PCAP II *Plus*

ENHANCED-CAPABILITY PERSONAL COMPUTER AUDIO PROCESSOR

USER'S MANUAL

DIGITAL AUDIO CORPORATION

A DRI COMPANY

PCAP II *Plus*

ENHANCED-CAPABILITY PERSONAL COMPUTER AUDIO PROCESSOR

User's Manual

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DIGITAL AUDIO CORPORATION
A DRI COMPANY

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ACKNOWLEDGEMENT

The Enhanced-Capability Personal Computer Audio Processor, model PCAP II *Plus,* is a fourth-generation PC-based digital audio filtering system. Like its predecessors, the original PCDF4096, PCAP, and MCAP products, this product was inspired by Mr. James “Jim” Foye of the United States Postal Service and Mr. John “Jack” Losinski of the Federal Bureau of Investigation. Both Jim and Jack have made numerous helpful suggestions that have been applied to the products during both the original technical development and their use in various applications over the years. For their invaluable contributions Digital Audio Corporation is most grateful.

WHAT'S NEW?

In addition to the base capabilities of the previous-generation PCAP II product, the PCAP II *Plus* system offers the following enhanced capabilities:

- More Processing Power
  - Two additional general filter stages now available (increased to 6 from 4)
  - Two new broadband filter stages now available
  - 50% increase in adaptive filter taps (increased to 6144 from 4096)
  - New 1.8GHz floating-point processing core for implementing enhanced filtering functions (e.g., broadband noise reduction) and future software upgrades

- Improved Spectrum Analyzer
  - 6x improvement in screen refresh rate

- Re-engineered Graphical User Interface
  - More intuitive look and feel
  - Improved visual status indications
  - Quicker response on all controls, graphs, and indicators
  - Direct "text box" numeric entry on all filtering parameters
  - Higher efficiency for improved productivity

- New Broadband Noise-Reduction Filters
  - NoiseEQ™ frequency-selective spectral subtraction filter
  - NoiseReducer general spectral subtraction filter
  - Both effective at reducing difficult random noises such as waterflow, wind, hiss, RF static, and GSM RF interference coupling

- Enhanced Digital Media Processing
  - AutoSYNC feature on digital audio I/O allows easy integration with PC-based audio workstations such as Digidesign Pro Tools® for direct processing of digital audio evidence
  - Supports all 16- and 24-bit, consumer and professional audio formats up to 96kHz sample rate

FOREWORD

The PCAP II *Plus* Enhanced-Capability Personal Computer Audio Processor is the latest generation of the most popular forensic audio analysis and processing system used by the law-enforcement and intelligence communities. The PCAP II *Plus* offers the following advantages over typical PC-based audio processors:

- Designed from the ground-up to implement generally-accepted forensic voice enhancement / clarification principles and methodologies; not originally designed as a commercial audio post-production, "sweetening", or phonograph audio restoration tool

- Powerful, interactive, real-time noise reduction processing using the latest DSP technology

- Compact, field-deployable packaging

- Direct digital audio input/output with AutoSYNC; interfaces directly with Digidesign Pro Tools® systems for direct processing of file-based digital audio media

- Both monaural and stereo signal processing

- Up to 12 sequential stages of enhancement processing

- All Windows™-based control: versatile configuration and easy, interactive adjustment

- All signal processing contained in an external unit; interfaces to PC serial port with a single cable

- Laptop or desktop computer control

- Built-in dual channel FFT spectrum analyzer

- Built-in digital filter coefficient display

- Disk file storage/recall of filter setups

- Stand-alone operation allowing the hardware processor to be operated at desired settings without PC

- Report generator for hardcopy printouts and word processor files of filter setups

- Unmatched flexibility and performance

The PCAP II *Plus* is applicable to a broad spectrum of voice and similar audio signals. It attacks a wide variety of noises in forensic applications. Body microphone, cassette, microcassette, telephone, broadcast, and hi-fi audio signals can all be processed efficiently, since the PCAP II can be set to operate at bandwidths of 3.2, 5.4, 6.5, 8.0, 11, and 16 kHz.

The PCAP II is designed to replace an entire rack of audio processing equipment. With monophonic signals, as many as 12 sequential stages of digital processing can be performed simultaneously. With stereo signals, nine sequential stages of stereo processing are available. A broad selection of filter types/options is available in each of the processing stages.

An easy-to-use Microsoft Windows-based Master Control program allows intuitive control of the entire process, including bandwidth selection, mono/stereo configuration, and number of digital processing stages. Individual process controls and filter modes for each digital stage are easily specified. Input and output level bargraphs are displayed, and the signal frequency spectrum at any point in the process can be displayed utilizing the built-in spectrum analyzer. Also, the built-in coefficient display allows the filter solutions for all adaptive filters to be viewed with ease.

1: . SYSTEM BASICS

1.1: SYSTEM CONFIGURATION

The basic configuration of the PCAP II *Plus* system is illustrated in the following (Figure 1-1):

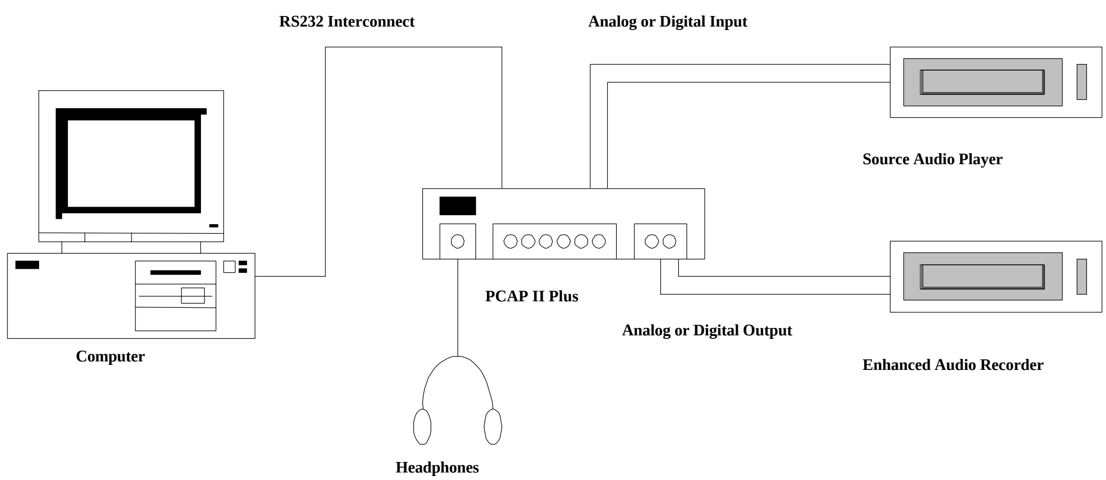

Figure 1-1 PCAP II *Plus* Basic System Configuration

For integration with a Digidesign Pro Tools system (e.g., Model Digi002), an alternative configuration is illustrated in the following (Figure 1-2):

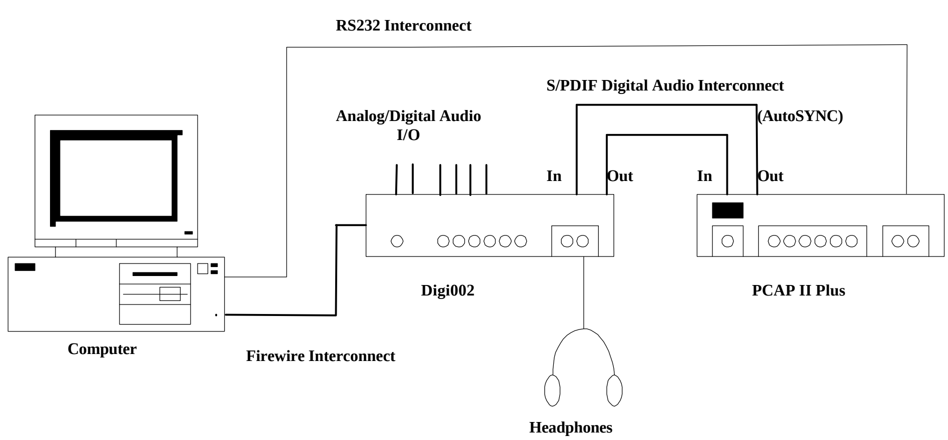

Figure 1-2 PCAP II *Plus* Alternative Pro Tools® Integration

Note that in this configuration the digital OUTPUT RATE should be configured for AutoSYNC for proper digital audio interaction with Pro Tools.

The PCAP II *Plus* Master Control program is written to be run on Windows® 98 Second Edition, Me, NT 4.0, 2000, and XP. For best performance the following *minimum* system configuration is recommended:

- Windows® XP operating system
- Intel Pentium® III CPU processor (at least 800 MHz)
- 256 Megabytes of RAM
- 10GB hard disk drive
- CD-ROM Drive
- 1024x768 color SVGA or LCD active-matrix display with 0.28 or better dot pitch
- Two-button mouse and/or touchpad
- At least one spare RS232 port
- Laser or dot-matrix printer

Performance will improve with higher speed CPUs and/or increased memory.

Please visit the Digidesign website at http://www.digidesign.com/ for complete information regarding computer/OS compatibility for Pro Tools products. Integration of the PCAP II *Plus* product with Pro Tools will require a computer platform that meets BOTH the requirements specified above and Digidesign's requirements.

1.2: EXTERNAL PROCESSOR CAPABILITY

The PCAP II *Plus EXTERNAL PROCESSOR* unit is a high-performance, self-contained digital signal processor and contains 38 DSP microprocessors, which are allocated as follows:

- 24 FIR filter processors which can be configured as 1 to 6 independent audio processors. These flexible processors may be combined into mono and stereo configurations. Any of these processors may be configured as an adaptive or adjustable digital FIR filter

- Two FIR spectral equalizer processors

- Four sample rate conversion processors

- Four FIR highpass filter processors

- Three general-purpose microprocessors which execute special DSP software, implement the dual FFT spectrum analyzer, communicate with the PC via RS232, and provide system control

- One high-performance floating-point DSP processor that performs broadband noise reduction functions and other advanced algorithms

Analog-to-digital and digital-to-analog conversion is performed by stereo, 24-bit, sigma-delta converters which perform 64x oversampling.

The base processing sample rate is adjustable from 7.2 kHz (3.2 kHz bandwidth) to 36 kHz (16 kHz bandwidth), ***regardless of the input sample rate when the digital input is used (sample rates from 25-108kHz are supported via asynchronous conversion)***. All processing sample rates are exact multiples of 50 Hz and 60 Hz, allowing maximum filter performance at harmonics of these frequencies.

Digital, 200Hz highpass filters are provided on all inputs (pre-process) and on all outputs (post-process) to remove rumble and other low-frequency noises.

Microprocessor-controlled limiters are provided on both analog inputs to prevent overload. When using the S/PDIF digital inputs, the limiters prevent the digital signal from exceeding a specified dB level.

Digital AGCs are provided to compensate for near party/ far party voice level differences.

Ten stand-alone nonvolatile memories are provided for onboard storage and recall of filter setups created by the PCAP II *Plus* Master Control Program. This allows the external processor to be operated in any of 10 previously-stored setups without being connected to a PC.

1.3: EXTERNAL PROCESSOR FRONT PANEL

The front panel controls are arranged into three logical groups: headphone MONITOR controls, STAND-ALONE controls, and INPUT LEVEL controls.

The MONITOR controls allow the user to listen to either the INPUT or OUTPUT signals with a pair of stereo headphones connected to the 1/4" PHONES jack. Switching the headphones between the INPUT and OUTPUT signals does not alter the signal flow to the LEFT OUT and RIGHT OUT line output RCA connectors. The VOLUME level can be adjusted to a comfortable listening level.

The STAND-ALONE CONTROLS allow the user to select and run one of ten previously-programmed filter setups stored in internal nonvolatile memory. For complete instructions on using the STAND-ALONE memories, see Sections 4.10.3: and 5: .0.

The front panel of the PCAP II *Plus* external processor appears as follows (Figure 1-):

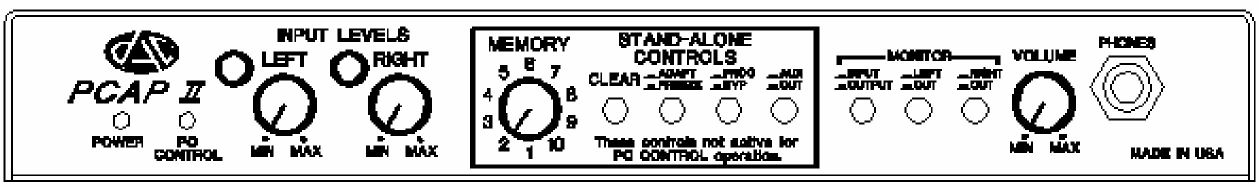

Figure 1-3 PCAP II *Plus* Front Panel

The LEFT and RIGHT INPUT LEVEL controls allow the user to adjust the input signals to the proper level for processing. Tricolor LEDs are provided to indicate signal levels; when the LEDs are GREEN, the signals are at the proper levels, while YELLOW indicates caution. RED indicates potential input overload. Input signal levels are also indicated by the displayed bargraphs on the PCAP II *Plus* Master Control Panel (See Section 3.2: PCAP II Tutorial).

The POWER led indicates when power is supplied to the unit, while the PC CONTROL led indicates when the unit is under control of the PCAP II *Plus* Master Control software.

1.4: EXTERNAL PROCESSOR REAR PANEL

The rear panel of the PCAP II *Plus* external processor appears as follows (Figure 1-):

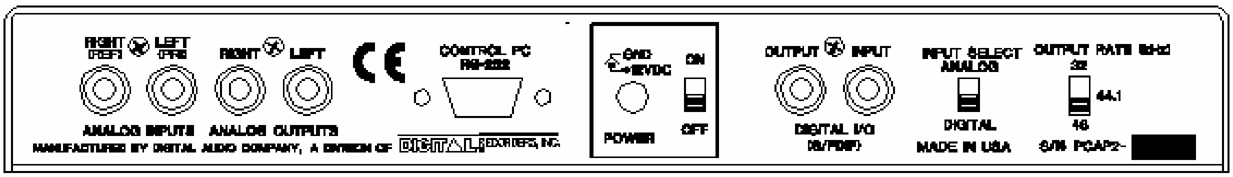

Figure 1-4 PCAP II *Plus* Rear Panel

DC power is provided to the unit through the external POWER jack by either the supplied external AC adaptor or direct connection to a 9 - 18 VDC source. The POWER switch must be switched to the ON position in order for the unit to operate.

A 9-pin RS232 jack is provided to connect the PCAP II *Plus* external processor to a computer using the supplied cable. The unit automatically senses RS232 activity and is put into PC CONTROL mode whenever the PCAP II *Plus* Master Control software establishes communication. After terminating the software, the unit will return to stand alone mode after about 10 seconds (PC CONTROL mode is indicated by the LED on the front panel).

Analog stereo input and output RCA jacks are provided on the rear panel. PCAP II *Plus* includes digital S/PDIF input and output jacks as well. Using the INPUT SELECT switch, the user can select which type of input (ANALOG or DIGITAL) to use. When DIGITAL input is selected, the PCAP II *Plus* will automatically synchronize to any legitimate digital input signal over a continuous range of sample rates from approximately 25kHz-108kHz. Both the ANALOG and DIGITAL outputs are always active.

The output rate of the digital output can be selected using the OUTPUT RATE switch. On PCAP II *Plus*, available output rate selections are 44.1kHz (standard, CD-compatible), 48kHz (professional, DAT-compatible), and AutoSYNC. The 44.1kHz and 48kHz selections force the digital outputs to the specified rate *regardless* of the input or processing sample rates; this is achieved through the use of asynchronous sample rate conversion technology. In the case of AutoSYNC, the digital output rate will synchronize to the digital input rate whenever the INPUT SELECT switch is set to DIGITAL, or be forced to 44.1kHz when ANALOG input is selected.

2: INSTALLATION INSTRUCTIONS

2.1: CAUTIONS TO USER

To install the PCAP II *Plus* hardware and software, the user must have a good working knowledge of IBM PC-compatible computers and the Microsoft Windows operating environment. Particularly, the user must know which RS232 COM ports are COM1, COM2, COM3, etc. On most modern PCs, there is a single COM port that is designated "COM1" that is suitable for connecting the PCAP II external processor; the standard installation procedure assumes that this is the case, that this single COM port is free (not connected to something else), and configures the software accordingly. However, if no free COM port is available, the user will need to create a free COM port via any of the following methods:

- Obtain a "USB to Serial" conversion module and install it onto any of your computer's available USB ports
- Obtain a "Serial Interface" card and install it in a free PCI slot inside your computer
- In the case of laptops, alternatively obtain a "Serial Interface" PCMCIA module and install it in a free PCMCIA slot in your laptop

WARNING: ***Should you instead choose to share a single COM port connection with a PDA, e.g. a Palm Pilot, connecting each to the COM port only when needed, be sure to shut down any "Hot Sync" manager application that might be running before attempting to run the PCAP II Plus Master Control software. Otherwise communication with the PCAP II Plus will fail due to conflict with the Hot Sync manager.***

Most likely, if any of these alternative options is used to create a free COM port, the new COM port will not be COM1, but rather might be COM2, COM3, COM4, etc. Therefore, it will likely be necessary to reconfigure the PCAP II *Plus* software COM port selection to the correct port by performing the procedure in Section 6.1: after completing the installation procedure in Section 2.2: .

The Microsoft Windows operating environment must be in place prior to installing the PCAP II *Plus* software. All video drivers, device drivers, etc. must be installed and operating properly.

For advanced users who need to configure the PCAP II *Plus* to operate at RS232 symbol rates slower than 115200 baud (factory default), please see Section 6.2: .

NOTE: Altering the baud rate setting is not recommended.

2.2: INSTALLATION PROCEDURE

1. Carefully remove the PCAP II *Plus* external processor from the shipping container. Confirm that the AC power adapter, RS232 cables (2), demonstrator audio CD, and software disk(s) are included. Also confirm that any optional accessories are included.

2. Connect the AC power cord to the PCAP II *Plus* rear panel POWER connector. Keep the POWER switch OFF for now.

3. With the PCAP II *Plus* POWER switch OFF, plug the AC power adapter into an AC outlet.

4. With the computer turned OFF, connect the supplied RS232 cable between the CONTROL PC RS232 connector on the PCAP II *Plus* rear panel and the desired computer COM port (You may need to purchase an adapter if the COM port has 25 pins).

5. Now that they are connected together, switch ON both the PCAP II *Plus* and the computer (sequencing is not critical).

6. Once Windows has booted up (Windows 95, 98, NT, 2000, or XP), insert the PCAP II *Plus* Master Control software CD into the CD-ROM drive in your PC. The installation program should automatically run. If not, use the **Run** command in the Windows **Start** menu to browse the CD-ROM and select the **setup.exe** installation program.

   NOTE: If for any reason you ever lose the software installation CD, or need to install upgrades and/or bug patches, the latest version software is always available for download at www.dacaudio.com; please contact DAC to obtain the proper access information

7. The Setup Utility will now install the PCAP II *Plus* Master Control software on your PC's hard disk. Please follow any instructions displayed by the Setup Utility.

   Once the Setup Utility has completed installing the software, the icon in Figure 2-1 should appear on your screen:

Figure 2-1 PCAP II *Plus* Master Control Icon

Double click on this icon now to run the program. A screen similar to Figure 2-2 should appear:

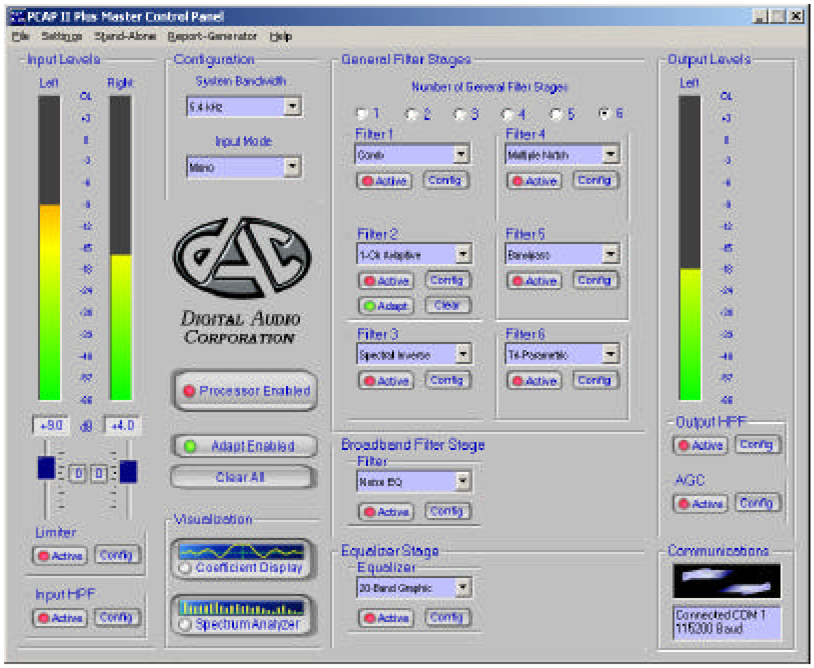

Figure 2-2 Master Control Panel

If an error message is displayed, it is possible that the software is not configured for the correct COM port (software defaults to COM1). If you know to which COM port the PCAP II *Plus* external processor is connected, configure the Master Control program for the correct COM port by following the procedure in Section 6.1: .

The PCAP II *Plus* system should now be installed and ready to run.

3: GETTING STARTED

Operation of the PCAP II *Plus* system is highly intuitive; most operators can quickly learn while using. The Fast Start procedure in Section 3.1: should allow first time users to quickly begin processing audio and utilizing the basic enhancement capabilities of the PCAP II *Plus*. For a lesson in operating the PCAP II *Plus* controls, it is recommended that the user also complete the Tutorial in Section 3.2: . Section 3.3: applies the PCAP II *Plus* to different noise problems contained on the PCAP II *Plus* Training CD.

In the following sections the manual will refer to both a Source Audio Player and an Enhanced Audio Recorder. The Source Audio Player can be any type of playback unit such as a Cassette, CD, Minidisc, Microcassette, or Digital Audio Tape (DAT) playback unit; for live processing applications, it could also be a live audio source. The Enhanced Audio Recorder can similarly be any type of audio recorder and/or output feed in the case of live processing.

In the case of a Digidesign Pro Tools® integration, Source Audio Player refers to the track(s) being played into the PCAP II *Plus* via the Digital Audio Interconnect, while Enhanced Audio Recorder refers to the track(s) being recorded from the PCAP II *Plus* output via the Digital Audio Interconnect.

Always consider that the quality of both the playback and the recording equipment will limit the quality of the processed and enhanced audio. Always try to use digital audio connections whenever they are available, as this avoids unnecessary analog-digital conversion losses. In the case of analog original material, e.g. cassette and microcassette, always use the highest quality machine you can obtain for playback, as everything else will be limited by this step. ***Direct digital recording of the enhanced audio is always recommended.***

3.1: FAST START

Fast start the PCAP II *Plus* as follows:

1. Set the rear panel INPUT SELECT switch to ANALOG and connect the LEFT and RIGHT channel line-level audio outputs (AUDIO OUT jacks) of your Source Audio Player to the LEFT (PRI) and RIGHT (REF) ANALOG INPUTS RCA jacks on the PCAP II *Plus* external processor rear panel as shown in Figure 3-1. Note that the RIGHT (REF) signal is only used in stereo configuration and with the 2CH Adaptive Filter in mono configuration. Alternatively, for direct digital input of digital audio material, the INPUT SELECT switch can be set to DIGITAL and the digital output of your Source Audio Player can be connected to the INPUT DIGITAL I/O RCA jack.

2. If you wish to record the enhanced audio, connect the line-level audio inputs (AUDIO IN jacks) of your Enhanced Audio Recorder to the LEFT and RIGHT ANALOG OUTPUTS RCA jacks on the PCAP II *Plus* external processor rear panel as shown in Figure 3-1. Alternatively, for direct digital recording of the enhanced audio, connect the OUTPUT

DIGITAL I/O RCA jack to the corresponding digital input jack of your digital Enhanced Audio Recorder.

3. Connect your stereo headphones to the PHONES jack on the PCAP II *Plus* external processor front panel as shown in Figure 3-1. Make sure that the MONITOR pushbuttons are set to OUTPUT, LEFT, and RIGHT (the three buttons are “out, in, and in”, respectively). Turn the phones VOLUME control to MIN. It is also recommended that headphones be connected to the Enhanced Audio Recorder's headphone jack to confirm that the output signal is being properly recorded.

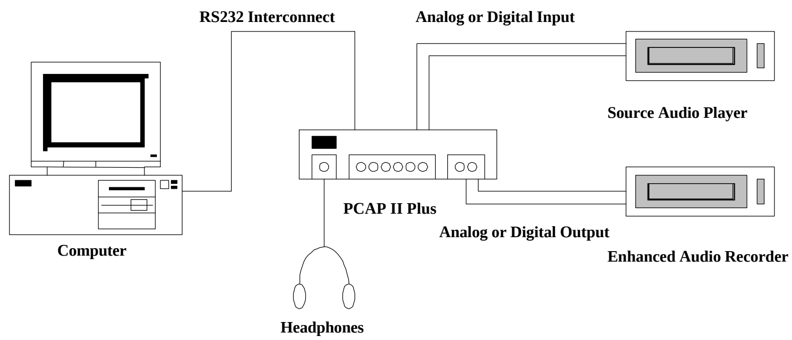

Figure 3-1 PCAP II *Plus* Connection Diagram

4. With the installation procedure in Section 2.2: complete, run the PCAP II *Plus* Master Control program by double-clicking on the screen icon. It may require several seconds for the program to load and initialize the external unit.

5. With the PCAP II *Plus* Master Control Panel displayed on your PC screen, insert some recorded audio material into the Source Audio Player, then press the PLAY button.

6. With the audio now flowing into the PCAP II *Plus*, adjust the LEFT and RIGHT INPUT LEVELS controls on the external processor front panel until the Input Levels bargraphs on the PCAP II *Plus* Master Control Panel indicate GREEN with occasional peaks in the YELLOW range. Note that some monaural material may not have audio present on the right channel.

7. With the phones VOLUME initially set to MIN, place the stereo headphones on your ears and slowly adjust the phones VOLUME for comfortable listening.

   NOTE: If you wish to listen to the unprocessed analog audio going into the PCAP II *Plus*, toggle the MONITOR switch on the external processor to INPUT (button “in”); this does NOT work for digital input material. To listen to the PCAP II *Plus* output, toggle the MONITOR switch back to OUTPUT.

If you wish at this point to discontinue the Fast Start procedure and experiment with the PCAP II *Plus* Master Control Panel on your own, please feel free to do so - you will not damage anything. If, however, you still feel unsure of what to do, please continue the Fast Start procedure as follows:

8. From the PCAP II *Plus* Master Control Panel menu bar, click on **File**. This will cause the following pulldown menu to appear:

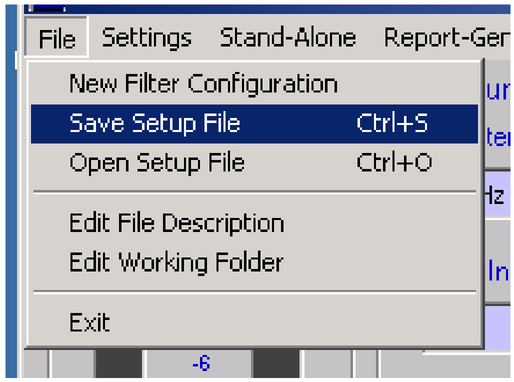

Figure 3-2 Fast Start File Pulldown Menu

9. Click on **Open Setup File** to bring up a window similar to Figure 3-3:

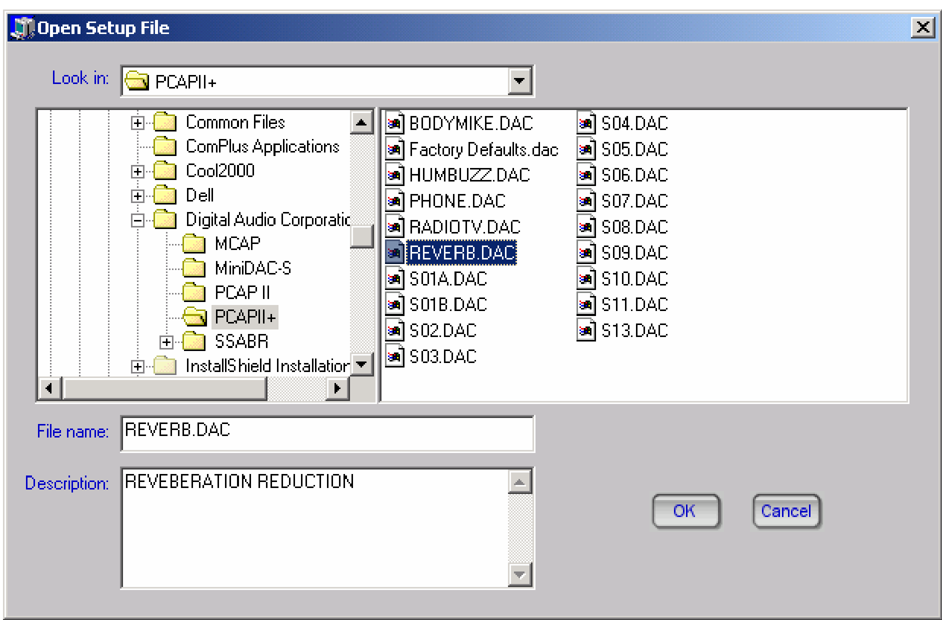

Figure 3-3 Fast Start Open Setup File Window

10. The **Setup Files** box should contain a list of all PCAP II *Plus* setup files on your hard disk. The installation procedure in Section 2.2: should have installed several setup files which have the **.DAC** extension onto your hard disk. Included in these setup files are:

File	Description
PHONE.DAC	"Landline" telephone audio enhancement
GSMBUZZ.DAC	GSM mobile phone RF interference noise reduction
BODYWIRE.DAC	"Body wire" receiver static noise reduction and voice enhancement
MOTOR.DAC	Motor/engine noise reduction
HUM60HZ.DAC	Powerline hum and buzz removal (North America)
HUM50HZ.DAC	Powerline hum and buzz removal (all countries outside North America)
JAILCELL.DAC	Cancellation of severe room echoes and reverberations (e.g., jail cell)
RADIOTV.DAC	Cancellation of radio and/or TV audio from live or recorded audio using a reference

These basic setups should be able to improve the voice quality of most typical law-enforcement recordings. To begin processing audio with one of these setups, open the desired setup file by clicking on its filename in the **Setup Files** box, then click on **OK**.

11. When the mouse cursor returns to normal, the setup file opening will be complete. Toggle the MONITOR switch on the external processor between INPUT and OUTPUT to hear the difference between the unprocessed input signal and the processed output signal (again, the MONITOR INPUT selection does not function when you are using the DIGITAL input).

Feel free to experiment with the PCAP II *Plus* Master Control Panel settings – the settings loaded from the setup file can usually be tweaked by the user to improve the processed result. Remember - you can always go back to the original setup at any time by repeating steps 8-12. Also, any setup you create can be saved for later reuse by clicking **File,** then **Save Setup File.** Or, if you wish to try loading any of the other basic setup files, simply repeat steps 8-11 and select a different file.

Once you become comfortable recalling the basic setup files, you are encouraged to complete the PCAP II *Plus* Tutorial in Section 3.2: .

3.2: PCAP II TUTORIAL

This brief tutorial should allow the user to quickly learn the basic operation of the PCAP II *Plus* controls. It should require about 1 hour for completion.

NOTE: A subset of the PCAP II *Plus* functions are utilized in this tutorial. Refer to Chapter 4: for detailed information on all of the PCAP II *Plus* functions.

<u>Tutorial Steps:</u>

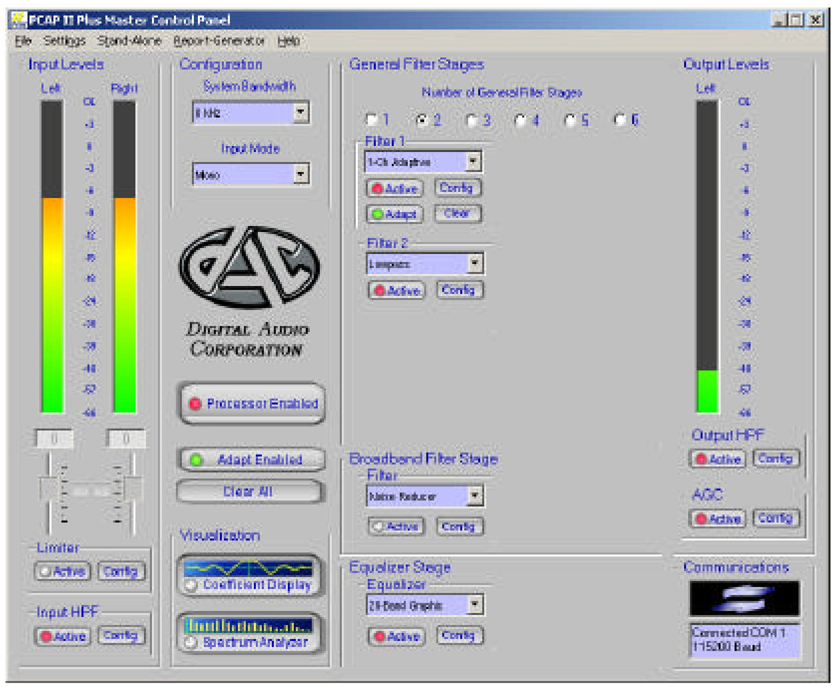

Figure 3-4 Tutorial Master Control Panel

1. With the installation procedure in Section 2.2: completed, run the PCAP II *Plus* Master Control program by clicking from the Windows Start menu on Start->Programs->PCAP II Plus->PCAP II Plus. A window similar to Figure 3-4 Tutorial Master Control Panel will appear:

   This is the Master Control Panel, from which all features can be accessed. The Master Control Panel is organized logically from left to right, with audio input controls at the far left, digital processing controls in the middle, and audio output controls at the far right.

2. Use the mouse* to set the **Input HPF, Limiter, Output HPF, AGC, System Bandwidth, Input Mode, Number of General Filter Stages, Adapt Enabled,** and **Processor Enabled** buttons as they are shown in Figure 3-4. Do not set up the Filter blocks and Equalizer blocks at this time.

3. Connect the LEFT and RIGHT outputs (AUDIO OUT jacks) of the audio player to the LEFT and RIGHT ANALOG INPUTS jacks on the PCAP II *Plus* external processor; make sure that the INPUT SELECT switch is set to ANALOG. Insert the PCAP II *Plus* Training CD and set to the first track.

4. Connect a pair of stereo headphones to the PHONES jack on the PCAP II *Plus* external processor. Switch the MONITOR switches to INPUT/LEFT/RIGHT and adjust the headphone LEVEL control to MIN. Mono headphones should not be used as possible damage to the headphone amplifier could occur.

5. Play the demonstration audio CD. Adjust the INPUT LEVELS controls on the PCAP II *Plus* external processor so that the peak audio level, as measured by the bargraphs on the Master Control Panel, is approximately -6 dB.

   NOTE: The audio needs to be playing throughout this tutorial. If you reach the end of the audio before completing the tutorial, please restart it (some CD players allow a track to be continually repeated.)

6. With the demonstration audio CD playing, slowly increase the headphone VOLUME until audio can be clearly heard in the left ear (no signal in the right ear).

7. Click on the **Active** button in both the Filter 1 block and the Filter 2 block until both buttons indicate that the filter is out (bypassed, RED LED unlit).

8. Click on the **Active** button in the Equalizer block until the button indicates equalizer is out (bypassed, RED LED unlit).

9. Switch the MONITOR switches on the PCAP II *Plus* external processor to OUTPUT/LEFT/RIGHT. Readjust the headphone VOLUME if necessary.

10. Switch the Input HPF in and out by clicking on the **Active** button in the Input HPF block. You should hear the low frequency effects of this control. Restore the **Active** button such that the Input HPF is in.

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* **Fast keys** are also available for buttons having an underlined letter. Pressing <Alt> in combination with the letter causes that button to toggle, *e.g.* pressing <Alt-P> toggles the **Processor Enabled** button in our out.

11. Switch the Output HPF in and out by clicking on the **Active** button in the Output HPF block. You should hear the low frequency effects of this control. Restore the button such that the Output HPF is in.

12. Click on the **Config** button in the Limiter block to activate the Limiter Setup Window. The following window (Figure 3-5) will appear:

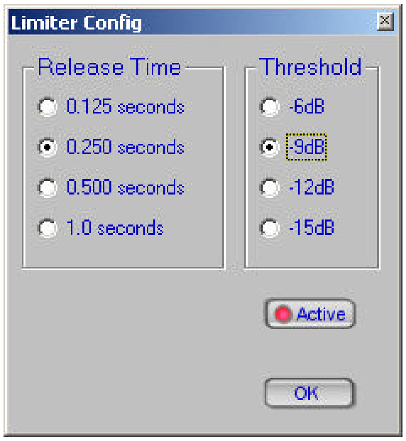

Figure 3-5 Tutorial Limiter Setup Window

When the limiter is active, red markers will appear in the Input Level bargraphs on the Master Control Panel. These markers indicate the limiter threshold setting.

13. Use the mouse to set the Release Time to 0.250 seconds and the Threshold to -9dB, as shown in Figure 3-5. Click on **OK** when done*.

14. Adjust the headphone VOLUME control to MIN. Increase both INPUT LEVELS controls fully clockwise to the **MAX** position. This should cause the PCAP II *Plus* audio inputs to overload (this will not damage the unit but will distort the audio). The tricolor level LED should indicate RED on peaks, and the Left bargraph in the Input Levels block should be frequently popping up into the RED zone, indicating overload.

15. Slowly increase the headphone VOLUME until distorted audio can be clearly heard.

16. Switch the Limiter In and Out by clicking on the **Active** button in the Limiter block. You should notice the indicated bargraph levels decrease to -9dB and the audio quality (as heard through the headphones) dramatically improve whenever the Limiter block **Active** button is lit RED. The input limiter is electronically lowering the INPUT LEVELS to avoid overload.

---

*If you try to click anywhere on the Master Control Panel while a control window (such as the Limiter Setup window) is displayed, a warning beep will sound, indicating that you need to first close the control window by clicking the **OK** button.

Restore the **Active** button to the out position (LED unlit) and restore the INPUT LEVELS controls to normal level.

17. Click on the **AGC Setup** button in the Output Levels block. The following window (Figure 3-6) will appear:

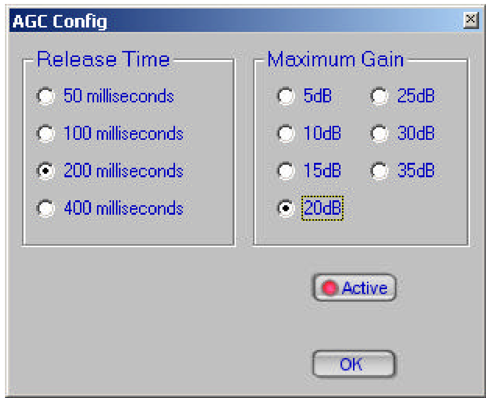

Figure 3-6 Tutorial AGC Setup Window

Use the mouse to set the Release Time to 200 milliseconds and the Maximum Gain to 20dB, as shown in Figure 3-6. Click on **OK** when done.

18. Reduce the INPUT LEVEL until the Output Level Bargraphs indicate a peak signal level of approximately -36dB.

19. Switch the AGC in and out by clicking on the **Active** button in the AGC Block. You should notice the output level bargraphs slowly increase to approximately -18dB indicated level and the audio level as heard through the headphones dramatically increase whenever the button indicates that the AGC is in. The AGC attempts to make the Output Level constant and is useful in near party/far party situations. Restore the **Active** button to the AGC is out, and restore the INPUT LEVELS controls to normal level.

20. Click the filter selection box within the Filter 1 block. A drop-down window similar to the one in Figure 3-7 will appear below the selector. Use the mouse to click on **1-Channel Adaptive**. The Filter 1 block should now indicate that the selected mode is 1-Channel Adaptive.

Figure 3-7 Tutorial Filter Selection Window

21. Click on the **Select** button in the Filter 2 block and select **Lowpass**.

22. Click on the **Control** button in the Filter 1 block. The window in Figure 3-8 will appear:

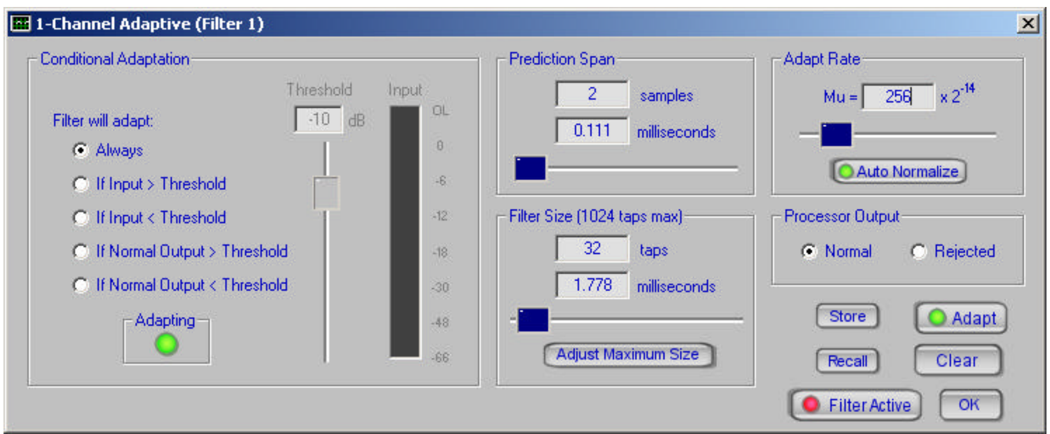

Figure 3-8 Tutorial 1-Channel Adaptive Filter Control Window

This is the 1-Channel Adaptive Filter Control Window. It is used to select Filter Size, Adapt Rate, and Prediction Span. It also permits configuration of the filter for Conditional Adaptation.

