PCAP II

PERSONAL COMPUTER AUDIO PROCESSOR

USER'S MANUAL

DIGITAL AUDIO CORPORATION

A DRI COMPANY

PCAP II

PERSONAL COMPUTER AUDIO PROCESSOR

User's Manual

November 2, 1999
Version 12.0

DIGITAL AUDIO CORPORATION
A DRI COMPANY

5121 Holly Ridge Drive
Raleigh, NC 27612
Phone: 919 782 6767
Fax: 919 782 6766

sales@dacaudio.com
www.dacaudio.com

TABLE OF CONTENTS

LIST OF FIGURES

ACKNOWLEDGEMENT

The Personal Computer Audio Processor, model PCAP II, is a second generation PC-based digital audio filtering system. Like its predecessor, the Personal Computer Digital Filter, this product was inspired by and significantly contributed to by Mr. James P. Foye of the United States Postal Service. Mr. Foye made numerous valuable technical and applications suggestions during the development of the PCAP. He also performed the critical tasks of beta testing. For those contributions Digital Audio Corporation is most grateful.

FOREWORD

The PCAP II Personal Computer Audio Processor is a new and powerful digital audio processor. The PCAP II offers the following advantages over other PC-based audio processors:

- Compact field-deployable packaging
- Direct digital audio input/output
- Both mono and stereo signal processing
- Up to nine stages of enhancement processing
- All Windows™-based control: versatile configuration and easy 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 PCAP II 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 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 nine sequential stages of processing (five of which are digital) can be performed simultaneously. With stereo signals, seven sequential stages of stereo processing (three of which are digital) for each channel are also available. The digital stages can implement a broad selection of filter types.

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 PCAP II's built-in spectrum analyzer.

1. SYSTEM BASICS

1.1 System Configuration

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

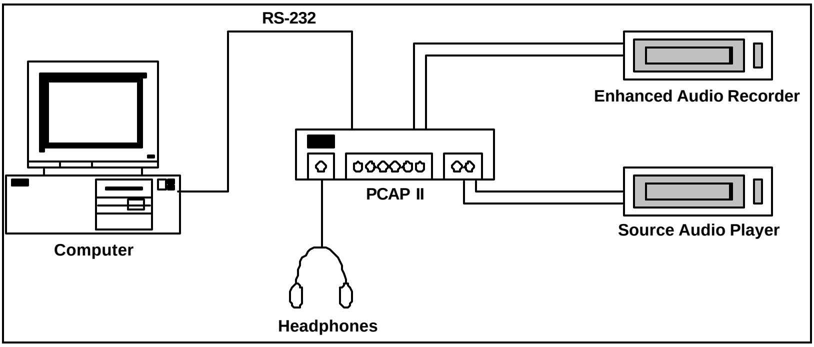


Figure 1-1 PCAP II System Configuration

The PCAP II Master Control program is written to be run on any Microsoft Windows 95/98/NT IBM PC-compatible computer. For best performance the following *minimum* system is recommended:

- Intel Pentium CPU processor (at least 166 MHz)
- 32 Megabytes of RAM
- 2.1 GB hard disk drive
- CD-ROM Drive
- 800x600 color SVGA display with 0.28 or better dot pitch
- Two-button mouse
- At least one spare RS232 port
- Laser or dot-matrix printer

Performance will improve with higher speed CPUs.

1.2 External Processor Capability

The PCAP II *EXTERNAL PROCESSOR* unit is a high-performance, self-contained digital signal processor and contains 27 DSP microprocessors, which are allocated as follows:

- Sixteen FIR filter processors which can be configured as 1, 2, 3, or 4 independent audio processors. These flexible processors may be combined into mono and stereo configurations. Any of the processors may be configured as an adaptive or adjustable digital filter.

- Two FIR fixed filter processors which are dedicated as output spectral equalizers.

- Two FIR fixed filter processors which perform interpolation and decimation filters.

- Four FIR fixed filter processors which perform input and output highpass filters.

- 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.

Analog-to-digital and digital-to-analog conversion is performed by stereo, 16-bit, sigma-delta converters which perform 64x oversampling.

The base sample rate is adjustable from 7.2 kHz (3.2 kHz bandwidth) to 36 kHz (16 kHz bandwidth). All 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 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 Master Control Program. This allows the external processor to be operated in any of four previously-stored setups without being connected to a PC.

For a simplified functional block diagram, see Figure 1-2.

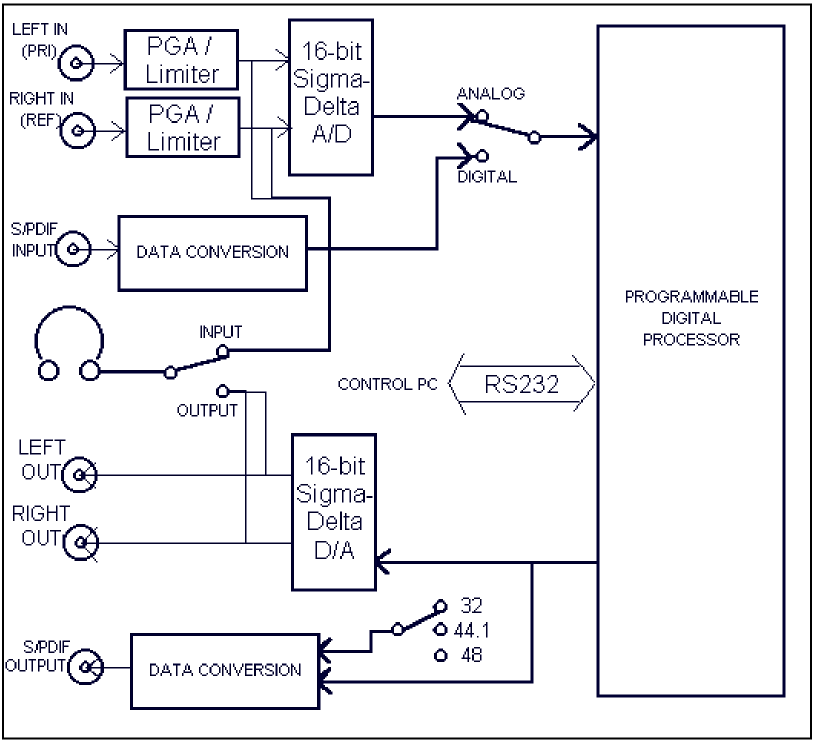


Figure 1-2 PCAP II Functional Block Diagram

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 4.8.4 and 5.0.

The front panel of the PCAP II external processor appears as follows (Figure 1-3):

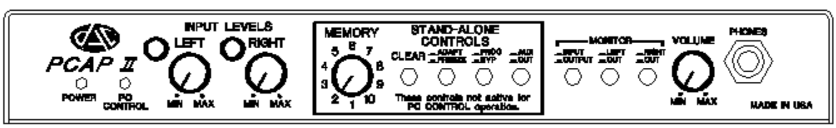


Figure 1-3 PCAP II Front Panel

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 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 Master Control software.

1.4 External Processor Rear Panel

The rear panel of the PCAP II external processor appears as follows (Figure 1-4):

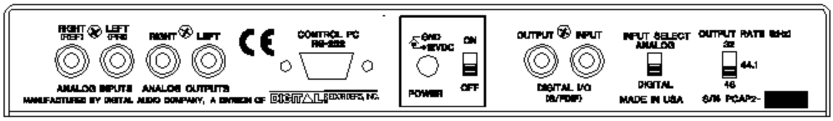


Figure 1-4 PCAP II Rear Panel

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 external processor to a computer using the supplied cable. In this new version of the PCAP, the unit is automatically put into PC CONTROL mode when characters are received over the RS232 connector using the PCAP II Master Control software. 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 inputs and outputs are provided on the rear panel via RCA jacks. As a new feature, the PCAP II 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. Both the ANALOG and DIGITAL outputs are always active. The output rate of the digital output can be selected using the OUTPUT RATE switch.

2. INSTALLATION INSTRUCTIONS

2.1 Cautions to User

To install the PCAP II 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 <u>must</u> know which RS232 COM ports are COM1, COM2, COM3, etc. Usually, a serial mouse will be installed on COM1, leaving COM2 available for connecting the PCAP II external processor. If COM2 is not available in the user's PC, it will be necessary to reconfigure the PCAP II software COM port selection by performing the procedure in Section 6.1 <u>after</u> completing the installation procedure in Section 2.2.

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

For advanced users who need to configure the PCAP II to operate at RS232 symbol rates slower than 38400 baud (factory default), please see Section 6.2. **Note: Altering the baud rate setting is <u>not</u> <u>recommended.</u>**

2.2 Installation Procedure

1. Carefully remove the PCAP II 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 rear panel POWER connector. Keep the POWER switch OFF for now.

3. With the PCAP II 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 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 and the computer.

6. Once Windows has booted up (Windows 95/98 or NT), insert the PCAP II 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 start the installation program.

   If the machine you wish to install the PCAP II Master Control Software does not have a CD-ROM then, on a machine that does have a CD-ROM, insert the CD and run Windows Explorer. *Note: The installation software will most likely run automatically. If it does exit it by clicking the "Cancel" button.* You can run Windows Explorer by clicking on the **Start** menu and then on **Run** and typing "explorer" or by clicking **Start->Programs->Windows Explorer**. Once Windows Explorer has opened, click on the CD-ROM drive that contains the PCAP II Master Control Software to view the contents of it. You will see a folder labeled "Installation Diskettes". Open this folder and you will see several other folders and one file labeled "MakeDisks.bat". Double-click on this file and a utility will run that will guide you through making installation diskettes. Make sure that the disks you use are formatted and do not contain any other files.

   To start the installation from a diskette, insert the first diskette into the floppy drive . Click on the **Start** Menu and then on **Run**. Type:

   A:SETUP<enter>

   in the Command Line text box.

7. The PCAP II Setup Utility will now install the PCAP II 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:

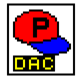


Figure 2-1 PCAP II Master Control Icon

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

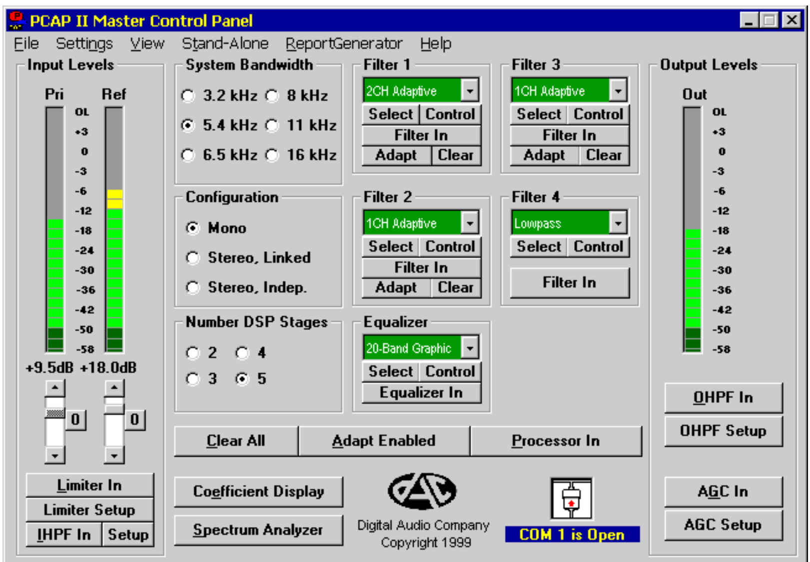


Figure 2-2 Master Control Panel

If an error message is displayed, it is possible that the software is not configured for the correct COM port (software defaults to COM2). If you know to which COM port the PCAP II external processor is connected, configure the PCAP II Master Control program for the correct COM port by following the procedure in Section 6.1.

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

3. GETTING STARTED

Operation of the PCAP II 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 system. For a lesson in operating the PCAP II controls, it is recommended that the user also complete the PCAP II Tutorial in Section 3.2. Section 3.3 applies the PCAP II to different noise problems contained on the PCAP II Training CD.

In the following sections the Source Audio Player and an Enhanced Audio Recorder will be referred to. The Source Audio Player can be any type of playback unit such as a CD, Mini-Disk, Micro-cassette, or Digital Audio Tape (DAT) playback unit. The Enhanced Audio Recorder can be any type of audio recorder such as a CD, Mini-disk, Micro-cassette, or DAT recording device.

Always consider that the quality of the playback and recording media will limit the quality of the processed and enhanced audio. The recommended device is a DAT recorder with digital inputs because the PCAP can transfer the audio digitally. DAT recorders have excellent bandwidth and signal-to-noise (SNR) ratio.

3.1 Fast Start

Fast start the PCAP II as follows:

1. Connect the LEFT and RIGHT channel line-level audio outputs (AUDIO OUT jacks) of your Source Audio Player to the LEFT IN and RIGHT IN RCA jacks on the PCAP II external processor rear panel as shown in Figure 3-1. Note that the RIGHT IN signal is only used in stereo configuration and with the 2CH Adaptive Filter in mono configuration.

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 OUT and RIGHT OUT RCA jacks on the PCAP II external processor rear panel as shown in Figure 3-1.

3. Connect your stereo headphones to the PHONES jack on the PCAP II external processor front panel as shown in Figure 3-1. 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.

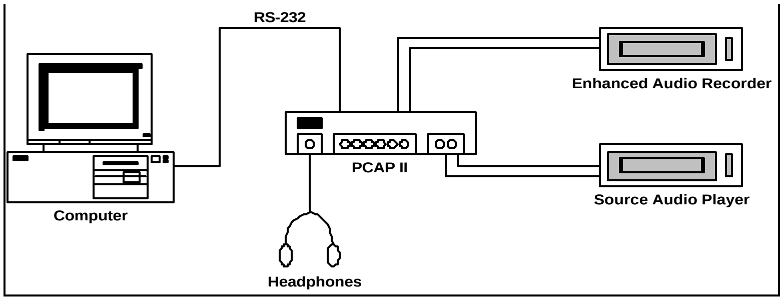


Figure 3-1 PCAP II Connection Diagram

4. With the PCAP II installation procedure in Section 2.2 complete, run the PCAP II 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 Master Control Panel displayed on your PC screen, insert the PCAP II Training CD into the Source Audio Player, then press the PLAY button.

6. Adjust the LEFT and RIGHT INPUT LEVEL controls on the PCAP II external processor front panel until the Input Levels bargraphs on the PCAP II Master Control Panel indicate GREEN with occasional peaks in the YELLOW range. Note that some mono segments of the audio may not have audio recorded on them.

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

   NOTE: If you wish to listen to the unprocessed audio going into the PCAP II, toggle the MONITOR switch on the PCAP II external processor to INPUT. To listen to the PCAP II output, toggle the MONITOR switch to OUTPUT.

If you wish at this point to discontinue the Fast Start procedure and experiment with the PCAP II 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 Master Control Panel menu bar, click on **File**. This will cause the following pulldown menu to appear

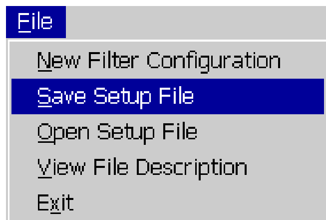


Figure 3-2 Fast Start File Pulldown Menu

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

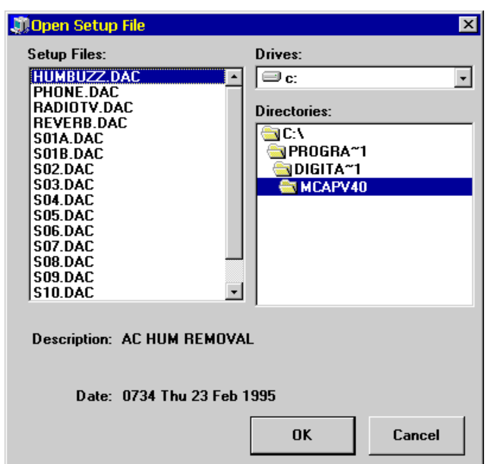


Figure 3-3 Fast Start Open Setup File Window

10. The **Setup Files** box should contain a list of all PCAP II 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:

PHONE.DAC	-	Telephone audio enhancement
BODYMIKE.DAC	-	Body microphone/recorder audio enhancement
HUMBUZZ.DAC	-	Powerline hum and buzz removal
REVERB.DAC	-	Cancellation of room echoes and reverberations
RADIOTV.DAC	-	Cancellation of radio and/or TV audio from live or recorded audio using a reference

These basic setups should be able to enhance most voice 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. The following message will now appear to alert you that 60 seconds may be required to open your setup file (Figure 3-4):

Figure 3-4 Fast Start Recall Setup Alert Message

Click on **OK** to commence the opening (recall) of the setup file. An "hourglass" mouse cursor will now appear, indicating that the PCAP II is busy recalling a setup file.

12. When the mouse cursor returns to normal, the setup file will be complete. Toggle the MONITOR switch on the PCAP II external processor between INPUT and OUTPUT to hear the difference between the unprocessed input signal and the processed output signal.

Feel free to experiment with the PCAP II Master Control Panel settings - you may open the original setup at any time by repeating steps 8-12. If you wish to try loading any of the other basic setup files, repeat steps 8-12.

Once you become comfortable recalling the basic setup files, it is strongly recommended that you complete the PCAP II 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 's controls. It should require about 1 hour to complete, yet it demonstrates the basic functionality.

