SOUNDSCOPE
USER'S MANUAL

**Speech Analysis Software
For the 21st Century**

GW Instruments, Inc.
THE COMPUTER INSTRUMENT COMPANY

Instruments

**Speech Analysis Software
For the 21st Century**

SoundScope

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User's Manual

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Instruments

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Chapter 1
Installation

Welcome to the wonderful world of SoundScope. We are extremely excited about this product and hope you can share in our enthusiasm. SoundScope is more than just 1 product, it is a core technology on which the following products are based:

SoundScope/16	*Base Package*	#GWI-SoS16
SoundScope/LC	*Novice Level System*	#GWI-Sos/LC
SoundScope/8	*Entry Level System*	#GWI-SoS8
SoundScope/16 Open	*C Programmer's Dream*	#GWI-Sos-C

SoundScope software fully supports the MacSpeech Lab I and II hardware for record, play and analysis. GWI is committed to the SoundScope platform and will continue to develop it throughout this decade and into the 21st Century.

TO GET STARTED

To get started, we recommend that you first Install the software and take a tour with *Chapter 2, Tutorial*. To learn about advanced features, please see *Chapter 3, Advanced Tutorial*, and *Chapter 4, Instrument Design*.

If you are working with a demonstration version of the software and want more information, please contact GW Instruments and order a copy of the SoundScope & SuperScope II Reference Manual, order #GWI-Demo-SS2-Ref. If you are interested in adding your own C source code to SoundScope and want more information, please order the SoundScope Open Programmer's Guide and ThinkC Interface Source Code, order #GWI-Demo-SS2-C.

COMPUTER REQUIREMENTS

SoundScope must be run on a machine with the following minimum requirements:

COMPUTER

SoundScope/16 and *SoundScope/16 Open* requires one Nubus slot and is therefore compatible with Macintosh II series and Quadra series computers. Note that the Macintosh IIsi requires a NuBus adapter card, available from Apple Computer.

SoundScope LC installs into a Macintosh LC, LCII or LCIII Processor Direct Slot.

SoundScope/8 attaches to the Modem port and is therefore compatible with Macintosh SE or later computers.

OPERATING SYSTEM

Apple System 7.x or System ≥6.0.7 is required.

MONITOR

Although it is not required, a color or grayscale monitor and graphics board capable of displaying 256 shades produces the best spectrograms.

HARD DISK

SoundScope requires a hard disk with at least 6 MB free.

RANDOM ACCESS MEMORY (RAM)

At least 4MB of RAM is required when running under System 6 Finder. At least 5MB is required when running under System 6 MultiFinder or System 7.0. More RAM is needed if your System Folder contains many extensions ("INITs"), or if you wish to record long utterances. In summary, 8MB of RAM is recommended. If you incur a memory error upon application launch, try setting the Memory Allocation field to 2.5MB or so (i.e. select the SoundScope application from the Finder and choose Get Info under File).

If you are running the SoundScope/16 Open system (used by C programmers) under Think C, 8MB of RAM is required, and 16MB is recommended.

SYSTEM COMPONENTS

The SoundScope product family includes the following systems:

SOUNDSCOPE/16   *BASIC PACKAGE*   *#GWI-SoS16*
◊ SoundScope/16 Software on floppy disk
◊ SoundScope User's Manual and SoundScope & SuperScope II Ref. Manual
◊ AudioMedia™ II 16-bit digitizer board, cables, AudioMedia software, and AudioMedia manuals. Technical support on AudioMedia products is available directly from Digidesign.
◊ RANE Microphone Amplifier and cables
◊ BOSE™ powered speakers and cables
◊ Microphone, cable and stand
◊ Eve™ security key

SOUNDSCOPE/LC   *Novice Level System*   *#GWI-SoS/LC*
◊ SoundScope/16 Software on floppy disk
◊ SoundScope User's Manual and SoundScope & SuperScope II Ref. Manual
◊ AudioMedia™ II 16-bit digitizer board for Macintosh LC, cables, AudioMedia software, and AudioMedia manuals. Technical support on AudioMedia products is available directly from Digidesign.
◊ RANE Microphone Amplifier and cables
◊ BOSE™ powered speakers and cables
◊ Microphone, cable and stand
◊ Eve™ security key

SOUNDSCOPE/8   *Entry Level System*   *#GWI-SoS8*
◊ SoundScope/8 Software on floppy disk. SoundScope/8 software is identical to SoundScope/16 software, except: it does not play/record to/from the 16bit digitizer board, it loads 16bit sound files with a maximum resolution of 12bits, it digitizes via the Sound Control Panel with a maximum resolution of 8bits, and it does not run Tasks.
◊ SoundScope User's Manual and SoundScope & SuperScope II Ref. Manual MacRecorder™ 8-bit digitizing microphone, MacRecorder software, and MacRecorder Manuals. Technical support on MacRecorder products is available directly from MacroMedia.

SOUNDSCOPE/16 OPEN   *C Programmer's Dream*   *#GWI-SoS-C*
◊ SoundScope/16, described above
◊ ≥40MB External Hard Disk
◊ SoundScope ThinkC Object Code & ThinkC Interface Source Code
◊ SoundScope Open Programmer's Documentation

SOUNDSCOPE/8 FIVE-PACK   *Multiple Systems*   *#GWI-SoS8-5x*
◊ Five additional copies of SoundScope/8 for a multi-station installation. At least
   one base system (e.g. Part #GWI-SoS8) must be purchased in order to qualify for
   the Five-Pack purchase. Five-Packs cannot be broken (e.g. you cannot buy 3 for
   60% of the Five-Pack price).

SOUNDSCOPE/16 FIVE-PACK   *Multiple Systems*   *#GWI-SoS16-5x*
◊ SoundScope/16, described above
◊ Four additional SoundScope/16 security keys for a multi-station installation. At
   least one base system (e.g. Part #GWI-SoS16) must be purchased in order to
   qualify for the Five-Pack purchase. Five-Packs cannot be broken (e.g. you
   cannot buy 3 for 60% of the Five-Pack price).

SOFTWARE INSTALLATION

SoundScope is shipped to you in a compressed "archive" form. The following installation procedure will create a new folder with the enclosed software.

#1 Insert the **GREEN Application Disk** and double-click on its .sea file.

#2 Press Continue when the Introduction dialog appears, as shown to the right.

#3 When the File Save dialog appears, specify a location for the new software (e.g. the top level of your hard disk). Press the Save button to commence with the installation procedure.

#4 When it's done with the first installment, a dialog will appear, as illustrated to the right. Press Quit to continue.

#5 Drag the GREEN Application Disk into the trash (to eject it).

#6 Insert the **GWI Application Disk #1** and double-click on it's "GWI Installer" file.

#7 When the Introduction dialog appears, as shown to the right, click a key to continue.

#8 When the File Save dialog appears and asks you for the destination of the next installment; locate the folder that was just created, enter it, and then Press Save.

#9 A progress dialog will appear, as illustrated to the right. As the decompression commences, it will ask you to insert GWI Application Disk #2, Disk #3, etc. When it's done, it will say "Install Successful".
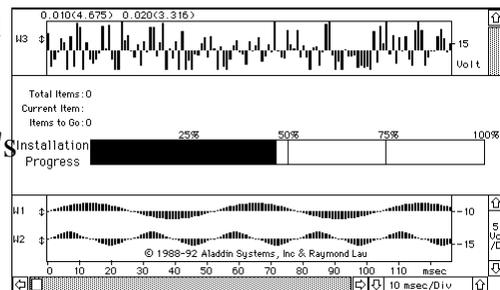
#10 The computer will then ask you to insert GWI Application Disk #1. After doing this, drag its icon into the trash.

Instruments   **1 - 5**

#11 LARGE-SCREEN MONITORS

If you are using a monitor that is 16" or greater (measured along the diagonal), SoundScope may run out of memory when computing large spectrogram or LPC history plots. You can prevent this problem by allocating additional memory for SoundScope, as explained below.

- Locate the "SoundScope/xx" or "GWI Application" file in the "SoundScope Application" folder (where xx is 8, 16, or Demo). Click the icon once to select it, and then choose Get Info under the File menu.
- In the lower right corner, type in the number "4000" (or larger). This number is the maximum amount of memory (in bytes) that the application will be able to use while it is running.
- Click the close box in the upper left corner.