23. Use the mouse to set all controls to match the settings shown in Figure 3-8*. Do <u>not</u> click on the **OK** button at this time.

24. Listen to the filter OUTPUT through your headphones. Note the effect on the signal as the **Filter Active** button is toggled between the in state (LED is lit RED) and the out state (LED is

---

*Scroll bars are used throughout the PCAP II to adjust various filter parameters. Drag the scroll box "slider" or click *within* the scroll bar to make coarse adjustments. Click on the scroll arrows on *either end* of the scroll bar to make fine adjustments. The text boxes also allow precise entry of values.

unlit). You should notice a reduction in background noise whenever the filter is in. Press the **Clear** button to cause Filter 1 to readapt to the input signal.

25. Restore the **Filter Active** button to the out state (LED unlit) and click on **OK** to exit the 1-Channel Adaptive Filter Control Window.

26. Click on the **Control** button in the Filter 2 block. The window in Figure 3-9 will appear:

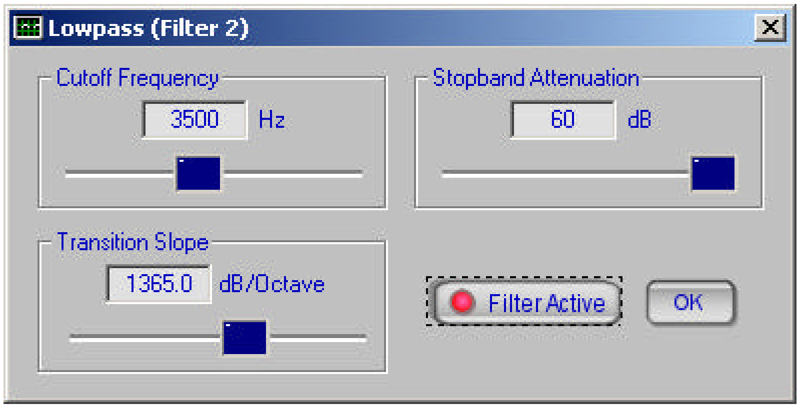

Figure 3-9 Tutorial Lowpass Filter Control Window

This is the Lowpass Filter Control Window. It is used to select Cutoff Frequency, Stopband Attenuation, and Transition Slope.

27. Use the mouse to set all controls to match the settings shown in Figure 3-9. Do <u>not</u> click on the **OK** button at this time.

28. Listen to the filter OUTPUT through your headphones. Note the effect on the signal as the **Filter Active** button is toggled in and out. You should notice a dramatic reduction in high-frequency sound whenever the button indicates that the filter is in. Try adjusting the Cutoff Frequency control to various settings; as the Cutoff Frequency is increased, you should hear high-frequency sounds increase, and as Cutoff Frequency is reduced, high-frequency sounds will be reduced.

29. Click on **OK** to exit the Lowpass Filter Control Window.

30. Click on the **Active** button in the Filter 2 block of the Master Control Panel until it indicates that the filter is out.

31. Click on the **Active** button in the Equalizer Block until it indicates that the Equalizer is in.

32. Click on the Equalizer selection box in the Equalizer Block. The window in Figure 3-10 will appear.

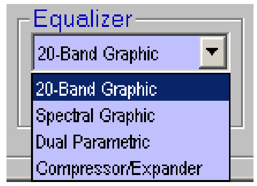

Figure 3-10 Tutorial Equalizer Selection Window

Use the mouse to click on the **20-Band Graphic**. The Equalizer block should now indicate that the selected mode is 20-Band Graphic.

33. Click on the **Control** button in the Equalizer block. The window in Figure 3-11 will appear.

This is the 20-Band Graphic Equalizer Control Window. The twenty vertical scroll bars, or "sliders", are used to set the equalizer attenuation for each of the 20 frequency bands.

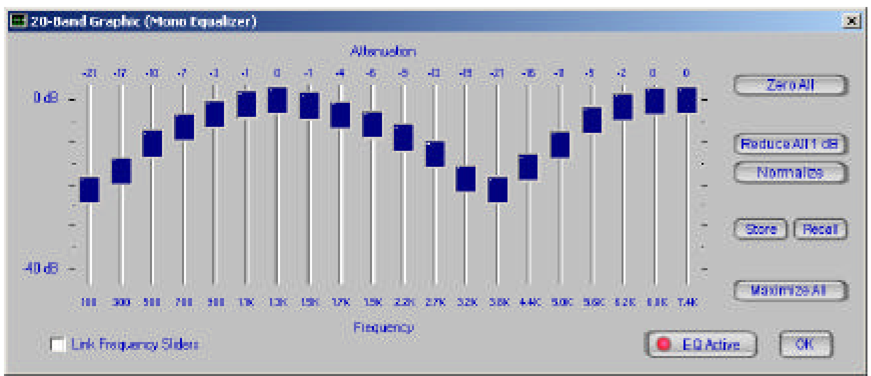

Figure 3-11 Tutorial 20-Band Graphic Equalizer Control Window

34. Use the mouse to set all sliders to match the settings shown in Figure 3-11.

35. Click on the **Store** button to bring up the Store 20-Band Graphic Equalizer window as in Figure 3-12.

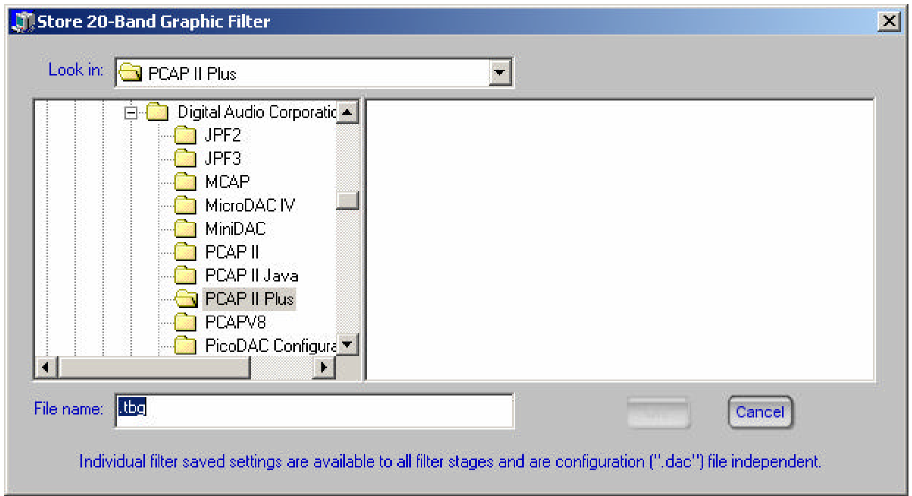

Figure 3-12 Tutorial Store 20-Band Graphic Equalizer Window

36. Enter a the filename "Test1.tbg" (the **OK** button will be enabled when a valid filename is entered). (We will recall these settings later.) Once you have clicked the **OK** button, the system will automatically return to the 20-Band Graphic Equalizer control window.

37. While listening to the OUTPUT audio with the headphones, adjust the sliders and listen to how each affects the sound. (Remember, you can drag the box, click in the bar, or click on the end arrows. Also note that the gain value is displayed above each bar.) Observe the effect on the signal as the **Active** button is toggled between having the equalizer in and out. Also, note that whenever a slider is adjusted, the filter will automatically be switched in (**Active** button illuminated).

38. Click on the **Reduce All 1dB** button a few times and see how all the sliders move down together 1dB at a time. (This enables you to increase the gain of a slider previously at 0 dB.) Click on the **Normalize** button and see how all the sliders will instantly move up together so that the highest slider is exactly 0dB. (Once the equalizer is adjusted, it should be normalized to minimize signal loss.)

39. Click on the **Zero All** button to instantly set all the sliders to the 0dB position. The original slider settings would now be lost had we not previously stored them in step 36.

40. Click on the **Recall** button to bring up the Recall 20-Band Graphic Equalizer window. The window in Figure 3-13 will appear.

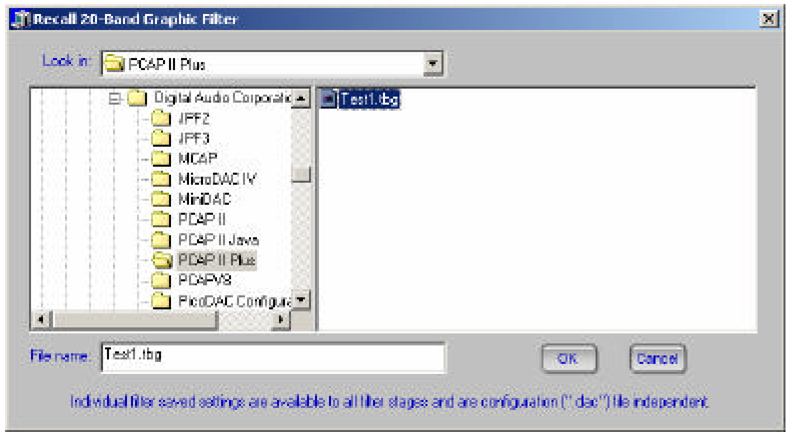

Figure 3-13 Tutorial Recall 20-Band Graphic Equalizer Window

41. Select the file "Test1.tbg" that was saved in Step 36 to recall the slider settings previously stored. Once you have clicked this button, the system will automatically return to the 20-Band Graphic Equalizer control window and the stored slider settings will be restored[*].

42. Click on **OK** to exit the 20-Band Equalizer Control Window.

43. Click on the **Spectrum Analyzer** button. This will cause the Spectrum Analyzer window (Figure 3-14) to appear as follows: (The actual spectral display will depend upon the audio being processed.)

    Use the mouse to set up all controls as they appear in Figure 3-14. Make sure that the **Left In** selection is chosen in the Trace 1 signal select box, that the **Line Output** selection is chosen in the Trace 2 signal select box, and both **Show Trace** boxes are checked.

[*] Any time a filter setting is stored, a file named "Previous Filter Settings.tbg" is created. If after recalling the settings, you wanted to return to the original settings, you can reload this file.

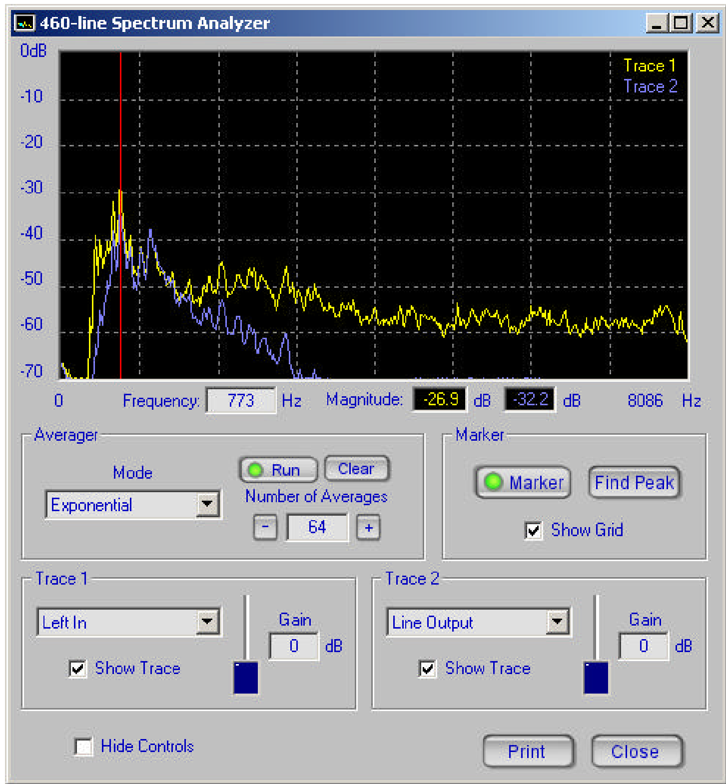

Figure 3-14 Tutorial Spectrum Analyzer Window

44. Try adjusting the **Number of Averages** control using the + and - buttons in the Averager block. Note that as the **Number of Averages** parameter is increased, the spectrum traces react more slowly and smoothly to the input signal. Return the **Number of Averages** setting to 8.

45. Click on the **Run** button in the Averager block until GREEN LED is no illuminated. You should see the spectrum waveform stop updating.* Click on the **Clear** button in the Averager block; both spectrum waveforms should now be cleared completely away. Restore the **Run** button to the active state to allow the spectrum waveform to resume responding to the input signal.

46. Click the mouse anywhere on the spectral waveforms to move the vertical red marker to that point. A readout of frequency (Frequency:) and magnitude (Magnitude:) of Trace 1and Trace 2 will be indicated below the display grid. Use the text entry box by the "Frequency:" indicator to more precisely control the marker.

---

*The spectral analysis freezes immediately; however, the display will briefly continue to update as remaining data is read from the PCAP II *Plus* external processor.

47. Click the **Find Peak** button to automatically move the marker to the largest magnitude displayed. Adjust the **Gain** controls for both Trace 1 and Trace 2 and note how the indicated magnitude for each trace is increased as gain is increased.

48. Set the **Gain** controls for the Yellow Trace to 40dB and the Blue Trace to 0dB. While the demo audio CD is playing, notice how the word **Gain** changes to **OVL**[1] (display overload) for Trace 1 and how the magnitude (Magnitude:) indication for the Trace 1 changes color from yellow to red whenever strong peaks occur. Reduce the **Gain** to avoid overload distortion of the spectral display.

49. Return both **Gain** controls to 0dB.

50. Click <u>outside</u> the Spectrum Analyzer window on the Master Control Panel window to bring up that window, **leaving the spectrum analyzer active**. Do not close the Spectrum Analyzer window. Make sure that the two HPFs, Limiter, AGC, Filters, and Equalizer are all switched **OUT**. Click on the Filter 2 **Config** button to bring up the Lowpass filter control window.

51. Notice how the Master Control Panel will disappear while the Lowpas Configuration Window and Spectrum Analyzer windows are displayed on the same screen. Click the **Filter Active** button in the Lowpass window until it indicates that the filter is in. Trace 1 now displays the signal going into the Lowpass filter, while Trace 2 displays the signal coming out of the Lowpass filter. Adjust any of the Lowpass filter controls, noting the effect on the Trace 2[2].

52. Click on the **OK** button in the Lowpass window to return to the Master Control Panel.

53. Note **Coefficient Display** and **Spectrum Analyzer** buttons on the Master Control Panel allows the user to access either the Coefficient Display or the Spectrum Analyzer display, but not both at the same time. Click on **Spectrum Analyzer** to return to the Spectrum Analyzer window if it is not already visible.

54. Now turn off the Spectrum Analyzer by clicking on the **Close** button in the Spectrum Analyzer window. The system will now return to the Master Control Panel.

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[1] The **GAIN** affects the display only, enabling weaker signals to be seen; an **OVL** indication will not distort the processed audio.

[2] The Spectrum Analyzer functions may be adjusted by moving the mouse cursor to that window. However, the Spectrum Analyzer window may not by closed at this time.

55. Access the Save Setup File feature by clicking on **File** in the menu bar. The pulldown menu in Figure 3-15 will appear.

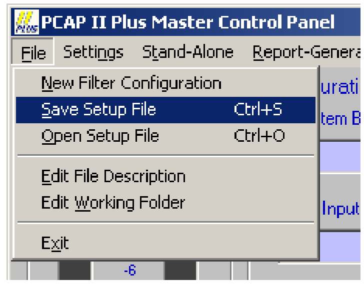

Figure 3-15 Tutorial File Pulldown Menu

56. Click on **Save Setup File** to bring up the Save Setup File window as shown in Figure 3-16.

For now, do not change the **Look In** or Folder Selection settings; just keep in mind that you could change these settings to allow setup files to be stored to any directory of any drive. This window will initialize the folder and drive to whatever was set in the **Edit Working Folder** Window.

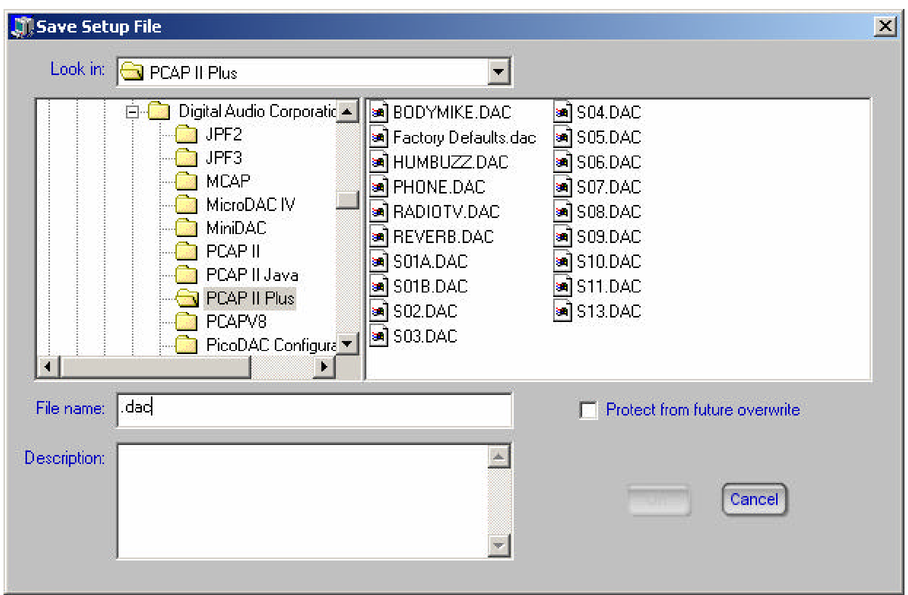

Figure 3-16 Tutorial Save Setup File Window

57. Click your mouse cursor on the **Description** text box. Using your keyboard type in "My First Setup". See Figure 3-17.

58. Now click your mouse cursor on the **File Name** text box. Use the left arrow key to move the cursor to just before the **.DAC** extension, then type "mysetup". These entries should appear as in Figure 3-17.

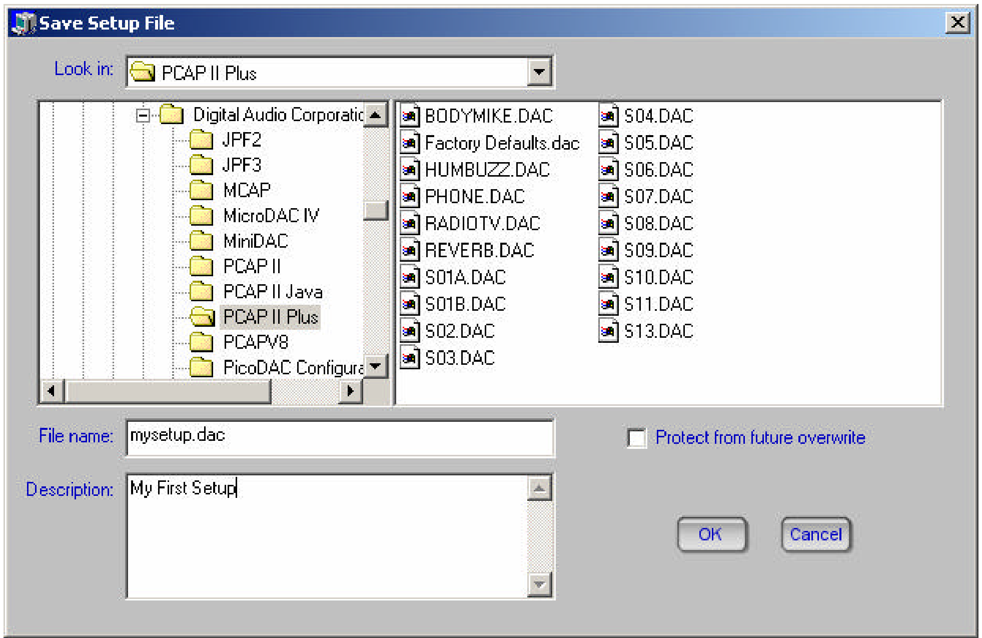

Figure 3-17 Tutorial Store in File Name Text Box

59. Click on **OK** to store your first setup file and return to the Master Control Panel.

60. Click on **File** to access the File Pulldown Menu as in step 56. Click on **Open Setup File** to bring up the Open Setup File window similar to Figure 3-3, but with **mysetup.dac** added to the files list.

    For now, do not change the path settings; just keep in mind that you could change these settings to allow setup files to be recalled from any folder of any drive.

61. Click on each file listed in the file list box on the right of the window. Notice how the **Description** for each appears at the bottom of the window as each is clicked.

62. Click on **PHONE.DAC**, then click on **OK** to open the recommended settings for processing telephone audio. If you do not have a printer installed on your computer, skip now to step 65.

63. Access the hardcopy report generator by clicking on **Report Generator** in the menu bar. The pulldown menu shown in Figure 3-18 will appear.

64. Make sure your printer is ready, then click on **Print Report to Printer**. A report listing all screen settings for the telephone filter setup will be printed (you will be prompted to select the printer to which to print to as well as whether or not to make the report single or double-spaced.)

65. Open your original filter settings by repeating steps 60 and 61 to bring up the **Open Setup File** window. Next, click on **mysetup.dac** in the list box, then click on **OK** to return to the Master Control Panel.

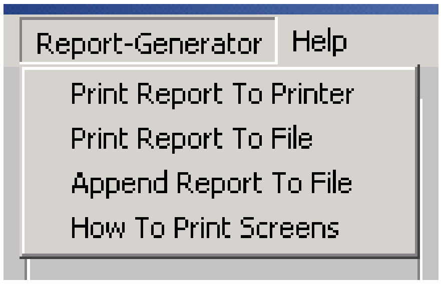

Figure 3-18 Tutorial Report Generator Pulldown Menu

This completes the PCAP II *Plus* Tutorial. Please feel free at this point to experiment with the on-screen filter settings. Remember you can always recall filter settings that will likely work using the **Open Setup File** feature. To exit the Master Control Panel, either double click on the **-** (minus) button at the upper left corner of the window, or click on the **File** menu bar option, then click on **Exit**.

Please refer to Chapter 4: for more detailed instructions on using the windows in the Master Control program.

3.3: TRAINING CD RECORDINGS

The PCAP II *Plus* Training CD supplied with the PCAP II *Plus* is a useful training tool. Each of its 13 noise sample recordings is accompanied by a several suggested setup files (".dac" files), installed on your computer along with the other software by the installation procedure.

Table 3.1 summarizes the contents of the training CD. The SETUP column contains the name of the setup files installed on the disk as part of the software installation procedure (Section 2.2:). To access a setup file, click on **File** in the Master Control menu bar (or press <Alt-F>). Next, click on **Open Setup File** (or press <Alt-O>). The available setup files should then appear in the **Setup Files** box; setup files have the extension "**.dac**". If the needed setup file is not displayed hen attempting to open a setup file, the path information may require changing. Refer to Section 4.10.2: for additional details.

According to Table 3.1, the first setup file "Training CD Track 3 Solution A.dac" is the suggested setup file for Track 3 (voice plus swept tones). In all cases the name of the proper setup file for a particular training track is *Training CD Track X Solution Y.dac*, where *X* is the track number and *Y* is the solution letter. Usually solutions "A" are the most simple solution, providing basic noise removal. Solution "B" is the more advanced noise solutions involving several filters to further improve the intelligibility of the speech. For more information about each solution file, view the **Description** from within the **Open Setup File** window as each file is selected. You may also use the **Edit File Description** option from within the **File Menu**.

The majority of tracks on the Training CD are real world examples chosen to demonstrate both the processing power of the PCAP II *Plus* as well as to show how difficult some noise problems can be. The intelligibility of a few of the more difficult tracks may be very low, even after enhancement. Remember, too, that the goal of forensic audio enhancement is intelligibility, not audio "quality". The aim is to understand what is said, not to make the tracks sound like they were recorded in a studio environment.

All recordings except Track 13 are monophonic, *i.e.*, the left and right channels have the same audio. The 13th recording is a Reference Canceller filtering example and has the primary microphone audio on the left channel and the TV reference audio on the right channel. Make sure to have the reference audio (right channel) slightly louder than the primary microphone audio for best results.

Playback the CD on a standard CD player connecting its Left and Right Line Outputs to the PCAP II *Plus* LEFT and RIGHT ANALOG MAIN INPUTS, respectively. As each track is played, open the suggested setup file. Try varying individual control parameters to observe overall processing effects.

Have fun!

Track	Time	Noise Description	Setup	Comments
1	02:30	500 Hz tone	Training CD Track 1 Solution A.dac Training CD Track 1 Solution B.dac	Adjust **Notch Freq** to null tone Single tone requires small filter size
2	02:20	400 Hz and 1100 Hz tones	Training CD Track 2 Solution A.dac Training CD Track 2 Solution B.dac	Adjust **Notch Freq** to null tones with notch filter(s) Compare adaptive and notch filters
3	01:11	Swept tones	Training CD Track 3 Solution A.dac Training CD Track 3 Solution B.dac	Adjust **Prediction Span** and **Adapt Rate** comparing effects Compare use of ASIF to balance audio
4	02:21	High frequency hiss	Training CD Track 4 Solution A.dac Training CD Track 4 Solution B.dac	Adjust LPF cutoff and attenuation Compare with addition of Adaptive and SIF to improve quality
5	00:44	Radio interference	Training CD Track 5 Solution A.dac Training CD Track 5 Solution B.dac	Combined use of small and large adaptive filters, small for tones, large for reverb Added Noise Reducer
6	01:14	A.C. hum	Training CD Track 6 Solution A.dac Training CD Track 6 Solution B.dac	Hi Res Graphic restores high frequencies. Adjust **Comb Frequency** to minimize 60 Hz hum Compare addition of SIF
7	02:40	Telephone A.C. hum	Training CD Track 7 Solution A.dac Training CD Track 7 Solution B.dac	Adjust **Comb Frequency** to minimize 60 Hz hum ASIF added to make audio sound more natural
8	02:24	Acoustic resonances	Training CD Track 8 Solution A.dac Training CD Track 8 Solution B.dac	SIF and Adaptive use to remove resonances ASIF used instead of SIF
9	02:34	Jail Cell	Training CD Track 9 Solution A.dac Training CD Track 9 Solution B.dac	Hi Res Graphic restores high frequencies Adaptive filters used to attack resonances
10	01:11	Body transmitter	Training CD Track 10 Solution A.dac Training CD Track 10 Solution B.dac	Hi Res, SIF, and Adaptive used to reduce reverb Adaptive used first
11	02:26	Bar music	Training CD Track 11 Solution A.dac Training CD Track 11 Solution B.dac	Filter cannot separate voices (spoken or sung) but does reduce muffling Compressor/Expander used in conjunction with ASIF
12	02:10	Interview and argument	Training CD Track 12 Solution A.dac	Very muffled audio with rumble noises, adaptive filters and NoiseEQ used to reduce
13	04:52	Radio/TV	Training CD Track 13 Solution A.dac Training CD Track 13 Solution B.dac	Ref Canceller adaptive filtering followed by Adaptive to reduce reverb Addition of Lowpass and ASIF to reduce high frequency music noises

Table 3.1 PCAP II *Plus* Training CD Summary

4: PCAP II *PLUS* SOFTWARE REFERENCE MANUAL

This portion of the user's manual is designed as a reference guide to which the user may refer for more detailed information regarding the PCAP II *Plus* Master Control program graphical user interface (GUI). It is assumed in this section that the user has a good working knowledge of Microsoft Windows, has properly installed the PCAP II *Plus* hardware and software using the installation instructions in Section 2.2: , and has either completed the Tutorial in section 3.2: or attended the “DAC School” training course. Please contact DAC if you would like to schedule participation in the next available DAC School session.

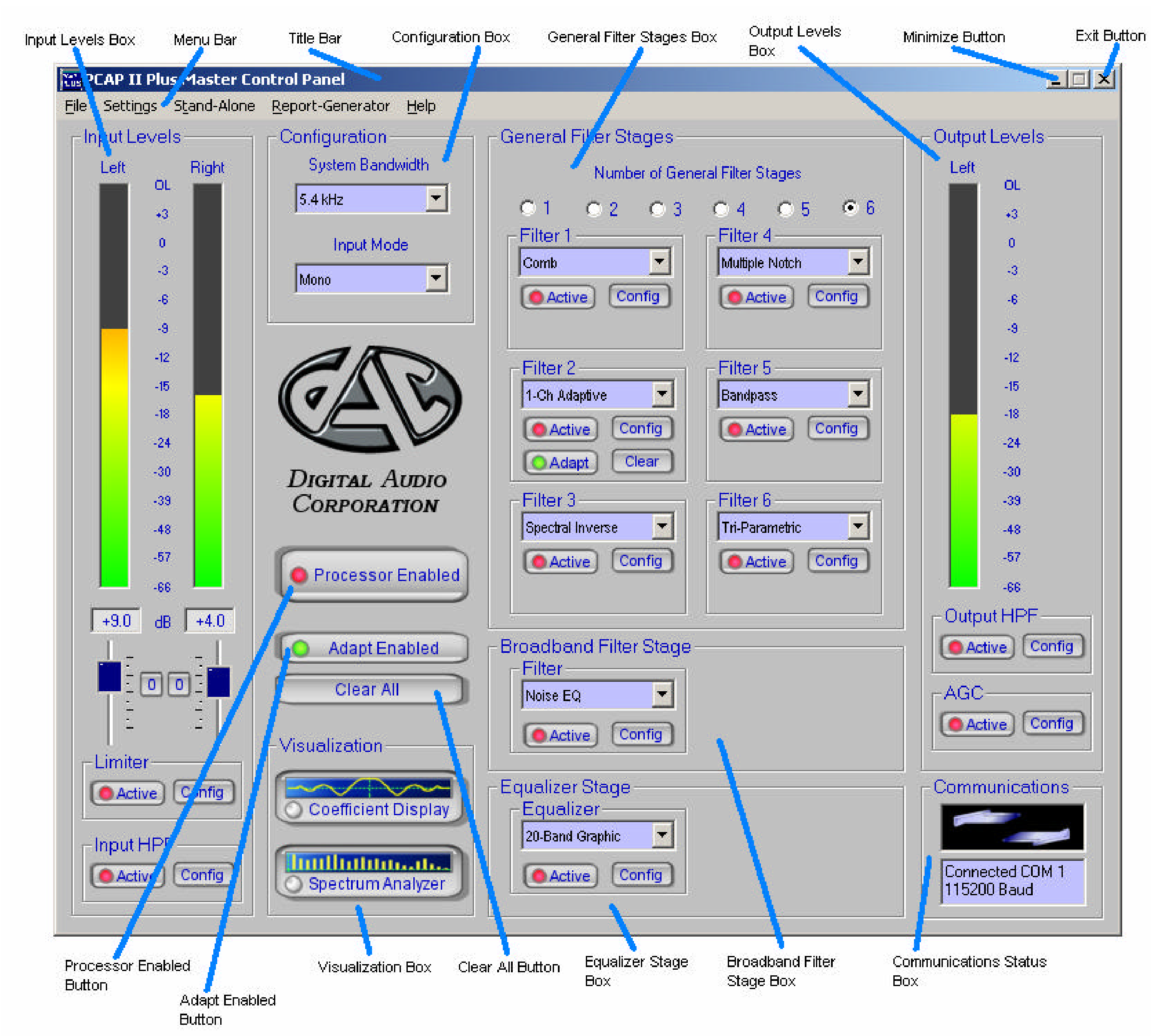

Figure 4-1 PCAP II *Plus* Master Control Panel

4.1: MASTER CONTROL PANEL

When the PCAP II *Plus* Master Control program is run from Microsoft Windows, the Master Control Panel appears. The Master Control Panel features are shown in Figure 4-1.

From this screen, all audio processing capabilities can be accessed. The Master Control Panel is organized logically from left to right, with audio input controls at the far left, digital signal processing controls in the middle, and audio output controls at the far right. Master Control functions are as follows:

Title Bar	Used to "drag" the Master Control window to the desired area of the display. Also indicates the filename for any .DAC setup file currently in use; see Section 4.10: for further details on filter setup files.
Menu Bar	Used to access the Master Control pulldown menus that allow saving and opening of filter setups from disk files, configuring RS232 communication with the external processor, storing filter setups in nonvolatile memories inside the external processor for Stand-Alone operation, generating hardcopy setup reports, and getting online help. See Section 4.10: for further details on these features.
Input Levels Box	Used to view input signal levels via bargraph display, adjust input levels via slider controls if desired, **Config**ure the Input HPF stage and toggle its **Active** status, and **Config**ure the input Limiter stage and toggle its **Active** status. Also, in the case where the DIGITAL input is used, the status of the channel swapping feature is indicated just above the bargraphs; see Section 4.10: for further details on these features.
Configuration Box*	Used to select the System Bandwidth for the external processor which most closely matches that of the input signals (physically adjusts the processing sampling rate of the PCAP II *Plus*). For speech processing applications, settings of 3.2kHz to 6.5kHz work best. For wider bandwidth applications, 8.0kHz to 16kHz is recommended. Also, allows specifying the Input Mode; **Mono** allows the user to enhance a single input signal (Left channel only) with as many as 12 successive stages of signal processing. **Stereo, Linked** allows the user to process two input signals

* Changing any of these settings will automatically clear all Ref Canceller, 1CH Adaptive, Noise Reducer, NoiseEQ, Multiple Notch, Multiple Slot, Spectral Inverse, and Hi-Res Graphic Filters.

simultaneously with as many as nine stages of digital signal processing; each stage is identical for both the Left and Right channels. **Stereo, Independent** is the same as **Stereo, Linked**, except that the processing stages may be setup and controlled independently for the Left and Right channels.

General Filter Stages Box — Selects the number of General Filter Stages that are to be used. Between 1 and 6 of these stages are available, depending upon the specified System Bandwidth and Input Mode. For each General Filter Stage, any of 17 different types of filters can be selected using the drop-down menu; also, dedicated Master Control buttons for each stage allow the user to toggle the **Active** status (filter is active when the LED illuminated RED, inactive when not illuminated) and open special dialogs that **Config**ure the detailed filtering parameters. For adaptive filter selections (1-Channel Adaptive and Ref Canceller), additional buttons allow the user to toggle the **Adapt** status (adaptation is enabled when the LED illuminated GREEN, inactive when not illuminated) and **Clear** the adaptive filter solution from memory. See Section 4.4: for detailed descriptions of the individual filter controls.

Output Levels Box — Used to view output signal levels via bargraph displays, **Config**ure the Output HPF stage and toggle its **Active** status, and **Config**ure the AGC stage and toggle its **Active** status. See Section 4.2.1: for further details on the Output HPF and AGC stages.

Minimize Button — Standard Windows control, allows the Master Control screen to be hidden while still running. To make the screen viewable again, simply click the PCAP II *Plus* icon on the Windows Task Bar, which is normally at the bottom of the Windows desktop.

Exit Button — Standard Windows control, allows the Master Control program to be terminated; alternatively, the user can utilize the Menu Bar and click on **File**, then **Exit**, or type <Alt-F4> using the keyboard. After termination, the PCAP II *Plus* External Processor will automatically switch back to STAND ALONE operation after 10 seconds.

Visualization Box — Allows user to activate either the built-in Coefficient Display (see Section 4.9: for more details) or the built-in Spectrum Analyzer (see Section 4.8.2: for more details).

Processor Enabled Button — Toggle control used to specify whether the original input signals (LED not illuminated) or the processed signals (LED illuminated RED) are routed to the ANALOG and DIGITAL outputs.

Adapt Enabled Button	Toggle control used either to freeze all adaptive filters (LED not illuminated) or to allow non-frozen filters (as specified by the **Adapt** button for each Filter) to adapt (LED illuminated GREEN).
Clear All Button	Clears the filter coefficient solutions of any adaptive filters selected. If no adaptive filters are present, then the **Clear All** button has no effect. **NOTE:** To individually clear any adaptive filter without disturbing other adaptive filters, click on its individual associated **Clear** button.
Broadband Filter Stage Box	Used to control the Broadband Filter Stage. In Mono and Stereo, Linked Input Modes, one Filter block will be displayed. In Stereo, Independent Input Mode, separate Filter blocks will be displayed for the Left and Right channels. Select the type of broadband filter to be implemented using the drop-down menu (See Section 4.3.3: for more details). Setup the selected filter's parameters using the **Config** button to view the control screen for that filter (See Section 4.5: for further details). Switch each filter stage In or Out of the process using the **Active** button.
Equalizer Stage Box	Used to control the equalizer stages. In Mono and Stereo, Linked Input Modes, one equalizer block will be displayed. In Stereo, Independent Input Mode, separate equalizer blocks will be displayed for the Left and Right channels. Select the type of equalizer to be implemented using the drop-down menu (See Section 4.3.3: for more details). Setup the selected equalizer's parameters using the **Config** button to view the control screen for that equalizer (See Section 4.5: for further details). Switch each equalizer stage In or Out of the process using the **Active** button.
Communications Box	Indicates the status of the RS232 communications between the Master Control software and the External Processor. Normally should indicate “Connected COM1”, “115200 Baud” if everything OK; if the word “Closed” appears in the box, troubleshoot the connection as described in Sections 6.1: and 6.2: .

4.2: INPUT AND OUTPUT PROCESSORS

4.2.1: Input Level Measurement/Adjustment

Application:

It is critical that the input signals coming into the PCAP II Plus be adjusted in level such that the Input Levels bargraphs on the Master Control screen are maximized in intensity without indicating an overload condition (display fully pegged in the RED zone). Therefore, the Input Levels controls are provided.

There are two ways that the user can adjust the levels of the Left and Right input signals: 1) knob control via the external processor front panel, and 2) software slider control via the Master Control screen. Note that in the STAND-ALONE processing mode, only the knob control is available.

Selection of the level control method is accomplished in the Settings menu as shown in Figure 4-2. When "Enable Software Input Level Controls" is checked, the sliders located under the Input Levels bargraphs on the Master Control screen are used; moving the sliders up and down increase and decrease, respectively, the input audio levels, and the "0" buttons are used to instantly set the levels precisely to the 0dB point (especially useful when the DIGITAL input is used and it is desirable not to change the audio level during the processing). Otherwise, if unchecked the INPUT LEVELS knobs on the external processor are used; rotating the knobs clockwise and counterclockwise increases and decreases the levels, respectively.

Figure 4-2: Software Input Level Control Enabling

Indication of the input audio levels is similarly accomplished via both the Master Control software screen and the external processor front panel. Interpretation of the Master Control bargraphs is illustrated in Figure 4-3; interpretation of the external processor front panel "tricolor" LEDs is illustrated in Figure 4-4.

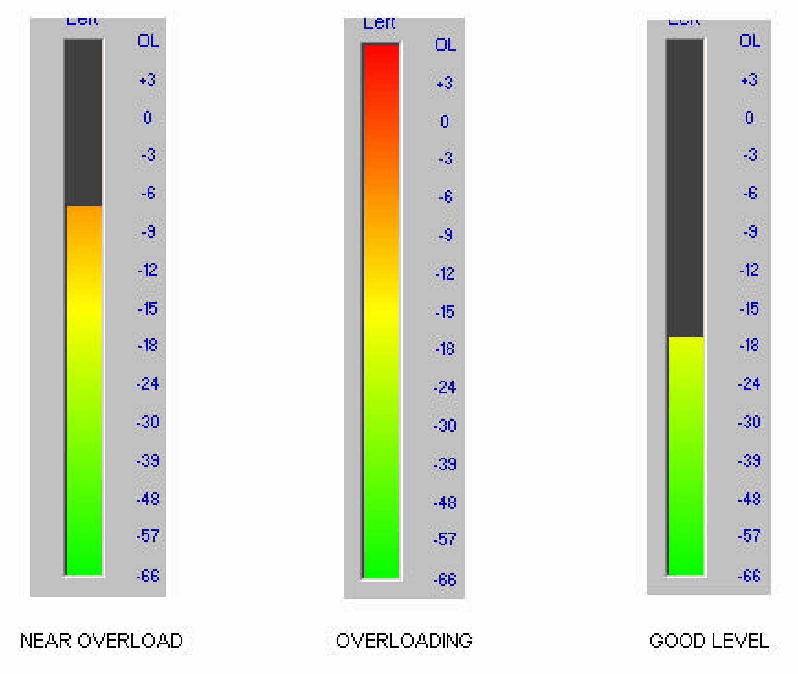

Figure 4-3: Master Control Bargraph Interpretation

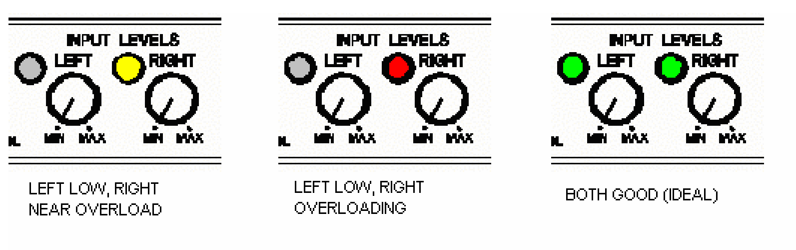

Figure 4-4: Tricolor LED Interpretation

4.2.2: Digital Input Channel Swapping

Application:

> *Particularly on occasions where a digital audio tape recorder (DAT) is used to record "Primary" (e.g., microphone) audio and "Reference" (e.g., TV) audio simultaneously onto the same tape, it is possible that the channels may be swapped; e.g., the Primary audio may be recorded onto the Right audio channel and the Reference audio recorded on the Left audio channel. If this has occurred, then when inputting the audio digitally to the PCAP II Plus, the Ref Canceller filter would normally not be able to provide cancellation of the Reference audio, as it would be on the wrong channel.*
>
> *Traditionally, it has been necessary to first convert such recorded signals to analog and then swap the cables going into the analog inputs; now, however, the new Channel Swap feature allows the channels to be swapped digitally without having to convert to analog, thereby improving the results of the Ref Canceller processing.*

The Channel Swap feature is accessed via the Settings menu as shown in Figure 4-5. When checked, "Swap Digital Input Channels" causes the Left and Right audio channels of the DIGITAL input to be transposed (Left -> Right, Right -> Left).

