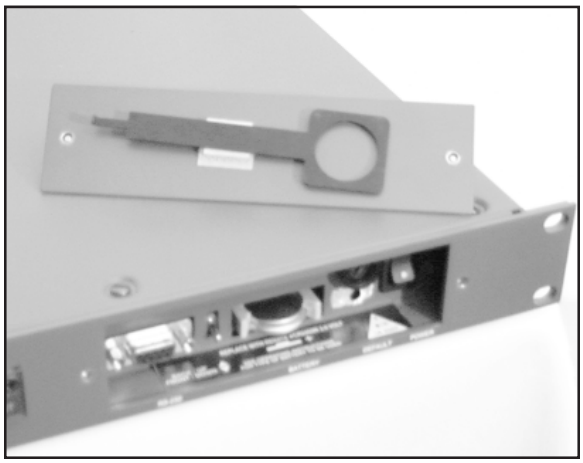
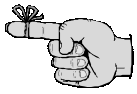


Figure 3-3 Battery/module removal tool.

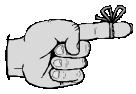
The memory backup battery located under the security cover on the front panel has an expected life of four years. If the Host LED is blinking the battery needs to be changed.

Leave the AC power connected and the ISP-100 turned on when changing a failed battery. Failure to do so will result in loss of data.



CAUTION: The battery used in this device may present a risk of fire or chemical burn if mistreated. Do not recharge, disassemble, heat above 212°F (100°C) or incinerate. Replace battery with Renata CR2450N or MERLIN 40-02-052055 only. Use of another battery may present a risk of fire or explosion.

Be sure to change the battery with the unit on. If the unit is turned off, all of the setup information stored in the ISP-100 will be lost (i.e. the unit will return to factory default mode).



Saving the ISP-100 configuration data to disk will guard against the loss of data in the event AC power fails before a failed battery has been changed.

Caution: Use only the tool attached to the inside of the security cover to change the battery. The tool is black anodized so it will not short out the battery or anything else. DO NOT under any circumstance use pliers to grip the battery. Doing so could cause damage. See Figure 3-3.

Replace the battery with a Renata CR2450N battery. Replacement battery/module removal tools as well as replacement batteries are available from Telex. Their part numbers are:

Battery/Module Removal Tool	**40-04-052619**
Replacement Battery	**40-02-052055**

Dispose of used battery promptly. Keep away from children. Do not disassemble and do not dispose of in fire.

Denmark

Advarsel!
Lithiumbatteri. Eksplosionsfare ved feijlagtig handtering.
af samme fabrikat og type.
Lever det brugte batteri tilbage till leverandoren.

FINLAND

VAROITUS: Paristo voi rajahtaa, jos se on virheellisesti asennettu. Vaihda paristo ainoastaan valmistajan suosittelemaan tyyppun. Havita kaytetty paristo valmistajan ohjeiden mukaisesti.

VARNING: Explosionsfara vid felaktigt batteribyte. Anvand samma batterityp eller en eller en ekvivalent typ som rekommenderas av tillverkaren. Kassera anvant batteri enligt fabrikantens instruktion.

SWEDEN

VARNING:
Felaktigt batteribyte kan medfora fara for explosion. Anvand darfor endast samma typ eller likvardig typ enligt apparattillverkarens rekommendation.

Kassera forbrukade batterier enligt tillverkarens anvisning.

Technical Support

Technical questions should be directed to:

Customer Service Department - Merlin
Telex Communications, Inc.
9600 Aldrich Avenue South
Minneapolis, MN 55420 U.S.A.
Telephone: (612) 884-4051
Fax: (612) 884-0043

Helpful Information

Be prepared to provide the following information (if possible):

1. A detailed description of the problem.
2. QuickMAP being used.
3. Serial Number(s)
4. Version numbers for application, OS, and DSP software.
5. Front panel LED information.
6. PC operating system used.
7. COM port used.
8. PC type and amount of memory installed.

Procedure for Returns

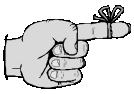
If a repair is necessary, contact the dealer where this unit was purchased.

If repair through the dealer is not possible, obtain a RETURN AUTHORIZATION from:

Customer Service Department
Telex Communications, Inc.
Telephone: 1-800-828-6107 or (612) 884-4051
Fax: 1-800-323-0498 or (612) 884-0043

DO NOT RETURN ANY EQUIPMENT DIRECTLY TO THE FACTORY WITHOUT FIRST OBTAINING A RETURN AUTHORIZATION.

Be prepared to provide the company name, address, phone number, a person to contact regarding the repair, the type and quantity of equipment, a description of the problem and the serial number(s).



Save the box the ISP-100 was shipped in. It makes a convenient way to package a repair unit.

Shipping to Manufacturer for Repair or Adjustment

All shipments of Merlin products should be made via United Parcel Service or the best available shipper prepaid. The equipment should be shipped in the original packing carton; if that is not available, use any suitable container that is rigid and of adequate size. If a substitute container is used, the equipment should be wrapped in paper and surrounded with at least four inches of excelsior or similar shock-absorbing material. All shipments must be sent to the following address and must include the Return Authorization.

Factory Service department
Telex Communications, Inc.
West 1st Street
Blue Earth, MN 56013 U.S.A.

Upon completion of any repair the equipment will be returned via United Parcel Service or specified shipper collect.

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SPECIFICATIONS

Components

Compressor

Property	Range	Default	Fine	Coarse
Bypass	in/out	out	n/a	n/a
Threshold	-60 dB to 0 dB	0 dB	.5 dB	3 dB
Compression Ratio	1.2:1 to 24:1	1.2:1	n/a	n/a
Attack Time	20 msec to 50 ms	20 msec	.02 ms	1 ms
Release Time	20 msec to 5 sec	5000 ms	1 ms	100 ms
Detection Window	20 msec to 5 sec	50 ms	1 ms	100 ms
Crest Factor	0.00 to 1.00	.70	.01	.05
Knee	hard/soft	hard	n/a	n/a
Sidechain Channel	self, max of both, or sidechain	depends	n/a	n/a

Crossover

Property	Range	Default	Fine	Coarse
Slope	6,12,18,24	12	n/a	n/a
Class	Bessel, Butterworth, or Linkwitz-Riley	Butterworth	n/a	n/a
Cutoff Frequency	20 Hz to 20 kHz	Depends	1/12 Oct.	1/3 Oct.
Gain	+0 dB to -96 dB	0 dB	0.5 dB	1 dB
Absolute Polarity	+/-	+	n/a	n/a

Delay*

Property	Range	Default	Fine	Coarse
Delay	0 ms to 2.7 s	0 ms	.02 ms	1 ms

Dither

Property	Range	Default	Fine	Coarse
Bypass	in/out	out	n/a	n/a
Level	16 to 24 bits	20 bits	1 bit	1 bit

Gain**

Property	Range	Default	Fine	Coarse
Input Gain	+18 dB to -96 dB	0 dB	0.5 dB	1 dB
Mute	on/off	off	n/a	n/a
Absolute Polarity	+/-	+	n/a	n/a

Gate

Property	Range	Default	Fine	Coarse
Bypass	in/out	out	n/a	n/a
Threshold	-60 dB to 0 dB	-60 dB	.5 dB	3 dB
Attenuation	0d B to -100 dB	0 dB	.5 dB	3 dB
Attack Time	20 msec to 50 ms	.02	.02 ms	1 ms
Release Time	20 msec to 5 sec	5000 ms	1 ms	100 ms
Detection Window	20 msec to 5 sec	50 ms	1 ms	100 ms
Key Channel	self, max of both, or sidechain	depends	n/a	n/a

* Depends on QuickMAP selected.
** In digital domain.

Limiter

Property	Range	Default	Fine	Coarse
Bypass	in/out	out	n/a	n/a
Threshold	-60 dB to 0 dB	0 dB	.5 dB	3 dB
Attack Time	20 msec to 50 ms	.02 ms	.02 ms	1 ms
Release Time	20 msec to 5 sec	5000 ms	1 ms	100 ms
Detection Window	20 msec to 5 sec	50 ms	1 ms	100 ms
Crest Factor	0.00 to 1.00	.70	.01	.05
Knee	hard/soft	hard	n/a	n/a
Sidechain Channel	self, max of both, or sidechain	depends	n/a	n/a

Filters

Property	Range	Default	Fine	Coarse
Bypass	in/out	out	n/a	n/a
Absolute Polarity	+/-	+	n/a	n/a
Gain Trim	+12 dB to -12 dB	0 dB	.1 dB	1 dB

LowPass Filter

Property	Range	Default	Fine	Coarse
Slope	6,12	12	n/a	n/a
Cutoff Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Class	Bessel, Butterworth, or Linkwitz-Riley	Butterworth	n/a	n/a

HighPass Filter

Property	Range	Default	Fine	Coarse
Slope	6,12	12	n/a	n/a
Cutoff Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Class	Bessel, Butterworth, or Linkwitz-Riley	Butterworth	n/a	n/a

AllPass Filter

Property	Range	Default	Fine	Coarse
Cutoff Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Filter Order	1,2	2	n/a	n/a

LowShelf Filter

Property	Range	Default	Fine	Coarse
Slope	6,12	12	n/a	n/a
Cutoff Frequency	20 Hz to 20k Hz	20	1/12 Oct.	1/3 Oct.
Boost/Cut	+12 dB to -12 dB	0 dB	.1 dB	1 dB

HighShelf Filter

Property	Range	Default	Fine	Coarse
Slope	6,12	12	n/a	n/a
Cutoff Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Boost/Cut	+12 dB to -12 dB	0 dB	.1 dB	1 dB

Notch Filter

Property	Range	Default	Fine	Coarse
Center Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Cut	0 dB to -50dB	0 dB	1 dB	6 dB
Bandwidth	.08 to 1.0 octaves	1.0	.01 Oct.	.1 Oct.

PEQ

Property	Range	Default	Fine	Coarse
Center Frequency	20 Hz to 20 kHz	20	1/12 Oct.	1/3 Oct.
Boost/Cut	+12 dB to -12 dB	0 dB	.1 dB	1 dB
Bandwidth	0.08 to 3.0 octaves	1.0	1/12 Oct.	1/3 Oct.

Combine

Property	Range	Default	Fine	Coarse
Channel “1” Entry Gain	+0 dB to -96 dB	0 dB	.5 dB	1 dB
Channel “1” Entry Polarity	+/-	+	n/a	n/a
Channel “2” Entry Gain	+0 dB to -96 dB	0 dB	.5 dB	1 dB
Channel “2” Entry Polarity	+/-	+	n/a	n/a

Input

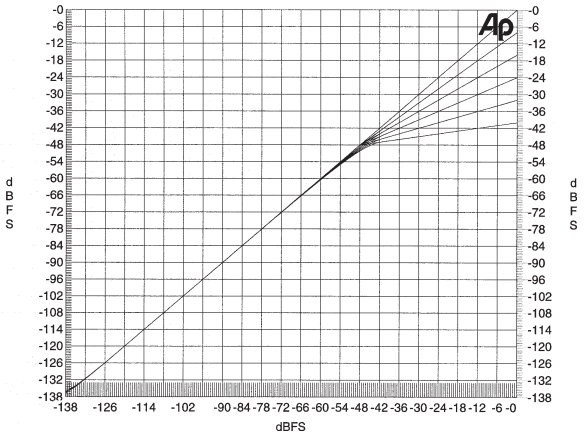
Property	Range	Default	Fine	Coarse
Left Gain (A)	0,2,4,8 dB	0 dB	n/a	n/a
Right Gain (B)	0,2,4,8 dB	0 dB	n/a	n/a
Left Pad (A)	-20 dB	Off	n/a	n/a
Right Pad (B)	-20 dB	Off	n/a	n/a

Output

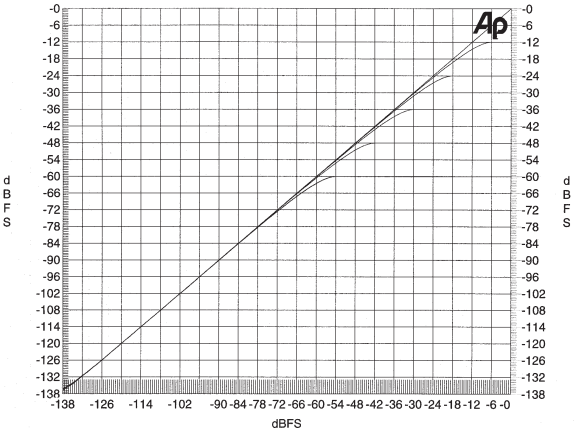
Property	Range	Default	Fine	Coarse
Left Gain (A)	-8,-6,-4,0,8 dB	0 dB	n/a	n/a
Right Gain (B)	-8,-6,-4,0,8 dB	0 dB	n/a	n/a
Left Mute (A)	on/off	off	n/a	n/a
Right Mute (B)	on/off	off	n/a	n/a
Relay Mute	on/off			

ISP-100 Component Plots

Compressor

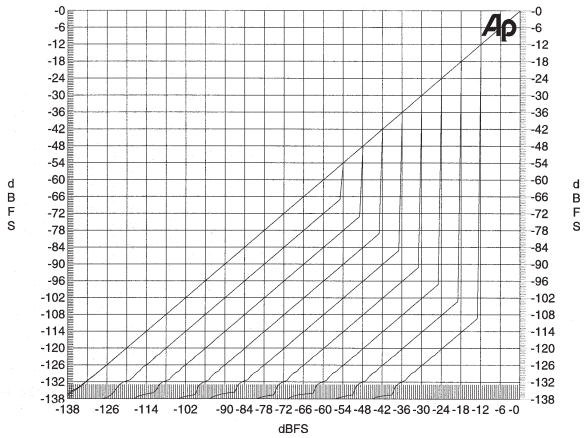


Limiter

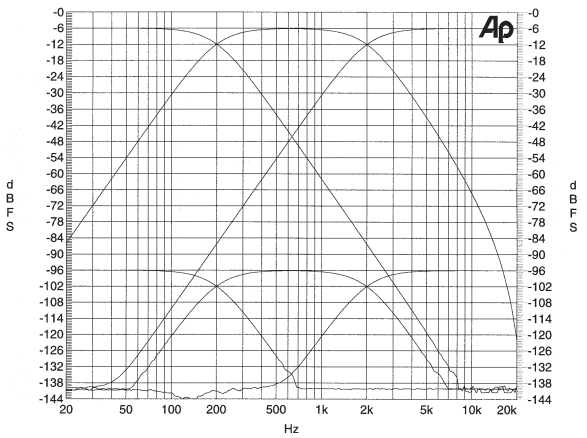


AP is a trademark of AudioPrecision, Inc.

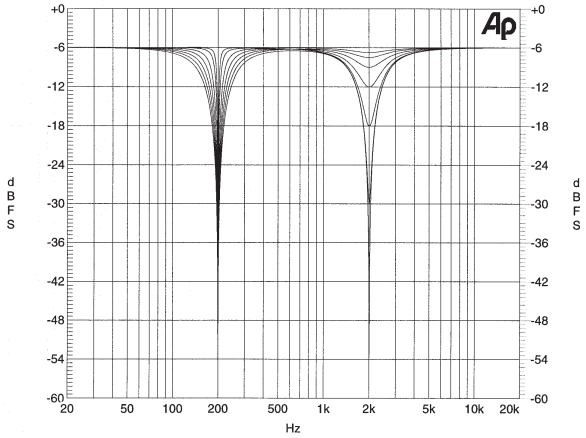
Gate



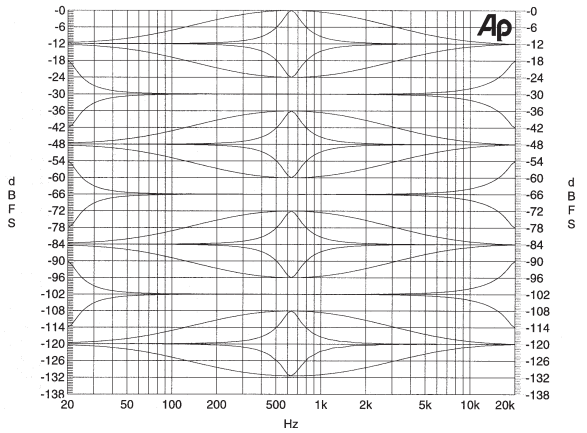
Linkwitz-Riley 3-way Crossover



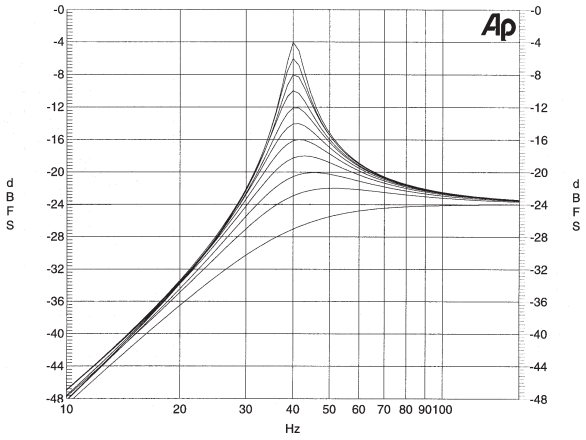
Notch Filter



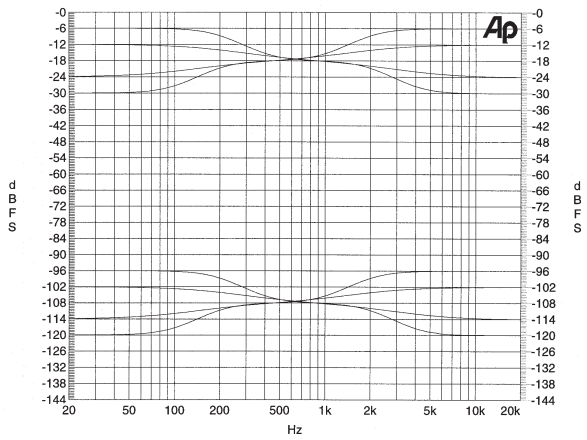
Parametric Equalizer Filter



Peaked HighPass Filter



Shelving Filter



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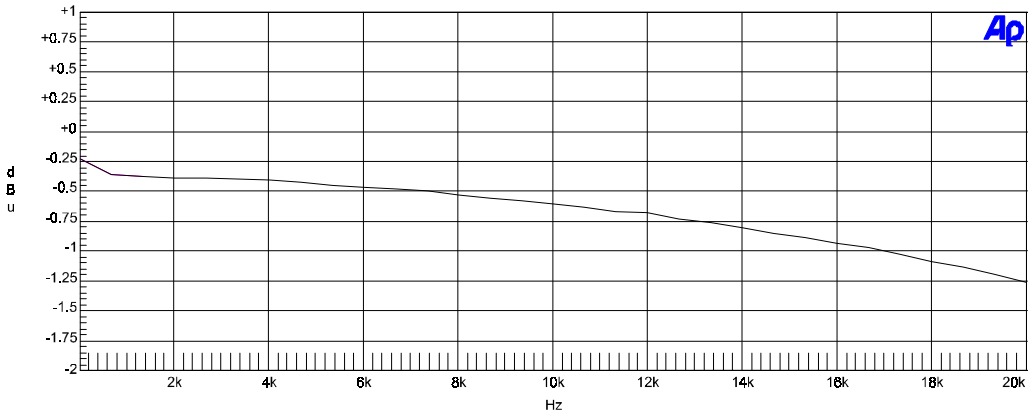
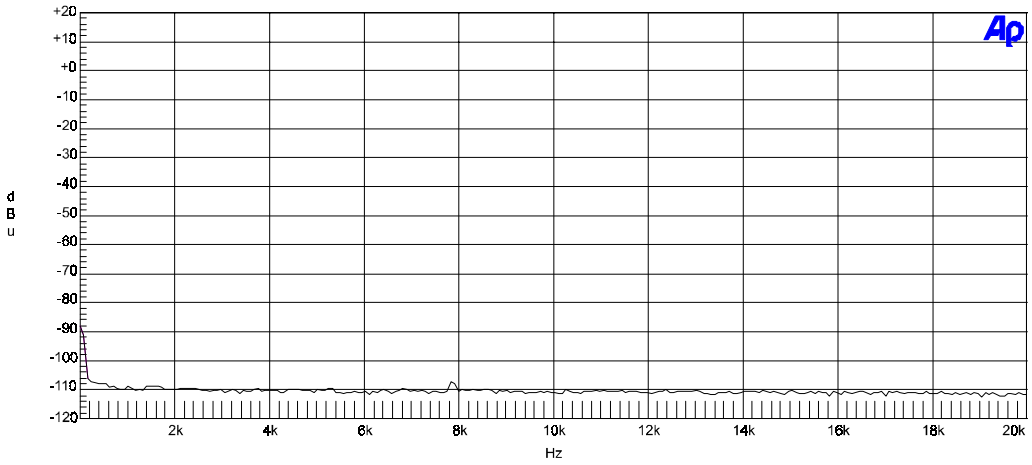
System Measurements

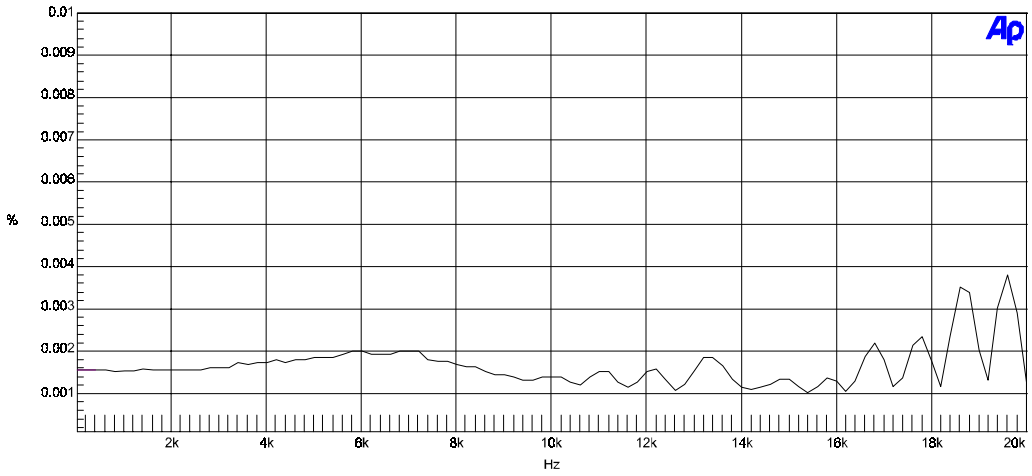
System Frequency Response



System Noise Floor



System THD+N



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MIM-1 & MIM-2* Analog Input Modules

Power

Symbol	Description	Minimum	Typical	Maximum	Units
VA18P	positive analog supply voltage	17.50	18.00	20.00	volts
VA18M	negative analog supply voltage	-17.50	-18.00	-20.00	volts
VD5P	positive digital supply voltage	4.75	5.00	5.50	volts
IA18P	positive analog supply current	77(43.7)	81(46.0)	85(48.3)	milliamps
IA18M	negative analog supply current	72(41.6)	76(43.75)	80(46.0)	milliamps
ID5P	positive digital supply current	113(114)	123(124)	133(134)	milliamps

Performance

Symbol	Description	Minimum	Typical	Maximum	Units
THD+N	Total Harmonic Distortion + Noise		0.003(0.002)	0.007(0.006)	%
IMD	Intermodulation Distortion[1]		0.004	0.007	%
DIM	Dynamic Intermodulation Distortion[2]		0.003	0.006	%
	Noise Floor[3]		110	107	dBu
	Channel Crosstalk		-100	-80	dB
N	Resolution		20(24)		bits
	Frequency Response		+0.01, - 0.1	+.02, -0.15	dB
	Delay[4]	18 (38.7)			WST
MCLK+	Master Clock	12.283	12.288	12.292	MHz
WST+	Word Clock	47.98	48.000	48.02	kHz
	Oversampling		64X(128X)		
	Stopband Attenuation	100(110)			dB
	Maximum Input Level[5]	26	30	32	dBu
	Interchannel Phase Deviation[6]		0.5	1.0	degrees
CMRR	Common Mode Rejection		52	40	dB
	Gain Accuracy[7]		±0.05	±0.25	dB
Z_{in}	Input Impedance[8]	11000	12000	13500	ohms
	Operating Temperature	50	75	100	°F
	Operating Humidity	20	50	80	%

Miscellaneous

Overall Size:[9]	8.40” long by 2.55” wide by 1.50” high
Weight:	4.0 ounces
Synchronous Sampling:	Multiple cards can be synchronized via DPD signal on connector
Input connector:	3 Pin standard female XLR
Mating connector:	3 Pin standard male XLR
Internal connector:	48 pin standard female “F” size per DIN 41 612
XLR pinout:	Pin 1 – shield, Pin 2 – positive signal, Pin 3 – negative signal
Serial Interface:	Master or slave, selectable via SPI control I/F.
Clock Mode:	256 or 384 times sample rate, selectable via SPI control I/F.
Digital Power Down:	Selectable via SPI control I/F.
Analog Gain:	Selectable: 0, -20, -18, -16, -12, 2, 4, 8 dB.

Notes

(Unless otherwise specified measurements are at unity gain with +16 dBu signal.

[1] Unity gain, SMPTE 4:1, 60 Hz & 7 kHz sine waves.

[2] Unity gain, DIM30 4:1, 3.15 kHz SQ wave & 15 kHz sine wave, 30 kHz LPF.

[3] Power spectral density.

[4] Group Delay inherent in conversion process. WST=$^1/_{48\ kHz}$

[5] 20 dB pad engaged.

[6] Maximum levels are differential and balanced. Limit set by protection circuit.

[7] Relative to unity gain, 1 kHz.

[8] Electronically balanced, RF surpressed, differential impedance 20- 20kHz.

[9] Maximum dimensions: dominated by panel size for height and width.

* MIM-2 differences parenthetically noted.

MOM-1 Analog Output Module

Power

Symbol	Description	Minimum	Typical	Maximum	Units
VA18P	positive analog supply voltage	17.50	18.00	20.00	volts
VA18M	negative analog supply voltage	-17.50	-18.00	-20.00	volts
VD5P	positive digital supply voltage	4.75	5.00	5.50	volts
IA18P	positive analog supply current	92	96	100	milliamps
IA18M	negative analog supply current	47	51	55	milliamps
ID5P	positive digital supply current	114	124	134	milliamps

Performance

Symbol	Description	Minimum	Typical	Maximum	Units
THD+N	Total Harmonic Distortion + Noise		0.003	0.007	%
IMD	Intermodulation Distortion[1]		0.004	0.007	%
DIM	Dynamic Intermodulation Distortion[2]		0.003	0.006	%
	Noise Floor		110	107	dBu
	Channel Crosstalk		-100	-80	dB
N	Resolution		20		bits
	Frequency Response		+0.1, - 0.5	+.2, -1.0	dB
	Delay[3]	25			WST
MCLK+	Master Clock	12.283	12.288	12.292	MHz
WST+	Word Clock	47.98	48.000	48.02	kHz
	Oversampling		128X		
	Stopband Attenuation	75			dB
	Maximum Output Level[4]	21	26		dBu
	Interchannel Phase Deviation		0.5	1.0	degrees
	Output Balance		0.5		dB
	Gain Accuracy[5]		±0.05	±0.25	dB
Z_{out}	Output Impedance[6]	594	600	606	ohms
	Operating Temperature	50	75	100	°F
	Operating Humidity	20	50	80	%

Miscellaneous

Overall Size:	8.40” long by 2.55” wide by 1.50” high
Weight:	4.3 ounces
Input connector:	3 Pin standard male XLR
Mating connector:	3 Pin standard female XLR
Internal connector:	48 pin standard female “F” size per DIN 41 612
XLR pinout:	Pin 1 – shield, Pin 2 – positive signal, Pin 3 – negative signal
De-emphasis:	48 kHz de-emphasis, selectable via SPI control I/F.
Relay mute:	Left and right output relay mute, selectable via SPI control I/F.
DAC mute:	Left and right individually or group selectable via SPI control I/F.
Serial Data Interface:	7 formats available, selectable via SPI control I/F.
Analog Gain:	Selectable: 0, -8, -6, -4, 8 dB via SPI control I/F.[7]

Notes

[1] Unity gain, SMPTE 4:1, 60 Hz & 7 kHz sine waves.
[2] Unity gain, DIM30 4:1, 3.15 kHz SQ wave & 15 kHz sine wave, 30 kHz LPF.
[3] Group Delay inherent in conversion process. WST=$1/_{48\ kHz}$
[4] Minimum levels are single ended, maximum levels are differential and balanced. Bridging load.
[5] Relative to unity gain, 1 kHz.
[6] Electronically balanced, RF surpressed, differential impedance 20- 20kHz.