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

Tutorial Steps:

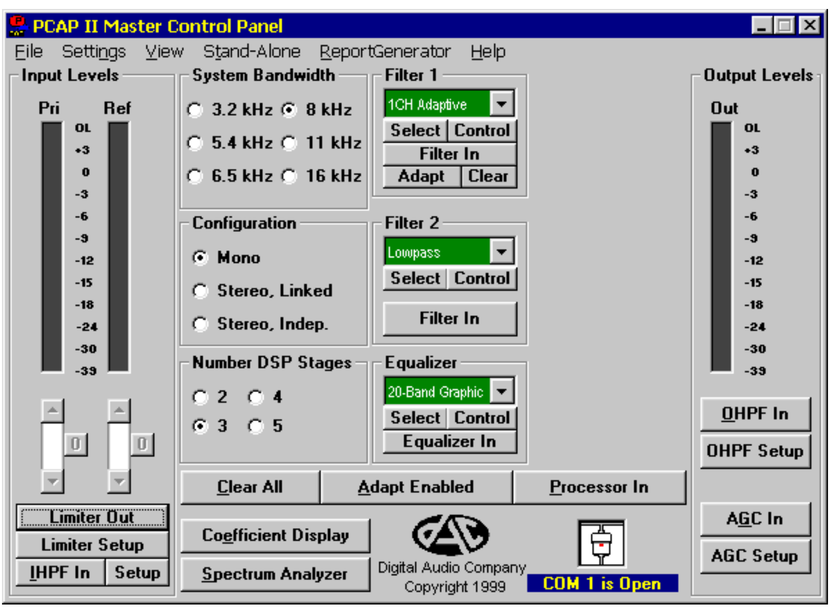


Figure 3-5 Tutorial Master Control Panel

1. With the installation procedure in Section 2.2 completed, run the PCAP II Master Control program by double clicking the PCAP II Master Control icon in Microsoft Windows. A window similar to Figure 3-5 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 In/Out, Limiter In/Out, Output HPF In/Out, AGC In/Out, System Bandwidth, Configuration, Number DSP Stages, Adapt Enabled/Disabled,** and **Processor In/Out** buttons as they are shown in Figure 3-5. 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 external processor; make sure that the INPUT SELECT switch is set to ANALOG. Insert the PCAP II Training Tape and rewind to the beginning.

4. Connect a pair of stereo headphones to the PHONES jack on the PCAP II 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 external processor so that the peak audio level, as measured by the bargraphs on the Master Control Panel, is approximately -6 dB.

---

*Additionally, the control buttons may be selected with the <Tab> key highlighting that button. Once selected, the button may be toggled with the <Space> or <Enter> keys. **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 In/Out** button.

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 **Filter** button in both the Filter 1 block and the Filter 2 block until both buttons indicate **Filter Out** (bypassed).

8. Click on the **Equalizer** button in the Equalizer block until the button indicates **Equalizer Out** (bypassed).

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

10. Switch the Input HPF in and out by clicking on the **Input HPF** button in the Input Levels block. You should hear the low frequency effects of this control. Restore the button to the **Input HPF Out** indication.

11. Switch the Output HPF in and out by clicking on the **Output HPF** button in the Output Levels block. You should hear the low frequency effects of this control. Restore the button to the **Output HPF Out** indication.

12. Click on the **Limiter Setup** button in the Input Levels block. The following window (Figure 3-6) will appear:

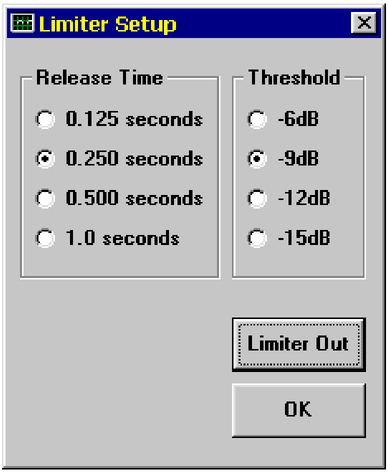


Figure 3-6 Tutorial Limiter Setup Window

13. Use the mouse to set the Release Time to 0.250 seconds and the Threshold to -9dB, as shown in Figure 3-6. 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 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 **Limiter** button in the Input Levels block. You should notice the indicated bargraph levels decrease to -9dB and the audio quality (as heard through the headphones) dramatically

---

*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.

improve whenever the button indicates **Limiter In**. The input limiter is electronically lowering the INPUT LEVELS to avoid overload. Restore the button to the **Limiter Out** position 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 3-7) will appear:

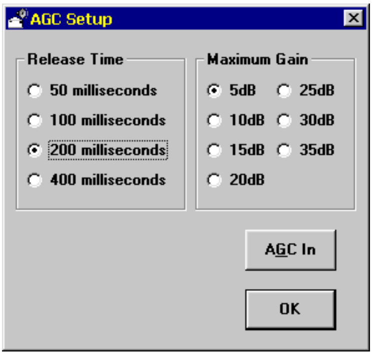


Figure 3-7 Tutorial AGC Setup Window

Use the mouse to set the Release Time to 200 milliseconds and the Maximum Gain to 20dB, as shown in Figure 3-7. 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 **AGC** button in the Output Levels 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 **AGC In**. The AGC attempts to make the Output Level constant and is useful in near party/far party situations. Restore the button to the **AGC Out** indication, and restore the INPUT LEVELS controls to normal level.

20. Click on the **Select** button in the Filter 1 block. A window similar to the one in Figure 3-8 will appear. Use the mouse to click on **1CH Adaptive**, then **OK***. The Filter 1 block should now indicate that the selected mode is 1CH Adaptive.

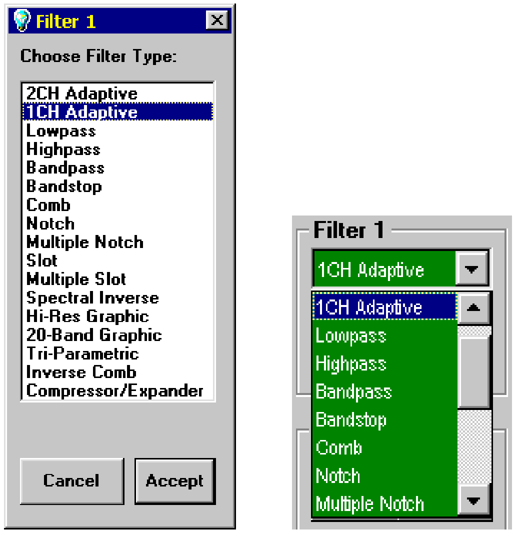


Figure 3-8 Tutorial Filter Selection Window

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 3-9 will appear:

---

*Double-clicking the desired filter mode will avoid having to click the **OK** button.

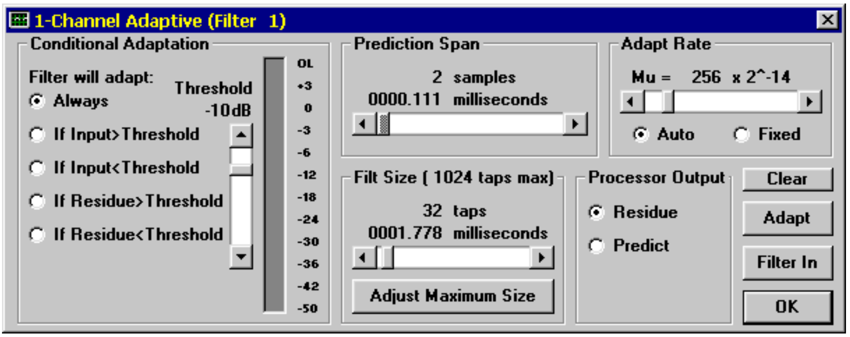


Figure 3-9 Tutorial 1CH Adaptive Filter Control Window

This is the 1CH Adaptive Filter Control Window. It is used to select Filter Size, Adapt Rate, Prediction Span, and Adapt Mode. 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 3-9*. 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** button is toggled between **Filter In** and **Filter Out**. You should notice a reduction in background noise whenever the button indicates **Filter In**. Press the **Clear** button to cause Filter 1 to readapt to the input signal.

25. Restore the **Filter** button to the **Filter Out** indication and click on **OK** to exit the 1CH Adaptive Filter Control Window.

26. Click on the **Control** button in the Filter 2 block. The window in Figure 3-10 will appear:

---

*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.

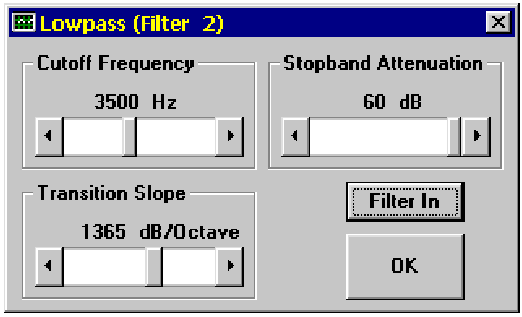


Figure 3-10 Tutorial Lowpass Filter Control Window

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 3-10. 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** button is toggled between **Filter In** and **Filter Out**. You should notice a dramatic reduction in high-frequency sound whenever the button indicates **Filter 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 **Filter** button in the Filter 2 block until it indicates **Filter Out**.

31. Click on the **Equalizer** button in the Equalizer block until it indicates **Equalizer In**.

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

Use the mouse to click (try double clicking to avoid clicking on **OK**) on **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 3-12 will appear.

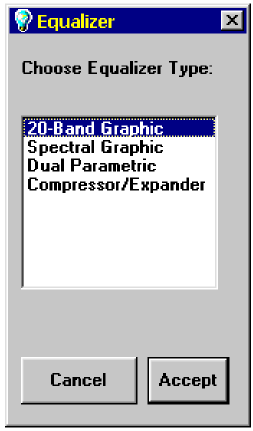


Figure 3-11 Tutorial Equalizer Selection Window

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.

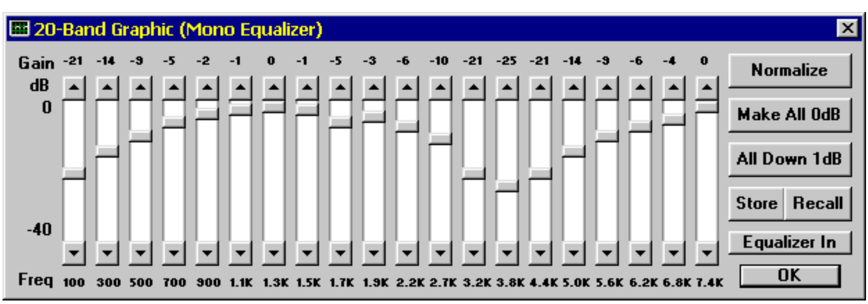


Figure 3-12 Tutorial 20-Band Graphic Equalizer Control Window

34. Use the mouse to set all sliders to match the settings shown in Figure 3-12.

35. Click on the **Store** button to bring up the Store 20-Band Graphic Equalizer window as in Figure 3-13.

36. Click on the **1** button to store your slider settings in Memory 1*. (We will recall these settings later.) Once you have clicked this button, the system will automatically return to the 20-Band Graphic Equalizer control window.

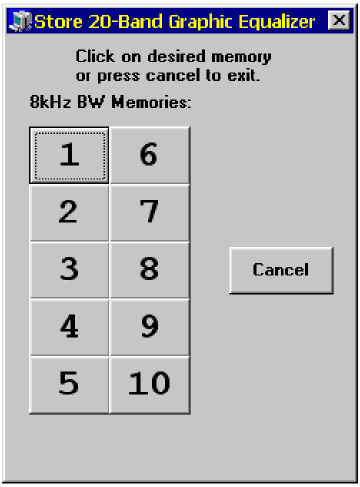


Figure 3-13 Tutorial Store 20-Band Graphic Equalizer Window

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 **Equalizer** button is toggled between **Equalizer In** and **Equalizer Out**. Also, note that whenever a slider is adjusted, the **Equalizer** button will automatically be switched from **Equalizer In**, if it was in **Equalizer Out**.

38. Click on the **All Down 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

---

*Each of the six system bandwidths has its own set of 10 memories.

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 **Make All 0dB** 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 3-14 will appear.

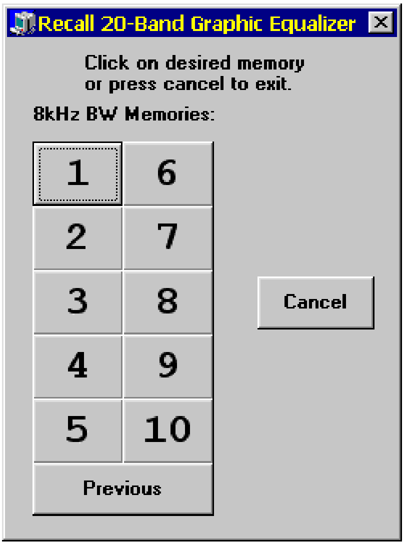


Figure 3-14 Tutorial Recall 20-Band Graphic Equalizer Window

41. Click on the **1** button to recall the slider settings previously stored in Memory 1. (The **Previous** button will recall the last settings, in case you intended to, but forgot to save them.) Once you have clicked this button, the system will automatically return to the 20-Band Graphic Equalizer control window and the original slider settings will be restored from Memory 1.

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 3-15) 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 3-15. Make sure that the **Primary In** selection is highlighted in the Yellow Trace signal select box, that the **Line Output** selection is highlighted in the Blue Trace signal select box, and that the Displayed Trace is set to **Both**.

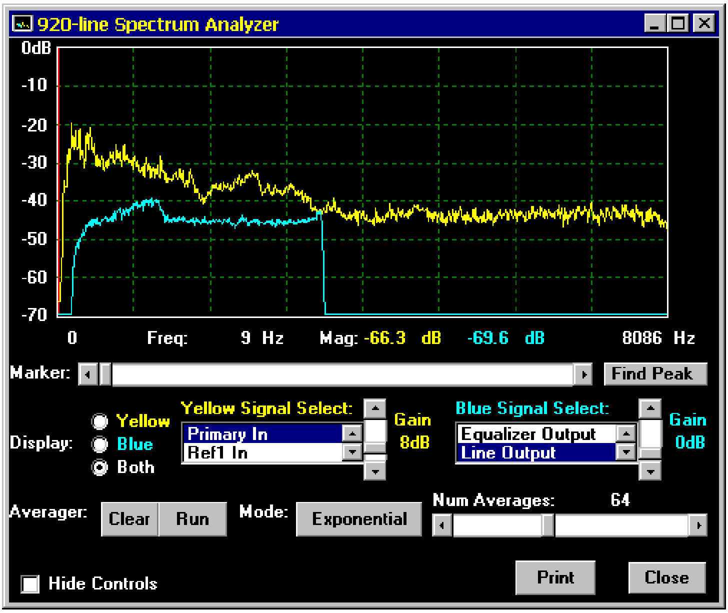


Figure 3-15 Tutorial Spectrum Analyzer Window

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

45. Click on the Run/Freeze button in the Averager block until **Freeze** is indicated. You should see the spectrum waveform stop updating.* Click on the **Clear** button in the Averager block; both spectrum waveforms should now be cleared

---

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

completely away. Restore the Run/Freeze button to the **Run** indication 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 (Freq:) and magnitude (Mag:) of the Yellow and Blue traces will be indicated below the display grid. Use the horizontal scroll bar in the Marker block for more precise marker control.

47. Click the **Find Peak** button to automatically move the marker to the largest magnitude displayed. Adjust the **Gain** controls for both the Yellow and Blue traces 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*** (display overload) for the Yellow Trace and how the magnitude (Mag:) indication for the Yellow Trace 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 **Control** button to bring up the Lowpass filter control window.

51. Notice how the Lowpass and Spectrum Analyzer windows are displayed on the same screen. Click the **Filter** button in the Lowpass window until it indicates **Filter In**. The Yellow trace now displays the signal going into the Lowpass filter, while the Blue trace displays the signal coming out of the

---

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

Lowpass filter. Adjust any of the Lowpass filter controls, noting the effect on the Blue trace* .

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

53. To return to the Spectrum Analyzer display, click on **View** in the menu bar to bring up the **Display Select** window as shown in Figure 3-16.

54. Note that this display allows the user to access either the Coefficient Display or the Spectrum Analyzer display, or to turn all Displays Off. Click on **Spectrum Analyzer** to return to the Spectrum Analyzer window. This two-step sequence is the same as clicking on the **Spectrum Analyzer** Button in the Master Control Panel (step 43).

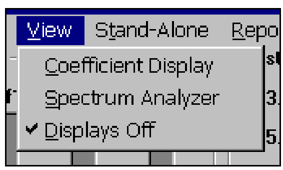


Figure 3-16 Tutorial Display Select Window

55. 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.

56. Access the Save Setup File feature by clicking on **File** in the menu bar. The pulldown menu in Figure 3-17 will appear.

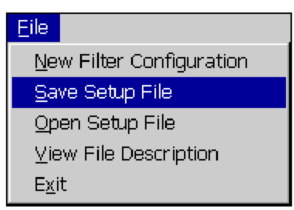


Figure 3-17 Tutorial File Pulldown Menu

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

57. Click on **Save Setup File** to bring up the Save Setup File window as shown in Figure 3-18.

For now, do not change the **Drives** or **Directories** settings; just keep in mind that you could change these settings to allow setup files to be stored to any directory of any drive.

Figure 3-18 Tutorial Save Setup File Window

58. Click your mouse cursor on the **Description** text box, which should currently contain the word **Unnamed**. Using your keyboard, delete all characters in the text box, then type in **My First Setup**. See Figure 3-19.

59. Now click your mouse cursor on the **Store in 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 3-19.

---

however.

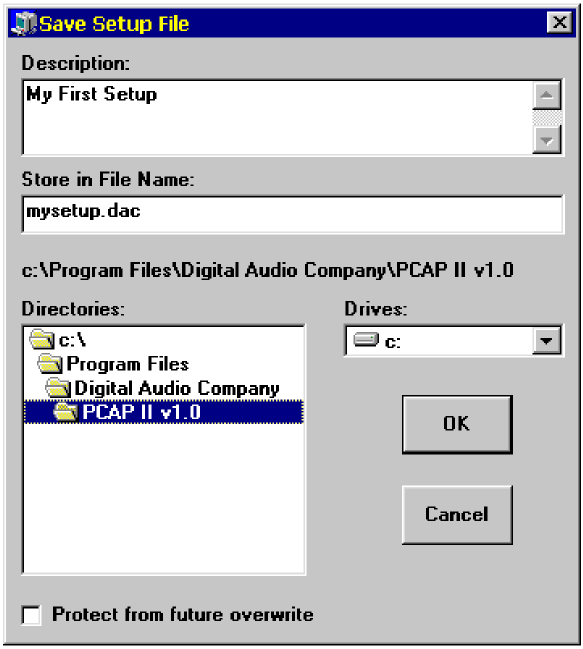


Figure 3-19 Tutorial Store in File Name Text Box

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

61. 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 **Setup Files** list.