NOTE: this ONLY affects the DIGITAL input, not the ANALOG INPUTS.

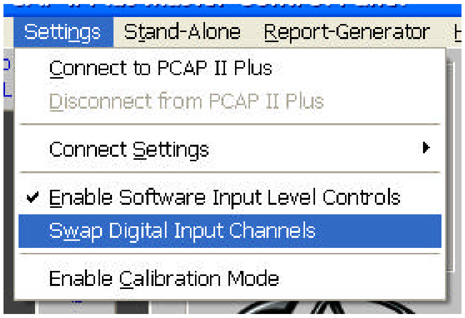

Figure 4-5: Channel Swap Feature Selection

When the Channel Swap feature is activated, the Input Levels section of the Master Control screen will appear as shown in Figure 4-6.

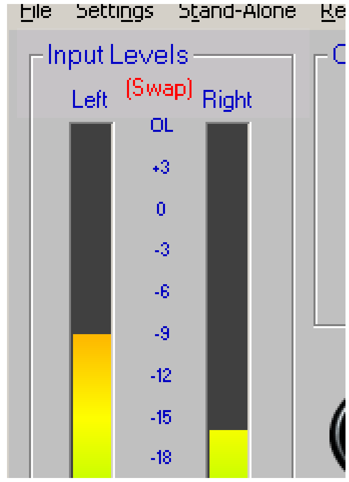

Figure 4-6: Channel Swap Feature Indication

4.2.3: Input and Output Highpass Filters (HPFs)

Application:

> *The Input HPFs are used to remove rumble or other low-frequency noises which occur below 100 Hz to 500 Hz (adjustable) from the input signals before they enter the digital processors. Since very little speech information is lost in bandlimiting below these frequencies, this filter is recommended for voice enhancement. However, for wide bandwidth and hifi signals, the low-frequency cutoff may result in some loss of desired signal components.*
>
> *The Output HPFs function the same as the Input HPFs, except that they remove low-frequency noise <u>after</u> the input signal has passed through the digital processors. (Signal processing may restore some of the low-frequency energy.)*

Clicking on the **Active** button within the **Input HPF** block will cause its RED LED to toggle between illuminated and not illuminated as shown in Figure 4-7; this indicates whether the Input HPFs are in (RED) or out (GRAY) of the process. Similarly, clicking on the **Active** button within the **Output HPF** block will toggle the Output HPFs in and out of the process.

Figure 4-7 Input/Output HPF Active Button

For fast access* to the Input and Output HPFs, pressing <Alt-I> or <Alt-O> may also be used to toggle the Input or Output HPFs, respectively, in or out of the process.

Adjustment of the Input HPF cutoff frequency is accomplished by clicking on the Input HPF **Config** button. When this button is pressed, the window shown in Figure 4-8 appears.

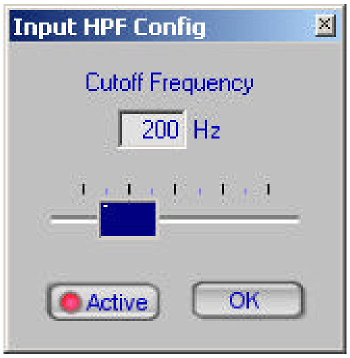

Figure 4-8 Input HPF Config Window

Specify the cutoff frequency in 10 Hz increments using the slider control. Minimum cutoff frequency is 100 Hz, while the maximum cutoff frequency is 500 Hz. Similarly, the Output HPF cutoff frequency can be adjusted by clicking on the Output HPF **Config** button to bring up the Output HPF Config Window (identical to the Input HPF Setup Window shown in Figure 4-8).

NOTE: Both the Input HPF and Output HPF **Active** buttons are overridden by the **Processor Enabled** button. When the **Processor Enabled** LED is not illuminated, both the Input HPF and the Output stages HPF will be bypassed, regardless of their buttons' status.

*Fast access keys use the Alternate key and the underlined letter simultaneously.

4.2.4: Digitally-Controlled Limiter

Application:

> *The dual-channel input Limiter automatically protects the input circuits from overload distortion by reducing input signal levels whenever loud sounds, such as door slams, exceed a specified Threshold. When the overload goes away, the Limiter returns the input signal levels to their original settings over the specified Release Time interval. If no loud sounds exceed the specified Threshold, the Limiter will not affect the input signals. In Stereo Input Modes, the two inputs may be linked, allowing an overload on any input channel to reduce the gain equally on all inputs, which minimizes the impact on signal processing and maintains proper stereo perspective.*

Clicking on the **Active** Button within the Limiter block will cause the RED LED to toggle on and off as shown in Figure 4-9, indicating whether the Limiter is in (RED) or out (GRAY) of the process.

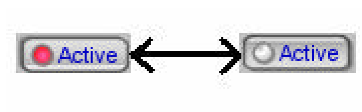

Figure 4-9 Limiter Active Button

For fast access to the Limiter, pressing <Alt-L> may also be used to toggle the Limiter in or out of the process.

NOTE: The Limiter **Active** Button is never overridden by the **Processor Enabled** button.

The Limiter **Config** button is used to view the dialog that adjusts the Limiter control settings. When this button is pressed, the window shown in Figure 4-10 appears.

This window allows the user to adjust the Release Time, Threshold, and Link settings of the Limiter.

Release Time specifies how quickly input signal levels will return to normal after an overload condition goes away; the shorter the Release Time, the more quickly the levels will return to normal. The Release Time options are 0.125 seconds (fastest), 0.250 seconds, 0.500 seconds, and 1.0 seconds (slowest). *The 0.250 seconds setting is recommended for voice applications*. Release Time settings longer than 0.250 seconds may result in excessively long periods of reduced signal level after an overload occurs.

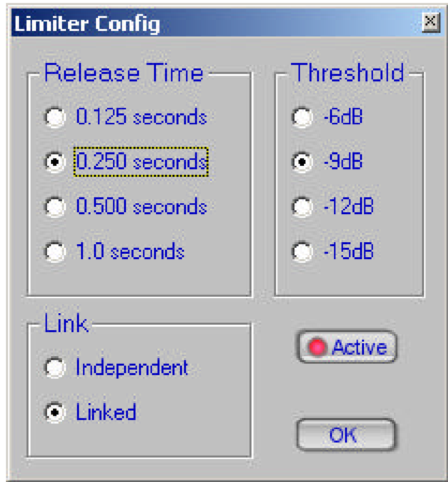

Figure 4-10 Limiter Config Window

Threshold specifies the input bargraph level which is considered to be an overload condition; any time an input signal exceeds this level, the gain will be decreased for that input until it no longer exceeds the Threshold. The Threshold options are -6dB (highest), -9dB, -12dB, and -15dB (lowest). *The -9dB setting is recommended for most applications*. In general, Threshold settings lower than -9dB will provide better overload protection, but with significant sacrifice in peak signal level (dynamic range).

The Link option is provided in Stereo Input Modes to give the user the option of making the input channels Linked or Independent (In Mono configuration, the extra inputs are used only with the Reference Canceller, and are always Linked). When Linked, the Limiter gains are equal at all times for all channels, and are reduced in response to an overload on any input channel. When Independent, the Limiter gains are not necessarily equal, each reduced in response to only its own input channel. For stereo or multiple microphone applications, the Limiter inputs should be Linked to avoid annoying Left/Right differential gain shifts and loss of stereo perspective; however, Independent Limiter gains should be used when processing two unrelated monophonic signals in a Stereo Input Mode.

When the limiter is active, red markers will appear in the Input Level bargraphs on the Master Control Panel. These markers indicate the limiter threshold setting.

4.2.5: Digitally-Controlled AGC

Application:

> *The dual output Automatic Gain Control automatically attempts to boost low-level output signals to a peak reference level (-18dB bargraph level) by gradually increasing output signal gain over a specified Release Time interval until either the proper level or Maximum Gain has been reached. This compensates for near party/far party conversations and for losses in signal level which may have occurred during the enhancement process. If the output signal levels are at or above the -18 dB reference level, the AGC will have no effect.*

Clicking on the **Active** Button within the AGC block will cause its RED LED to toggle on and off as shown in Figure 4-9, indicating whether the AGC is in (RED) or out (GRAY) of the process.

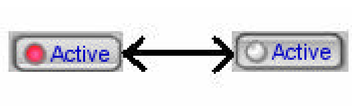

Figure 4-11 AGC Active Button

For fast access to the AGC, pressing <Alt-G> may also be used to toggle the AGC in or out of the process.

NOTE: The AGC's **Active** Button is overridden by the **Processor Enabled** button. When the LED on the **Processor Enabled** button is not illuminated, the AGC stage will be bypassed.

When the AGC **Config** button is clicked, the window in
Figure 4-12 will appear. This window allows the user to adjust the Release Time, Maximum Gain, and Link settings of the AGC.

Release Time specifies how quickly the AGC will react to decreases in output signal level. The shorter the Release Time, the more quickly the AGC will react. The Release Time options are 50 milliseconds (fastest), 100 milliseconds, 200 milliseconds, and 400 milliseconds (slowest). *For most voice applications, the 200 milliseconds setting is recommended*. Release Time settings less than 200 milliseconds may result in annoying "pumping" sounds as the AGC changes gain during rapid-fire conversations.

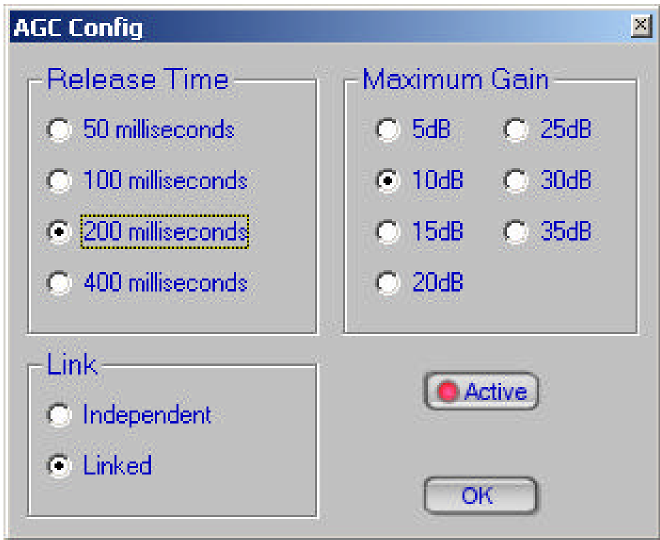

Figure 4-12 AGC Config Window

Maximum Gain specifies how much gain the AGC can apply in its attempt to bring the output signal up to the proper level. The greater the Maximum Gain, the lower the output signal level that can be brought up to proper level. The Maximum Gain options are 5dB, 10dB, 15dB, 20dB, 25dB, 30dB, and 35dB. *For most near party/far pary applications, the 10dB setting is recommended*. Maximum Gain settings greater than 10dB may elevate background noise during pauses in speech. A "soft AGC" using 5dB is often useful even when large voice level differences are not present.

The Link option is provided in Stereo Input Mode settings to give the user the option of making the AGC gains on the two output channels Linked or Independent (In Mono Input Mode, the linking option is irrelevant). When Linked, the AGC gains are equal at all times, and are increased in response to the largest signal level of the two output channels. When Independent, the AGC gains are not necessarily equal, each increased in response to only its own output channel. For most applications, the AGC outputs should be Linked to avoid annoying Left/Right differential gain shifts and loss of stereo perspective; however, Independent AGC gains should be used when processing two unrelated monophonic signals in a Stereo Input Mode.

4.3: DIGITAL PROCESSING SELECTION

4.3.1: General Filter Stages Filter Selection

To select the type of filtering to be applied by any of the General Filter Stages, use the filter select combo box (or "drop-down menu") shown in Figure 4-13. To select the filter, first click on the small arrow to the right of the combo box and then, using the menu that appears, click on the desired filter.

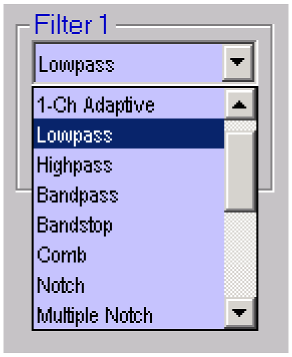

Figure 4-13 General Filter Stages Filter Selection Combo Box

Clicking the **Config** button in any General Filter blocks causes the control window for the selected filter type to appear. See Section 4.4: for detailed information on control windows for each filter type.

4.3.2: Broadband Filter Stage Filter Selection

The Broadband Filter Stage follows the General Filter Stages in the processing chain. To select the type of filtering to be applied by the Broadband Filter Stage, use the filter select combo box (or "drop-down menu") shown in Figure 4-13. To select the filter, first click on the small arrow to the right of the combo box and then, using the menu that appears, click on the desired filter.

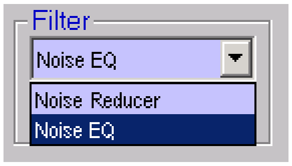

Figure 4-14 Broadband Filter Stage Filter Selection Combo Box

Clicking the **Config** button causes the control window for the selected Broadband filter type to appear. See Section 4.4: for detailed information on control windows for each filter type.

4.3.3: Equalizer Stage Processing Selection

The Equalizer Stage follows the Broadband Filter Stage in the processing chain. To select the type of processing to be applied by the Equalizer Stage, use the filter select combo box (or "drop-down menu") shown in Figure 4-13. To make a selection, first click on the small arrow to the right of the combo box and then, using the menu that appears, click on the desired selection.

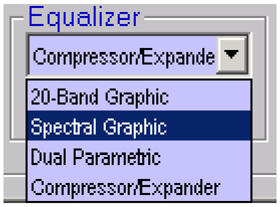

Figure 4-14 Equalizer Stage Processing Selection Combo Box

Clicking the **Config** button causes the control window for the selected Equalizer type to appear. See Section 4.5: for detailed information on control windows for each type of Equalizer stage.

4.4: GENERAL FILTER CONTROL WINDOWS

This section provides detailed description of the control window for each type of processing that the PCAP II *Plus* implements in the General Filter, Broadband Filter, and Equalizer processing stages. For any Filter or Equalizer block, the control window for the selected filter is accessed by clicking on the Config button.

4.4.1: Ref Canceller Filter

Application:

The Ref Canceller adaptive filter is used to automatically cancel from the Left channel (or "Primary") input any audio which matches the Right channel (or "Reference") input. For example, the Primary (Left) input is microphone audio with desired voices masked by radio or TV sound. The radio/TV interference can be cancelled in real-time if the original broadcast audio, usually available from a second receiver, is simultaneously connected to the Reference (Right) input.

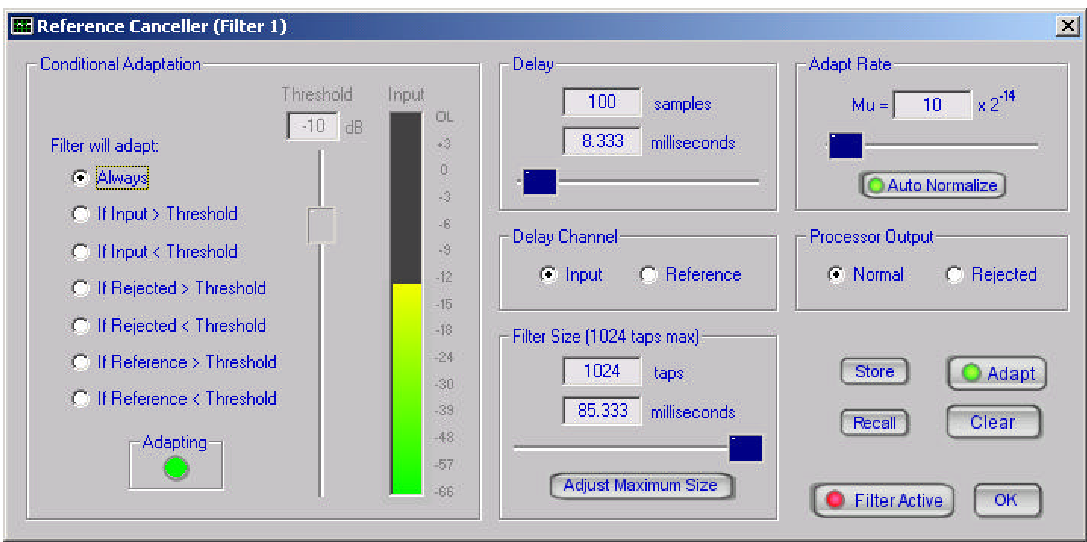

Figure 4-15 Reference Canceller Filter Control Window

Description of controls (Figure 4-15) is as follows:

Conditional Adaptiation: For advanced users only. *Novice users should keep **Filter will adapt** set to **Always***. The **Threshold** setting has no effect in this case.

Conditional Adaptation allows the adaptive filter to automatically Adapt/Freeze based upon Master Control Panel bargraph levels. This can be very useful in situations where there are pauses, or breaks, in the speech being processed.

Hint: Conditional adaptation is useful in *maintaining* adaptation once the filter has converged. Motion in the room and air temperature changes affect the filter's operation. First allow the filter to converge in **Always** and the click on **If Normal Output < Threshold**. Adjust the threshold for adaptation (**Adapting** LED is GREEN when adapting) by observing the bargraph levels when the voice is not present; note that the threshold is shown on the bargraph along with the audio level, and that the bargraph corresponds with the Left channel audio in the Stereo, Linked Input Mode selection.

Click on the **Clear** button if you desire the filter to completely readapt based upon the new Conditional Adaptation settings.

The **Adapting** LED indicator shows the current adaptation status of the filter; if the LED is illuminated GREEN, the filter is adapting, and when the LED is not illuminated the filter is not adapting. In Stereo, Linked Input Mode, only the Left channel adaptation status is displayed.

Filter Size: Used to set the number of FIR filter taps in the adaptive filter. Filter size is indicated both in taps (filter order) and in milliseconds. Minimum Filter Size is 4 taps, but can be set to as high as 6144 taps depending on System Bandwidth, Input Mode, and Number of General Filter Stages settings. *Normally, the maximum filter size possible is used in the Ref Canceller adaptive filter*

Adapt Rate: Used to set the rate at which the adaptive filter adapts to changing signal conditions (mathematically known as Mu). A Mu of 1×2^{-14} provides very slow adaptation, while a Mu of 3808 x 2^{-14} provides fastest adaptation.* *As a rule set this rate to approximately 100-200 initially, to establish convergence, then back off to a mid value to maintain cancellation.*

Auto Normalize: Selects Normalized (LED illuminated GREEN) or Fixed (LED not illuminated) adaptation rate. *Normalized is recommended*. When Auto Normalize is enabled, the specified Adapt Rate is continuously power scaled based upon the input signal level. *This generally results in faster convergence for a given Mu*. When Auto Normalize is not enabled, the specified Adapt Rate is utilized at all times without power scaling

Delay Channel: Specifies whether the delay line is to go into either the Primary (Left) channel or the Reference (Right) channel. *For most applications , a slight delay (typically 5 msec) is placed in the Primary channel*, For applications with long distances between the mike and radio/TV, a delay in the Reference channel may be required. Extreme caution should be exercised when using reference channel delay; excessive delay in that channel will not allow cancellation to take place.

Delay: Sets the number of audio samples in the delay line. Delay is indicated both in samples and in milliseconds. Minimum Delay is 1 sample, but can be set to as high as 32768 samples depending on Configuration and Number DSP Stages settings.

Processor Output: Selects Residue or Filter output. *The Residue output is the normal output selection*, which is the signal left over after the Reference signal has been cancelled from the Primary signal. The Filter ouput is the modified Reference signal being subtracted from the Primary signal.

Filter Button: Used to switch the Reference Canceller filter in and out of the process without affecting the other filters in the process. The Filter LED indicates that the filter is "in" when it is lit RED. The filter is out of the process when the Filter LED is unlit.

Adapt/Freeze Button: Used to freeze this adaptive filter independently. The filter is adapting when the Adapt LED is lit. The filter is frozen when the Adapt LED is unlit.

Clear Button: Used to reset the coefficients of the Reference Canceller to zero without

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*2^{-14} is a means of expressing 2 raised to the -14^{th} power, which equals 0.000061; 256 x 2^{-14} is, therefore, equal to 0.016.

affecting any other adaptive filters in the process.

Store Button: Used to store adaptive filter settings and coefficients. This button is disabled in Stereo-Linked mode. See section 4.4.1.1: for more information.

Recall Button: Used to load previously stored adaptive fitler settings and coefficients. This button is disabled in Stereo-Linked mode. See section 4.4.1.2: for more information.

NOTE: The same Filter, Clear, and Adapt/Freeze buttons are also available in the Filter block for each filter on the Master Control Panel.

4.4.1.1: Storing Adaptive Filter Settings and Coefficients

This section relates to both the Reference Canceller and the 1-Channel Adaptive Filter. Each of these filters allows the user to store the filter settings of the adaptive filter as well as the coefficients stored within the filter.

When clicking the **Store** button a window will appear (Figure 4-16) displaying the current working folder and showing all the previously stored adaptive filter settings files. If there have been no previous settings stored then no other files will be listed. These files have the extension "*.cof". Coefficients stored from a 1-Channel filter can be loaded by a Reference Canceller and vice versa, although this is not recommended.

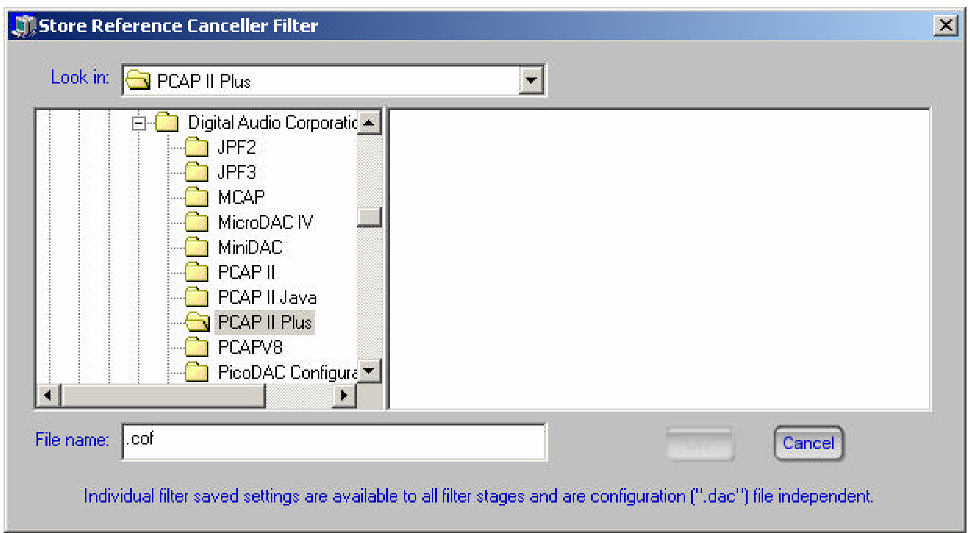

Figure 4-16 Adaptive Filter Store Settings Window

Before the **OK** button will be enabled a valid settings file name must be entered. Once a valid name is entered, click **OK** to store the adaptive filter settings. All details about the filter will be stored, including it's number of taps and the value of each coefficient. After clicking **OK,** wait a few moments as the coefficients are downloaded from the PCAP II *Plus*. The filter will be frozen (adaptation will be disabled) while this download takes place.

4.4.1.2: Recalling Adaptive Filter Settings and Coefficients

This section relates to both the Reference Canceller and the 1-Channel Adaptive Filter. Each of these filters allows the user to recall the filter settings of the adaptive filter as well as the coefficients stored within the filter.

To recall an adaptive filter's settings, click the **Recall** button on the adaptive filter's configuration window. After clicking the button the Recall window will appear (Figure 4-17) displaying the current working folder and showing all the previously stored adaptive filter settings files. If there have been no previous settings stored then no other files will be listed. These files have the extension "*.cof". Coefficients stored from a 1 Channel filter can be loaded by a Reference Canceller and vice versa, although this is not recommended.

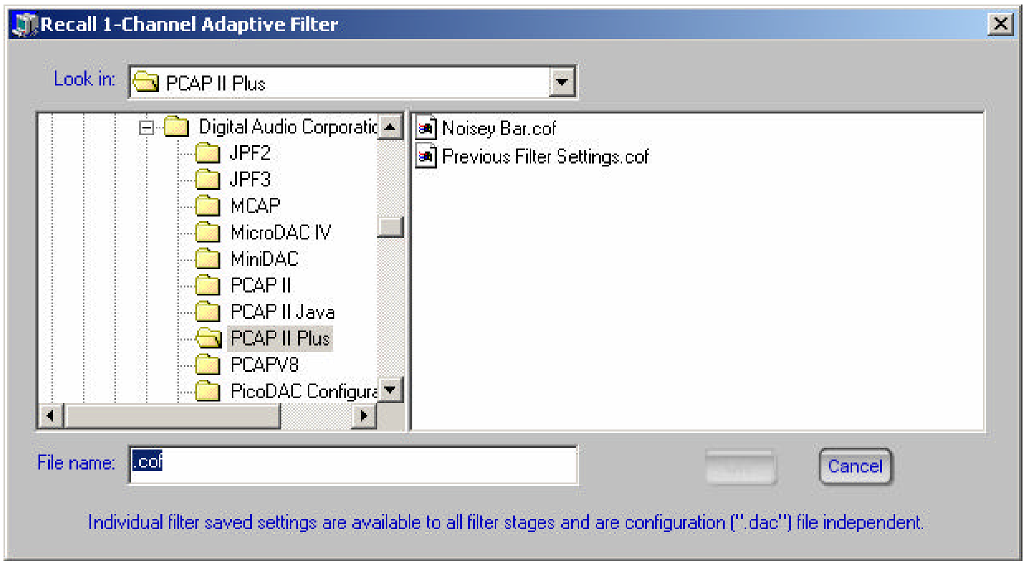

Figure 4-17 Adaptive Filter Recall Settings Window

Before the **OK** button will be enabled a valid settings file name must be entered. Once a valid name is entered, click **OK** to store the adaptive filter settings. All details about the filter will be recalled, including it's number of taps and the value of each coefficient. After clicking **OK,** wait a few moments as the coefficients are uploaded from the PCAP II *Plus*.

NOTE: If the recalled adaptive filter settings require additional PCAP II *Plus* resources that might be used by other filter stages, a warning will appear and the PCAP II *Plus* filter stages will be reconfigured.

When recalling any filter settings, the existing filter settings will be stored in a file entitled "Previous Filter Settings.cof". This feature will allow you to reload the previous settings should you decide they are needed.

Other files may be listed in the working folder entitled "Previous Filter 1 Settings.cof" or "Previous Left Filter 1 Settings.cof". These files are created when loading a PCAP II *Plus* settings (".*.dac") file that contains adaptive filters. These file contain the settings and coefficients from when this settings file was created.

Upon loading the adaptive filter settings and coefficients the filter will be frozen (the adaptation will be disabled).

4.4.2: 1-Channel Adaptive Filter

Application:

The 1-Channel Adaptive filter is used to automatically cancel predictable and convolutional noises from the input audio. Predictable noises include tones, hum, buzz, engine/motor noise, and, to some degree, music. Convolutional noises include echoes, reverberations, and room acoustics.

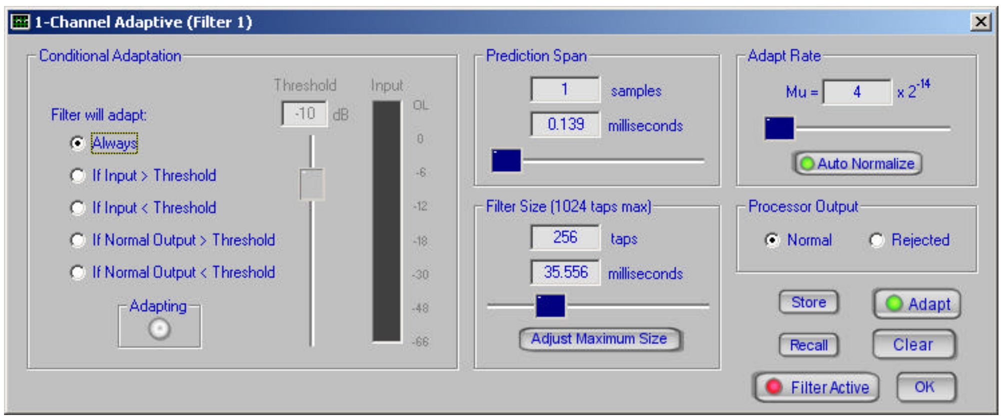

Figure 4-18 1-Channel Adaptive Filter Control Window

Description of controls (Figure 4-18) is as follows:

Conditional Adaptation: The **Threshold** setting has no effect in this case.

Conditional Adaptation allows the adaptive filter to automatically Adapt/Freeze based upon Master Control Panel bargraph levels. This can be very useful in situations where there are pauses, or breaks, in the speech being processed.

Hint: Conditional adaptation is useful in *maintaining* adaptation once the filter has converged. Motion in the room and air temperature changes affect the filter's operation. First allow the filter to converge in **Always** and the click on **If Normal Output** < **Threshold**. Adjust the threshold for adaptation (**Adapting** LED is GREEN when adapting) by observing the bargraph levels when the voice is not present; note that the threshold is shown on the bargraph along with the audio level, and that the bargraph corresponds with the Left channel audio in the Stereo, Linked Input Mode selection.

Click on the **Clear** button if you desire the filter to completely readapt based upon the new Conditional Adaptation settings.

The **Adapting** LED indicator shows the current adaptation status of the filter; if the LED is illuminated GREEN, the filter is adapting, and when the LED is not illuminated the filter is not adapting. In Stereo, Linked Input Mode, only the Left channel adaptation status is displayed.or GRAY when the filter is frozen. In Stereo, Linked configuration, two indicators are provided for each filter block (one LED for each channel).

Filter Size: Used to set the number of FIR filter taps in the adaptive filter. Filter size is indicated both in taps (filter order) and in milliseconds. Minimum Filter Size is 4 taps, but can be set to as high as 6144 taps depending on System Bandwidth, Input Mode, and Number of General Filter Stages settings.

Small filters are most effective with simple noises such as tones and music. Larger filters should be used with complex noises such as severe reverberations and raspy power hums. *A nominal filter size of 512 to 1024 taps is a good overall general recommendation.*

Adapt Rate: Used to set the rate at which the adaptive filter adapts to changing signal conditions (mathematically known as Mu). A Mu of 1 x 2^{-14} provides very slow adaptation, while a Mu of 3808 x 2^{-14} provides fastest adaptation.* *As a rule set this rate to approximately 100-200 initially, to establish convergence, then back off to a mid value to maintain cancellation.*

Larger adapt rates should be used with changing noises such as music; whereas, smaller adapt rates are acceptable for stable tones and reverberations. Larger adapt rates sometimes affect voice quality, as the filter may attack sustained vowel sounds.

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*2^{-14} is a means of expressing 2 raised to the –14th power, which equals 0.000061; 256 x 2^{-14} is, therefore, equal to 0.016.

Auto Normalize: Selects Normalized (LED illuminated GREEN) or Fixed (LED not illuminated) adaptation rate. *Normalized is recommended*. When Auto Normalize is enabled, the specified Adapt Rate is continuously power scaled based upon the input signal level. *This generally results in faster convergence for a given Mu*. When Auto Normalize is not enabled, the specified Adapt Rate is utilized at all times without power scaling

Prediction Span: Sets the number of samples in the prediction span delay line. Prediction span is indicated both in samples and in milliseconds, and can be adjusted from 1 to 4096 samples. Shorter prediction spans allow maximum noise removal, while longer prediction spans preserve voice naturalness and quality. *A prediction span of 2 or 3 samples is normally recommended.*

Filter Button: Used to switch the 1-Channel Adaptive filter in and out of the process without affecting the other filters in the process. The Filter LED indicates that the filter is "in" when it is lit RED. The filter is out of the process when the Filter LED is unlit.

Adapt/Freeze Button: Used to freeze this adaptive filter independently. The filter is adapting when the Adapt LED is lit. The filter is frozen when the Adapt LED is unlit.

Clear Button: Used to reset the coefficients of the 1-Channel Adaptive Fitler to zero without affecting any other adaptive filters in the process.

Store Button: Used to store adaptive filter settings and coefficients. This button is disabled in Stereo-Linked mode. See section 4.4.1.1: for more information.

Recall Button: Used to load previously stored adaptive fitler settings and coefficients. This button is disabled in Stereo-Linked mode. See section 4.4.1.2: for more information.

NOTE: The Filter, Clear, and Adapt/Freeze buttons are also available in the Filter block for each filter on the Master Control Panel.

4.4.3: Lowpass Filter

Application:

The Lowpass filter is used to decrease the energy level (lower the volume) of all signal frequencies above a specified Cutoff Frequency, thus reducing high-frequency noises, such as tape hiss, from the input audio. The Lowpass filter is sometimes called a "hiss filter."

The Cutoff Frequency is usually set above the voice frequency range so that the voice signal will not be disturbed. While listening to the filter output audio, the Cutoff Frequency can be incrementally lowered from its maximum frequency until the quality of the voice just begins to be affected, achieving maximum elimination of high-frequency noise.

The amount of volume reduction above the Cutoff Frequency can further be controlled by adjusting the Stopband Attenuation setting (maximum volume reduction is 60dB). The slope at which the volume is reduced from normal (at the Cutoff Frequency) to the minimum volume (specified by Stopband Attenuation) can also be controlled by adjusting the Transition Slope setting.

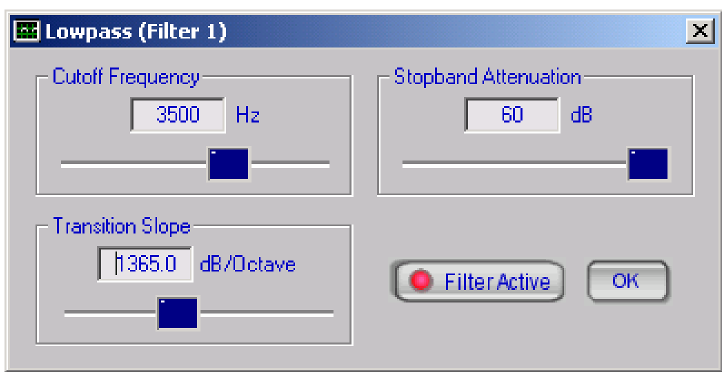

Figure 4-19 Lowpass Filter Control Window

Description of controls is as follows:

Cutoff Frequency: Specifies frequency in Hertz above which all signals are attenuated. Frequencies below this cutoff are unaffected. Minimum Cutoff Frequency is 100 Hz, while the maximum Cutoff Frequency depends upon the System Bandwidth setting. Cutoff Frequency can be adjusted in 1 Hz steps.

Stopband Attenuation: Specifies amount in dB by which frequencies above the Cutoff Frequency are ultimately attenuated. Stopband attenuation is adjustable from 10dB to 60dB in 1 dB steps.

Transition Slope: Specifies slope at which frequencies above the Cutoff Frequency are rolled off in dB per octave. Sharpest roll off occurs when Transition Slope is set to maximum, while gentlest roll off occurs when Transition Slope is set to minimum. Sharp rolloffs may cause the voice to sound hollow but will allow more precise removal of high frequency noises. Note that the indicated value changes depending upon the Cutoff Frequency and System Bandwidth settings.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the lowpass filter and its controls is given in Figure 4-20.

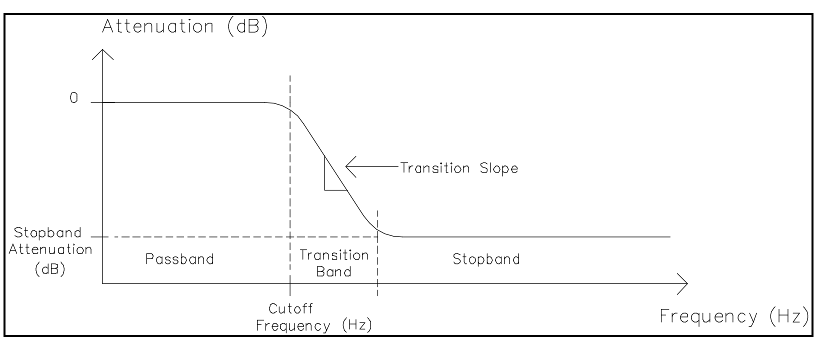

Figure 4-20 Lowpass Filter Graphical Description

4.4.4: Highpass Filter

Application:

The Highpass filter is used to decrease the energy level (lower the volume) of all signal frequencies below a specified Cutoff Frequency, thus reducing low-frequency noises, such as tape or acoustic room rumble, from the input audio (The Highpass filter is sometimes called a "rumble filter").

The Cutoff Frequency is usually set below the voice frequency range (somewhere below 300 Hz) so that the voice signal will not be disturbed. While listening to the filter output audio, the Cutoff Frequency, initially set to 0 Hz, can be incrementally increased until the quality of the voice just begins to be affected, achieving maximum elimination of low-frequency noise.

The amount of volume reduction below the Cutoff Frequency can further be controlled by adjusting the Stopband Attenuation setting (maximum volume reduction is 60dB). The slope at which the volume is reduced from normal (at the Cutoff Frequency) to the minimum volume (specified by Stopband Attenuation) can also be controlled by adjusting the Transition Slope setting.

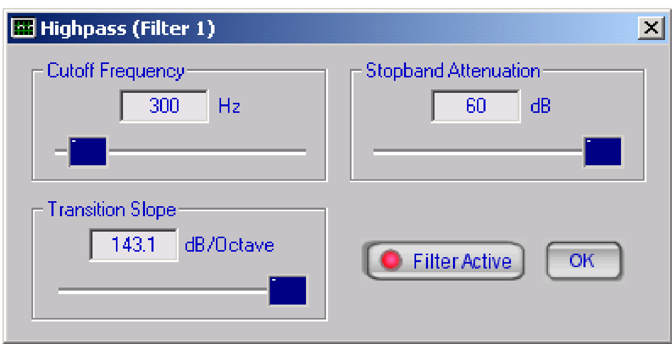

Figure 4-21 Highpass Filter Control Window

Description of controls is as follows:

Cutoff Frequency: Specifies frequency in Hertz below which all signals are attenuated. Frequencies above this cutoff are unaffected. Minimum Cutoff Frequency is 0 Hz (no frequencies attenuated), while the maximum Cutoff Frequency depends upon the System Bandwidth setting. Cutoff Frequency can be adjusted in 1 Hz steps

Stopband Attenuation: Specifies amount in dB by which frequencies below the Cutoff Frequency are ultimately attenuated. Stopband attenuation is adjustable from 10dB to 60dB in 1 dB steps.

Transition Slope: Specifies slope at which frequencies below the Cutoff Frequency are attenuated in dB per octave. Sharpest attenuation occurs when Transition Slope is set to maximum, while gentlest attenuation occurs when Transition Slope is set to minimum. Note that the indicated value changes depending upon the Cutoff Frequency and System Bandwidth settings.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the “X” button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the highpass filter and its controls is as follows in Figure 4-22.

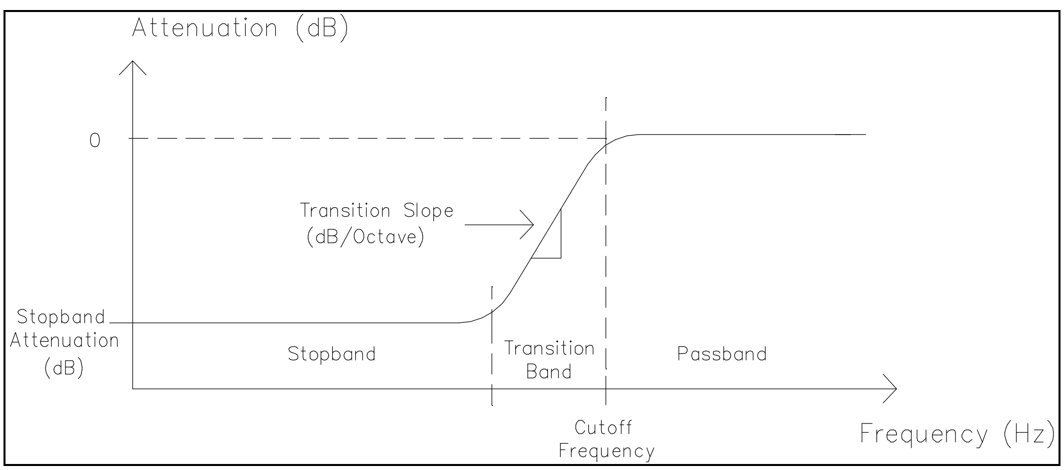

Figure 4-22 Highpass Filter Graphical Description

4.4.5: Bandpass Filter

Application:

The Bandpass filter is used to decrease the energy level (lower the volume) of all signal frequencies below a specified Lower Cutoff Frequency and above a specified Upper Cutoff Frequency, thus combining the functions of a Lowpass and Highpass filter connected in series into a single filter. The signal region between the Lower Cutoff Frequency and the Upper Cutoff Frequency is called the passband region. The Bandpass filter is useful for simultaneously reducing both low-frequency rumble and high-frequency hiss.