#12 SYSTEM EXTENSIONS AND CONTROL PANELS

Several system Extension and Control Panel files are included with SoundScope as shown in the following table. These files can be found within the "System Extensions" and "Control Panels" folders, inside the "Goodies" folder at the same level as the SoundScope application.

Extension	Manufacturer	Purpose
SSProtect INIT	GWI	SoundScope/16 copy protection
Digidesign Sound Input	DigiDesign	16bit AudioMedia board driver
DigiSystem INIT	DigiDesign	16bit AudioMedia board driver
MacRecorder® Driver	MacroMedia	8bit MacRecorder driver
XLink Extensions	CEL	HyperCard XCMD Interface
XLink Manager	CEL	HyperCard XCMD Interface

Control Panel	Manufacturer	Purpose
XLink Setup	CEL	HyperCard XCMD Interface
Serial Switch	Apple	Fast Computer serial switch

These files add features to your computer's operating system, and are necessary in some applications. For example, in order to run SoundScope/16 software, the *SSProtect INIT* Extension must be installed. Each Extension consumes some memory; therefore you should only install those that are needed. To install an Extension, place its file into your System folder (and inside the "Extensions" folder, if running under System 7). To install a Control Panel, place its file into your System folder (and inside the "Control Panels" folder, if running under System 7). To activate an Extension or Control Panel, Restart the computer.

- If you have SoundScope/LC, SoundScope/16, or SoundScope/16 Open, install the *SSProtect INIT* Extension.
- If you have a 16bit SoundScope/16 or SoundScope/LC digitizer board, install the *Digidesign Sound Input* and *DigiSystem INIT* Extensions.
- If you working with the demonstration version of SoundScope, please install the following Extensions: *XLink Extensions*, *XLink Manager*. Also install the *XLink Setup* Control Panel.
- If you have a SoundScope/8 8bit digitizer, install the *MacRecorder® Driver* Extension.
- If you want to run HyperCard XCMDs (e.g. in the TriVowelgram instrument), install the *XLink Extensions* and *XLink Manager* Extensions; and then install the *XLink Setup* Control Panel.
- If you want to run an 8bit digitizer, or a MacSpeech Lab I digitizer with a Macintosh IIfx or Quadra computer, install the *Serial Switch* Control Panel, double-click on it, and then choose Compatible.

#13 **SECURITY KEY**
In order to run SoundScope/LC, SoundScope/16 or SoundScope/16 Open, the Eve security hardware key must be attached to your Macintosh. Locate the key (a small gray box with a short cable) and plug the key into any available Apple Desktop Bus (ADB) port. Most Macintosh computers have 2 ADB ports on the back of the computer, and two on the keyboard. ADB devices can be "daisy-chained", such as plugging the mouse into the Eve security key, which is in turn plugged into the keyboard. SoundScope/8 does not require this key.
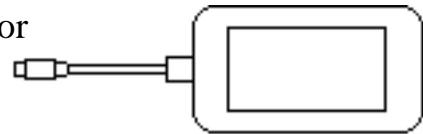
#14 Restart your computer to activate the Extensions and Control Panels, if you have not already done so.

#15 Double-click on the "GWI Application" file (in the recently installed folder). Please insert the GREEN Application Disk when requested. After the application appears with an empty front panel, it will ask you to exit (to reset the desktop) and re-enter by double-clicking on the application file.

#16 Create a new folder outside the SuperScope II application folder and label it "SS2 Instruments & Data". Please keep your personal instrument and data files in this folder, separate from the SuperScope II files, to reduce the chance of deleting an important file during a software update.

You have now completed the installation procedure -- congratulations! If you want a tour of the software, please proceed to *Chapter 2, Tutorial*.

SOUNDSCOPE/16 AND SOUNDSCOPE/LC HARDWARE INSTALLATION

This section describes how to install the SoundScope/16 and SoundScope/LC hardware, as illustrated in the following figure. The recommended wiring is not mandatory since the AudioMedia II board can attach to any standard audio source or input with "line-level" signals and RCA phono connectors (e.g. tape deck, CD player, stereo amplifier, DAT tape recorder).

DAT Recorder

Mic

AudioMedia II

Macintosh Computer

in
out DIGITAL AUDIO TAPE
L
R ANALOG INPUT
L
R ANALOG OUTPUT

Mic In
Line Out
Power In

Amplifier

Red
Black

Red
Black

Red
Black

Powered Speakers

Power In
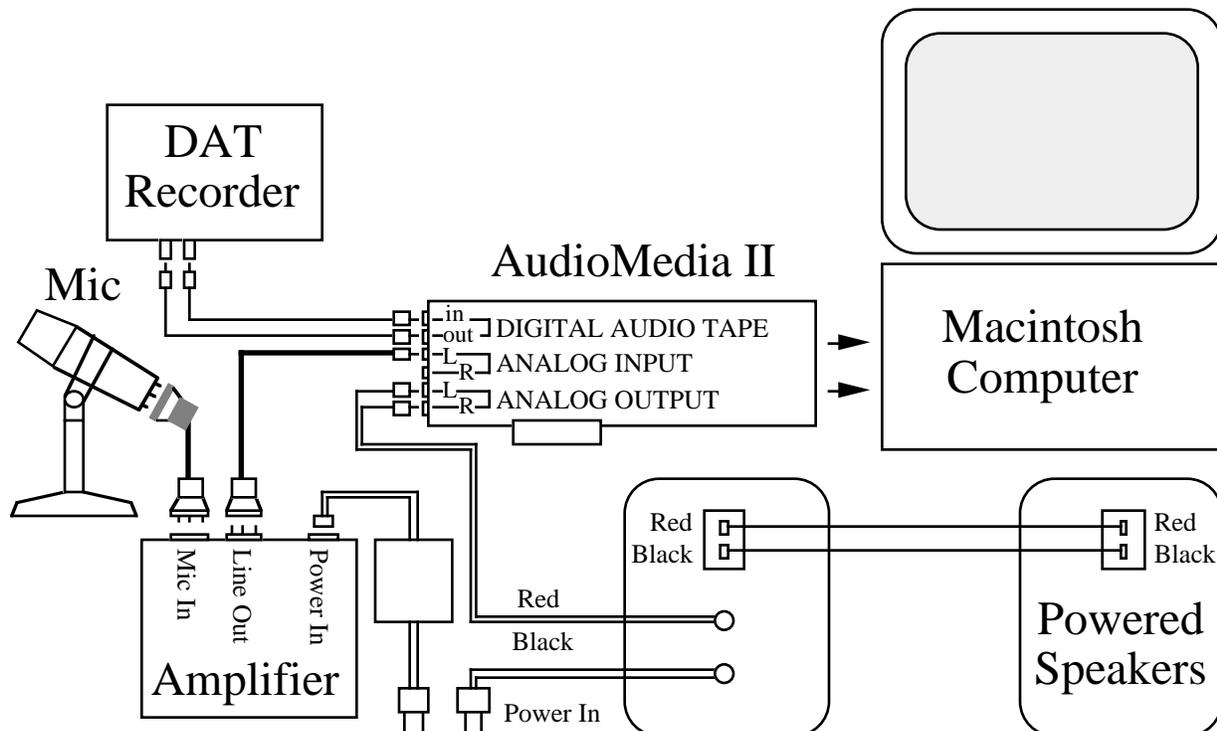
#1   Before beginning, select Shutdown from the Finder's Special menu, and then turn off the power switches for your Macintosh and all peripherals. It may be wise to unplug the power cord as well.

#2   Locate the 16-bit digitizer board, in the AudioMedia box.

If you are installing the board into a Macintosh IIsi: make sure that you have a Nubus Adapter Card (available from Apple Computer). Follow the instructions in the "Installing an expansion card" section of the Getting Started With Your Macintosh IIsi manual, then skip ahead to Step #10 Bose™ Powered speakers.

If you are installing a SoundScope/LC board into a Macintosh LC, follow the board installation instructions in your Macintosh LC manual, and then skip ahead to Step #10 Bose™ Powered speakers.

#3   Remove the top cover. There are several NuBus connectors (often called "slots") on the circuit board, aligned with removable panels (or empty holes) along the back side. The slots are numbered according to the following scheme, proceeding from left to right:

1 through 6	Macintosh II, IIx, IIfx
1 through 3	Macintosh IIcx
4 through 6	Macintosh IIci

#4   Choose an empty slot for your digitizer and remove the plastic or metal panel (if any). For easy reference, write the slot number here: ___ .