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

62. Click on each file listed in the **Setup Files** list box. Notice how the **Description** and **Date** for each appears at the bottom of the window as each is clicked.

63. 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 66.

64. Access the hardcopy report generator by clicking on **Report Generator** in the menu bar. The pulldown menu shown in Figure 3-20 will appear.

65. 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 now be printed.

66. Open your original filter settings by repeating steps 61 and 62 to bring up the **Open Setup File** window. Next, click on **mysetup.dac** in the **Setup Files** list box, then click on **OK** to return to the Master Control Panel.

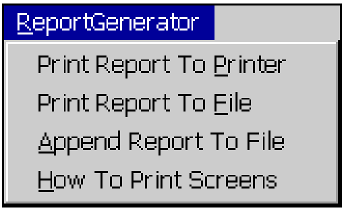


Figure 3-20 Tutorial Report Generator Pulldown Menu

This completes the PCAP II 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 PCAP II 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 4.0 for more detailed instructions on using the windows in the PCAP II Master Control program.

3.3 Training CD Recordings

The PCAP II Training CD supplied with the PCAP II is a useful training tool. Each of its 13 noise sample recordings is accompanied by a suggested Setup File, installed on your comptuer along with the other PCAP II software by the installation procedure.

Table 3.1 summarizes the contents of the training CD. The SETUP column contains the name of the setup file installed on the disk as part of the PCAP II 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, the **Drive** or **Directories** may require changing. Refer to Section 4.8.2 for additional details.

According to Table 3.1, the setup file s03.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 in the for *sXX.dac*, where *XX* is the track number.

All recordings except track 13 are monophonic, *i.e.*, the left and right channels have the same audio. The 13th recording is a 2CH adaptive (Ref Canceller) filtering example and has the microphone audio on the left channel and the TV reference audio on the right channel.

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

TK	TIME	NOISE DESCRIPTION	SETUP	COMMENTS
1	00:00	500 Hz tone	s01a.dac s01b.dac	Adjust **Notch Freq** or recorder **Pitch** to null tone Single tone requires small filter size
2	02:30	400 Hz and 1100 Hz tones	s02a.dac s02b.dac	Adjust recorder **Pitch** to null tones Compare adaptive and notch filters
3	05:30	Swept tones	s03.dac	Adjust **Prediction Span** and **Adapt Rate** comparing effects
4	06:20	High frequency hiss	s04.dac	Adjust LPF cutoff and attenuation
5	08:50	Radio interference	s05.dac	
6	09:40	A.C. hum	s06.dac	Hi Res Graphic restores high frequencies. Adjust **Comb Frequency** or recorder pitch to minimize 60 Hz hum
7	11:00	Telephone A.C. hum	s07.dac	Adjust **Comb Frequency** or recorder **Pitch** to minimize 60 Hz hum
8	13:55	Acoustic resonances	s08.dac	
9	16:30	Jail Cell	s09.dac	Hi Res Graphic restores high frequencies
10	19:20	Body transmitter	s10.dac	
11	20:45	Bar music	s11.dac	Filter cannot separate voices (spoken or sung) but does reduce muffling
12	23:25	Interview and argument	s12.dac	
13	25:50	Radio/TV	s13.dac	2CH adaptive (Ref Canceller) filtering using a reference. Cancellation limited by analog tape recorder's wow/flutter

Tabel 3.1 PCAP II Training CD Summary

4. PCAP II 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 on specific windows in the PCAP II Master Control program. It is assumed in this section that the user has a good working knowledge of Microsoft Windows, has properly installed the PCAP II hardware and software using the installation instructions in Section 2.2, and has completed the PCAP II Tutorial Section in 3.2.

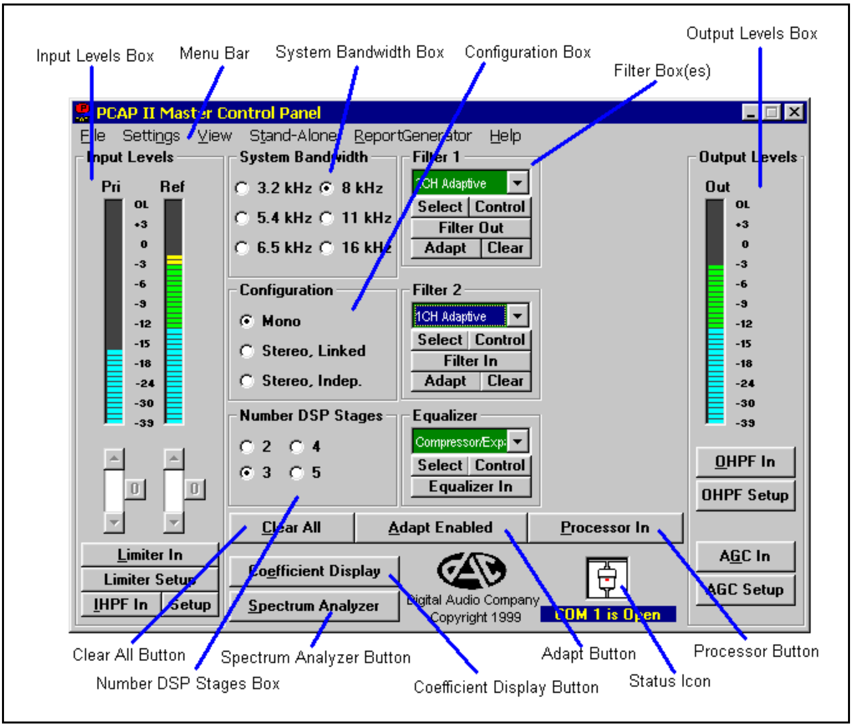


Figure 4-1 PCAP II Master Control Panel

4.1 Master Control Panel

When the PCAP II 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 external processor 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:

Menu Bar: Used to access the Master Control pulldown menus which allow saving and opening of filter setups from disk files, configuring RS232 communication with the external processor, viewing signal spectra and filter coefficients, storage of filter setups in nonvolatile memories inside the external processor for Stand-Alone operation, generation of hardcopy setup reports, and getting online help. See Section 4.8 Master Control Pulldown Windows for further details on these features.

Input Levels: Used to view input signal levels via bargraph display, adjust input levels via slider controls if desired, setup Input HPFs, and switch them In and Out of the process, and setup input Limiters and switch them In and Out of the process.

System Bandwidth Block*: Used to select the bandwidth of the audio processor which most closely matches that of the input signals (physically adjusts the sampling rate of the PCAP II). For speech processing applications, settings of 3.2kHz to 6.5kHz work best. For wider bandwidth signals applications, 8.0kHz to 16kHz is recommended.

---

* Changing any of these settings will automatically clear all 2CH Adaptive, 1CH Adaptive, Multiple Notch, Multiple Slot, Spectral Inverse, and Hi-Res Graphic Filters.

Configuration Block**:	Selects mono or stereo filter configurations. **Mono** configuration allows the user to enhance a single input signal with as many as five successive stages of digital signal processing. **Stereo, Linked** configuration allows the user to process two input signals simultaneously with as many as three stages of digital signal processing; each stage is identical for both the left and right channels. **Stereo, Indep.** configuration is the same as **Stereo, Linked**, except that the processing stages may be set up differently for the left and right channels.
Number DSP	Selects the number of DSP (Digital Signal Processing) processing Stages Block*:stages that are to be used. DSP stages consist of both Filter stages and Equalizer stages. At least two stages are always available (1 DSP Filter + 1 DSP Equalizer), but depending on System Bandwidth and Configuration, as many as five stages can be available.
Filter Block(s):	Used to control each DSP Filter Stage. In Mono and Stereo, Indep. configurations, as many as four Filter blocks will be displayed; in Stereo, Linked configuration, either one or two filter blocks will be displayed. Select the type of filter to be implemented by each stage by using the **Select** button to pull up the filter selection window (See Section 4.3.1 Filter Selection Window.) Set up the selected filter's parameters by using the **Control** button to pull up the control screen for that filter (See Section 4.4 DSP Filter Control Windows). Switch each filter stage In or Out of the process using the **Filter In/Out** button. Reset and freeze each adaptive filter stage without affecting the other stages using the **Clear** and **Adapt/Freeze** buttons.

**Ibid.

*Ibid.

Equalizer Block(s): Used to control each equalizer stage. In Mono and Stereo, Linked configurations, one equalizer block will be displayed. In Stereo, Indep. configuration, a separate equalizer block will be displayed for each channel. Select the type of equalizer to be implemented by using the **Select** button to pull up the equalizer selection window (See Section 4.3.2 Equalizer Selection Window.) Set up the selected equalizer's parameters by using the **Control** button to pull up the control screen for that equalizer (See Section 4.5 DSP Equalizer Control Windows.) Switch each equalizer stage In or Out of the process using the **Equalizer In/Out** button.

Clear All Button: Clears the filter coefficients 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 the **Clear** button in its Filter Block.

Adapt Button: Toggle control used either to freeze <u>all</u> adaptive filters (**Adapt Disable** displayed) or to allow non-frozen filters (as specified by the **Adapt/Freeze** button in each Filter block) to adapt.

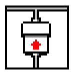
Processor Button: Toggle control used to specify whether the original input signal(s) is (are) routed to the line output(s) (**Processor Out** displayed) or whether the product(s) of the digital processor is (are) routed to the line output(s) (**Processor In** displayed).

Coefficient Button: Allows user to quickly activate built-in coefficient display. (See Display Button: Section 4.7 Coefficient Display Window)

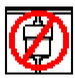
Spectrum Analyzer Button: Allows user to quickly activate the built-in spectrum analyzer display. (See Section 4.6.3 Spectrum Analyzer Window.)

Output Levels Block: Used to view output signal levels via bargraph display, switch Output HPFs In and Out of the process, switch output automatic gain controls (AGCs) In and Out of the process, and to setup AGC parameters. See Section 4.2.1 for further details on the Output HPFs and AGCs

Status Box: Used to indicate RS232 COM port. The messages that can be displayed by the Status Box are:



Selected COM port located and PCAP external processor communication established.



PC is unable to establish a link with the PCAP on the selected COM port at the selected Baud Rate

4.2 Input and Output Processors

4.2.1 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 **Input HPF** Button will cause its caption to toggle between **Input HPF In** and **Input HPF Out** as shown in Figure 4-2, indicating whether the Input HPFs are in or out of the process. Similarly, clicking on the **Output HPF** Button will cause its caption to toggle between **Output HPF In** and **Output HPF Out** as shown in Figure 4-2, indicating whether the Output HPFs are in or out of the process.

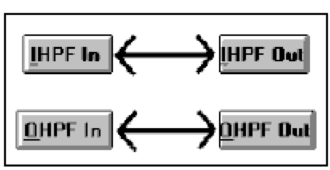


Figure 4-2 Input and Output HPF Buttons

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 IHPF **Setup** button. When this button is pressed, the window shown in Figure 4-3 appears.

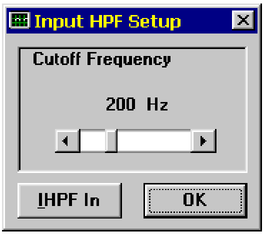


Figure 4-3 Input HPF Setup Window

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 **Setup** button to bring up the Output HPF Setup Window (identical to the Input HPF Setup Window shown in Figure 4-3).

NOTE: Both the Input HPF and Output HPF buttons are overridden by the **Processor** button. When the **Processor** button is set to **Processor Out**, both the Input HPF and the Output HPF will be bypassed, regardless of their buttons' status.

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

4.2.2 Digitally-Controlled Limiter

Application:

> *The four-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 configurations, the four 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.*

Clicking on the **Limiter** Button will cause its caption to toggle between **Limiter In** and **Limiter Out** as shown in Figure 4-4, indicating whether the Limiter is in or out of the process.

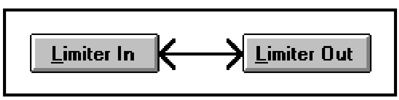


Figure 4-4 Limiter Button

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** Button is never overridden by the **Processor** In/Out button.

The **Limiter Setup** button is used to adjust the control settings for the Limiter. When this button is pressed, the window shown in Figure 4-5 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.

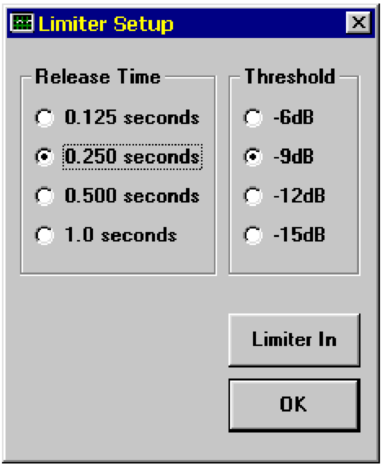


Figure 4-5 Limiter Setup Window

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 configurations 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; however, Independent Limiter gains should be used when processing two unrelated monophonic signals in a Stereo configuration.

4.2.3 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 **AGC** Button will cause its caption to toggle between **AGC In** and **AGC Out** as shown Figure 4-6, indicating whether the AGC is in or out of the process.

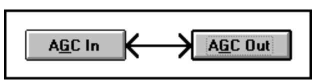


Figure 4-6 AGC Button

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** Button is overridden by the **Processor** button. When the **Processor** button is set to **Processor Out**, the AGC will be bypassed, regardless of its button status.

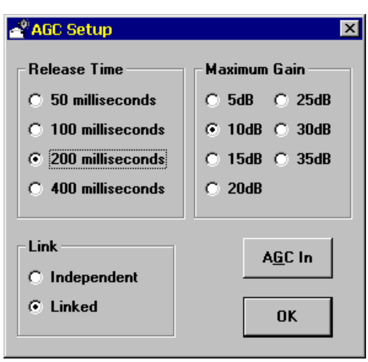


Figure 4-7 AGC Setup Window

When the **AGC Setup** button is pressed. The window in Figure 4-7 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,.

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 configurations to give the user the option of making the two output channels Linked or Independent (In Mono configuration, outputs are always Linked). 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; however, Independent AGC gains should be used when processing two unrelated monophonic signals in a Stereo configuration.

4.3 DSP Processor Selection

4.3.1 Filter Selection Window

Clicking the **Select** button in any Filter block (not an Equalizer block) causes the following window (Figure 4-8) to appear.

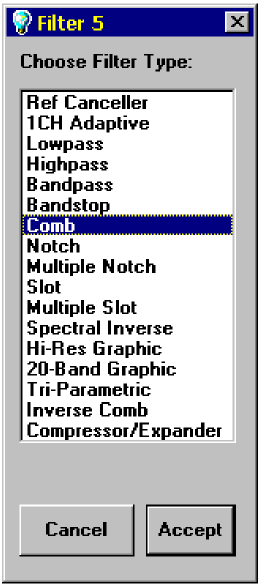


Figure 4-8 Filter Selection Window

Click the mouse on the desired filter type then click on **OK** (or, simply double-click the desired filter type) to select it for the current Filter block. Another way of selecting the filter is by using the filter select combo box shown in Figure 4-9. To select the filter, first click on the small arrow to the right of the combo box and then, using the drop down menu that appears, click on the desired filter.

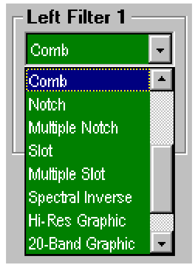


Figure 4-9 Filter Select Combo Box

Clicking the **Control** button in any Filter block 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 Equalizer Selection Window

The last DSP stage is always an Equalizer stage. Clicking the **Select** button for any Equalizer block (not a Filter block) causes the following window (Figure 4-10) to appear.

Click the mouse on the desired equalizer type then click on **OK** (or, simply double-click the desired equalizer type) to select it for the current Equalizer block. Alternately the equalizer select combo box can be used in the same manner as with the Filter blocks.

Clicking the **Control** button for any Equalizer block causes the control window for the selected equalizer mode to appear. See Section 4.5 for detailed information on control windows for each equalizer mode.

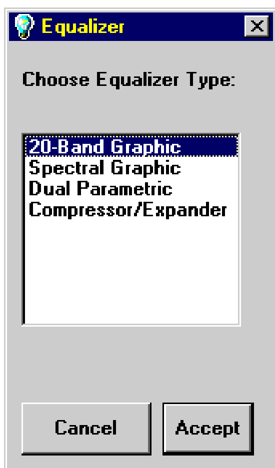


Figure 4-10 Equalizer Selection Window

4.4 DSP Filter Control Windows

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

4.4.1 2CH Adaptive Filter

Application:

The 2CH Adaptive filter is used to automatically cancel from the Primary (Left) input any audio which matches the Reference (Right) 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.

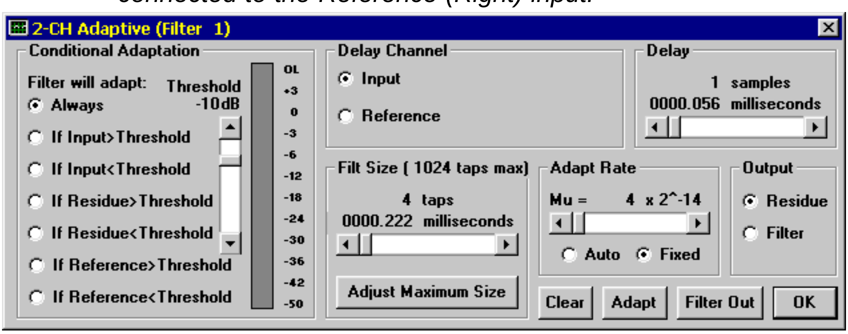


Figure 4-11 2CH Adaptive Filter Control Window

Description of controls (Figure 4-11) is as follows:

Conditional Adaptation: 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 filters operation. First allow the filter to converge in **Always** and the click on **If Output < Threshold**. Adjust the threshold for adaptation (LED is red) by observing the bargraph levels when the voice is not present.