The Lower Cutoff Frequency is usually set below the voice frequency range (somewhere below 300 Hz) so that the voice signal will not be disturbed. While listening to the filter output audio, the Lower Cutoff Frequency, initially set to 0 Hz, can be incrementally increased until the quality of the voice just begins to be affected, achieving maximum elimination of low-frequency noise.

The Upper Cutoff Frequency is usually set above the voice frequency range (somewhere above 3000 Hz) so that the voice signal will not be disturbed. While listening to the filter output audio, the Upper Cutoff Frequency, initially set to its maximum frequency, can be incrementally lowered until the quality of the voice just begins to be affected, achieving maximum elimination of high-frequency noise.

The amount of volume reduction outside the passband region can further be controlled by adjusting the Stopband Attenuation setting (maximum volume reduction is 60dB). The slope at which the volume is reduced from normal (at each Cutoff Frequency) to the minimum volume (specified by Stopband Attenuation) can also be controlled by adjusting the Transition Slope setting.

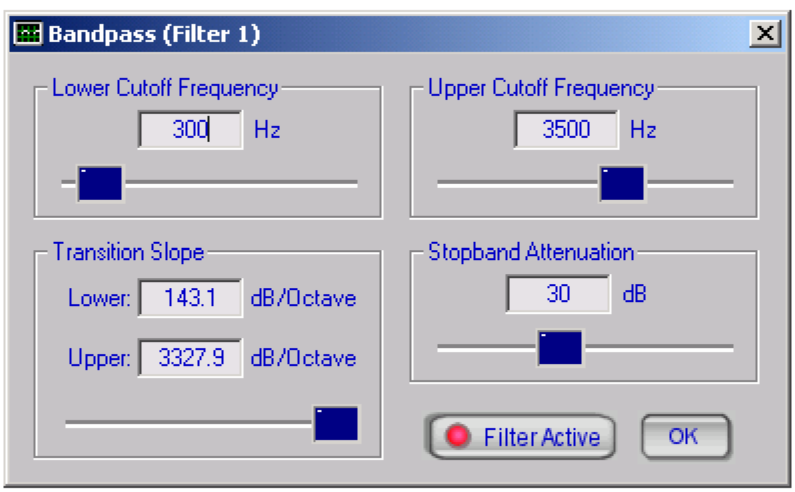

Figure 4-23 Bandpass Filter Control Window

Description of controls is as follows:

Lower Cutoff Frequency:	Specifies frequency in Hertz below which all signals are attenuated. Frequencies between this cutoff and the Upper Cutoff Frequency are unaffected. Minimum Lower Cutoff Frequency is 0 Hz, while the maximum Lower Cutoff Frequency is 10 Hz below the Upper Cutoff Frequency. Lower Cutoff Frequency can be adjusted in 1 Hz steps. **NOTE:** The Lower Cutoff Frequency can <u>never</u> be set higher than 10 Hz below the Upper Cutoff Frequency.
Upper Cutoff Frequency:	Specifies frequency in Hertz above which all signals are attenuated. Frequencies between this cutoff and the Lower Cutoff Frequency are unaffected. Minimum Upper Cutoff Frequency is 10 Hz above the Lower Cutoff Frequency, while the maximum Upper Cutoff Frequency depends upon the System Bandwidth setting. Upper Cutoff Frequency can be adjusted in 1 Hz steps. **NOTE:** The Upper Cutoff Frequency can <u>never</u> be set lower than 10 Hz above the Lower Cutoff Frequency.
Transition Slope:	Specifies slope at which frequencies below the Lower Cutoff Frequency and above the Upper Cutoff Frequency are attenuated in dB per octave. Sharpest attenuation occurs when Transition Slope is set to maximum, while gentlest attenuation occurs when Transition Slope is set to minimum. Note that the indicated value changes depending upon the Cutoff Frequency and System Bandwidth settings. Also, note that the Lower and Upper Transition Slopes always have different values; this is because the frequency width of an octave is proportional to Cutoff Frequency.
Stopband Attenuation:	Specifies amount in dB by which frequencies below the Lower Cutoff Frequency and above the Upper Cutoff Frequency are ultimately attenuated. Stopband Attenuation is adjustable from 10dB to 60dB in 1 dB steps.
Filter Active Button:	Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process. **NOTE:** The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the “X” button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the Bandpass filter and its controls follows in Figure 4-24.

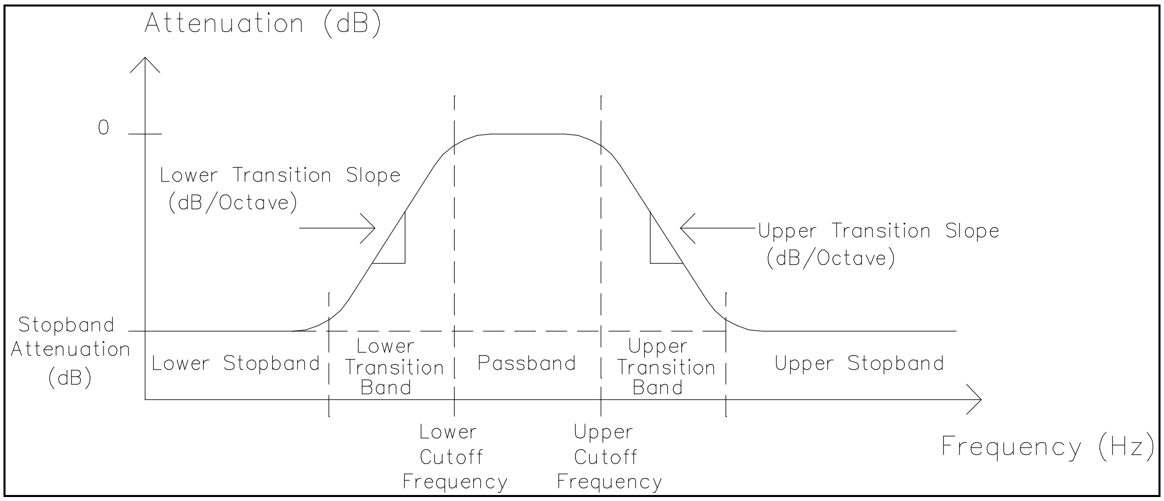

Figure 4-24 Bandpass Filter Graphical Description

4.4.6: Bandstop Filter

Application:

The Bandstop filter is used to decrease the energy level (lower the volume) of all signal frequencies above a specified Lower Cutoff Frequency and below a specified Upper Cutoff Frequency. The signal region between the Lower Cutoff Frequency and the Upper Cutoff Frequency is called the <u>*stopband*</u> *region. The Bandstop filter is useful for removing in-band noise from the input signal.*

The Lower Cutoff Frequency is usually set below the frequency range of the noise, while the Upper Cutoff Frequency is set above the frequency range of the noise. While listening to the filter output audio, the Lower and Upper Cutoff Frequencies can be incrementally adjusted to achieve maximum elimination of noise while minimizing loss of voice.

The amount of volume reduction in the stopband region can further be controlled by adjusting the Stopband Attenuation setting (maximum volume reduction is 60dB). The slope at which the volume is reduced from normal (at each Cutoff Frequency) to the minimum volume (specified by Stopband Attenuation) can also be controlled by adjusting the Transition Slope setting.

Description of controls is as follows:

Lower Cutoff Frequency: Specifies frequency in Hertz below which no signals are attenuated. Frequencies between this cutoff and the Upper Cutoff Frequency are attenuated. Minimum Lower Cutoff Frequency is 0 Hz, while the maximum Lower Cutoff Frequency is 10 Hz below the Upper Cutoff Frequency. Lower Cutoff Frequency can be adjusted in 1 Hz steps.

NOTE: The Lower Cutoff Frequency can <u>never</u> be set higher than 10 Hz below the Upper Cutoff Frequency.

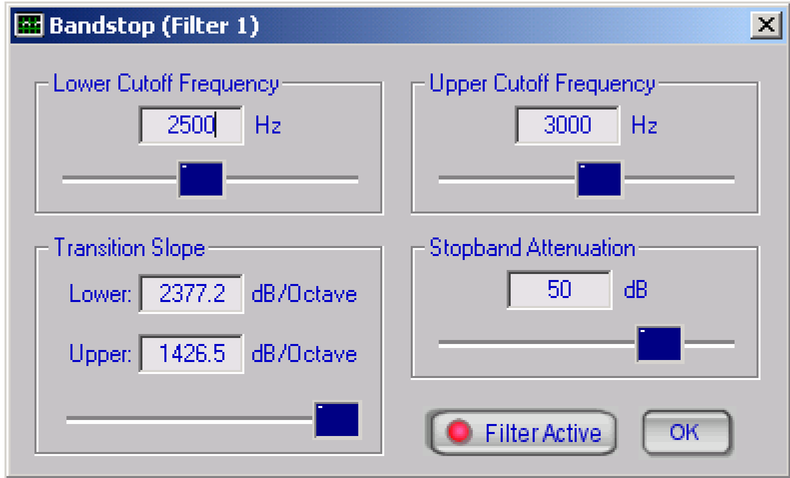

Figure 4-25 Bandstop Filter Control Window

Upper Cutoff Frequency: Specifies frequency in Hertz above which no signals are attenuated. Frequencies between this cutoff and the Lower Cutoff Frequency are attenuated. Minimum Upper Cutoff Frequency is 10 Hz above the Lower Cutoff Frequency, while the maximum Upper Cutoff Frequency depends upon the System Bandwidth setting. Upper Cutoff Frequency can be adjusted in 1 Hz steps.

NOTE: The Upper Cutoff Frequency can never be set lower than 10 Hz above the Lower Cutoff Frequency.

Transition Slope: Specifies slope at which frequencies above the Lower Cutoff Frequency and below the Upper Cutoff Frequency are attenuated in dB per octave. Sharpest attenuation occurs when Transition Slope is set to maximum, while gentlest attenuation occurs when Transition Slope is set to minimum. Note that the indicated value changes depending upon the Cutoff Frequency and System Bandwidth settings. Also, note that the Lower and Upper Transition Slopes always have different values; this is because the frequency width of an octave is proportional to Cutoff Frequency.

Stopband Attenuation: Specifies amount in dB by which frequencies above the Lower Cutoff Frequency and below the Upper Cutoff Frequency are attenuated. Stopband attenuation is adjustable from 10dB to 60dB in 1 dB steps.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the Bandstop filter and its controls follows in Figure 4-26.

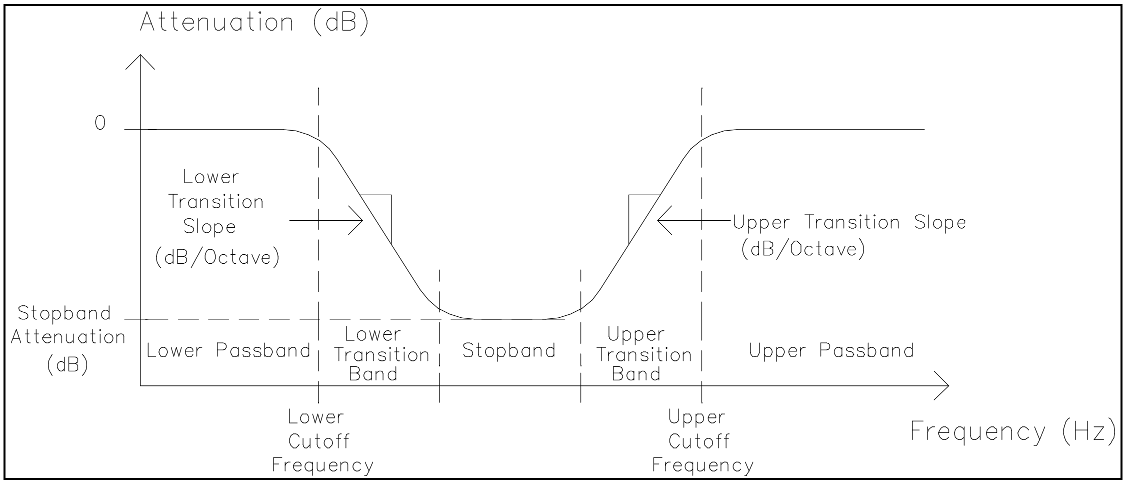

Figure 4-26 Bandstop Filter Graphical Description

4.4.7: Comb Filter

Application:

The Comb filter is used to remove, or "notch out", harmonically related noises (noises which have exactly equally-spaced frequency components), such as power-line hum, constant-speed motor/generator noises, etc., from the input audio. The filter response consists of a series of equally-spaced notches which resemble a hair comb, hence the name "Comb filter".

Adjust the Comb Frequency to the desired spacing between notches (also known as "fundamental frequency"). Set the Notch Limit to the frequency beyond which you do not want any more notches. Set the Notch Depth to the amount in dB by which noise frequency components are to be reduced.

Normally, the Notch Harmonics option will be set to ***All****, causing frequencies at* *<u>all</u>* *multiples of the Comb Frequency (within the Notch Limit) to be reduced. However, certain types of noises have only the odd or even harmonic components present. In these situations, set the Notch Harmonics option to either* ***Odd*** *or* ***Even****.*

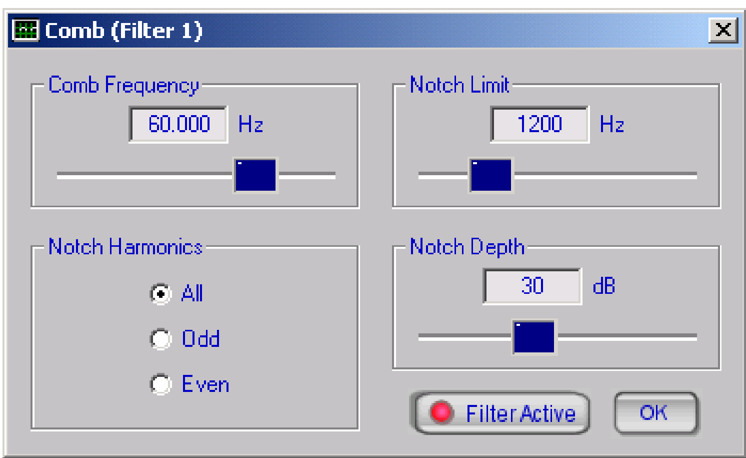

Figure 4-27 Comb Filter Control Window

Description of controls is as follows:

Comb Frequency:	Specifies fundamental frequency in Hertz of comb filter. Notches are generated at multiples, or harmonics, of this frequency. **NOTE:** Comb Frequency changes whenever the System Bandwidth setting is altered; if you change the System Bandwidth setting, you will need to readjust the Comb Frequency for any Comb Filters selected.
Notch Limit:	Specifies frequency in Hertz above which no notches are generated. Minimum Notch Limit is 100 Hz, while maximum Notch Limit depends upon the System Bandwidth setting. Notch Limit is adjustable in 50 Hz steps.
Notch Depth:	Depth of notches that are generated. Notch Depth is adjustable from 10 dB to 60 dB in 1 dB steps.
Notch Harmonics:	Specifies whether notches will be generated at All, Odd, or Even multiples, or harmonics, of the Comb Frequency. If, for example, the Comb Frequency is set to 60.000 Hz, then selecting **All** will generate notches at 60 Hz, 120 Hz, 180 Hz, 240 Hz, 300 Hz, etc. Selecting **Odd** will generate notches at 60 Hz, 180 Hz, 300 Hz, etc. Selecting **Even** will generate notches at 120 Hz, 240 Hz, 360 Hz etc.
Filter Active Button:	Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process. **NOTE:** The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.
OK Button:	Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

Hint: *A comb filter is adjusted in the following manner. Set the Notch Limit and Notch Depth to their maximum positions; set notch harmonics to **All**. Next adjust the Comb Frequency to achieve maximum hum removal; normally this will be in the vicinity of 60 or 50 Hz. (Analog recordings will seldom be exactly 50 or 60 Hz due to tape speed errors.*

*Next, adjust the Notch Limit down in frequency until the hum is bearly heard, then increase it 100 Hz. Adjust the Notch Depth up following the same procedure. Finally, select the **Odd** or **Even** if they do not increase the hum level; otherwise, use **All**.*

This procedure minimizes the filtering to only that needed for the hum. Since a comb filter is a reverberator, a 1-Channel Adaptive Filter is often placed after it to reduce the reverberation and clean up any residual noises escaping the comb filter.

A graphical description of the Comb filter and its controls follows in Figure 4-28.

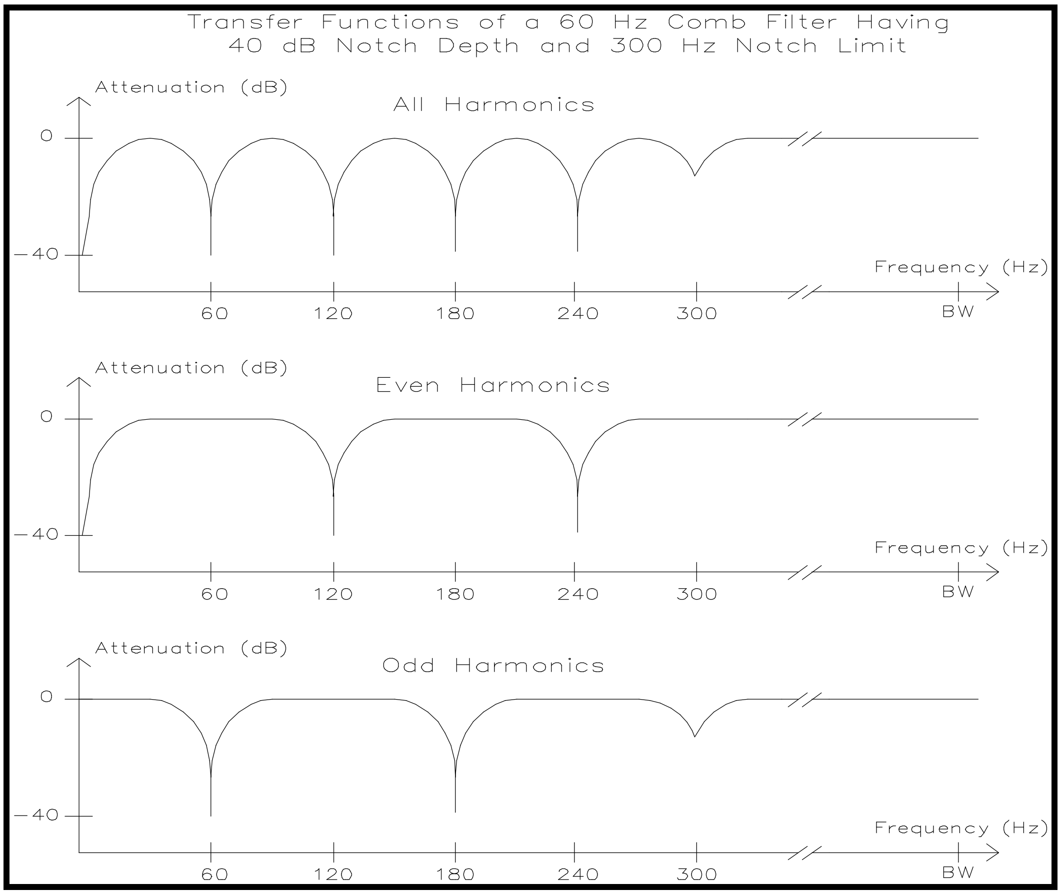

Figure 4-28 Comb Filter Graphical Description

4.4.8: Notch Filter

Application:

The Notch filter is used to remove, or "notch out", a narrow-band noise, such as a tone or a whistle, from the input audio with minimal effect to the remaining audio. The Notch filter works best with stable noise sources which have constant frequency; if the frequency of the noise source varies, then the 1-Channel Adaptive filter is recommended.

To properly utilize the Notch filter, you will first need to identify the frequency of the noise; this is best done using the Spectrum Analyzer window. See Section 4.8.2: for complete instructions on operating the Spectrum Analyzer window.

Initially set the Notch Depth to 60 dB and the Notch Width to the narrowest possible value. Next, set the Notch Frequency to the noise frequency. Fine adjustment of the Notch Frequency may be necessary to place the notch precisely on top of the noise signal and achieve maximum reduction of the noise. This is best done by adjusting the Notch Frequency up or down 1 Hz at a time while listening to the Notch filter output on the headphones.

Often, the noise frequency will not remain absolutely constant but will vary slightly due to modulation, recorder wow and flutter, and acoustic "beating." Therefore, you may need to increase the Notch Width from its minimum setting to keep the noise within the notch.

For maximum noise reduction, set the Notch Depth to 60dB. It is best to adjust the Notch Depth up from 60 dB until the tone is observed, then increase the depth 5 dB.

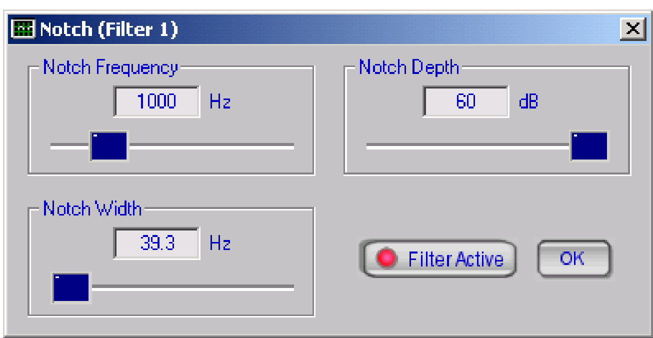

Figure 4-29 Notch Filter Control Window

Description of controls is as follows:

Notch Frequency: Specifies frequency in Hertz which is to be removed from the input audio. Minimum Notch Frequency is 10 Hz, while maximum Notch Frequency depends upon the System Bandwidth setting. Notch Frequency is adjustable in 1 Hz steps.

Notch Depth: Depth of the notch that is generated. Notch Depth is adjustable from 10 dB to 60 dB in 1 dB steps.

Notch Width: Width of the generated notch in Hertz.

NOTE: Notch Width varies with the System Bandwidth setting.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

Hint: *A notch filter is best for stable tones, as it has a sharp, or "V", bottom. If a flat-bottom, or "square", notch is needed, the bandstop or Multiple Notch filter may be preferred. Also, a 1-Channel Adaptive filter (Section 4.4.2:) is useful for automatically tracking varying tones.*

A graphical description of the Notch filter and its controls follows in Figure 4-30.

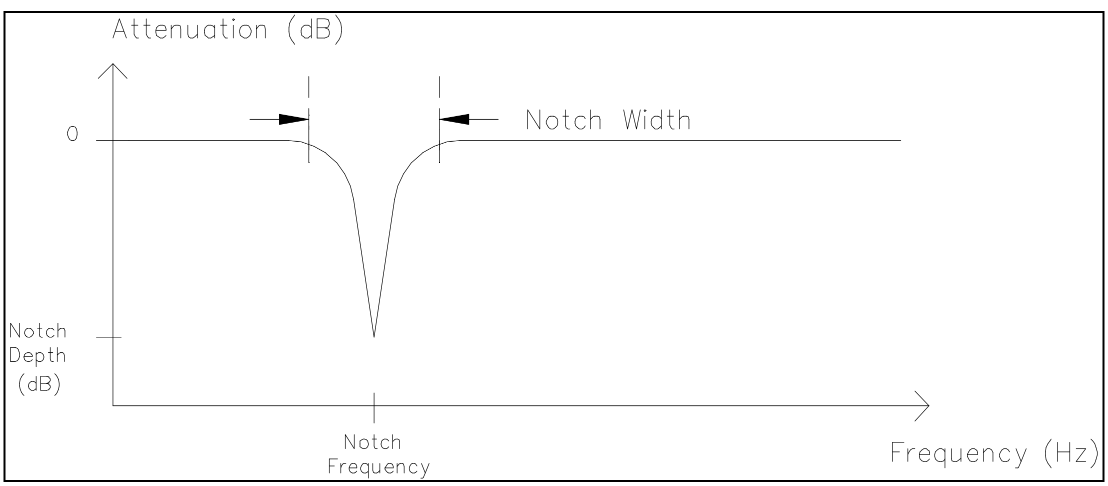

Figure 4-30 Notch Filter Graphical Description

4.4.9: Multiple Notch Filter

Application:

The Multiple Notch filter is used to remove, or "notch out", up to 16 single-frequency noises, such as tones or whistles, from the input audio with minimal effect to the remaining audio. This is accomplished using a frequency-sampling-synthesized 1024-tap FIR filter which is calculated in the PC by the PCAP II Plus Master Control program. The Multiple Notch filter, unlike the Notch filter, is able to tolerate moderate wow and flutter variances in the frequency of a noise source as it implements a "square" notch, not a "V" notch. If the frequency of a noise source varies excessively, then the 1-Channel Adaptive filter is recommended.

To properly utilize the Multiple Notch filter, you will first need to identify the noise frequencies; this is best done using the Spectrum Analyzer window. See Section 4.8.2: for complete instructions on operating the Spectrum Analyzer window.

Once the noise frequencies have been identified, set the Notch Freq(uency) of each notch to be used to the desired noise frequency.

Usually, the noise frequencies will not remain constant but will vary slightly due to modulation, wow and flutter, and acoustic "beating". Therefore, you may need to increase the Notch Width of each notch from its minimum setting to avoid having the noise move in and out of the notch.

*Once all the Notch Freqs and Notch Widths have been entered, you will need to build the actual filter using the **Build** command. Unlike the Notch Filter, which responds to controls immediately, the Multiple Notch Filter must be constructed in the computer and transferred to the external unit. A brief delay thus occurs before the filter is implemented.*

Figure 4-31 depicts the control window for notches 1 through 8. A second window for 9 through 16 is button accessed.

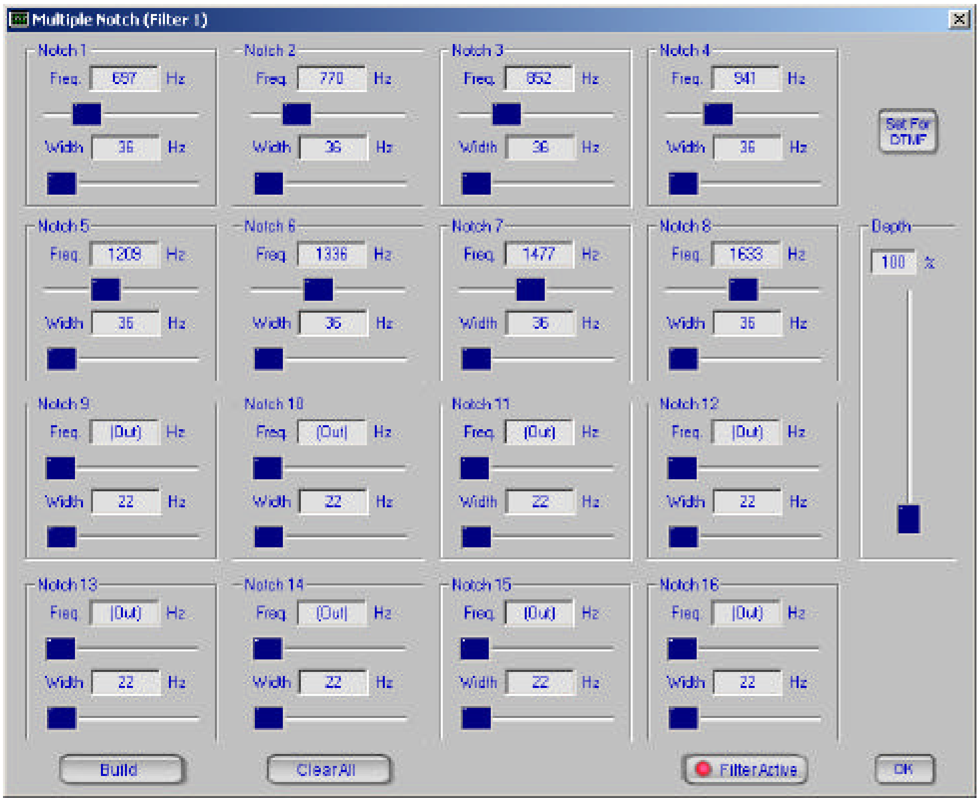

Figure 4-31 Multiple Notch Filter Control Window

Description of controls is as follows:

Notch Freq (1-16): Specifies frequency in Hertz which is to be removed from the input audio by each of the 16 notches. Minimum Notch Freq is 1 Hz, while maximum Notch Freq depends upon the System Bandwidth setting. Set Notch Freq to **Out** (scroll box in full left position) if the notch is not desired. Notch Freq is adjustable in 1 Hz steps.

Notch Width (1-16): Specifies width in Hertz for each of the 16 notches. Minimum and maximum Notch Widths and adjustment resolution depend upon the System Bandwidth setting.

Build Button: Causes the FIR filter coefficients for the Multiple Notch filter to be calculated and loaded into the external processor for implementation. While this is occurring, an "hourglass" mouse cursor will appear.

NOTE: You must click the Build button after any Notch Freq or Notch Width is changed in order for the change to take effect. You must also click the Build button after any change in System Bandwidth or Input Mode in order to rebuild the filter.

Depth Control: Used to specify the dB attenuation to be applied for all notches currently implemented. However, unlike the Notch Freq and Width controls, the Depth control is fully interactive and has instantaneous effect at all times; clicking the Build button is not necessary for the Depth control to have effect. Range of adjustment is 0 to -90.3dB.

Clear All Button: Clears all Notch Freq and Notch Width settings for all 16 notches and restores the Multiple Notch filter coefficients to an allpass filter (no notches).

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

Set for DTMF Button: This button will set the first eight notches to each of the eight specific DTMF tones. This is useful for cancelling noise introduced into phone conversation by the DTMF tones. After the button is clicked the filter must be rebuilt.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the Multiple Notch filter and its controls follows in Figure 4-32:

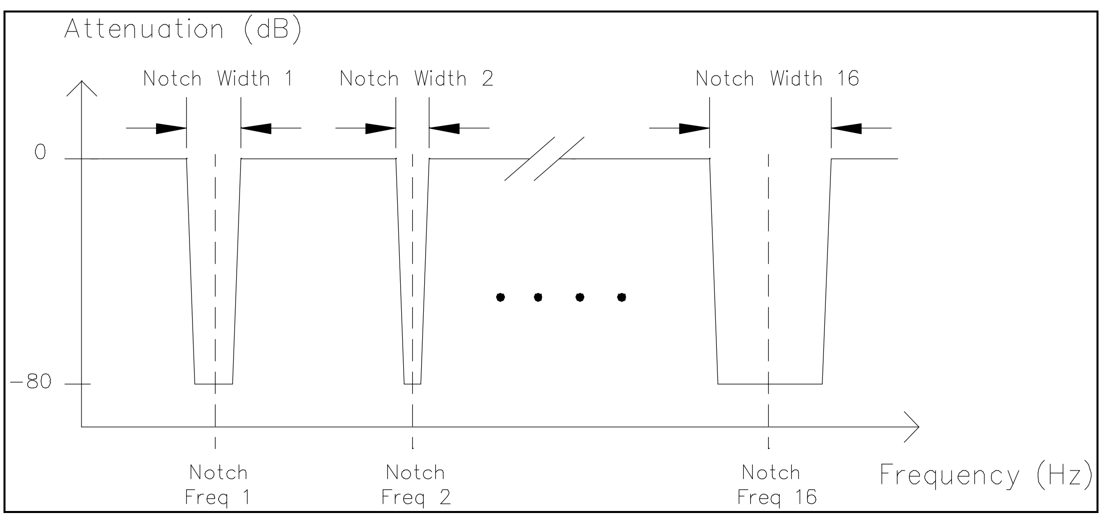

Figure 4-32 Multiple Notch Filter Graphical Description

4.4.10: Slot Filter

Application:

NOTE: The Slot filter has very little use in speech enhancement applications; the main value is in isolating other types of signals that are non-speech in nature.

The Slot filter is used to isolate, or "slot", a single-frequency signal, such as a tone or a whistle, in the input audio, attenuating all other audio. This is the exact opposite of the Notch filter function.

To properly utilize the Slot filter, you will first need to identify the frequency of the signal to be isolated; this is best done using the Spectrum Analyzer window. See Section 4.8.2: for complete instructions on operating the Spectrum Analyzer window.

Once the frequency of the signal has been identified, initially set Stopband Attenuation to 60 dB and the Slot Width to the narrowest possible value. Next, set the Slot Frequency to the signal frequency. Fine adjustment of the Slot Frequency may be necessary to place the slot right on top of the signal. This is best done by adjusting the Slot Frequency up or down 1 Hz at a time while listening to the Slot filter output on the headphones.

Usually, the signal frequency will not remain constant but will vary slightly due to modulation, recorder wow and flutter, and acoustic "beating". Therefore, you may need to increase the Slot Width from its minimum setting to avoid having the signal move in and out of the slot.

To optimize background noise reduction for your application, set the Stopband Attenuation to 60dB. If, however, you wish to leave a small amount of the background noise mixed in with the isolated signal, adjust the Stopband Attenuation to the desired value.

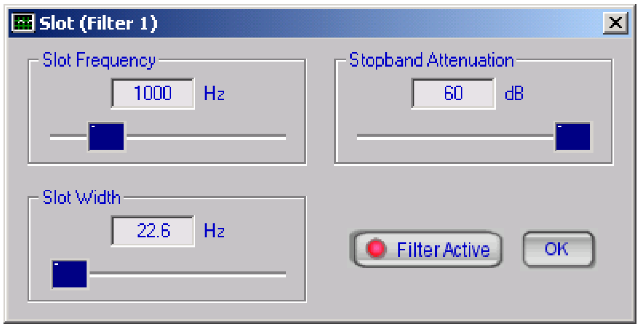

Figure 4-33 Slot Filter Control Window

Description of controls is as follows:

Slot Frequency: Specifies frequency in Hertz which is to be enhanced in the input audio. Minimum Slot Frequency is 10 Hz, while maximum Slot Frequency depends upon the System Bandwidth setting. Slot Frequency is adjustable in 1 Hz steps.

Stopband Attenuation: Specifies amount in dB by which frequencies other than the Slot Frequency are attenuated. Stopband attenuation is adjustable from 10dB to 60dB in 1 dB steps.

Slot Width: Width of the generated slot in Hertz.

NOTE: Slot Width varies with the System Bandwidth setting.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the Slot filter and its controls follows in Figure 4-34. Note that the slot width is defined at its -6 dB points.

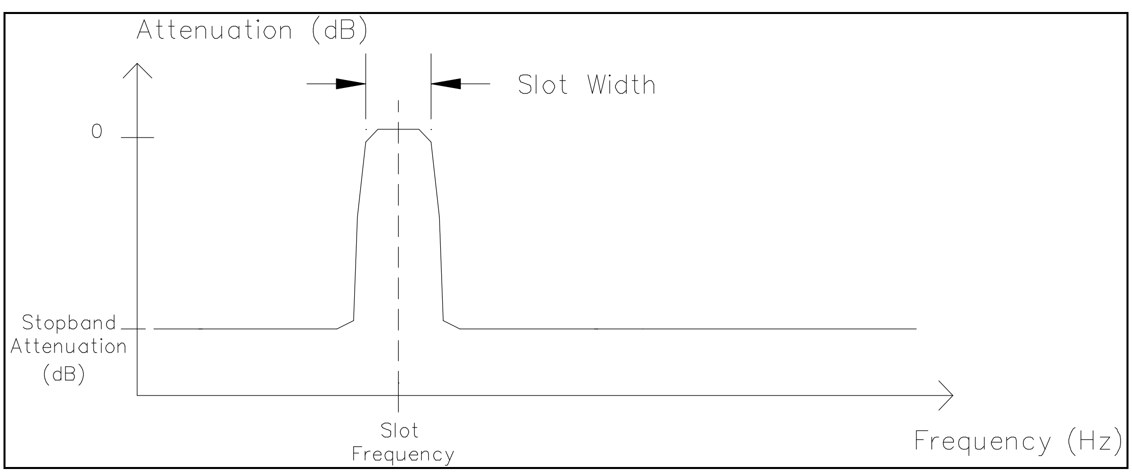

Figure 4-34 Slot Filter Graphical Description

4.4.11: Multiple Slot Filter

Application:

NOTE: The Slot filter has very little use in speech enhancement applications; the main value is in isolating other types of signals that are non-speech in nature, e.g. telephone "DTMF" tones.

The Multiple Slot filter is used to isolate, or "slot", up to 16 single-frequency signals, such as tones or whistles, from the input audio, eliminating all other audio. This is the exact opposite of the Multiple Notch filter function, and is accomplished using a frequency-sampling-synthesized 1024-tap FIR filter which is calculated in the PC by the PCAP II Plus Master Control program.

To properly utilize the Multiple Slot filter, you will first need to identify the frequencies of all signals to be isolated; this is best done using the Spectrum Analyzer window. See Section 4.8.2: for complete instructions on operating the Spectrum Analyzer window.

Once the frequencies of the signals have been identified, set the Slot Freq(uency) of each slot to be used to the desired signal frequency.

Usually, the signal frequencies will not remain constant but will vary slightly due to modulation, wow and flutter, and acoustic "beating". Therefore, you may need to increase the Slot Width of each slot from its minimum setting to avoid having the signal move in and out of the slot.

Once all the Slot Freqs and Slot Widths have been entered, you will need to build the actual filter using the ***Build*** *command. Like the Multiple Notch Filter, this filter is also constructed in the computer. A brief delay occurs after* ***Build*** *before it takes effect.*

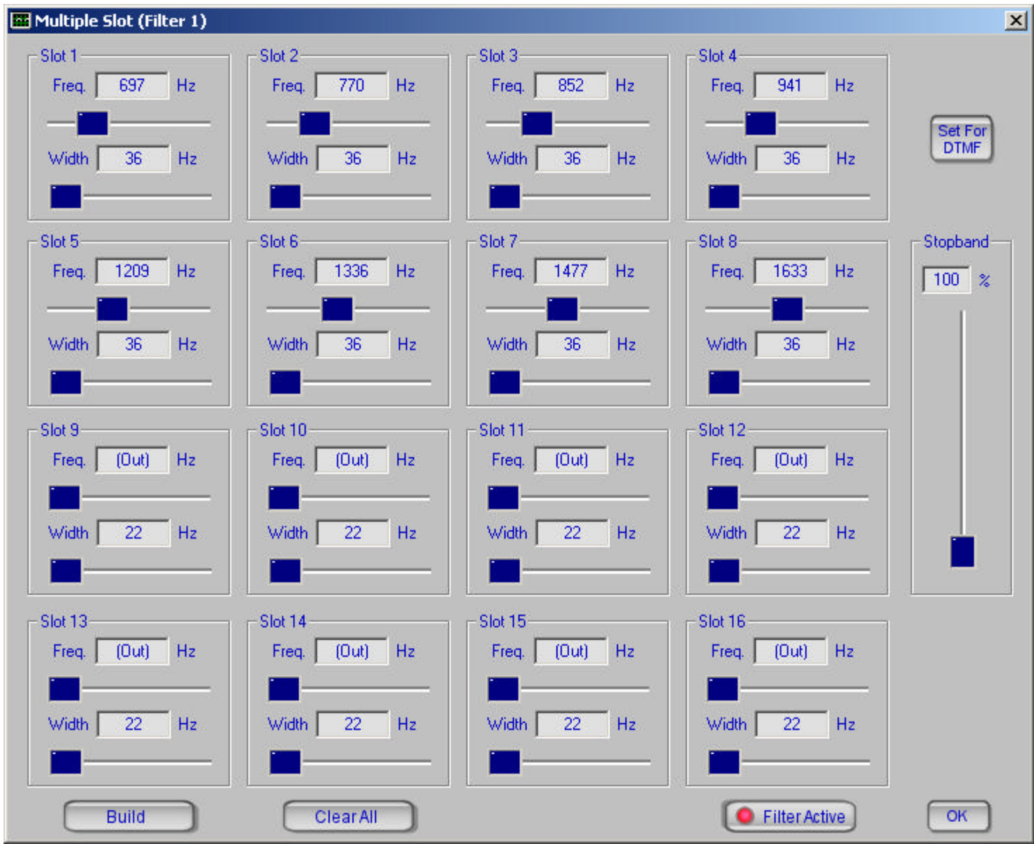

Figure 4-35. Multiple Slot Filter Control Window

Description of controls is as follows:

Slot Freq (1-16): Specifies frequency in Hertz which is to be isolated from the input audio by each of the 16 slots. Minimum Slot Freq is 1 Hz, while maximum Slot Freq depends upon the System Bandwidth setting. Set Slot Freq to **Out** (scroll box in full left position) if the slot is not desired. Slot Freq is adjustable in 1 Hz steps.

Slot Width (1-16): Specifies width in Hertz for each of the 16 slots. Minimum and maximum Slot Widths and adjustment resolution depend upon the System Bandwidth setting.

Build Button: Causes the FIR filter coefficients for the Multiple Slot filter to be calculated and downloaded to the external processor for implementation. While this is occurring, an "hourglass" mouse cursor will appear.

NOTE: You must click the Build button after any Slot Freq or Slot Width is changed in order for the change to take effect. You must also click the Build button after any change in System Bandwidth or Input Mode in order to rebuild the filter.

Stopband Control: Used to specify the dB attenuation to be applied in between the slots currently implemented. However, unlike the Slot Freq and Width controls, the Stopband control is fully interactive and has instantaneous effect at all times; clicking the Build button is not necessary for the Stopband control to have effect. Range of adjustment is 0 to -90.3dB.