MDM-1 AES/EBU Digital I/O Module

Power

Symbol	Description	Minimum	Typical	Maximum	Units
VA18P	Pos Analog Supply Voltage	17.50	18.00	20.00	Volts
VA18M	Neg Analog Supply Voltage		Not Used		Volts
VD5P	Pos Digital Supply Voltage	4.75	5.00	5.50	Volts
IA18P	Pos Analog Supply Current	8	9	10	mA
IA18M	Neg Analog Supply Current		Not Used		mA
ID5P	Pos Digital Supply Current	122	155	168	mA

Performance (Sample Rate Converter Engaged)[1]

Symbol	Description	Minimum	Typical	Maximum	Units
F_s	Sample Rate for Digital Input Data[2]	32	48	>54	kHz
THD+N	Total Harmonic Distortion plus Noise[3] (RELATIVE)		0.00002	0.00016	%
IMD	Intermodulation Distortion[4] (SMPTE 4:1)			TBD	%
	Jitter in Output Clock[5] (700 Hz - 100KhZ)			415	pS
	Tolerable Jitter in Input Clock before Loss of Lock[6]			90	nS
	Resolution (input)[7]	4		20	bits
	Resolution (output)[8]	24	24	24	bits
	Frequency Response[9]			+/-0.01	dB
	Dynamic Range[10]		117	120	dB
	Noise Floor[11]		-117		dBFS
	Delay through transmitter[12]		$^{43}/_{64}$		WST+ periods
	Interchannel Phase Deviation[13]		0		Degrees
MCLK/2	Sample Rate Converter Reference Clock Source[14]	4.333	6.144	6.667	MHz
MCLK+	Master Clock	TBD	12.288	TBD	MHz
WSR+	Receive Word Clock[15]		(MCLK+)/256		Hz
WST+	Transmit Word Clock[16]		(MCLK+)/256		Hz
SCKR+	Receive Bit Clock[17]		(MCLK+)/4		Hz
SCKT+	Transmit Bit Clock[18]		(MCLK+)/4		Hz

Performance
(Sample Rate Converter bypassed)

Symbol	Description	Minimum	Typical	Maximum	Units
F_s	Sample Rate for Digital Input Data[19]	TBD	48	96	kHz
THD+N	Total Harmonic Distortion plus Noise[20] (RELATIVE)			0.00002	%
IMD	Intermodulation Distortion[21] (SMPTE 4:1)			0	%
	Jitter in Output Clock		415		pS
	Tolerable Jitter in Input Clock before Loss of Lock[22]		35		nS
	Resolution (input)		24	24	bits
	Resolution (output)	24	24	24	bits
	Frequency Response[23]			+/-0	dB
	Dynamic Range[24]	Equal to Dynamic Range of input signal			dB
	Noise Floor[25]	Equal to Noise Floor of input signal			dB
	Delay through receiver[26]		2		WSR+periods
	Delay through transmitter[27]		$^{43}/_{64}$		WST+periods
	Interchannel Phase Deviation[28]			0	Degrees
MCLK+	Master Clock[29]	TBD	12.288	24.576	MHz
WSR+	Receive Word Clock[30]		(MCLK+)/256		Hz
WST+	Transmit Word Clock[31]		(MCLK+)/256		Hz
SCKR+	Receive Bit Clock[32]		(MCLK+)/4		Hz
SCKT+	Transmit Bit Clock[33]		(MCLK+)/4		Hz
Z	Input Impedance		110		Ohms
Z_{out}^{in}	Output Impedance		110		Ohms

Miscellaneous

Overall Size:	8.40" long by 2.55" wide by 1.50" high
Weight:	3.8 Ounces
Input Connector:	3 Pin standard female XLR
Output Connector:	3 Pin standard male XLR
XLR pinout:	Pin 1 shield, Pin 2 positive input, Pin 3 negative input.
Input Data Format:	AES/EBU (AES3) Professional and S/PDIF. (IEC958) Consumer data formats are automatically identified and correctly decoded.
Output Data Format:	AES/EBU Professional, regardless of input data format.
De-emphasis:	Not implemented; emphasis information from Channel Status Bits is relayed to host via Serial Peripheral Interface (SPI).
Mutes:	Linked Left and Right input mute selectable via SPI control; linked Left and Right output mute selectable via SPI control.
Serial Data Interface:	TBD
Synchronous Sampling:	Multiple outputs may be synchronized via DPDB signal on connector. MDM-1 transmitter is slaved to MCLK+, WST+, and SCKT+ signals. MDM-1 receiver may be configured as Master, to generate MCLK+ (via AES_CLK), WSR+, and SCKR+ signals derived from input data stream, or as Slave, to receive WSR+ and SCKR+ signals from card-edge connector, either with or without the Sample Rate Converter. With sample rate converter engaged, the MDM-1 cannot be configured as the master.
Unless otherwise noted:	
	Signal measurements are at unity gain, +16 dBu and 20 Hz to 20 kHz bandwidth.
	Power measurements are quiescent.
	Operating temperature with non condensing humidity

Notes

[1] Per Analog Devices AD1890 data sheet, where applicable.
[2] With 6.144 MHz MCLK/2 input.
[3] 20 Hz to 20 kHz, full-scale input, through sample rate converter, to output via passthrough.
[4] From input, through sample rate converter, to output via passthrough.
[5] When installed in ISP-100.
[6] When installed in ISP-100.
[7] Input samples longer than 20 bits are truncated to 20 bits.
[8] Least four significant bits are zero filled.
[9] DC to 20 kHz input, through sample rate converter, to output via passthrough.
[10] 20 Hz to 20 kHz, -60 dB input, through sample rate converter, to output via passthrough.
[11] From input, through sample rate converter, to output via passthrough.
[12] From SDO0 pin on card connector, through transmitter, to output. The fractional-sample delay is inherent in the Crystal CS8404 transmitter.
[13] From input, through sample rate converter, to output via passthrough.
[14] Frequency is tripled by a PLL on the MDM-1 before being applied to the sample rate converter.
[15] Must be derived from MCLK+.
[16] Must be derived from MCLK+.
[17] Must be derived from MCLK+.
[18] Must be derived from MCLK+.
[19] Must be nominally 48 kHz when installed in ISP-100.
[20] From input to output via passthrough.
[21] From input to output via passthrough.
[22] When installed in ISP-100.
[23] DC to Fs/2.
[24] From input to output via passthrough.
[25] From input to output via passthrough.
[26] From input, through receiver, to SDI0 pin on card connector.
[27] From SDO0 pin on card connector, through transmitter, to output. The fractional-sample delay is inherent in the Crystal CS8404 transmitter.
[28] From input to output via passthrough.
[29] Must be Fs x 256.
[30] Must be derived from MCLK+.
[31] Must be derived from MCLK+.
[32] Must be derived from MCLK+.
[33] Must be derived from MCLK+.

Physical

Height
1.75 inches (44.5 mm)

Width
19.0 inches (483 mm)

Depth
14.5 inches+0.5” (368 mm + 12.7 mm) for protrusions

Weight
Approx.12.9 lbs. (5.9 kg)

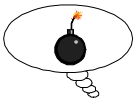
Included Accessories
IEC power cord, user’s manual, software, spare fuse

Power

Universal AC input voltage
90-264 VAC; 50-60 Hz; 35 W Maximum
Internal Power Supply Sync. Frequency - 153.6 kHz

INSTALLING VUE-IT™

Minimum System Requirements



VUE-IT is currently not available for Windows NT.

CPU: Pentium® processor.

OS: Microsoft® Windows 3.1x, Windows 95, or Windows 98

Memory: 8MB RAM

Hard Drive: 5MB of available space (beyond swap file needs)

Disk Drive: 3.5 inch high-density

Monitor: 640X480, 256 color

Pointing Device: mouse or other Windows compatible pointing device

Communications: free COM port (1-4) with 16550 UART

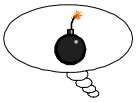
Identifying a Serial Port

The ISP-100 requires a serial port to communicate with the PC. You must determine if you have a free 16550 serial port on the PC you intend to use with the ISP-100. You can do this by looking at the back of the computer for an empty 9-pin D-sub male (DB-9) or a 25-pin D-sub male (DB-25) connector. If you are unable to find an available serial port, then you must either disconnect a device from one of the ports that is currently being used, or install an additional serial port.

If you are unfamiliar with this, please contact your local computer dealer for assistance.

NOTE: It may be necessary to remove device drivers attached to serial ports in Windows 95. A good example of this would be a device driver for an external modem. This will eliminate the possibility of a software conflict between the VUE-IT software and other device drivers.

Loading the Software



Leaving programs running while trying to install VUE-IT could cause problems.

1. Close all programs before beginning installation.

2. Insert the disk labeled “VUE-IT FOR THE ISP-100” into your 3.5-inch floppy drive.

3. Windows 95 users should click on the **Start** button and select **Run**.

4. Windows 3.1x users should click on **File** and select **Run**.

5. Type “A:\setup.exe” (Substitute the appropriate drive letter for your system.)

6. Click on the **OK** button and follow the instructions as they appear on the screen.

7. Reboot the computer.

Updating the ISP-100's Operating System

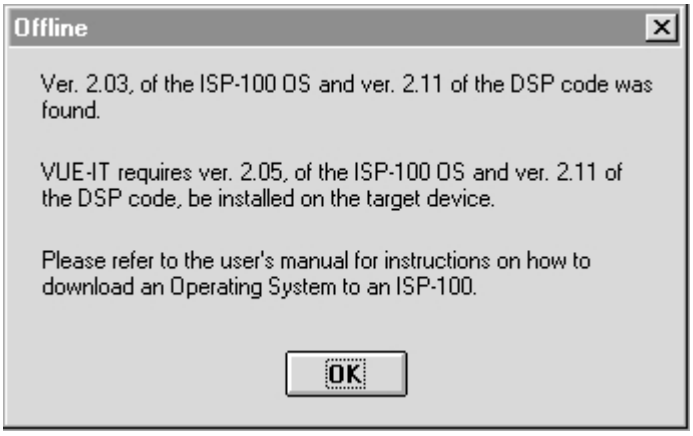


Figure 5-1 OS mismatch dialog.

Figure 5-2 OS download request dialog.

The updating of the operating system contained in the ISP-100's flash memory is usually required after loading a new version of VUE-IT on the computer. A dialog box such as those in Figures 5-1 and 5-2 usually indicates this.

The procedure is as follows:

1. Make sure both the computer and the ISP-100 are both turned off.

2. Connect the computer to the ISP-100.

3. Hold in the default switch located to the left of the power switch and turn on the ISP-100. Continue to hold in the default switch until the LEDs turn red and begin rotating. See Figure 2-1 item 6.

4. Turn on the computer and start the VUE-IT software. You should see a dialog box similar to that in Figure 5-2.

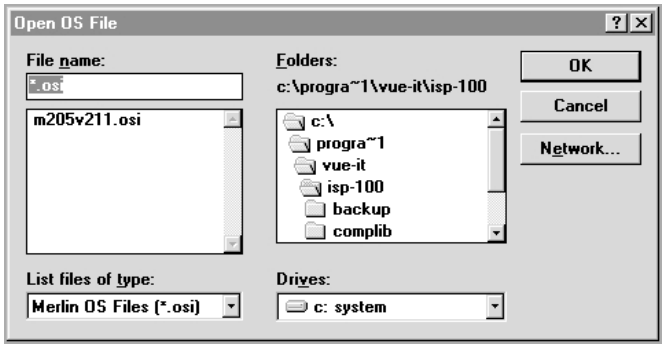


Figure 5-3 Open OS file dialog.

5. Download the OS software file you need via the application when prompted. See Figure 5-3 for more information. The default location for OS files is "**c:\program files\vue-it\isp-100**".

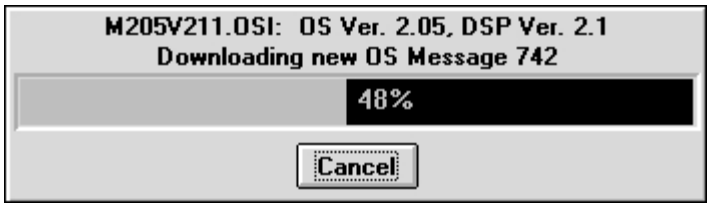


Figure 5-4 Download progress dialog.

6. The lights will continue to rotate with only brief pauses. You will also see a progress indicator on the PC's screen similar to the one in Figure 5-4.

7. When the download is complete, the ISP-100 will automatically restart.

Figure 5-5 ISP-100 detected dialog.

8. Wait for the HOST LED to light yellow and the rest of the LEDs to light green. At this point a window similar to the one in Figure 5-5 will appear. The ISP-100 is now ready for use.

Configuring the Software

Turn on the ISP-100. The power switch is located at the far right under the front access panel. This is accomplished by placing the switch in the down position. See Figure 2-1 for the location of the power switch.

Check to make sure that the LEDs light in a percentage complete fashion.

Launch the VUE-IT application on the PC.

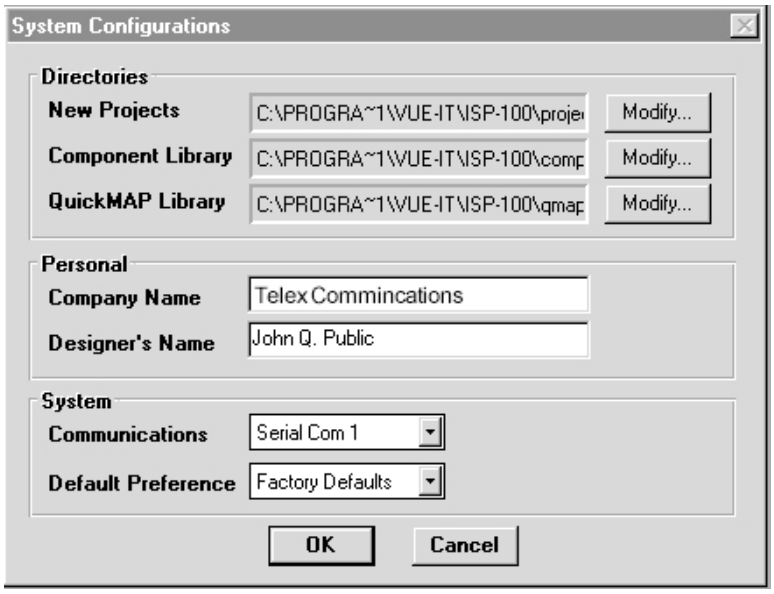


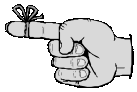
Figure 5-6 System configuration dialog.

A dialog box similar to the one in the Figure 5-6 should appear.

The System Configuration dialog box (see Figure 5-6) allows the user to select the serial port used, modify directories used, and enter information such as the company's and designer's names. It is recommended that you do not change the directory names unless you are an experienced user and know the implications of doing so. See the entry titled "VUE-IT File System" in section 6 for more information. After the proper information has been entered or selected, click on the **OK** button.

Figure 5-7 ISP-100 detected dialog.

After clicking on the **OK** button a dialog box like the one in the Figure 5-7 should appear. Click on the **OK** button. The ISP-100 and the PC are now ready for use together.



You will not be prompted again for this information. You can change it at any time by clicking on the **Option** menu and selecting **Configure**.

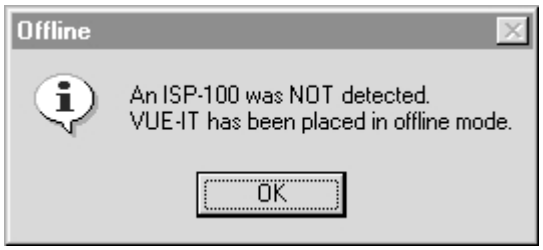


Figure 5-8 ISP-100 not detected dialog.

If a dialog box like the one in Figure 5-8 appears in place of the one in Figure 5-7, the cable connecting the computer to the ISP-100 should be checked. If no problem is found there, try selecting a different serial port.

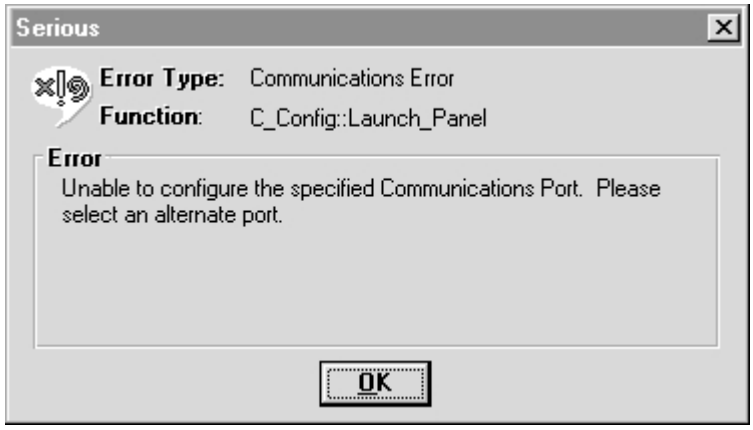


Figure 5-9 Communications error dialog.

If a dialog box similar to the one in the Figure 5-9 appears, then a serial port that does not exist has been selected. Another serial port must be selected.

You can change the serial port selection at any time by clicking on the **Option** menu and selecting **Configure**.

If you continue to have problems please contact Technical Support. See section 3 for contact information.

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GETTING STARTED

Starting VUE-IT

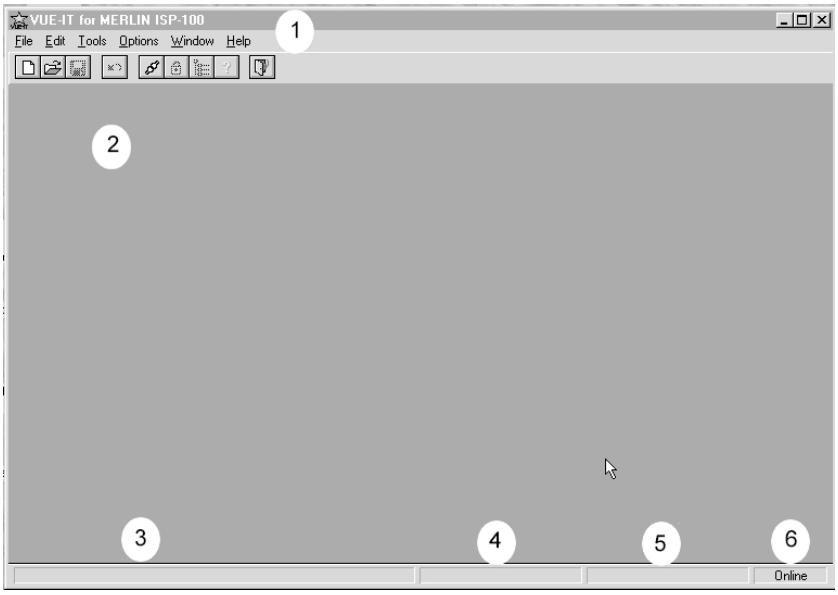


Figure 6-1 VUE-IT workspace.

When VUE-IT is started, you will see a screen similar to the one in Figure 6-1. The window displayed in Figure 6-1 is the basic environment in which all interactions with VUE-IT take place. There are six key features of this window that users should be aware of:

1. Menu Bar - This is where the user will find features that are either integral to VUE-IT or standard in windows. When you see a reference to a command located under a menu, this is where you should look. Below the menu names are short-cut buttons that provide access to common commands via a single mouse click. The buttons are (from left to right): New Project, Open Project, Save Project, Revert, Scan, Project Manager, Help, and Exit.

2. Work Space - This is where the QuickMAP, Project Manager, QuickSET Manager, and component boxes appear. This is the main work area for the user.

3. General Help - This box displays helpful messages about commands and functions.

4. Component/Parameter Type - This box displays the name of the component or parameter selected.

5. Component/Parameter Input Range - This box displays the input value range for a component or parameter.

6. Online/Offline Status - This box displays the status of the connection between the ISP-100 and the PC running VUE-IT.

Starting a New Project

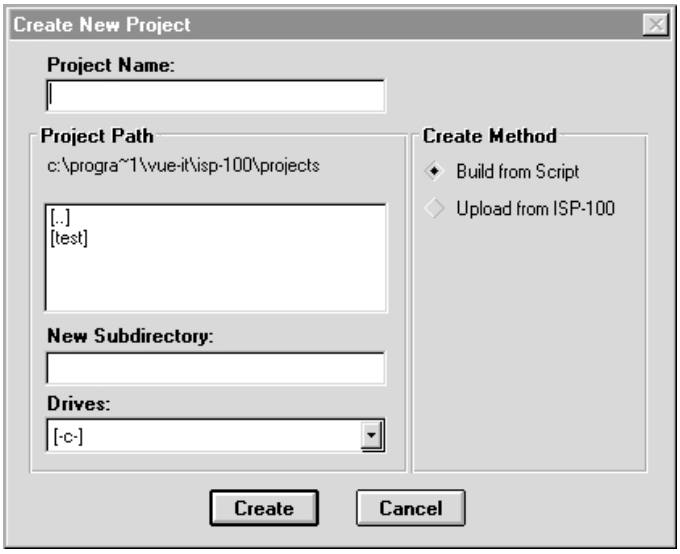


Figure 6-2 Create new project dialog.

To start a new project select **New Project** from the **File** menu. A panel similar to the one in Figure 6-2 will appear.

You are allowed to select a name of up to eight characters for the project, assign a path for the project to be stored, and create a new subdirectory if needed. You also will need to select the method by which the new project will be created. To select the method, click on the diamond to the left of either the **Build from Script** or **Upload from ISP-100** options. The **Build from Script** option uses files stored on your hard drive to create the QuickMAP. The **Upload from ISP-100** option creates the QuickMAP based on the information stored in an ISP-100 that is connected to the PC. When you have given the project a name, assigned a path or directory, and selected the creation method, click on the **Create** button to start the project creation process.

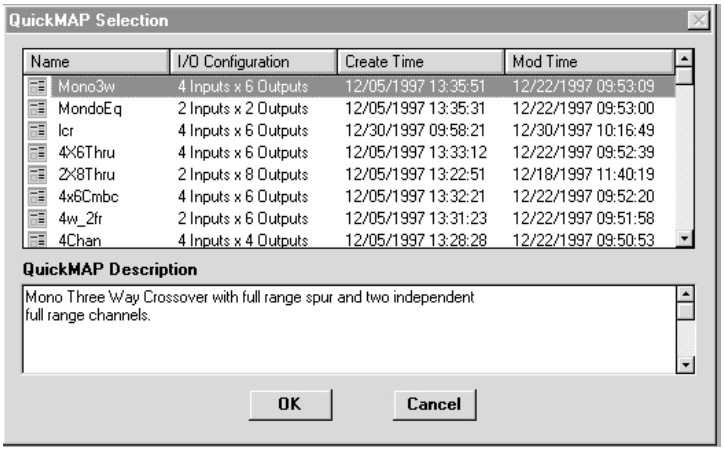


Figure 6-3 QuickMAP selection dialog.

If the **Build from Script** option is selected a dialog similar to the one in Figure 6-3 will appear. Here you will be able to select the script that will be

used to create the QuickMAP. The QuickMAPs can be sorted by Name, I/O Configuration, Creation Time, and Modification Time. To sort the QuickMAPs, click on the heading button of the property you wish to sort by (i.e. **Name**, **I/O Configuration**, etc.). The first click of the mouse will sort the QuickMAPs in ascending order and the second click will sort them in descending order.

If you select the **Upload from ISP-100** option there will be a pause while the QuickMAP in the ISP-100 is loaded. Using the **Upload from ISP-100** option is a good way to retrieve the settings of an ISP-100 when the original project file does not exist. Please note that this feature is NOT backward compatible. An ISP-100 with a project created using a previous version of VUE-IT will NOT be able to upload the project to your computer.

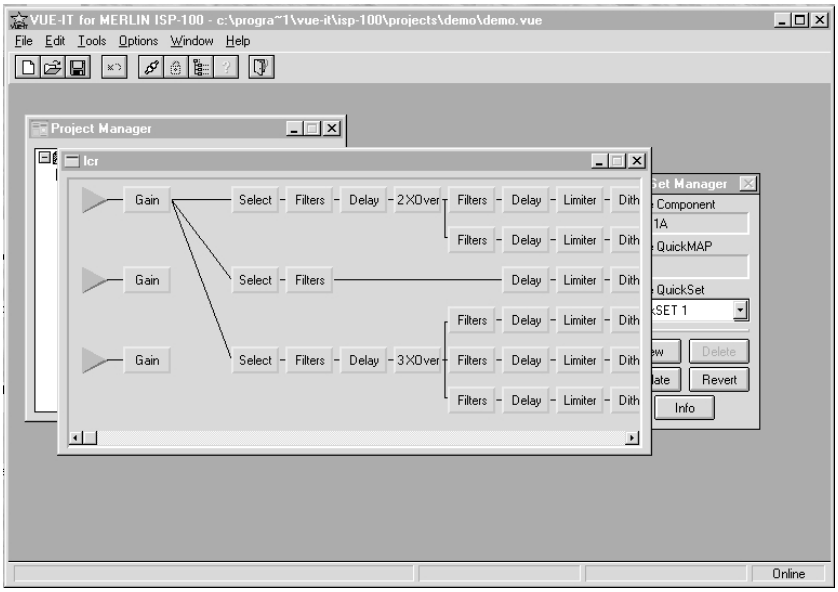


Figure 6-4 New project created.

CAUTION:
Trying to connect to an ISP-100 with a VUE-IT version different from the one used to create the project stored in it is not allowed. Do not update the OS. Updating the OS will erase the project stored in the ISP-100. Use the version of VUE-IT that created the project to Upload the project from the ISP-100 to the PC and save it. Then update the OS and restore the project..

After the QuickMAP has been created three panels will appear on the screen. They are: Project Manager, QuickSET Manager, and QuickMAP. See Figure 6-4.

VUE-IT File System

When a new project is created a number of files and directories are created. By default, the directory structure used by VUE-IT projects is as follows:

C:\PROGRAM FILES\VUE-IT\ISP-100\PROJECTS\<Directory Name>\<File Name>.<Extension>

The files created by a new project are stored in the project directory. The file names are the same as the name given the project when it was created. The file name formats are as follows:

Project File:	<File Name>.VUE
Project QuickMAP File:	<File Name>.QMS
Project QuickSET File:	<File Name>.QM0
Project GPI Settings:	<File Name>.GP0

Archiving Projects
To save a project for archival purposes do the following:

1. Exit VUE-IT.

2. Save the project directory and the four files mentioned previously to the archival media (i.e. floppy disk, zip™ disk, tape, CD, etc...). The directory name will usually be the name you assigned the project.

Mounting Projects
To install a project from an archive do the following:

1. Exit VUE-IT.

2. Create a directory under the C:\PROGRAMS\VUE-IT\ISP-100\PROJECTS\ directory.

3. Copy the four files mentioned previously above to the new project directory created in step 2.

4. Restart VUE-IT.

QuickMAP Template Files
QuickMAP template files are, by default, located under the following directory:

C:\PROGRAM FILES\VUE-IT\ISP-100\QMAPS\

The QuickMAP template file location is important to users who either have access to QuickMAP creation software or have been sent a new QuickMAP not contained in a standard VUE-IT installation.

Component Directory
Component files created by the Component Export command and used by the Component Import command are, by default, located under the following directory:

C:\PROGRAM FILES\VUE-IT\ISP-100\COMPLIB\

The files have the same name as the one assigned during export. The format of the component file name is as follows:

<File Name>.CMP

Operating System (OS) Files
OS files are loaded into the ISP-100's flash memory by VUE-IT. For the procedure on updating the ISP-100' s OS, see section 5. The OS files are, by default, located under the following directory:

C:\PROGRAM FILES\VUE-IT\ISP-100\

The format of the OS file name is as follows:

<File Name>.OSI

Using VUE-IT

VUE-IT has a variety of commands available to the user. These commands are accessible via the menu items along the top of the VUE-IT workspace window.

File (see Figure 7-1)

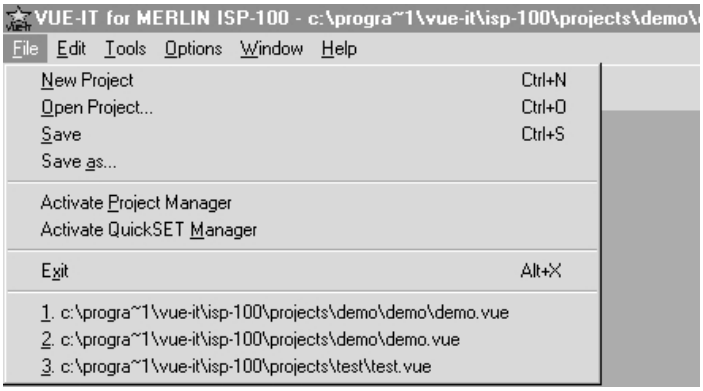


Figure 7-1 File menu.

New Project - Creates a new project file.

Open Project - Loads an existing project.

Save - Saves a project.

Save as... - Saves an existing project under another name.

Activate Project Manager - Opens the project manager window after it has been closed.

Activate QuickSET Manager - Opens the QuickSET Manager window after it has been closed. This function only works when one or more components are open.

Exit - Shuts the program down.

The last eight projects used are listed at the bottom of the menu if the **Most Recently Used files (MRU)** option is checked in the **Preferences** panel under the **Options Menu**. If the project no longer exists you are asked if you want to remove the item from the list.

Edit (see Figure 7-2)

Figure 7-2 Edit menu.

Cut - Removes the selected item(s) and places them on the clipboard.

Copy - Places a copy of the selected item(s) on the clipboard.

Paste - Inserts item(s) from the clipboard.

Tools (see Figure 7-3)

Figure 7-3 Tools menu.

Refresh Directories - Updates the directory list for QuickMAP and component library files.

Scan - Looks to see if an ISP-100 is connected to the computer.

Options (see Figure 7-4)

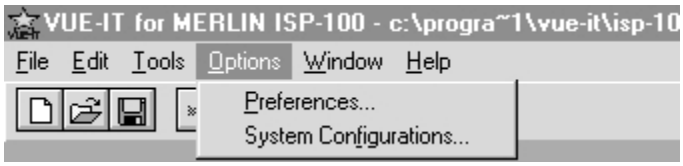


Figure 7-4 Options menu.

Preferences - Allows the user to setup preferences available within VUE-IT.

System Configurations - Allows the user to setup project directories, company name, designer's name, communications settings, etc.

Window (see Figure 7-5)

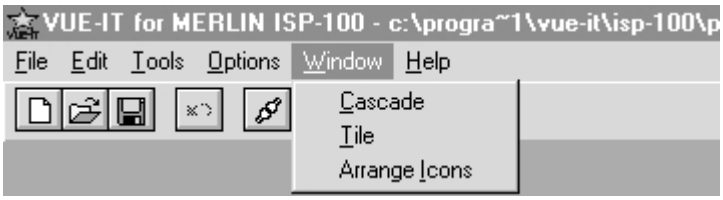


Figure 7-5 Windows menu.

Cascade - Arranges panels in overlapping layers.

Tile - Arranges panels side-by-side.

Arrange Icons - Arranges minimized panels side-by-side.

Help (see Figure 7-6)

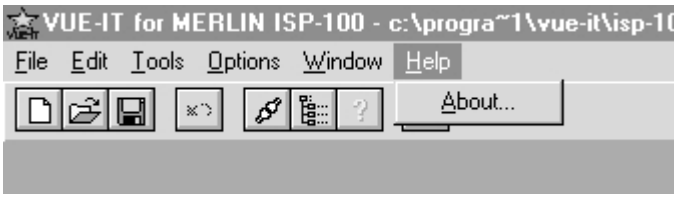


Figure 7-6 Help menu.

About - Lists the current application software version, OS version, and DSP version if an ISP-100 is connected and online. If an ISP-100 is not online, only the application software version will be reported.

Preferences (see Figure 7-7)

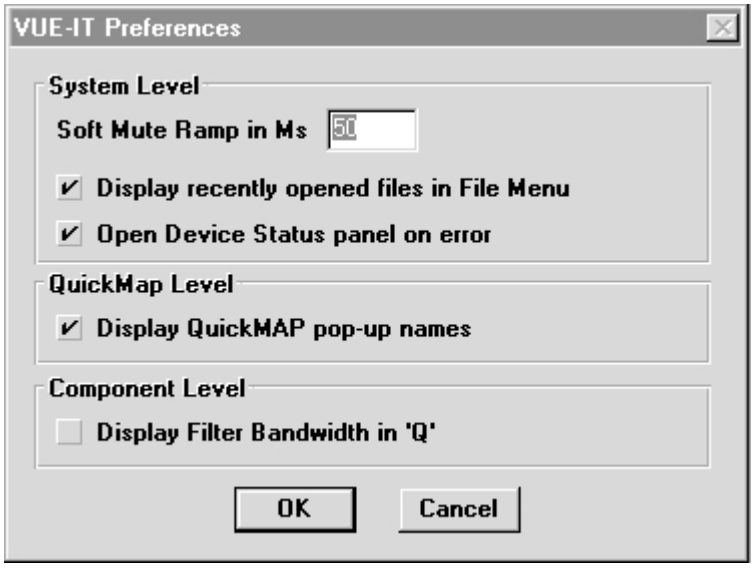


Figure 7-7 VUE-IT preferences dialog.

Soft Mute Ramp in Ms
This sets the amount (0 ms - 1000 ms) of ramp-up and ramp-down

there is when changing between QuickSETs or using a GPI mute. The total time for ramp-up and ramp-down is twice the time specified. (i.e. If 50 ms of time is specified, the ramp-down will be 50 ms and the ramp-up will be 50 ms. Thus, the total time between the start of the ramp-down to the end of the ramp-up is 100 ms.)

Display recently opened files in the File Menu
The check box turns on and off the Most Recently Used files (MRU) function. Removing the check mark will disable the function and clear the list.

Open Device Status panel on error
When a device status message is received, the status panel is automatically opened.

Display QuickMAP pop-up names
If selected the pop-up component nametags in the QuickMAP layout display are shown.

Display Filter Bandwidth in "Q"
If the box is checked filter bandwidth to be displayed in terms of Q. If it is unchecked bandwidth will be displayed in terms of octave fraction.

System Configurations (see Figure 7-8)

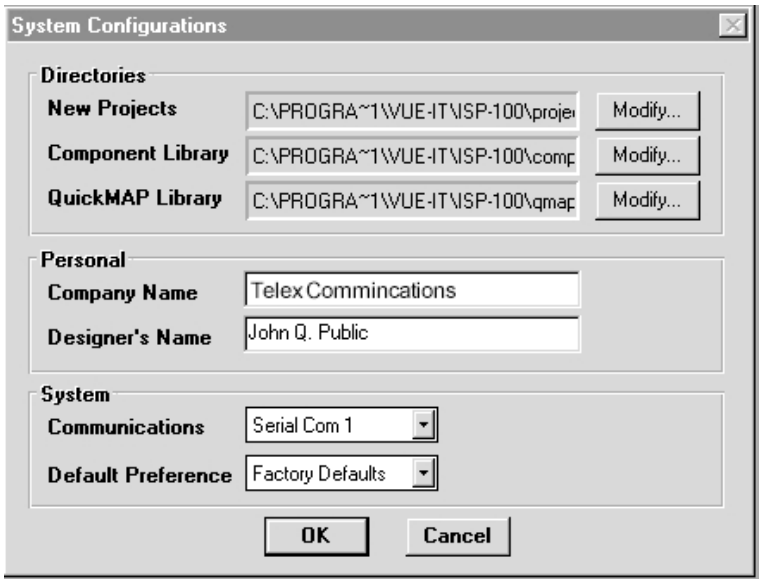


Figure 7-8 System configuration dialog.