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

The Master Control Panel will display an Adapting indicator for each filter block whenever **Filter will adapt** is set to anything other than **Always**. 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 32 taps, but can be set to as high as 4096 taps depending on System Bandwidth, Configuration, and Number DSP Stages settings. *Normally, the maximum filter size possible is used in the 2CH 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 x 2^-14 provides very slow adaptation, while a Mu of 256 x 2^-14 provides fastest adaptation.* *As a rule set this rate to maximum initially, to establish convergence, then back off to a mid value to maintain cancellation.*

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.

Adapt Mode: Selects Auto(matic) or Fixed adaptation rate. *Auto is recommended*. When Fixed is selected, the specified Adapt Rate Mu is applied to the filter at all times. When Auto is selected, the specified Adapt Rate is continuously power normalized depending upon the input signal level. *The Auto mode generally results in faster convergence for a given Mu*.

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

---

*2^-14 is an alternate means of expressing $2^{-14} = 0.000061$; 256 x 2^-14 is, therefore, equal to 0.016.

	the modified Reference signal being subtracted from the Primary signal.
Filter Button:	Used to switch the 2CH Adaptive filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.
Adapt/Freeze Button:	Used to freeze this adaptive filter independently.
Clear Button:	Used to reset the coefficients of the 2CH Adaptive filter to zero without affecting any other adaptive filters in the process.

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

4.4.2 1CH Adaptive Filter

Application:

> *The 1CH 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.*

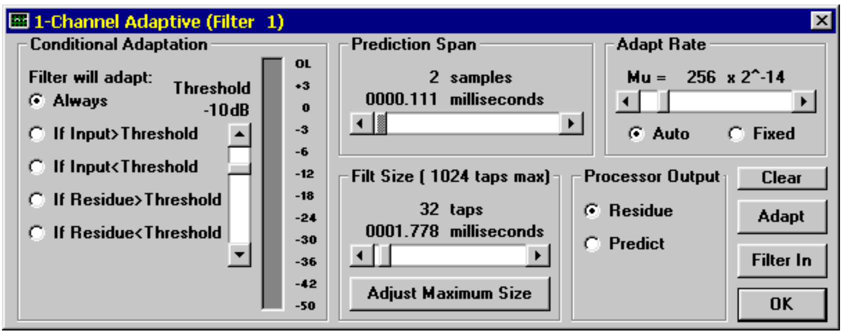


Figure 4-12 1CH Adaptive Filter Control Window

Description of controls (Figure 4-12) is as follows:

Conditional Adaptation: For advanced users only. *Novice users should keep* ***Filter will adapt*** *set to* ***Always***. The **Threshold** setting is meaningless in this case.

Conditional Adaptation allows the adaptive filter to automatically Adapt/Freeze based upon bargraph levels as displayed on the Master Control Panel. This can be very useful in situations where there are pauses, or breaks, in the speech being processed. For example, if you only desire the filter to adapt when no one is talking, you will first need to observe the **Input** bargraph levels to determine the dB level below which no talking occurs. Then, click on **If Input<Threshold** and set the **Threshold** to

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

The Master Control Panel will display an Adapting indicator for each filter block whenever **Filter will adapt** is set to anything other than **Always**. The simulated LED will be RED when the filter is adapting 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 (filter order) in the adaptive filter. Filter size is indicated both in taps and in milliseconds. Minimum Filter Size is 32 taps, but can be set to as high as 4096 taps depending on System Bandwidth, Configuration, and Number DSP 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 256 x 2^-14 provides fastest adaptation.

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.

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 10 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.*

Adapt Mode: Selects Auto(matic) or Fixed adaptation rate. *Auto is recommended*. When Fixed is selected, the specified Adapt Rate Mu is applied to the filter at all times. However, when Auto is selected, the specified Adapt Rate is continuously rescaled depending upon the input signal level. Overall convergence rate is faster with Auto.

Filter Button: Used to switch the 1CH Adaptive filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

Adapt/Freeze Button: Used to freeze this adaptive filter independently.

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

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.

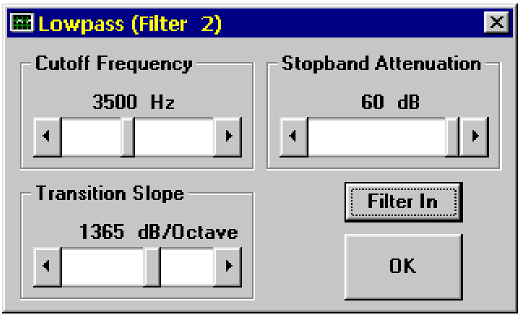


Figure 4-13 Lowpass Filter Control Window

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 Cutoff Frequency, System Bandwidth, Configuration, and Number DSP Stages settings.
Filter Button:	Used to switch the Lowpass filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process. **NOTE:** The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the lowpass filter and its controls is given in Figure 4-14.

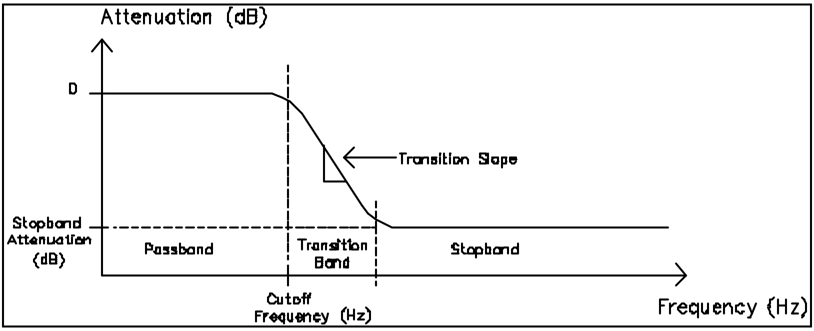


Figure 4-14 Lowpass Filter Graphical Description

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.

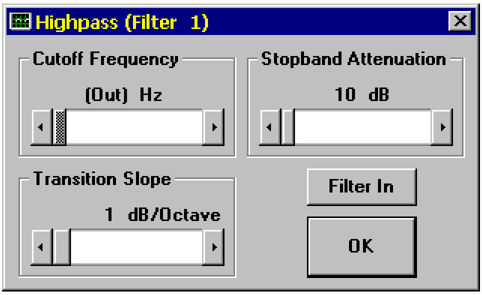


Figure 4-15 Highpass Filter Control Window

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 Cutoff Frequency, System Bandwidth, Configuration, and Number DSP Stages settings.
Filter Button:	Used to switch the Highpass filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process. **NOTE:** The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the highpass filter and its controls is as follows in Figure 4-16.

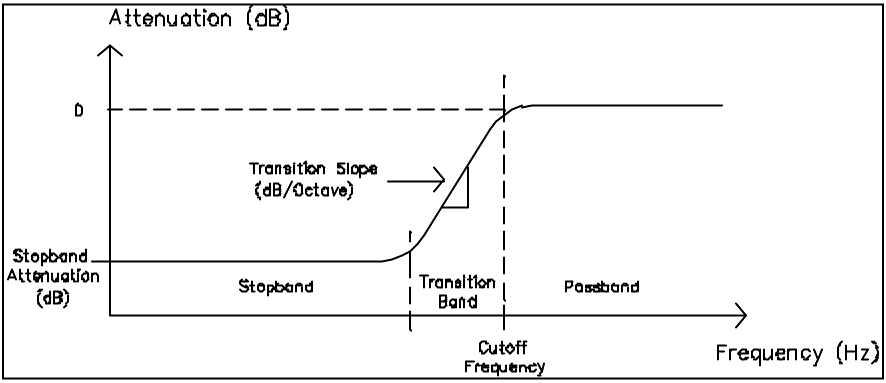


Figure 4-16 Highpass Filter Graphical Description

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 seriesed Lowpass and Highpass filter into a single filter. The signal region between the Lower Cutoff Frequency and the Upper Cutoff Frequency is called the <u>passband</u> 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.

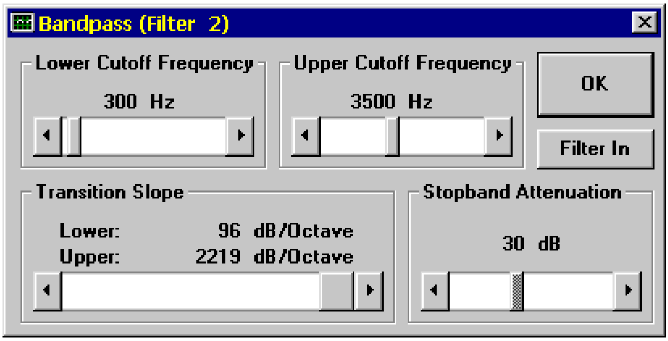


Figure 4-17 Bandpass Filter Control Window

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 never 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 never 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 Cutoff Frequency, System Bandwidth, Configuration, and Number DSP Stages 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 Button:	Used to switch the Bandpass filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process. **NOTE:** The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the Bandpass filter and its controls follows in Figure 4-18.

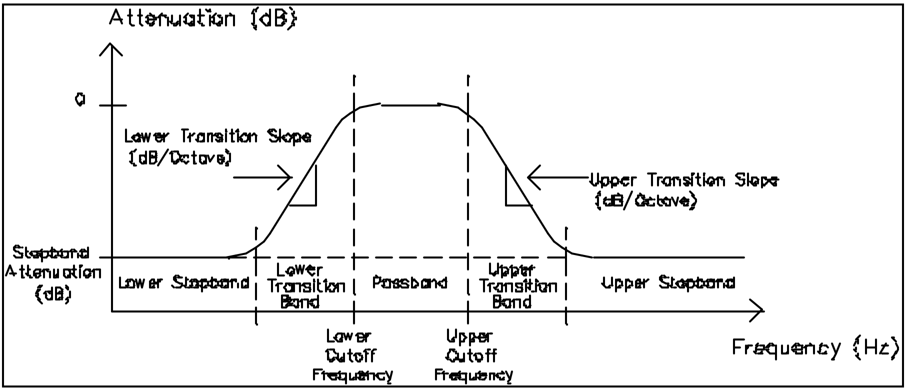


Figure 4-18 Bandpass Filter Graphical Description

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 stopband 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 never be set higher than 10 Hz below the Upper Cutoff Frequency.

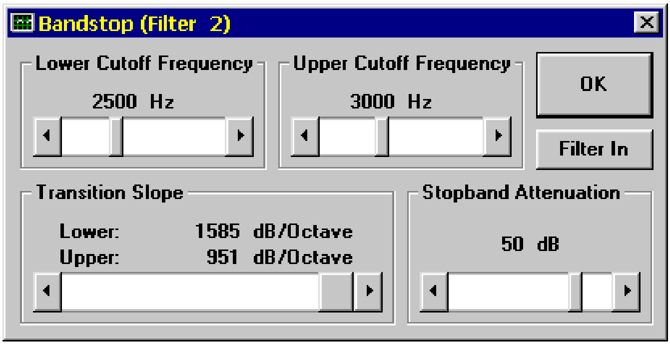


Figure 4-19 Bandstop Filter Control Window

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 <u>never</u> 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 Cutoff Frequency, System Bandwidth, Configuration, and Number DSP Stages 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 Button: Used to switch the Bandstop filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the Bandstop filter and its controls follows in Figure 4-20.

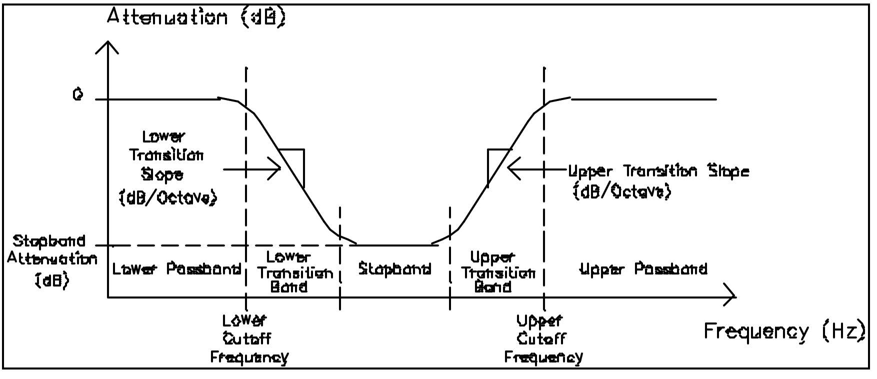


Figure 4-20 Bandstop Filter Graphical Description

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* *all* *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****.*

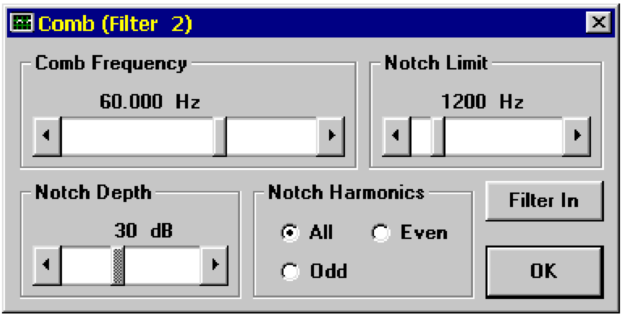


Figure 4-21 Comb Filter Control Window

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 Button: Used to switch the Comb filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

NOTE: The Filter button is also available in the Filter block for each filter on 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 <u>down</u> 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 1CH 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 4-22.

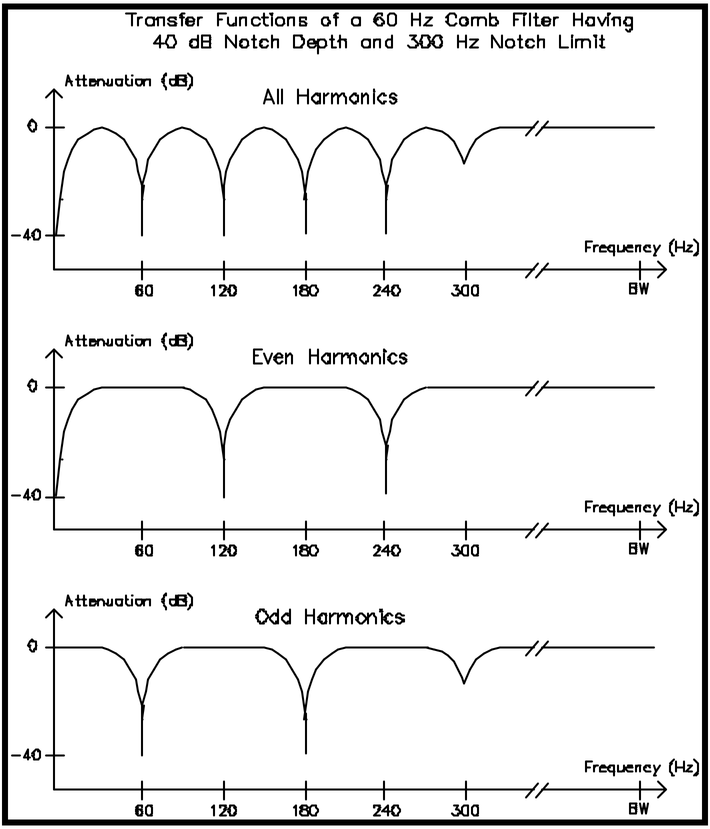


Figure 4-22 Comb Filter Graphical Description

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 1CH 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 4.6.3 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.

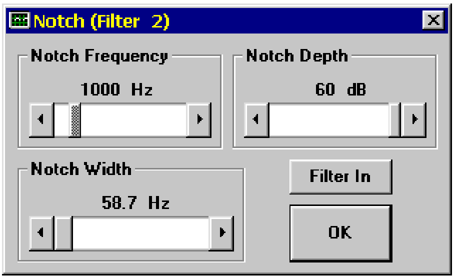


Figure 4-23 Notch Filter Control Window

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: Minimum Notch Width varies with System Bandwidth, Configuration, and Number DSP Stages.

Filter Button: Used to switch the Notch filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

Hint: A notch filter is best for stable tones. It has a sharp bottom. If a flatter bottom (stopband) is needed the bandstop filter (Section 0) may be prefered. Also, a 1CH Adaptive filter (Section 4.4.2) is useful in tracking varying tones.

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

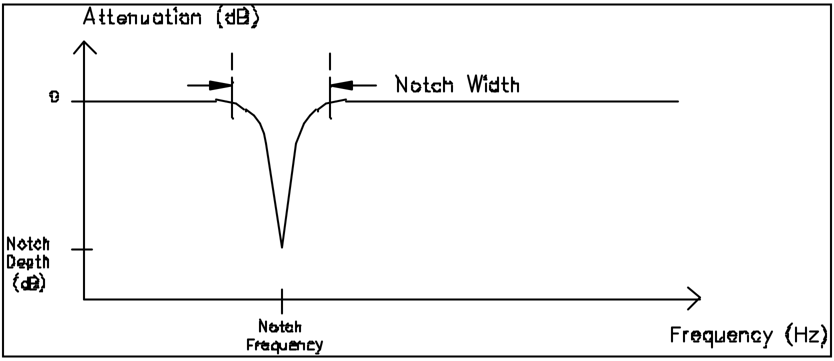


Figure 4-24 Notch Filter Graphical Description

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 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; if the frequency of a noise source varies excessively, then the 1CH 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 4.6.3 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 4-25 depicts the control window for notches 1 through 8. A second window for 9 through 16 is button accessed.