#5   Touch the Macintosh power supply box (the shiny metal object in the corner) to discharge any static electricity that your body might have accumulated.

#6   Unwrap the digitizer board, carefully handling it by the edges only.

#7   Orient the board so that the RCA phono jacks are facing the rear of the computer and the NuBus connector on the board is aligned with the connector on the computer.

#8   Carefully press down on the top edge of the digitizer, slowly sliding it into the NuBus slot. If it does not insert fairly easily, take it out and try again. As you insert the board, make sure its rear I/O panel sits well into the bracket at the rear of the computer, and snaps into the small receptacle at the base of this bracket. The RCA phono jacks should protrude slightly through the slot's I/O window at the rear of the computer.

#9   Replace the Macintosh cover. Do not power on the computer system yet.

#10 **BOSE™ POWERED SPEAKERS**
- Unpack both speakers. The back of each speaker identifies whether it is for the left or right channel.
- Locate the 8-foot black speaker wire (this wire is composed of two insulated wires glued together, with exposed wire leads at each end). Notice that one of the insulated wires bears a red stripe.
- Connect the two speakers together with the speaker wire. For both speakers, the wire with the red stripe connects to the red receptacle, the other wire to the black receptacle.
- Any time you plug anything into or out of these audio jacks, make sure the computer and your audio equipment are powered off. Connect the gray Audio Input wire from the left speaker to the two stereo outputs at the base of the digitizer I/O panel, with the red (Right channel) plug on the bottom.
- Finally, connect the power cord from the left speaker to a power outlet.

#11 **MICROPHONE AND AMPLIFIER**
- Make sure the computer and your audio equipment are powered off.
- Unpack the microphone and microphone cable.
- Insert the microphone cable's three-prong XLR plug into the base of the microphone, and insert the other end of the cable into the Amplifier's "Mic Input".
- Insert the amplifier cable's three-prong XLR plug into the Amplifier's "Line Out" jack, and insert the other end of the cable into the Left Audio Input connector at the digitizer's I/O panel (3rd RCA connector from top).
- Unpack the microphone stand and slide the microphone into the receptacle at the top of the stand.

SOUNDSCOPE/8 HARDWARE INSTALLATION

This section describes how to install the SoundScope/8 hardware, as illustrated in the figure below. You may skip this section if you purchased SoundScope/8 without the 8-bit digitizer. SoundScope/8 software records from the internal microphone included with many Macintosh computers, and from MacSpeech Lab I & II hardware.

MacRecorder — MODEM or PRINTER port — Macintosh Computer
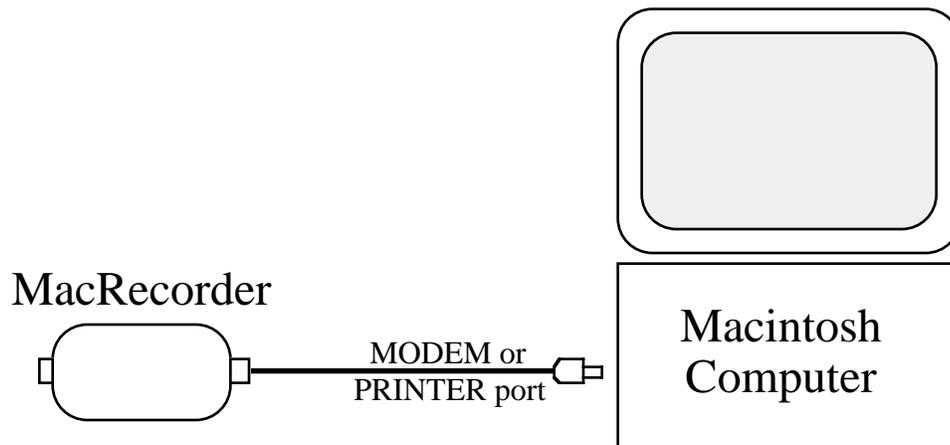
#1   Locate the 8-bit digitizer, in the MacRecorder box. It is a small gray box, slightly larger than a mouse, with a six-foot cord affixed to one end.

#2   Plug the 8-bit digitizer's connector into the MODEM or PRINTER port on the back of your computer. The arrow imprinted on the plug should be facing up as it is inserted.

SOUNDSCOPE/8 LIMITATIONS
SoundScope/8 software is identical to SoundScope/16 software except for the following:

- SoundScope/8 does not play/record to/from the 16bit digitizer board.
- SoundScope/8 loads 16bit sound files with a maximum resolution of 12bits.
- SoundScope/8 digitizes via the Sound Control Panel with a maximum resolution of 8bits.
- SoundScope/8 does not run Tasks.

OUR MISSION FOR THE 90'S

The world of computing is currently evolving at lightning speed with computers the size of notebooks doing what mainframes did only 10 years ago. And we believe this spectacular growth will continue at a 5%/month increase in price-to-performance-ratio well into the 21st Century (it's exciting). This, coupled with the fact that the manufacturing cost of software is negligible, has led us to believe that companies who invest heavily in developing ultra advanced software will be the winners at the turn of the century.

In 1989, we undertook the challenge of developing a highly advanced software product family. This involved over 30 developers and three years of painstaking effort before it produced it's first fruit. SoundScope is more than one computer instrument -- it is an environment in which the end user can design their own instruments. We believe that the next revolution in computing will involve the end user taking on the role of the developer; designing, prototyping, learning, and implementing new and unique methods in the field. Subsequently the SoundScope product family allows the end user to "program" with an easy to use mouse/dialog box user interface.

Our Company Mission is to change the way scientists, engineers and medical researchers work in the laboratory by developing computer based instrumentation which is faster, stronger, more accurate, more capable, more flexible, less expensive and easier to use than the traditional benchtop counterparts. We hope this product made a dent in this quest and wish to hear from you if you have ideas concerning future releases.

IDEAS FOR NEW FEATURES

If you have any suggestions for improvements, please phone or write us *Attention: New Products*. We develop products for you, the user, and want your feedback.

APPLICATION NOTES

GW Instruments collects 2 page summaries of common instruments. If you would like to write about your instrument, or would like to read about someone else's, please call or phone GW Instruments, attention *Application Note Program Coordinator*.

BIG BAD BUGS

To learn about dead bugs, known bugs and areas under development please open file (choose Open under Journal) "!Development Status.c" in the "Programmer's Notes" folder inside the "Goodies" folder. This file expects tabs every 4 characters (i.e. choose Options under Journal and then set the Tabs to "4" characters). To report a bug, please:

#1 If the bug causes an Error Alert to appear (i.e. one with a Bug Report button), press the magic button to open the Bug Report dialog, pictured to the right. Alternatively, if the bug does not cause an Error Alert to appear, press *Option* 'y', at any time, to invoke the Bug Report dialog.

> **Bug Report Bureau**
>
> Dear End User:
>
> If you found a bug and would like to inform the manufacturer, please proceed with the following steps:
>
> #1 Press OK to exit this dialog.
> #2 Press OK to COPY this message to the clipboard.
> #3 Exit this application and PASTE the clipboard contents (which include these instructions) to a word processor. If your bug is so awful it doesn't allow you to do this, you can restart your computer, launch this application, and press OPTION 'y' to get to this point, and proceed from there.
> #4 Fill out the form below.
> #5 Delete this letter and print your report in a small fixed character
>
> [OK]

#2 Read the displayed instructions.
#3 Press OK to exit the dialog.
#4 Press OK to copy the bug report information to the clipboard.
#5 Paste the clipboard text into a new word processor window.
#6 Describe the sequence of steps required to reproduce the bug (this is crucial). If we cannot see the problem at GW Instruments, we probably cannot fix it.
#7 Delete the Dear End User letter at the top of the file, to save trees.
#8 <u>Print in a small font</u> and send to GW Instruments via fax, mail, or electronic mail. If you phone us, please do so while in front of your computer, if possible.
#9 If your bug is real mean, please refer to Tip #413 in Appendix E of the SoundScope & SuperScope II Reference Manual.
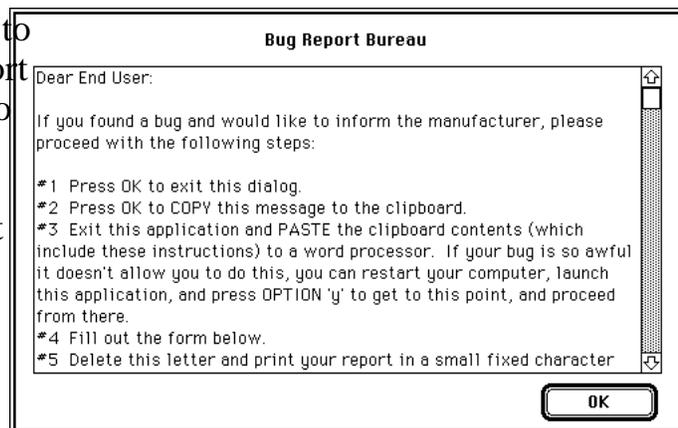
IF YOU ARE NEW TO THE MACINTOSH

If you are new to the Macintosh, we recommend that you keep a good reference at your side. An outstanding one is *The Macintosh Bible*, 3rd Edition, by Sharon Zardetto (Goldstein & Blair, ISBN 0-940235-11-0). Another wonderful reference is *Everything You Wanted To Know About the Mac*, by Larry Hanson (Hayden, ISBN 0-672-30142-3).