New Projects
This is the directory under which new projects are stored. To modify the path, click on the **Modify...** button located to the right of the directory path entry.

Component Library
This is the directory under which the component presets are stored. To modify the path, click on the **Modify...** button located to the right of the directory path entry.

QuickMAP Library
This is the directory under which QuickMAP templates are stored. To modify the path, click on the **Modify...** button located to the right of the directory path entry.

Company Name
This entry is reserved for possible future use in documenting projects. You can enter the company name, but it currently will not be saved in a project.

Designer's Name
This entry is reserved for possible future use in documenting projects. You can enter your name here, but it currently will not be saved in a project.

Communications
This list allows you to select which COM port (1-4 and offline) on the PC will be used for communications with the ISP-100. Once a new COM port is selected you may need to select **Scan** from the **Tools** menu in order for the software to identify an ISP-100 connected to the new COM port.

Default Preference
This list is reserved for future program expansion. Via this list you would be able to change settings based on a designer's preferences. This would allow for multiple designers using the same program on the same PC.

Project Manager

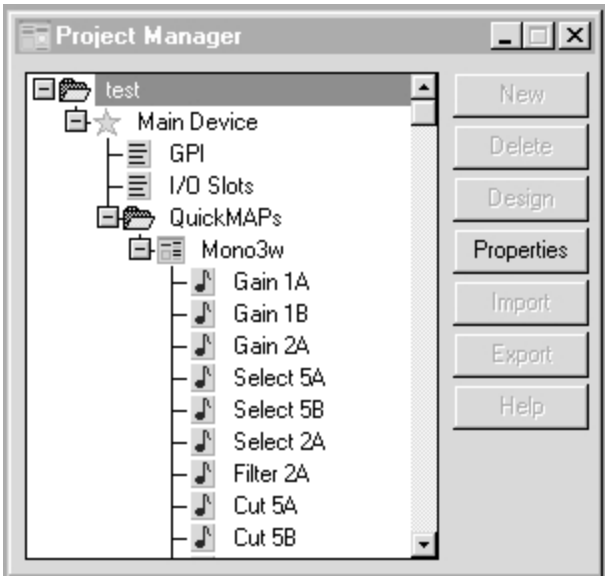


Figure 7-9 Project manager dialog.

The Project Manager (see Figure 7-9) contains information related to the project design. Via the manager, the user is able to define GPI inputs, configure inputs and outputs, import and export component settings, and adjust component settings. In addition to these functions, the user is also

able to access ISP-100 statistics such as free memory, number of QuickSETs, etc.

Properties for the ISP-100

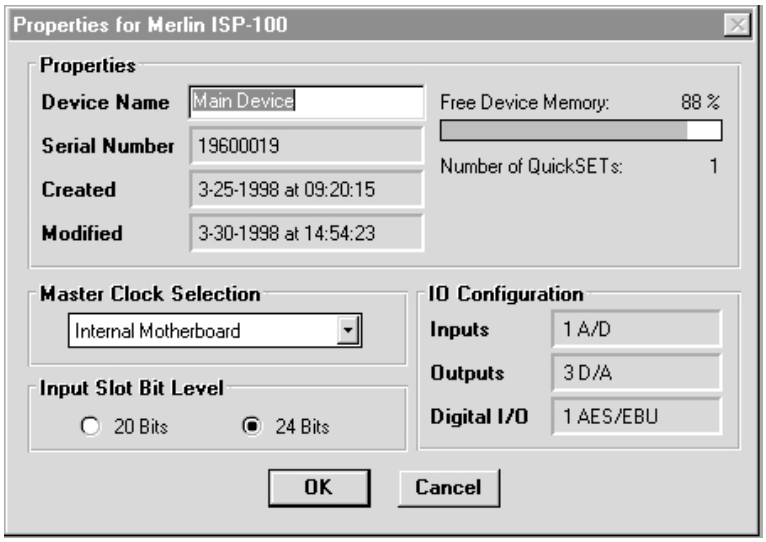


Figure 7-10 ISP-100 properties dialog.

Properties for the ISP-100 (see Figure 7-10) reports statistics on the ISP-100. It is accessed by double clicking on the entry next to the star icon (see Figure 7-9), or by clicking on the entry next to the star icon (see Figure 7-9) and then clicking on the **Properties** button in the Project Manager.

Master Clock Selection

There are four location selections for master clock:
1. Main Board – no reference frequency. (default).
2. Slot #1 – 12.288 MHz reference frequency (AES/EBU only).
3. Slot #2 – 12.288 MHz reference frequency (AES/EBU only).
4. External Reference – 12.288, 6.144, or 3.072 MHz via external input BNC.

An error will be returned if either slot 1 or slot 2 is selected as the clock master while the slot contains a card configured to use the sample rate converter (SRC).

The default clock master is the main board. If the user places an AES/EBU card in an input slot the user can manually change the clock master to select either the main, external sync, or input card slot.

Input Slot Bit Level

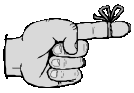
The input slot bit level can be set to either 20-bit or 24-bit. 20-bit is usually associated with consumer grade (SPDIF) digital audio. 24-bit is usually associated with professional grade (AES/EBU) digital audio. However, it is possible to have consumer grade 24-bit digital audio. For optimal performance be sure to set the proper bit level for the audio data stream.

Project Properties

Figure 7-11 Project properties dialog.

Project properties (see Figure 7-11) provides the user with information on the name of the project, project location, and notes related to the project. It is accessed by double clicking on the project name entry, or by clicking on the project name entry and then clicking on the **Properties** button in the Project Manager.

QuickMAP Selection



A QuickMAP is a signal processing topology used with an ISP-100. The QuickMAP sets the signal paths and processing from input to output. An ISP-100 can hold one QuickMAP at a time.

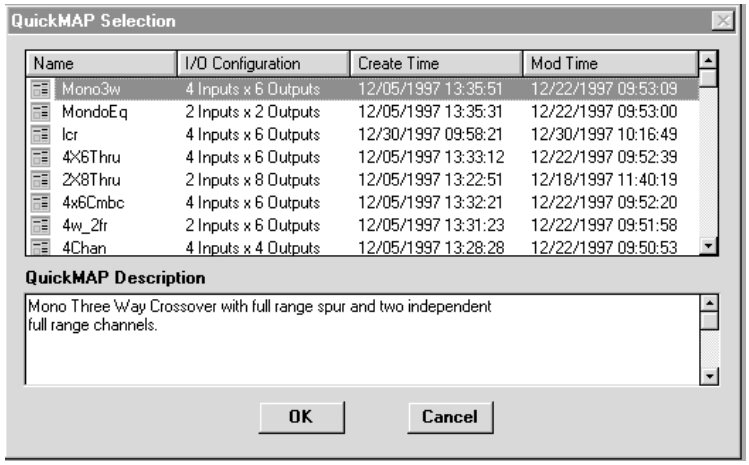


Figure 7-12 QuickMAP selection dialog.

QuickMAP Selection (see Figure 7-12) is where a project's QuickMAP is chosen. It appears when the **Build From Script** option is selected when setting up a new project. Here you will be able to select the script that will be used to create the QuickMAP. The QuickMAPs can be sorted by Name, I/O Configuration, Creation Time, and Modification Time. To sort the QuickMAPs, click on the heading button of the property you wish to sort by (i.e. **Name**, **I/O**

Configuration, etc.). The first click of the mouse will sort the QuickMAPs in ascending order and the second click will sort them in descending order. See section 6 for the location of the QuickMAP directory.

QuickMAP Properties

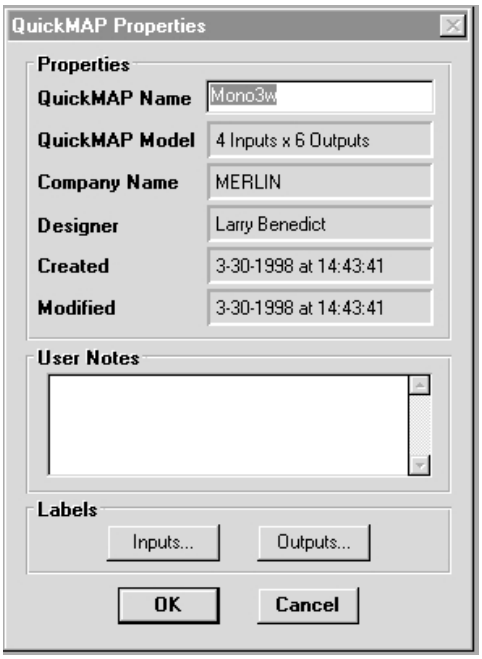


Figure 7-12 QuickMAP properties dialog.

QuickMAP Properties (see Figure 7-13) appear by clicking on the QuickMAP entry in the Project Manager and then clicking on the Properties button. From this dialog, you can change the QuickMAP name. You can also change the names of the input and output labels by clicking on either the **Inputs** or **Outputs** buttons.

Component Properties

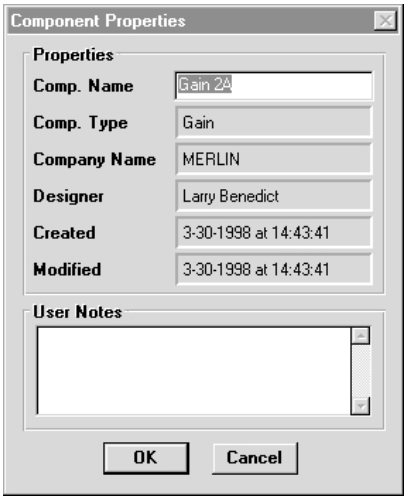


Figure 7-14 Component properties dialog.

Clicking on a component (i.e. **Cut**, **Compressor**, **Limiter**, etc.) in the Project Manager (see Figure 7-9) and then clicking on the **Properties** button displays the properties for the selected component. Component Properties allow the user to name the component and enter notes related to that component. See Figure 7-14.

Component Export

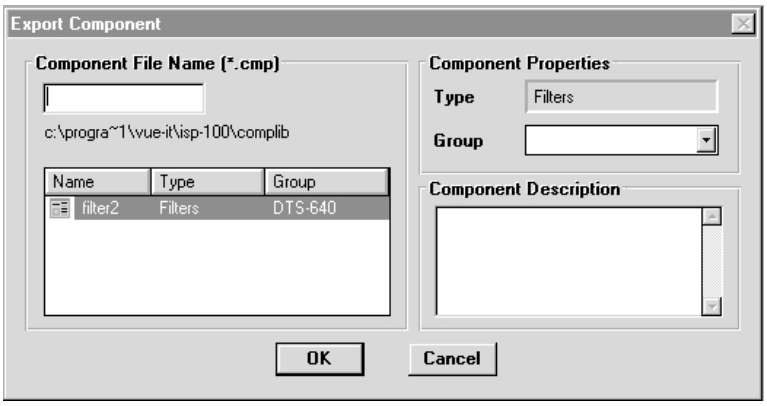


Figure 7-15 Component export dialog.

To export component settings from the Project Manager, click on the desired component entry in the Project Manager (see figure 7-9) and then click on the **Export** button. You will then see a dialog box like the one in Figure 7-15.

The Export Component dialog displays currently existing component presets' filenames in the lower left corner of the dialog. These files can be sorted by clicking on the **Name**, **Type**, and **Group** heading buttons. The first mouse click on one of these buttons will sort the list in ascending order and the second click will sort the list in descending order.

The Export Component dialog requires you to supply a name for both the component settings and the group to which it is assigned. In addition to these entries, you are also allowed to provide a brief description of the settings. Once the component name, group, and description have been entered, click on the **OK** button. Clicking on the **OK** button causes the program to create a file holding the component settings and description in the COMPLIB directory. The full path to the COMPLIB directory is assigned when you first installed VUE-IT. See the System Configurations entry for information on changing the COMPLIB directory's path.

Component Import

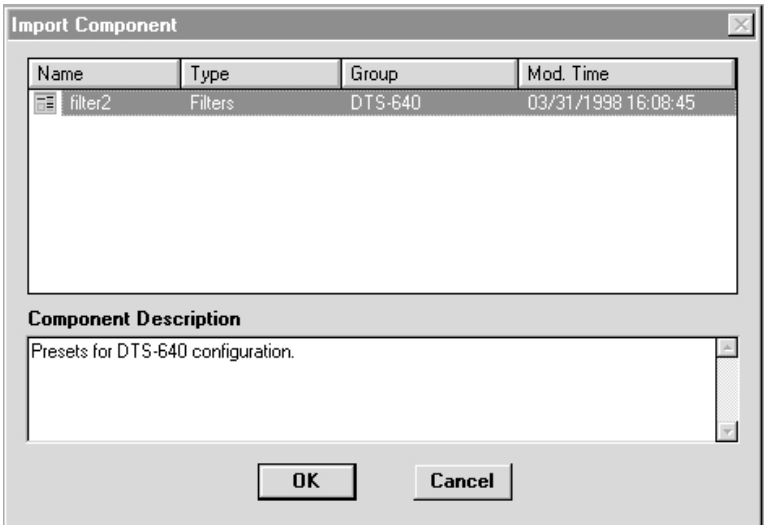


Figure 7-16 Component import dialog.

To import component settings from the Project Manager, click on the desired component entry in the project manager (see Figure 7-9) and then click on the **Import** button. You will then see a dialog box like the one in Figure 7-16.

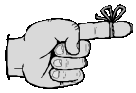
The Import Component dialog displays currently existing component presets' filenames in the upper half of the dialog box. These files can be sorted by clicking on the **Name**, **Type**, **Group**, and **Mod Time** heading buttons. The first mouse click on one of these buttons will sort the list in ascending order and the second click will sort the list in descending order.

The lower half of the dialog box displays notes on the component preset currently selected in the upper half of the dialog box.

QuickSET Manager

Figure 7-17 QuickSET manager dialog.

The QuickSET Manager (see Figure 7-17) allows the user to develop and manage multiple groups of component settings. The QuickSET Manager is opened when a component is opened.



A QuickSET is made up of the settings of all of the components in a QuickMAP. Multiple QuickSETs allow users to change the component settings for a QuickMAP. QuickSETs can be changed without a computer by assiging a QuickSET to a GPI input. When that input receives a contact closure, the QuickMAP will change its component settings to those contained in the QuickSET assigned to the GPI pin.

Creating New QuickSETs
The user can name a group of settings by clicking on the **New** button and then entering the name in the pop-up window and clicking on the **OK** button.

Deleting QuickSETs
If at least two QuickSETs exist, the user can delete an inactive one by clicking on the **Delete** button, selecting the QuickSET from the list, and then clicking on the **OK** button.

Selecting the Active QuickSET
The user can select a QuickSET from the list of available QuickSETs by clicking on the arrow next to the Active QuickSET entry and selecting the desired QuickSET from the list.

Updating a QuickSET
The user can use the **Update** button to save changes to a QuickSET that has already been created. Using Update overwrites the information previously saved in the selected QuickSET.

Discarding Changes
The Revert button allows the user to make changes to a QuickSET, and then go back to the original settings. This button only works when changes have been made and the **Update** button has not been used.

GPI

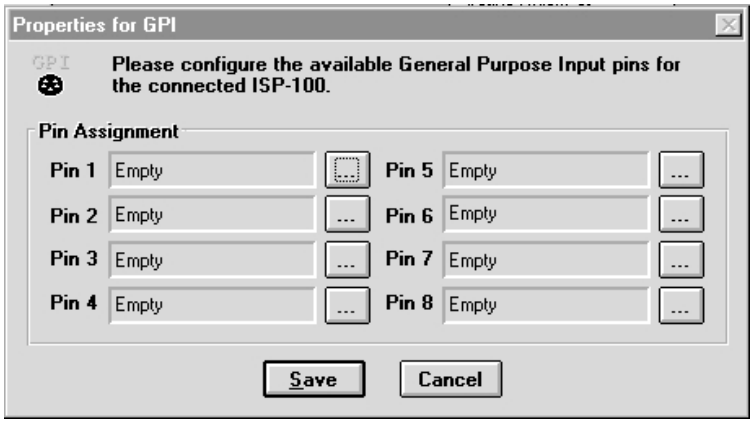


Figure 7-18 GPI properties dialog.

Clicking on the **GPI** entry in the Project Manager and then clicking on the **Properties** button will open the General Purpose Interface (see Figure 7-18). From this panel you can configure the GPI input pins.

Configuring Input Pins
Each input pin can be assigned a command by clicking on the button to the right of the entry for that pin (see Figure 7-18). A pop-up menu

will appear, and the proper command can be selected from the list by clicking on it. The list displays the following options:

Empty - No command assigned to input pin.

Mute Combo - The initial contact closure causes the outputs to mute. The subsequent contact closure causes the output to un-mute.

Mute Enable - Contact closure causes the outputs to mute.

Mute Disable - Contact closure causes the outputs to un-mute.

QuickSET Increment - Contact closure causes the next QuickSET stored in memory to be loaded into the QuickMAP.

QuickSET Decrement - Contact closure causes the previous QuickSET stored in memory to be loaded into the QuickMAP.

QuickSET Select - Contact closure causes the specified QuickSET to be loaded into the QuickMAP.

Testing the GPI Assignments

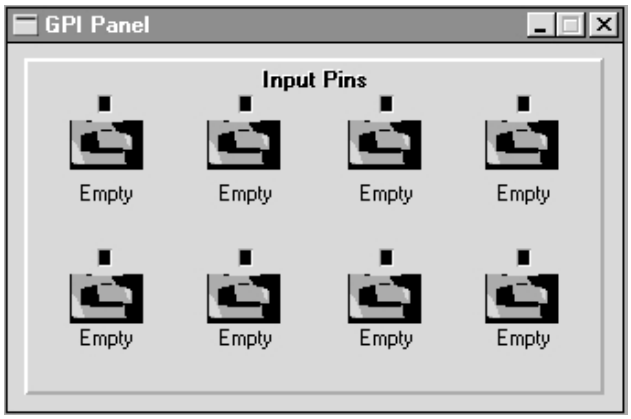
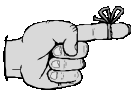


Figure 7-19 GPI panel.

If you click on the **GPI** entry and then click on the **Design** button, you will see a panel like the one in Figure 7-19. The panel in Figure 7-19 will allow you to simulate GPI contact closures without having to hookup external contact closures to the GPI inputs of the ISP-100.

QuickMAP



A QuickMAP is a signal processing topology used with an ISP-100. The QuickMAP sets the signal paths and processing from input to output. An ISP-100 can hold one QuickMAP at a time.

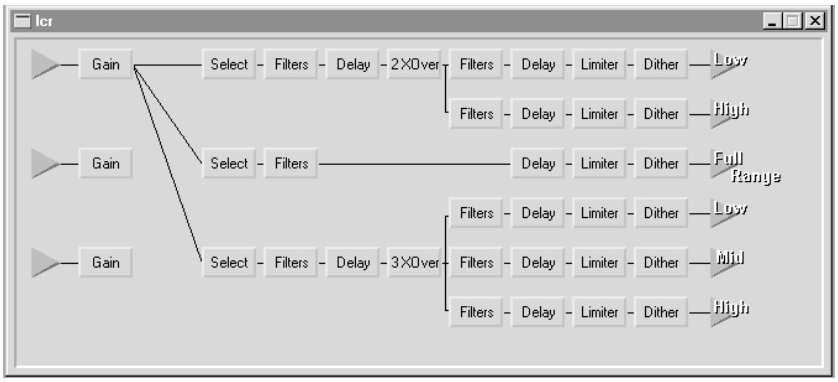


Figure 7-20 QuickMAP panel.

Double clicking on the QuickMAP (name) entry or by clicking on the entry and then clicking on the **Design** button can access a graphical representation of the selected QuickMAP (see Figure 7-20).

Clicking on the green input triangles in the QuickMAP will open the I/O Properties dialog (see Figure 7-21).

Clicking on the green output triangles will open the Output Meters (see Figure 7-24).

Clicking on the component boxes will open the component. Placing the pointer on a component, holding down the left mouse button, and dragging the component onto a component of the same type and releasing the mouse button will copy the component settings from the component being dragged to the component it is dropped on.

QuickMAP Component Bypassing

Right clicking on the component boxes will produce a pop-up menu. The user is able to modify component settings (this is the same as left clicking on the component), access its properties, import or export settings, and in selected cases bypass the component. A fully bypassed component will appear red in the QuickMAP. Please note that this only occurs when the main bypass is activated for that component.

I/O Properties (see Figure 7-21)

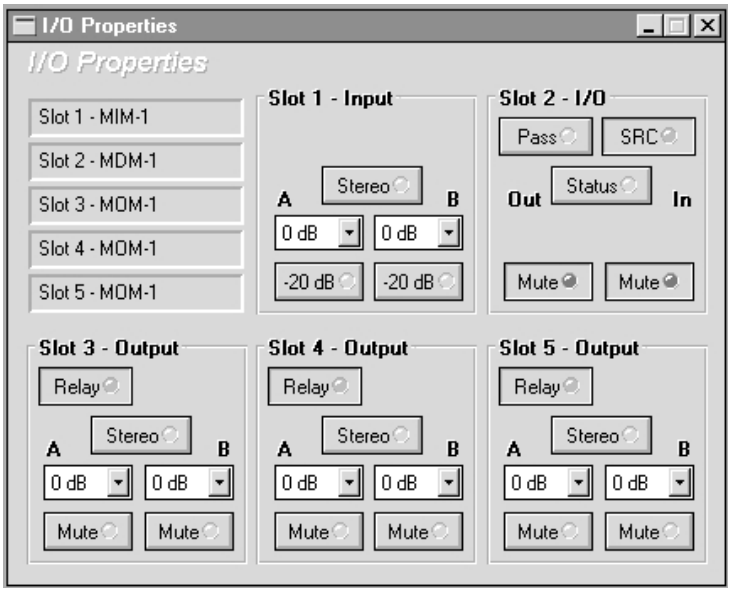


Figure 7-21 I/O properties dialog.

Input
The Input Function controls the levels entering the unit. This function can be accessed in any of the following three ways:

1. Clicking on the green input triangles.

2. Right clicking on either the green input or green output triangles and selecting **Properties** from the pop-up list.

3. Clicking on the I/O Slots entry in the Project Manager and then clicking on the **Properties** button.

Using the Stereo Control
This button controls the tracking of the Gain/Attenuator levels. If the LED is gray, then the two channels do not track each other. If the LED is blue, then the channels track each other. Clicking on the button will toggle the setting.

Gain
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. Click on the desired entry to make a selection.

Using the Pad Control
This button controls the insertion of a -20dB pad in the signal path. A gray LED on the button indicates the pad is not inserted. A red LED on the button indicates that the pad is inserted. Clicking on the button will toggle the setting.

Output
The Output function controls the output levels from the unit.

Using the Stereo Control
This button controls the tracking of the Gain/Attenuator levels. If the LED is gray, then the two channels do not track each other. If the LED is blue, then the channels will track each other. Clicking on the button will toggle the setting.

Mute Control
This button determines if the output signal will be muted. A gray LED indicates that the signal is not muted, while a red LED indicates that the signal is muted. Clicking on the button will toggle the setting.

Using the Relay Control
This button activates and deactivates the output relays. A gray LED indicates that the audio path is not complete. A green LED indicates that the audio path is complete. Clicking on the button will toggle the setting.

Gain
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. Click on the desired entry to make a selection.

Digital I/O

In Mute (RX)
Mutes the AES receiver data stream.

Out Mute (TX)
Mutes the AES transmitter data stream

Pass Through
Routes the input from the receiver directly through the transmitter to the card's output (the output data from the DSP is bypassed). Either the In-Mute or Out-Mute will mute the output when Pass Through is on. Data may still be received when Pass Through is on. Pass Through is only valid with a 48 kHz bit stream, or with a non-48 kHz bit stream and Sample Rate conversion enabled. A non-48 kHz sample rate with no sample rate conversion will result in an error when the Pass Through option is selected.

SRC
Clicking on the **SRC** button turns on the Sample-Rate-Converter. The SRC is used to convert non-48 kHz data streams into 48 kHz data streams compatible with the ISP-100.

Note: Turning on the SRC will result in a 20-bit output data stream except in the case of Pass Through.

Status Information

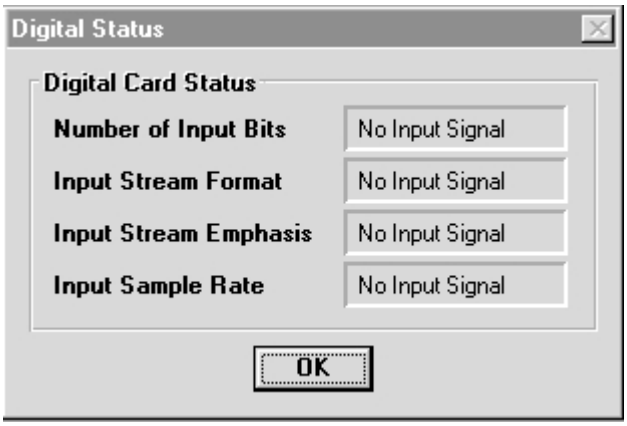


Figure 7-22 Digital I/O status panel.

Status information can be obtained for card slots 1 and 2 by clicking on the **Status** button in the desired slot. See Figure 7-22.

Sample Rate Converter
On indicates that the sample rate conversion option is selected on the card. The card cannot be selected as the clock master if the SRC is turned on.

Off indicates that the sample rate conversion option is not selected on the card.

Input Stream Format
Pro indicates that the digital receiver has detected an AES-EBU formatted data stream.

Consumer indicates that the digital receiver has detected an SPDIF formatted data stream.

Input Stream Emphasis
Emphasis indicates that the sending device has added emphasis to the incoming data stream.

None indicates that the sending device has not emphasized the incoming data stream.

Input Sample Rate
Indicates the sample rate of the data stream in kHz. There are three possible frequencies **48.0 kHz**, **44.1 kHz**, and **32.0 kHz**. All other frequencies above and below these are listed as **Out of Range**.

Note: The emphasis information in the output data stream will be set to no emphasis, regardless of the input emphasis status.

Device Status Panel

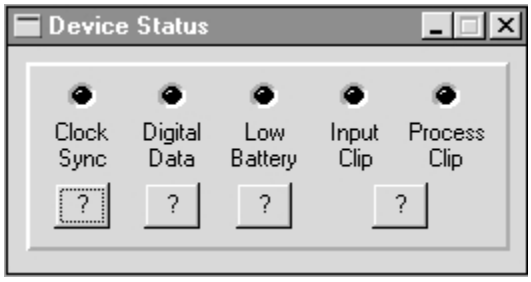


Figure 7-23 Device status panel.

Clicking on the entry next to the star icon (see Figure 7-9) and then clicking on the **Design** button in the Project Manager opens the Device Status Panel. See Figure 7-23.

The device status panel has five simulated LEDs:

Clock Sync (red)
The main board cannot lock to specified location or frequency.

Digital Data (red)
1. Data validity error. Card position dependent.

2. A non-48 kHz Sample Rate is being used, and no Sample Rate Converter is selected. Card position dependent.

3. Non-Audio Input Stream. Card position dependent.

4. Slot Cannot be chosen as master and have sample rate converter present. Card position dependent.

5. An input slot must have an AES card to be master. Card position dependent.

Low Battery (yellow)
Please check battery.

Input Clip (red)
Input audio has clipped before the DSPs.

Process Clip (red)
Which slot and channel caused the clip.

The specifics of an error condition can be found by clicking on the **?** button. You are also able to clear the error indication. If the error condition still exists or reoccurs, the error LED will turn on again.

Output Meters

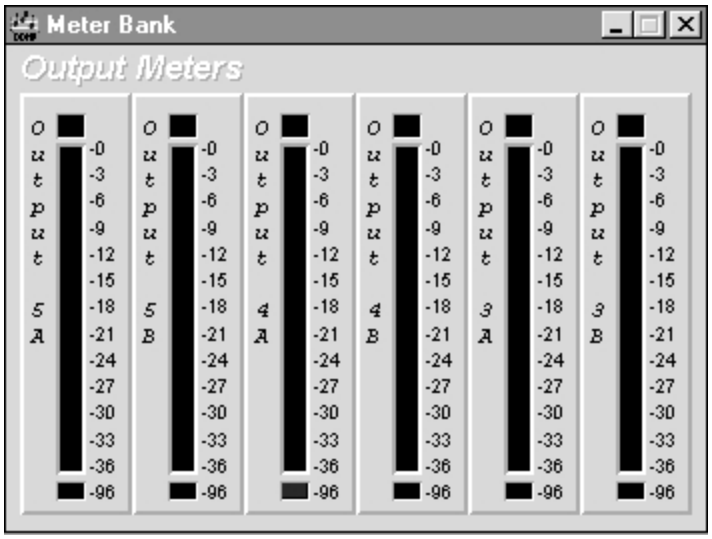


Figure 7-24 Output meter panel.

The Output Meters (see Figure 7-24) provide monitoring of the outputs of the DSPs before digital-to-analog conversion. To activate the Meters, click on the green output triangles in the QuickMAP, or click on the Meter Bank entry in the Project Manager and then click on the **Design** button.

PROCESSING COMPONENTS

The components used in the VUE-IT interface have a wide variety of controls for adjusting settings.

Each component description begins with a full explanation of the component including all of the parameters that can be adjusted. In some areas the explanations may go into more depth than the average user would need. This was done in order to satisfy the advanced user who feels a need to understand the inner workings of the component. References to specific journal articles and other supplementary sources are provided for further reading.

An explanation of how to manipulate the various controls follows the component explanation. In many instances a specific parameter may have multiple means of adjustment. An example of this would be the gain control in the Gain component. A user accustomed to traditional hardware would probably prefer the virtual faders and knobs, whereas a user accustomed to computers and exact entries would probably prefer the text box where the amount of gain or attenuation can be directly entered.

Combine

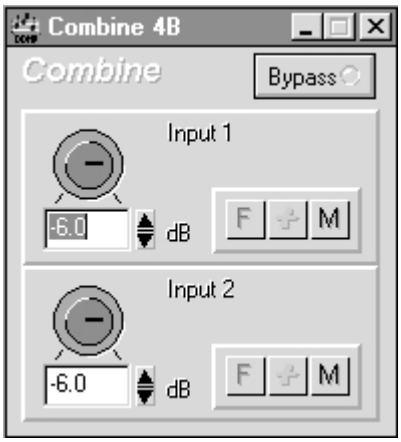


Figure 8-1 2-to-1 combine component.

The Combine component (see Figure 8-1) adds multiple input signals together and sends the sum to a single output. In QuickMAPs, 2-to-1, 3-to-1, and 4-to-1 combines may exist. The combine's default attenuation is set so that the combine's output level is unity. In every case the attenuation applied to each input channel may be independently adjusted between -96 dB and 0 dB, or muted, and the polarity applied to each input channel may be independently set to + or - (in-phase or out-of-phase). Attenuation and polarity are applied to the input signals before they are added to the combination.

Bypassing a combine sets all attenuations to 0 dB and all polarities to + (in-phase).

Input Attenuators

The input attenuators can be adjusted three ways:

Using the Knob

Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box

Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows

To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Polarity Control (+/-)
This button determines the polarity of the signal. A + indicates that the signal will not have it's phase altered. A - indicates that the signal's phase will be changed 180°. Clicking on the button will toggle the setting.