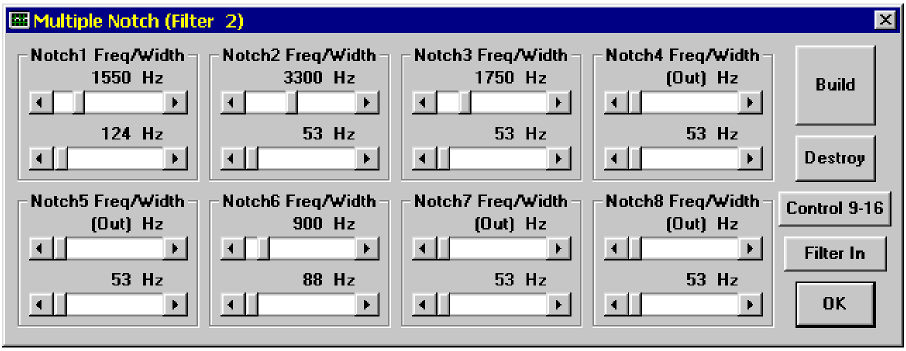


Figure 4-25 Multiple Notch Filter Control Window

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 downloaded to the PCAP II 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, Configuration, or Number DSP Stages in order to rebuild the filter.

Destroy 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).

Control Button: The Multiple Notch control window is only capable of displaying the settings for eight notches at a time; Notches 1-8 or Notches 9-16 may be displayed. Click on the **Control** button to toggle between **Control 1-8** (Notches 1-8 displayed) and **Control 9-16** (Notches 9-16 displayed).

Filter Button: Used to switch the Multiple Notch filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the Multiple Notch filter and its controls follows in Figure 4-26:

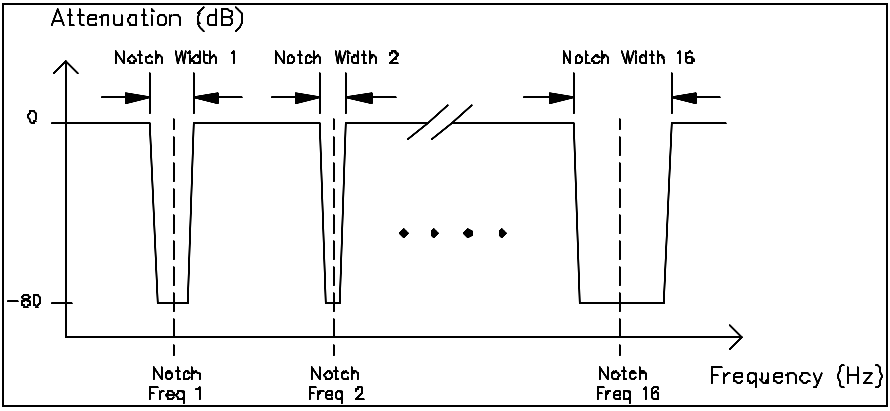


Figure 4-26 Multiple Notch Filter Graphical Description

4.4.10 Slot Filter

Application:

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 4.6.3 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.

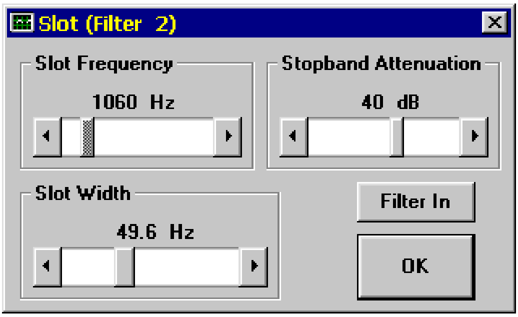


Figure 4-27 Slot Filter Control Window

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 System Bandwidth, Configuration, and Number DSP Stages.

Filter Button: Used to switch the Slot filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.

NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

A graphical description of the Slot filter and its controls follows in Figure 4-28. Note that the slot width is defined at its -6 dB points.

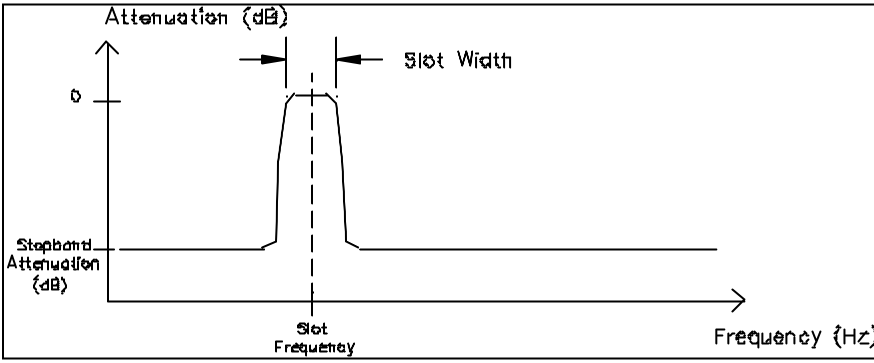


Figure 4-28 Slot Filter Graphical Description

4.4.11 Multiple Slot Filter

Application:

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 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 4.6.3 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.*

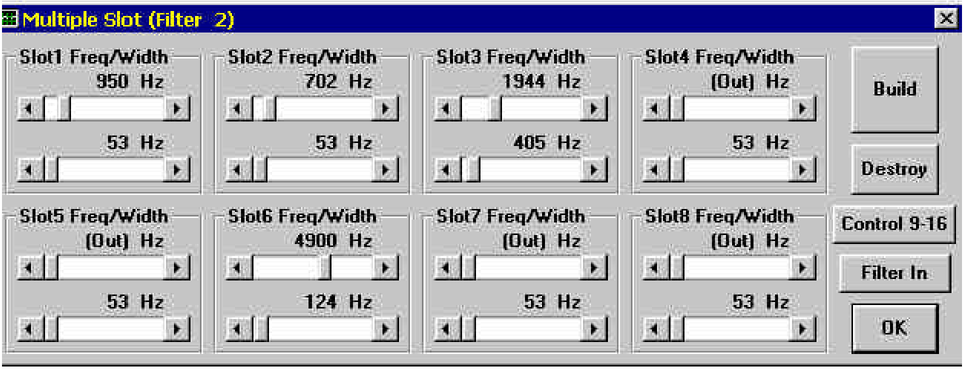


Figure 4-29. Multiple Slot Filter Control Window

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 PCAP II 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, Configuration,

	or Number DSP Stages in order to rebuild the filter.
Destroy 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).
Control Button:	The Multiple Slot control window is only capable of displaying the settings for eight slots at a time; Slots 1-8 or Slots 9-16 may be displayed. Click on the **Control** button to toggle between **Control 1-8** (Slots 1-8 displayed) and **Control 9-16** (Slots 9-16 displayed).
Filter Button:	Used to switch the Multiple Slot filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process.
	NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

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

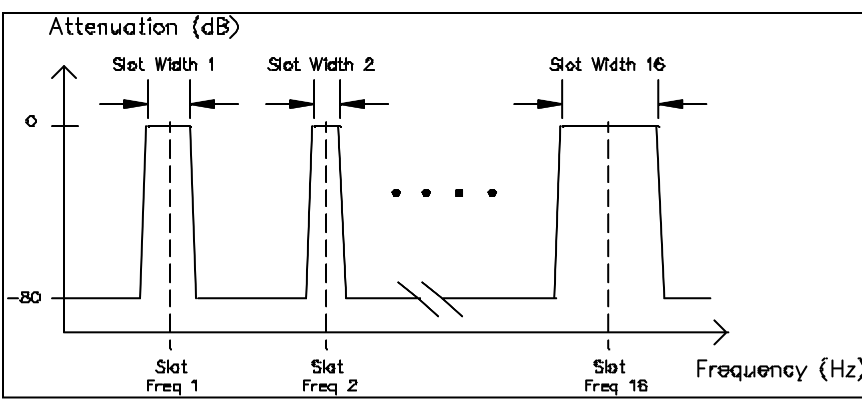


Figure 4-30 Multiple Slot Filter Graphical Description

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.

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 4-31 *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, or white, spectrum.*

This mode of operation is ***Equalized Voice.*** *Available controls permit the operator to reshape the output audio to white (shown), pink, or voice-like 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 (range) is adjustable using another control.*

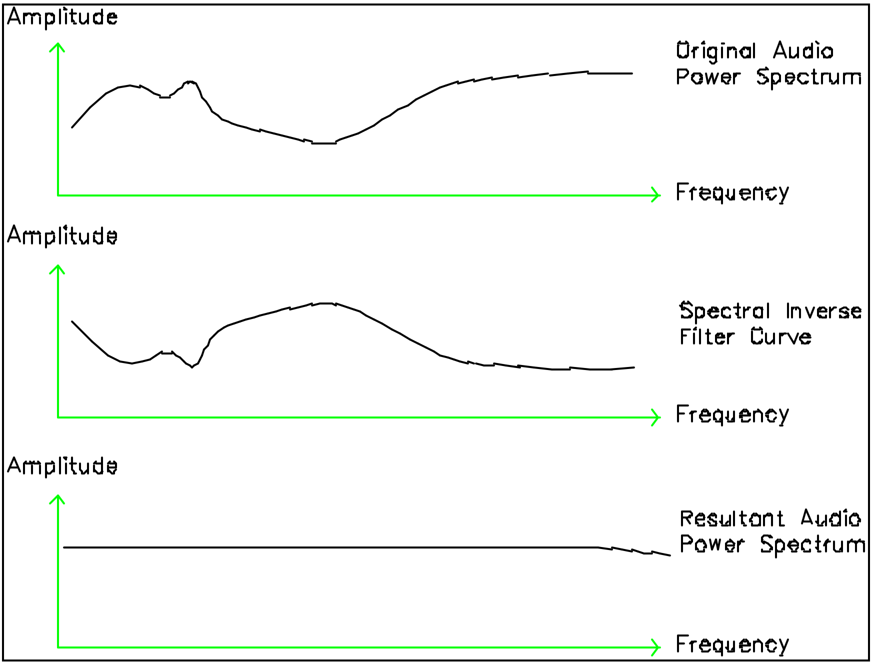


Figure 4-31 Basic Process of Spectral Inverse Filter

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 1CH 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. Audio outside these limits is unaffected.*

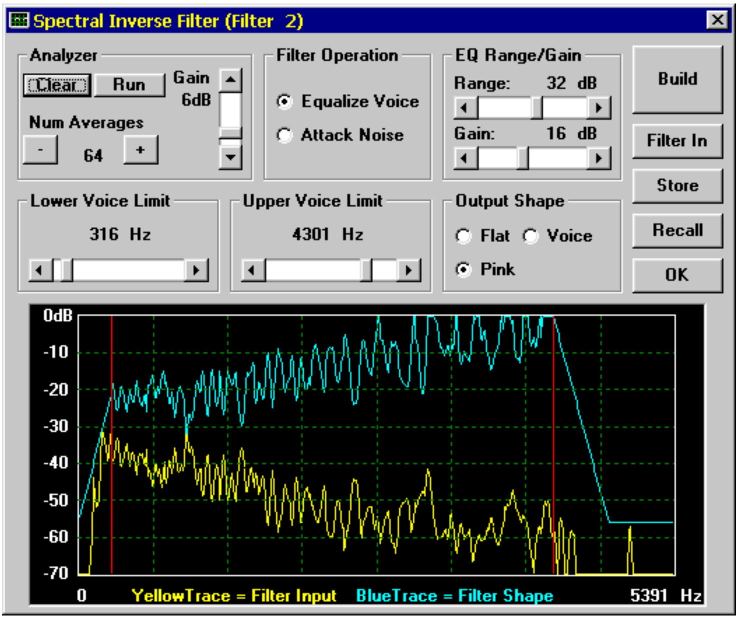


Figure 4-32 SIF Control Window When **Equalize** Voice Selected

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 green 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/Freeze** button which allows the user to start (**Run** indication) or stop (**Freeze** indication) update of the averaged spectrum.

- **Num 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 4-32. When **Attack Noise** is selected, the SIF Control Window appears as shown in Figure 4-33.

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).

Range/ Gain Block: For **Equalize Voice** operation, specifies EQ **Range** and **Gain**. For **Attack Noise** operation, specifies Attack **Range** and **Gain**.

EQ or Attack **Range** 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. Maximum **Range** is **50dB**.

NOTE: Maximum **Range** 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.

EQ or Attack **Gain** specifies the digital boost to be applied to the entire spectral inverse filter curve. Normally, **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. **Gain** should be initially set to approximately 50 percent of the **Range** setting. If filter output is distorted, reduce **Gain** setting and re-Build the filter; if filter output level is too low, try increasing the **Gain** setting.

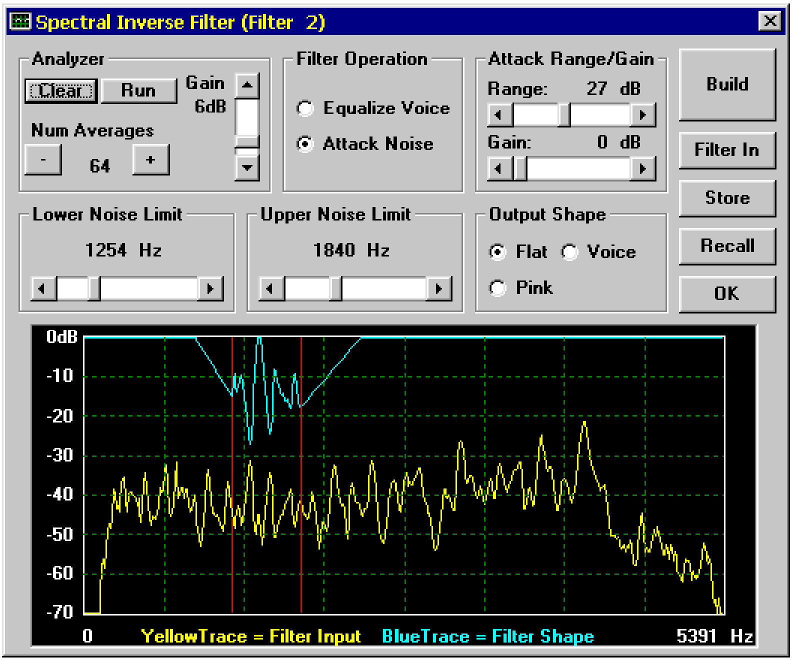


Figure 4-33 SIF Control Window When **Attack Noise** Selected

Lower and Upper Frequency Limits:	For **EQ 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 Attack Limit** and **Upper Attack 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.

NOTE: You must use the scroll bars; you cannot click the mouse at the desired points in the Filter Display area.

Output Shape Block:

Specifies the final reshaping curve to be applied to the entire SIFfilter. For **Attack Noise** only **Flat** should be used. For **Equalize Voice** three 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). 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.

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 Button: Used to switch the Spectral Inverse filter in and out of the process without affecting the other filters in the process. Button indicates **Filter In** when the filter is in the process, or **Filter Out** when the filter is out of the process. Whenever the **Build** button is clicked, this button will always be set to **Filter In**.

NOTE: The Filter button is also available in the Filter block for each filter on the Master Control Panel.

Store Button: This button allows the user to store a calculated spectral inverse filter curve to any of 60 disk memories (10 for each of the six **System Bandwidth** settings) which will not be lost when the computer is turned off. Clicking this button brings up the following window:

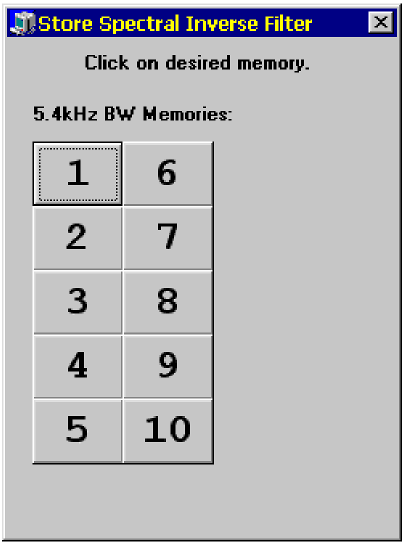


Figure 4-34 Spectral Inverse Filter Store Window

Clicking one of the 10 "jukebox-style" buttons will cause the current filter curve to be stored in that disk memory for the current **System Bandwidth**.

Recall Button: This button allows the user to recall a previously stored spectral inverse filter curve from one of 11 disk memories for the currently selected **System Bandwidth**. Clicking this button brings up the window in Figure 4-35.

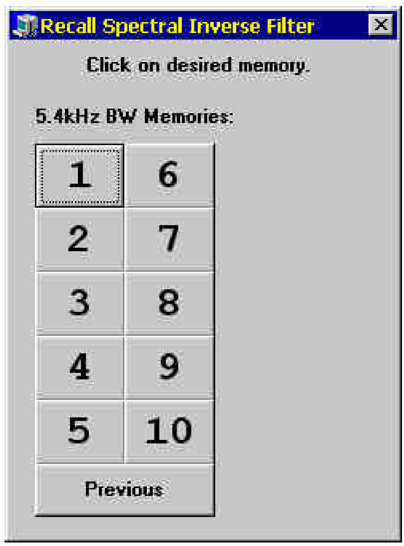


Figure 4-35 Spectral Inverse Filter Recall Window

Clicking one of the 10 "jukebox-style" buttons will cause the filter curve to be configured as previously stored. Clicking the eleventh button, labelled **Previous**, will cause the curve to be restored to the shape it had prior to the last memory recall at the current **System Bandwidth** setting. Successive clicks of **Recall**, then **Previous**, perform A/B comparison of two filter curves.

<u>Examples of Spectral Inverse Filters:</u>

In the examples below, SIF will equalize a voice spectrum using various **EQ Ranges** and **Output Shapes**. When the Range is small, only the peaks in the spectrum are flattened. As the Range 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 4-36, Figure 4-37, and Figure 4-38 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 **EQRange** is increased, more peaks are attenuated. Figure 4-38 completely compensates all peaks and valleys. Note also the 40 dB attenuation outside the two Limits.