Clear All Button: Clears all Slot Freq and Slot Width settings for all 16 notches and restores the Multiple Slot filter coefficients to an allpass filter (no slots).

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

Set for DTMF Button: This button will set the first eight notches to each of the eight specific DTMF tones. This is useful for isolating the tones in a telephone call. After the button is clicked the filter must be rebuilt.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

A graphical description of the Multiple Slot filter and its controls follows in Figure 4-36.

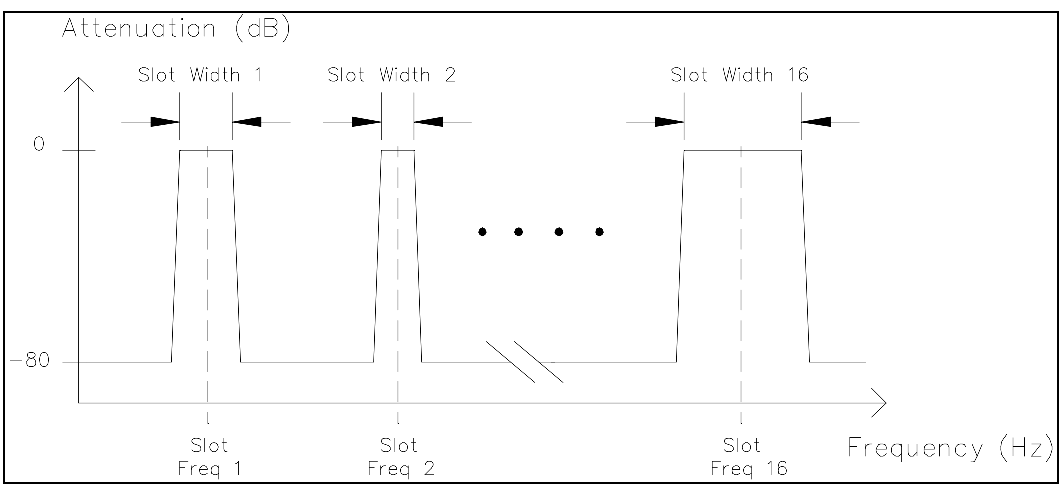

Figure 4-36 Multiple Slot Filter Graphical Description

4.4.12: Spectral Inverse Filter

Application:

The Spectral Inverse Filter (SIF) is an equalization filter which automatically readjusts the spectrum to reduce noise and muffling effects. It is especially useful when the voice has been exposed to reverberations and bandlimited noises. For an automatic implementation of the SIF, use the ASIF (see Section 4.5.3:)

SIF measures the signal's spectrum and uses this information to implement a high-resolution digital filter for correcting spectral irregularities and reduce added noises. Figure 4-37 *illustrates the process. The original audio spectrum (top trace) is inverted (middle trace). A digital filter is implemented which has the shape of this middle trace. When the original spectrum (top trace) is modified by this filter, low energy frequencies are boosted and high energy frequencies are attenuated. The resulting "filtered" audio has a flat spectrum.*

This mode of operation is called ***Equalize Voice.*** *Available controls permit the operator to reshape the output audio to flat, pink, voice-like, or custom spectrum. The operator also specifies the spectral range to be equalized using upper and lower frequency limits; audio outside these limits is attenuated. The amount of spectral correction is adjustable using the* ***Filter Amount*** *control.*

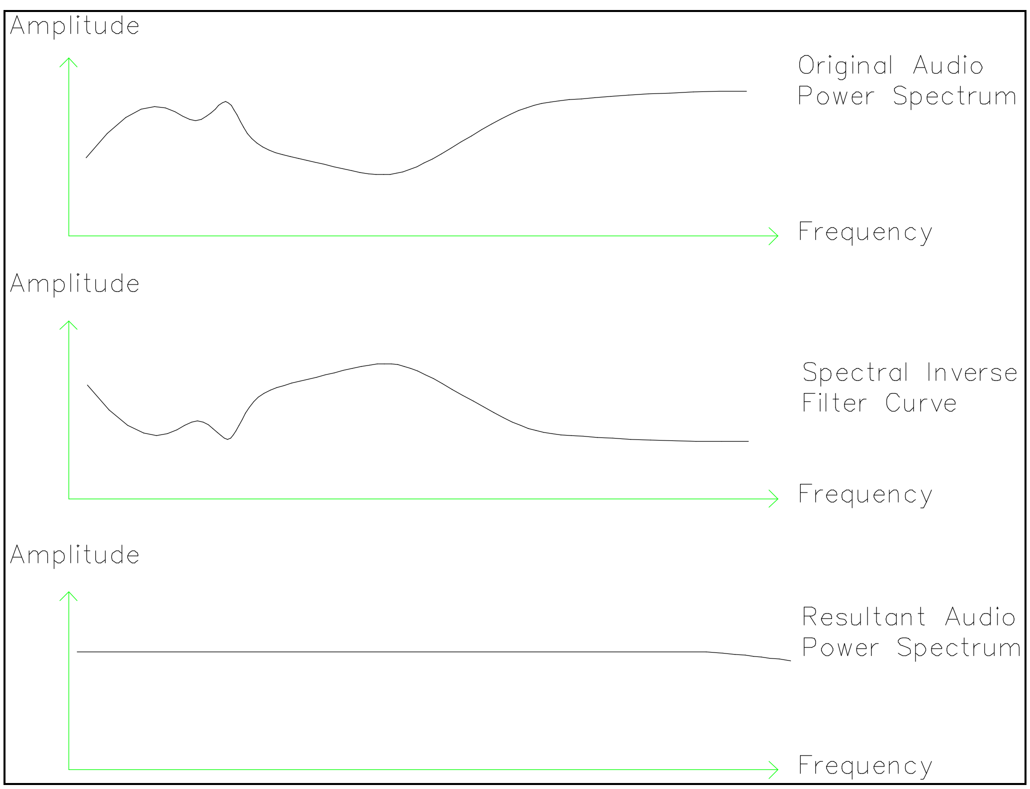

Figure 4-37 Basic Process of Spectral Inverse Filter

The equalization effect of SIF is very beneficial with reverberant audio and recordings exposed to substantial recorder wow and flutter. The noise sources must remain stationary for SIF to be effective. SIF cannot readjust itself to changing noises, such as music. In such cases, the 1-Channel adaptive filter is recommended.

*A second SIF equalization mode is **Attack Noise**. This mode is especially useful in reducing band limited noises such as horns and mechanically induced noises. The operator isolates the spectral region where the noise is present with limit cursors and the noise is precisely flattened within that region; audio outside these limits is unaffected.*

Figure 4-38 SIF Control Window When **Equalize** Voice Selected

Description of controls/indicators is as follows:

Filter Display: Used to display the original audio spectrum Input (Yellow Trace = Filter) and the spectral inverse filter curve (Blue Trace = Filter Shape). For each trace, 460 spectral lines and 70dB of dynamic range are displayed. A grid is superimposed to aid the user in determining frequency and amplitude.

Analyzer Block: Used to control the spectrum analyzer which acquires the original audio power spectrum; this spectrum is displayed and continuously updated in the Filter Display area as a yellow trace. Analyzer controls include:

- **Clear** button which is used to zero the averager memory and cause the averaged spectrum to be recalculated anew.

- **Run** button which allows the user to start (GREEN LED indication) or stop (LED unlit) update of the averaged spectrum.

- **Number of Averages** setting which allows the user to specify the degree of smoothing of the original audio power spectrum. For minimum smoothing, set to **1**; for maximum smoothing, set to **128**. A long-term power spectrum (64 to 128 averages) is best for setting up the filter.

- **Gain** control which allows the user to apply a digital gain of up to 40dB to the analyzer input, allowing low-level spectrum components to be displayed; however, if excessive gain is applied, the analyzer input will overload, causing the **Gain** label to change to **OVL** (analyzer overload) and the captions at the bottom of the Filter Display area to change color to red.

Filter Operation Block: Specifies whether SIF is to be used to **Equalize Voice** or **Attack Noise**. When **Equalize Voice** is selected, the SIF control window appears as shown in Figure 4-38. When **Attack Noise** is selected, the SIF Control Window appears as shown in Figure 4-39.

Equalize Voice operation is used to reshape the original input voice audio to a more natural-sounding spectral shape over a specified frequency range. All audio outside this frequency range is attenuated by 40dB.

Attack Noise operation is used to attack large-magnitude narrow-band noises (such as motor noises) over a specified frequency range. Audio outside this frequency range remains unaffected (0dB attenuation).

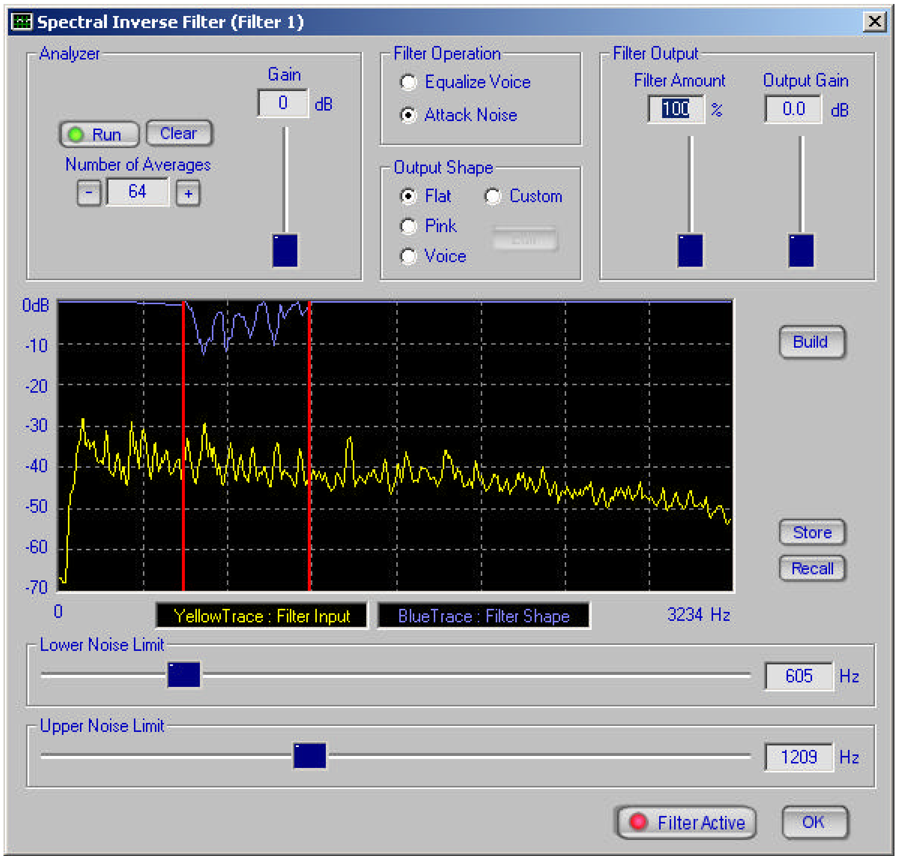

Figure 4-39 SIF Control Window When **Attack Noise** Selected

Filter Amount Block

Specifies **Filter Amount** and **Output Gain**.

Equalize Voice or Attack Noise **Filter Amount** specifies the maximum amount of volume reduction that can be applied by the inverse filter within the specified frequency limits; this may be set to the approximate difference in amplitude between the largest and smallest input spectral components within the frequency limits. This value varies between 0 and 100%. 0 indicates no filtering, 100 indicates full filtering. Varying the **Filter Amount** will update the blue trace in the Filter Display to show how the filter is affected.

NOTE: Maximum **Fitler Amount** should only be used when necessary; it may excessively elevate background noises

For **Equalize Voice** operation, the inverse filter response rolls off to -40dB outside the frequency limits. For **Attack Noise** operation, the inverse filter response rolls up to 0dB (no attenuation) outside the frequency limits.

Equalize Voice or Attack Noise **Output Gain** specifies the digital boost to be applied to the entire spectral inverse filter curve. Normally, **Output Gain** is applied in the **Equalize Voice** mode; the gain is usually 0 dB in the **Attack Noise** mode. This boost is necessary to make up for the volume reduction performed by the inverse filter. **Output Gain** should be initially set to approximately 0 dB. If **Output Gain** is applied and the filter output is distorted, reduce **Output Gain** setting and re-**Build** the filter; if filter output level is too low, try increasing the **Output Gain** setting.

Lower and Upper Voice/Noise Limits

For **Equalize Voice** operation, specifies **Lower Voice Limit** and **Upper Voice Limit**. These are the lower and upper frequency limits over which the input voice audio is equalized. Audio outside these limits is rolled off and ultimately attenuated by 40dB. A Lower Limit above 300 Hz and an Upper Limit below 3000 Hz is not recommended, as voice intelligibility may suffer.

For **Attack Noise** operation, specifies **Lower Noise Limit** and **Upper Noise Limit**. These are the lower and upper frequency limits over which noise in the input audio is attacked. These values should be set to "bracket" any noise spikes in the original audio power spectrum.

To set the upper and lower frequency limits, use the horizontal scroll bars to position the two red markers on the Filter Display to the desired frequency positions. You may also click-and-drag the red markers in the Filter Display to the desired frequency as well as enter the frequency amount in the frequency entry fields.

Output Shape Block:

Specifies the final reshaping curve to be applied to the entire SIF filter. For **Attack Noise** only **Flat** should be used. For **Equalize Voice** four curves are available and include **Flat** (no reshaping), **Voice** (6dB/octave rolloff above and below 500 Hz), and **Pink** (3dB/octave rolloff above 100 Hz), and **Custom**. The **Voice** and **Pink** curves are provided to reshape the resultant audio power spectrum to that of a typical voice spectrum; the **Voice** curve provides "hard" reshaping, while the **Pink** curve provides softer reshaping of the spectrum.

When selecting the **Custom** option the **Edit** button will be enabled. Clicking the **Edit** button will display the **SIF Custom Curve** edit window (Figure 4-40). This window operates exactly the same way as the Hi-Res Graphic filter. For operation see Section 4.4.13: .

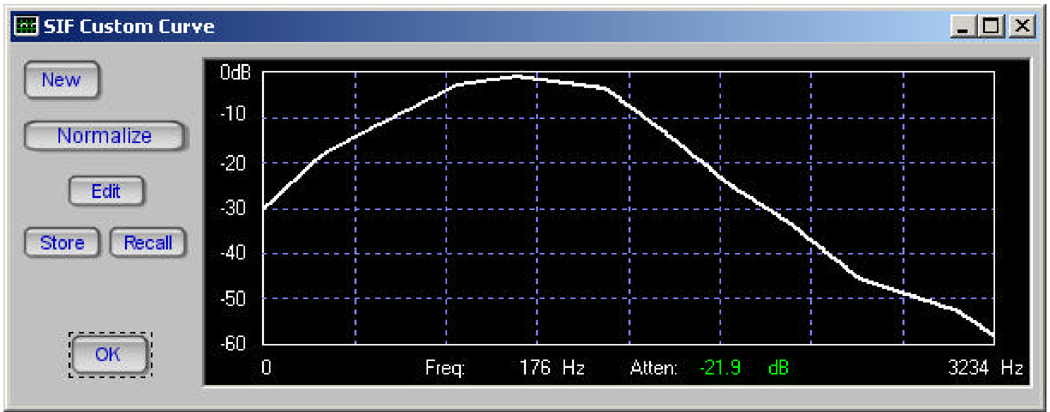

Figure 4-40 SIF Custom Curve Window

Build Button: Builds the spectral inverse filter based on the original input audio spectrum and the SIF control settings. When clicked, the mouse cursor will change to an "hourglass" shape, indicating that the PC is busy calculating the spectral inverse filter coefficients and sending them to the external processor. When the filter build is complete, the mouse cursor will return to normal and the calculated spectral inverse filter curve will be displayed as a blue trace in the Filter Display area.

Hint: Before clicking the **Build** button, it is recommended that the spectrum analyzer be set to **Freeze** to allow experimentation with the control settings for the same input spectrum.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Active button is also available in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the “X” button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

Store Button: This button allows the user to store a calculated spectral inverse filter curve to a file, which will not be lost when the computer is turned off. Clicking this button brings up the **Store Spectral Inverse Filter** window (Figure 4-41) referencing the current working folder. Once a valid file name is entered in the **File Name** field the **OK** button in this window will be enabled. Clicking the **OK** button will recall the selected file from disk. These stored settings are bandwidth independent, although they will will only be relevant for the bandwidth in which they were originally created

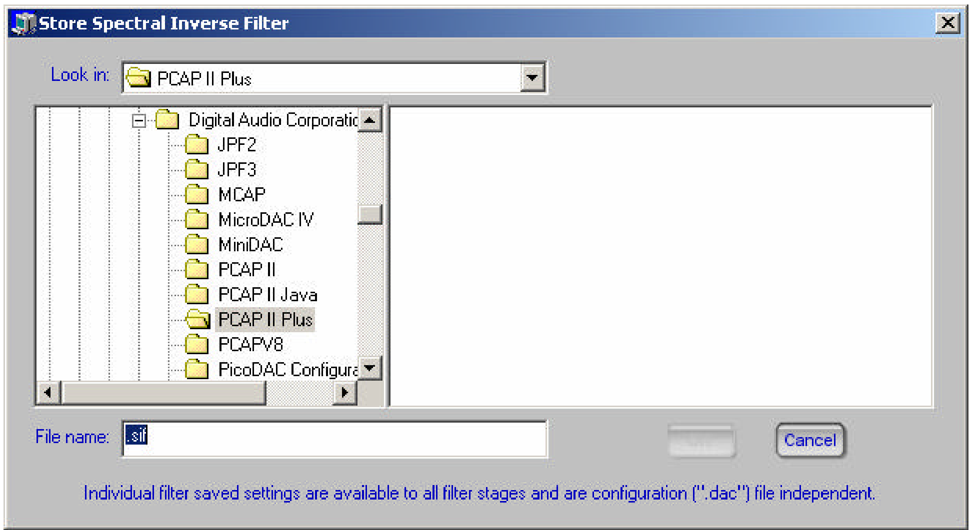

Figure 4-41 SIF Store Settings Window

Recall Button: This button allows the user to recall a previously stored spectral inverse filter curve from a file, Clicking this button brings up the **Recall Spectral Inverse Filter** window (Figure 4-42) referencing the current working folder. Once a valid file name is entered in the **File Name** field the **OK** button in this window will be enabled. Clicking the **OK** button will recall the selected file from disk. These stored settings are bandwidth independent, although they will will only be relevant for the bandwidth in which they were originally created.

NOTE: When recalling an stored filter the current filter settings will be stored to a file entitled “Previous Filter Settings.sif”.

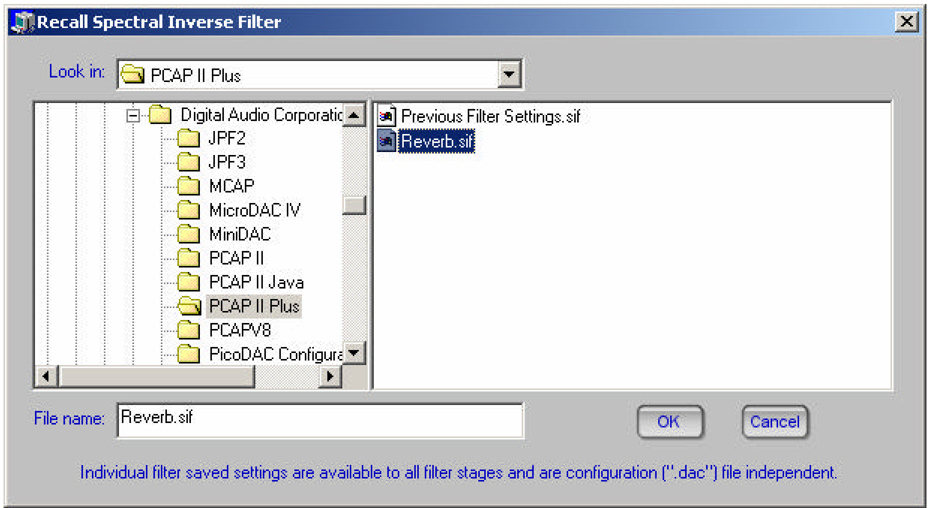

Figure 4-42 SIF Recall Settings Window

Examples of Spectral Inverse Filters:

In the examples below, SIF will equalize a voice spectrum using various **Equalize Voice Filter Amounts** and **Output Shapes**. When the **Filter Amount** is small, only the peaks in the spectrum are flattened. As the **Filter Amount** is increased, lower energy segments are equalized. The top trace in each of the figures below gives the filter curve and the bottom trace gives the original input spectrum. In Figure 4-43, Figure 4-44, and Figure 4-45 the EQ Range is increased. Each increase in Range is accompanied by an increase in compensating Gain. Compare the filter curve (top) to the original input spectrum (bottom). As the **Filter Amount** is increased, more peaks are attenuated. Figure 4-45 completely compensates all peaks and valleys. Note also how the 40 dB attenuation outside the two Limits is affected.

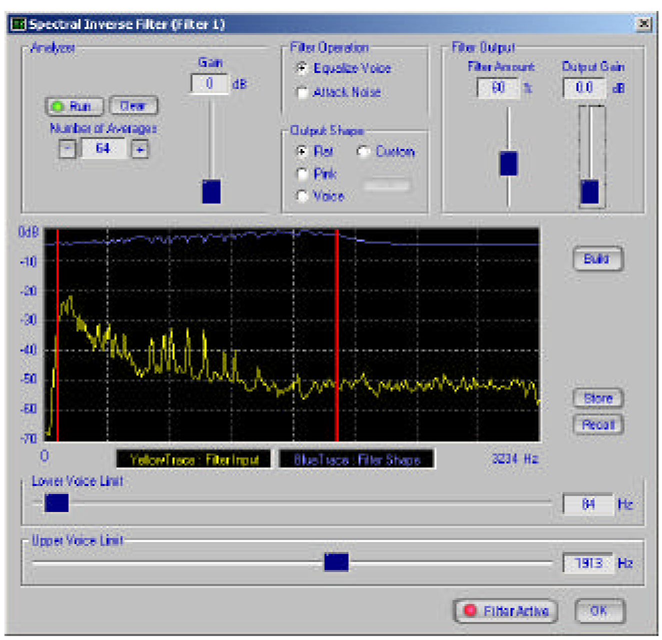

Figure 4-43 **Equalize Voice** operation, **Filter Amount** set to 60%, **Output Shape** set to **Flat**

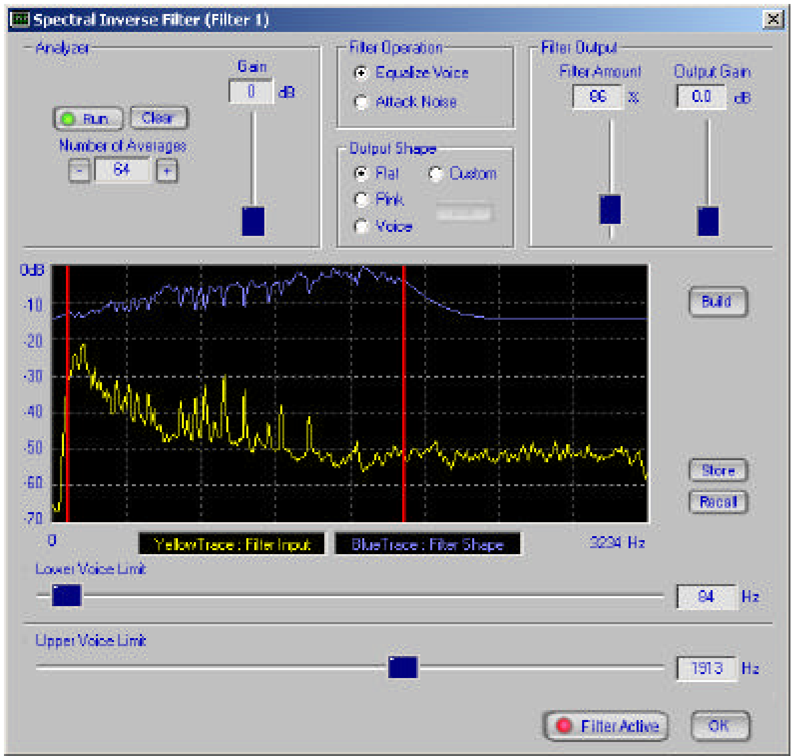

Figure 4-44 SIF with **Filter Amount** set to 85%

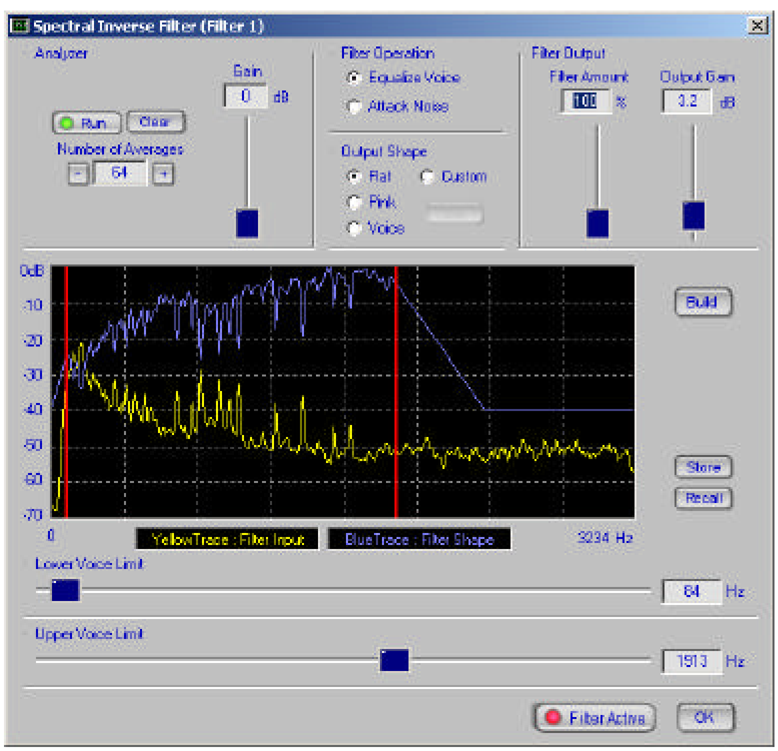

Figure 4-45 SIF with **Filter Amount** set to 100%, **Output Gain** set to 3.2 dB

The Output spectral Shape may be selected as Flat, illustrated in Figure 4-43, Figure 4-44, and Figure 4-45 above. It may also be set to Voice or Pink. See Figure 4-47.

The **Attack Noise** mode does not attenuate Out-of-Limits signal, but equalizes and attenuates in-Limits signal frequencies. Figure 4-46 illustrates.

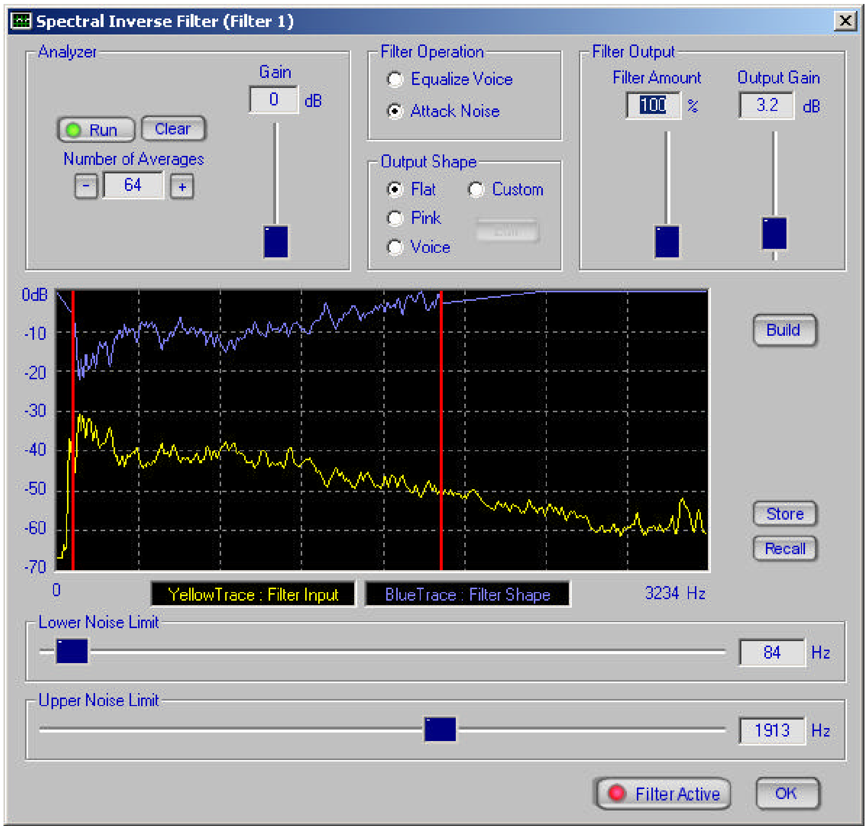

Figure 4-46 **Attack Noise** operation, **Filter Amount** set to 100%

Applications Suggestions

The following suggestions may be beneficial in setting up and operating the Spectral Inverse filter.

Analyzer: The FFT spectrum analyzer automatically produces the power spectrum of the signal entering the SIF. For SIF to be effective, a smoothed spectrum is necessary; SIF adjusts for long-term stable noises including resonances and steady noises. Short-term effects of voice and non-stationary noises will decrease the filter's effectiveness; therefore, the **Number of Averages** should be set large. At least 32 averages are recommended, but more will give a smoother spectrum from which to build a filter.

The Analyzer **Gain** should be increased to display weaker energy components. Do not overload (OVL) the analyzer, as the spectral information would become corrupted.

Try to capture a representative spectrum using the **Run** button. Once a stable, representative spectrum is obtained on the display, freeze the analyzer (GREEN **Run** LED unlit). This same spectrum may be used for several different variations of the filter (changing Limits or **Filter Amount**, as an example).

NOTE: Immediately following a freeze, the screen will continue to update briefly, as the PC is receiving the final prefrozen spectral data from the external unit.

Limits: In the **Equalize Voice** operation, the **Upper** and **Lower Voice Limits** should bracket the voice signal. Outside these limits, the audio is bandpass-filtered. Setting the **LowerVoice Limit** above 300 Hz and the Upper Limit below 3000 Hz may adversely affect intelligibility. Try several sets of Limits; build the SIFs; Store the filters, and Recall and compare in an A/B fashion.

In the **Attack Noise** operation, bracket the noise band surgically with these Limits. If there are several disjoint noise bands, series additional SIFs. (The PCAP II *Plus* will series up to four). The audio outside those limits is unfiltered.

Filter Amount and **Output Gain**: SIF is a spectral attenuator. The spectral peaks are pressed down toward the spectral valleys. The amount of reduction is limited by the **Fitler Amount** scroll bar. If, however, all peaks are reduced to the *lowest* valley, no more reduction takes place.

For example, if the distance from the highest peak to the lowest valley *between the Upper and Lower Limits* is 30dB, the maximum range actually used will be 30dB. Normally a **Range** of 60-90% is adequate, as attacking the stronger spectral peaks will provide the necessary enhancement. Go gently at first and compare several SIFs with different **Filter Amounts**. The **Store** and **Recall** memories are useful in saving candidate filter solutions.

Since the SIF is an attenuator, the audio level should be restored with the **Output Gain** scroll bar. Normally, setting this to 3-10 dB. If output distortion occurs, reduce the **Output Gain**. The **Output Gain** may be increased to make up for losses in previous Filter stages. Note that the Gain is usually 0 dB in the **Attack Noise** mode.

Output Shape: Four output spectral shapes are selectable by the user. The **Flat** shape requires the SIF to produce a uniform (to the degree possible) long-term output spectrum. This can be subsequently *reshaped* by the Hi-Res Graphic Filter or Output Equalizer to a more natural-sounding spectrum. Alternately, the **Voice** or **Pink** shapes may be selected which have built-in output shaping. *The **Pink** shape is often the most pleasing.* The **Custom** shape is used when the other three predefined shapes are not adequate.

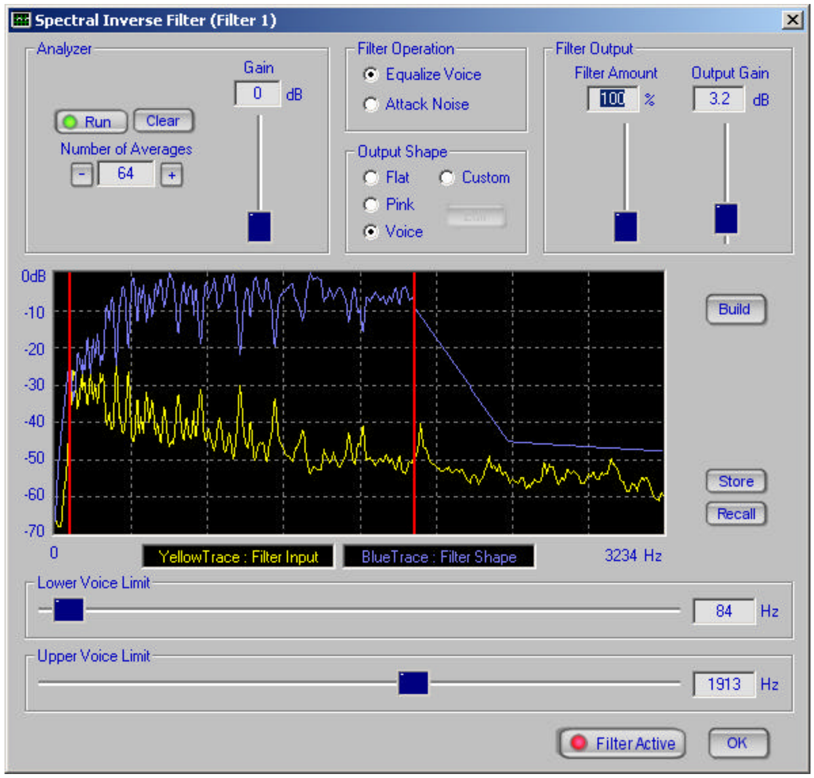

Figure 4-47 SIF with **Output Shape** set to **Voice**

Experiment: Select a section of audio and display its spectrum. *Freeze* the analyzer and vary different control settings, storing built filters. Compare the results and determine the best solution. Always compare these different filter solutions using the same input audio.

4.4.13: Hi-Res Graphic Filter

Application:

In some applications, it may be necessary to precisely reshape the spectrum of input audio prior to passing it through successive filter stages. For example, if the audio is from a microphone which has an unusual frequency response curve (for example, a microphone acoustically modified as a result of concealment), a compensation filter that reshapes the audio to a normal spectral shape might be desirable.

The Hi-Res Graphic Filter is essentially a 460-band graphic equalizer; however, instead of having 460 separate slider controls, it allows the user to precisely draw the desired filter shape on the computer screen, using the mouse, with as much or as little detail as desired. Once the filter shape has been drawn, a linear-phase digital filter is constructed in the PC and transferred to the external processor.

The ***Edit*** *feature allows the user to make readjustments to the filter shape, while the* ***Normalize*** *button allows the user to shift the entire filter curve up until the highest point is at 0dB.*

A ***Store*** *and* ***Recall*** *capability is also provided to allow the user to store commonly-used filter shapes to disk memories so that they can be recalled later.*

Figure 4-48 Hi-Res Graphic Filter Window

Description of controls/indicators is as follows:

Filter Display: Graphically displays the current shape of the filter. Also used in conjunction with the mouse to draw a new filter shape or to edit an existing one (see **New**, **Edit**, and **Normalize** button descriptions). A grid is provided to assist the user in visually judging frequency and attenuation at any point in the display.

Freq and Atten Readouts: Used to precisely readout the frequency in Hertz and attenuation in dB at any point in the filter curve. Hold the left Mouse button down while editing the curve or drawing a new curve. These readouts below the grid indicate precise frequencies and attenuations. Releasing the button draws the segment.

Store Button: This button allows the user to store a curve to a user-specified disk file that will not be lost when the computer is turned off. Clicking this button brings up the dialog shown in Figure 3-12. Save the configuration to any specified filename; clicking the **Cancel** button exits the Store window without storing the curve.

Recall Button: This button allows the user to recall a previously stored curve from any of the saved disk files previously generated using the **Store** button. Clicking this button brings up the window given in Figure 3-13.

NOTE: The curve that is in memory prior to recalling any specified file **EXCEPT** the one named “Previous Filter Settings” will be stored in “Previous Filter Settings” as part of the recall operation. Therefore, you can reload the previous curve at any time by recalling the “Previous Filter Settings” file.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the equalizer is in the process, and is not illuminated (GRAY) when the filter is out of the process.

NOTE: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the “X” button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

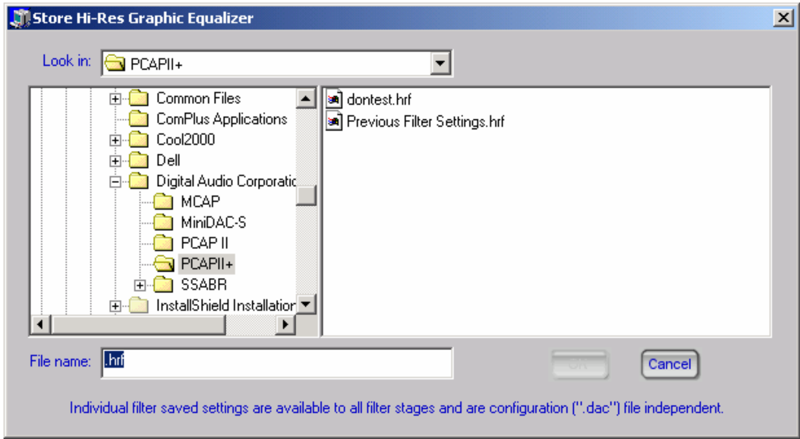

Figure 4-49 Hi-Res Filter Store Window

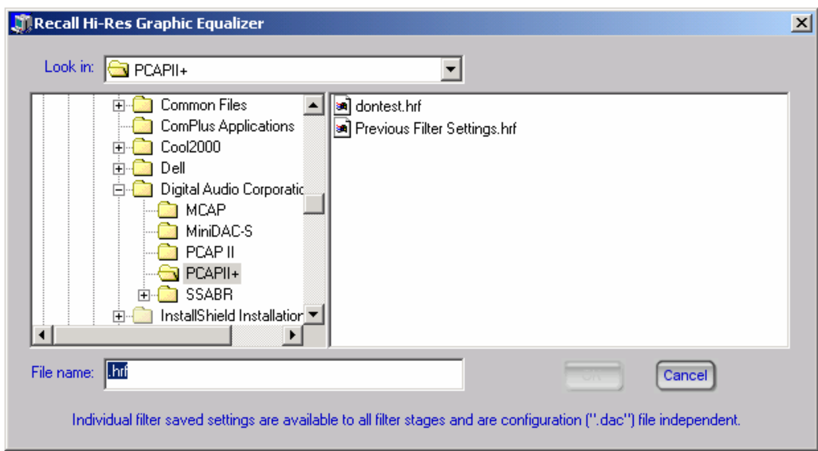

Figure 4-50 Hi-Res Graphic Filter Recall Window

4.4.13.1: Hi-Res Graphic Mini-Tutorial

The **New**, **Edit**, and **Normalize** buttons are used to graphically manipulate the shape of the filter curve. Their functions are complex, and thus are best illustrated in the following mini-tutorial:

1. Master Control Panel, set **System Bandwidth** to **5.4 kHz**, **Input Mode** to **Mono**, and **Number of General Filter Stages** to **1**. Go to the **Filter 1** block, then select **Hi-Res Graphic** from the drop-down menu.

2. Click on the **Config** button to bring up the Hi-Res Graphic Filter control window. When used for the first time, the control window will be the that of the previous Figure 4-48.

3. Click on the **New** button to draw a new filter. The screen will now appear as follows:

Figure 4-51 New Hi-Res Graphic Filter Display

Had you accidentally clicked the **New** button, you could click on **Abort** to restore the previous filter.

4. You should now notice that all the buttons on the control window have been replaced by a simple **Abort** botton. Clicking on **Abort** at any time prior to completing the curve restores the previous filter.

To draw the new filter curve, you will need to carefully click the mouse cursor on points within the filter display area which correspond to the desired attenuations at the desired frequencies.

While the mouse click button is held down, the **Freq** and **Atten** readouts will be updated as the mouse is moved; you can use this feature to place points in the filter curve at exact frequencies and attenuations. When the mouse click button is released, a line segment will be drawn from the last defined point on the curve to the current mouse cursor position.

For this example, placing points at precise frequencies and attenuations is not required; draw the curve approximately as shown in Figure 4-52 using mouse clicks.

Note that the very first click always sets the 0 Hz attenuation starting point.

Hint: To advance the frequency a single step move the cursor to the left of the last frequency position and click to mouse. The curve will advanced 12 Hz at the specified attenuation.

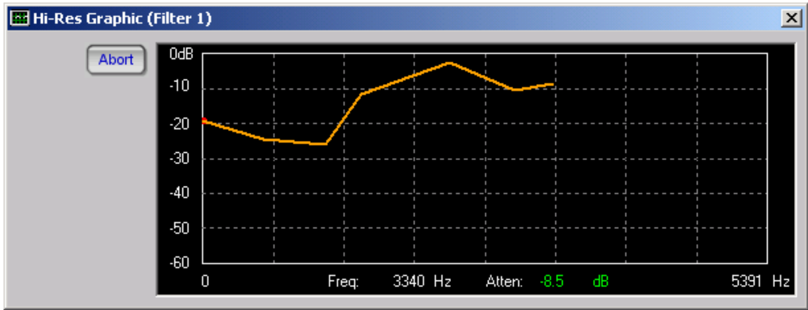

Figure 4-52 Hi-Res Graphic Draw in Progress

5. Complete drawing the filter curve as shown below (Figure 4-53) by drawing points all the way to the right edge of the filter display area.