Mute Control (M)
This button determines if the input signal will be muted. A black M indicates that the signal is not muted, while a red M indicates that the signal is muted. Clicking on the button will toggle the setting.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Compressor

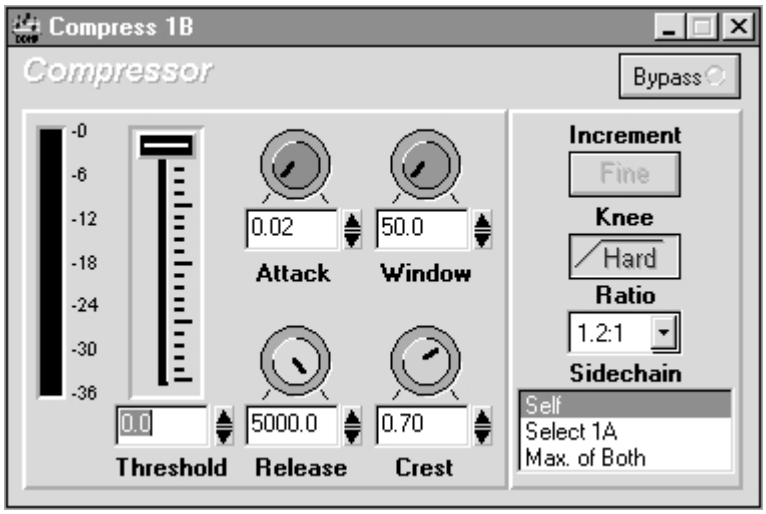


Figure 8-2 Compressor component.

A compressor (see Figure 8-2) is used in situations where the input signal has a greater dynamic range, or ratio of loudest to softest, than is desired in the output signal. Compression is accomplished by preventing the output signal from changing in level as much as the input signal, and can be thought of as a form of automatic gain riding.

Familiar Controls

Most of the controls in the ISP-100 Compressor are common to nearly all traditional dynamic range compressors. These controls should be familiar to anyone with experience in the operation of any compressor.

Compression Ratio

The Compression Ratio control adjusts the amount by which the output signal level changes for a given change in input signal level. When compressing, the output level change in dB will be equal to the input level change in dB divided by the compression ratio:

$$\text{output level change} = \frac{\text{input level change}}{\text{compression ratio}}$$

For example, if the compression ratio is 2:1 and the input level changes by +10 dB, then:

$$\text{output level change} = \frac{\text{input level change}}{\text{compression ratio}} = \frac{+10\ \text{dB}}{2} = +5\ \text{dB}$$

Any compressor is essentially a limiter once the compression ratio is increased beyond approximately 6:1.

Compression Ratio is selectable from {1.2:1, 1.5:1, 2:1, 3:1, 4:1, 6:1, 8:1, 12:1, 16:1, 24:1}, though any compressor is essentially a limiter once the compression ratio is increased beyond approximately 6:1.

Threshold

The Threshold control adjusts the input signal level above which compression is used. When the input signal level is below the

threshold value, no compression takes place and the change in level of the output signal exactly matches the change in level of the input signal. When the input signal level is above the threshold value, compression is used and the change in level of the output signal is less than the change in level of the input signal by the amount dictated by the selected compression ratio.

Threshold is adjustable in the range -60 dBFS to 0 dBFS.

Attack Time
A compressor monitors the level of the input signal and controls the level of the output signal accordingly. When the input signal level suddenly increases, a compressor allows the output signal level to also increase, but only as much as is appropriate for the selected compression ratio setting. Desirable sound effects can sometimes be created if a compressor is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Attack Time control adjusts the duration of this ramp. Attack Time is adjustable in the range 20 µSec (one sample) to 50 mSec.

Release Time
A compressor monitors the level of the input signal and controls the level of the output signal accordingly. When the input signal level suddenly decreases, a compressor allows the output signal level to also decrease, but only as much as is appropriate for the selected compression ratio setting. A compressor generally sounds better if it is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Release Time control adjusts the duration of this ramp. Release Time is adjustable in the range 20 µSec (one sample) to 5 mSec.

Knee Selection
Below the threshold level, no compression is taking place and any changes in input level are matched exactly by changes in output level. Above the threshold level, compression is taking place and changes in output level are less than the corresponding changes in input level. The transition from no compression to compression can be set to take place abruptly or gradually.

Under hard knee conditions the output level tracks the input level perfectly as it rises up to the threshold. Above the threshold the output level continues to rise as dictated by the compression ratio. There is an abrupt transition from no compression below the threshold to compression above.

Under soft knee conditions, there is a smooth, gradual transition from no compression below the threshold to compression above.

Under soft knee conditions the output level and the input level track until approximately 12-18 dB below the threshold, beyond which they slowly begin to diverge. Depending upon the compression ratio selected, the output level may be as much as

2.5 dB below the input level at the threshold. The divergence continues to increase gradually until approximately 6 dB above the threshold, beyond which the output level rise is dictated solely by the selected compression ratio. There is a smooth, gradual transition from no compression below the threshold to compression above.

The sonic differences between the two knee types are often subtle and are very much signal-dependent. Experimentation is recommended.

Sidechain Source

There are situations in which compression is applied to one channel based upon the level in another channel. This might be used in a stereo signal pair to prevent image shift; one of the channels would be considered to be the level reference and the compressors in both channels would be set up identically. Another example might be ducking, where the compression applied to background music in one channel would be controlled by the level of the announcer's voice in another.

The Sidechain Source channel selection indicates the channel whose level is to be used to control the compressor. If the input channel is to control itself, then it should be selected as the Sidechain Source. Otherwise the appropriate other channel should be selected from the set of available Sidechain Sources in the menu.

Unfamiliar Controls

There are two controls on the ISP-100 compressor that are not available in traditional dynamic range compressors. These controls provide much greater flexibility and utility to this compressor than can be achieved with traditional compressors.

Detection Window

The detection window is the period of time over which the compressor computes the average signal level. Every dynamic range compressor that operates upon average signal level uses a detection window. In traditional compressors this window is of fixed length, and the operator may not even be aware of that length.

Some traditional compressors provide switching between average and peak detection. This is actually just a selection between a relatively long detection window of generally a few tens or hundreds of milliseconds, and a very short detection window of generally a few tens or hundreds of microseconds.

The ISP-100 compressor provides for operator control of the length of the detection window. Very dynamic signals and signals containing a lot of high frequency energy require shorter detection windows. Non-dynamic signals and signals containing mostly low frequency energy require longer detection windows. Detection Window is adjustable in the range 20 μSec (one sample) to 5 Sec.

Crest Factor Sensitivity

A signal's crest factor is the ratio of a its peak level to its average level. The signal having the lowest possible crest factor is the square wave, whose peak and average levels are equal and whose crest factor is therefore 1.0. The peak level of a pure sinewave tone is 1.414 times its average level; a sinewave therefore has a crest factor of 1.414. Most music and speech signals contain peaks that are very much higher than their average levels; therefore their crest factors are very much greater than one. The important points to remember are that a signal's peak value is always greater than or equal to its average value, and that the crest factor of any signal is therefore always greater than or equal to 1.0.

The Crest Factor Sensitivity adjustment on the ISP-100 compressor function controls its sensitivity to the crest factor of the input signal. It works like this:

The signal level detection algorithm used in the ISP-100 compressor consists of two separate detectors operating in parallel. One of the detectors computes the average level over the selected detection window; the other computes the peak level of each input sample.

The peak level is multiplied by the setting of the Crest Factor Sensitivity control, which is always a number between zero and one. The two quantities are then compared and the larger is returned as the signal level.

When the Crest Factor Sensitivity is set to 0.00 the ISP-100 compressor operates as a traditional compressor responding to average signal levels; when the Crest Factor Sensitivity is set to 1.00 the ISP-100 compressor operates as a traditional compressor responding to peak signal levels. Values in-between represent various hybrids of the two. If the compressor is to ignore peaks and respond only to average levels, then the Crest Factor Sensitivity may be set near zero. If the compressor is to respond only to peaks, then the Crest Factor Sensitivity may be set near one.

Good results may be obtained by following these steps:

1. Set the Crest Factor Sensitivity to 0.00. This will cause the ISP-100 compressor to behave like a traditional compressor responding to average signal levels.

2. Set all of the other compressor controls as appropriate for the signal being compressed, just as they would be set in a traditional compressor.

3. Adjust the Crest Factor Sensitivity control so that the compressor responds appropriately to short signal peaks. If it were possible to know, in advance, what the crest factor of the

input signal was going to be, then the Crest Factor Sensitivity control would ideally be set to:

$$\frac{1}{\text{Crest Factor of Input Signal}}$$

Since this quantity is not generally known, it is best to experiment until satisfactory results are obtained.

Metering

The compressor meter shows the attenuation applied to the signal passing through the compressor at any given moment in time. When the signal is being compressed the meter indicates negative values (gain less than 1.0). When the signal is not being compressed the meter indicates 0 dB (gain of 1.0). Meter values are raw, no averaging or ballistics are applied.

Notes on Operation of the Compressor

Bypass

Bypassing a compressor causes the signal to pass through unaltered.

Overshoot

The compressor algorithm used in the ISP-100 allows for a small amount of overshoot on transients, regardless of the settings of the Detection Window and Crest Factor Sensitivity controls. The amount of overshoot is somewhat dependent upon the setting of the Threshold control; the lower the threshold the greater the overshoot. This represents normal operation for the compressor.

Computation of Average Level

There are two common methods of computing average signal level; one is based upon RMS (Root Mean Square) calculations, and the other is based upon Absolute Value calculations. Each of these methods produces a slightly different value for signal level, and the difference depends upon the characteristics of the signal itself. The ISP-100 compressor uses Absolute Value for its computation of signal level, so the level reported by the compressor may differ slightly from the RMS value expected for a given signal.

Crest Factor Sensitivity and Sinewaves

The peak level of a pure sinewave tone is 1.414 times its average level. The ISP-100 compressor will therefore apply approximately 3 dB more compression to a pure sinewave signal when the Crest Factor Sensitivity is set to 1.00 (peak-responding) than when it is set to 0.00 (average-responding). This represents normal operation for the compressor.

Sidechaining

When the sidechain input to an ISP-100 compressor is selected to be a channel other than its input, the compressor controls must still be set. The sidechain selects only the channel's data,

not its compressor control settings. If multiple compressors utilize the same channel as their sidechain sources, then all controls on all compressors must still be configured.

Meter
The meter located at the left of the component indicates the amount of attenuation taking place in dB.

Threshold
The Threshold can be adjusted three ways:

Using the Slider
Click on the indicator on the slider and while continuing to hold down the mouse button, move the mouse up and down. The numbers in the box below the slider will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Attack, Window, Release, and Crest factor controls.
All of these controls can be adjusted in three ways:

Using the Knob
Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Knee Control
This button determines the type of transition that occurs when the component goes from inactive to active. A Hard knee causes the response by the component to be abrupt, while a Soft knee causes a gentle response. Clicking on the button will toggle the setting.

Ratio Control
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Sidechain
To make a selection click on the desired entry.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Crossover

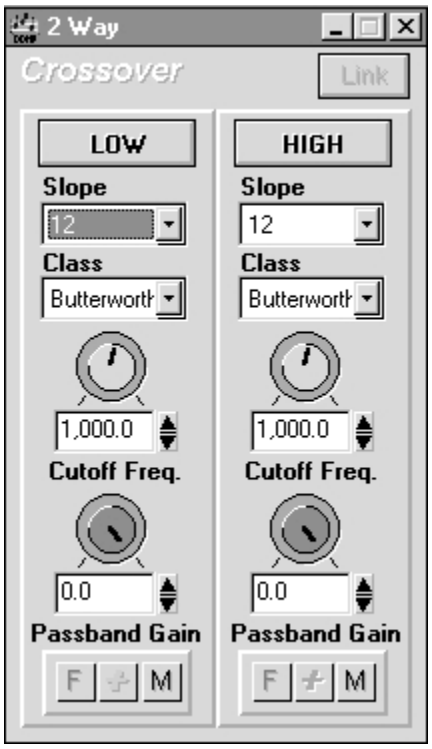


Figure 8-3 Two-way crossover component.

Custom crossovers other than those available from the crossover component can be constructed by using the filter component.

Crossovers (see Figure 8-3) are used to divide the audio signal into multiple bands appropriate for different types of loudspeakers or loudspeaker elements. In QuickMAPs, 2-way, 3-way, and 4-way crossovers may exist. Each of these crossovers may be implemented using Bessel or Butterworth filters with 1st- through 4th-order slopes, or Linkwitz-Riley filters with 2nd- or 4th-order slopes, or combinations thereof.

Two-Way Crossovers

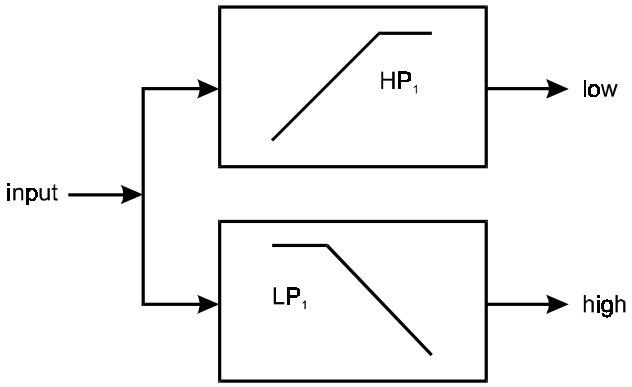


Figure 8-4 Two-way crossover implementation.

In the ISP-100, 2-way crossovers are implemented as shown in Figure 8-4.

The lowpass filter LP_1 and the highpass filter HP_1 may each be independently configured as:

Bessel 6 dB/octave (1st-order)
Bessel 12 dB/octave (2nd-order)
Bessel 18 dB/octave (3rd-order)
Bessel 24 dB/octave (4th-order)

Butterworth 6 dB/octave (1st-order)
Butterworth 12 dB/octave (2nd-order)
Butterworth 18 dB/octave (3rd-order)
Butterworth 24 dB/octave (4th-order)

Linkwitz-Riley (Butterworth-Squared) 12 dB/octave (2nd-order)
Linkwitz-Riley (Butterworth-Squared) 24 dB/octave (4th-order)

In addition, the cutoff frequency, attenuation, and polarity in the low band and the high band may be independently adjusted, and the filters in either band may be bypassed entirely.

The cutoff frequency in each filter is continuously variable from 20 Hz to 20 kHz.

The attenuation in each band is continuously variable from -96 dB to 0 dB, plus mute.

The polarity in each band is selectable as + or - (in-phase or out-of-phase).

Each band may be bypassed, in which case all of the crossover filters in that band will be bypassed while the crossover filters in the other band will not be affected.

Two-Way Bessel and Butterworth Crossovers

Bessel crossovers utilize Bessel filters. Bessel filters are designed for maximally-flat group delay, meaning that they have nearly linear phase and excellent transient response. Their amplitude response is not as flat and their cutoff is not as sharp as those of Butterworth filters.

Butterworth crossovers utilize Butterworth filters. Butterworth filters are designed for maximally-flat amplitude response, meaning that the amplitude response in the passband is as flat as possible without ripple. Their phase response is not nearly as linear and their transient response is not as good as those of Bessel filters.

The Bessel filters used in the ISP-100 crossovers are automatically scaled so that their amplitude response is -3 dB at the selected cutoff frequency, regardless of the filter order.

The amplitude response of a Butterworth filter is naturally -3 dB at the selected cutoff frequency, regardless of the filter order.

Two-way Linkwitz-Riley Crossovers

Linkwitz-Riley designs constitute a special, even-order in-phase class of two-way sum-to-allpass response crossovers, and they must be configured in accordance with very strict phase and polarity requirements.

A properly implemented 2-way, 12 dB/octave Linkwitz-Riley crossover is characterized by:

- 2nd-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 2nd-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of highpass band opposite that of lowpass band

A properly implemented 2-way, 24 dB/octave Linkwitz-Riley crossover is characterized by:

- 4th-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 4th-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of highpass band the same as that of lowpass band

Summarizing: In a 2-way, 12 dB/octave Linkwitz-Riley crossover the high band must be out-of-phase with the low band; in a 2-way, 24 dB/octave Linkwitz-Riley crossover the high band must be in-phase with the low band.

Three-Way Crossovers

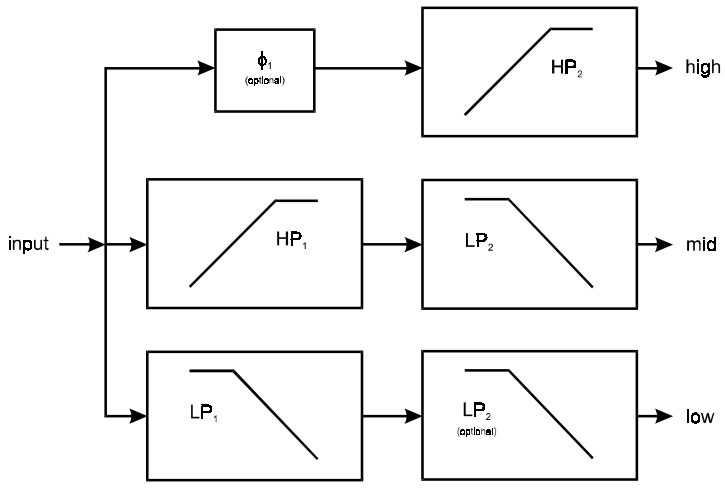


Figure 8-5 Three-way crossover implementation.

In the ISP-100, 3-way crossovers are implemented as shown in Figure 8-5.

This parallel structure was selected over the more common binary-tree structure for its suitability to the distribution of filters among multiple DSPs. The parallel structure is also capable of perfect

implementation of Linkwitz-Riley crossovers, for which the ISP-100 has been optimized. The filters marked optional in Figure 8-5 will be explained in the Linkwitz-Riley Crossover section, below.

The lowpass filters LP_x (except for the optional LP_2) and the highpass filters HP_x may each be independently configured as:

Bessel 6 dB/octave (1st-order)
Bessel 12 dB/octave (2nd-order)
Bessel 18 dB/octave (3rd-order)
Bessel 24 dB/octave (4th-order)

Butterworth 6 dB/octave (1st-order)
Butterworth 12 dB/octave (2nd-order)
Butterworth 18 dB/octave (3rd-order)
Butterworth 24 dB/octave (4th-order)

Linkwitz-Riley (Butterworth-Squared) 12 dB/octave (2nd-order)
Linkwitz-Riley (Butterworth-Squared) 24 dB/octave (4th-order)

In addition, the cutoff frequency, attenuation, and polarity in the low band, the mid band, and the high band may be independently adjusted, and all of the filters in any band may be bypassed entirely.

The cutoff frequency in each filter is continuously variable from 20 Hz to 20 kHz.

The attenuation in each band is continuously variable from -96 dB to 0 dB, plus mute.

The polarity in each band is selectable as + or - (in-phase or out-of-phase).

Each band may be bypassed, in which case all of the crossover filters in that band will be bypassed while the crossover filters in the other bands will not be affected.

Three-way Bessel and Butterworth Crossovers

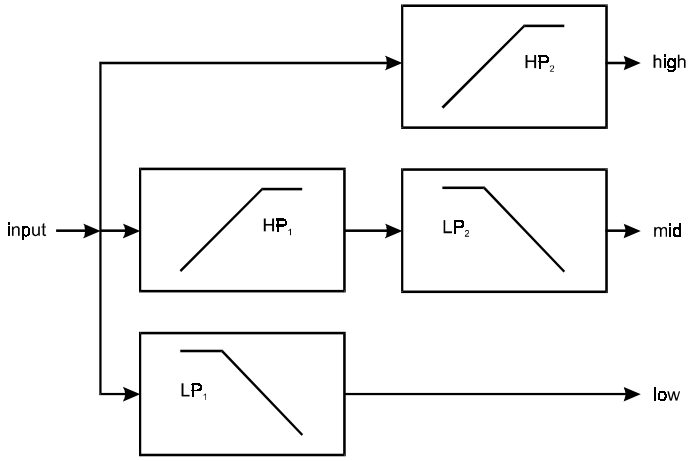


Figure 8-6 Three-way Bessel & butterworth crossover implementation.

In the ISP-100, 3-way Bessel and Butterworth crossovers are implemented as shown in Figure 8-6.

Bessel crossovers utilize Bessel filters. Bessel filters are designed for maximally-flat group delay, meaning that they have nearly linear phase and excellent transient response. Their amplitude response is not as flat and their cutoff is not as sharp as those of Butterworth filters.

Butterworth crossovers utilize Butterworth filters. Butterworth filters are designed for maximally-flat amplitude response, meaning that the amplitude response in the passband is as flat as possible without ripple. Their phase response is not nearly as linear and their transient response is not as good as those of Bessel filters.

The Bessel filters used in the ISP-100 crossovers are automatically scaled so that their amplitude response is -3 dB at the selected cutoff frequency, regardless of the filter order.

The amplitude response of a Butterworth filter is naturally -3 dB at the selected cutoff frequency, regardless of the filter order.

Three-way Linkwitz-Riley Crossovers

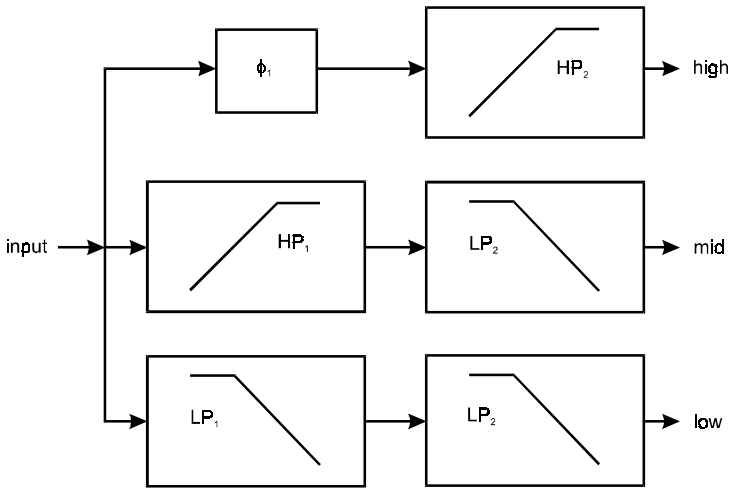


Figure 8-7 Three-way Linkwitz-Riley crossover implementation.

In the ISP-100, 3-way Linkwitz-Riley crossovers are implemented as shown in Figure 8-7.

Linkwitz-Riley designs constitute a special, even-order in-phase class of two-way sum-to-allpass response crossovers, and they must be configured in accordance with very strict phase and polarity requirements. Because the basic Linkwitz-Riley crossover is a 2-way design, extra steps must be taken in 3-way Linkwitz-Riley crossovers to ensure that these phase and polarity requirements are met.

Note the presence of the unexpected θ_1 allpass filter in the high

band and LP_2 lowpass filter in the low band. The inclusion of these filters is mandatory for proper implementation of the 3-way Linkwitz-Riley crossover; they ensure that the phase shift and magnitude response in each band match those of the other bands to maintain the fundamental Linkwitz-Riley sum-to-allpass quality. A detailed explanation of the use of compensation filters such as these may be found in “Active Realization of Multiway All-Pass Crossover Systems” by Joseph A. D’Appolito; *Journal of the Audio Engineering Society*, Volume 35, Number 4, April 1987.

A 3-way Linkwitz-Riley crossover is essentially a combination of two 2-way Linkwitz-Riley crossovers. This means that the polarity requirements associated with a 2-way Linkwitz-Riley crossover carry over to a 3-way crossover.

A properly implemented 2-way, 12 dB/octave Linkwitz-Riley crossover is characterized by:

- 2^{nd}-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 2^{nd}-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of highpass band opposite that of lowpass band

In a 3-way crossover the bands associate in non-obvious ways (see Sum-to-Allpass Characteristics of Linkwitz-Riley Crossovers), and the polarity reference for the high band is unexpectedly the low band, rather than the adjacent mid band. Thus, if a 12 dB/octave filter is used for LP_1 and HP_1, then the mid band must be out-of-phase with the low band. If a 12 dB/octave filter is used for LP_2 and HP_2, then the high band must be out-of-phase with the low band, not the mid band.

A properly implemented 2-way 24 dB/octave Linkwitz-Riley crossover is characterized by:

- 4^{th}-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 4^{th}-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of high band the same as that of low band

In a 3-way crossover the bands associate in non-obvious ways (see Sum-to-Allpass Characteristics of Linkwitz-riley Crossovers), and the polarity reference for the high band is unexpectedly the low band, rather than the adjacent mid band. Thus, if a 24 dB/octave filter is used for LP_1 and HP_1, then the mid band must be in-phase with the low band. If a 24 dB/octave filter is used for LP_2 and HP_2, then the high band must be the in-phase with the low band, not the mid band.

Four-Way Crossovers

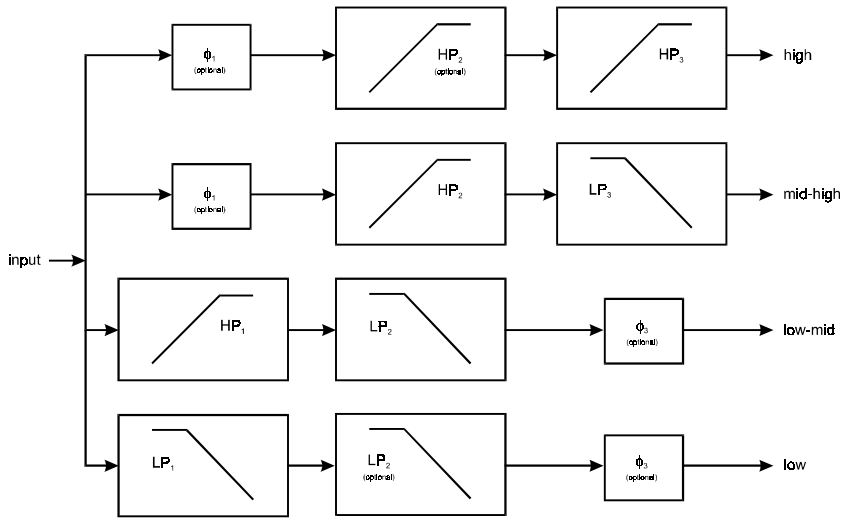


Figure 8-8 Four-way crossover implementation.

In the ISP-100, 4-way crossovers are implemented as shown in Figure 8-8.

This parallel structure was selected over the more common binary-tree structure for its suitability to the distribution of filters among multiple DSPs. The parallel structure is also capable of perfect implementation of Linkwitz-Riley crossovers, for which the ISP-100 has been optimized. The filters marked optional in Figure 8-8 will be explained in the Linkwitz-Riley Crossover section.

The lowpass filters LP_x (except for the optional LP_2) and the highpass filters HP_x (except for the optional HP_2) may each be independently configured as:

Bessel 6 dB/octave (1st-order)
Bessel 12 dB/octave (2nd-order)
Bessel 18 dB/octave (3rd-order)
Bessel 24 dB/octave (4th-order)

Butterworth 6 dB/octave (1st-order)
Butterworth 12 dB/octave (2nd-order)
Butterworth 18 dB/octave (3rd-order)
Butterworth 24 dB/octave (4th-order)

Linkwitz-Riley (Butterworth-Squared) 12 dB/octave (2nd-order)
Linkwitz-Riley (Butterworth-Squared) 24 dB/octave (4th-order)

In addition, the cutoff frequency, attenuation, and polarity in the low band, the low-mid band, the mid-high band, and the high band may be independently adjusted, and all of the filters in any band may be bypassed entirely.

The cutoff frequency in each filter is continuously variable from 20 Hz to 20 kHz.

The attenuation in each band is continuously variable from -96 dB and 0 dB, plus mute.

The polarity in each band is selectable as + or - (in-phase or out-of-phase).

Each band may be bypassed, in which case all of the crossover filters in that band will be bypassed while the crossover filters in the other bands will not be affected.

Four-way Bessel and Butterworth Crossovers

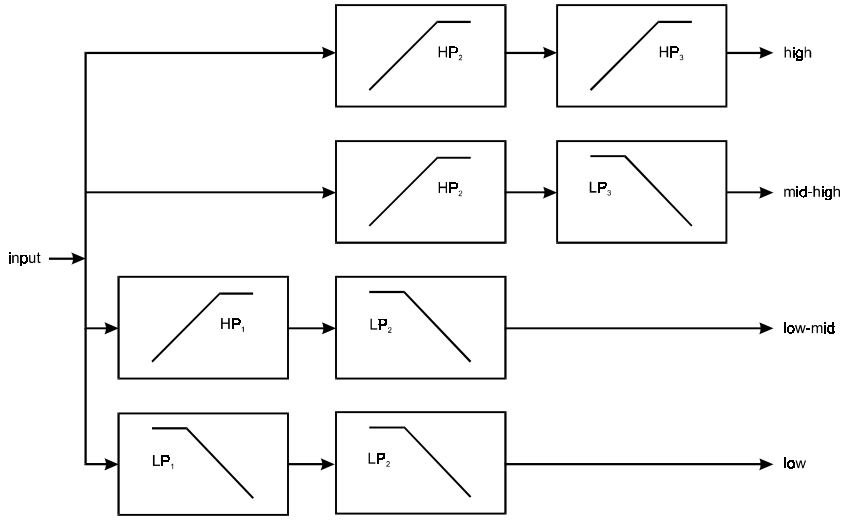


Figure 8-9 Four-way Bessel & Butterworth crossover implementation.

In the ISP-100, 4-way Bessel and Butterworth crossovers are implemented as shown in Figure 8-9.

Bessel crossovers utilize Bessel filters. Bessel filters are designed for maximally-flat group delay, meaning that they have nearly linear phase and excellent transient response. Their amplitude response is not as flat and their cutoff is not as sharp as those of Butterworth filters.

Butterworth crossovers utilize Butterworth filters. Butterworth filters are designed for maximally-flat amplitude response, meaning that the amplitude response in the passband is as flat as possible without ripple. Their phase response is not nearly as linear and their transient response is not as good as those of Bessel filters.

The Bessel filters used in the ISP-100 crossovers are automatically scaled so that their amplitude response is -3 dB at the selected cutoff frequency, regardless of the filter order.

The amplitude response of a Butterworth filter is naturally -3 dB at the selected cutoff frequency, regardless of the filter order.

Four-way Linkwitz-Riley Crossovers

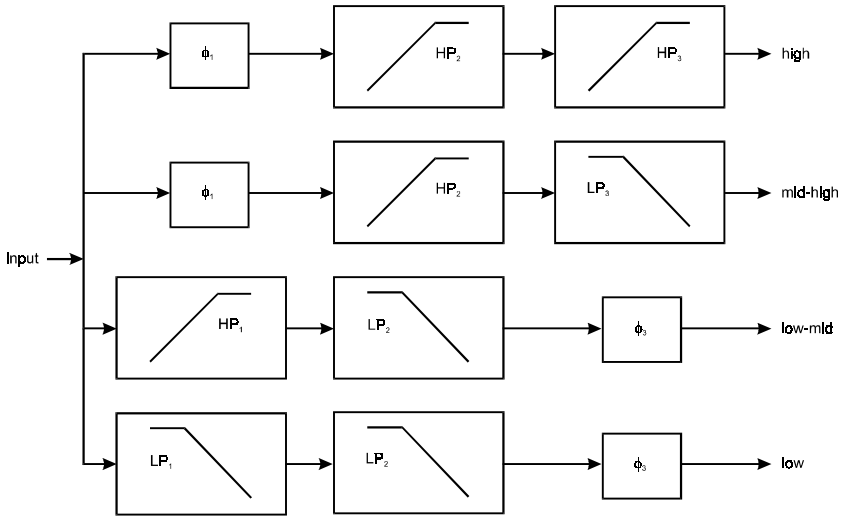


Figure 8-10 Four-way Linkwitz-Riley crossover implementation.

In the ISP-100, 4-way Linkwitz-Riley crossovers are implemented as shown in Figure 8-10.

Linkwitz-Riley designs constitute a special, even-order in-phase class of 2-way sum-to-allpass response crossovers, and they must be configured in accordance with very strict phase and polarity requirements. Because the basic Linkwitz-Riley crossover is a 2-way design, extra steps must be taken in 4-way Linkwitz-Riley crossovers to ensure that these phase and polarity requirements are met.