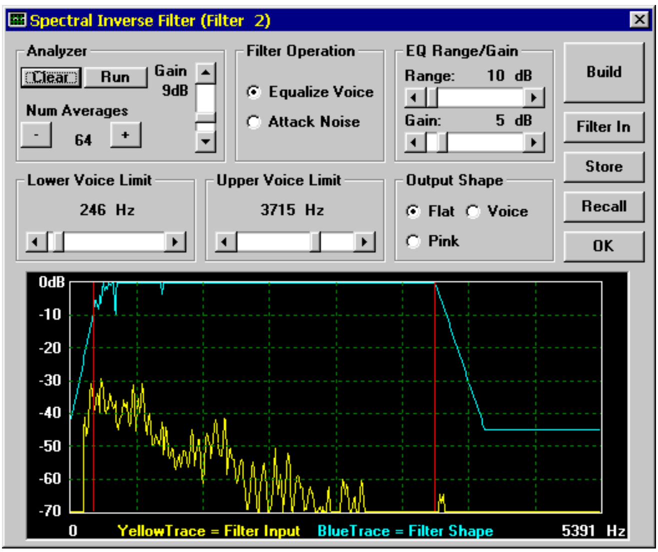
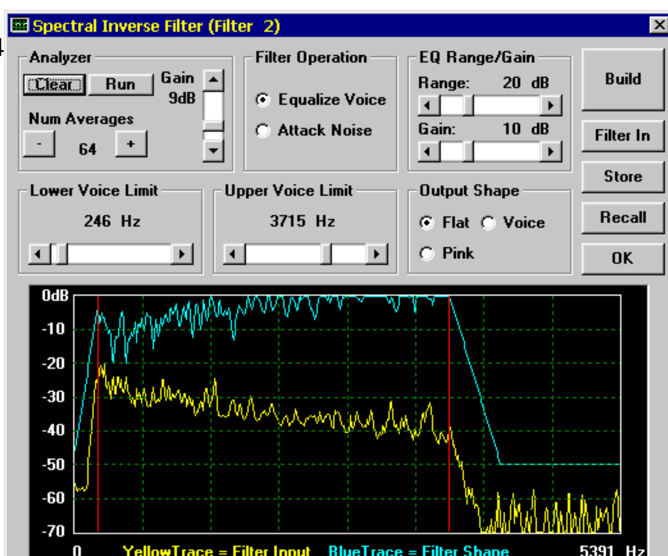


Figure 4 **Shape**



The Output spectral Shape may be selected as Flat, illustrated in Figure 4-36, Figure 4-37, and Figure 4-38, above. It may also be set to Voice or Pink. See Figure 4-40.

The Attack Noise mode does not attenuate Out-of-Limits signal, but equalizes and attenuates in-Limits signal frequencies. Figure 4-39 illustrates.

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 **Num 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.

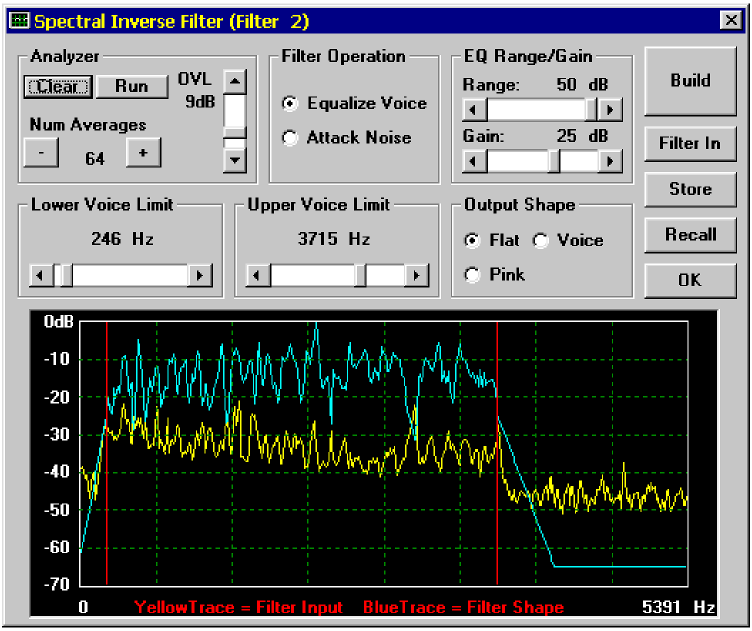


Figure 4-38 SIF with **EQ Range** set to **50dB**

Try to capture a representative spectrum using the Freeze/Run button. Once a stable, representative spectrum is obtained on the display, Freeze the analyzer. This same spectrum may be used for several different variations of the filter (changing Limits or Range, as an example).

NOTE: Immediately following 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 Lower 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 will series up to four). The audio outside those limits is unfiltered.

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

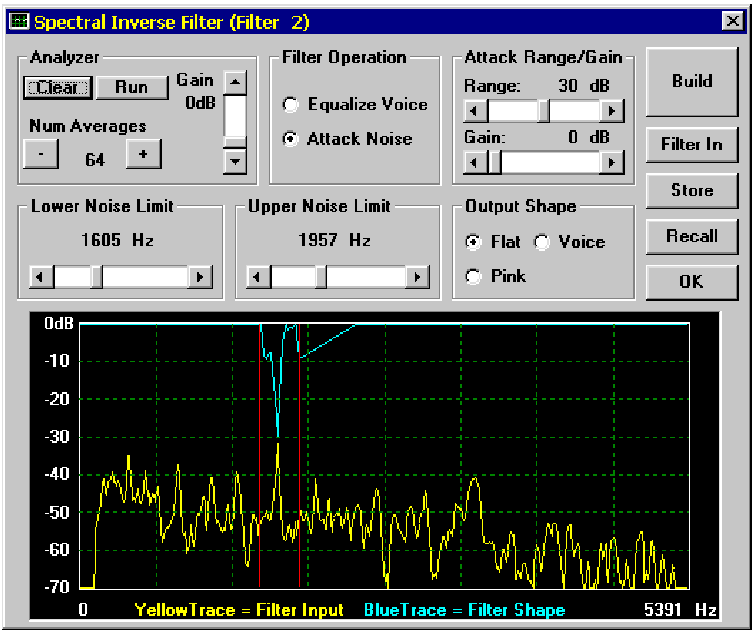


Figure 4-39 **Attack Noise** operation, **Attack Range** set to **30dB**

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, even if the scroll bar is set to a larger **Range**. Normally a **Range** of 10-20dB is adequate, as attacking the stronger spectral peaks will provide the necessary enhancement. Go gently at first and compare several SIFs with different **Ranges**. 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 **Gain** scroll bar. Normally, setting this to one-half the **Range** is adequate. If output distortion occurs, reduce the **Gain**. The **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: Three 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.*

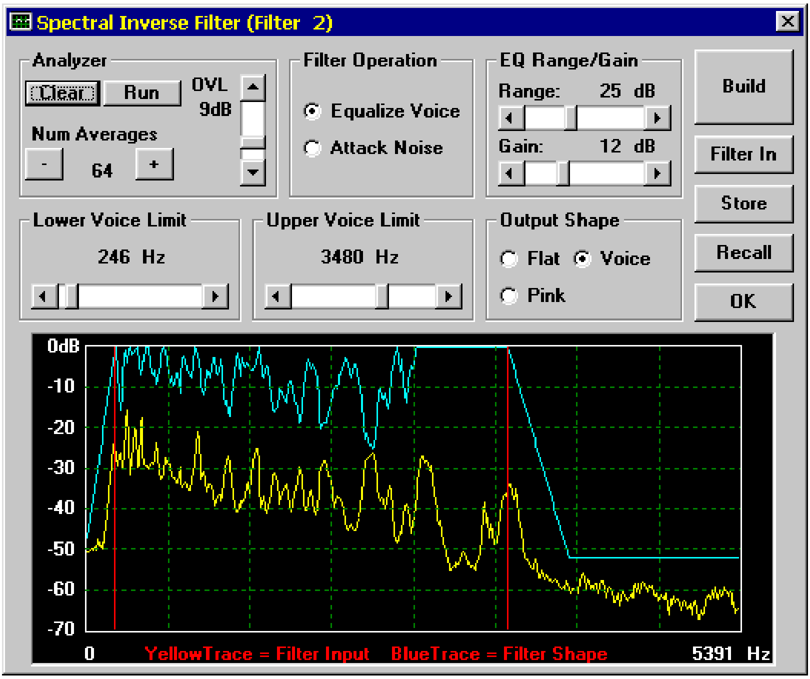


Figure 4-40 SIF with **Output Shape** set to **Voice**

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 DSP 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 PCAP II unit.

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.*

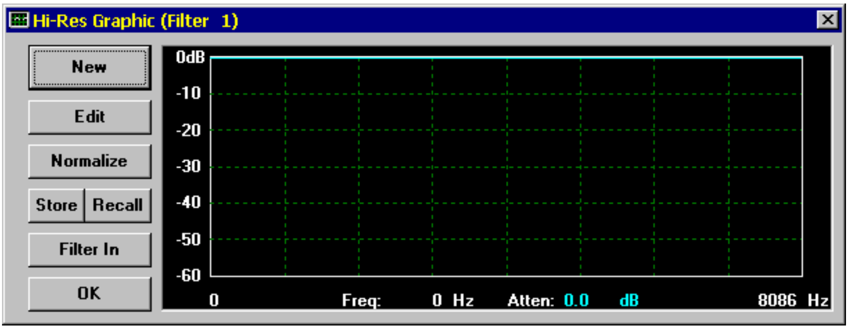


Figure 4-41 Hi-Res Graphic Filter Window

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 filter curve to any of 60 disk memories (10 for each of the six **System Bandwidth** settings) which will not be lost when the computer is turned off. Clicking this button brings up the following window (Figure 4-42):

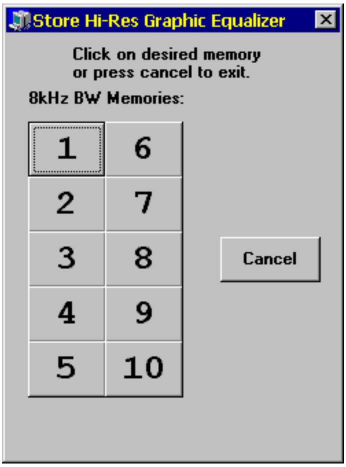


Figure 4-42 Hi-Res Filter Store Window

Clicking one of the 10 "jukebox-style" buttons will cause the current filter curve to be stored in that disk memory for the current **System Bandwidth**. Clicking the **Cancel** button exits the Store window without storing the filter curve in any of the memories.

Recall Button: This button allows the user to recall a previously stored filter curve from one of 11 disk memories for the currently selected **System Bandwidth**. Clicking this button brings up the following window (Figure 4-43).

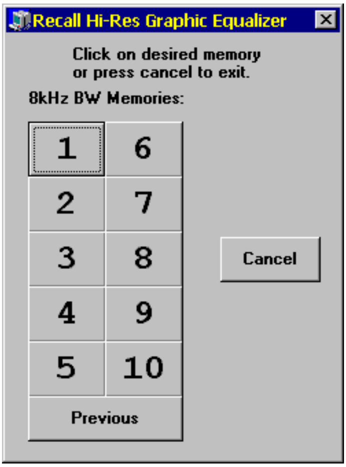


Figure 4-43 Hi-Res Graphic Filter Recall Window

Clicking one of the 10 "jukebox-style" buttons will cause the filter curve to be configured as previously stored. Clicking the eleventh button, labelled **Previous**, will cause the curve to be restored to the shape it had prior to the last memory recall at the current **System Bandwidth** setting. Successive clicks of **Recall**, then **Previous**, perform A/B comparison of two filter curves.

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. From the PCAP II Master Control Panel, set **System Bandwidth** to **5.4 kHz**, **Configuration** to **Mono**, and **Number DSP Stages** to **2**. Click on the **Select** button in the **Filter 1** block, then select **Hi-Res Graphic** from the selection window.

2. Click on the **Control** button in the **Filter 1** block 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 4-41.

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

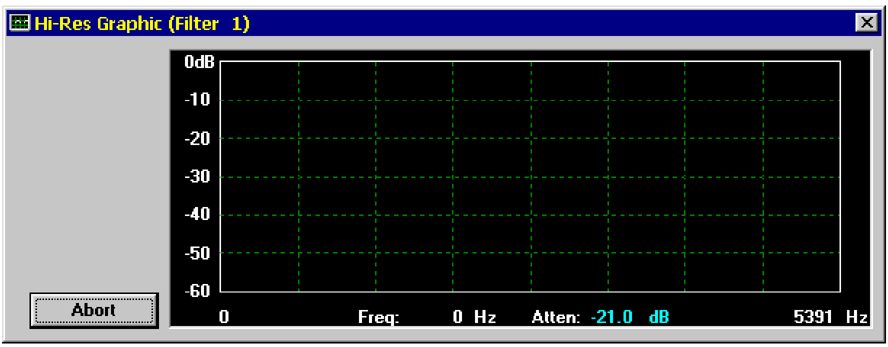


Figure 4-44 New Hi-Res Graphic Filter Display

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 draw 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 4-45 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.

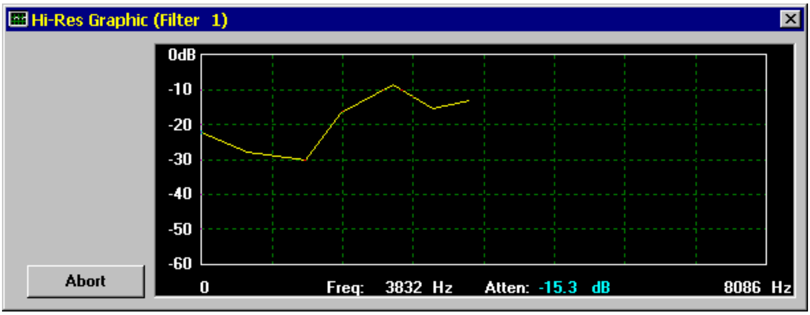


Figure 4-45 Hi-Res Graphic Draw in Progress

5. Complete drawing the filter curve as shown below (Figure 4-46) by drawing points all the way to the right edge of the filter display area.

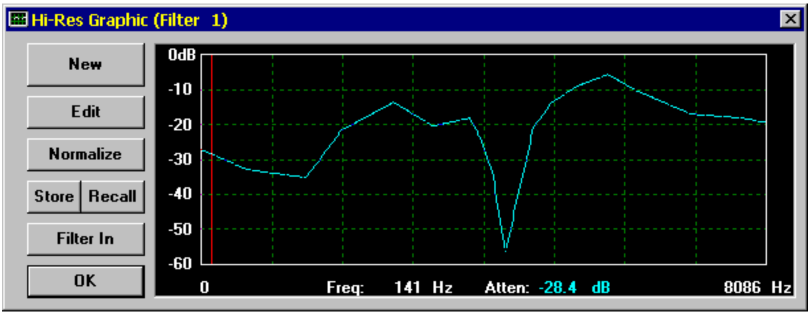


Figure 4-46 Completed Hi-Res Graphic Draw

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 3000 Hz in Figure 4-46, above. Click on **Edit** to bring up the following display (Figure 4.45):

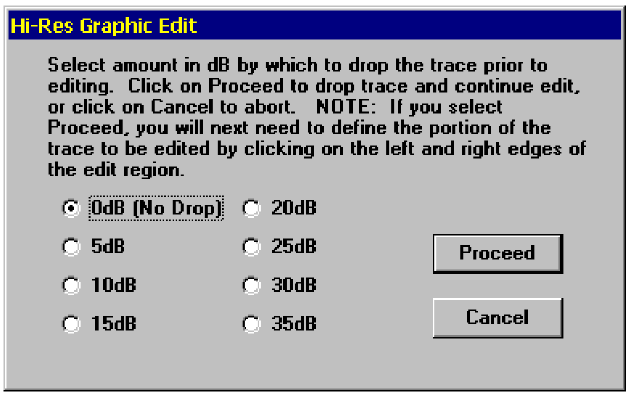


Figure 4-47 Hi-Res Graphic Edit Window

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 4-48):

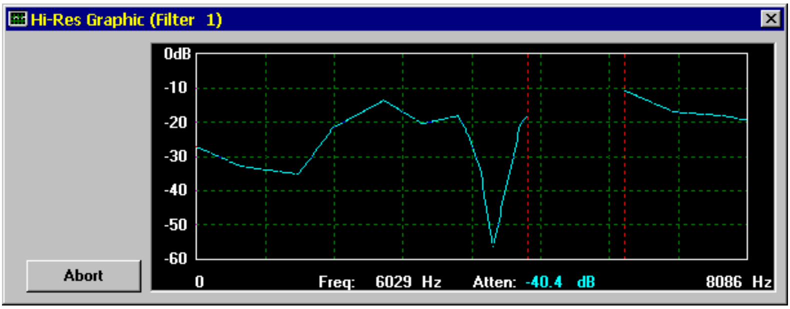


Figure 4-48 Hi-Res Graphic Define Edit Region

8. Now, draw in the new portion of the filter curve using mouse clicks as in Step 4, above, roughly as shown below (Figure 4-49).