Figure 4-53 Completed Hi-Res Graphic Draw

When you have drawn the last point (must be at or beyond the right edge of the filter display area), the mouse cursor will change to an "hourglass" shape for a few seconds while the filter is being calculated. When the calculations are complete, the mouse cursor and the buttons in the Hi-Res Graphic control window will return to normal appearance.

6. Suppose you decide that you would like to remove the "dip" which occurs in the filter curve at approximately 3500 Hz in Figure 4-53, above. Click on **Edit** to bring up the following display (Figure 4.51):

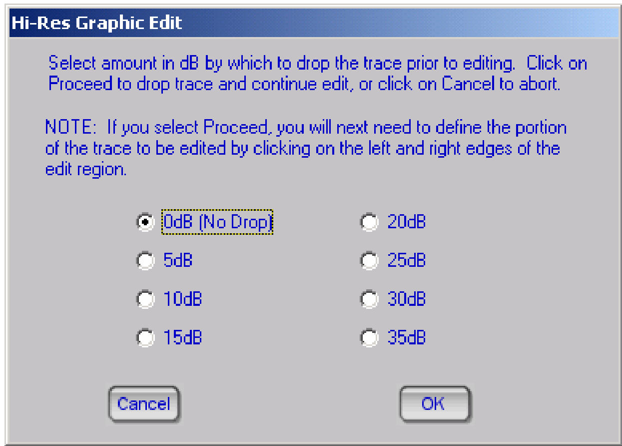

Figure 4-54 Hi-Res Graphic Edit Window

In this window, you can make the entire filter curve drop by a specified amount prior to editing the curve. This can be used to create headroom which can be used to increase the gain (decrease the attenuation) in one portion of the curve relative to the rest of the curve. For now, select a drop of **0dB (No Drop)** and click on **Proceed**.

7. You should now notice that all the buttons on the control window have been replaced with a single **Abort** button, which permits returning to the pre-Edit filter.

   To edit out the dip, you will first need to define the edit region by carefully specifying the left and right edges of the portion of the filter curve that you wish to modify. Click your mouse to the left and to the right of the dip to produce the following display (Figure 4-55):

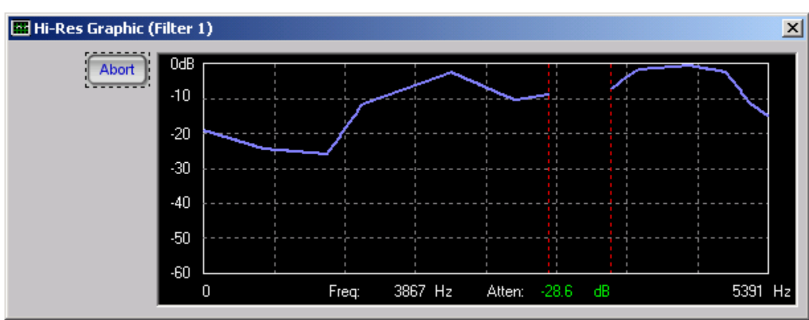

Figure 4-55 Hi-Res Graphic Define Edit Region

8. Now, draw in the new portion of the filter curve using mouse clicks as in Step 4, above, roughly as shown below (Figure 4-56).

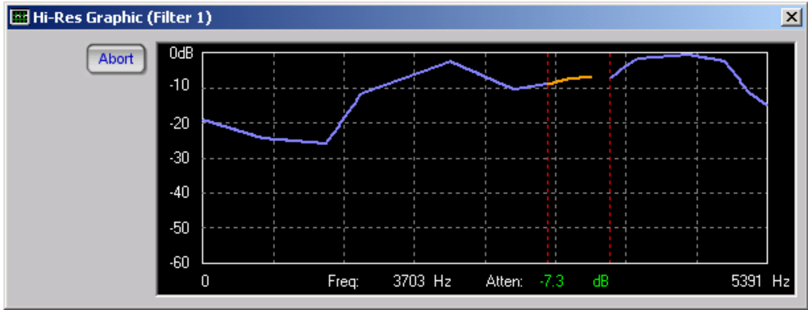

Figure 4-56 Hi-Res Edit In Progress

9. Complete drawing the new portion of the filter curve as shown below (Figure 4-57) by drawing points all the way to the right edge of the edit region:

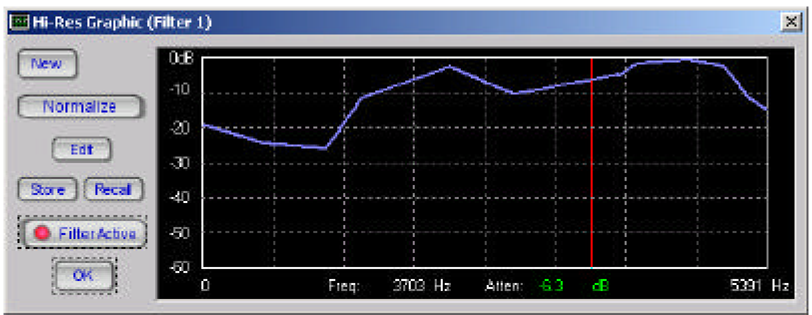

Figure 4-57 Completed Hi-Res Graphic Edit

When you have drawn the last point (must be at or beyond the right edge of the edit region), the mouse cursor will change to an "hourglass" shape for a few seconds while the filter is being recalculated. When the calculations are complete, the mouse cursor and the buttons in the Hi-Res Graphic control window will return to normal appearance.

10. Normalizing the filter places the highest point on the filter curve at 0 dB. Doing so minimizes loss in the filter and preserves system dynamic range. Now normalize the filter curve to 0dB by clicking the **Normalize** button. You should see the mouse cursor change to the "hourglass" shape for a few seconds; when the normalization calculations are complete, the filter shape should appear as follows (Figure 4-58):

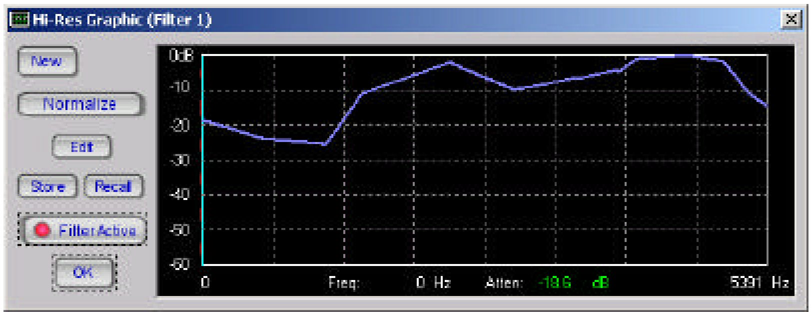

Figure 4-58 Normalized Hi-Res Graphic Filter

This completes the Hi-Res Graphic Filter mini-tutorial.

4.4.14: 20-Band Graphic Filter

The 20-band Graphic Filter is an easy-to-use linear-phase FIR digital filter that is used to reshape the spectrum of the final output signal. Reshaping is accomplished with twenty vertical scroll bars (also called "slider" controls) which adjust the attenuation of each frequency band. These controls are very similar to the slider controls found on analog graphic equalizers found on many consumer stereo systems, and thus should be very familiar to even the novice user.

However, unlike analog graphic equalizers, this digital equalizer has some very powerful additional capabilities. For example, the ***Normalize*** *button allows the user to instantly move all slider controls up until the top slider is at 0dB. The* ***Zero All*** *button instantly sets all the sliders to 0dB, while the* ***Maximize*** *button instantly sets all the sliders to -40dB. The* ***All Down 1dB*** *button instantly moves all sliders down in 1dB increments. The* ***Link Frequency Sliders*** *feature allows all the sliders to move together in sync whenever one is moved. None of these functions is available in an analog graphic equalizer! Notice also that the 20 sliders are spread across the selected Bandwidth and that the frequency spacing is optimized for voice processing.*

Additionally, since a computer with a disk drive operates the equalizer, a ***Store*** *and* ***Recall*** *capability is available. This allows the user to store commonly-used slider configurations in disk memories so that they can be instantly recalled later whenever they are needed, without having to manually adjust the slider controls.*

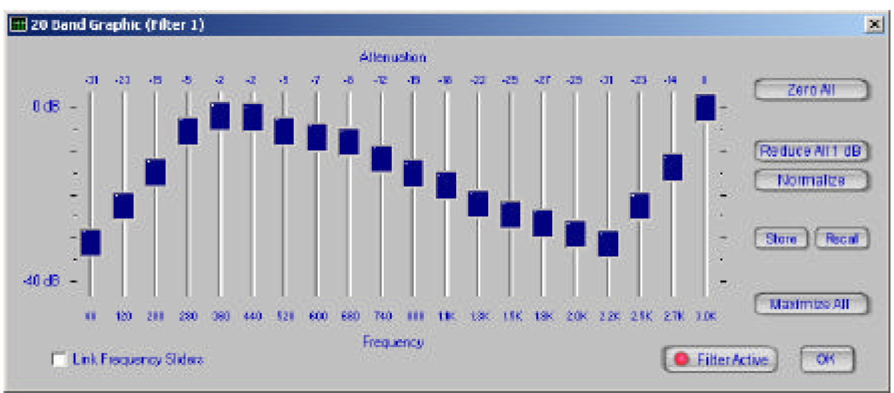

Figure 4-59 20-Band Graphic Filter Control Window

Description of controls/indicators is as follows:

Slider controls:	The twenty vertical scroll bar "slider" controls are used to set the frequency response of the equalizer. Each slider can set the gain of its frequency band to any value between 0dB and -40 dB in 1dB steps.

Link Frequency Sliders Checkbox: When checked, this feature links the 20 slider controls together such that when any one of them is moved, all others move together in sync. Uncheck the box to move the sliders without affecting the others.

Center Frequency: Note that the Center Frequency of each band is labelled underneath each slider, and that bands are more closely spaced at low frequencies

Gain Indication: Above each slider control, the gain for that frequency band is given. The gain can also be visualized graphically by the position of the slider control.

Normalize Button: This button instantly shifts all slider controls up together until the top slider is at 0dB. After normalization, the relative positioning of the sliders remains the same. This allows the digital equalizer to implement the desired equalization curve with minimum signal loss.

Zero All Button: This button instantly moves the slider controls for all bands to 0dB, defeating the entire equalizer. This is a useful feature when it is desired to reset all sliders from scratch.

Maximize All Button: This button instantly moves the slider controls for all bands to -40dB, maximizing the attenuation for all bands. This is a useful feature when it is desired to quickly adjust the sliders such that only a few bands are passed with all others rejected.

All Down 1dB Button: This button shifts all sliders down by 1dB from their current position; no slider, however, will be allowed to go below -40dB. This button allows the user to shift the entire equalizer curve down so that there will be room to move one or more sliders up relative to the others.

Store Button: This button allows the user to store a slider configuration to a user-specified disk file that will not be lost when the computer is turned off. Clicking this button brings up the dialog shown in Figure 3-12 of Chapter 3: .0. Save the configuration to any specified filename; clicking the **Cancel** button exits the Store window without storing the slider configuration in any of the memories.

Recall Button: This button allows the user to recall a previously stored slider configuration from any of the saved disk files previously generated using the **Store** button. Clicking this button brings up the window given in Figure 3-13 of Chapter 3: .0.

Note: The curve that is in memory prior to recalling any specified file **EXCEPT** the one named “Previous Filter Settings” will be stored in “Previous Filter Settings” as part of the recall operation. Therefore, you can reload the previous curve at any time by recalling the “Previous Filter Settings” file.

Filter Active Button:	Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filteris out of the process. **Note:** The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.
OK Button:	Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

4.4.15: Tri Parametric Filter

Application:

The Tri Parametric Filter consists of three adjustable IIR filter "substages", connected in series, which can be used for both ***peaking*** *and* ***nulling*** *portions of the input signal's frequency spectrum. These substages can be configured independently, and are typically used as precision notch filters, which perform* ***nulling*** *of the input signal at a specified Center Freq over a specified width. The Boost/Cut parameter can be set to a* <u>*negative*</u> *value (-1dB to –60dB) for this type of operation. However, a substage is also capable of* ***peaking****, which allows a* <u>*positive*</u> *gain (Boost/Cut of 1dB to +16dB) to be placed at the Center Frequency over the specified Width.*

The filter input signal is first attenuated as specified by the ***Input Atten*** *control, then the attenuated signal flows sequentially through Substage 1, Substage 2, then finally Substage 3. Controllable parameters for each seriesed substage include* ***Center Freq****uency****, Width,*** *and* ***Boost/Cut.***

Each substage can be independently bypassed using the individual ***Active*** *buttons in order to allow the user to more quickly and easily adjust each stage for optimum results. Simply click in the substage that is to be adjusted while the other stages are clicked out; repeat this process for each substage that is used. Also, an input* ***Input Atten****uation control allows the user to quickly reduce the overall gain of the entire parametric filter without altering the overall shaping function.*

An bargraph ***Output Level*** *indicator, as well as an independent Internal* ***Overload*** *indicator, assists the user in avoiding output distortion by preventing the dynamic range limits of the filter substages from being exceeded.*

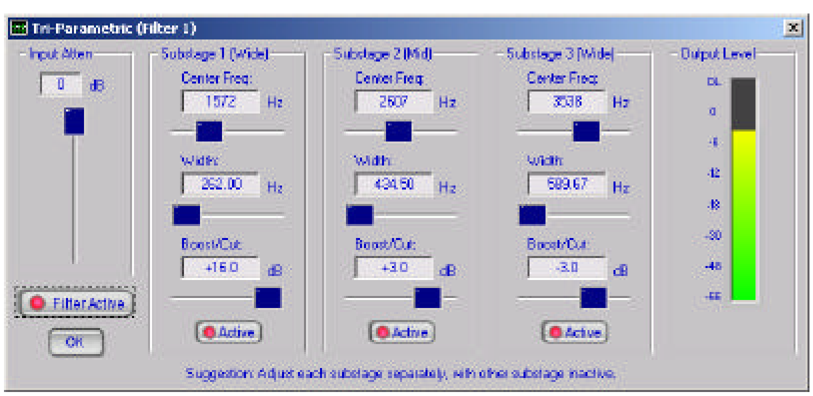

Figure 4-60 Tri Parametric Filter

Description of operation:

The **Center Freq** text box / slider controls allow the user to either type in the desired Center Frequency of each stage directly, or to make mouse adjustments. Center Frequency can be specified in 1 Hz steps.

The **Width** text box / slider controls work similarly to the Center Freq controls. The displayed Width, however, is proportional to the Center Frequency, and will thus be automatically changed whenever the Center Freq controls are adjusted.

The **Boost/Cut** controls work similarly to the Center Freq and Width controls, as the Boost/Cut can be typed directly or selected with the slider control. The Boost/Cut adjustment ranges from -60dB to +16dB in 1dB increments.

> **Warning:** Pay particular attention to the Output Level and Stage Overload indicators when positive Boost/Cut values are used, as the maximum digital signal range for the system can easily be exceeded.

The individual Substage **Active** buttons allow the user to quickly switch any substage in (LED illuminated RED) or out (LED not illuminated) of the process. These function in the same manner as the **Filter Active** button, except they a

The **Input Atten** text box and slider control is normally set to 0dB, but can be used to insert an attenuation of as much as 60dB in 1dB increments. This is a useful method for quickly reducing the overall gain of the four combined parametric stages without having to manually adjust all four Boost/Cut controls (for example, when a Stage Overload occurs), allowing the overall shaping characteristics of the parametric filter to be retained. Type **0** in the text box to quickly restore the input attenuation to 0dB.

The **Output Level** bargraph, which indicates the actual parametric filter output signal level, is provided for convenience. It functions identically to the signal bargraphs on the Master Control Panel.

The **Filter Active** button allows the entire parametric filter to be switched in or out of the process at any time. The button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process. **NOTE:** The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

The **OK** button, or the “X” button at the upper right corner of the window, is used to exit the dialog and return to the Master Control panel.

A graphical description of a parametric filter and its controls follows in Figure 4-61.

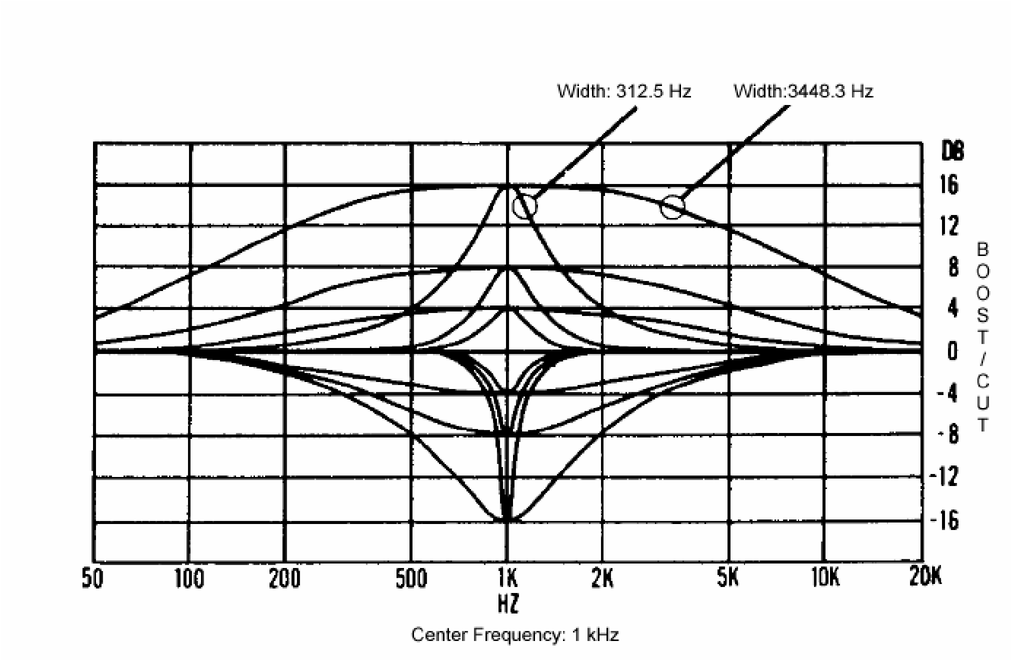

Figure 4-61 Graphical Representation of a Parametric Filter

4.4.16: Inverse Comb

Application:

The Inverse Comb filter is used to indentify harmonically related noises (noises which have exactly equally-spaced frequency components), such as power-line hum, constant-speed motor/generator noises, etc., from the input audio. The filter response consists of a series of equally-spaced slots. This filter is the exact opposite of the Comb Filter.

Adjust the Inverse Comb Frequency to the desired spacing between slots (also known as "fundamental frequency"). Set the Harmonic Limit to the frequency beyond which you do not want any more slots. Set the Attenuation to the amount in dB by which noise frequency components are to be reduced.

Normally, the Enhance Harmonics option will be set to All, causing frequencies outside all multiples of the Fundamental Frequency (within the Slot Limit) to be reduced However, certain types of noises have only the odd or even harmonic components present. In these situations, set the Enhance Harmonics option to either Odd or Even.

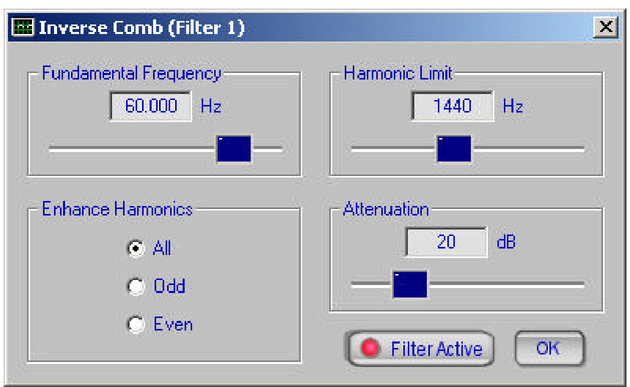

Figure 4-62 Inverse Comb Filter

Description of controls is as follows:

Fundamental Frequency: Specifies fundamental frequency in Hertz of inverse comb filter. Slots are generated at multiples, or harmonics, of this frequency.

Note: Inverse Comb Frequency changes whenever the System Bandwidth setting is altered; if you change the System Bandwidth setting, you will need to readjust the Inverse Comb Frequency for any Inverse Comb Filters selected.

Harmonic Limit:	Specifies frequency in Hertz above which no slots are generated and the signal is attenuated. Minimum Harmonic Limit is 100 Hz, while maximum Harmonic Limit depends upon the System Bandwidth setting. Harmonic Limit is adjustable in 50 Hz steps.
Attenuation:	Depth of the attenuation region around the slots that is generated. Attenuation is adjustable from 10 dB to 60 dB in 1 dB steps.
Enhanced Harmonics:	Specifies whether slots will be generated at All, Odd, or Even multiples, or harmonics, of the Inverse Comb Frequency. If, for example, the Inverse Comb Frequency is set to 60.000 Hz, then selecting All will generate slots at 60 Hz, 120 Hz, 180 Hz, 240 Hz, 300 Hz, etc. Selecting Odd will generate slots at 60 Hz, 180 Hz, 300 Hz, etc. Selecting Even will generate slots at 120 Hz, 240 Hz, 360 Hz etc.
Filter Active Button:	Used to switch the Inverse Comb filter in and out of the process without affecting the other filters in the process. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process. **Note**: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

4.4.17: Limiter/Compressor/Expander

Application:

The LCE (Limiter/Compressor/Expander) is a three-section dynamic signal level processor, recommended for advanced users only. Dynamic signal level processing enables the user to manipulate the overall dynamic range of a signal, generally to correct for near-party/far-party and/or "quiet talker" scenarios.

The three types of level processing available (three sections) are limiting, compression, and expansion. Unlike an analog implementation of this process, this digital implementation is substantially easier to set up and operate and is far more accurate. All matching of amplifier gains is carried out automatically by the processor. Logarithmic level conversions are carried out precisely in real time.

These dynamic processes modify the ***amplitude*** *of the signal using a variable-gain digital amplifier. The* ***amplitude*** *is a rectified and smoothed version of the signal wave form, as measured by a real-time digital envelope detector.*

The operation of the envelope detector is governed by its ***Attack Time*** *and* ***Release Time****, which are adjustable. Normally a fast attack time and slow release time are used with speech.*

Setting a fast attack time (less than 10 milliseconds) causes the processor to rapidly respond to sharp sounds. The level detector will be more peak sensitive to fast attack time and more average-value sensitive to longer attack times. An ***Attack Time*** *of 2-5 milliseconds is recommended for Speech applications.*

Short release times, less than 100 milliseconds, may make the level detector too responsive to intra-syllabic pause creating an annoying "pumping" artifact. Conversely, long release times, greater than 500 milliseconds for example, may fail to respond to breath group pauses and exchanges between speakers. Therefore a ***Release Time*** *of 200-400 milliseconds is generally recommended for speech applications.*

The gain algorithm dynamically adjusts the signal amplification based on input signal level and its ***Gain Region*** *(limiting, compression, or expansion). If the input signal level exceeds the* ***Limit Threshold****, it is in the* ***Limit*** *region. If the input signal level exceeds the* ***Compression Threshold****, but not the* ***Limit Threshold****, it is in the* ***Compression*** *region. And, lastly, if the input signal level is below the* ***Compression Threshold****, it is in the* ***Expansion*** *region. The three* ***Gain Regions*** *are described below:*

> ***Limit Region****: In the limit region the output signal level is "damped" or kept from exceeding the specified* ***Limit Threshold*** *by applying attenuation.*

Compression Region: In this region the output signal level changes at a fraction of the rate of the increase of the input signal level. The dynamic range of the output signal is thus is reduced with respect to the input signal. As an example, a 2:1 compressor would produce an output level change of only 10 dB when the input signal changes by 20 dB. Compression allows signals of wide dynamic ranges to be squeezed into more limited dynamic ranges of recording media and transmission channels. Compression also eases listening, especially for noisy audio. Compressors are generally preferred over AGC's since input signal level differences are more modestly preserved.

Expansion Region: In this region the output signal has a wider dynamic range than the input signal. Expansion is the opposite of compression. As an example, with an expansion ratio of 1:3, the output signal will change by 30 dB with only a 10dB change in the input signal. Expansion may be used to restore a signal's dynamic range following compression; for example, if a 2:1 compression has taken place, a 1:2 expansion would restore the signal to its original dynamic range. Expansion is also used to attenuate objectionable low-level background sounds that are below voices.

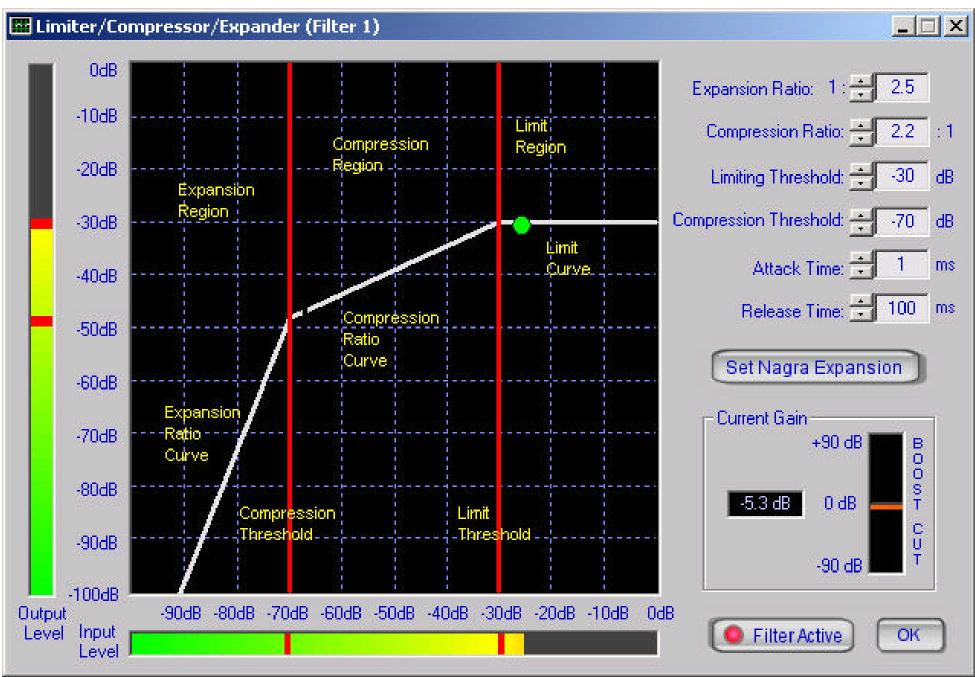

Figure 4-63 Limiter/Compressor/Expander Control Window

Description of operation (Six different parameters are adjustable on the LCE Control Window:)

The **Expansion Ratio** can be adjusted through a range of 1:1 to 1:100 by using the scroll buttons beside its text entry box, by entering the value in its text entry box, or by clicking on the **Expansion Ratio Curve** using the mouse pointer and adjusting it to the desired value. *NOTE: Only the portion of the ratio to the right of the colon is specified.*

The **Compression Threshold** can be adjusted through a range of –90dB to 0dB by using the scroll buttons beside its text entry box, by entering the value in its text entry box, or by clicking on the **Compression Threshold** line using the mouse pointer and adjusting it to the desired value.

The **Compression Ratio** can be adjusted through a range of 1:1 to 100:1 by using the scroll buttons beside its text entry box, by entering the value in its text entry box, or by clicking on the **Compression Ratio Curve** using the mouse pointer and adjusting it to the desired value. *NOTE: Only the portion of the ratio to the left of the colon is specified.*

The **Limit Threshold** can be adjusted through a range of –90dB to 0dB by using the scroll buttons beside its text entry box, by entering the value in its text entry box, or by clicking on the **Limit Threshold** line using the mouse pointer and adjusting it to the desired value.

The **Attack Time** can be adjusted through a range of 1 to 250 milliseconds by using the scroll buttons beside its text entry box or by entering the value in its text entry box.

The **Release Time** can be adjusted through a range of 50 to 2000 milliseconds by using the scroll buttons beside its text entry box or by entering the value in its text entry box.

Other indicators are the Input and Output Level Displays. The Input Level Display shows the current envelope level for the stage. The Output Level Display shows the output level of the stage calculated by multiplying the input envelope by the gain. The current position on the LCE graph is shown via a “dancing” orange dot. The Current Gain display shows the current instantaneous gain value of between –90dB and 90dB, as calculated by the external processor.

4.5: BROADBAND FILTER CONTROL WINDOWS

4.5.1: Noise Reducer

Application:

The Noise Reducer is a frequency-domain spectral-subtraction filter that implements automatic noise reduction over 512 separate frequency bands. It operates by continually measuring the spectrum of the input signal and attempting to identify which portions of the signal are voice and which portions are non-voice (or noise). All portions determined to be noise are used to continually update a ***noise estimate*** *calculation; this is used to calculate the equalization curve that needs to be applied to the input signal to reduce each band's energy by the amount of noise energy calculated to be in that band.*

The net result is an output signal that has all non-voice signals reduced in level as much as possible, thereby "polishing" the enhanced voice signal as much as possible prior to final equalization and AGC.

Operation of the Noise Reducer is governed by two primary controls: the Master Attenuation Control and the Noise Smoothing button. Adjusting the Master Attenuation Control allows the user to precisely control the amount of noise reduction being applied; the greater the value, the more aggressive the operation of the Noise Reducer.

Because large amounts of noise reduction invariably create audible "birdy noise" artifacts in the output audio due to the nature of adaptive frequency-domain processing, the user should always try to minimize the amount of noise reduction being applied to achieve the best balance between maximal noise reduction and minimal audible artifacts.

In cases where a large amount of noise reduction needs to be applied, but audible artifacts are unacceptable, it is recommended that the Noise Smoothing feature be activated in order to try to minimize the audible artifact as much as possible.

Finally, for convenience an Output Gain control and Output level bargraph are provided to enable the user to adjust the processed output signal to maximum level for better listening and recording.

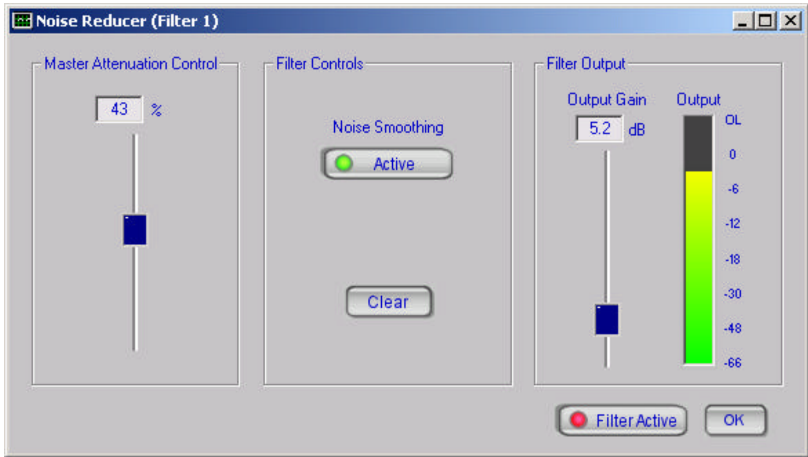

Figure 4-64: Noise Reducer Control Window

Description of controls/indicators is as follows:

Master Attenuation Control: Used to specify the amount of noise reduction that the spectral subtraction attempts to apply to the input signal. Adjustment range is 0 (no attenuation) to 100% (maximal attenuation) in 1% increments.

Noise Smoothing Button: Used to activate or deactivate a special feature that minimizes the amount of audible artifacts that occur when aggressive settings are used for the Master Attenuation Control. Feature is either on (LED illuminated GREEN) or off (LED not illuminated).

Output Gain: Allows user to apply between 0 and 30dB of makeup gain to the processed output signal to maximize the signal level prior to final equalization, AGC, and listening/recording. The associated Output bargraph shows the actual output signal level after the gain has been applied.

Clear Button: Used to clear the spectral subtraction solution currently in memory and restart the algorithm from scratch.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

Note: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

4.5.2: NoiseEQ™

Application:

Like the Noise Reducer tool, the NoiseEQ™ is a frequency-domain spectral-subtraction filter that implements automatic noise reduction over 512 separate frequency bands. It operates by continually measuring the spectrum of the input signal and attempting to identify which portions of the signal are voice and which portions are non-voice (or noise). All portions determined to be noise are used to continually update a ***noise estimate*** *calculation; this is used to calculate the equalization curve that needs to be applied to the input signal to reduce each band's energy by the amount of noise energy calculated to be in that band.*

The net result is an output signal that has all non-voice signals reduced in level as much as possible, thereby "polishing" the enhanced voice signal as much as possible prior to final equalization and AGC.

Operation of the NoiseEQ is governed by two primary controls: the Frequency Specific Noise Reduction (FSNR) control group and the Noise Smoothing button. Adjusting the control sliders within the FSNR section allows the user to precisely control the amount of noise reduction being applied within each of 20 distinct groups of frequency bands, offering much more precise control of the spectral subtraction than is available in the Noise Reducer tool, though it does take more time to setup.

The idea is to tailor the FSNR controls to minimize the amount of noise reduction applied within the speech frequency groups while maximizing it in other frequency groups. For each slider control, the greater the value, the more aggressive the operation of the NoiseEQ will be within that group of frequencies. Because large amounts of noise reduction invariably create audible "birdy noise" artifacts in the output audio due to the nature of adaptive frequency-domain processing, the user should always try to minimize the amount of noise reduction being applied in each band to achieve the best balance between maximal noise reduction and minimal audible artifacts.

In cases where large amounts of noise reduction need to be applied, but audible artifacts are unacceptable, it is recommended that the Noise Smoothing feature be activated in order to try to minimize the audible artifact as much as possible.

Finally, for convenience an Output Gain control and Output level bargraph are provided to enable the user to adjust the processed output signal to maximum level for better listening and recording.

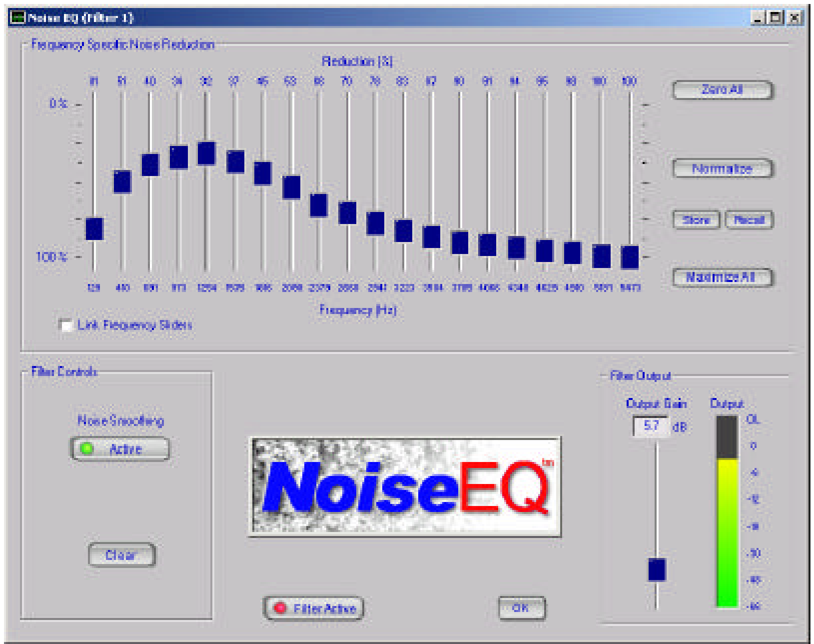

Figure 4-65: NoiseEQ™ Control Window

Description of controls/indicators is as follows:

Frequency Specific Noise Reduction: Used to specify the amount of noise reduction that the spectral subtraction attempts to apply to the input signal within each of 20 separate groups of frequency bands. Within each band, adjustment range is 0 (no attenuation) to 100% (maximal attenuation) in 1% increments. In much the same manner as the 20-Band Graphic Equalizer, special extra controls allow the user to Link Frequency Sliders, Zero All, Maximize All, Normalize, and Store/Recall complete curves to/from disk files.

Noise Smoothing Button: Used to activate or deactivate a special feature that minimizes the amount of audible artifacts that occur when aggressive settings are used for the Master Attenuation Control. Feature is either on (LED illuminated GREEN) or off (LED not illuminated).

Output Gain: Allows user to apply between 0 and 30dB of makeup gain to the processed output signal to maximize the signal level prior to final equalization, AGC, and listening/recording. The associated Output bargraph shows the actual output signal level after the gain has been applied.

Clear Button: Used to clear the spectral subtractionsolution currently in memory and restart the algorithm from scratch.

Filter Active Button: Used to switch the filter in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the filter is in the process, and is not illuminated (GRAY) when the filter is out of the process.

Note: The Filter Active button is equivalent to, and linked with, the Active button in the associated Filter block on the Master Control Panel.

OK Button: Use either this button or the "X" button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

4.5.3: Adaptive Spectral Inverse Filter (ASIF)

Application:

The Adaptive Spectral Inverse Filter (ASIF) is an equalization filter that automatically readjusts the spectrum to match an expected spectral shape. It is especially useful when the target voice has been exposed to spectral coloration (i.e. muffling, hollowness, or tinniness), but it can also be used to remove bandlimited noises. This filter is much like the Spectral Inverse Filter (SIF, See Section 4.4.12:), except it continually updates the spectral solution, whereas the SIF only updates the solution when it is "built".

The ASIF maintains an average of the signal's spectrum and uses this information to implement a high-resolution digital filter for correcting long-term spectral irregularities. The goal of the filter is to reshape the overall spectral envelope of the audio, not to respond to transient noises and characteristics.

Several user controls are available for refinement of ASIF operation. The user can specify the expected spectrum so that the output audio is reshaped to a flat, pink, voice-like, or custom curve. An adapt rate setting controls the update rate for the spectral average, which in turn determines how quickly the filter responds to changes in the input audio. Upper and lower limit controls allow the user to specify the range over which equalization is applied, and a mode setting controls whether frequencies outside the equalization range are attenuated or left unaffected. The amount of spectral correction is adjustable using the Filter Amount control. The user can enable the auto-gain functionality to ensure that the output audio level is maintained at approximately the same as the input audio level. If the user disables the auto-gain, an output gain slider is available to manually boost the level of the output signal.

As an aid to visualizing the filter operation, the user can choose to view the input and output audio traces or the filter coefficient trace.

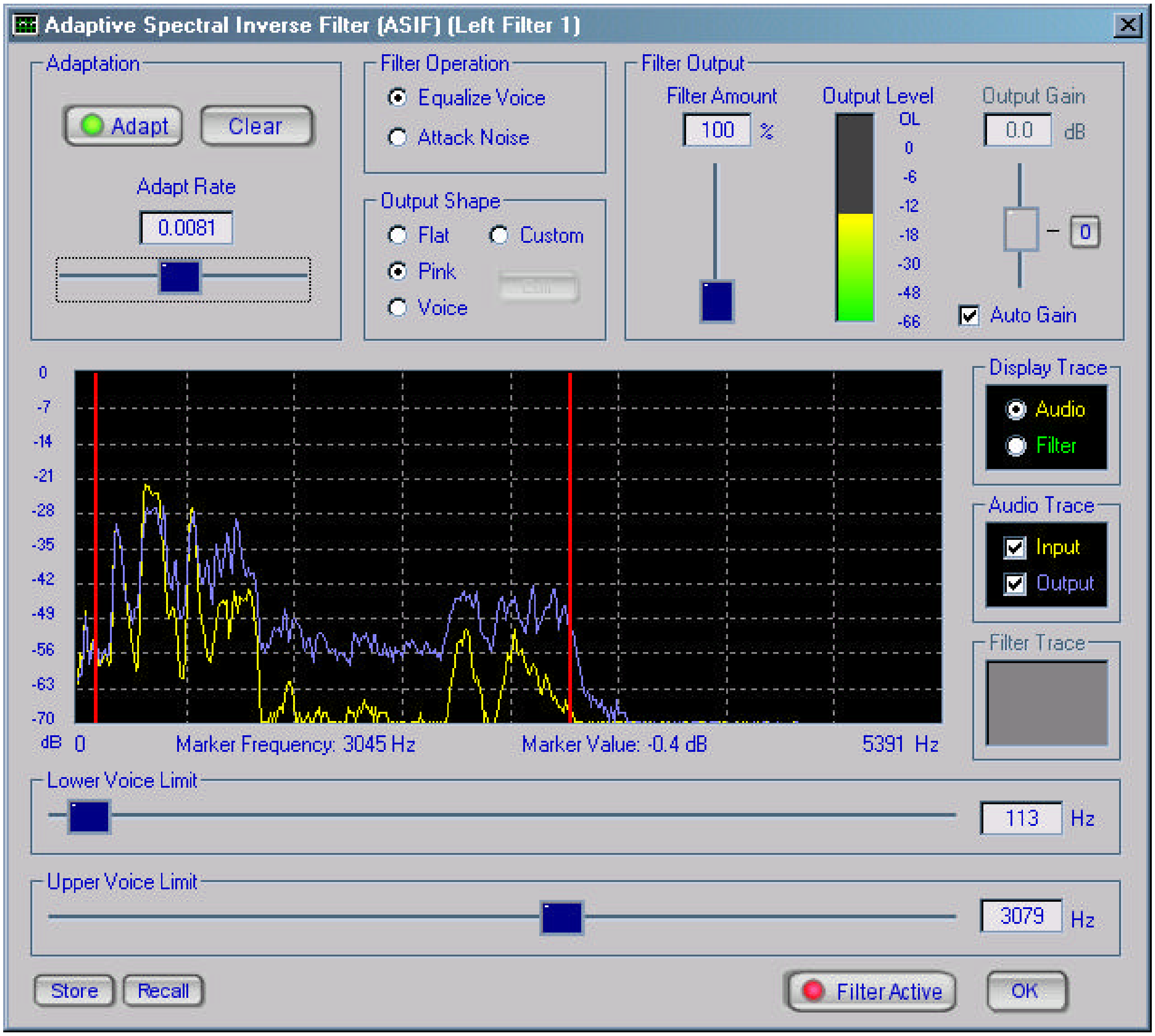

Figure 4-66: ASIF control window

Description of controls/indicators is as follows:

Display Trace and Display Controls: The display trace can be used to view either the filter input and output audio or the ASIF filter response. The radio buttons in the **Display Trace** block allow the user to select which type of trace is shown.