Note the presence of the unexpected θ_1 allpass filter and HP_2 highpass filter in the high band, the θ_1 allpass filter in the mid-high band, the θ_3 allpass filter in the low-mid band, and the LP_2 lowpass filter and θ_3 allpass filter in the low band. The inclusion of these filters is mandatory for proper implementation of the 4-way Linkwitz-Riley crossover; they ensure that the phase shift and magnitude response in each band match those of the other bands to maintain the fundamental Linkwitz-Riley sum-to-allpass quality. A detailed explanation of the use of compensation filters such as these may be found in “Active Realization of Multiway All-Pass Crossover Systems” by Joseph A. D’Appolito; *Journal of the Audio Engineering Society*, Volume 35, Number 4, April 1987.

A 4-way Linkwitz-Riley crossover is essentially a combination of three 2-way Linkwitz-Riley crossovers. This means that the polarity requirements associated with a 2-way Linkwitz-Riley crossover carry over to a 4-way crossover.

A properly implemented 2-way, 12 dB/octave Linkwitz-Riley crossover is characterized by:

- 2^{nd}-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 2^{nd}-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of highpass band opposite that of lowpass band

In a 4-way crossover the bands associate in non-obvious ways (see Sum-to-Allpass Characteristics of Linkwitz-Riley Crossovers). As expected, the polarity reference for the low-mid band is the low band, and the polarity reference for the high band is the mid-high band. However, the polarity reference for the mid-high band is unexpectedly the low band, rather than the adjacent low-mid band. Thus, if a 12 dB/octave filter is used for LP_1 and HP_1, then the low-mid band must be out-of-phase with the low band. If a 12 dB/octave filter is used for LP_3 and HP_3, then the high band must be out-of-phase with the mid-high band. And if a 12 dB/octave filter is used for LP_2 and HP_2, then the mid-high band must be out-of-phase with the low band, not the low-mid band.

A properly implemented 2-way 24 dB/octave Linkwitz-Riley crossover is characterized by:

- 4^{th}-order Linkwitz-Riley (Butterworth-Squared) lowpass filter, -6 dB magnitude response at crossover frequency
- 4^{th}-order Linkwitz-Riley (Butterworth-Squared) highpass filter, -6 dB magnitude response at crossover frequency
- polarity of high band the same as that of low band

In a 4-way crossover the bands associate in non-obvious ways (see Sum-to-Allpass Characteristics of Linkwitz-riley Crossovers). As expected, the polarity reference for the low-mid band is the low band, and the polarity reference for the high band is the mid-high band. However, the polarity reference for the mid-high band is unexpectedly the low band, rather than the adjacent low-mid band. Thus, if a 24 dB/octave filter is used for LP_1 and HP_1, then the low-mid band must be in-phase with the low band. If a 24 dB/octave filter is used for LP_3 and HP_3, then the high band must be the in-phase with the mid high band. And if a 24 dB/octave filter is used for LP_2 and HP_2, then the mid high band must be the in-phase with the low band, not the low-mid band.

Sum-to-Allpass Characteristics of Linkwitz-Riley Crossovers

A characteristic of 2-way Linkwitz-Riley crossovers is that the output bands sum to an allpass filter—the frequency response of the sum has a constant magnitude of one and only phase shift occurs. From Figure 8-4, this means that

$$LP_x - HP_x = \theta_x$$

or

$$LP_x + HP_x = \theta_x$$

depending upon whether 12 dB/octave or 24 dB/octave filters are used, respectively.

The following sections will show this sum-to-allpass trait to hold true for the ISP-100 Linkwitz-Riley 3-way and 4-way crossover implementations.

Three-way
From Figure 8-6

$$\begin{aligned}\text{Summed Response} &= LP_1LP_2 \pm HP_1LP_2 \pm \theta_1HP_2\\ &= (LP_1 \pm HP_1)LP_2 \pm \theta_1HP_2\\ &= \theta_1LP_2 \pm \theta_1HP_2\\ &= \theta_1(LP_2 \pm HP_2)\\ &= \theta_1\theta_2 \text{ Q.E.D.}\end{aligned}$$

There is an alternate configuration, not implemented in the ISP-100, that also has this property:

$$\begin{aligned}\text{Summed Response} &= LP_1\theta_2 \pm HP_1LP_2 \pm HP_1HP_2\\ &= LP_1\theta_2 \pm HP_1(LP_2 \pm HP_2)\\ &= LP_1\theta_2 \pm HP_1\theta_2\\ &= (LP_1 \pm HP_1)\theta_2\\ &= \theta_1\theta_2 \text{ Q.E.D.}\end{aligned}$$

Four-way
From Figure 8-9

$$\begin{aligned}\text{Summed Response} &= LP_1LP_2\theta_3 \pm HP_1LP_2\theta_3 \pm \theta_1HP_2LP_3 \pm \theta_1HP_2HP_3\\ &= (LP_1 \pm HP_1)LP_2\theta_3 \pm \theta_1HP_2(LP_3 \pm HP_3)\\ &= \theta_1LP_2\theta_3 \pm \theta_1HP_2\theta_3\\ &= \theta_1(LP_2 \pm HP_2)\theta_3\\ &= \theta_1\theta_2\theta_3 \text{ Q.E.D.}\end{aligned}$$

Low, Low-Mid, Mid, Mid-High and High Controls
The **Low**, **Low-Mid**, **Mid**, **Mid-High** and **High** buttons determine if that section of the crossover is bypassed. If the word Low, Low-Mid, Mid, Mid-High or High is displayed on the button, then that portion of the crossover is not bypassed. If the word Bypassed is displayed on the button's face, then that portion of the crossover is bypassed. Clicking on the buttons will toggle their settings.

Link/Unlink Control
The **Link/Unlink** button determines if the settings in the crossover track their corresponding partners. A simple rule of thumb is that when the sections are linked, the component on the right will track its partner on the left. Clicking on the button(s) will toggle the settings.

Slope
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Class
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Cutoff Frequency
The Cutoff Frequency can be adjusted three ways:

Using the Knob
Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Passband Gain
The Passband Gain can be adjusted three ways:

Using the Knob
Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Polarity Control (+/-)
This button determines the polarity of the signal. A + indicates that the signal will not have it's phase altered. A - indicates that the signal's phase will be changed 180°. Clicking on the button will toggle the setting.

Mute Control (M)
This button determines if the input signal will be muted. A black M indicates that the signal is not muted, while a red M indicates that the signal is muted. Clicking on the button will toggle the setting.

Cut

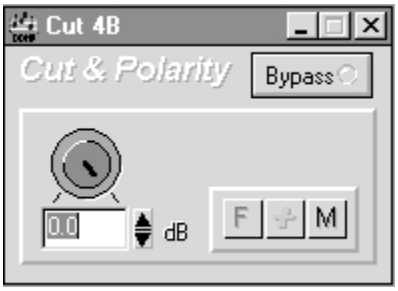


Figure 8-11 Cut component.

The Cut (see Figure 8-11) can apply as much as 96 dB attenuation, or mute completely, and invert polarity.

Attenuation and Polarity

Application of attenuation is accomplished by multiplying each data sample by a number whose absolute value lies between 1.0 (0 dB) and 0.000015625 (-96 dB). Muting is accomplished by multiplying each sample by zero. Polarity inversion is accomplished by multiplying by a negative number. All gain and polarity changes are soft, meaning that they are gradually changed from the old setting to the new in order to help prevent the occurrence of clicks or pops.

Bypass

Bypassing cut is equivalent to setting its attenuation to 0 dB and its polarity to + (in-phase). Bypassing a muted attenuator un-mutes it.

Attenuator

The attenuator can be set three ways:

Using the Knob

Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box

Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows

To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Polarity Control (+/-)
This button determines the polarity of the signal. A + indicates that the signal will not have it's phase altered. A - indicates that the signal's phase will be changed 180. Clicking on the button will toggle the setting.

Mute Control (M)
This button determines if the input signal will be muted. A black M indicates that the signal is not muted, while a red M indicates that the signal is muted. Clicking on the button will toggle the setting.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Delay

Figure 8-12 Delay component.

Delay (see Figure 8-12) is used to compensate for differences in the physical alignment of loudspeakers or loudspeaker drivers.

In the ISP-100, the delay in each output channel is adjustable in the range 0 – 2,500 milliseconds (depending on QuickMAP used) in 20.833 microsecond (single sample) increments. This translates to approximately (0 - 2750 feet) in 0.275 inch increments, assuming that the speed of sound is approximately 1100 ft/sec.

Setting Delay

The Delay control can be adjusted two ways:

Using the Text Box

Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows

To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine (F) or Coarse (C) Control

This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Bypass Control (master)

This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Dither

Figure 8-13 Dither component.

Dithering (see Figure 8-13) is used when shortening high resolution digital audio data words to a lower resolution; for example, when sending 24-bit ISP-100 audio data to 20-bit D/A converters, or directly to a 16-bit Digital Audio Tape recorder through a digital interface. Simply truncating, or discarding, the unused bits results in audible artifacts that are generally objectionable. Dithering does not eliminate these artifacts, it changes their characteristics so that they are less noticeable to the human ear.

If, after reducing the number of bits in a signal, the original high-resolution (more bits) signal is subtracted from the final lower-resolution (fewer bits) signal, the difference is called the error signal:

(Final Signal) - (Original Signal) = (Error Signal)

Some very simple algebra shows that the final signal can be thought of as the sum of the error signal and the original signal:

(Final Signal) = (Error Signal) + (Original Signal)

When audio data words are shortened by simple truncation, it turns out that the error signal is correlated with itself and with the original signal. This simply means that the error signal is not pure white noise, and that the frequency and amplitude of the error signal is in some way related to the frequency and amplitude of the original signal.

If noise with some very specific amplitude and spectral characteristics is added to the original signal before truncation, then the correlation is destroyed and the error signal sounds like low level noise instead of tones. Even though the added noise actually raises the noise floor in the final low-resolution signal, it sounds better because human hearing finds low level broadband noise to be much less objectionable than spurious tones.

This addition of noise before truncation is called dithering. The noise that is added is called the dither signal.

In summary, dithering is a way of making digital audio better by making it worse.

Bit Level

In the ISP-100 the bit level of the dither signal is selectable as whole

numbers between 16 and 24 bits, inclusive. The bit level of the dither signal should be set to the bit resolution of the destination for the data. The resulting dither signal will have an amplitude equal to ±1.0 quantization level (peak-to-peak) at the selected Bit Level.

Using the examples, the bit level for the dither signal should be set to 20 bits when the data are destined for the 20-bit D/A converters, and to 16 bits when the data are destined for the DAT recorder.

Dither Signal Characteristics
The dither signal used in the ISP-100 has a highpass spectral shape for reduced audibility. It is characterized by approximately a +6 dB/ octave slope in the audio band.

The dither signal used in the ISP-100 has a triangular probability density function (TPDF) for the minimization of noise modulation effects.

Bypass
Bypassing dither results in no dither whatsoever being added to the signal. This is different than setting the dither to the 24th bit level.

Dither Bit Level
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Filter

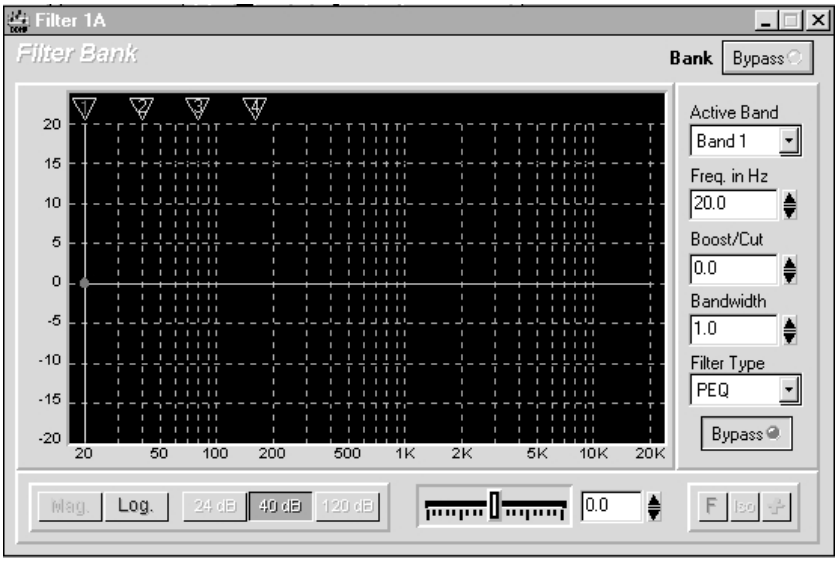


Figure 8-14 Filter component.

The filter component can be used to implement custom crossovers.

Filters (see Figure 8-14) are used to modify the spectral balance, frequency content, or phase response of the signals upon which they operate.

Filter Bands

The filters in the ISP-100 are arranged in bands, each of which represent a single 2nd-order filter in the DSP code. Each band can be configured as one of seventeen different filter types:

LowPass Filter, 6 dB/octave
LowPass Filter, 12 dB/octave*

HighPass Filter, 6 dB/octave
HighPass Filter, 12 dB/octave*

LowShelf Filter, 6 dB/octave
LowShelf Filter, 12 dB/octave

HighShelf Filter, 6 dB/octave
HighShelf Filter, 12 dB/octave

Notch Filter

Parametric EQ Filter

Peaked HighPass Filter

AllPass Filter, 1st-order
AllPass Filter, 2nd-order

* More filters available in the Crossover filters.

All filters in the ISP-100 are derived from analog prototypes. The analog filters are approximated by the Bilinear Transform method, prewarped to match response at the critical frequency appropriate for the type of filter (cutoff frequency for LowPass/HighPass and Peaked HighPass; hinge frequency for LowShelf/HighShelf; center frequency for all others).

Each filter band may be bypassed; so as to pass the signal unaltered.

LowPass Filters

LowPass Filters pass low frequencies and stop high frequencies. Since it is not realistically possible to create a perfect filter that passes low frequencies totally unaltered and stops high frequencies completely, lowpass filter design involves compromises that allow some rounding of the corner at the filter cutoff frequency and some slope in the transition to the high frequency stopband. Different compromise schemes are given different names; the ISP-100 incorporates Bessel, Butterworth, and Linkwitz-Riley lowpass filter types.

The cutoff frequency is defined as the frequency at which the magnitude of the filter response has fallen to -3 dB relative to the unfiltered signal in the Bessel and Butterworth types, and to -6 dB in the Linkwitz-Riley types. In the ISP-100 lowpass filters the cutoff frequency is continuously variable from 20Hz to 20 kHz.

In the ISP-100 lowpass filters the passband slope is selectable from -6 dB/octave to -12 dB/octave for the Bessel and Butterworth types.

NOTE: Linkwitz-Riley filters are also known as Butterworth-Squared filters because they are the product of two Butterworth filters. A -12 dB/octave Linkwitz-Riley lowpass filter may be constructed by cascading two identical -6 dB/octave Butterworth lowpass filters; a -24 dB/octave Linkwitz-Riley lowpass filter may be constructed by cascading two identical -12 dB/octave Butterworth lowpass filters.

HighPass Filters

HighPass Filters pass high frequencies and stop low frequencies. Since it is not realistically possible to create a perfect filter that passes high frequencies totally unaltered and stops low frequencies completely, highpass filter design involves compromises that allow some rounding of the corner at the filter cutoff frequency and some slope in the transition to the low frequency stopband. Different compromise schemes are given different names; the ISP-100 incorporates Bessel, Butterworth, and Linkwitz-Riley lowpass filter types.

The cutoff frequency is defined as the frequency at which the magnitude of the filter response has fallen to -3 dB relative to the unfiltered signal in the Bessel and Butterworth types, and to -6 dB in the Linkwitz-Riley types. In the ISP-100 highpass filters the cutoff frequency is continuously variable from 20Hz to 20 kHz.

In the ISP-100 highpass filters the passband slope is selectable from 6 dB/octave, 12 dB/octave, 18 dB/octave, and 24 dB/octave for the Bessel and Butterworth types, and from 12 dB/octave and 24 dB/octave for the Linkwitz-Riley type.

NOTE: Linkwitz-Riley filters are also known as Butterworth-Squared filters because they are the product of two Butterworth filters. A 12 dB/octave Linkwitz-Riley highpass filter may be constructed by cascading two identical 6 dB/octave Butterworth highpass filters; a 24 dB/octave Linkwitz-Riley highpass filter may be constructed by cascading two identical 12 dB/octave Butterworth highpass filters.

LowShelf Filters

LowShelf Filters raise or lower the magnitude response at frequencies below the hinge frequency without altering the response at frequencies above the hinge frequency. Since it is not realistically possible to create a perfect filter that alters only low frequencies without affecting high frequencies, lowshelf filter design involves compromises that allow some rounding of the corner at the filter hinge frequency and some slope in the transition to the unaltered high frequencies.

The hinge frequency is defined as the frequency at which the magnitude of the filter response is changed by (approximately) 3 dB relative to the unfiltered signal. In the ISP-100 lowshelf filters the hinge frequency is continuously variable from 20Hz to 20 kHz.

In the ISP-100 lowshelf filters the transition band slope is selectable from 6 dB/octave and 12 dB/octave.

In the ISP-100 lowshelf filters the boost/cut is continuously variable from -12 dB to +12 dB.

NOTE: When a lowshelf filter with boost is selected, the expected filter frequency response is characterized by boosted low frequencies and unaltered high frequencies. In the ISP-100 this filter will exhibit unaltered response at low frequencies and cut response at high frequencies. The shape of the frequency response will be exactly as expected, but the level of the response will be shifted downward by an amount exactly equal to the selected boost quantity. This is necessary to prevent numerical overflow problems in the DSP filter coefficients, and represents normal operation for the lowshelf filter.

HighShelf Filters

HighShelf Filters raise or lower the magnitude response at frequencies above the hinge frequency without altering the response at frequencies below the hinge frequency. Since it is not realistically possible to create a perfect filter that alters only high frequencies without affecting low frequencies, highshelf filter design involves compromises that allow some rounding of the corner at the filter hinge frequency and some slope in the transition to the unaltered low frequencies.

The hinge frequency is defined as the frequency at which the magnitude of the filter response is changed by (approximately) 3 dB relative to the unfiltered signal. In the ISP-100 highshelf filters the hinge frequency is continuously variable from 20Hz to 20 kHz.

In the ISP-100 highshelf filters the transition band slope is selectable from 6 dB/octave and 12 dB/octave.

In the ISP-100 highshelf filters the boost/cut is continuously variable from -12 dB to +12 dB.

NOTE: When a highshelf filter with boost is selected, the expected filter frequency response is characterized by boosted high frequencies and unaltered low frequencies. In the ISP-100 this filter will exhibit unaltered response at high frequencies and cut response at low frequencies. The shape of the frequency response will be exactly as expected, but the level of the response will be shifted downward by an amount exactly equal to the selected boost quantity. This is necessary to prevent numerical overflow problems in the DSP filter coefficients, and represents normal operation for the highshelf filter.

Notch Filters

Notch Filters pass all frequencies except for the notch frequency, which they stop completely. Since it is not realistically possible to create a perfect filter that stops one frequency completely and passes all other frequencies totally unaltered, notch filter design involves compromises that allow some width in the notch and less than infinite attenuation at the notch frequency.

In the ISP-100 notch filters the notch frequency is continuously variable from 20 Hz to 20 kHz.

In the ISP-100 notch filters the notch width is continuously variable from 1/12 octave to 1 octave.

In the ISP-100 notch filters the notch depth (attenuation at the notch frequency) is continuously variable from -50 dB to 0 dB.

NOTE: The width of a notch filter is defined by the points on either side of the notch at which the response is -3 dB relative to the unaltered signal, not at the points where the response has risen +3 dB relative to the notch depth. The ISP-100 notch filters use the correct definition of width, so a $^{1}/_{12}$ octave notch in the ISP-100 is considerably narrower than a $^{1}/_{12}$ octave parametric EQ filter whose width is defined relative to the notch depth.

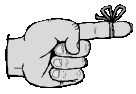
Parametric EQ Filters

Equalization (EQ) filters are used to modify the frequency response in a band of frequencies. Parametric EQ filters provide for adjustment of all filter parameters (center frequency, boost or cut, and bandwidth), as opposed to graphic EQ filters which have fixed frequency and bandwidth and provide only for adjustment of boost or cut.

The EQ filters in the ISP-100 are full parametric. Each may be adjusted for center frequency, boost or cut, and bandwidth.

Center Frequency
The center frequency is the frequency at which the maximum boost or cut takes place. In the ISP-100 EQ filters the center frequency is continuously variable from 20 Hz to 20 kHz.

Bandwidth
The bandwidth is measured at the points on either side of the center frequency where the boost is 3 dB less than the maximum boost or the cut is 3 dB less than the maximum cut, whichever is appropriate. In the ISP-100 filters, bandwidth can be expressed in either octave fraction (fractions of an octave, or number of octaves if the bandwidth is greater than a full octave) or quality (Q).



Bandwidth can be displayed in terms of Q or octave fraction. See page 7-3 for information on changing the way filter bandwidth is displayed.

The conversions between octave fraction and Q are:

$$\text{octave fraction} = 2\left\{\log_2\left[\frac{1}{Q}+\sqrt{\frac{1}{Q^2}+4}\right]-1\right\}$$

and

$$Q = \frac{1}{2^{\frac{\text{octave fraction}}{2}} - 2^{\frac{\text{-octave fraction}}{2}}}$$

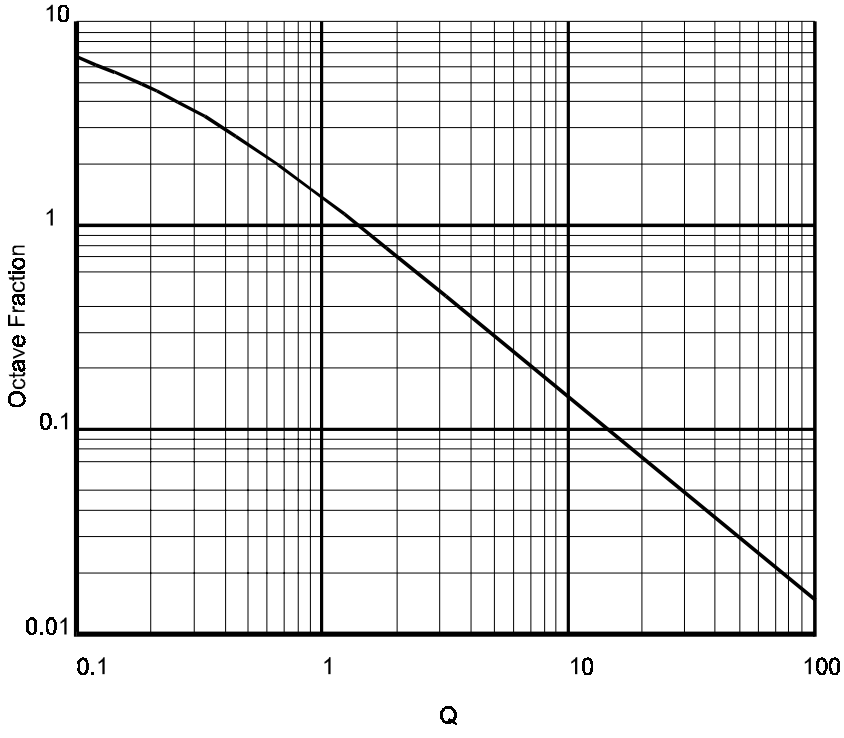


Figure 8-15 Octave Fraction vs. Q relationship.

The relationship between octave fraction and Q is shown in Figure 8-15.

In the ISP-100 EQ filters the bandwidth is continuously variable from 1/12 octave to 3 octaves (which translates to Q between 17.3 and 0.404).

Boost and Cut
In the ISP-100 EQ filters the boost/cut is continuously variable from +12 dB to -12 dB.

Peaked HighPass Filters
The Peaked HighPass filter supplied with the ISP-100 is designed to provide bass boost appropriate for low frequency extension in vented-box loudspeakers. As the input frequency is reduced, the filter response rises toward the selected peak magnitude at approximately the selected cutoff frequency, and then returns to a 12 dB/octave rolloff characteristic below.

Peak Magnitude
In the ISP-100 Peaked HighPass filter the peak magnitude of the filter response is continuously variable from 0 dB to +20 dB. The peak occurs at approximately the selected cutoff frequency when the selected Peak Magnitude is large; it occurs at somewhat higher frequency when the selected Peak Magnitude is small. (If the peak magnitude is selected to be 0 dB, then the peak occurs at infinite frequency and the actual magnitude at the cutoff frequency is -3 dB.)

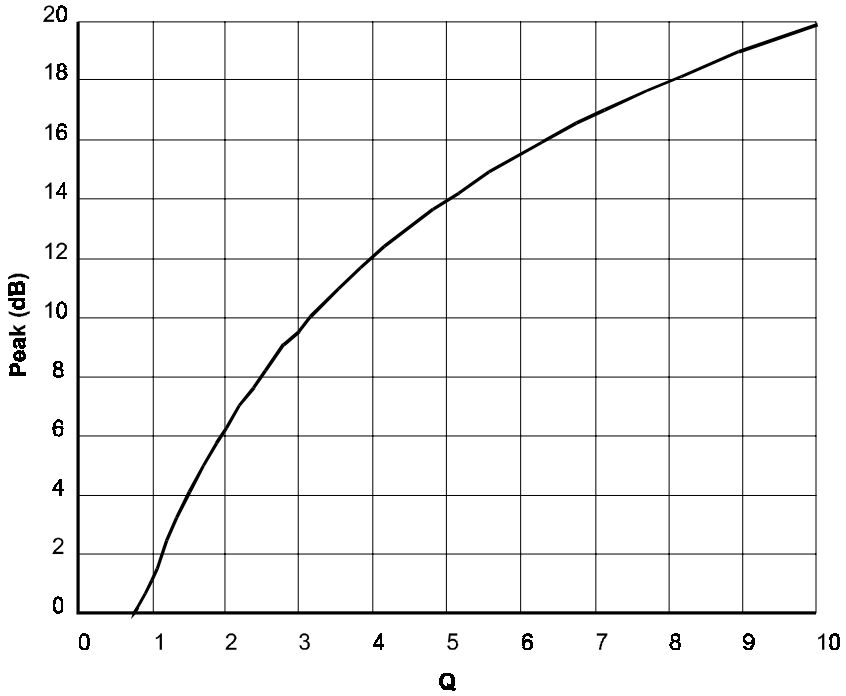


Figure 8-16 Peak Magnitude vs. Q relationship.

The relationship between Peak Magnitude and filter Q is shown in Figure 8-16.

Cutoff Frequency
The cutoff frequency is the design frequency of the highpass filter, and is only approximately representative of the location of the magnitude peak. In the ISP-100 the cutoff frequency is continuously variable from 20 Hz to 20 kHz.

AllPass Filters
AllPass Filters shift phase without altering magnitude. The allpass filters supplied with the ISP-100 are designed to provide phase compensation for multiway Linkwitz-Riley crossover networks. The 1st-order allpass filters compensate for 2nd-order (12 dB/octave) Linkwitz-Riley crossovers; the 2nd-order allpass filters compensate for 4th-order (24 dB/octave) Linkwitz-Riley crossovers.

A detailed explanation of the use of allpass compensation may be found in “Active Realization of Multiway All-Pass Crossover Systems” by Joseph A. D’Appolito; *Journal of the Audio Engineering Society*, Volume 35, Number 4, April 1987.

In the ISP-100 allpass filters the filter order is selectable from 1st-order and 2nd-order.

In the ISP-100 allpass filters the center frequency is continuously variable from 20 Hz to 20 kHz.

Filter Type
To change settings, click on the arrow next to the entry in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Active Band
To change settings, click on the arrow next to the number in the text box. A selection list will appear. The list may or may not be scrollable. To make a selection click on the desired entry.

Bypass
The Bypass control bypasses an individual filter band. If the LED on the button is gray, then the filter band is active. If the LED on the button is red, then the filter band is bypassed.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Frequency in Hz
The center frequency of a filter can be adjusted three ways:

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the

text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Using the Filter Tool
Click on the triangle associated with the filter band that needs its center frequency adjusted. While holding the mouse button down, drag the mouse from side to side. A display indicating the current center frequency will appear. When the desired frequency is reached, release the mouse button. Please note that exact center frequencies may be impossible to obtain using this method.

Filter bands can be stacked for increased boost/cut.

Boost/Cut
The Boost (gain) or Cut (attenuation) of a filter can be adjusted three ways:

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Using the Filter Tool
Click on the red dot on the active filter band. While holding down the mouse button, move the mouse up and down. A display will appear indicating the amount of boost or cut. When the desired level is reached, release the mouse button.

Bandwidth
The Bandwidth of the filter can be adjusted in three ways:

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted.

Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Using the Filter Tool
Right-click on the red dot on the active filter band. While holding down the right-mouse button, move the mouse from side-to-side. A display will appear indicating the bandwidth of the filter. When the desired level is reached, release the right-mouse button.

Using the Magnitude (Mag.) or Phase Control
This button allows the user to view the composite filter response in either a magnitude vs. frequency or phase vs. frequency environment. Clicking on the button will toggle the setting.

Using the Logarithmic (Log.) or Linear Control
This button allows the user to view the composite filter response in either a logarithmic or linear environment. Clicking on the button will toggle the setting.

Using the 24 dB, 40 dB, 120 dB Controls
These buttons allow the user to view the composite filter response in three different dynamic ranges. Clicking on one of the buttons will set the dynamic range.

Gain Trim
The Gain Trim can be used to shift response up or down. This is especially useful with shelving filters. With shelving filters in a boost situation, an amount of gain trim equal to the amount of boost should be added. Gain trim does not need to be added in the case of a shelving filter with cut. The Gain Trim can be adjusted two ways:

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the Slider
Click on the indicator on the slider and while continuing to hold down the mouse button, move the mouse side-to-side. The numbers in the box next to the slider will change. When the desired level is reached, release the mouse button.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Isolate (Iso) Control
This button allows the user to toggle between viewing either the composite response of all active filters in the filter bank or the composite response of an individual filter. Clicking on the button will toggle the setting.

Polarity Control (+/-)
This button determines the polarity of the signal. A + indicates that the signal will not have it's phase altered. A - indicates that the signal's phase will be changed 180°. Clicking on the button will toggle the setting.

Gain

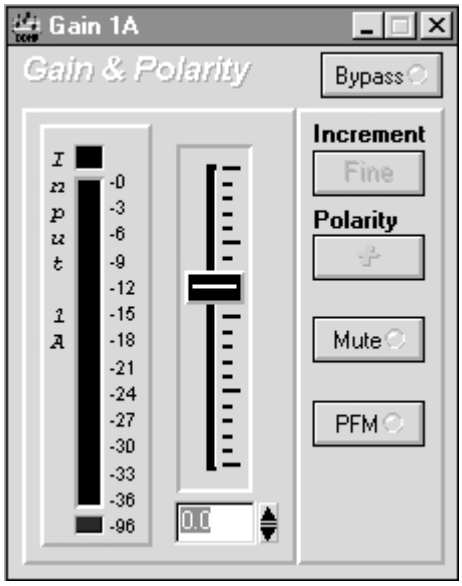


Figure 8-17 Gain component.

Gain (see Figure 8-17) is the first function applied to all ISP-100 input data. The gain component performs many operations, including metering of raw input data (pre-fader), application of up to 18 dB gain, 96 dB attenuation, or full-muting, metering of data after the level has been changed (post-fader), and polarity inversion.

Metering

Gain includes peak-reading meters. The meters show the amplitude of the individual digital samples having the largest absolute value. The meters are graduated in dB and referenced to digital clipping (dBFS; dB relative to Full Scale).