SOFTWARE UPDATES

To receive the latest version of the SoundScope software, please contact GW Instruments. Updates are free during the first 12months, and are then a small fee to cover shipping and handling costs. To be fair, the update fee is waived if triggered by a software bug (i.e. it's our fault). Problems due to new computers and new system software do not qualify as "bugs".

SoundScope/8 Software Update............................*Part #GWI-SoS8-Upd*
SoundScope/16 Software Update.......................*Part #GWI-SoS16-Upd*
SoundScope Open Software Update..................*Part #GWI-SoS-C-Upd*

CUSTOMER SERVICE

GW Instruments is committed to providing high-quality service to its customers. For technical support, please contact GW Instruments by phone, fax, AppleLink, Internet or in writing. If you phone us, please do so while in front of your computer, if possible. Technical Support is open between 9 and 5 Eastern Standard Time.

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35 Medford Street   Fax: 617/625-1322
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United States   Internet: D∅268@AppleLink.Apple.COM

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Chapter 2
Tutorial

This chapter describes the main features of SoundScope in tutorial fashion. From here, you can take advantage of the familiar Macintosh interface to explore additional SoundScope capabilities.

SoundScope was created to help you record, analyze, manipulate, and play speech and other sounds. Both recorded sounds and the result of sound calculations are stored in the computer as **waves**. Specific time values in a wave may be indicated by **markers** that can be moved with the mouse. The waves and markers are shown on the screen in rectangular areas called **displays**. The number, size, type and layout of displays is completely controlled by the user. There are four types of displays, listed below.

- **Wave Plot** shows the timewave (voltage as a function of time).
- **Analysis** shows a Spectrogram, LPC History, Horizontal Spectral Slice, Long-term Average Spectrum, Fo (fundamental frequency), Jitter, Shimmer, HNR (Harmonic-to-noise ratio), Envelope, Energy, Zero Crossing rate, Spline, or LPC Residual.
- **Snapshot** shows a close-up of the timewave at a specific point in time, showing either a 10, 20, 50 or 100 msec window, or an "instantaneous" FFT (Fast Fourier Transform), LPC (Linear Predictive Coding) or Cepstrum.
- **XY Plot** shows a wave plotted against another wave (e.g. F1 vs. F2), rather than as a function of time.

SoundScope also features integrated text editors called **journals**. You can type notes directly into a journal, using it like a simple word processor. To record numerical values, simply select the appropriate menu command and click the mouse on specific points of interest in any wave. SoundScope can automatically pick out peaks or other key parameters, and log them to the journal at the click of a button.

Waves, markers, displays, and journals can be combined like building blocks to create software **instruments**. A few pre-defined instruments have been included with SoundScope. You can use these as is, customize them to suit your specific needs, or build new instruments from scratch.

In fact, SoundScope is aimed at two different audiences, instrument <u>users</u> and instrument <u>designers</u>. Instrument users can begin work right away, using software instruments that were created by GWI or by colleagues. Instrument designers will

invest more time learning how to customize SoundScope, to create new instruments for their own use and for use by others.

This chapter guides you through a "hands-on" interaction with SoundScope. The Tour is divided into two chapters, each of which can be reviewed separately. In the following instructions, each step is preceded by a bullet symbol (•). It is important to not miss a step, since many depend on previous steps. SoundScope makes extensive use of hierarchical menus (i.e. submenus) to provide direct access to waves, displays, journals and other objects. This manual uses the ▸ character to indicate a submenu. For example, "Select Load Data ▸ A from the Wave menu" is shorthand for "Pull down the Wave menu, drag down to the Load Data submenu, drag across to 'A', and then release the mouse button to select it."

PART 1: GETTING STARTED

SoundScope was designed to handle two channels of sound, labelled A and B. With the appropriate hardware, SoundScope can record and play these channels simultaneously (in stereo). For many applications, one channel of data is sufficient. In the first part of this tutorial, we will look at an instrument designed for a single channel. To begin, make sure your computer is on and that you have installed the SoundScope software, as described in *Chapter 1, Installation.*

• Inside the "SoundScope Application", folder on your hard disk, locate the instrument file called "1 Channel Analyzer". Double-click on the file's icon to launch SoundScope with this instrument. (Or, click once on the icon and select Open from the Finder's File menu.)

Note: You will get an error message if there is not enough memory (RAM) to launch SoundScope. As shown in it's Get Info box (select from the Finder's File menu), SoundScope prefers 3 MB (3000 K) or more, but will run with as little as 2000 allocated. To determine if your machine has enough memory, select About this Macintosh (or About the Finder) from the Finder's Apple (⌘) menu. If you are running MultiFinder or System 7, quit out of other applications to increase the memory available. Please see *Chapter 1, Installation* for a detailed discussion of memory issues.

After a few seconds, SoundScope's menubar will appear at the top of the screen, and the instrument will then be loaded. The screen will look something like this:

Marker "A1"   Marker "A2"

Analysis
Display
"Calc_D
"

Wave Plot
Display
"Time_D"

Snapshot
Display
"Snap_D"

Journal
"Notes"

Marker "A1"   Marker "A2"   Wave "A" will appear here
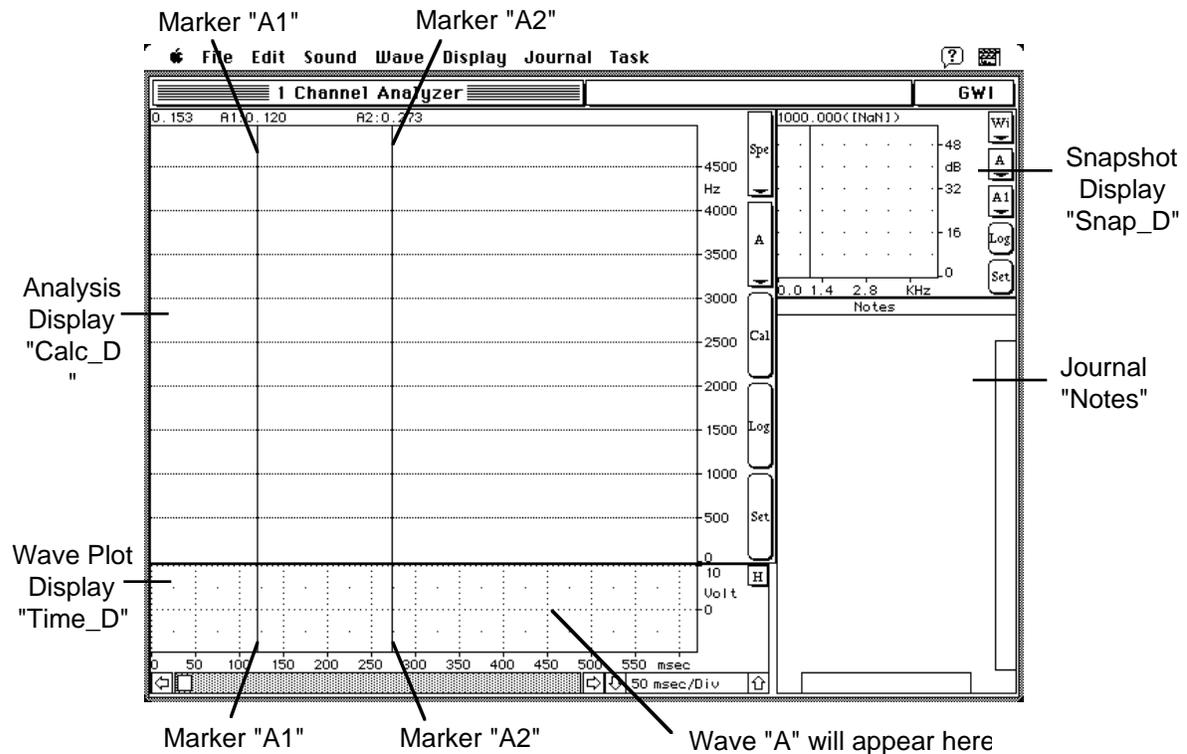
This instrument uses SoundScope's "Short" menubar, which provides every command required by the typical instrument user. Additional commands for instrument designers are available in the "Full" menubar, available from Choose Menubar in the Edit menu. Let's begin with something tangible, recording and playing sounds. For this tutorial, we will take them in reverse order.