When viewing audio on the display, the user can choose to view the **Input Audio** only, the **Output Audio** only, or both input and output audio. This selection is made using the checkboxes in the **Audio Trace** block. The input audio is always shown in yellow and the output audio is always shown in blue. The audio trace shows 460 spectral lines and a dynamic range from 0 to -70 dB.

When viewing the filter response on the display, the user can choose the dynamic range of the trace. This selection is made using the drop-down menu in the **Filter Trace** block. Because the filter can both attenuate and boost frequencies, the trace grid does not have hard range limits. Instead, the trace limits are automatically adjusted to "good" settings based on the user-specified range and the filter response values. The trace will always obey the user-specified range, but the minimum and maximum values on the vertical axis will adjust as necessary to accommodate the filter response values. The filter response trace reflects the actual coefficients that are applied to the input audio to get the output audio. This means that the trace will reflect the voice limits, output shape, filter amount, operation mode, and the applied auto gain or manual gain.

A marker is available to help identify the value of a specific point on the trace. The marker, which looks like a set of crosshairs, can be placed anywhere on the trace and its frequency (horizontal position) and amplitude (vertical position) can be read just below the trace display. The marker is available on both the audio and filter response traces.

Adaptation: The controls in this block are used to specify the adaptation rate of the averager on which the ASIF is based

Adapt Button: When the green light is lit (and the button depressed), the ASIF is adapting in response to incoming audio. When the light is unlit (and the button unpressed), the ASIF response is frozen.

Clear Button: This button allows the user to re-initialize the ASIF response and restart adaptation.

Note: After a **Clear** operation or after re-enabling adaptation, there will be an adaptation period while the filter adapts to the current input signal. The length of this adaptation period depends on the **Adapt Rate** control setting.

Adapt Rate Control: This control allows the user to select the rate of adaptation for the spectral average on which the ASIF response is based. The spectral averager uses an exponential average of the form $H_{i+1} = (a)(X_{i+1}) + (1-a)(H_i)$. The value shown in the display box corresponds to the averaging constant a in the exponential average. The lower the adapt rate value, the slower the filter will respond to changes in the input audio.

Note: "Fast response" sounds like a good thing, so it can be tempting to set the adapt rate to a high value. However, the goal of the ASIF is not to remove transient noises, but rather to reshape the long-term spectral envelope of the signal. If the adapt rate is too fast, the filter will respond too quickly to transient audio characteristics, which will produce artifacts in the output audio and will prevent the filter from settling on a good average solution. *For this reason, most applications will work best with adapt rates at the low end of the available range.* If you hear tonal artifacts that come and go in the output audio, or if the filter trace display coefficients seem to be changing rapidly, you probably need to reduce the adapt rate.

Filter Operation: In this block, the user can select the operational mode of the filter. If the filter is being used to correct spectral coloration, the **Equalize Voice** mode should be selected. If the filter is being used to remove bandlimited noise, the **Attack Noise** mode should be selected.

Note: The **Filter Operation** mode selection only affects the behavior of the filter *outside* the range selected by the upper and lower limits. In **Equalize Voice** mode, the frequency ranges outside the limits are attenuated. In **Attack Noise** mode, the frequency ranges outside the limits are left unaffected (subject to a transition region near the limits). If the auto gain is disabled and the manual gain is set to 0 dB, frequencies outside the limits and transition regions will be unaffected. However, if gain is applied, the gain will be reflected over the entire frequency range. See the section on **Upper** and **Lower Voice Limits** for more information on selecting the range.

Note: Changing the filter operation mode does not require an adaptation period to arrive at a "good" solution. Because a full average spectrum is maintained regardless of the mode setting, the new mode takes effect instantaneously in both the output audio and the display traces. However, since the auto gain adapts based on the actual applied filter with operational mode taken into account, there may be some adaptation time required to reach a stable auto gain value after the mode is changed.

Output Shape: In this block, the user can select the target spectral shape that the filter attempts to achieve. The ASIF has an inherent spectral flattening effect on the audio. The selected spectral shape is applied to further reshape the audio spectrum. The following output shapes are available:

- **Flat** – no additional shaping after ASIF flattening
- **Pink** -- 3 dB per octave rolloff above 100 Hz is applied in addition to ASIF flattening
- **Voice** – 6 dB/octave rolloff above and below 500 Hz in addition to ASIF flattening

- **Custom –** user draws custom curve to be applied in addition to ASIF flattening

Note: Changing the output shape does not require an adaptation period to arrive at a "good" solution. Because a full average spectrum is maintained regardless of the output shape setting, the new output shape takes effect instantaneously in both the output audio and the display traces. However, since the auto gain adapts based on the actual applied filter with the shaping curve taken into account, there may be some adaptation time required to reach a stable auto gain value after the shaping curve is changed.

Custom Curve: To draw a custom curve, select **Custom** and then click the **Edit** button beneath the Custom selection button; the ASIF Custom Curve window will open. Click **New** to begin a new curve. To draw the curve, use your mouse to make single clicks within the axes; the curve is built piecewise by "connecting the dots" at the click locations. For fine control of the click locations, use the cursor coordinate information located at the bottom of the drawing axes. The ASIF custom curve drawing window is identical to the **Hi-Res Graphic Filter** drawing window. For more information on drawing a custom curve, see Section 4.4.13: .

You can save custom curves for later use. Click the Store button to save your custom curve to a file. Click the Recall button to load a custom curve that was previously saved to a file.

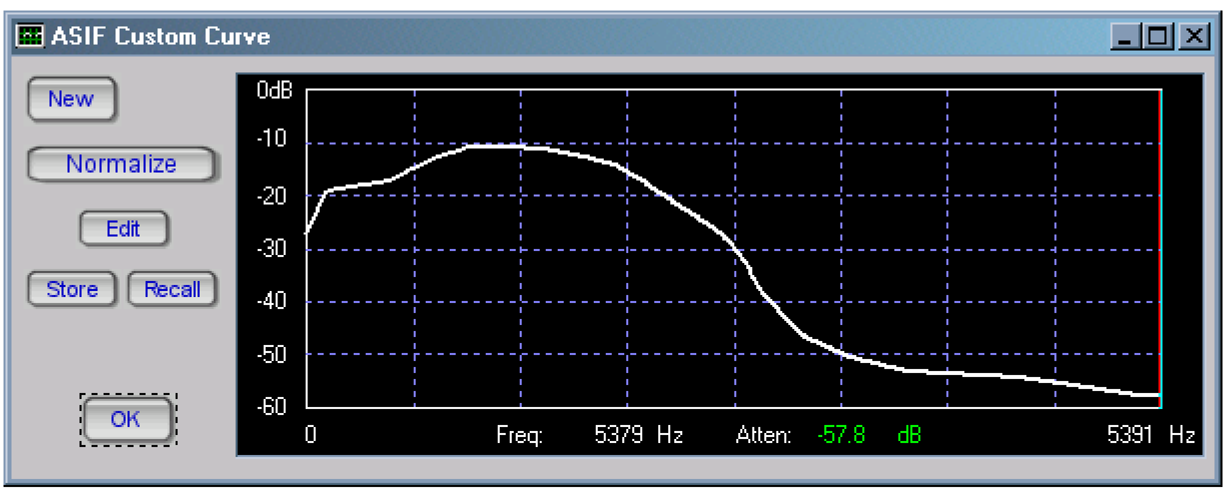

Figure 4-67: ASIF custom curve drawing window

Filter Output: The controls in this block allow the user to make adjustments to the filter output. An output level bargraph is shown as an aid to determining the output level.

Filter Amount — This setting controls the degree to which the ASIF can affect the signal, with 0% corresponding to no filtering and 100% corresponding to full filtering. In general, it is best to use the minimum Filter Amount setting that produces the desired result. When **Equalize Voice** mode is used, a lower **Filter Amount** can reduce artifacts that result from a fast adapt rate, so the **Filter Amount** can be used to help strike a balance between responsiveness and stability. When **Attack Noise** mode is used to reduce bandlimited noise, a lower **Filter Amount** setting will often be a better choice to prevent the elevation of background noises.

Note: Changing the **Filter Amount** setting does not require an adaptation period to arrive at a "good" solution. Because a full average spectrum is maintained regardless of the setting, the new filter amount setting takes effect instantaneously in both the output audio and the display traces. However, since the auto gain adapts based on the actual applied filter with filter amount taken into account, there may be some adaptation time required to reach a stable auto gain value after the filter amount is adjusted.

Example 2 below illustrates the effect of the Filter Amount control.

Output Gain and Auto Gain: — These controls provide two options for adjusting the level of the ASIF output. When **Auto Gain** is enabled, the ASIF automatically monitors the input and output levels and applies a gain value that matches the output level to the input level. When **Auto Gain** is disabled, the user can use the **Output Gain** setting to specify the amount of boost applied to the ASIF output.

The Auto Gain is an adaptive value whose rate of change depends on the same Adapt Rate slider setting that controls filter coefficient averaging. This means that when the filter response changes rapidly and dramatically, the auto gain will take some time to "catch up" to these changes. In particular, the output audio may clip when user settings are changed in a ways that have a boosting effect, such as switching from a pink to a flat shaping curve, adjusting the filter amount, or increasing the size of the ASIF region in Equalize Voice mode so that some frequencies that had been heavily attenuated are now present. While these settings changes will take effect immediately, the Auto Gain may take some time to adapt to the change. For this reason, when the user expects to be making many changes in the settings, it is often better to disable Auto Gain and instead choose a manual gain setting that avoids clipping.

Lower and Upper — The **Lower** and **Upper Voice Limits** allow the user to specify the

Voice Limits: frequency range, or "ASIF region," over which the ASIF is applied. Two red markers, controlled by the sliders below the display trace, indicate where the lower and upper voice limits are located. The markers may also be adjusted by clicking and dragging within the display trace. Viewing audio on the display trace while manipulating the markers is an easy way to identify where your ASIF region limits should fall.

In **Equalize Voice** mode, the ASIF region is typically chosen to be the range over which speech frequencies are found. Setting a **Lower Limit** above 300 Hz or an **Upper Limit** below 3000 Hz is not recommended in equalize voice mode, as intelligibility may suffer. When in **Equalize Voice** mode, all frequencies outside the ASIF region are assumed to be non-speech and are therefore attenuated.

In **Attack Noise** mode, the ASIF region is typically chosen to "bracket" the bandlimited noise as closely as possible. Frequencies outside the ASIF region will be "passed through," i.e. there will be little or no effect outside the ASIF region except for a narrow transition band between the ASIF region and the passbands.

Note: Changing the Voice Limits does not require an adaptation period to arrive at a "good" solution. Because a full average spectrum is maintained regardless of the Voice Limit settings, the new Voice Limits will take effect instantaneously in both the output audio and the display traces. However, since the auto gain adapts based on the actual applied filter with voice limits taken into account, there may be some adaptation time required to reach a stable auto gain value after the limits are changed.

Store/Recall Buttons: The **Store** and **Recall** buttons allow the user to save the state of the ASIF to be recalled for later use. After clicking the Store button, the user selects an ".asf" filename under which the ASIF state will be stored. Upon clicking **OK**, the system takes a snapshot of the filter state and saves that information into the specified file.

To restore a saved ASIF file, the user clicks the **Recall** button, selects the desired ".asf" file, and then clicks **OK** for the settings to be loaded into the ASIF module.

The **Store** and **Recall** functionality saves the adapted state of the filter in addition to all the user settings. This means that the stored file contains a filter shape that is adapted to whatever audio was running through the system at the time of the store. When the filter is recalled, it opens with **Adapt** disabled so that the state of the filter is preserved until the user wishes it to begin adapting.

To begin adapting from the previously adapted filter state (i.e. if the current input audio is similar to the store-time input), simply click the **Adapt** button to enable filter adaptation. To use the saved settings but re-start filter adaptation from an initialized state (i.e. if the current input audio is different from the store-time input), click Clear to clear the filter, then click **Adapt** to enable filter adaptation.

Filter Active Button: This button allows the user to switch the filter in and out of the processing chain. When the **Filter Active** LED is red, the filter is active. When the LED is gray, the filter is not active.

Note: The filter continues to run and adapt even when it is inactive. The input rather than the filtered audio is routed to the output. This means that the filter continues to maintain a solution to the current audio input, assuming that adaptation is enabled. When the filter is made active again, the current solution will take effect instantaneously with no adaptation period.

OK Button: To exit the ASIF Configuration window, click **OK**. The filter will continue to run using the current settings.

Examples:

1. This example shows a typical use of the ASIF. The target audio is speech, but the speech is severely muffled. The muffling can be visualized on the audio trace, where the yellow input trace shows a sharp drop-off for frequencies above approximately 600 Hz. The ASIF boosts these muffled frequencies. The blue output trace shows that the muffled frequencies have been boosted so that the overall signal spectrum looks more like the selected pink output shape.

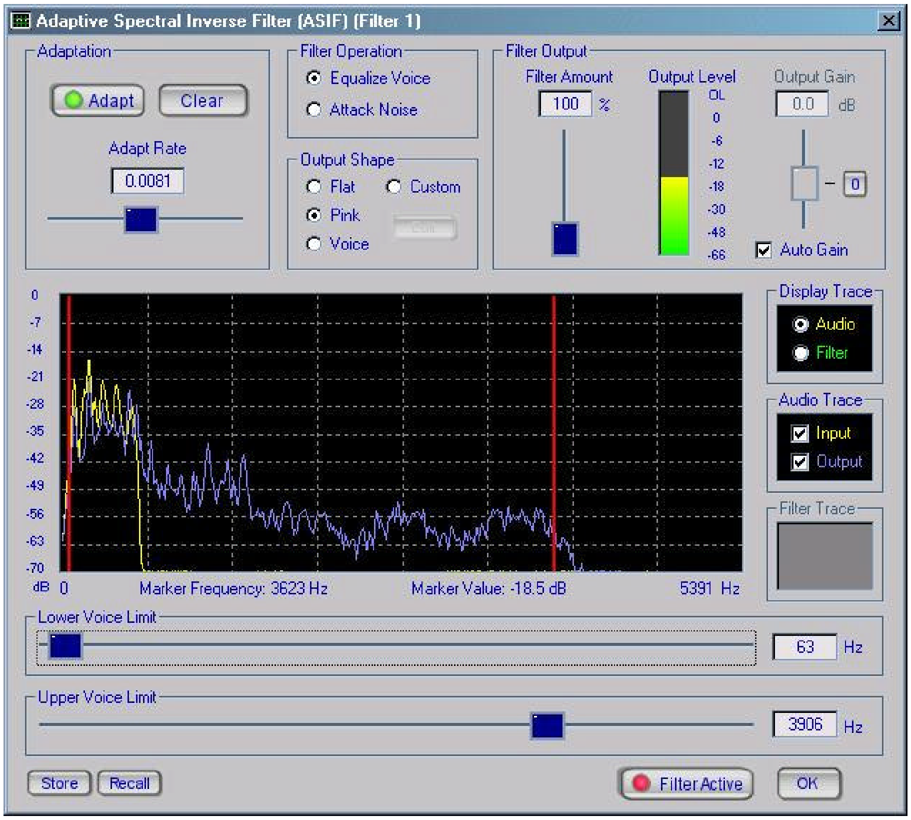

Figure 4-68: Audio trace view for ASIF Example 1

The filter trace view shows what the ASIF filter response looks like. Within the ASIF region, frequencies below 600 Hz are basically unaffected, while frequencies above 600 Hz are boosted considerably. Outside the ASIF region, the frequencies are attenuated since we are in Equalize Voice mode.

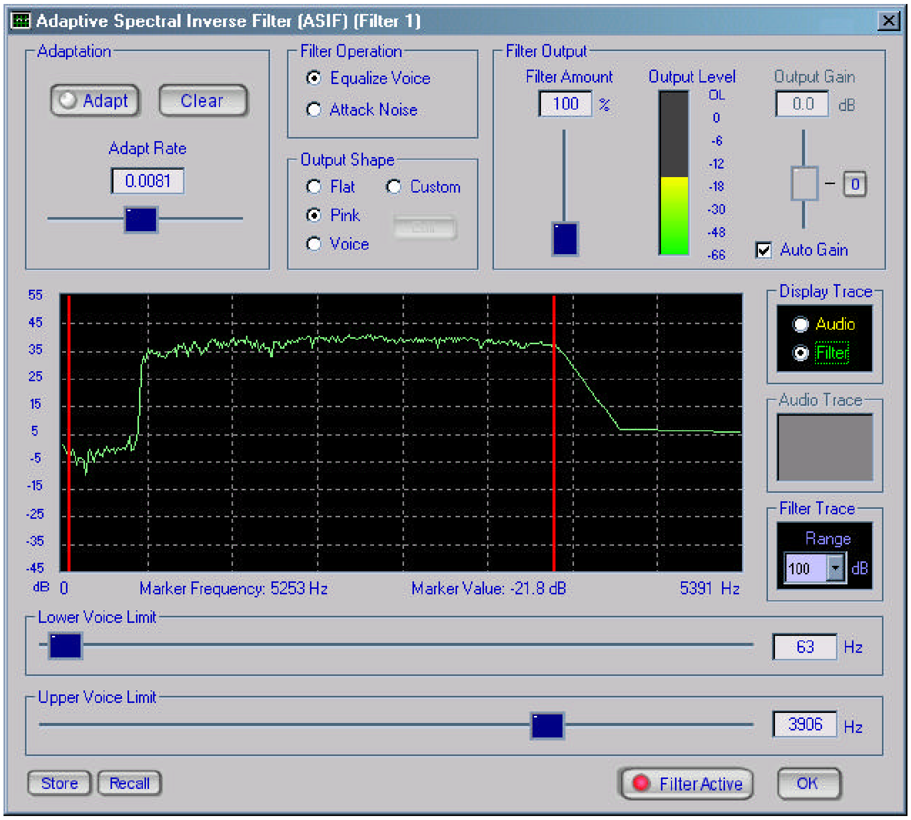

Figure 4-69: Filter trace view for ASIF Example 1

2. The Filter Amount controls the degree of filtering that is applied to the signal. In Example 1, the Filter Amount was set to 100%, or full filtering. The figures below show the same filter at other filter amount settings. Notice that, as the filter amount is reduced, the filter shape converges to the selection output shape (with gain applied).

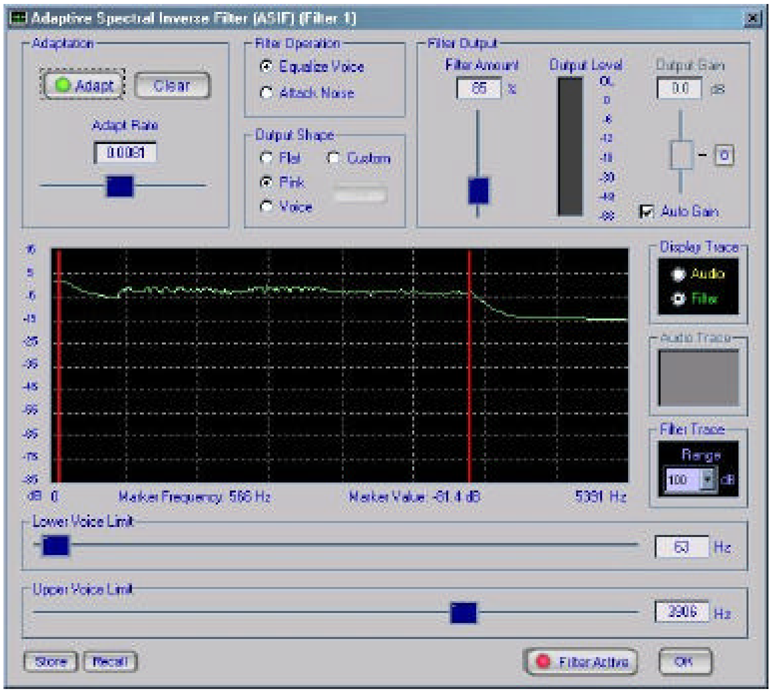

Figure 4-70: Filter response from Example 1 with Filter Amount 85%

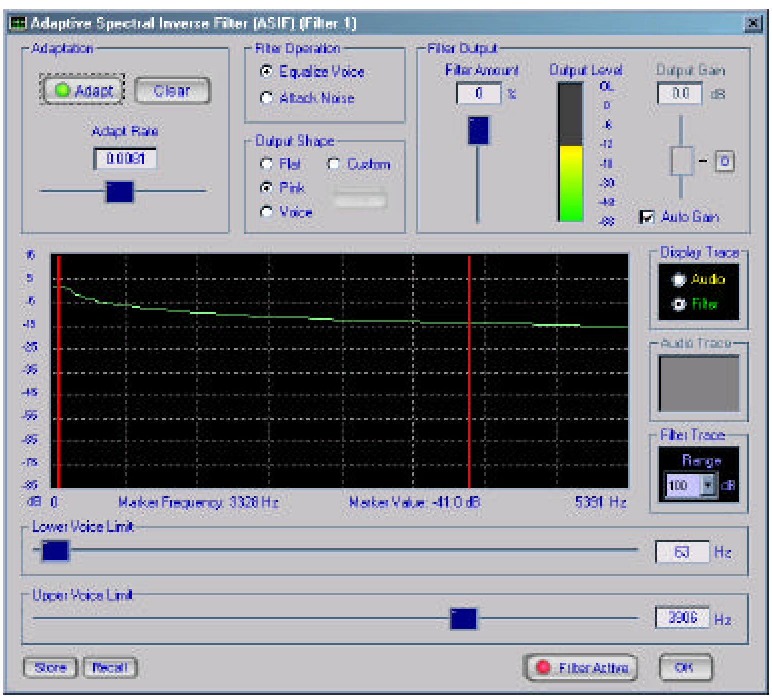

Figure 4-71: Filter response from Example 1 with Filter Amount 0%

4.6: EQUALIZER CONTROL WINDOWS

This section provides detailed description of the control window for each equalizer mode. For any digital Equalizer block, the control window for the selected equalizer is accessed by clicking on the Control button.

The output equalizer is used to reshape the noise-reduced digitally filtered signal. In the process of enhancement, high frequencies might be boosted (this is common with both the 1-Channel adaptive and Spectral Inverse filters). The output equalizer enables the operator to restore naturalness to the speech and de-emphasize residual high-frequency hiss.

4.6.1: 20-Band Graphic Equalizer

Application:

> *The 20-band Graphic Equalizer is an easy-to-use linear-phase FIR digital filter that is used to reshape the spectrum of the final output signal. Reshaping is accomplished with twenty vertical scroll bars (also called "slider" controls) which adjust the attenuation of each frequency band. These controls are very similar to the slider controls found on analog graphic equalizers found on many consumer stereo systems, and thus should be very familiar to even the novice user.*
>
> *However, unlike analog graphic equalizers, this digital equalizer has some very powerful additional capabilities. For example, the* ***Normalize*** *button allows the user to instantly move all slider controls up until the top slider is at 0dB. The* ***Zero All*** *button instantly sets all the sliders to 0dB, while the* ***Maximize*** *button instantly sets all the sliders to -40dB. The* ***All Down 1dB*** *button instantly moves all sliders down in 1dB increments. The* ***Link Frequency Sliders*** *feature allows all the sliders to move together in sync whenever one is moved. None of these functions is available in an analog graphic equalizer! Notice also that the 20 sliders are spread across the selected Bandwidth and that the frequency spacing is optimized for voice processing.*
>
> *Additionally, since a computer with a disk drive operates the equalizer, a* ***Store*** *and* ***Recall*** *capability is available. This allows the user to store commonly-used slider configurations in disk memories so that they can be instantly recalled later whenever they are needed, without having to manually adjust the slider controls.*

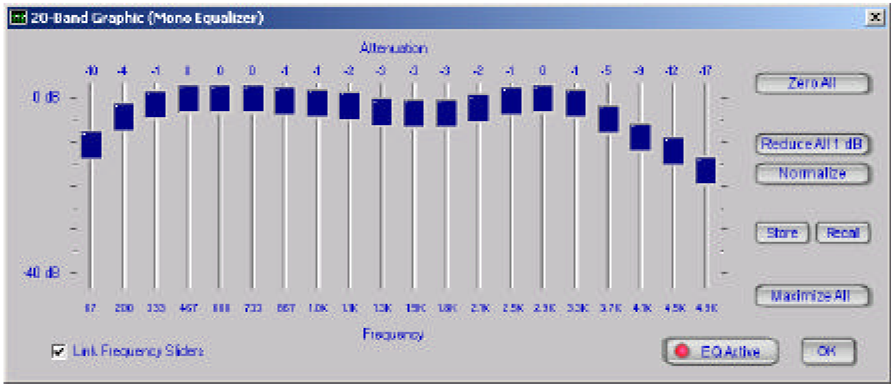

Figure 4-72 20-Band Graphic Equalizer Control Window

Description of controls/indicators is as follows:

Slider controls:	The twenty vertical scroll bar "slider" controls are used to set the frequency response of the equalizer. Each slider can set the gain of its frequency band to any value between 0dB and -40 dB in 1dB steps.
Link Frequency Sliders Checkbox:	When checked, this feature links the 20 slider controls together such that when any one of them is moved, all others move together in sync. Uncheck the box to move the sliders without affecting the others.
Center Frequency:	Note that the Center Frequency of each band is labelled underneath each slider, and that bands are more closely spaced at low frequencies
Gain Indication:	Above each slider control, the gain for that frequency band is given. The gain can also be visualized graphically by the position of the slider control.
Normalize Button:	This button instantly shifts all slider controls up together until the top slider is at 0dB. After normalization, the relative positioning of the sliders remains the same. This allows the digital equalizer to implement the desired equalization curve with minimum signal loss.
Zero All Button:	This button instantly moves the slider controls for all bands to 0dB, defeating the entire equalizer. This is a useful feature when it is desired to reset all sliders from scratch.
Maximize All Button:	This button instantly moves the slider controls for all bands to -40dB, maximizing the attenuation for all bands. This is a useful feature when it is desired to quickly adjust the sliders such that only a few bands are passed with all others rejected.

All Down 1dB Button: This button shifts all sliders down by 1dB from their current position; no slider, however, will be allowed to go below -40dB. This button allows the user to shift the entire equalizer curve down so that there will be room to move one or more sliders up relative to the others.

Store Button: This button allows the user to store a slider configuration to a user-specified disk file that will not be lost when the computer is turned off. Clicking this button brings up the dialog shown in Figure 3-12 of Chapter 3: .0. Save the configuration to any specified filename; clicking the **Cancel** button exits the Store window without storing the slider configuration in any of the memories.

Recall Button: This button allows the user to recall a previously stored slider configuration from any of the saved disk files previously generated using the **Store** button. Clicking this button brings up the window given in Figure 3-13 of Chapter 3: .0.

NOTE: The curve that is in memory prior to recalling any specified file **EXCEPT** the one named “Previous Filter Settings” will be stored in “Previous Filter Settings” as part of the recall operation. Therefore, you can reload the previous curve at any time by recalling the “Previous Filter Settings” file.

EQ Active Button: Used to switch the equalizer in and out of the process without affecting the operation of the other filters. Button LED is illuminated RED when the equalizer is in the process, and is not illuminated (GRAY) when the equalizer is out of the process.

NOTE: The EQ Active button is equivalent to, and linked with, the Active button in the associated Equalizer block on the Master Control Panel.

OK Button: Use either this button or the “X” button at the upper right corner of the window to exit the dialog and return to the Master Control panel.

4.6.2: Spectral Graphic Equalizer

Application:

For many output equalization requirements, the 20-Band Graphic Equalizer described in Section 4.6.1: should be adequate. Its resolution is limited, having only 20 coarsely-spaced slider controls; some applications require finer control of the output spectrum shape. For this reason, the 115-band Spectral Graphic Equalizer is provided (a 460-band version is also available as the Hi-Res Graphic Filter described in Section 4.4.13:).

The Spectral Graphic Equalizer is essentially a 115-band graphic equalizer, but does not have 115 separate slider controls. Rather, it allows the user to precisely draw the desired equalizer shape on the computer screen, using the mouse, with as much or as little detail as desired. Once the equalizer shape has been drawn, a linear-phase digital filter is created and loaded into the digital equalizer stage for implementation.

*The **Edit** feature allows the user to make adjustments to the equalizer shape, while the **Normalize** button allows the user to shift the entire equalizer curve up until the highest point is at 0dB.*

*A **Store** and **Recall** capability is also provided to allow the user to store commonly-used equalizer shapes to disk memories so that they can be recalled later.*

Description of operation:

The Spectral Graphic Equalizer operation is identical to the Hi-Res Graphic Filter of Section 4.4.13: , with the singular exception that its resolution is one-fourth as great. Please refer to this section for operating instructions and illustrations.

4.6.3: Dual Parametric Equalizer

The Dual Parametric Equalizer is identical to the Tri-Parametric Filter described in Section 4.4.14: , except that only two substages are available instead of three. Please refer to that section for operating instructions.

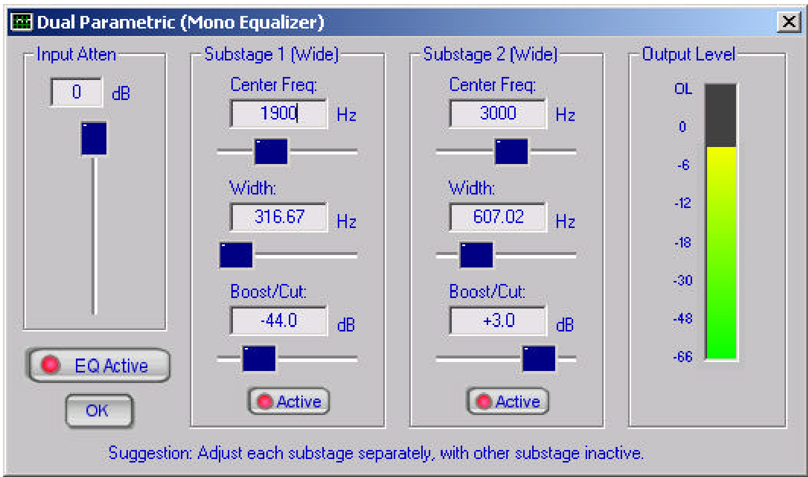

Figure 4-73 Dual Parametric Filter

4.6.4: Limiter/Compressor/Expander

The Equalizer Stage Limiter/Compressor/Expander is identical to that found in the General Filter Stages. Please refer to Section 4.4.17: for more details on this tool.

4.7: STORING AND RECALLING INDIVIDUAL FILTER SETTINGS

Several of the filters and equalizers provide the ability to store their individual settings to a disk file. These settings can be recalled for later use with other configurations. These individual filter settings files are in addition to the setup files that store the entire configuration. Filters or equalizers that provide these features will have a **Store** or **Recall** button on their control windows. Table 4-1 shows the different filters that allow store and recall of their individual settings.

Table 4-1 Individual Filter Settings File Extensions

Filter Type	Individual Settings File Extension
20-Band Graphic Filter and Equalizer	.tbg
Spectral Inverse Filter	.sif
Spectral Inverse Filter Custom Curve	.scc
Hi-Res Graphic Filter	.hrf
Spectral Graphic Equalizer	.edf
NoiseEQ Broadband Filter	.neq
Adaptive Filter Coefficients	.cof

4.7.1: Storing Individual Settings Files

Upon clicking the **Store** button from the respective filter control window the Store Filter window will appear. This window will be intialized to open in the current working folder (See Section 6.3: for more information about the current working folder). The appropriate file extension will be placed in the **File Name** entry field. If you leave the file extension off of the file name it will be automatically appended for you. Clicking on a valid individual settings file from within the file selection window will fill the **File Name** entry field with the name of the file clicked. If the file already exists a window will appear requesting verfication before the file is overwritten. Once a valid file name has been entered clicking the **OK** button will store the settings file and dismiss the Store Filter window.

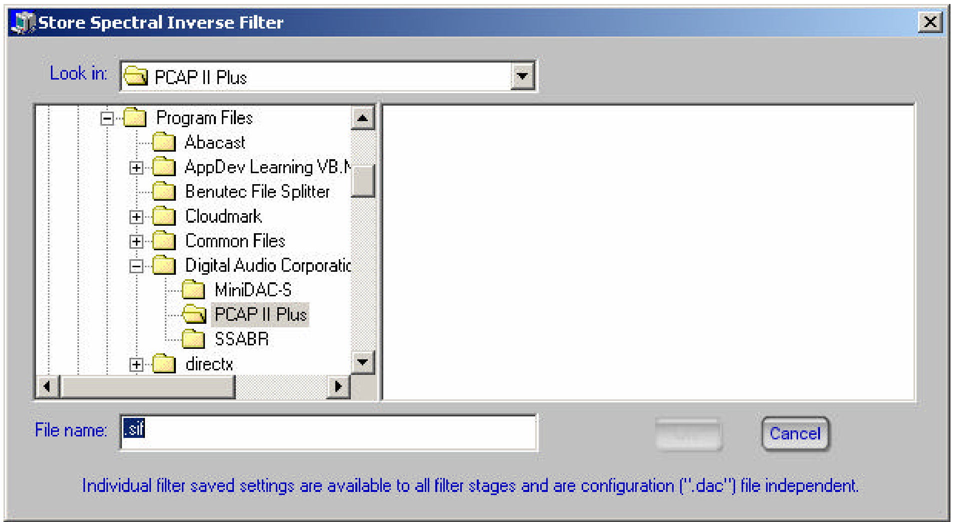

Figure 4-74 Store Individual Settings Window for the SIF

4.7.2: Recalling Individual Settings Files

Upon clicking the **Recall** button from the respective filter control window the Recall Filter window will appear. This window will be intialized to open in the current working folder (See Section 6.3: for more information about the current working folder). The appropriate file extension will be placed in the **File Name** entry field. If you leave the file extension off of the file name it will be automatically appended for you Clicking on a valid individual settings file from within the file selection window will fill the **File Name** entry field with the name of the file clicked. Once a valid file name has been entered clicking the **OK** button will store the settings file and dismiss the Recall Filter window.

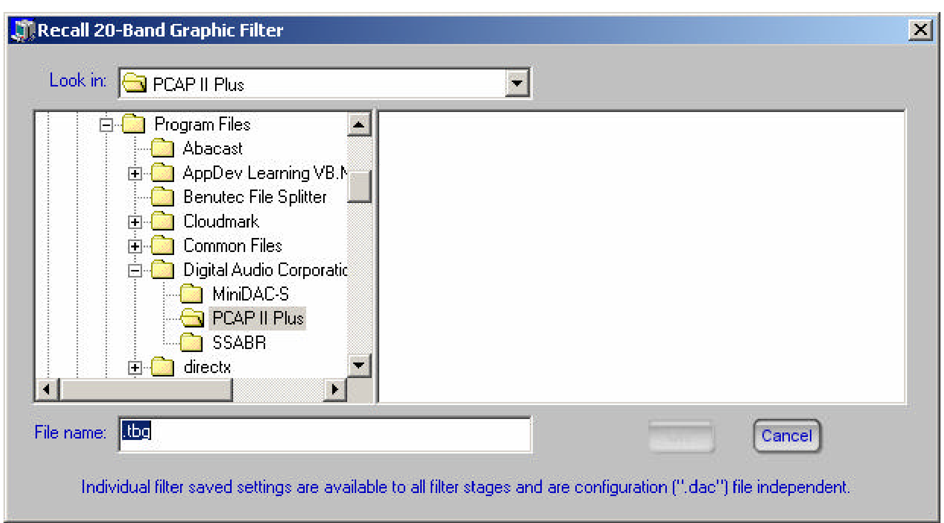

Figure 4-75 Recall Individual Settings Window for the 20-Band Graphic Filter

4.8: VISUALIZATION TOOLS

4.8.1: Spectrum Analyzer and Coefficient Display Buttons

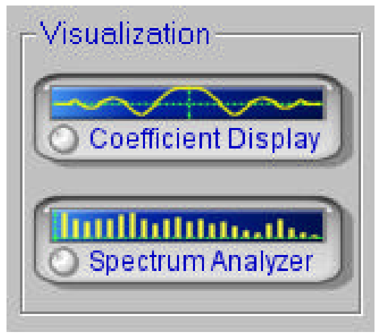

Figure 4-76 Spectrum Analyzer and Coefficient Display Buttons

Clicking the **Spectrum Analyzer** button until the LED is illuminated BLUE brings up the Spectrum Analyzer window; clicking until the LED is not illuminated closes the window. For fast access to the Spectrum Analyzer window, press <Alt-S> on the keyboard. See Section 4.8.2: for complete instructions on operating the controls in the Spectrum Analyzer window.

Similarly, clicking the Coefficient Display button until the LED is illuminated BLUE brings up the Coefficient Display window; clicking until the LED is not illuminated closes the window. For Fast access to the Coefficient Display, press <Alt-E> on the keyboard. See Section 4.9: for complete instruction on operating the controls in the Coefficient Display window.

4.8.2: Spectrum Analyzer Window

Application:

To properly utilize the processing tools, it is often necessary to measure the frequency characteristics of the input signal. This assists in determining the type of filtering needed. Also, after processing the signal, it may be desirable to compare the frequency characteristics of each digital filter output to those of the input signal, thus determining the effectiveness of each digital filter. A dual-channel FFT spectrum analyzer with selectable inputs is ideal for accomplishing these tasks.

*The dual-channel FFT spectrum analyzer is used to view the frequency spectrum of the signal at any stage of the enhancement process. Two traces, **Trace 1** and **Trace 2**, can be displayed either simultaneously or separately. Either trace can be configured to the signal spectrum at any point in the processing chain. The **Averager** feature combines successive spectra to achieve a slower, smoother display. Each trace consists of 460 spectral lines with a useable dynamic range of 70dB. Adjustable **Gain** controls allow up to 40dB of digital gain to be applied to each trace to boost low level signals to better fit within the this dynamic range. An overall dynamic range of 110 dB is thus available.*

*A moveable **Marker** allows frequency and magnitude readout at any point in the two spectra. The **Find Peak** feature allows the marker to be moved instantly to the largest magnitude displayed.*

*Finally, the Spectrum Analyzer window is fully sizeable, and can utilize all the available display area for viewing if desired. Controls can be hidden using the **Hide Controls** checkbox, and printouts can be generated using the **Print** button.*

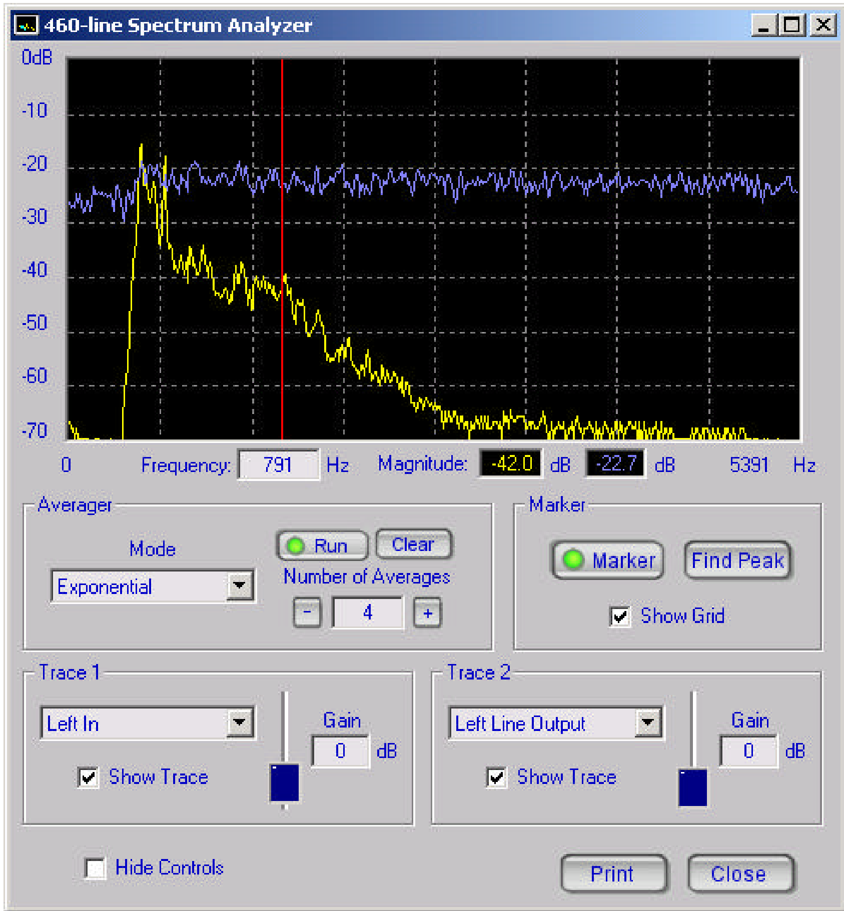

Figure 4-77 Spectrum Analyzer Window

Description of controls/indicators is as follows:

Averager Block: Controls averaging of successive FFT spectral traces.