Because of their peak-reading nature, gain meters are not useful for monitoring average signal levels as the peak level of a signal can exceed its average by a tremendous amount. The advantage of true peak-indicating meters in a digital signal processor like the ISP-100 lies in their ability to indicate digital clipping. Regardless of the average level of a signal, if any peaks in the signal are clipped then the Gain meter reads 0 dB.

Gain and Polarity

Application of attenuation within the gain component is accomplished by multiplying each data sample by a number whose absolute value lies between 8.0 (+18 dB) and 0.000015625 (-96 dB). Muting is accomplished by multiplying each sample by zero. Polarity inversion is accomplished by multiplying by a negative number. All gain and polarity changes are soft, meaning that they are gradually changed from the old setting to the new in order to help prevent the occurrence of clicks or pops.

Bypass

Bypassing the gain component is equivalent to setting its attenuation

to 0 dB and its polarity to positive (in-phase). Bypassing in a muted channel un-mutes it.

Metering
The meter located at the left of the component measures level in dBFS. The meter's level will turn red 3 dB before clipping. The red clip light is located at the top of the meter. The green signal present light is located at the bottom of the meter.

Master Attenuator
The Master Attenuator can be adjusted three ways:

Using the Slider
Click on the indicator on the slider and while continuing to hold down the mouse button, move the mouse up and down. The numbers in the box below the slider will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine or Coarse Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Polarity Control (+/-)
This button determines the polarity of the signal. A + indicates that the signal will not have it's phase altered. A - indicates that the signal's phase will be changed 180°. Clicking on the button will toggle the setting.

Mute Control
This button determines if the input signal will be muted. A gray LED indicates that the signal is not muted, while a red LED indicates that the signal is muted. Clicking on the button will toggle the setting.

Pre-Fader Metering (PFM)
This button determines if the meter is getting its data pre- or post-fader. A gray LED indicates that the meter information is post-fader

(i.e. after the master attenuator). A green LED indicates that the meter information is pre-fader. Clicking on the button will toggle the setting.

Gate

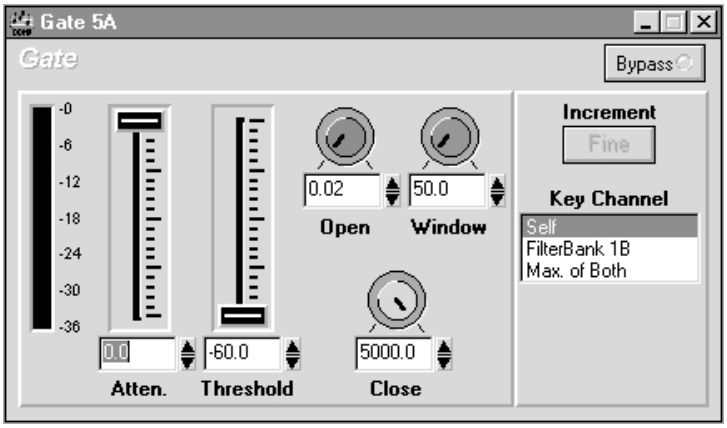


Figure 8-18 Gate component.

A gate (see Figure 8-18) is used in situations where the input signal is characterized by long periods of silence and a noticeably high noise floor, such as a signal taken from magnetic tape or from an open microphone. Gating is accomplished by attenuating the output signal during the periods of input silence in order to reduce the audibility of the noise.

Familiar Controls

Most of the controls in the ISP-100 Gate are common to nearly all traditional noise gates. These controls should be familiar to anyone with experience in the operation of any gate.

Threshold

The Threshold control adjusts the input signal level below which attenuation is applied. When the average input signal level is above the threshold value, no attenuation is applied and the level of the output signal exactly matches the level of the input signal. When the average input signal is below the threshold value, attenuation is applied and the level of the output signal is less than the level of the input signal by the amount dictated by the selected gated attenuation value.

Threshold is adjustable in the range {-60 dBFS to 0 dBFS}.

Opening Time

A gate monitors the level of the input signal and controls the level of the output signal accordingly. When the average input signal level increases from below threshold to above, a gate opens (removes all attenuation) so that the output signal level matches the input signal level. Desirable sound effects can sometimes be created if a gate is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Opening Time control adjusts the duration of this ramp.

Opening Time is adjustable in the range 20 µSec (one sample) to 50 mSec.

Closing Time

A gate monitors the level of the input signal and controls the level of the output signal accordingly. When the average input signal level decreases from above threshold to below, a gate closes (applies attenuation) so that the output signal level is less than the input signal level. A gate generally sounds better if it is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Closing Time control adjusts the duration of this ramp.

Closing Time is adjustable in the range 20 µSec (one sample) to 5 Sec.

Key Channel

There are situations in which gating is applied to one channel based upon the level in another channel. This might be used in a stereo signal pair to prevent image shift; one of the channels would be considered to be the level reference and the gates in both channels would be set up identically.

The Key Channel selection indicates the channel whose level is to be used to control the gate. If the input channel is to control itself, then it should be selected as the Key Channel. Otherwise the appropriate other channel should be selected from the set of available Key Channels in the menu.

Unfamiliar Controls

There are two controls on the ISP-100 gate that are not available in traditional noise gates. These controls provide much greater flexibility and utility to this gate than can be achieved with traditional gates.

Gated Attenuation

The gated attenuation is the amount of attenuation that is applied to a signal whose average level falls below the selected threshold. Most traditional noise gates use a fixed attenuation or simply mute the channel completely. The ISP-100 gate provides for operator control of the gated attenuation.

Gated Attenuation is adjustable in the range -100 dB to 0 dB.

Detection Window

The detection window is the period of time over which the gate computes the average signal level. Every noise gate that operates upon average signal level uses a detection window. In traditional gates this window is of fixed length, and the operator may not even be aware of that length.

The ISP-100 gate provides for operator control of the length of the detection window. Very dynamic signals and signals containing a lot of high frequency energy require shorter detection windows. Non-dynamic signals and signals containing mostly low frequency energy require longer detection windows.

Detection Window is adjustable in the range 20 µSec (one sample) to 5 Sec.

Metering
The gate meter shows the attenuation applied to the signal passing through the gate at any given moment in time. When the signal is below threshold and the gate is closed, the meter indicates negative values (gain less than 1.0). When the signal is above threshold and the gate is open, the meter indicates 0 dB (gain of 1.0). Meter values are raw, no averaging or ballistics are applied.

Notes on Operation of the Gate

Bypass
Bypassing a gate causes the signal to pass through unaltered.

Computation of Average Level
There are two common methods of computing average signal level; one is based upon RMS (Root Mean Square) calculations, and the other is based upon Absolute Value calculations. Each of these methods produces a slightly different value for signal level, and the difference depends upon the characteristics of the signal itself. The ISP-100 gate uses Absolute Value for its computation of signal level, so the level reported by the gate may differ slightly from the RMS value expected for a given signal.

Keying from Other Channels
When the key channel input to a ISP-100 gate is selected to be a channel other than its input, the gate controls must still be set. The key channel selects only the channel's data, not its gate control settings. If multiple gates utilize the same channel as their key sources, then all controls on all gates must still be configured.

Metering
The meter located to the left of the component indicates the amount of attenuation in dB.

Attenuation and Threshold Controls
These controls can be adjusted three ways:

Using the Slider
Click on the indicator on the slider and while continuing to hold down the mouse button, move the mouse up and down. The numbers in the box below the slider will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted.

Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Open, Window, Close Controls
These controls can be adjusted three ways:

Using the Knob
Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Key Channel
To change settings click on the desired entry.

Fine (F) or Coarse (C) Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Limiter

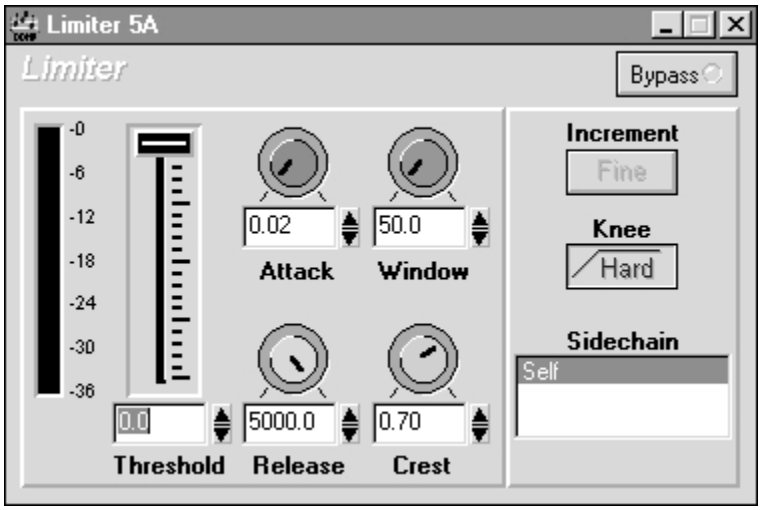


Figure 8-19 Limiter component.

A limiter (see Figure 8-18) is used in situations where the output signal must not exceed a given level, regardless of any increase in the level of the input signal. Limiting is often used for system protection purposes, such as the avoidance of amplifier clipping or the prevention of mechanical or thermal damage to loudspeaker components.

Familiar Controls

Most of the controls in the ISP-100 Limiter are common to nearly all traditional dynamic range limiters. These controls should be familiar to anyone with experience in the operation of any limiter.

Threshold

The Threshold control adjusts the input signal level above which limiting is applied. When the input signal level is below the threshold value, no limiting takes place and the change in level of the output signal exactly matches the change in level of the input signal. When the input signal level is above the threshold value, limiting is applied such that the output signal level does not increase, regardless of any increase in the input signal level.

Threshold is adjustable in the range {-60 dBFS to 0 dBFS}.

Attack Time

A limiter monitors the level of the input signal and controls the level of the output signal accordingly. When the input signal level suddenly increases, a limiter allows the output signal level to also increase, but only up to the threshold setting and not beyond. Desirable sound effects can sometimes be created if a limiter is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Attack Time control adjusts the duration of this ramp.

Attack Time is adjustable in the range {20 uSec (one sample) to 50 mSec}.

Release Time
A limiter monitors the level of the input signal and controls the level of the output signal accordingly. When the input signal level suddenly decreases, a limiter allows the output signal level to also decrease only as much as is appropriate, given the relationship between the new input signal level and the threshold setting. A limiter generally sounds better if it is forced to ramp the output signal level to its new value gradually rather than instantaneously. The Release Time control adjusts the duration of this ramp.

Release Time is adjustable in the range {20 μSec (one sample) to 5 Sec}.

Knee Selection
Below the threshold level, no limiting is applied and any changes in input level are matched exactly by changes in output level. Above the threshold level, limiting is applied and changes in the output level are not allowed regardless of the input level. The transition from no limiting to limiting can be set to take place abruptly or gradually.

Under hard knee conditions the output level tracks the input level perfectly as it rises up to the threshold. Above the threshold the output level does not change. There is an abrupt transition from no limiting below the threshold to limiting above.

Under soft knee conditions the output level and the input level track until approximately 12-18 dB below the threshold, beyond which they slowly begin to diverge. The output level is approximately 2.5 dB below the input level at the threshold. The divergence continues to increase gradually until approximately 6 dB above the threshold, at which point the output level is equal to the threshold value and beyond which the output level no longer rises. There is a smooth, gradual transition from no limiting below the threshold to limiting above.

The sonic differences between the two knee types are often subtle and are very much signal-dependent. Experimentation is recommended.

Sidechain Source
There are situations in which limiting is applied to one channel based upon the level in another channel. This might be used in a stereo signal pair to prevent image shift; one of the channels would be considered to be the level reference and the limiters in both channels would be set up identically.

The Sidechain Source channel selection indicates the channel whose level is to be used to control the limiter. The choices for sidechain source are the input channel itself, one other channel, or the maximum amplitude of these two at any given moment. This makes stereo linking possible by configuring the limiters identically and utilizing the same sidechain source for both.

Unfamiliar Controls

There are two controls on the ISP-100 limiter that are not available in traditional dynamic range limiters. These controls provide much greater flexibility and utility to this limiter than can be achieved with traditional limiters.

Detection Window

The detection window is the period of time over which the limiter computes the average signal level. Every dynamic range limiter that operates upon average signal level uses a detection window. In traditional limiters this window is of fixed length, and the operator may not even be aware of that length.

Some traditional limiters provide switching between average and peak detection. This is actually just a selection between a relatively long detection window of generally a few tens or hundreds of milliseconds, and a very short detection window of generally a few tens or hundreds of microseconds.

The ISP-100 limiter provides for operator control of the length of the detection window. Very dynamic signals and signals containing a lot of high frequency energy require shorter detection windows. Non-dynamic signals and signals containing mostly low frequency energy require longer detection windows.

Detection Window is adjustable in the range {20 uSec (one sample) to 5 Sec}.

Crest Factor Sensitivity

A signal's crest factor is the ratio of a its peak level to its average level. The signal having the lowest possible crest factor is the square wave, whose peak and average levels are equal and whose crest factor is therefore 1.0. The peak level of a pure sinewave tone is 1.414 times its average level; a sinewave therefore has a crest factor of 1.414. Most music and speech signals contain peaks that are very much higher than their average levels, therefore their crest factors are very much greater than one. The important points to remember are that a signal's peak value is always greater than or equal to its average value, and that the crest factor of any signal is therefore always greater than or equal to 1.0.

The Crest Factor Sensitivity adjustment on the ISP-100 limiter function controls its sensitivity to the crest factor of the input signal. It works like this:

The signal level detection algorithm used in the ISP-100 limiter consists of two separate detectors operating in parallel. One of the detectors computes the average level over the selected detection window; the other computes the peak level of *each* input sample.

The peak level is multiplied by the setting of the Crest Factor Sensitivity control — always a number between zero and

one. The two quantities are then compared and the larger is returned as the signal level.

When the Crest Factor Sensitivity is set to 0.00 the ISP-100 limiter operates as a traditional limiter responding to average signal levels; when the Crest Factor Sensitivity is set to 1.00 the ISP-100 limiter operates as a traditional limiter responding to peak signal levels. Values in-between represent various hybrids of the two. If the limiter is to ignore peaks and respond only to average levels, then the Crest Factor Sensitivity may be set near zero. If the limiter is to respond only to peaks, then the Crest Factor Sensitivity may be set near one.

Good results may be obtained by following these steps:

1. Set the Crest Factor Sensitivity to 0.00. This will cause the ISP-100 limiter to behave like a traditional limiter responding to average signal levels.

2. Set all of the other limiter controls as appropriate for the signal being limited, just as they would be set in a traditional limiter.

3. Adjust the Crest Factor Sensitivity control so that the limiter responds appropriately to short signal peaks. If it were possible to know, in advance, what the crest factor of the input signal was going to be, then the Crest Factor Sensitivity control would ideally be set to 1/(the crest factor of the input signal).

Since this quantity is not generally known, it is best to experiment until satisfactory results are obtained.

Metering
The limiter meter shows the attenuation applied to the signal passing through the limiter at any given moment in time. When the signal is being limited the meter indicates negative values (gain less than 1.0). When the signal is *not* being limited the meter indicates 0 dB (gain of 1.0). Meter values are raw, no averaging or ballistics are applied.

Notes on Operation of the Limiter:

Bypass
Bypassing a limiter causes the signal to pass through unaltered.

Overshoot
The limiter algorithm used in the ISP-100 allows for a small amount of overshoot on transients, regardless of the settings of the Detection Window and Crest Factor Sensitivity controls. The amount of overshoot is somewhat dependent upon the setting of the Threshold control; the lower the threshold the greater the overshoot. This represents normal operation for the limiter.

Computation of Average Level
There are two common methods of computing average signal level; one is based upon RMS (Root Mean Square) calculations, and the other is based upon Absolute Value calculations. Each of these methods produces a slightly different value for signal level, and the difference depends upon the characteristics of the signal itself. The ISP-100 limiter uses Absolute Value for its computation of signal level, so the level reported by the limiter may differ slightly from the RMS value expected for a given signal.

Crest Factor Sensitivity and Sinewaves
The peak level of a pure sinewave tone is 1.414 times its average level. The ISP-100 limiter will therefore apply approximately 3 dB more limiting to a pure sinewave signal when the Crest Factor Sensitivity is set to 1.00 (peak-responding) than when it is set to 0.00 (average-responding). This represents normal operation for the limiter.

Sidechaining
When the sidechain source for an ISP-100 limiter is selected to be a channel other than its own input, the limiter controls must still be set. The sidechain selects only the channel's data, not its limiter control settings. If multiple limiters utilize the same channel as their sidechain sources, then all controls on all limiters must still be configured.

Meter
The meter located at the left of the component indicates the amount of attenuation taking place in dB.

Threshold
The Threshold can be adjusted three ways:

Using the Slider
Click on the indicator on the slider and while continuing to hold down the mouse button, move the mouse up and down. The numbers in the box below the slider will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of

increase or decrease is controlled by the fine/coarse control.

Attack, Window, Release, and Crest factor controls.
All of these controls can be adjusted in three ways:

Using the Knob
Click on the indicator on the knob and while continuing to hold down the mouse button, move the mouse in an arc. The numbers in the box below the knob will change. When the desired level is reached, release the mouse button.

Using the Text Box
Click in the box. Using the **Back Space** and/or **Delete** keys erases the current entry and enters the new level. Be sure to press the **Enter** key after the entry is made. Clicking outside the text box without pressing the **Enter** key will cause the setting to return to the number that it was before a change was attempted. Alternatively, click and drag across the number in the box to highlight the number in it. Type the new setting and press the **Enter** key when finished.

Using the UP/DOWN Arrows
To increase the level, click on the up arrow. To decrease the level attenuation, click on the down arrow. The amount of increase or decrease is controlled by the fine/coarse control.

Fine or Coarse Control
This button determines the increment/decrement associated with the UP/DOWN arrows. Fine produces small steps in value, while Coarse provides large steps in value. Clicking on the button will toggle the setting.

Knee Control
This button determines the type of transition that occurs when the component goes from inactive to active. A Hard knee causes the response by the component to be abrupt, while a Soft knee causes a gentle response. Clicking on the button will toggle the setting.

Bypass Control (master)
This button determines if the component is in the processing chain. If the LED on the button is gray, then the component is inserted into the signal processing chain. If the LED is red, then the component is bypassed in the signal processing chain. A bypassed component will be colored red in the QuickMAP.

Select

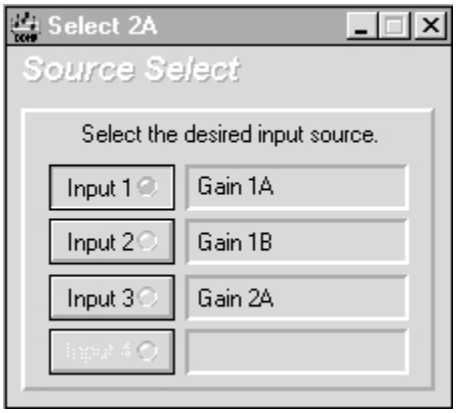


Figure 8-20 Select component.

The select function (see Figure 8-20) allows you to set the source that feeds portions of a QuickMAP. The user is allowed to select from as many inputs as the QuickMAP allows. An entire QuickMAP can be fed from a single source or multiple sources depending on the settings of the Selects within the QuickMAP.

Note: The amount of available sources is determined by the QuickMAP and not the hardware configuration. For example: A particular QuickMAP has three source selects in it. This will require two input cards. The user is only able to select from three inputs (as determined by the QuickMAP) even though two input cards provide a total of four inputs in terms of hardware.

Selecting the Input

To select the input click on the button next to the input that you want to make active. The LED on the button should turn from gray to green. If a particular button is grayed out then it is not available for selection.

GLOSSARY

A to D	Analog to Digital. Conversion of an analog signal to a digital representation.
Amplitude factor	See CREST FACTOR.
Attack	The rise of a musical note from zero to full volume.
Attack time	The time required for an applied signal that suddenly increases in amplitude to reach 1-(1/e), that is, 63.2% of its final, stable value.
Attenuate	To reduce in amplitude.
Attenuation characteristic	The decrease of signal amplitude versus frequency. It is usually expressed in decibels per octave.
Attenuation constant	See ATTENUATION CHARACTERISITIC.
Attenuator	A device for reducing signal amplitude in precise, predetermined steps, or smoothly over a continuous range.
Audibility	The quality of being able to be heard. In a healthy listener, the threshold of audibility is extremely low; at the threshold, the pressure of a sound wave varies from normal by approximately 10^{-4} pascals. The frequency range of human audibility extends roughly from 20 Hz to 20 kHz.
Audible	Hear or capable of being heard.
Boost	To increase the overall level of a signal.
Clipper	A circuit whose output voltage is fixed at a value for all input-voltages higher than a predetermined value. Clippers can flat-top the positive, negative, or both positive and negative peaks of an input voltage.
Clipping	1. Leveling off (flat-topping) a signal peak at a predetermined level. Also see CLIPPER. 2. In audio practice, the loss of syllables or words because of cutoff periods in the operation of the circuit (usually caused by overdriving a stage).
Compress	To reduce the bandwidth or dynamic range of a signal.
Compression	1. The reduction of output signal amplitude as input-signal amplitude rises. 2. Processing of a signal to increase low-level components and thereby raise the average power level. Usually a logarithmic function.
Compression ratio	In a system containing compression, the ratio A_1/A_2, where A_1 is the gain at a reference-signal level and A_2 is the gain at a specified higher signal level.
Compressor	A circuit or device which limits the amplitude of its output signal to a predetermined value in spite of wide variations in input signal amplitude. Unlike a CLIPPER a compressor does not flat-top output signal peaks.
Crest factor	For an AC wave, the ratio of the peak value to the rms value. The crest factor of a sine wave is equal to the square root of 2, or 1.4142126.

Cut	To reduce a signal's overall amplitude.
D to A	Digital to Analog. Conversion of a digital representation to an analog equivalent signal.
dBFS	Decibels with respect to digital full scale. The full scale amplitude (zero dBFS value) is the peak value of a sinewave whose positive or negative peak just reaches full scale.
dBm	dB relative to a reference value of 1 milliwatt. dBm is thus a power unit and requires knowledge of power levels (voltage and current, voltage and impedance, or current and impedance) rather than just voltage.
dBr	Relative dB; dB relative to an arbitrary reference value. The reference value must be stated for this to be a meaningful unit.
dBu	dB relative to a reference of 0.7746 Volts. Since 0.7746 Volts is the voltage across a 600 Ohm resistor when one milliwatt is being dissipated in that resistor, dBu and dBm are numerically equal in a 600 Ohm impedance environment.
dBV	dB relative to a reference value of 1 Volt.
Delay	The interval between the instant at which a signal or force is applied or removed and the instant at which a circuit or device subsequently responds in a specified manner.
Dither	The reduction of bits in a digital audio signal for use on a target system that uses a smaller number of bits than the originating system.
DSP	Digital Signal Processing. The art and science of digitally encoding analog signals producing raw data, performing one or more mathematical routines on the raw data to produce final data for use by other digital systems, and/or conversion back to analog signals.
Filter	A circuit or device which passes one frequency or frequency band while blocking others, or vice versa.
Filter attenuation	In a selective filter, the power, current, or voltage loss, in decibels, that takes place within the passband.
Filter attenuation band	The frequency band rejected by a band-suppression filter.
Filter cutoff	The frequency at which the transmission figure of a filter is below its maximum value by a prescribed amount (typically -3 dB).
Filter pass band	The frequency band defined by the upper and lower cutoff frequencies of a bandpass filter.
Filter stop band	The attenuation band of a band-suppression filter. See FILTER ATTENUATION BAND.
Gain	See BOOST

Gate	A device or circuit which has no output until it is triggered into operation by one or more enabling signals, or until an input signal exceeds a predetermined threshold amplitude.
Limiter	A device or circuit whose output-signal amplitude remains at some predetermined level in spite of wide variations of input-signal amplitude.
Limiting	The restriction of the maximum peak amplitude of a signal to a designated level.
Muting	The act of preventing signal from being output.
Phase shift	1. A change in the displacement, as a function of time, of a periodic disturbance having constant frequency. 2. The magnitude of such a change, measured in fractions of a wavelength or in electrical degrees.
Polarity	See PHASE SHIFT
QuickMAP	The diagram outlining the processing components in a particular processing setup.
QuickSET	The group of component settings associated with a QuickMAP.
RS-232	A standard covering serial communications between computers and other equipment.
Serial cable	A cable used to connect between two serial ports. The connectors used are typically DB-9, DB-25, or RJ-45.
Serial port	A communications port located either on a computer or on computer controlled equipment.
Threshold	1. The initial (observable) point of an effect, e.g., threshold of hearing. 2. A predetermined point, such as a minimum current or voltage, for the start of operation of a circuit or device.
Window	An interval during which a circuit is gated open to permit signal sampling. During this interval, a window is figuratively OPEN to the signal.

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STANDARD QUICKMAPS™

VUE-IT ships with eighteen standard QuickMAPs. These mappings were designed to be flexible enough to be applied in a wide variety of signal processing applications. This appendix provides drawings of the QuickMAPs for reference purposes. The drawings begin on page B-2.

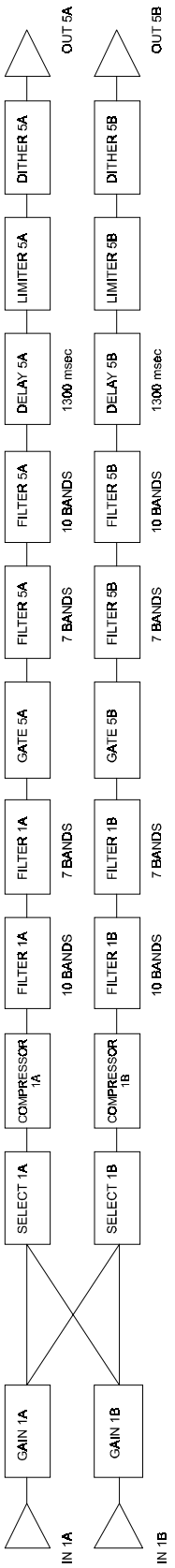
Identification of Slots and Connectors

The slots and connectors used are indicated by either the word IN in the case of an input or OUT in the case of an output. In both cases the word IN or OUT is followed by a number (1-5) and a letter (A or B). The number indicates the slot which is used and the letter indicates the connector used. For example: IN 1A would indicate that it is an input using connector A of slot 1.

Applying QuickMAPs

When looking for a QuickMAP that fits your application it is important that you keep in mind that not all portions of a QuickMAP must be used. It is possible to load a QuickMAP that requires more cards than you have or need. The slots that do not contain cards may give an error indication (yellow flashing LED). As long as the slots corresponsiding with the portion(s) of the QuickMAP you are using are filled there will not be a problem.

MONDOEQ.QMS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GAIN 1A
GAIN 1B
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SELECT 1B
COMPRESSOR 1A
COMPRESSOR 1B
FILTER 1A
FILTER 1B
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FILTER 1A
FILTER 1B
7 BANDS
GATE 5A
GATE 5B
FILTER 5A
FILTER 5B
7 BANDS
FILTER 5A
FILTER 5B
10 BANDS
DELAY 5A
DELAY 5B
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LIMITER 5A
LIMITER 5B
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DITHER 5B
OUT 5A
OUT 5B

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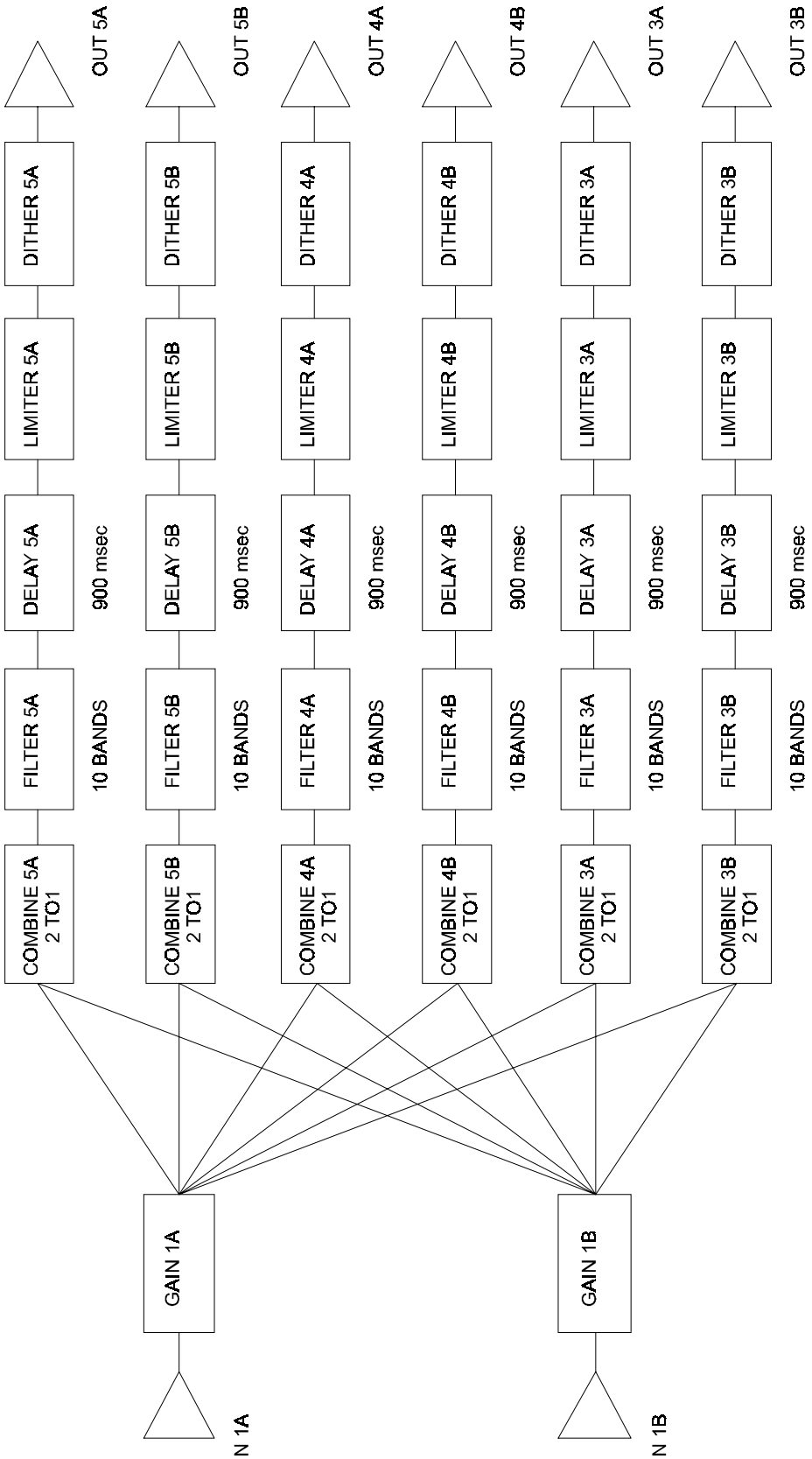
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GAIN 1A
GAIN 1B
COMBINE 5A 2 TO1
COMBINE 5B 2 TO1
COMBINE 4A 2 TO1
COMBINE 4B 2 TO1
COMBINE 3A 2 TO1
COMBINE 3B 2 TO1
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
10 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
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OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