LOADING SOUND DATA FROM DISK
To get quick results, we will start with a pre-recorded sound that was
saved to disk.

- Load a sound into channel A by selecting Load Data
  ▸ A from the Wave menu.

- In the standard open file dialog that appears, select
  the "were away yr ago (22.2)" file in the "Waves" folder.
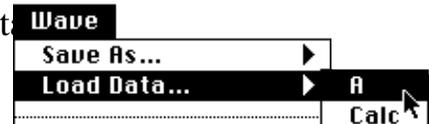
This soundwave will appear in the Wave Plot display named Time_D, which runs along the bottom of the screen. To provide unique but related names for displays and the waves inside, display names have "_D" at the end.

Wave Plot Display "Time_D"
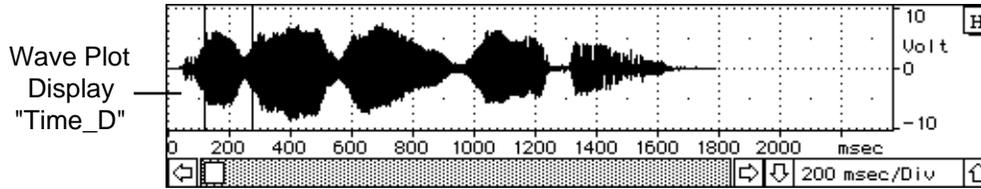
HARDWARE SETUP

In order to use the play and record commands, SoundScope must know which sound hardware is connected to the Macintosh. SoundScope does its best to determine this information each time it is launched; however, we will double-check to make sure that it is correct. If you are using the demo version of SoundScope, you can play pre-recorded sound through the speaker in your Macintosh.

- Select Setup from the Sound menu. A dialog will appear, as shown to the right.
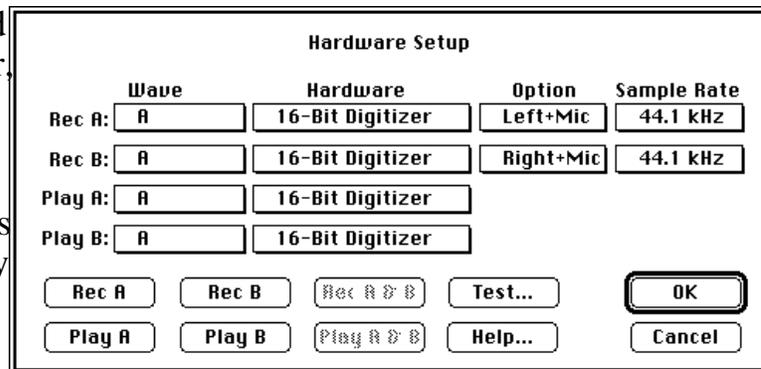
- Under the title Hardware, make sure the pop-up menus (Rec A, Rec B, Play A, Play B) show the name of your hardware: 16-Bit Digitizer, Apple (Modem, Printer, or Internal), MacSpeech Lab II, or MacSpeech Lab I (SoundScope/12). If you want to play through the internal speaker, select the Internal Speaker Hardware (this is handy if you are working with the demonstration version of the software). If you want to record from the microphone selected in the *Sound* Control Panel, select Apple with the Internal Option. If you want to record from an 8bit MacRecorder, select Apple with the Modem or Printer Option (depending on which port the MacRecorder is connected to).

  The 16-Bit Digitizer Hardware option pertains to an AudioMedia I board, an AudioMedia II board, or a MediaTime board. The MediaTime board (available from RasterOps Corporation for ~$2000) includes a 24bit video circuit, a real time frame grabber, two 16bit record channels, and two 16bit play channels -- it is truly amazing.

- Please set the Sample Rate and Option Pop-up menus as desired. For more information on these settings, see *Chapter 6, Front Panel*.

- Since you have already loaded sound data into channel A, test the hardware by clicking on the Play A button in the lower left corner.

- If you could clearly hear the words "We were away a year ago", then everything is fine. If not, verify that the hardware is correctly installed, and that the external volume control (if any) is set loud enough. If you hit "This sample rate is not supported by your Hardware", then you can either play a sound that you record with your hardware, or choose a wave in the "Wave" folder that is more compatible.

- Simply click the OK button to exit the dialog box.

PLAYING A SOUND
Although sounds can be played from the Hardware Setup dialog, menu commands (and their command-key equivalents) are usually easier.

- Select Play A from the Sound menu. You should again hear "We were away a year ago".

You can also play just a portion of the sound.

- Click once inside Time_D (at the bottom of the screen) to make it active.

- Drag with the mouse over any portion of the wave. The selected portion will be highlighted.

- Select Play Selected from the Sound menu.

RECORDING A SOUND
Having chosen the appropriate hardware, you can now record a sound to replace the current contents of the timewave. If you are using the demo version of SoundScope, you cannot record new sounds, and subsequently should skip ahead to *Saving Sound Data* on the next Page.

- Select Record A from the Sound menu. Begin speaking into the microphone immediately after the mouse button is released. (The cursor will change into the standard wristwatch icon when SoundScope begins to record data.) SoundScope will record for two seconds, and then update Time_D with the new wave.

- Select Play A from the Sound menu. If the sound is too soft, try holding the microphone closer to your mouth and recording again. (If you have SoundScope/8, SoundScope/12, or the MacSpeech Lab I or II hardware, adjust the hardware volume control and record again.)

- Select Controls from the Sound menu. Change the length of recording to one second by typing "1" into the Duration text box.

- Exit the dialog and record a new sound 1 second in duration.
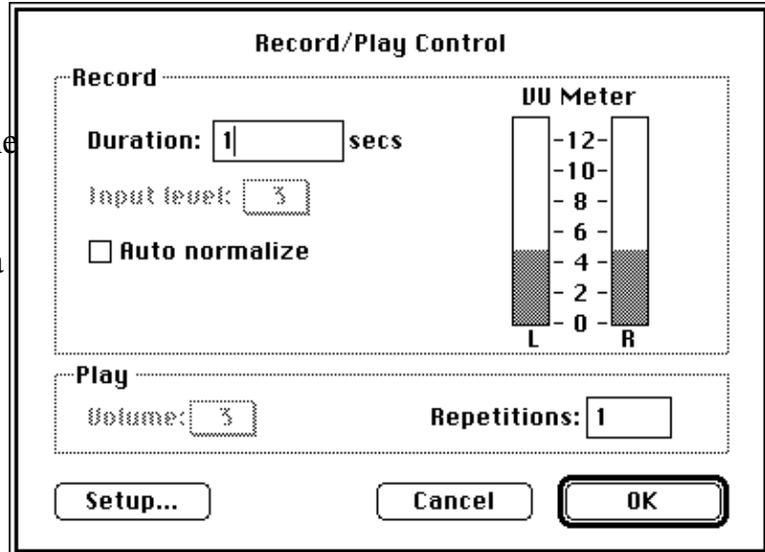
SAVING SOUND DATA
Before proceeding, save this sound to a new file.

Record/Play Control

Record

Duration: 1   secs

Input level: 3

☐ Auto normalize

VU Meter
-12-
-10-
- 8 -
- 6 -
- 4 -
- 2 -
- 0 -
L   R

Play

Volume: 3   Repetitions: 1

[Setup...]   [Cancel]   [OK]

- Select Save As ▶ A from the Wave menu.