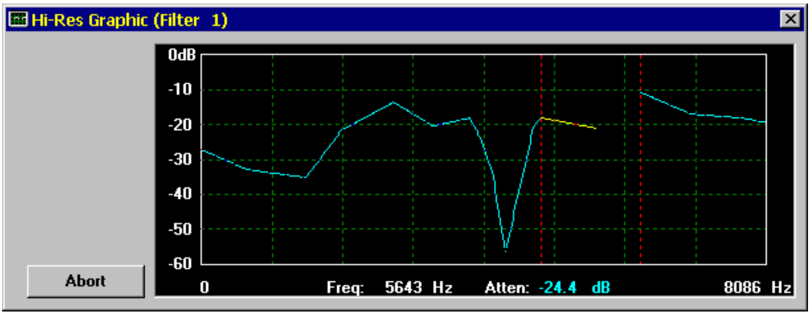


Figure 4-49 Hi-Res Edit In Progress

9. Complete drawing the new portion of the filter curve as shown below (Figure 4-50) by drawing points all the way to the right edge of the edit region:

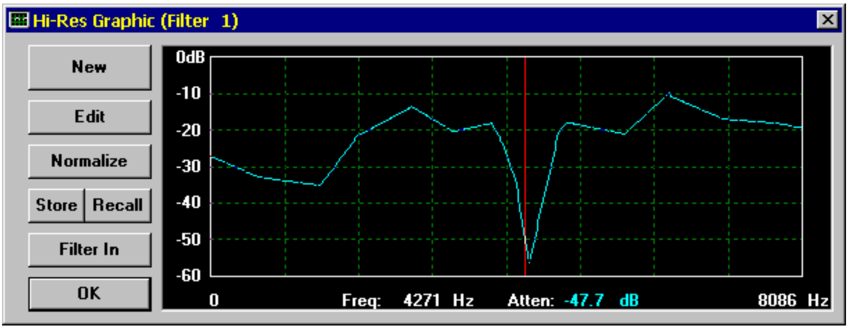


Figure 4-50 Completed Hi-Res Graphic Edit

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 4-51):

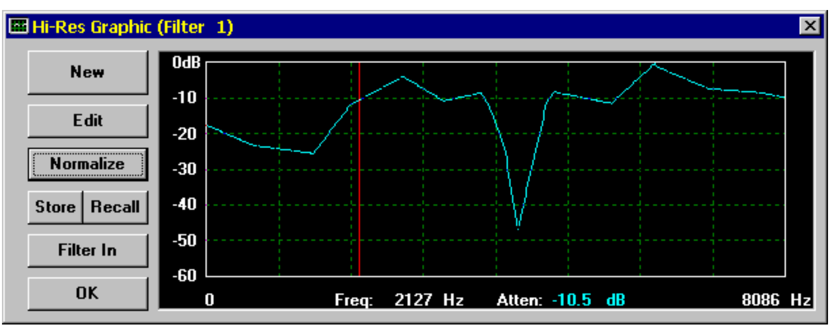


Figure 4-51 Normalized Hi-Res Graphic Filter

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

4.4.14 Tri Parametric Filter

Application:

The Tri Parametric Filter is a three-substage adjustable IIR filter which can be used for both ***peaking*** *and* ***nulling*** *portions of the input signal's frequency spectrum. Controllable parameters for each seriesed substage include* ***Center Frequency, Width,*** *and* ***Boost/Cut.***

Typically, parametric equalizers are used as notch filters, which perform ***nulling*** *of the input signal at a specified Center Frequency 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, the parametric filter 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 Each filter provides three peaking or* ***nulling*** *stages that can be configured independently.*

Each substage can be independently bypassed using the ***Substage In/Out*** *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*** *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.*

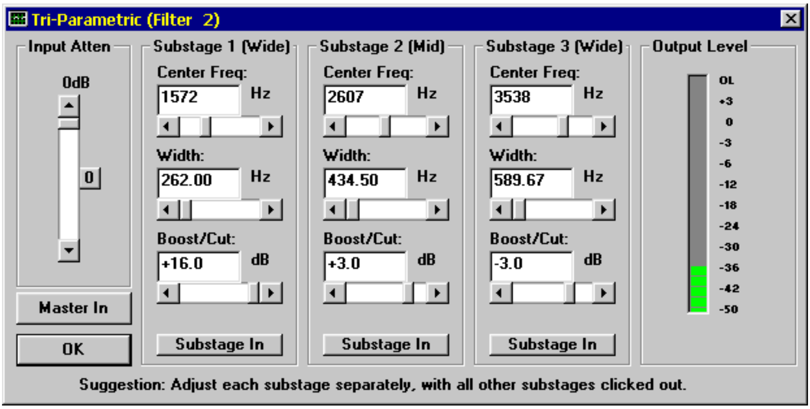


Figure 4-52 Tri Parametric Filter

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 manual adjustments. Click on the left/right arrow button at either end of the slider to make fine adjustments to center frequency, or drag the scroll box to make coarse 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 of the PCAP can easily be exceeded.

The **Substage In/Out** buttons allow the user to quickly bypass (**Substage Out**) any stage.

The **Input Atten** 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. Use the **0** button 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 PCAP Master Control Panel.

The **Master In/Out** button allows the entire parametric filter to be clicked **In** or **Out** of the process at any time. A graphical description of a parametric filter and its controls follows in Figure 4-53.

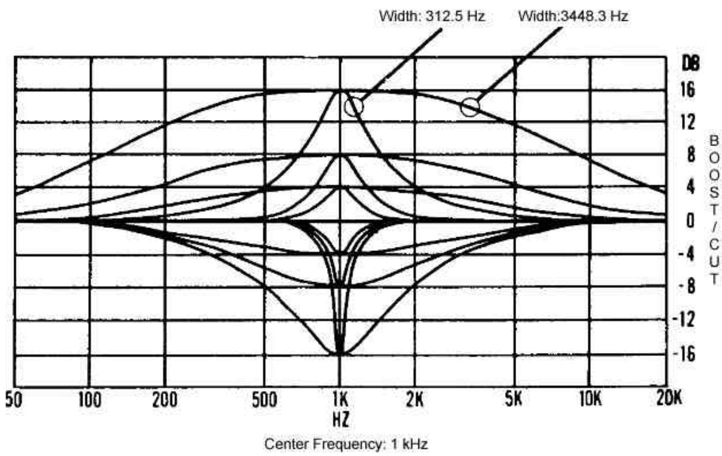


Figure 4-53 Graphical Representation of a Parametric Filter

4.4.15 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) in the PCAP II are limiting, compression, and expansion. Unlike an analog implementation of this process, the PCAP II 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 PCAP II's microprocessor. 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 envelope detector is controlled 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:*

> ***<u>Limit Region</u>**: In the limit region the output signal level is "damped" or kept from exceeding the specified **Limit Threshold** by applying attenuation.*
>
> ***<u>Compression Region</u>**:* 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 decrease of only 10 dB when the input signal decreases 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 are modestly preserved.
>
> ***<u>Expansion Region</u>**:* In the 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 decrease 30 dB when the input signal decreases only 10 dB. 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.

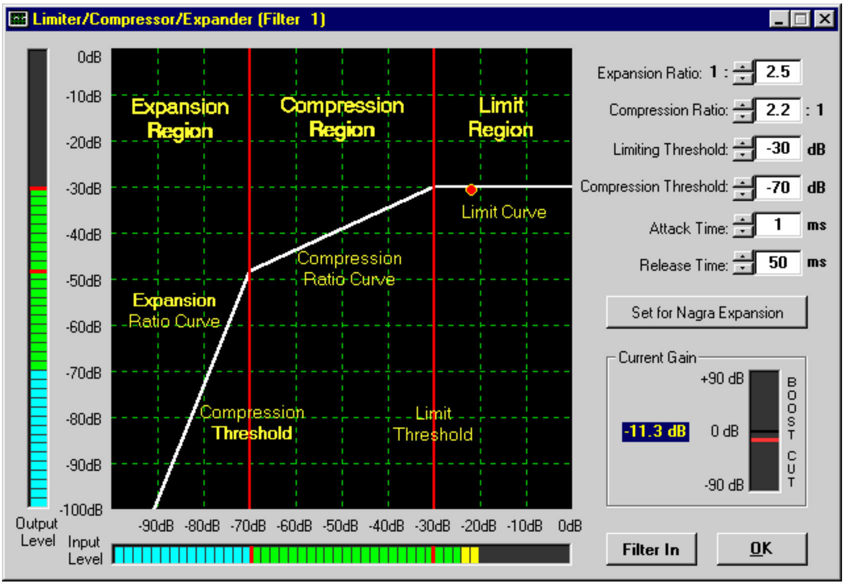


Figure 4-54 Limiter/Compressor/Expander Control Window

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 PCAP.

4.5 DSP 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 in 1CH adaptive filters.) The output equalizer enables the operator to restore naturalness to the speech and deemphasize residual high-frequency hiss.

4.5.1 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* ***Make All 0dB*** *button instantly sets all the sliders to 0dB. The* ***All Down 1dB*** *button instantly moves all sliders* ***Down 1dB****. 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.*

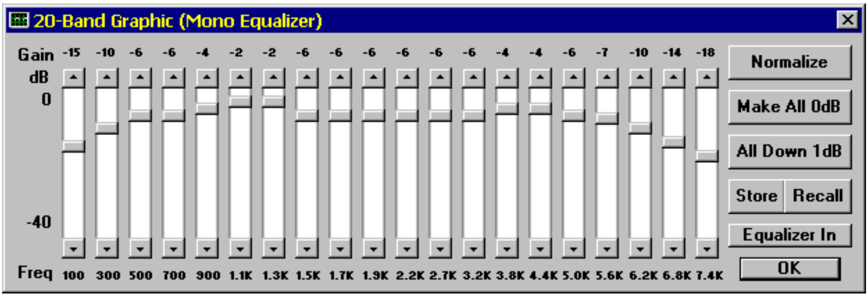


Figure 4-55 20-Band Graphic Equalizer Control Window

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.

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.

Make All 0dB 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.

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 any of 60 disk memories (10 for each of the 6 **System Bandwidth** settings) which will not be lost when the computer is turned off. Clicking this button brings up the window shown in Figure 3-13 of Chapter 3.0.

Clicking one of the 10 "jukebox-style" buttons will cause the current slider configuration to be stored in that disk memory for the current **System Bandwidth**. 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 one of 11 disk memories for the currently selected **System Bandwidth**. Clicking this button brings up the window given in Figure 3-14 of Chapter 3.0.

Clicking one of the 10 "jukebox-style" buttons will cause the sliders to be configured to the previously stored settings for that memory. Clicking the eleventh button, labelled **Previous**, will cause the sliders to be restored to the settings they had prior to the last memory recall at the current **System Bandwidth** setting. Successive clicks of **Recall**, then **Previous**, perform A/B comparison of two equalizer configurations.

4.5.2 Spectral Graphic Equalizer

Application:

For many output equalization requirements, the 20-Band Graphic Equalizer described in Section 4.5.1 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 of the PCAP II 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.

4.5.3 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.

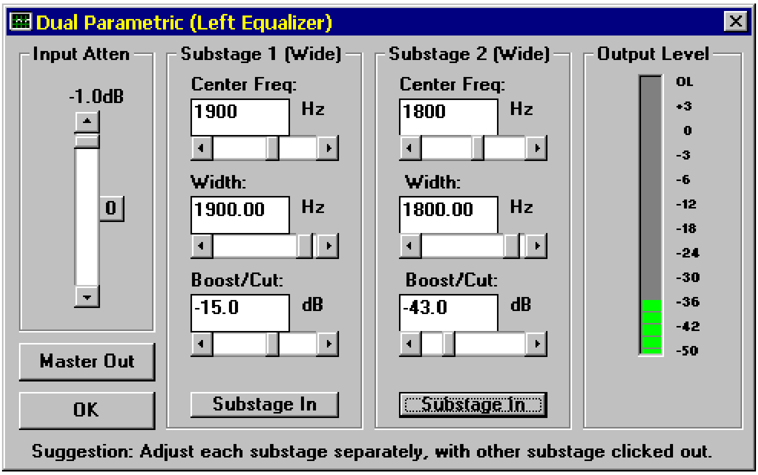


Figure 4-56 Dual Parametric Filter

4.6 DSP Display Selection

4.6.1 Spectrum Analyzer and Coefficient Display Buttons

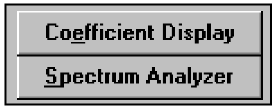


Figure 4-57 Spectrum Analyzer and Coefficient Display Buttons

Pressing the **Spectrum Analyzer** button at any time brings up the Spectrum Analyzer window. For fast access to the Spectrum Analyzer window, press <Alt-S> on the keyboard. See Section 4.6.3 for complete instructions on operating the controls in the Spectrum Analyzer window. Pressing the Coefficient Display button at any time brings up the Coefficient Display window. For Fast access to the Coefficient Display, press <Alt-E> on the keyboard. See Section 4.7 for complete instruction on operating the controls in the Coefficient Display window.

4.6.2 Display Select Window

Clicking on **View** in the menu bar causes the following window (Figure 4-58) to appear:

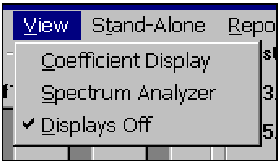


Figure 4-58 Display Select Window

Click the mouse on the desired type of display. Only one display will operate at a time. The display currently active is identified in the **Status** box of the Master Control Panel. Currently, the only supported displays are the Spectrum Analyzer and the filter Coefficient Display. Select **No Display** to turn off any display currently operating.

For fast access to the Display Select window, press <Alt-V> on the keyboard.

See Section 4.6.3 for complete instructions on operating the controls in the Spectrum Analyzer window. For complete instructions on operating the Coefficient Display window, see Section 4.7.

4.6.3 Spectrum Analyzer Window

Application:

To properly utilize the DSP Filter and Equalizer stages, 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, a ***Yellow Trace*** *and a* ***Blue Trace****, can be displayed either simultaneously or separately. Either trace can be configured to display any signal. The* ***Averager*** *is used to average 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.*

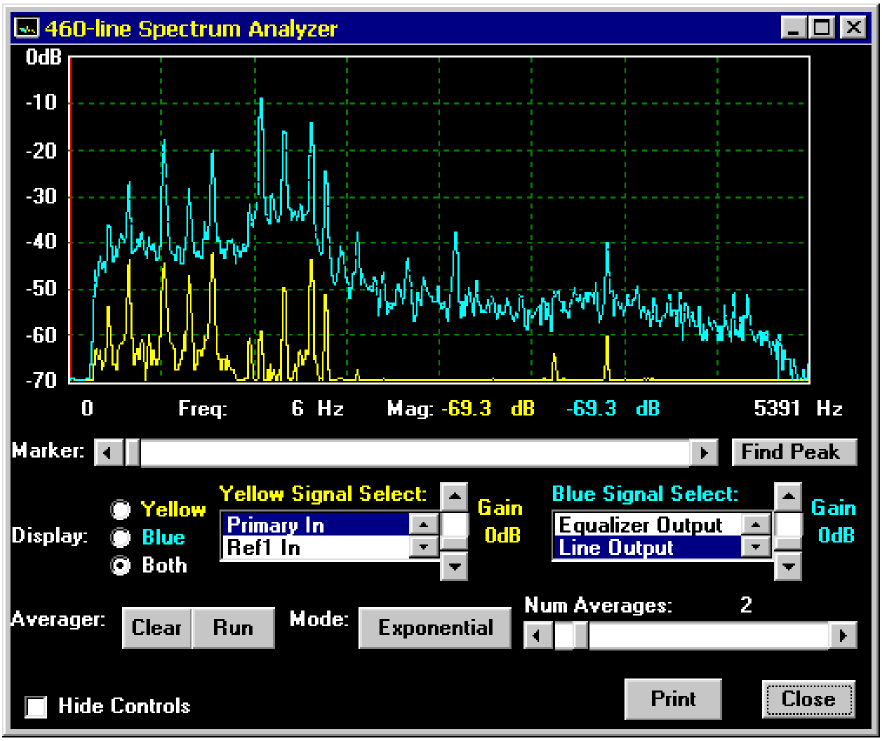


Figure 4-59 Spectrum Analyzer Window

Description of controls/indicators is as follows:

Averager Block: Controls averaging of successive FFT spectral traces.

Num Averages selects the number of averages to be applied to spectral traces. The more averages applied, the smoother the displayed spectrum waveforms will be; however, the waveforms will also update more slowly as Num Averages is increased. For voice filtering applications 8 to 32 averages are recommended. The user is allowed to freeze an averaged trace by clicking the Run/Freeze button between **Run** and **Freeze**. The user may clear the averaged trace(s) and restart the average at any time by clicking on the **Clear** button.

Yellow Trace Block: Defines what is displayed by the yellow-colored spectrum trace. The **Signal Select** box highlights the currently selected signal and allows the user to scroll through the available signals and select a different one with a mouse click.

The **Gain** scroll bar allows the user to apply digital gain to the Yellow Trace 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 Yellow Trace 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.

Blue Trace Block: Defines what is displayed by the blue-colored spectrum trace.

The **Signal Select** box highlights the currently selected signal and allows the user to scroll through the available signals and select a different one with a mouse click.

The **Gain** scroll bar allows the user to apply digital gain to the Blue Trace 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 Blue Trace input may be overloaded, causing a distorted spectrum display. When an overload does occur, the **Gain** label will change to **OVL** and the blue magnitude readout at the bottom of the display will change from blue to red.

Displayed Trace: Block: Allows the user to display the Yellow and Blue spectral traces simultaneously or separately.

Selecting **Yellow** displays the Yellow Trace only, selecting **Blue** displays the Blue Trace only, and selecting **Both** displays both the Yellow Trace and the Blue Trace.

It is recommended that the **Both** setting be used only when necessary. Since the trace update will require twice the time to complete as the **Yellow** or **Blue** setting, the display will be more sluggish.

Marker Block: Used to move the vertical red marker in the Spectrum Display area (described below), allowing frequency (**Freq**) and magnitude (**Mag**) readout of any point in the spectra.

The horizontal scroll bar is used to precisely place the marker at the desired point in the spectra. Make coarse adjustments to marker position by dragging the scroll box ("slider") with the mouse. Click on the scroll bar or scroll arrows at either end of the scroll bar to make fine adjustments.

Clicking the **Find Peak** button instantly moves the marker to the largest magnitude in the Spectrum Display.

For faster placement of the marker, click the mouse cursor on the desired point in the Spectrum Display area.

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. A green grid is 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 move 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.

Close Button: Clicking on this button turns off the Spectrum Analyzer display.

4.7 Coefficient Display Window

Application:

In setting up a 2CH Adaptive filter it is sometimes 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 digital Filter stages. For these reasons, the Coefficient Display window has been provided.