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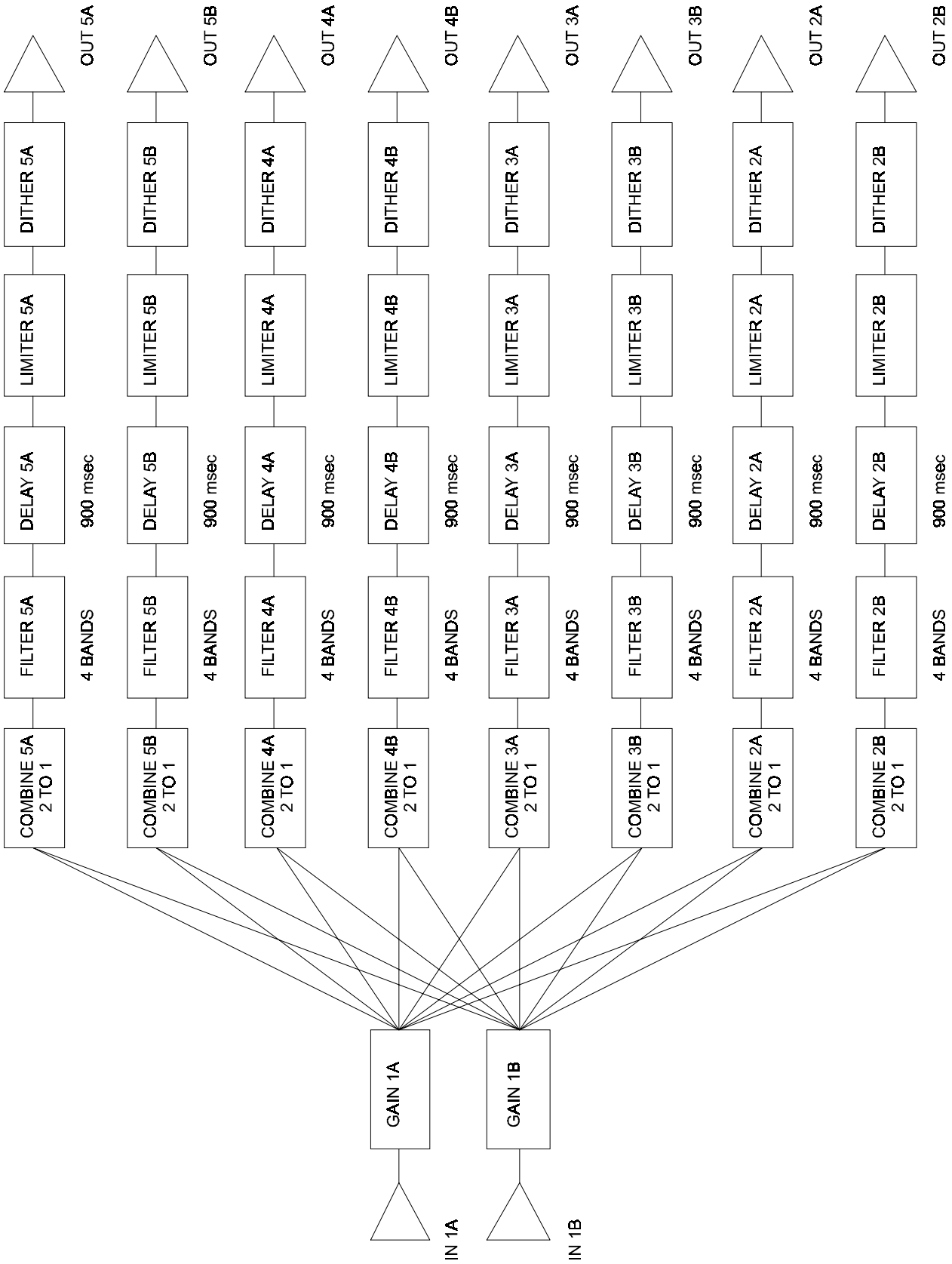
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IN 1B
GAIN 1A
GAIN 1B
COMBINE 5A 2 TO 1
COMBINE 5B 2 TO 1
COMBINE 4A 2 TO 1
COMBINE 4B 2 TO 1
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COMBINE 3B 2 TO 1
COMBINE 2A 2 TO 1
COMBINE 2B 2 TO 1
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
FILTER 2A
FILTER 2B
4 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
DELAY 2A
DELAY 2B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
LIMITER 2A
LIMITER 2B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
DITHER 2A
DITHER 2B
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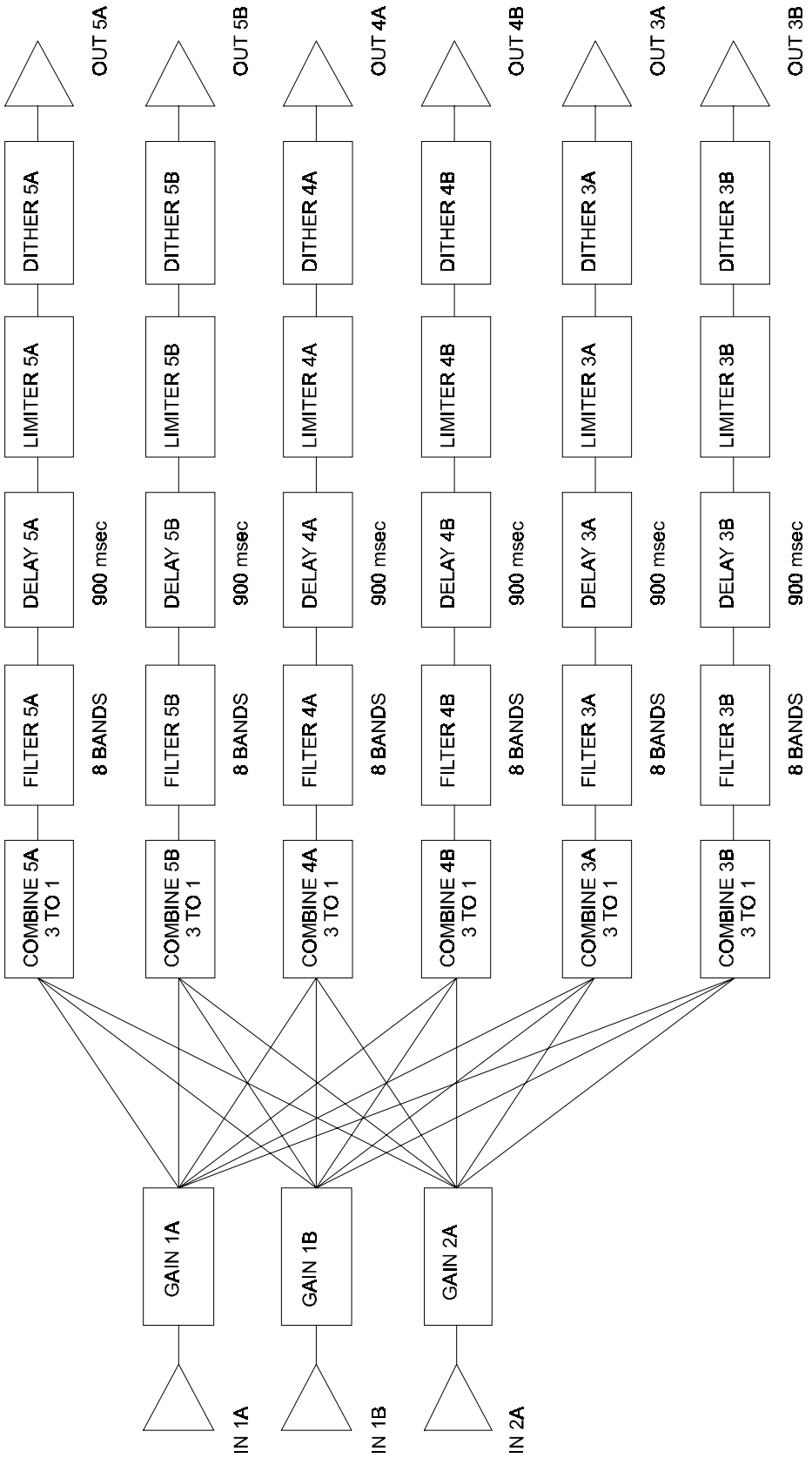
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
OUT 2A
OUT 2B

3X6CMBC.QMS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IN 1B
IN 2A
GAIN 1A
GAIN 1B
GAIN 2A
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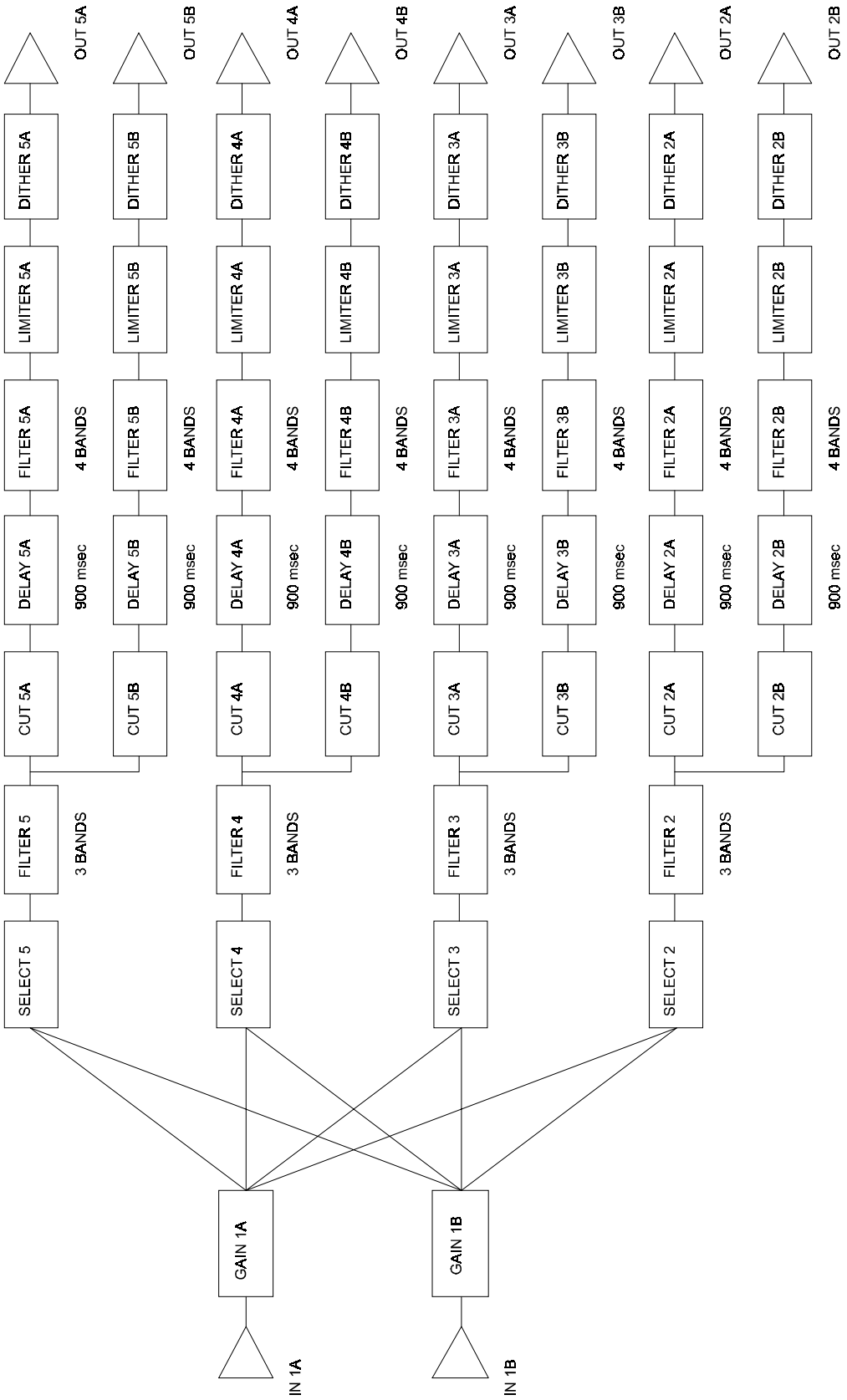
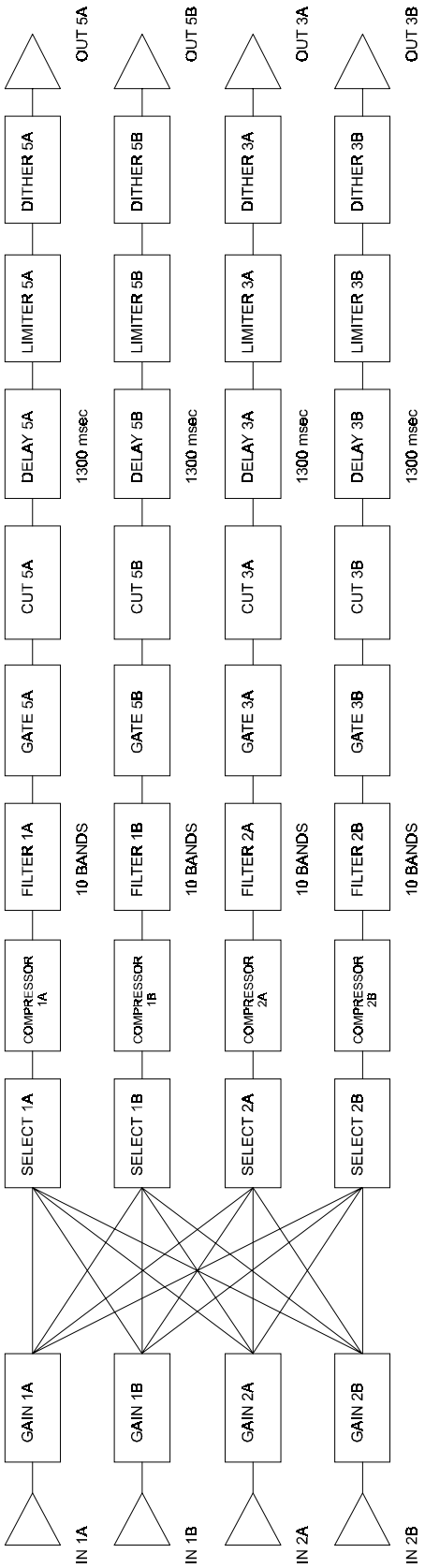
COMBINE 5B 3 TO 1
COMBINE 4A 3 TO 1
COMBINE 4B 3 TO 1
COMBINE 3A 3 TO 1
COMBINE 3B 3 TO 1
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
8 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

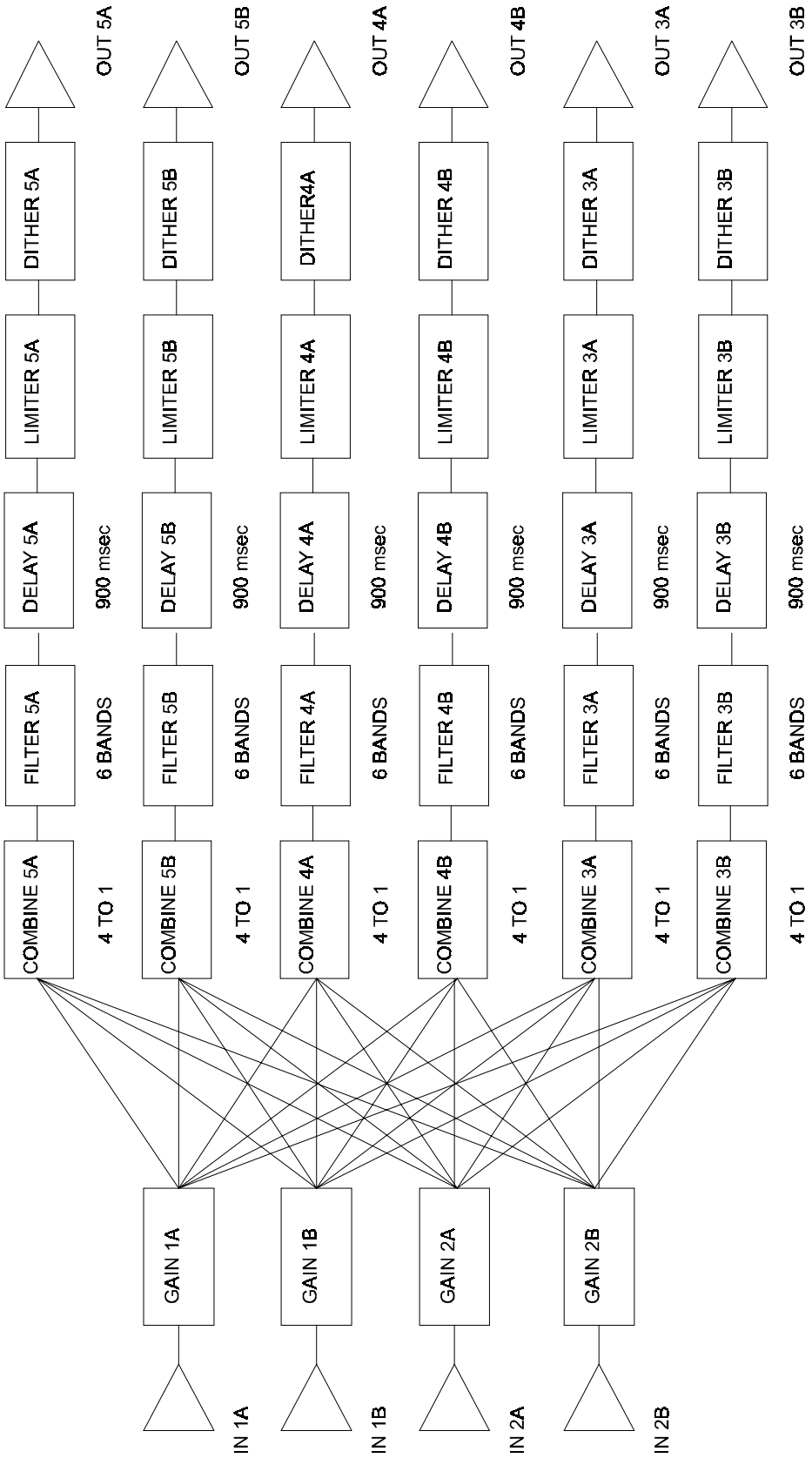
2X8THRU.QMS


IN 1A
IN 1B
GAIN 1A
GAIN 1B
SELECT 5
SELECT 4
SELECT 3
SELECT 2
FILTER 5
FILTER 4
FILTER 3
FILTER 2
3 BANDS
CUT 5A
CUT 5B
CUT 4A
CUT 4B
CUT 3A
CUT 3B
CUT 2A
CUT 2B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
DELAY 2A
DELAY 2B
900 msec
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
FILTER 2A
FILTER 2B
4 BANDS
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
LIMITER 2A
LIMITER 2B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
DITHER 2A
DITHER 2B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
OUT 2A
OUT 2B

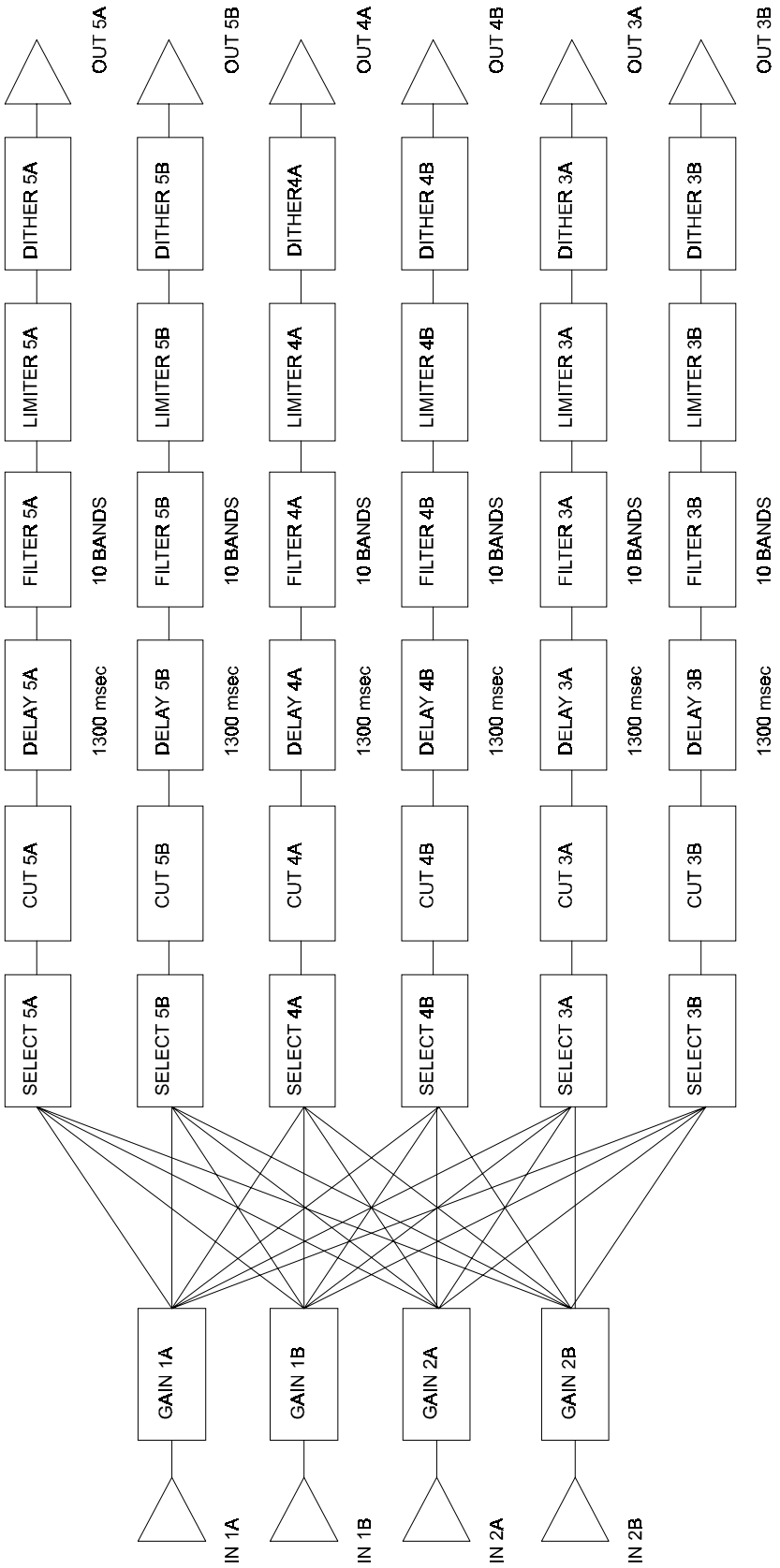
4CHAN.QMS


IN 1A
IN 1B
IN 2A
IN 2B
GAIN 1A
GAIN 1B
GAIN 2A
GAIN 2B
SELECT 1A
SELECT 1B
SELECT 2A
SELECT 2B
COMPRESSOR 1A
COMPRESSOR 1B
COMPRESSOR 2A
COMPRESSOR 2B
FILTER 1A
FILTER 1B
FILTER 2A
FILTER 2B
10 BANDS
10 BANDS
10 BANDS
10 BANDS
GATE 5A
GATE 5B
GATE 3A
GATE 3B
CUT 5A
CUT 5B
CUT 3A
CUT 3B
DELAY 5A
DELAY 5B
DELAY 3A
DELAY 3B
1300 msec
1300 msec
1300 msec
1300 msec
LIMITER 5A
LIMITER 5B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 3A
OUT 3B

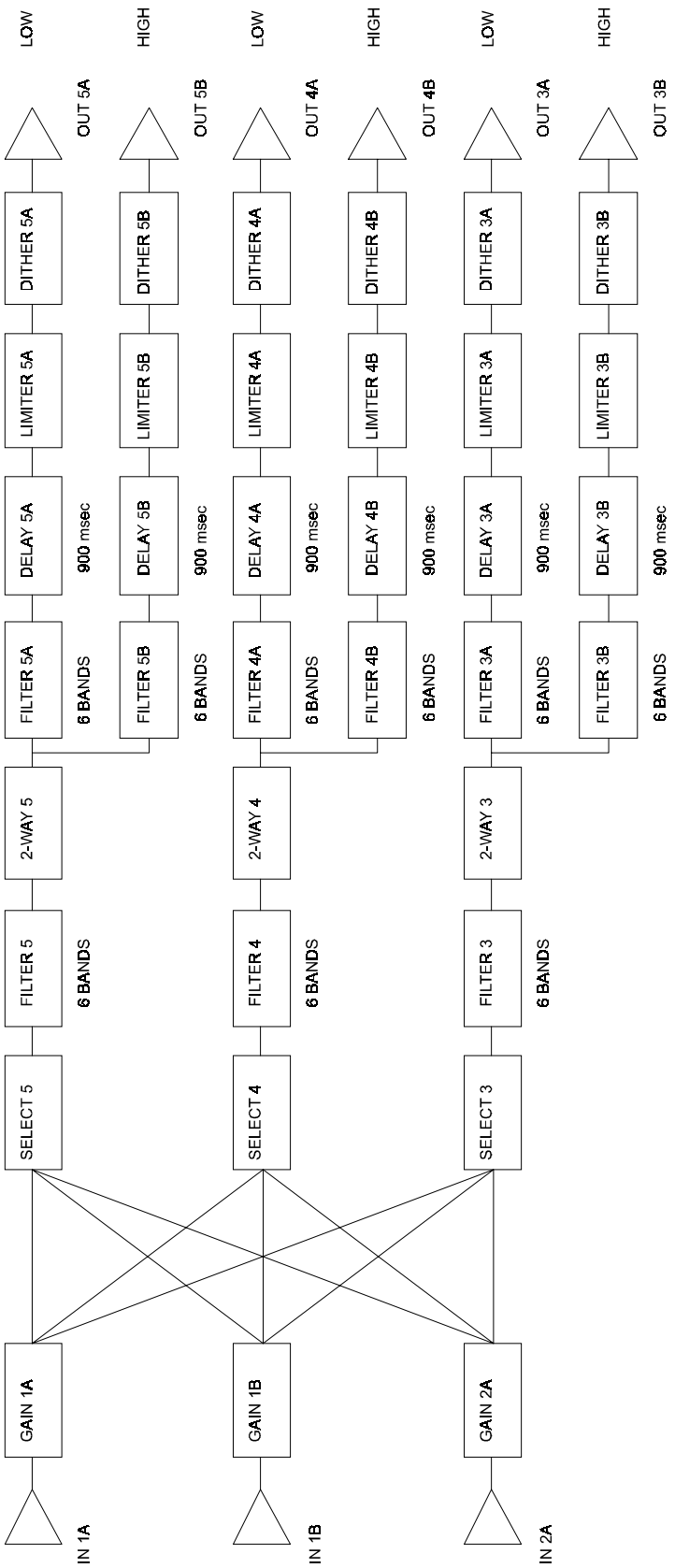
4X6CMBC.QMS


IN 1A
IN 1B
IN 2A
IN 2B
GAIN 1A
GAIN 1B
GAIN 2A
GAIN 2B
COMBINE 5A
COMBINE 5B
COMBINE 4A
COMBINE 4B
COMBINE 3A
COMBINE 3B
4 TO 1
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
6 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER4A
DITHER 4B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

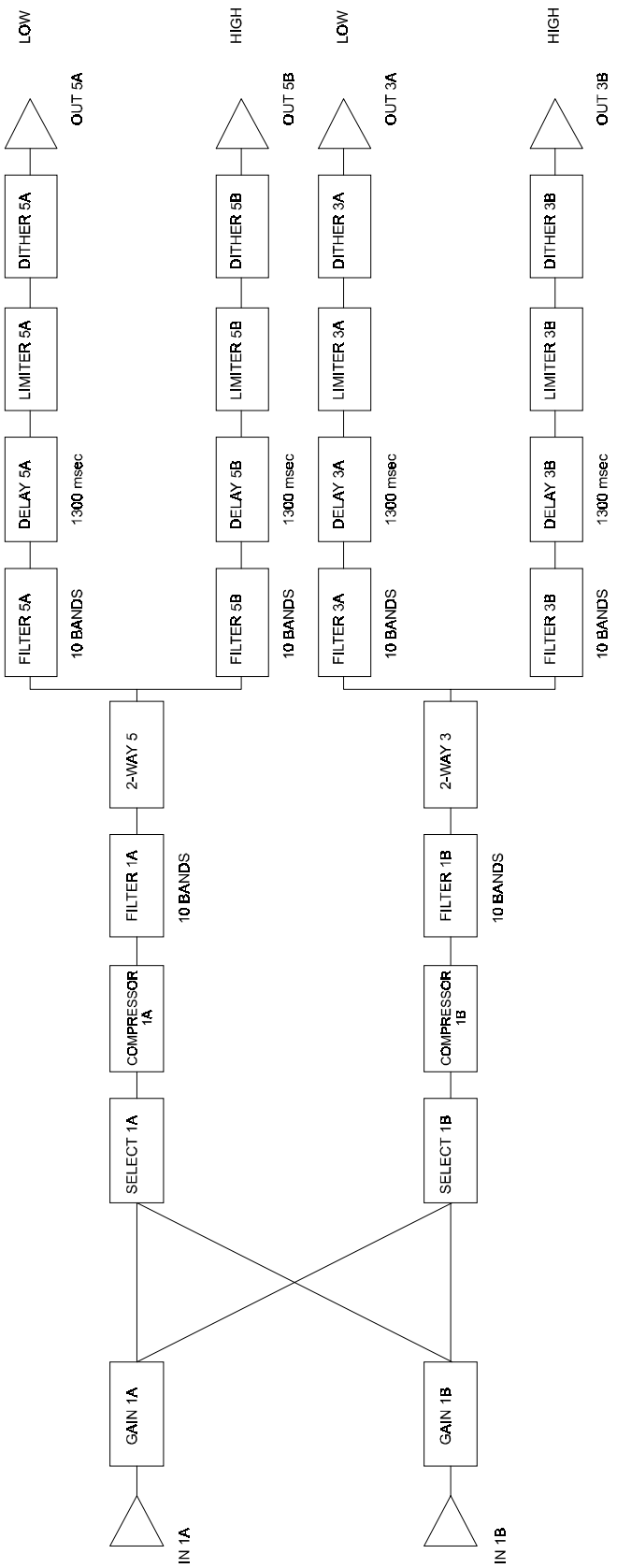
4X6THRU.QMS


IN 1A
IN 1B
IN 2A
IN 2B
GAIN 1A
GAIN 1B
GAIN 2A
GAIN 2B
SELECT 5A
SELECT 5B
SELECT 4A
SELECT 4B
SELECT 3A
SELECT 3B
CUT 5A
CUT 5B
CUT 4A
CUT 4B
CUT 3A
CUT 3B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
1300 msec
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
10 BANDS
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER4A
DITHER 4B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

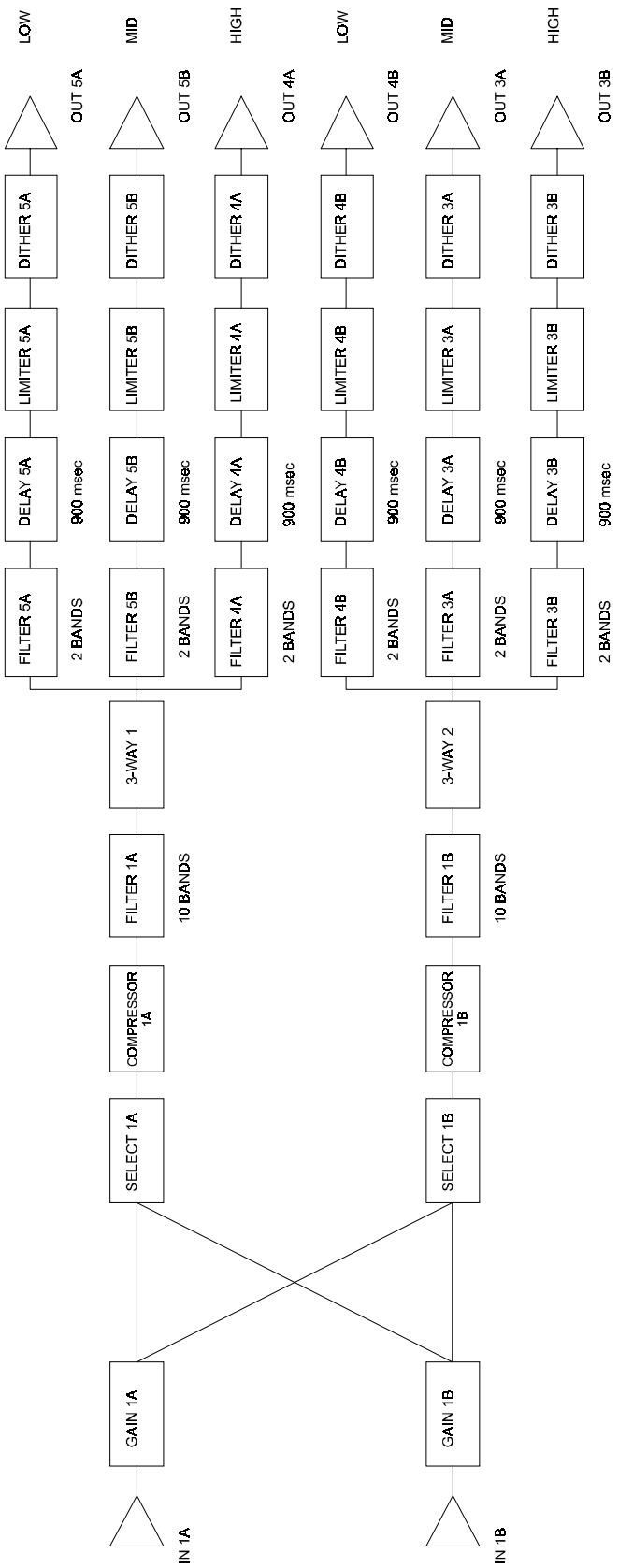
3_2W.QMS


LOW
HIGH
LOW
HIGH
LOW
HIGH
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
900 msec
900 msec
900 msec
900 msec
900 msec
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
6 BANDS
6 BANDS
6 BANDS
6 BANDS
6 BANDS
6 BANDS
2-WAY 5
2-WAY 4
2-WAY 3
FILTER 5
FILTER 4
FILTER 3
6 BANDS
6 BANDS
6 BANDS
SELECT 5
SELECT 4
SELECT 3
GAIN 1A
GAIN 1B
GAIN 2A
IN 1A
IN 1B
IN 2A