- Type "first wave" into the standard file dialog, and click on the Save button. You may have noticed that the files names of the supplied waves are suffixed with the sample rate (e.g. "(22.1)" means 22.1Ksamples/sec). This helps distinguish wave files from instrument files and is useful when working with different sample rates.

SAVING THE HARDWARE SETUP
To preserve changes in the hardware setup, save the instrument to disk.

- Select Save As from the File menu.

- Type "my 1 Ch Analyzer" into the standard file dialog, and click on the Save button.

That concludes Part 1 of the tour. Stay tuned for a review of all displays that are on the screen.

PART 2: EXPLORING THE FRONT PANEL

This section of the tour describes each of the displays that you see on the screen. In SoundScope, a screen is like an instrument front panel. Before starting, reload the wave used in Part 1.

* Select Load Data ▶ A from the Wave menu, and choose "were away yr ago (22.2)" in the "Waves" folder.

The **Wave Plot** display, named Time_D, shows a plot of the wave amplitude (in volts) as a function of time (in milliseconds). Sound data varies in amplitude from -10 volts to +10 volts. Depending on the size of your screen and the software instrument that you select, the Wave Plot display may not show the entire range.
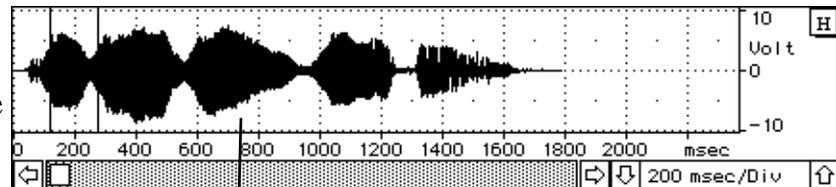
SCROLLING AND CHANGING SCALE

If only a portion of the timewave is visible in the display, use the scroll bar to move around:

* Click or press on the right and left arrows for precise movement, drag the scroll box to move to a specific location, or click on either side of the scroll box to move by about 75% of a screenful.

To change the amount of time that is displayed, change the horizontal scale (milliseconds per division):

* Click the down timescale arrow several times to zoom in on the timewave. Notice that the msec/Div number decreases each time the left time scale arrow is clicked. Then click the up timescale arrow until you can see the entire waveform.

* As a shortcut, click on the H button in the upper right hand corner of this display. SoundScope will automatically set the horizontal scale so that the entire waveform fits exactly in the display.

MOVING MARKERS
Specific time values in a wave may be indicated by **markers** that can be moved with the mouse. Two markers, A1 and A2, are currently visible in both Time_D and Calc_D above it. Each marker can be moved independently of the other, and will stay put until it is moved again.

- Click once inside Time_D (at the bottom of the screen) to make it active.

- Move the mouse pointer to the left-most marker, A1. When the mouse pointer lies directly on the marker, hold down the *Option* key (which changes the cursor to ⬌), press the mouse button and drag the marker about half an inch in either direction. Release the *Option* key and the mouse button. If you have trouble moving the marker, make sure you are placing the cursor exactly on top of the marker in Time_D. Then press the mouse button and hold it down as you move the mouse to the right or left.

THE MARKER LABEL AREA
The row of text and numbers at the top of Calc_D is called the marker label area, and shows information pertaining to marker positions. In this instrument, the marker names and time positions within the wave (in seconds) are shown directly above each marker. The time difference is also shown, on the far left side of the marker label area.

- In Time_D, move markers A1 and A2 in turn and notice how the numbers in the marker label area reflect the positions of the markers.

Because the SoundScope front panel is completely customizable, an instrument designer could show a marker label line directly above Time_D instead of or in addition to the one above Calc_D.

Our **Analysis Display**, named Calc_D, shows the result of a calculation performed on an entire wave, a segment of a wave (the portion of a wave between two markers), or the currently selected portion of a wave (selected by the mouse; will be highlighted). SoundScope puts a complete set of analysis routines at your fingertips, with convenient controls on the right-hand side of the display, the analysis option pop-up menu, source wave pop-up menu, Calc button, Log button, and Setup button. This display takes the source wave (or segment of a wave) as input, computes the specified analysis option, produces a result wave and displays that wave. The Calc button recalculates the specified analysis. The Log button performs actions such as logging data to a journal, or executing a task. The Setup button lets you alter the parameters of each analysis option, and select the events invoked by the Log button.

Analysis Options

Source Wave

Calc Button

Log Button

Setup Button
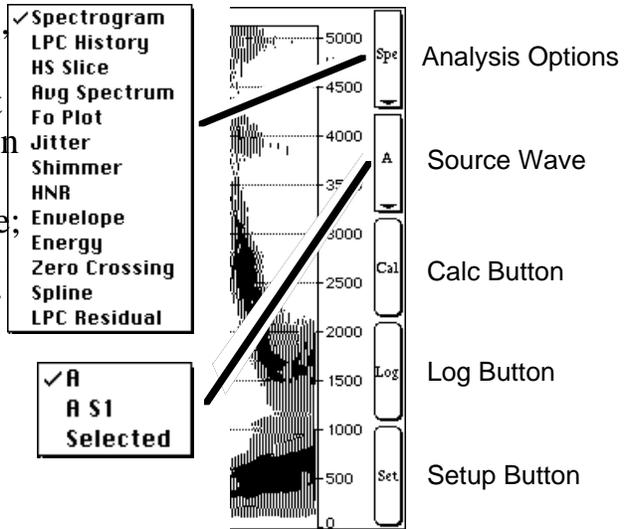
PRODUCING A SPECTROGRAM
As shown in the analysis pop-up in the upper right corner, Calc_D is currently set to compute a spectrogram ("Spe").

- Click the Calc button (the middle control on the right side). A spectrogram will be drawn from left to right, as illustrated to the right.

The spectrogram shows time along the horizontal axis and frequency along the vertical axis. The relative magnitude of the frequency components at each time is indicated by the darkness of the spectrogram. In this instrument, the time axis of Calc_D is **linked** to the time axis in Time_D below it. Scrolling or changing the scale of Time_D will also change Calc_D.
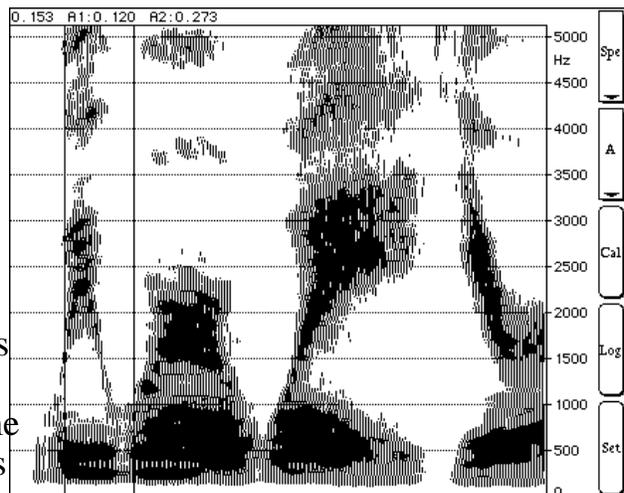
- To see a portion of the spectrogram in greater detail, click the time scale down arrow a few times. Scroll the display to an interesting part of the sound, and click the Calc button.

- Change the scale to 10 msec per division and click the Calc button again. Notice that the spectrogram is very rough, computed in discrete "blocks" or **frames**. This parameter can be adjusted, as described below.

CHANGING SPECTROGRAM SETUP

SoundScope gives you direct access to the key parameters behind each calculation. A spectrogram is computed one frame at a time. The amount the frame is advanced can be reduced to create a smoother plot.

- Click the Setup button on the right side of the analysis display. A dialog will appear as shown to the right.

- Enter "0.5" in the Frame advance textbox. Click the OK button in the lower right corner to confirm this setting.

- Click the Calc button to plot a new spectrogram. Notice that the display is much smoother.

Spectrogram Setup

Analysis
FFT Points: 256
Filter: 300 Hz (5ms)
Darkness: Medium | Max: 47 dB Min: 20 dB
Display range: 0.000 to 5.000 kHz
Frame advance: 2.000 ms (0.1...50)
☒ Use 6 dB pre-emphasis
☐ After calculating, run task: ◇

Log
☐ Run task: ◇

[Help...] [Color...] [Options...] [Cancel] [OK]

The Spectrogram Setup dialog contains many pop-up menus, radio buttons and edit fields. One especially important control that you may have noticed is the Display range pop-up, which specifies the lower and upper frequency bounds of the spectrogram display.