Number of Averages selects the number of averages to be applied to spectral traces in the Exponential and Peak Hold averaging modes (explained below). The more averages applied, the smoother the displayed spectrum waveforms will be; however, the waveforms will also update more slowly as Number of Averages is increased. For voice filtering applications 8 to 32 averages are recommended. The user can freeze an averaged trace by clicking **Run** button until the LED is not illuminated; resume the analyzer screen update by clicking **Run** button until the LED is illuminated GREEN. The user may clear the averaged trace(s) and restart the average at any time by clicking on the **Clear** button. The **Mode** selection allows the user to switch the averaging mode between **No Average** (non-averaged spectrum displayed for fastest response), **Exponential** averaging (normally recommended – spectrum builds both up and down using specified Number of Averages), or **Peak Hold** averaging (spectrum only builds up using specified Number of Averages, can never come down, thus capturing any strong peaks that might occur).

Marker Block: Used to turn the vertical red marker in the Spectrum Display area on and off via the **Marker** button; GREEN LED indicates the marker is on, LED OFF indicates the marker is off. The marker allows frequency (**Freq**) and magnitude (**Mag**) readout of any point in the spectra. Also, the **Show Grid** checkbox can be used to turn the grid markings on and off as needed for viewing the spectral graph most clearly.

To move the marker, simply click and/or drag the mouse cursor to any desired point within in the Spectrum Display area. Clicking the **Find Peak** button instantly moves the marker to the largest magnitude in the Spectrum Display.

Trace 1 Block: Defines what is displayed by the yellow-colored spectrum trace. The signal selection combo box allows the user to place the analysis for Trace 1 at any point in the processing chain; available options depend upon the **System Bandwidth** and **Input Mode** settings.

The **Gain** scroll bar allows the user to apply digital gain to the Trace 1 input signal prior to FFT analysis, allowing signals which fall below the 70dB dynamic range of the analyzer to be viewed. **Gain** may be adjusted from 0dB (no gain) to 40dB.

If the **Gain** setting is too large, the Trace 1 input may be overloaded, causing a distorted spectrum display. When an overload does occur, the **Gain** label will change to **OVL** and the yellow magnitude readout at the bottom of the display will change from yellow to red. Overloads affect only the display and have no effect on the processed signal.

The **Show Trace** checkbox allows Trace 1 to be turned on and off as necessary; note that showing both Trace 1 and Trace 2 will tend both to clutter the grid and slow down the screen update; it is therefore recommended that only one or the other be checked at any given time, unless overlayed viewing is needed.

Trace 2 Block: Identical to the Trace 1 Block, except that it manages the blue-colored spectrum trace.

Spectrum Display: Spectra are displayed in Cartesian (X,Y) format, with the Y axis representing magnitude in dB and the X axis representing frequency in Hertz. An optional grid can be superimposed on the black background to assist the user in visually judging frequency and magnitude of spectral components.

A vertical red marker is used to read out the exact magnitude(s) (**Mag**) of any frequency (**Freq**) in the spectrum display. To manage the marker, simply click the mouse cursor on the desired point in the Spectrum Display area, or utilize the controls in the Marker Block, described above.

Hide Controls Checkbox:	Unchecking this box allows the Spectrum Display to utilize the full window space; recheck the box if access to the controls is required.
Print Button:	Generate a hardcopy printout of the spec- trum at any time by clicking this button.
Close Button:	Clicking on this button or the "X" button at the upper right corner of the window turns off the Spectrum Analyzer display.

4.9: COEFFICIENT DISPLAY WINDOW

Application:

Particularly when setting up the Ref Canceller filter, it is often useful to display the impulse response (filter coefficients) of the filter. Additionally, it is sometimes desirable to know the precise time-domain response of any of the General Filter stages. For these reasons, the Coefficient Display window has been provided.

The Filter stage to be displayed is specified in the ***Filter*** *combo box within the* ***Display*** *block by clicking on the desired Filter. The number of filter coefficients to be displayed is specified by the* ***Number of Coefficients*** *control by clicking on the desired number; this causes the specified number of coefficients to be horizontally scaled to fit within the available display area. Regardless of the length of the Filter, the last selection in the combo box causes all coefficients in that Filter to be displayed.*

Vertical scaling of the Filter's coefficients for display is accomplished by clicking on the desired ***Zoom*** *factor. Supported* ***Zoom*** *factors range from* ***1X*** *to* ***200X****.*

A moveable ***Marker*** *allows* ***Time, Value****, and* ***Coefficient*** *number readout at any point in the Coefficient Display. The marker can be turned on and off; also, the visualization grid can be turned on and off as needed.*

Finally, the Coefficient Display window is fully sizeable, and can utilize all the available display area for viewing if desired. Controls can be hidden using the ***Hide Controls*** *checkbox, and printouts can be generated using the* ***Print*** *button.*

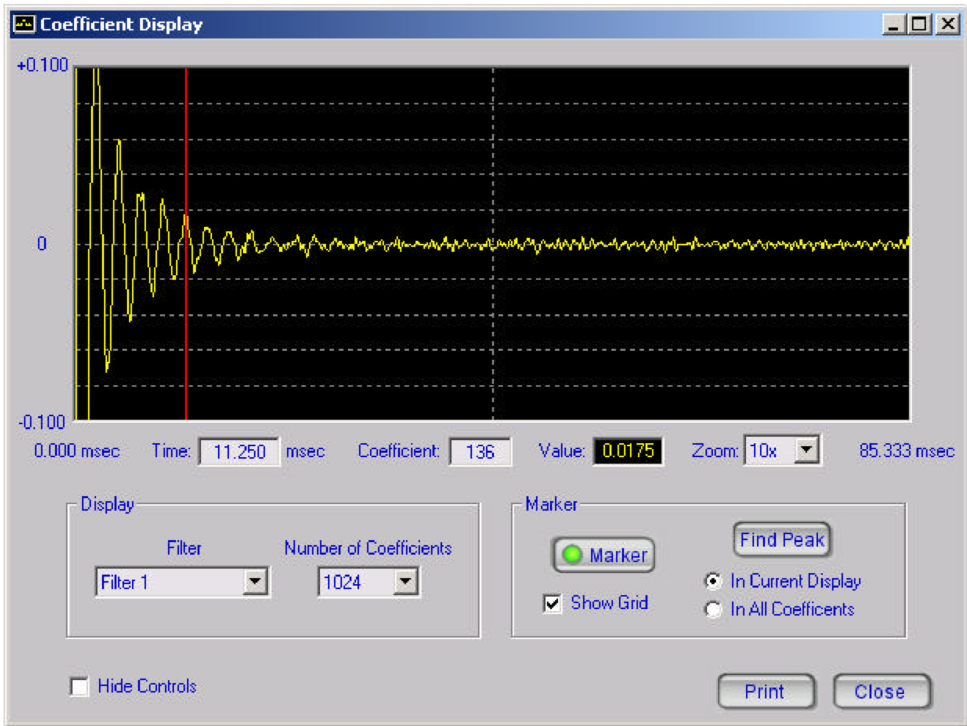

Figure 4-78 Coefficient Display Window

Description of controls is as follows:

Display Block: Used to select the General Filter stage whose coefficients are to be displayed via the **Filter** combo box; available options depend upon the **System Bandwidth** and **Input Mode** settings. Also, the **Number of Coefficients** to be displayed can be specified; options are power-of-two steps from 32 up to the Filter Size as specified in the control window for the filter.

Marker Block: Used to turn the vertical red marker in the Coefficient Display area on and off via the **Marker** button; GREEN LED indicates the marker is on, LED OFF indicates the marker is off. The marker allows **Time**, **Coefficient** number, and **Value** readout of any coefficient in the filter. Also, the **Show Grid** checkbox can be used to turn the grid markings on and off as needed for viewing the impulse response curve most clearly.

To move the marker, simply click and/or drag the mouse cursor to any desired point within in the Coefficient Display area. The Coefficient Display area will "pan", or shift, left or right whenever the marker selects a coefficient that is outside the displayed area; click on the left or the right side of the Coefficient Display grid to pan left or right. Clicking the **Find Peak** button instantly moves the marker to the largest magnitude in the Spectrum Display. Alternatively, entering the desired coefficient number in the text box moves the marker instantly to the desired point.

Zoom Combo: Used to specify the vertical zoom factor to be used when displaying coefficients. Coefficients may be zoomed from **1X** (no scaling) to a maximum of **200X**. If the scaled coefficient exceeds the maximum vertical display limits, it will be clipped prior to display. **NOTE:** This clipping does not affect signal processing.

Coefficient Display: Coefficients are displayed in Cartesian (X,Y) format, with the Y axis representing **Value** in a range from -1.0 to +1.0. The X axis represents **Time** in seconds. A yellow grid is superimposed on the black background to assist the user in visually judging **Time** and **Value** of Filter coefficients.

A vertical red marker is used to read out the exact **Value** and **Time** of any Filter coefficient. To move the marker, simply click the mouse cursor on the desired point in the Coefficient Display area, or utilize the controls in the Marker Block, described above.

Hide Controls Checkbox: Unchecking this box allows the Coefficient Display to utilize the full window space; recheck the box if access to the controls is required.

Print Button: Generate a hardcopy printout of the impulse response at any time by clicking this button.

Close Button: Clicking on this button or the "X" button at the upper right corner of the window turns off the Coefficient Display.

4.10: MASTER CONTROL PULLDOWN WINDOWS

4.10.1: Saving Setups to Disk Files

Application:

To save time configuring detailed control settings, complete setups may easily be saved to disk setup files for future reuse. These files are particularly handy when making presentations which require multiple setups, or when it is desired to precisely duplicate the enhancement procedure at some point in the future. Also, this feature allows easy transfer of enhancement setups between PCAP II Plus systems simply by exchanging the ".DAC" setup files.

***Note**: Any ".DAC" setup files previously stored by the Digital Audio Corporation **PCAP, PCAP II, or MCAP** software can be loaded by the **PCAP II Plus** software.*

Save a setup to a disk file as follows:

1. Click on **File** on the Master Control menu bar. When the pulldown menu appears, click on **Save Setup File**. This will cause the window in Figure 4-79 to appear:

2. Normally, it will not be necessary to change the path setting; if, however, you desire to place the setup file into a different drive (such as a floppy drive) or folder, utilize the explorer to navigate the available drives and folders and make the desired selection.

3. Each stored setup will have a user-entered text **Description** to uniquely identify it. You must enter the **Description** text before storing the file. To enter or edit the **Description** text, click on the **Description** text box and type in any text desired. It is recommended that the text be descriptive of the setup's application.

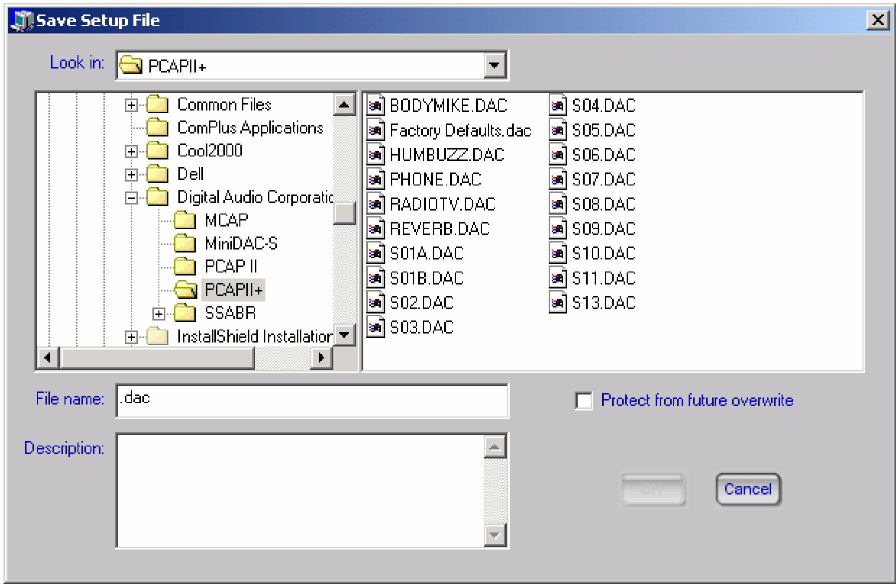

Figure 4-79 Save Setup File Window

4. You will need to specify a filename for the setup. Click on the **File Name** text box, then type the desired filename (up to 256 characters). All setup filenames **must** have the **.DAC** extension; thus, the **.DAC** extension is automatically included in the text box. If you delete the **.DAC** extension, an error message will be generated.

5. If you wish to protect your setup file from being accidentally overwritten in the future by the **Save Setup File** window, click the **Protect from Future Overwrite** box. This does not prevent deletion or overwrite by Windows Explorer or other applications, but merely sets a special flag bit inside the file.

6. Click on **OK** to save the setup file with the selected filename to the specified drive and directory.

4.10.2: Recalling Setups from Disk Files

Application:

To save time configuring detailed control settings, complete setups may easily be saved to disk *<u>setup files</u>* *for future reuse. These files are particularly handy when making presentations which require multiple setups, or when it is desired to precisely duplicate the enhancement procedure at some point in the future. Also, this feature allows easy transfer of enhancement setups between PCAP II Plus systems simply by exchanging the ".DAC" setup files.*

Note*: Any ".DAC" setup files previously stored by the Digital Audio Corporation* ***PCAP, PCAP II, or MCAP*** *software can be loaded by the* ***PCAP II Plus*** *software.*

Open a setup from a disk file as follows:

1. Click on **File** on the Master Control menu bar. When the pulldown menu appears, click on **Open Setup File**. This will cause the following window (Figure 4-80) to appear:

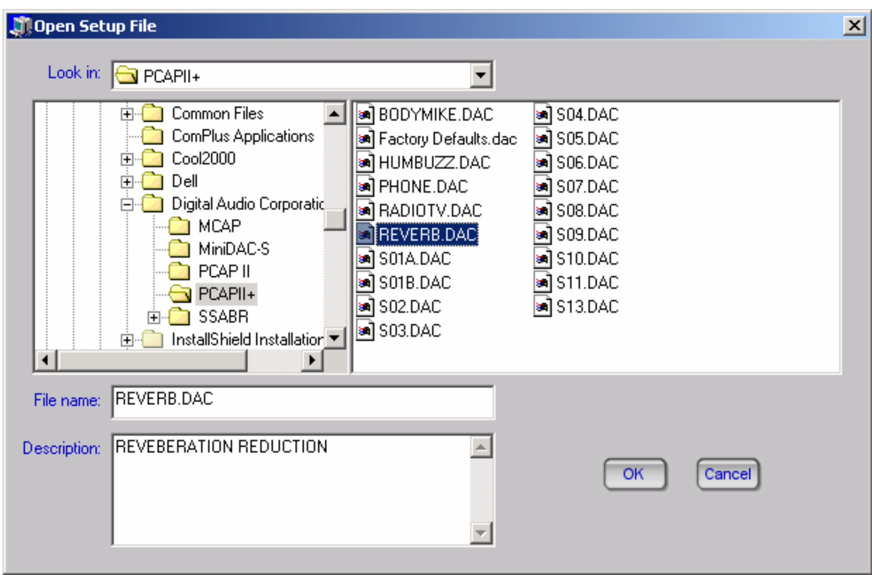

Figure 4-80 Open Setup File Window

2. Normally, it will not be necessary to change the path setting; if, however, you desire to open a setup file from a different drive (such as a floppy drive) or folder, utilize the explorer to navigate the available drives and folders and make the desired selection.

3. Each stored setup has a user-entered text **Description** to uniquely identify it. These are easily browsed prior to recalling a setup, thus preventing time from being wasted if the

wrong setup file is selected. To browse a **Description**, scroll to and click on the desired file listed in the **Setup Files** list box.

4. Once the desired setup file name has been found, either double-click it or click on **OK** to open it. A message will alert the user that 60 seconds may be required to completely open the setup.

An "hourglass" mouse cursor will now appear, indicating that the PCAP II *Plus* is busy configuring itself with the recalled settings from disk.

If a settings file contained any adaptive filters (1 Channel Adaptive or Reference Canceller), individual files entitled “Previous Filter X Settings.cof” (where X is the filter stage containing an adaptive filter) will be created. If the configuration of the PCAP II *Plus* is in Stereo-Independent mode the files will be entitled “Previous Left Filter X Settings.cof” or “Previous Right Filter X Settings.cof”. This feature allows recalling the coefficient solution from the previous session. See section 4.4.1.2: for more information.

4.10.3: Using Calibration Mode

Application:

The PCAP II Plus calibration mode allows verification of the PCAP II Plus functionality.

When enabled the PCAP II *Plus* will output on both digital and analog outputs the signal type selected in the **Signal Generator Output** area. One of a number of tests can be selected and run via the **Run Test** button. To download a complete calibration and test procedure in ".pdf" format, please visit the DAC website at www.dacaudio.com.

Figure 4-81 Calibration Mode Option in Settings Menu

Upon enabling the Calibration Mode, the Software Input Level Controls will be enabled to allow precise control of input levels. The AGC will be disabled to allow a constant output level. Note that if you use an RCA cable to loop the output to the input of the PCAP II *Plus*, you will be able to view the effects of filters through the signal path with the Spectrum Analyzer. However, you will not be able to listen to the filter effects as the PCAP II *Plus* will always output the pure generated signal while in Calibration Mode.

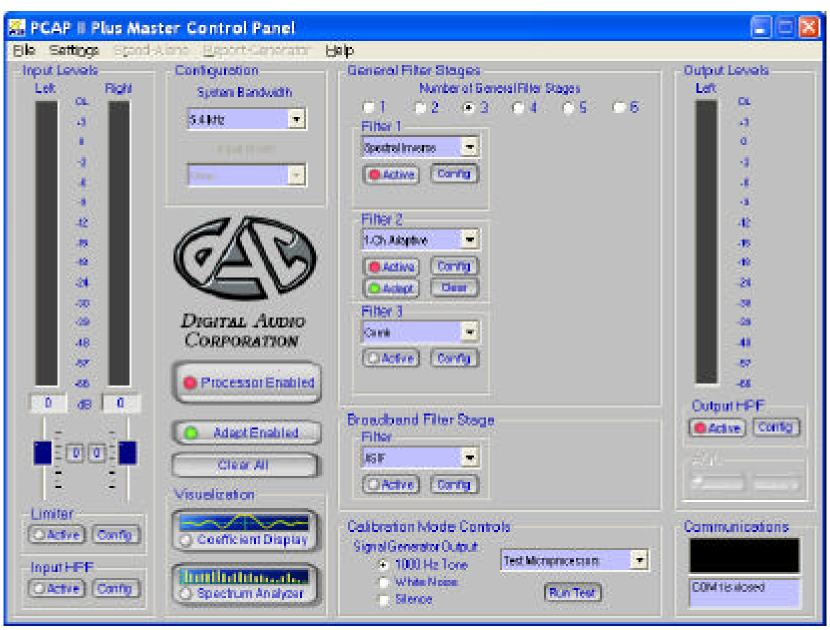

Figure 4-82 The Master Control Panel in Calibration Mode

4.10.4: Storing Setups to External Processor Stand-Alone Memories

Application:

The PCAP II Plus external processor has the capability to store up to ten enhancement setups internally in nonvolatile ***Stand-Alone*** *memories, allowing the external processor to be operated in those setups without needing to be connected to a PC. This makes it possible to store sophisticated enhancement setups in one or more external processors in the laboratory, then use them as portable field processors which have simple controls that can be operated by anyone. Thus, with this feature, the complete capabilities of the enhancement laboratory and the expertise of the laboratory personnel can be extended to field operations.*

For all memories, the function of the ***AUX*** *switch on the external processor front panel can be specified.*

Store a setup in a nonvolatile memory as follows:

1. Adjust all Master Control, General Filter Stages, Broadband Filter Stages, and Equalizer settings as desired to achieve the desired enhancement setup. This may also be accomplished using the **Open Setup File** feature described in Section 4.10.2: .

   It is strongly suggested that you save this enhancement setup to disk using the **Save Setup File** feature described in Section 4.10.1: . *NOTE: There is no way for the Master Control program to retrieve from the PCAP II Plus and interpret the settings and coefficients stored in the Stand-Alone memories.*

2. Click on **Stand-Alone** on the Master Control Panel menu bar to access the following pulldown menu (Figure 4-83):

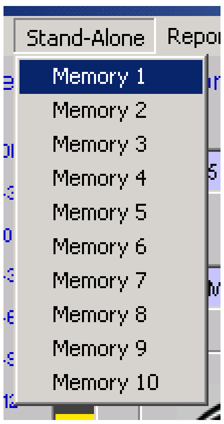

Figure 4-83 Stand-Alone Pulldown Menu

3. Click on the selected memory. The following window (Figure 4-84) will now appear:

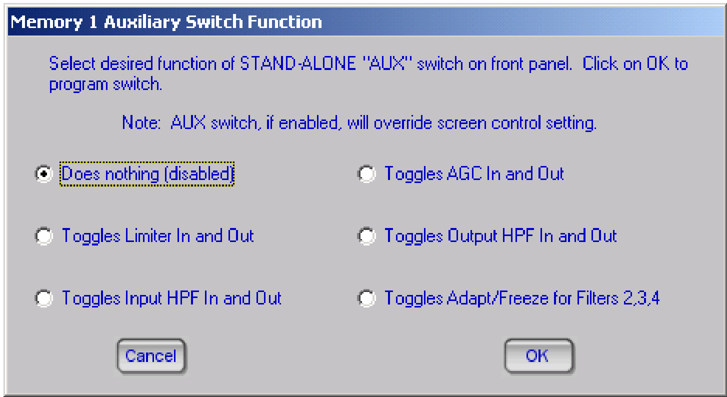

Figure 4-84 Auxiliary Switch Function Window

This window allows the function of the **AUX** switch on the external processor front panel to be programmed. Click on the program option desired, then click on **OK**.

4. A message window will now appear to alert you that the previous contents of the selected memory will be lost, and that 30 seconds could be required to transfer all settings to the memory. Additionally, the window will suggest that you also save your setup in a disk file. Click on **Yes** if you wish to proceed, or click on **No** to cancel without affecting the current memory contents.

An "hourglass" cursor will now appear, indicating that the external processor is busy storing the setup in the selected Stand-Alone memory. For complete instructions on operating the PCAP II *Plus* external processor from the Stand-Alone memories, please consult Section 5: .

4.10.5: Generating Setup Reports

The Master Control program has a **Report-Generator** feature which allows hardcopy printouts of all PCAP II *Plus* control settings to be generated. This is useful when the enhancement procedure needs to be documented. To access this feature, click on **Report-Generator** in the Master Control menu bar. The following pulldown menu (Figure 4-85) will be displayed:

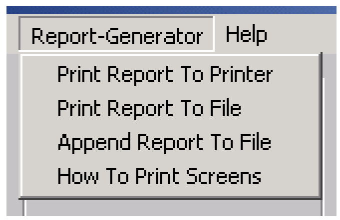

Figure 4-85 Report-Generator Pulldown Menu

Clicking the **Print Report to File** option prints the control settings to a user-specified ASCII text file, which could later be imported to a word processor (such as Microsoft Word, or the *Write* utility shipped with Microsoft Windows) for editing. All previous contents of the specified file will be overwritten. However, this option is limited, since no graphics can be stored in a text file. If you wish to transfer graphical information to a word processor, click on **How to Print Screens** for instructions.

Clicking the **Append Print Report to File** option causes the printed control settings to be tacked on to the end of any existing text file, without overwriting the previous contents. As with the **Print Report to File** option, no graphics can be stored.

Clicking the **Print Report to Printer** option sends all control settings, including graphs for certain Filter and Equalizer modes, to the Windows Print Manager for printing. For this feature to work properly, a printer connection needs to be available to your computer.

4.10.6: Getting Online Help

Context-sensitive online help is available to the PCAP II *Plus* Master Control program at any time by pressing the <F1> key. This is a standard feature of Microsoft Windows programs, and is the recommended method for accessing help.

The term "context-sensitive" means that the help text that is displayed depends upon the window from which you are currently operating. For example, if you are operating the Spectrum Analyzer controls and you press the <F1> button, help text for the Spectrum Analyzer will be displayed.

An alternative method for accessing help is to click on **Help** in the Master Control menu bar. The following pulldown menu (Figure 4-86) will be displayed:

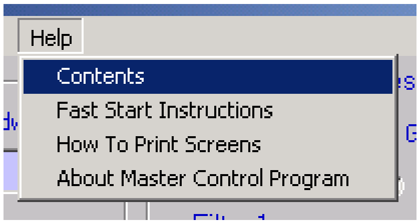

Figure 4-86 Help Pulldown Menu

Click on **Contents** to access the help Contents window. The Contents window will display all the subjects for which help is available for the PCAP II *Plus* Master Control program. You may browse through the displayed subjects and select help for a particular subject by double-clicking the desired subject.

The help utility is, in fact, a separate Windows program from the Master Control software. Once it has displayed the help text, it will remain on the screen as a separate window until it is closed by the user.

5: . OPERATING PCAP II STAND-ALONE

Before the PCAP II *Plus* external processor can be operated Stand-Alone, at least one of the Stand-Alone setup memories must be programmed using the procedure in Section 4.10.1: .

Operate the PCAP II external processor as a Stand-Alone audio processor as follows:

1. Connect 12VDC power, and switch the POWER switch ON.

2. For analog sources, connect the signals to be processed to the rear panel LEFT and RIGHT ANALOG INPUTS RCA jacks and set the INPUT SELECT switch to ANALOG for most filtering applications. For Ref Canceller adaptive filtering applications, the original (PRI) audio is input to the LEFT (white) RCA, while the noise reference (REF) audio is input to the RIGHT (red) RCA. Alternatively, connect a digital signal source to the S/PDIF digital INPUT (white) RCA jack and set the INPUT SELECT jack to DIGITAL.

3. Adjust the front panel LEFT and RIGHT INPUT LEVELS controls such that the tricolor LEDs indicate GREEN most of the time, with occasional YELLOW peaks.

4. Connect any recorder with analog inputs to the rear panel LEFT (white) and RIGHT (red) ANALOG OUTPUTS RCA jacks if recording the processed audio is desired. Alternatively, connect a recorder with digital input to the rear panel S/PDIF digital OUTPUT (red) RCA jack and select the desired recording sample rate using the OUTPUT SAMPLE RATE switch (44.1kHz is normally recommended, especially if the material is to be transferred to compact disc).

5. Plug stereo headphones into the front panel PHONES jack. Adjust the level control to a comfortable listening VOLUME.

6. Select the desired Stand-Alone memory by switching the MEMORY switch to one of the ten positions.

7. Initially, set the front panel ADAPT/FREEZE, PROCESS/BYPASS, and AUX/OFF switches to ADAPT, PROCESS, and AUX (all switches in).

While listening with the stereo headphones, switch the MONITOR switch between INPUT and OUTPUT to hear the difference between the original and processed audio, respectively. Normally, the MONITOR LEFT and RIGHT pushbuttons are kept pushed in, but either (or both) can be pushed out at any time to listen only to either the left or right channel audio, or to mute all headphone audio. You may make the following adjustments to the processor at any time:

1. To freeze all adaptive filters in the process, switch the ADAPT/FREEZE switch to FREEZE. To allow the adaptive filters to adapt to changing signals, switch to ADAPT.

2. To bypass all digital filters, allowing the original unfiltered audio to pass through to the LEFT and RIGHT ANALOG OUTPUTS and S/PDIF digital OUTPUT RCAs, switch the PROCESS/BYPASS switch to BYPASS. To filter the audio, switch to PROCESS.

3. The AUX switch is programmed to have one of the following functions when switched to AUX:

    a. No effect at all (default)

    b. Input Limiter IN

    c. Input HPF IN

    d. Output AGC IN

    e. Output HPF IN

    f. Right channel Adapt/Freeze (ADAPT/FREEZE switch used for Left channel only)

    g. Filter 2-6 Adapt/Freeze (ADAPT/FREEZE switch used for Filter 1 only)

When switched to OFF, the AUX switch always has no effect on the signal.

4. To clear all filters, press the CLEAR button. This will cause all adaptive filter coefficients to be reset to zero, allowing them to re-converge anew (assuming the ADAPT/FREEZE switch is in ADAPT).

6: . PCAP II *PLUS* CUSTOMIZATION OPTIONS

6.1: CONFIGURING PCAP II FOR DIFFERENT COM PORTS

WARNING: ***Should you choose to share a single COM port connection on your computer with a PDA, e.g. a Palm Pilot, connecting each to the COM port only when needed, be sure to shut down any "Hot Sync" manager application that might be running before attempting to run the PCAP II Plus Master Control software. Otherwise communication with the PCAP II Plus will fail due to conflict with the Hot Sync manager.***

The PCAP II *Plus* Master Control software is factory-configured to communicate with the external processor over RS232 communications port #1 (COM1). This will work for the vast majority of modern PC systems. However, in some cases where COM1 is not available, it may be necessary to reconfigure the software to operate on a different COM port as follows:

1. On the PCAP II *Plus* Master Control menu bar, click on the "Settings" menu bar option.

2. When the pulldown menu appears, click on "Com Port". The display will now appear as follows (Figure 6-1).

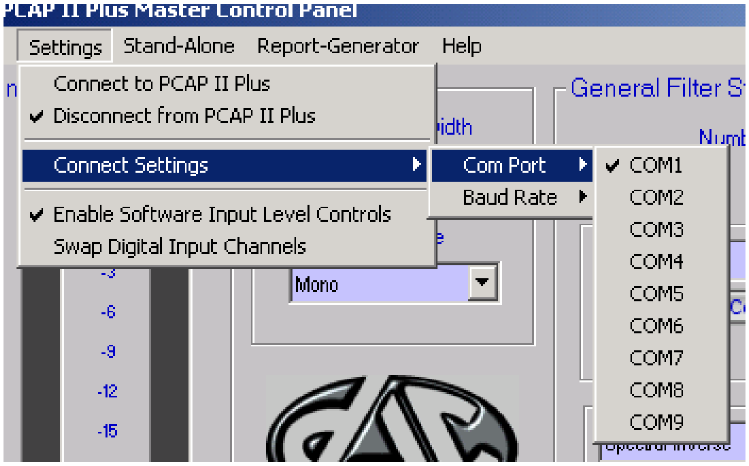

Figure 6-1 Com Port Selection Menu

3. Now click on the desired COM port. After clicking the desired COM port again click on the "Settings" menu bar option and then click on "Connect to PCAP II Plus". If any error messages appear, then you have misidentified the COM port. Repeat steps 1 through 3 until the message "Connected COM x" appears in the Communications Box. Once it does, proper communication has been achieved.

4. Click on "File", then "Exit" (or click the “X” button at the upper right corner of the Master Control screen) to exit the Master Control program. This will save the new COM port setting.

6.2: CONFIGURING PCAP II *PLUS* FOR DIFFERENT BAUD RATES

This section is intended for advanced users, only.

The PCAP II *Plus* system is capable of operating at RS232 symbol rates of 9600, 14400, 19200, 38400, and 115200 baud. Each PCAP II *Plus* is factory-configured to operate at 115200 baud, which works well for most modern PCs. However, some applications (such as modem or network links) may require the use of a different baud rate.

Change the PCAP II *Plus* baud rate as follows:

1. First, reset the internal DIP switches of the external processor for the new baud rate as follows:

    a. Power off the external processor and disconnect all cables, especially the EXTERNAL POWER and RS232 cables.

    b. Carefully remove the external processor top cover.

    c. Locate the 5-position DIP switch at the front of the internal printed-circuit board. Set DIP switch positions 1-3 to the desired baud rate as follows:

Baud Rate	DIP1	DIP2	DIP3
9600	OFF	OFF	ON
14400	ON	ON	OFF
19200	OFF	ON	OFF
38400	ON	OFF	OFF
115200 (default)	OFF	OFF	OFF

Table 6.1 Baud Rate DIP Switch Settings

    d. Replace top cover and reconnect all cables.

2. Next, configure the Master Control program for the new baud rate as follows:

    a. Make sure that the external processor is powered OFF.

    b. Power ON the computer (not the external processor) and load Windows. Run the PCAP II Master Control program by double-clicking the icon; an error message should appear indicating no communication with the box (it is powered off). Click OK and enter the program anyway.

c. On the Master Control menu bar, click on the "Settings" menu bar option.

d. When the pulldown menu appears, click on “Connect Settings”, then "Baud Rate". The display will now appear as shown in Figure 6-2.

e. Click on the baud rate setting which corresponds to the new external processor DIP switch settings. After clicking the desired baud rate again click on the "Settings" menu bar option and then click on "Connect to PCAP II Plus". If any error messages appear, recheck COM port setting, Baud Rate setting, connections, and external processor DIP switch settings.

f. Click on "File", then "Exit" to exit the PCAP II *Plus* Master Control program and save the new baud rate setting.

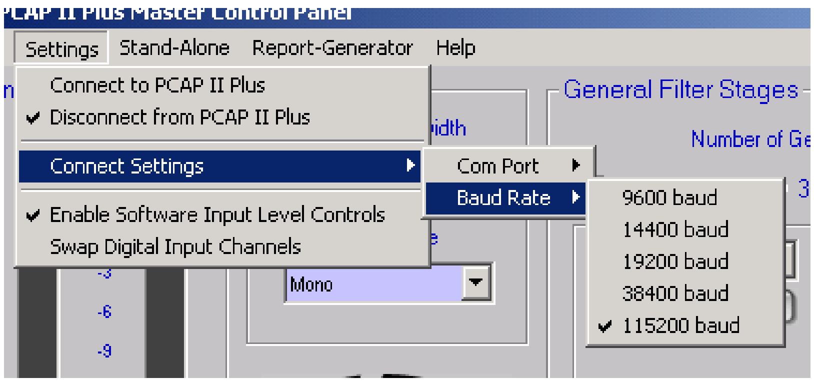

Figure 6-2 Baud Rate Selection Menu

3. After storing the new baud rate by exiting the Master Control program, test the new baud rate as follows:

a. Switch on the PCAP II external processor. Make sure that the RS232 cable is properly connected between the external processor and the PC.

b. Double-click on the PCAP II *Plus* Master Control icon to run the program.

c. Allow several seconds for the program to load and to initialize the external processor. The message "Connected COM x" should appear in the Communications Box at the lower left corner of the Master Control window.

d. Connect an audio signal (either analog or digital – make sure the correct INPUT SELECT switch setting is used!) and adjust the input level so that the tricolor level LEDs indicate GREEN.

If the Input Level bargraphs on the Master Control screen properly respond to the input signal, then the hardware and software are correctly configured for the new baud rate.

6.3: WORKING FOLDER

Within the **File** menu on the Master Control Panel is the option for **Edit Working Folder**. This option provides a way of designating where individual filter settings as well as ".dac" files will be stored.

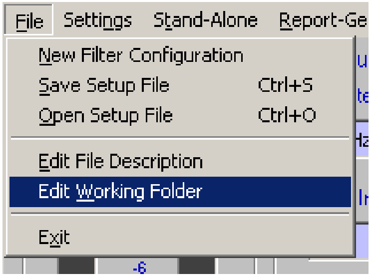

Figure 6-3 Working Folder Menu Option

Once you have clicked on the **Edit Working Folder** menu option the Default Working Folder Window will appear. The working folder selection area will default to begin set to the current working folder. To change this setting, navigate to the folder of your choice by clicking on the displayed folder. Select the new desired working folder by highlighting the desired folder.

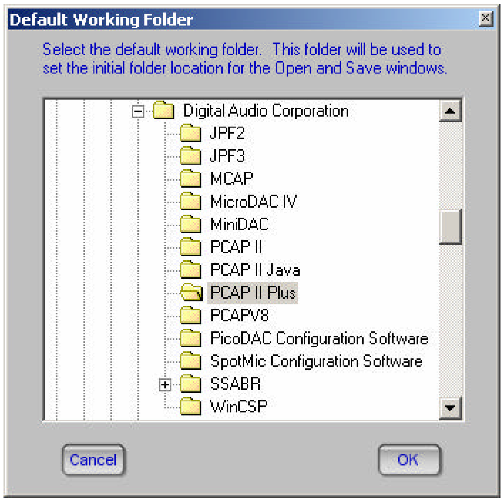

Figure 6-4 Default Working Folder Window

7: DAC1024T AND DAC4096T ADAPT RATE SETTINGS

The DAC1024T and DAC4096T both had 12 position Adapt Rate Switches on their front panels. The lower the number the switch setting was turned to the faster the filter coefficients adapted. For those users who are used to using those settings the table listed below shows how the switch settings correspond to the (ADAPT RATE) x 2^{-14} value shown on the filter control screen.

Table 7-1 Adapt Rate Switch Table

Switch Position	**Adapt Rate**
1	1023
2	512
3	256
4	128
5	64
6	32
7	18
8	9
9	6
10	4
11	2
12	1

8: PCAP II SPECIFICATIONS

Analog

Line Inputs (2)	• Two rear panel RCAs: left for mono operation, left and right for stereo operation, and for primary and reference for 2CH adaptive filter operation. • Z_{in} = 50kΩ, level adjustable -8 to +15 dBm.
Line Outputs (2)	• Two rear panel RCAs: left and right for stereo operation, single output for mono and 2CH adaptive filter operation. • Z_{out} = 100Ω, V_{out} = +8 dBm.
Headphone Output	• Panel stereo jack and volume control. Suitable for 8 ohm stereo headsets. • Monitor switches for selecting PCAP *Plus* inputs or outputs, as well as left / right / stereo channels.
Input Level Indicators	• Two tricolor LEDs. Red (-6 dB), orange (-12 dB), green (-18 dB).
Bandwidth	• Adjustable 3.2, 5.4, 6.5, 8.0, 11.0, and 16.0 kHz. • 35 Hz AC input coupling.
Analog Conversion	• 24-bit stereo A/D converter. 64X oversampling, sigma-delta technology. • 24-bit stereo D/A converter. 64X oversampling, delta-sigma technology.
Dynamic Range	• >90 dB.

Digital

Inputs / Outputs	• S/PDIF format RCA connectors • Output rate selectable between 44.1kHz and 48kHz, or AutoSYNC to digital input sample clock, if DIGITAL input

selected (if ANALOG input, output rate forced to 44.1kHz in AutoSYNC position)

- Accommodates any valid S/PDIF digital input signal over a sample rate range of 25-108kHz

Digital Processing

Digital Filters

- 6144-tap flexible adaptive/fixed filter module, dynamically allocable in 1024-tap sections to as many as 6 filter stages.
- Two 256-tap FIR filters for output equalizers.
- Four 256-tap FIR filters for input and output highpass filters.
- Four 128-tap FIR filters for interpolation / decimation.
- High-performance floating-point DSP processor that performs broadband noise reduction functions and other advanced algorithms

Limiter

- Microprocessor controlled
- Adjustable release time and threshold

AGC

- Digital Implementation
- Adjustable release time and gain range

Control Microprocessor

- TMS320C50 20 MIPS host processor with 32k x 16 program RAM, 64k x 8 data EPROM, 64k x 8 boot EPROM, and 512k x 8 flash memory.

Slave Microprocessors

- Two TMS320C50 20 MIPS processors with 32k x 16 program RAM and 32k x 16 data (delay) RAM
- One TMS320C6713 1.8 GFLOPS floating-point processor with 1280MB data/program/delay RAM and 1MB Flash memory

Computers Interface

- Standard RS232 serial interface for digital filter control, coefficient transfer, and spectral analysis data. 9.6k to 115.2k baud transfer rate, adjustable. (115.2k default)

Construction

Packaging
- 1.5" H x 10.0" W x 9.75" D, 3 lbs. Rugged aluminum enclosure with black powder-coat finish.

Power
- 10 - 16 VDC @ 2.0A, maximum.
- Universal AC adaptor supplied.

Panel Functions
- Front: phones jack, monitor select, and volume control; stand-alone mode select, filter clear, adapt/freeze, process/bypass, aux/off switches; two input level controls and tricolor level LEDs.
- Rear: Barrel +12VDC power connector and switch, RS232 interface connector, two analog input and two analog output RCA connectors, one digital input and one digital output RCA connector, digital output sample rate selector switch, analog / digital input select switch.

Host Computer
- Intel Pentium III 800MHz (or higher) desktop or laptop PC with mouse, SVGA monitor, 256 MB RAM, CD-ROM, 10 GB HD, and Windows XP recommended. Active matrix LCD display recommended if notebook used.

Signal Processing Functions

Filter Input Modes
- Mono, stereo, stereo independent

Adaptive Filters
- One-channel (1-Channel) adaptive predictive deconvolution and Ref Canceller (2CH) adaptive noise canceller.

Adjustable Filters
- Lowpass, highpass, bandpass, bandstop, notch, slot, comb, inverse comb.

Special Filters
- Hi-res graphic, spectral inverse filters (both 460 lines resolution and 60dB range).
- Limiter/Compressor/Expander
- Parametric Equalizer
- Noise Reducer frequency-domain spectral subtraction filter
- NoiseEQ™ frequency-domain spectral subtraction filter with user-defined frequency-selectivity

Output Equalizers	• 20-band graphic, spectral graphic (115 lines resolution), and parametric equalizers
Level Control	• Microprocessor-controlled input limiter and output AGC
Spectral Analysis	• Dual-channel FFT with exponential averaging • 460 line resolution • 70 dB display and 110 dB scalable dynamic range