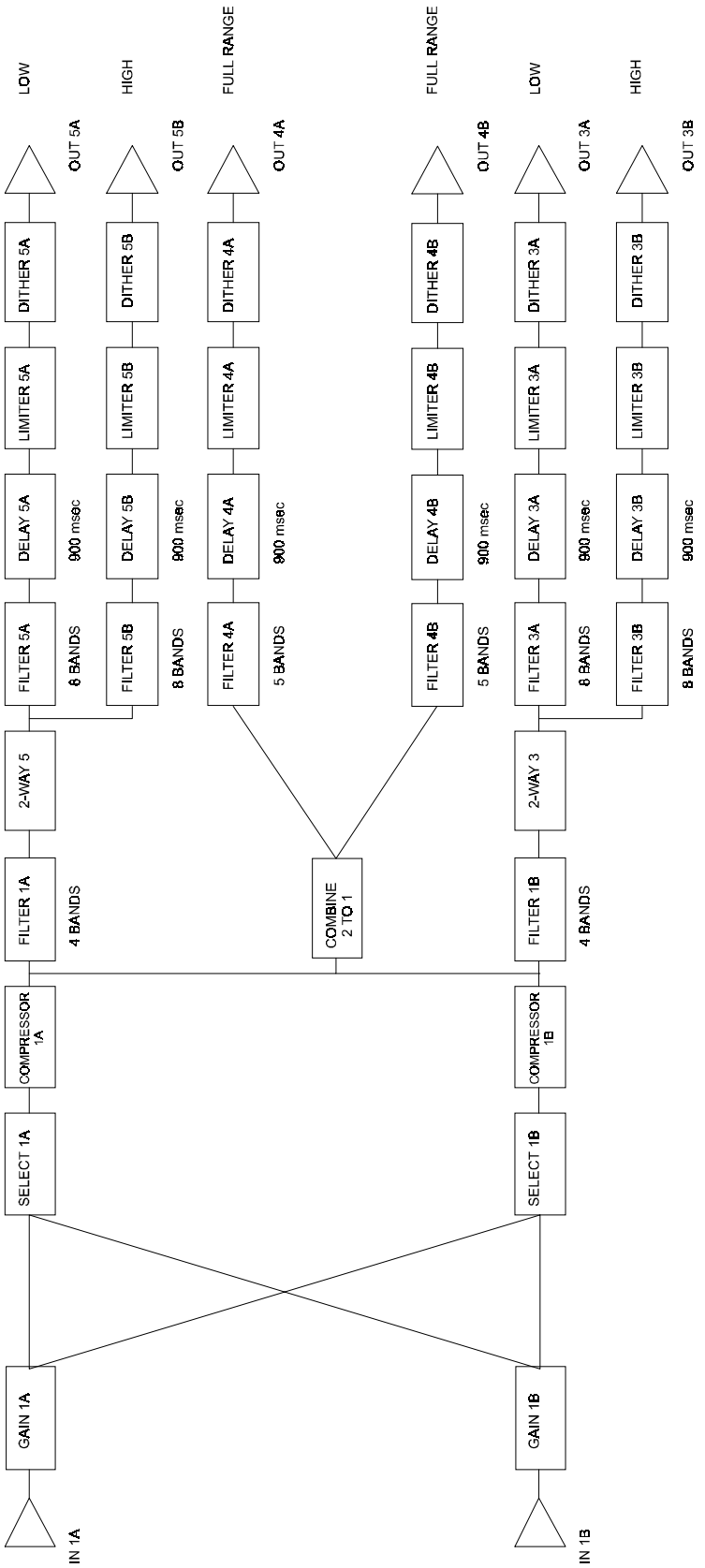
2WAYS.QMS


LOW
OUT 5A
DITHER 5A
LIMITER 5A
DELAY 5A
1300 msec
FILTER 5A
10 BANDS
HIGH
OUT 5B
DITHER 5B
LIMITER 5B
DELAY 5B
1300 msec
FILTER 5B
10 BANDS
LOW
OUT 3A
DITHER 3A
LIMITER 3A
DELAY 3A
1300 msec
FILTER 3A
10 BANDS
HIGH
OUT 3B
DITHER 3B
LIMITER 3B
DELAY 3B
1300 msec
FILTER 3B
10 BANDS
2-WAY 5
FILTER 1A
10 BANDS
COMPRESSOR 1A
SELECT 1A
2-WAY 3
FILTER 1B
10 BANDS
COMPRESSOR 1B
SELECT 1B
GAIN 1A
GAIN 1B
IN 1A
IN 1B

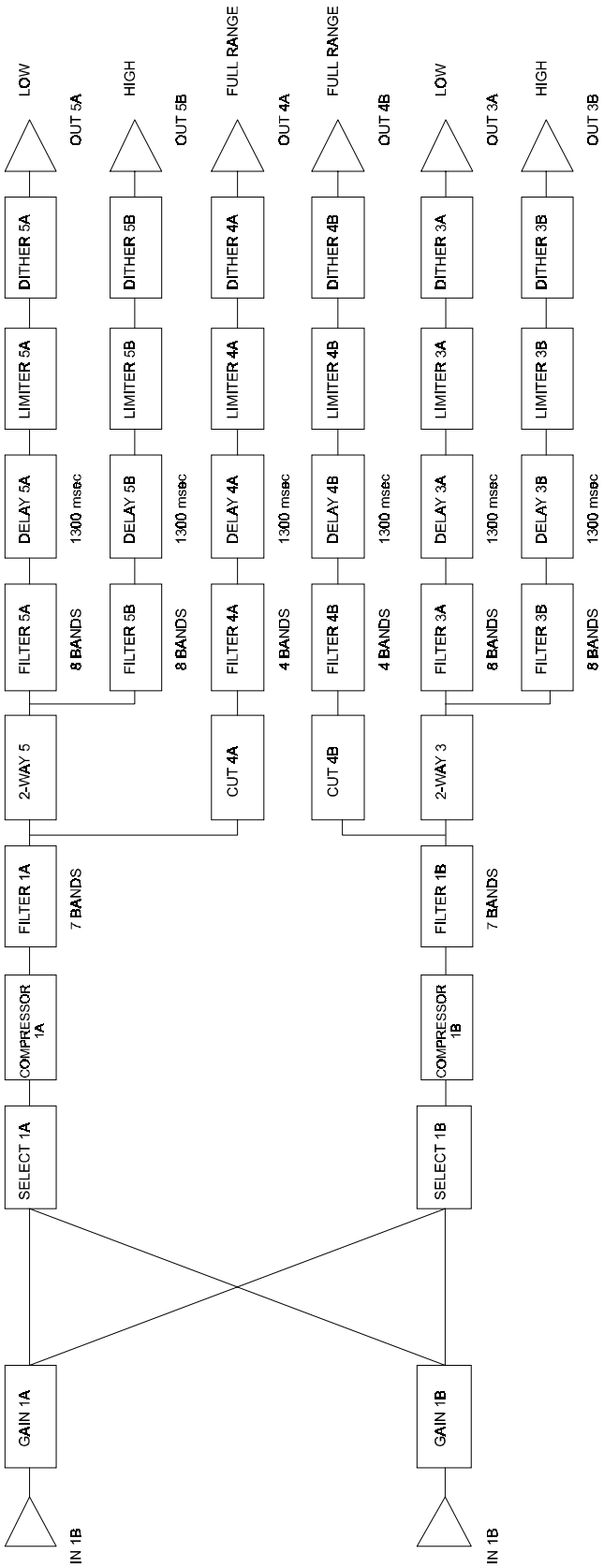
2_3WAY.QMS


IN 1A
IN 1B
GAIN 1A
GAIN 1B
SELECT 1A
SELECT 1B
COMPRESSOR 1A
COMPRESSOR 1B
FILTER 1A
FILTER 1B
10 BANDS
3-WAY 1
3-WAY 2
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
2 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
LOW
MID
HIGH
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

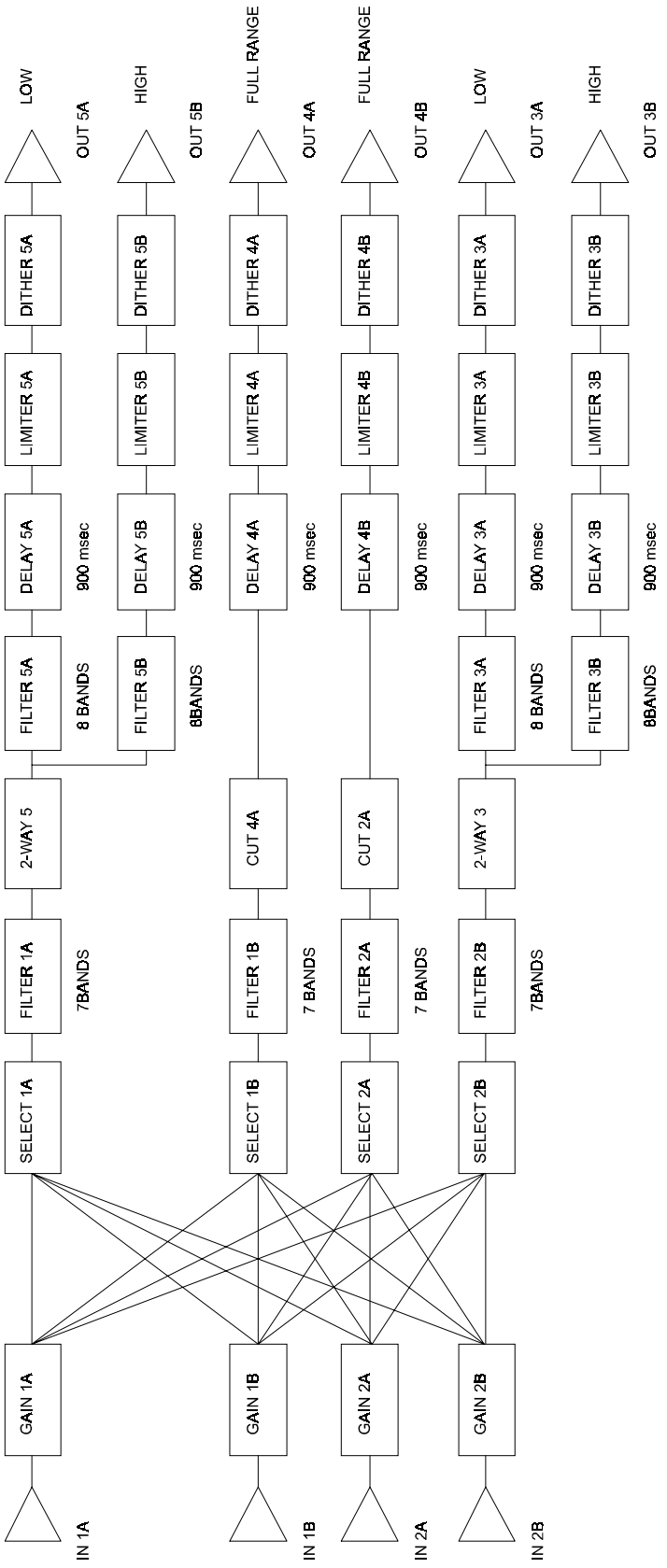
2_2W_SUB.QMS


LOW
HIGH
FULL RANGE
FULL RANGE
LOW
HIGH
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
900 msec
900 msec
900 msec
900 msec
900 msec
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
6 BANDS
8 BANDS
5 BANDS
5 BANDS
6 BANDS
8 BANDS
2-WAY 5
2-WAY 3
FILTER 1A
FILTER 1B
4 BANDS
4 BANDS
COMBINE
2 TO 1
COMPRESSOR
1A
COMPRESSOR
1B
SELECT 1A
SELECT 1B
GAIN 1A
GAIN 1B
IN 1A
IN 1B

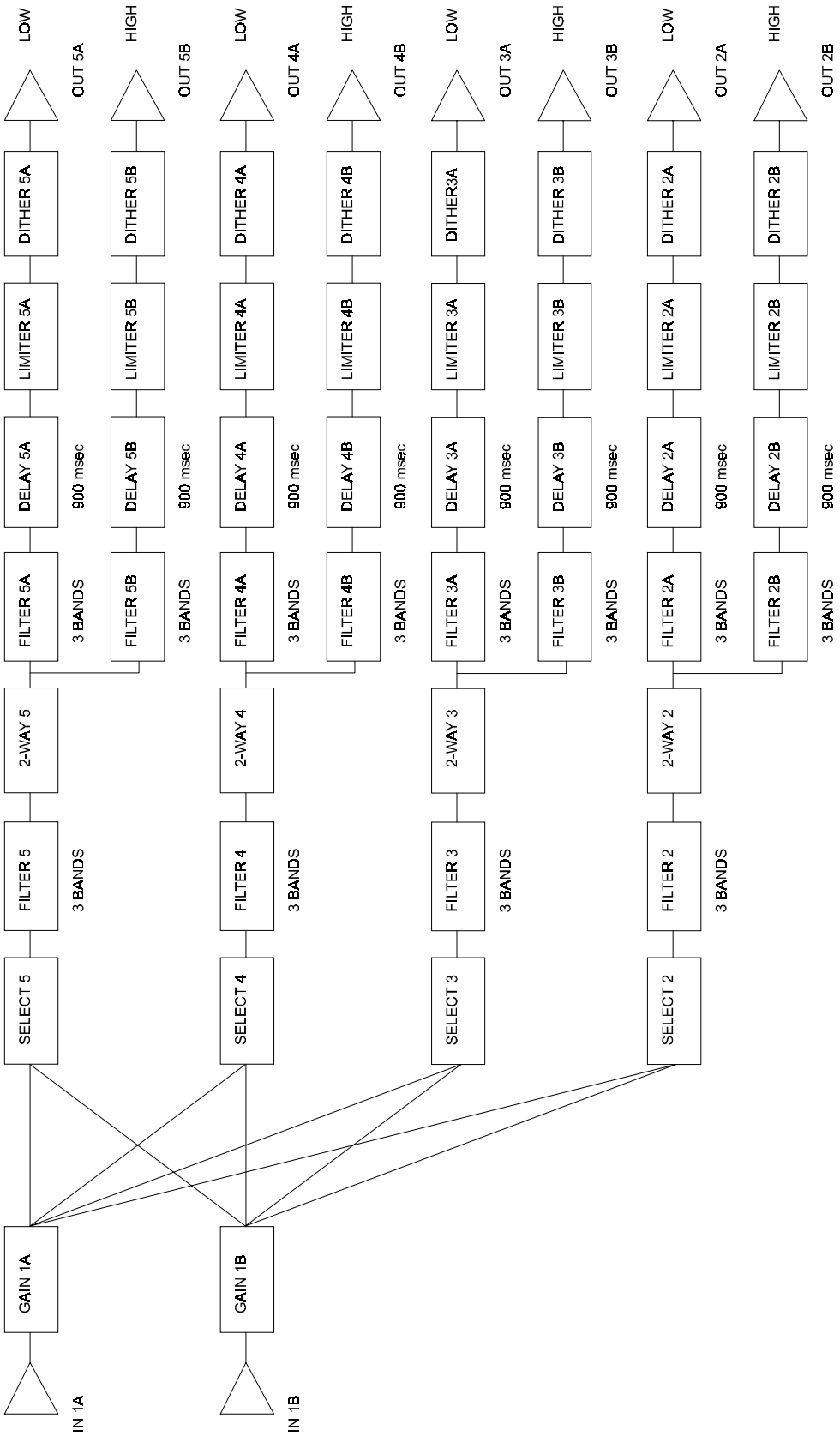
2_2W_FR.QMS


IN 1B
GAIN 1A
SELECT 1A
COMPRESSOR 1A
FILTER 1A
7 BANDS
2-WAY 5
FILTER 5A
8 BANDS
DELAY 5A
1300 msec
LIMITER 5A
DITHER 5A
OUT 5A
LOW
FILTER 5B
8 BANDS
DELAY 5B
1300 msec
LIMITER 5B
DITHER 5B
OUT 5B
HIGH
CUT 4A
FILTER 4A
4 BANDS
DELAY 4A
1300 msec
LIMITER 4A
DITHER 4A
OUT 4A
FULL RANGE
CUT 4B
FILTER 4B
4 BANDS
DELAY 4B
1300 msec
LIMITER 4B
DITHER 4B
OUT 4B
FULL RANGE
IN 1B
GAIN 1B
SELECT 1B
COMPRESSOR 1B
FILTER 1B
7 BANDS
2-WAY 3
FILTER 3A
8 BANDS
DELAY 3A
1300 msec
LIMITER 3A
DITHER 3A
OUT 3A
LOW
FILTER 3B
8 BANDS
DELAY 3B
1300 msec
LIMITER 3B
DITHER 3B
OUT 3B
HIGH

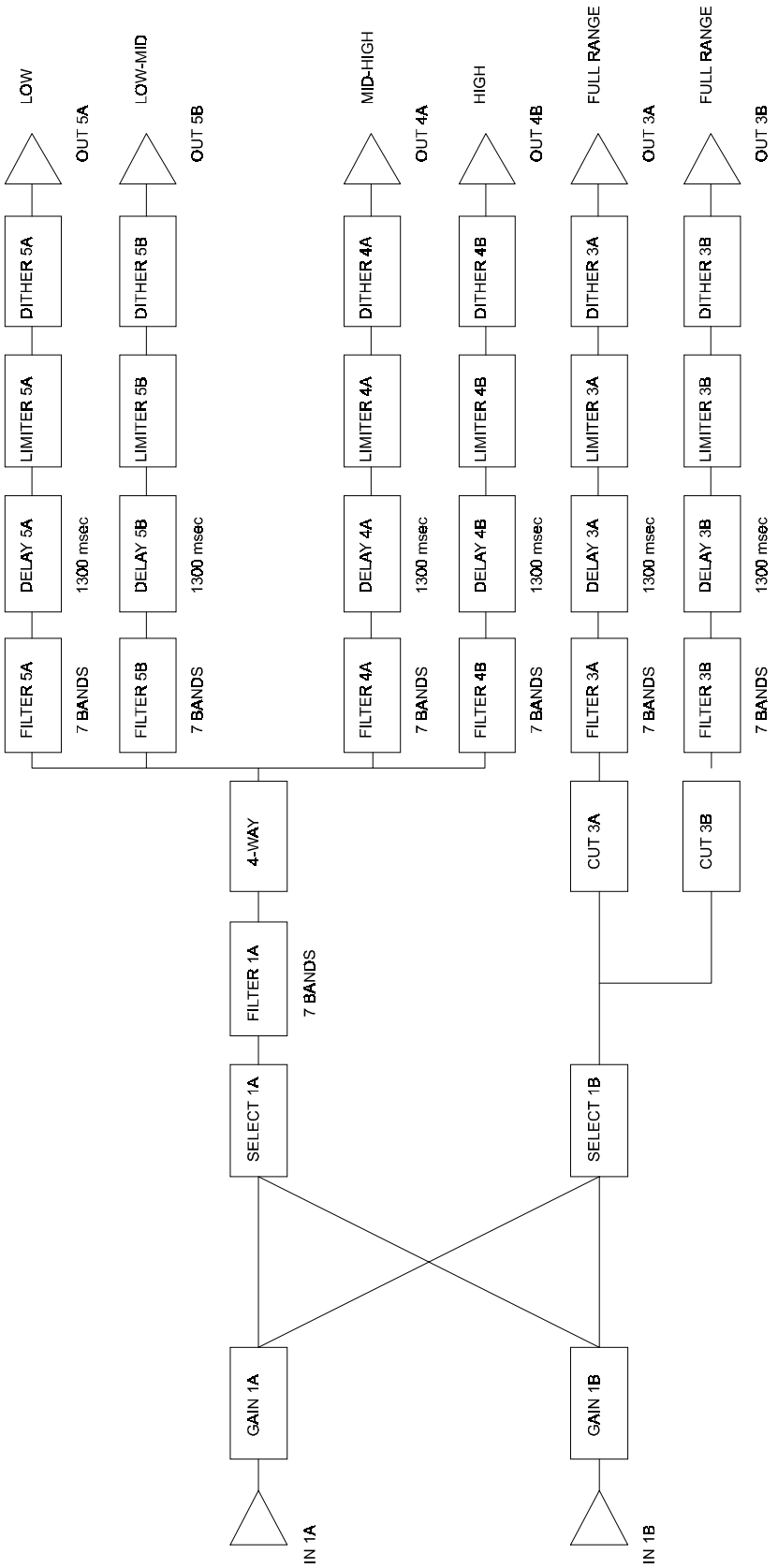
2_2W_2ST.QMS


IN 1A
IN 1B
IN 2A
IN 2B
GAIN 1A
GAIN 1B
GAIN 2A
GAIN 2B
SELECT 1A
SELECT 1B
SELECT 2A
SELECT 2B
FILTER 1A
FILTER 1B
FILTER 2A
FILTER 2B
7 BANDS
2-WAY 5
CUT 4A
CUT 2A
2-WAY 3
FILTER 5A
FILTER 5B
FILTER 3A
FILTER 3B
8 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
LOW
HIGH
FULL RANGE
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B

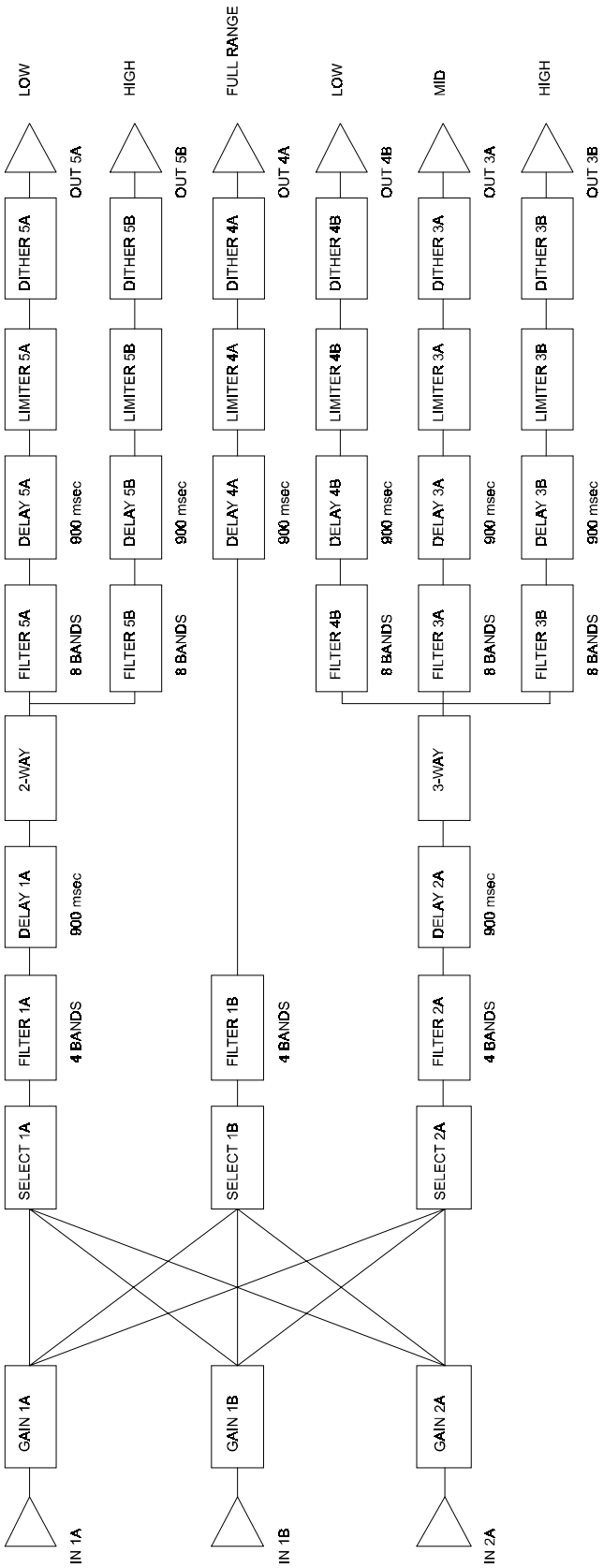
4_2WAYS.QMS


IN 1A
IN 1B
GAIN 1A
GAIN 1B
SELECT 5
SELECT 4
SELECT 3
SELECT 2
FILTER 5
FILTER 4
FILTER 3
FILTER 2
3 BANDS
2-WAY 5
2-WAY 4
2-WAY 3
2-WAY 2
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
FILTER 2A
FILTER 2B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
DELAY 2A
DELAY 2B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
LIMITER 2A
LIMITER 2B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER3A
DITHER 3B
DITHER 2A
DITHER 2B
LOW
HIGH
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
OUT 2A
OUT 2B

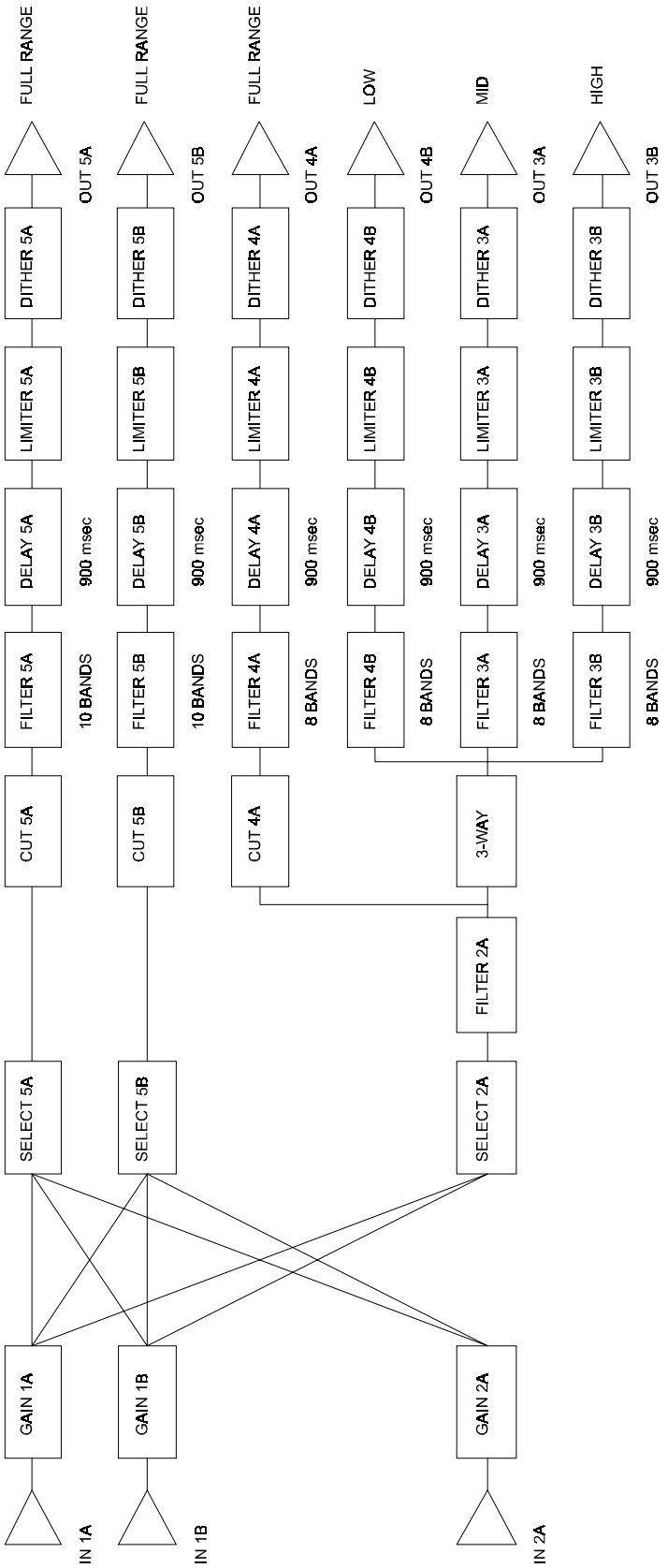
4W_2FR.QMS


LOW
OUT 5A
LOW-MID
OUT 5B
MID-HIGH
OUT 4A
HIGH
OUT 4B
FULL RANGE
OUT 3A
FULL RANGE
OUT 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
1300 msec
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
7 BANDS
4-WAY
CUT 3A
CUT 3B
FILTER 1A
SELECT 1A
SELECT 1B
GAIN 1A
GAIN 1B
IN 1A
IN 1B

LCR.QMS


IN 1A
IN 1B
IN 2A
GAIN 1A
GAIN 1B
GAIN 2A
SELECT 1A
SELECT 1B
SELECT 2A
FILTER 1A
4 BANDS
FILTER 1B
4 BANDS
FILTER 2A
4 BANDS
DELAY 1A
900 msec
DELAY 2A
900 msec
2-WAY
3-WAY
FILTER 5A
8 BANDS
FILTER 5B
8 BANDS
FILTER 4B
8 BANDS
FILTER 3A
8 BANDS
FILTER 3B
8 BANDS
DELAY 5A
900 msec
DELAY 5B
900 msec
DELAY 4A
900 msec
DELAY 4B
900 msec
DELAY 3A
900 msec
DELAY 3B
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
LOW
HIGH
FULL RANGE
LOW
MID
HIGH

MONO3W.QMS


IN 1A
IN 1B
IN 2A
GAIN 1A
GAIN 1B
GAIN 2A
SELECT 5A
SELECT 5B
SELECT 2A
CUT 5A
CUT 5B
CUT 4A
3-WAY
FILTER 2A
FILTER 5A
FILTER 5B
FILTER 4A
FILTER 4B
FILTER 3A
FILTER 3B
10 BANDS
10 BANDS
8 BANDS
8 BANDS
8 BANDS
8 BANDS
DELAY 5A
DELAY 5B
DELAY 4A
DELAY 4B
DELAY 3A
DELAY 3B
900 msec
900 msec
900 msec
900 msec
900 msec
900 msec
LIMITER 5A
LIMITER 5B
LIMITER 4A
LIMITER 4B
LIMITER 3A
LIMITER 3B
DITHER 5A
DITHER 5B
DITHER 4A
DITHER 4B
DITHER 3A
DITHER 3B
OUT 5A
OUT 5B
OUT 4A
OUT 4B
OUT 3A
OUT 3B
FULL RANGE
FULL RANGE
FULL RANGE
LOW
MID
HIGH

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