CHANGING THE ANALYSIS OPTION

Calc_D contains many analysis options, of which the spectrogram is merely one example.

- Select Fo Plot from the analysis pop-up menu and then Click the Calc button. After several moments, you will see a plot of the fundamental frequency (in Hz). Remember that the analysis display is setup to analyze the portion of soundwave A that is visible in the time display.

```
✓Spectrogram
 LPC History
 HS Slice
 Avg Spectrum
 Fo Plot
 Jitter
 Shimmer
 HNR
 Envelope
 Energy
 Zero Crossing
 Spline
 LPC Residual
```

- Reset the horizontal scale by clicking the H button in the time display, then change the analysis option back to Spectrogram. Click Calc to see the spectrogram again.

As you may have noticed while changing back and forth between Spectrogram and Fo Plot, many other options are available in the analysis pop-up. Each is described in detail in *Chapter 6, Front Panel*.

A **Snapshot display** shows information pertaining to a small interval (or frame) of time, centered about a marker. For example, Snap_D currently shows a wideband FFT, which shows the relative strength (in dB) of the frequency components present in the timewave near marker A1. A snapshot display

```
10ms
20ms
50ms
100ms
.....
Wideband FFT
Narrowband FFT
✓LPC
Cepstrum
```

```
✓A
 A S1
 Selected
```

```
✓A1
 A2
 Snap M1
```

(39.820)

88.6
dB
66.5

44.3

22.2

0

1.1   2.2   3.3   KHz

LP — Analysis Options

A — Source Wave

A1 — Source Marker

Log — Log Button

Set — Setup Button
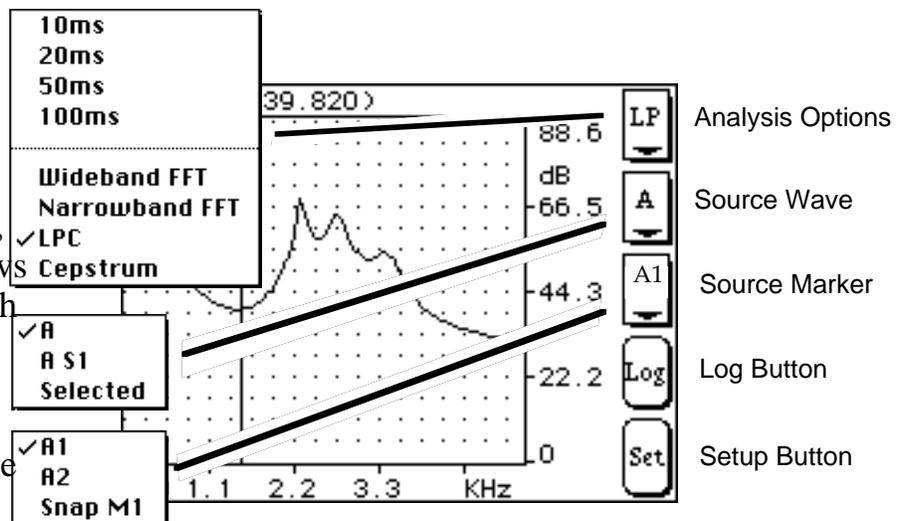
contains controls similar to an analysis display, with the addition of a marker pop-up menu. Because the computations involve only a small portion of the soundwave, they are performed immediately and no Calc button is required. The snapshot display takes the source wave as input, computes the specified snapshot option for the frame centered about the source marker, produces a result wave and displays that wave. Each time one of these items changes, the display automatically recalculates and displays the new result. The Log button performs actions such as logging data to a journal, or executing a task. The Setup button lets you alter the parameters of each analysis option, and select the events invoked by the Log button.

CHANGING THE SNAPSHOT OPTION

Wideband FFT is one of several options available. For information on all options, please see *Chapter 6*.

- In Snap_D, select Narrowband FFT from the options pop-up menu. The plot will resemble the wideband FFT, but with rougher, more jagged features.

- Click once inside Time_D to make it active.

- Move marker A1 a few times in the time display to get a feel for the narrowband FFT plot. Then go to the options pop-up menu again and select Wideband FFT to return to initial option.

SNAPSHOT MARKER LABEL AREA

The snapshot display has its own marker and label area to identify specific values within the snapshot result wave. In contrast, markers A1 and A2 point to values within the original soundwave.

- Click once inside Snap_D to make it active.

- Move the marker a few times, by holding down the *Option* key and dragging the mouse (press and hold the mouse button, move mouse, release button). There are two numbers displayed at the top of the marker. The first indicates the frequency in Hz (X axis value), the second (in parentheses) shows the magnitude in dB (Y axis value) at that frequency.

JOURNALS
SoundScope journals are simple text editors for storing key values and typing notes.

- Click once inside Time_D to make it active.

- Holding down the *Option* key, move marker A1 to 0.120 seconds.

- In Snap_D, click on the Setup button (the bottom of five controls). In the dialog that appears, make sure that the Send up to 5 peaks to journal option is checked. Click OK to exit the dialog.

- Now click the Log button, which will send information to the journal, as shown in the figure to the right.
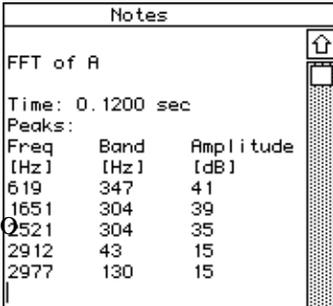
```
           Notes
FFT of A

Time: 0.1200 sec
Peaks:
Freq      Band      Amplitude
[Hz]      [Hz]      [dB]
619       347       41
1651      304       39
2521      304       35
2912      43        15
2977      130       15
```

Because journals are text editors, it is easy to reformat the data, as shown in the illustration to the right. After selecting a journal with the mouse, the user can easily delete, add and rearrange text using standard Macintosh editing techniques. You can also save journal data for use in a spreadsheet or word processor.

```
           Notes
Peaks from a Wideband FFT
of "SoundScope 16"
at 0.1200 sec

Freq      Band      Amplitude
[Hz]      [Hz]      [dB]
------------------------
619       347       41
1651      304       39
2521      304       35
2912      43        15
2977      130       15
```

- Select Save As ▶ Notes from the Journal menu.

That concludes Part 2 of the tour. The next section will cover waves, and introduce two different software instruments.

PART 3: WORKING WITH WAVES

Both soundwaves and the result of sound calculations are stored in objects known as waves. This section will demonstrate how to edit waves, compute statistics, and filter data. Before starting, we will open a different instrument and then load a new wave.

- Select Open from the File menu, open the "Instruments" folder (double-click or with the Open button), and choose "1 Channel (dual time)".

- Select Load Data ▸ A from the Wave menu, switch folders up to "SoundScope" and then down to "Waves", and select "phonetic I-A-U (10.4)".

This instrument is similar to the 1 Channel Analyzer, but adds a second time display at the top that automatically rescales to the size of the soundwave. The first time display (at the bottom of the screen) can be used to view any part of the time wave at any scale.
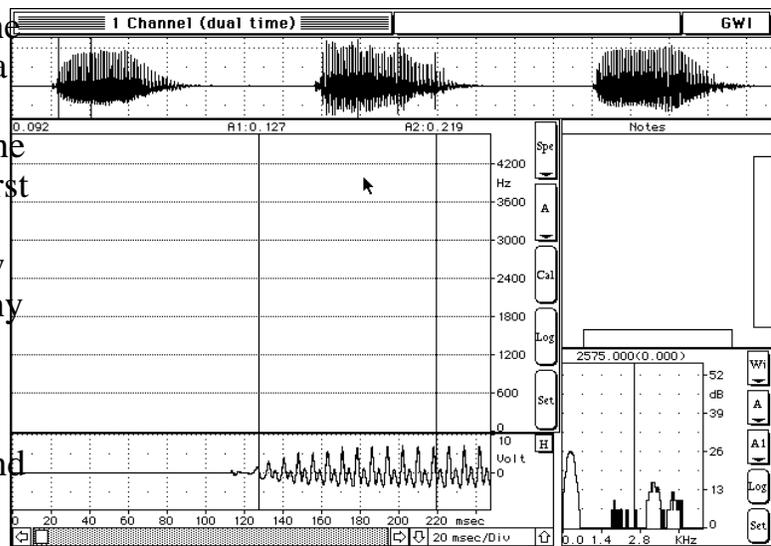
- To hear the soundwave, select Play A from the Sound menu. If you have SoundScope/16, you may need to first select Setup (due to incompatible sample rates) from the Sound menu, and change Play A and Play B hardware to Internal Speaker, as described in the Hardware Setup discussion earlier.