The Filter stage to be displayed is specified in the ***Filter Select*** *block by clicking on the desired Filter.*

The number of coefficients in the Filter to be displayed is specified in the ***Coefficients to Display*** *block by clicking on the desired number; this causes the specified number of coefficients to be horizontally scaled to fit within the display area. Regardless of the length of the Filter, clicking on* ***All*** *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 ***Scale*** *factor. Supported* ***Scale*** *factors range from* ***1X*** *to* ***200X****.*

A moveable ***Marker*** *allows* ***Time*** *and* ***Value*** *readout at any point in the Coefficient Display.*

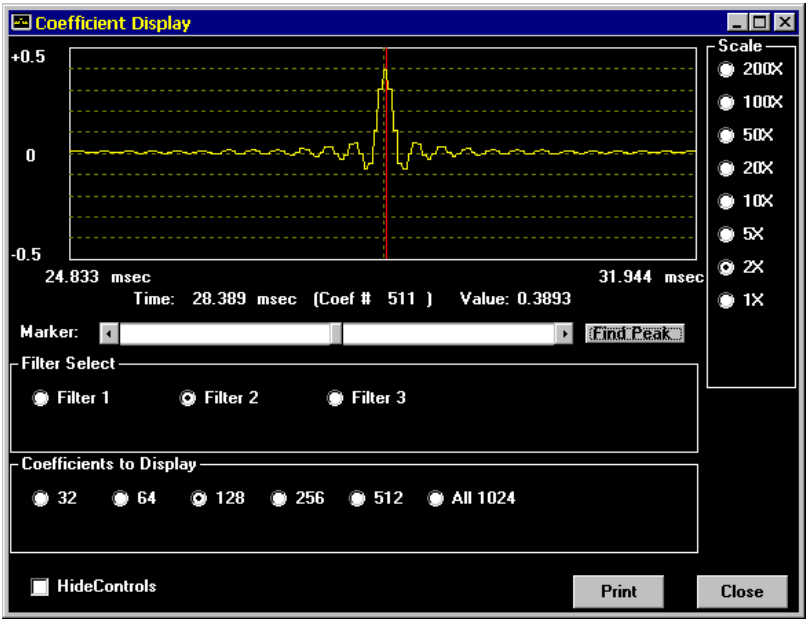


Figure 4-60 Coefficient Display Window

Description of controls is as follows:

Filter Select Block: Used to select the DSP Filter stage whose coefficients are to be displayed. Available options depend upon the **System Bandwidth**, **Configuration**, and **Number DSP Stages** settings.

Coefficients to Display Block: Used to select how many coefficients in the selected Filter are to be horizontally scaled to fit the display area. The minimum number of coefficients that can be displayed is 32, while the maximum number of coefficients varies according to the actual size of the Filter. To display all the coefficients in a Filter stage, click on **All**.

The maximum number of coefficients that can be displayed within the horizontal resolution of the Coefficient Display area is 512. When **Coefficients to Display** is set to **512** or less, coefficients are accurately displayed; however, when **Coefficients to Display** is set larger than **512**, the message **(Max of x)** will appear beside the **Value** indication, informing the user that only the largest value of every group of **x** coefficients is displayed. As an example, if 2048 coefficients are selected only 512 points are actually displayed. Each point is the largest magnitude in its group of four coefficients.

Scale Block: Used to specify the vertical scale factor to be used when displaying coefficients. Coefficients may be scaled 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. This clipping does not affect signal processing.

Marker Block: Used to move the vertical red marker in the Coefficient Display area (described below), allowing **Time** and **Value** (bottom of windows) readout of any coefficient to be displayed.

The horizontal scroll bar is used to precisely place the marker at the desired filter coefficient. Make coarse adjustments to marker position by dragging the scroll box ("slider") with the mouse. Click on the scroll bar or scroll arrows at either end of the scroll bar to make fine adjustments.

The Coefficient Display area will "pan", or shift, left or right whenever the marker selects a coefficient that is outside the displayed area.

For faster placement of the marker, click the mouse cursor on the desired point in the Coefficient Display area. Click on the left or the right side of the Coefficient Display grid to pan left or right.

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.

Close Button: Clicking on this button turns off the Coefficient Display.

4.8 Master Control Pulldown Windows

4.8.1 Saving Setups to Disk Files

Application:

To save time configuring PCAP II control settings, complete setups may easily be saved to disk setup files for future opening. 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 systems using floppy disks.

*Note: Any setup files previously stored by the Digital Audio Corporation **PCAP** software can be loaded by both the PCAP II and **MCAP** software. The PCAP II software is also capable of operating the **MCAP** unit.*

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 4-61 to appear:

2. Normally, it will not be necessary to change the **Drive** or the **Directories** settings; if, however, you desire to place the setup file into a different drive (such as a floppy drive) or directory, scroll through the lists of available drives and directories and select the desired drive or directory by double-clicking it. If you select a drive that is not ready, an error message will be generated.

3. Each stored setup will have a time stamp and a user-entered text **Description** to uniquely identify it. The time stamp will be set automatically when the file is stored, but 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.

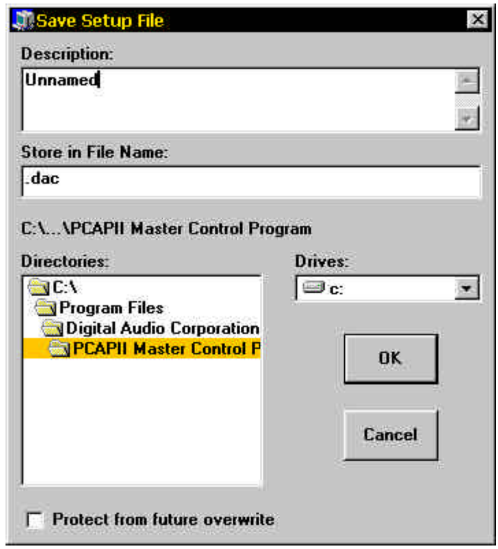


Figure 4-61 Save Setup File Window

4. You will need to specify a filename for the setup. Click on the **Store in 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 the Windows Explorer or by MS-DOS.

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

4.8.2 Recalling Setups from Disk Files

Application:

To save time configuring control settings in the PCAP II Master Control program, complete setups previously stored may be recalled from disk files with a few simple mouse clicks. This is particularly handy when making presentations which require multiple setups, or when it is desired to precisely duplicate a previous enhancement procedure. Also, this feature allows easy transfer of enhancement setups between PCAP II systems using floppy disks.

Note: Any setup files previously stored by the Digital Audio Corporation ***PCAP*** *software can be loaded by both the PCAP II and* ***MCAP*** *software. The PCAP II software is also capable of operating the* ***MCAP*** *unit.*

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 4-62) to appear:

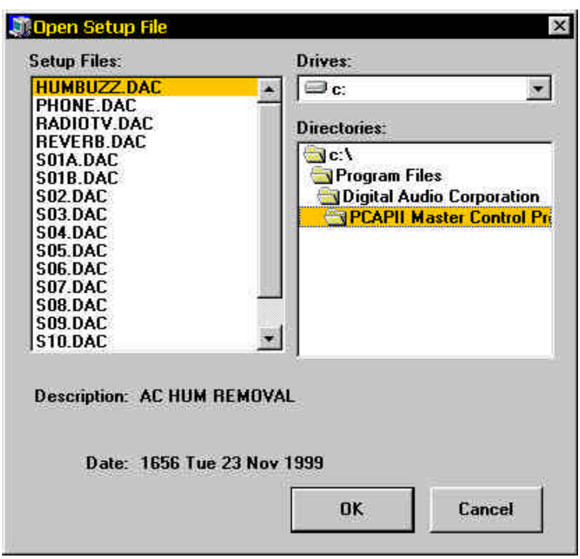


Figure 4-62 Open Setup File Window

2. Normally, it will not be necessary to change the **Drive** or the **Directories** settings; if however, you need to open a setup file from a different drive (such as a floppy drive) or directory, scroll through the lists of available drives and directories and select the desired drive or directory by double-clicking it. If you select a drive that is not ready or a directory which has no setup files in it, an error message will be generated.

3. Each stored setup has a time stamp and 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 setup time stamp and **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 is busy configuring itself with the recalled settings from disk.

4.8.3 Viewing DSP Displays

Click on **View** to access the **Display Select** window shown in Figure 4-58 on page 120. See Section 4.6.2 for further details.

4.8.4 Storing Setups to External Processor Stand-Alone Memories

Application:

*The PCAP II 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, Filter, 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 4.8.2.

   It is strongly suggested that you save this enhancement setup to disk using the **Save Setup File** feature described in Section 4.8.1. *NOTE: There is no way for the Master Control program to retrieve from the PCAP II 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 4-63):

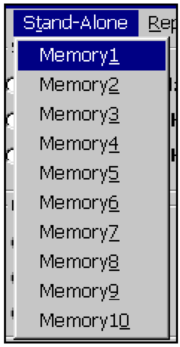


Figure 4-63 Stand-Alone Pulldown Menu

3. Click on the selected memory. The following window (Figure 4-64) will now appear:

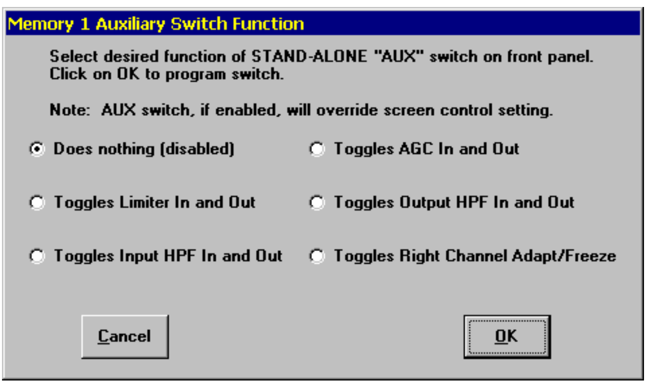


Figure 4-64 Auxiliary Switch Function Window

This window allows the function of the **AUX** switch on the PCAP II 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 PCAP II is busy storing the setup in the selected Stand-Alone memory. For complete instructions on operating the PCAP II external processor from the Stand-Alone memories, please consult Section 5.0.

4.8.5 Generating Setup Reports

The PCAP II Master Control program has a **Report-Generator** feature which allows hardcopy printouts of PCAP II 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 4-65) will be displayed:

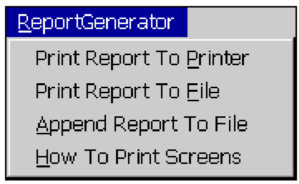


Figure 4-65 Report-Generator Pulldown Menu

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 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 (with paper) needs to be connected to your computer.

4.8.6 Getting Online Help

Context-sensitive online help is available to the PCAP II 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 4-66) will be displayed:

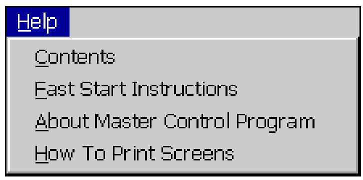


Figure 4-66 Help Pulldown Menu

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

The WinHelp utility is a separate Windows program. Once it has displayed the help text, it will remain on the screen as a separate window until it is closed by the user.

For further information on the WinHelp utility, please consult your Microsoft Windows User's Guide.

5. OPERATING PCAP II STAND-ALONE

Before the PCAP II external processor can be operated Stand-Alone, at least one of the Stand-Alone setup memories must be programmed using the procedure in Section 4.8.1.

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 2CH 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 an analog tape recorder to the rear panel LEFT (white) and RIGHT (red) ANALOG OUTPUTS RCA jacks if recording the processed audio is desired. Alternatively, connect a digital recorder 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, 3, and 4 Adapt/Freeze (ADAPT/FREEZE switch used for Filter1 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 CUSTOMIZATION OPTIONS

6.1 Configuring PCAP II for Different COM Ports

The PCAP II Master Control software is factory-configured to communicate with the external processor over RS232 communications port #2 (COM2). This is likely to work for most systems. However, in some cases where a serial mouse is connected to COM2 instead of COM1, or in general when the available COM port is not COM2, it may be necessary to reconfigure the software to operate on a different COM port as follows:

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

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

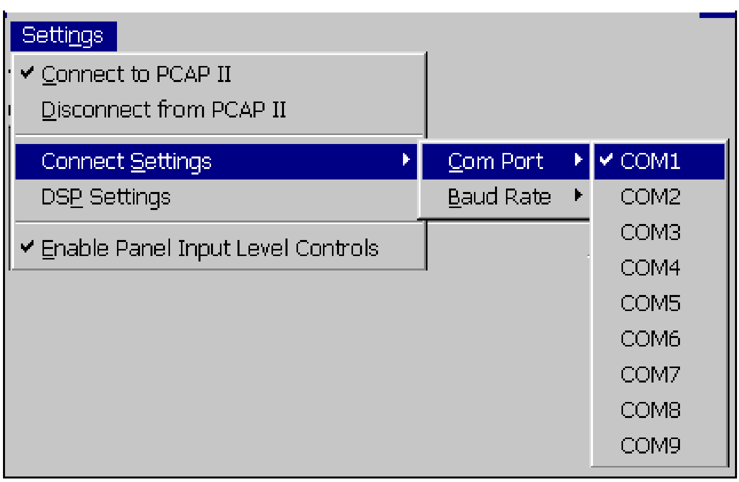


Figure 6-1 ComPort Selection Menu

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". If any error messages appear, then you have misidentified the COM port. Repeat steps

through 3 until the message "COMx is Open" appears. When it does, wait a few moments for the program to reset the external processor. If the Input Level bargraphs eventually respond to the input signal, then proper communication has been achieved.

4. Click on "File", then "Exit" to exit the PCAP II Master Control program. This will save the new COM port setting.

6.2 Configuring PCAP II for Different Baud Rates

This section is intended for advanced users, only.

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

Change the PCAP II 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 PCAP II 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-

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

Table 6.1 Baud Rate DIP Switch Settings

3 to desired baud rate as follows:

   d. Replace top cover and reconnect all cables.

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

   a. Make sure that the PCAP II 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.

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

d. When the pulldown menu appears, click on "BaudRate". The display will now appear as follows (Figure 6-2):

e. Click on the baud rate setting which corresponds to the PCAP II external processor DIP switch settings. If any error messages appear, recheck COM port setting, Baud Rate setting, connections, and DIP switch settings.

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

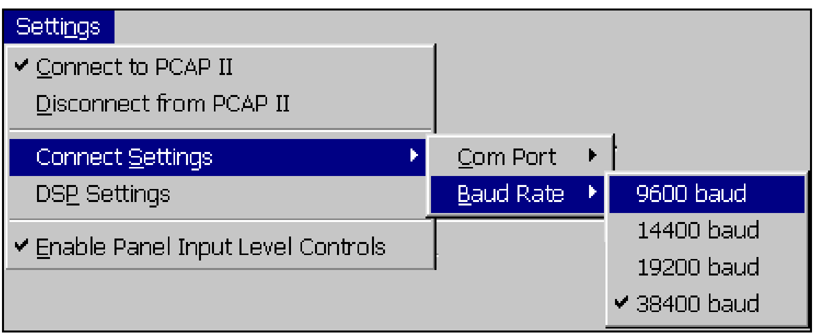


Figure 6-2 Baud Rate Selection Menu

3. After storing the new baud rate by exiting the PCAP II 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 Master Control icon to run the program.

c. Allow several seconds for the program to load and to initialize the external processor. The message "COMx is

Open" should appear in the Status box at the lower left corner of the PCAP II Master Control window.

d. Connect a line-level audio signal to either of the LEFT or the RIGHT ANALOG INPUTS RCA jacks and adjust the input level so that the tricolor level LEDs indicate GREEN.

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

6.3 Customizing Screen Colors

Many of the PCAP II Master Control program screen colors are set by the Windows environment. To change these colors, first open the **Control Panel** by clicking on the **Start** button on the **Taskbar** then clicking on the **Settings** menu option. There you will see the **Control Panel** icon. Double click on the **Display** icon and select the **Appearance** tab. In this window, you can select a **Scheme** from the list box and view the appearance of each, or you can set colors manually by selecting the item in the window example. If you do set colors manually, make sure that you click the **Save As** button to save your custom color scheme to a disk file. Finally, when you have all colors set the way that you like, click on **OK** to reset the Windows environment colors to these values.

For complete details on using the **Control Panel**, please refer to the Microsoft Windows User's Guide.

7. 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 II 's 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 32kHz, 44.1kHz, and 48kHz.

Digital Processing

Digital Filters

- 4096 tap adaptive/fixed filter. Splittable into two 2048 tap filters, one 2048 tap plus two 1024, or four 1024 tap filters.
- 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.

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 Microprocessor	•	TMS320C50 20 MIPS processors with 32k x 16 program RAM and 32k x 16 data (delay) RAM.
Computers Interface	•	Standard RS232 serial interface for digital filter control, coefficient transfer, and spectral analysis data. 9.6k to 38.4k baud transfer rate, adjustable. (38.4k default)

<u>Construction</u>

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 (or equivalent) CPU Processor PC with mouse, VGA, 32 MB RAM, CD-ROM,

		and 2.1 GB HD. Windows 95/98/NT/2000 required Color active matrix display recommended if notebook used.

<u>Signal Processing Functions</u>

Filter Configurations	•	Mono, stereo, stereo independent
	•	Max of five DSP stages mono (four at 16 kHz BW), Max of three DSP stages stereo (two at 16 kHz BW)
Adaptive Filters	•	One-channel (1CH) adaptive predictive deconvolution and two-channel (2CH) adaptive noise canceller.
Adjustable Filters	•	Lowpass, highpass, bandpass, bandstop, notch, slot, comb.
Special Filters	•	Hi-res graphic, spectral inverse filters (both 460 lines resolution and 60dB range).
	•	Limiter/Compressor/Expander
	•	Parametric Equalizer
Output Equalizers	•	20-band graphic and spectral graphic (115 lines resolution)
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