SELECTING A PORTION OF A WAVE

It is often useful to work with a small portion of a soundwave. The standard Macintosh technique is to use the mouse.

- Click once inside Time2_D (at the top of the screen) to make it active.

- Drag with the mouse over the first vowel. The selected portion will be highlighted.

- To hear the vowel, select Play Selected from the Sound menu.

WAVEFORM EDITING
The standard Macintosh Cut, Copy & Paste commands make editing a breeze.

- Select Cut from the Edit menu. Notice that the top display changes scale to accommodate the shorter wave.
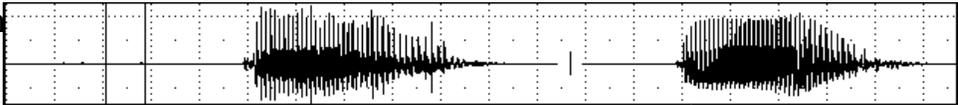
- Click the mouse between the two remaining vowels, as shown above. Select Paste from the Edit menu. Again, the display will rescale.

Next we will bring the vowels closer together by removing the silence at the beginning and in between.

- Select the flat portion of the utterance, before the first vowel.
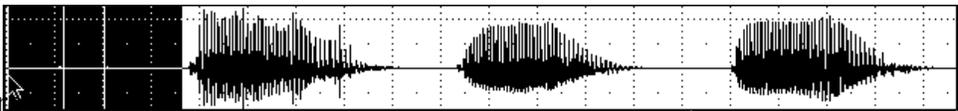
- Select Cut or Clear from the Edit menu. Note that Clear will prompt you with a dialog, to prevent you from accidently erasing valuable data.

- Select the flat portion between the second & third vowels.
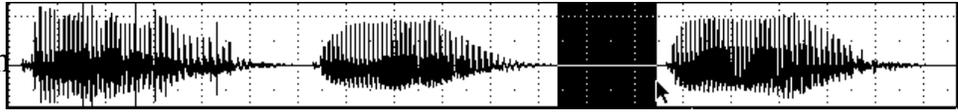
- Select Cut or Clear from the Edit menu.
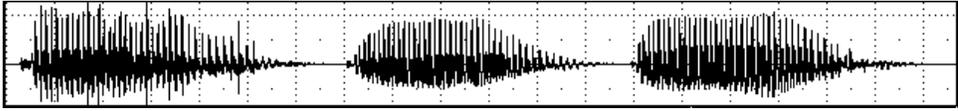
- To hear the result, select Play from the Sound menu. If you like, you can save this new sound.

AMPLIFY AND NORMALIZE

SoundScope lets you easily change the volume (loudness) of an entire wave, or of any portion.

- Select Amplify ▸ A from the Wave menu. Type "50%" to reduce the volume to half its current level. Click OK. Note that the wave amplitude drops in half in Time_D (on the bottom). The amplitude does not appear to change in Time2_D (on the top) because the display is set to automatically rescale in the vertical as well as horizontal direction.

- Now select an interesting portion of the wave with the mouse, perhaps the third vowel. Choose Normalize ▸ Selected from the Wave menu. Click OK. Note that the wave amplitude for the selected portion increases to fill the available ± 10 volt range.

- To hear the result, select Play A from the Sound menu.

WAVEFORM SEGMENTS
SoundScope offers a second technique for defining a portion of the wave: any two markers can define a segment.

- Click once inside Time_D (on the bottom) to make it active.

- Change the scale to about 50 msec per division (depending on your screen size), and scroll until you can see the second vowel. Note that the markers are not visible in this display. Fortunately, the top display always shows both markers, since it shows the entire soundwave.
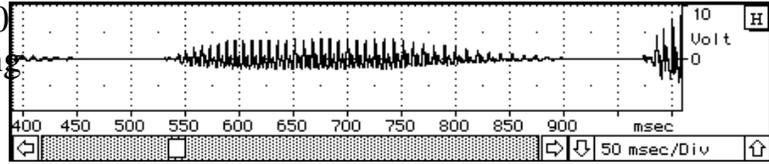
- Click once inside Time2_D (on the top) to make it active.

- Holding down the *Option* key (which will change the cursor), move one of the markers to the left of the middle vowel, and one to the right. Note that the markers appear in the bottom display as well.
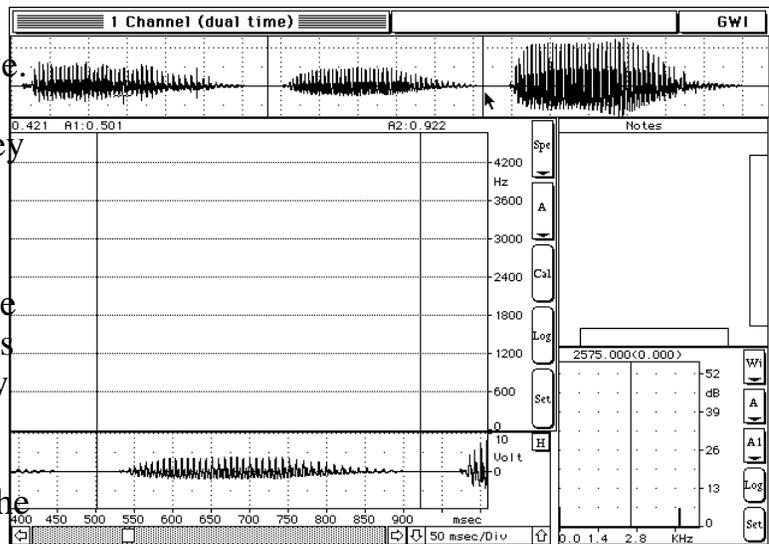
- To increase the volume of the segment between the two markers, choose Normalize ▶ A_S1 from the Wave menu. Click OK.

Segments are particularly useful for tasks, as you will see in the next chapter. In many cases, however, the choice between a segment and the Selected portion is simply a matter of preference.

EDITING WAVEFORM VALUES

On the computer, a sound is represented as a series of numbers or data points, often called samples. Each sample represents the digitized voltage (typically in the range of ±10 V) at a particular point in time. It is sometimes useful to examine and edit the actual data values. For example, it appears that the sound volume is zero at the beginning of segment A_S1, i.e. immediately following the left marker.

- Choose Edit Values ▶ A_S1 from the Wave menu. The time value is shown in the left column. Note that the segment starts at about 0.5 seconds (depending on exactly how much was cut out earlier). The samples of the wave are read left to right, top to bottom, with five samples on each row. Notice that the values are not exactly zero.

Edit Values:	A_S1				
sec					
0.120	0.219	0.684	1.238	1.770	2.060
0.120	2.303	2.629	3.047	3.539	3.923
0.120	4.185	4.366	4.419	4.362	4.287
0.121	4.207	4.093	3.918	3.600	3.246
0.121	2.960	2.620	2.236	1.879	1.583
0.121	1.400	1.218	0.865	0.458	0.104
0.121	-0.200	-0.406	-0.634	-0.840	-0.888
0.122	-0.887	-0.945	-1.025	-1.044	-0.937
0.122	-0.716	-0.541	-0.374	-0.085	0.198
0.122	0.406	0.511	0.605	0.859	1.143

Options

Index	Format	Units	Typing
⦿ Time	⦿ Dec	⦿ Volt	⦿ Overwrite
○ Point	○ Hex	○ Internal	○ Insert

OK

Help

The value of any sample may be changed by typing a new number into its cell. Ranges of cells can also be cut, copied and pasted within the table editor, and to and from SoundScope journals and displays, or third-party spreadsheets, word processors or other Macintosh software.

STATISTICS & SOUND STATISTICS

Two sets of statistics are available in SoundScope: sound statistics that are useful to speech scientists and speech pathologists, and general waveform statistics which are primarily intended for engineering applications.

• Click once inside Time_D (on the bottom) to make it active.

• Select a small portion (about 50 msec) of soundwave A, from the middle of the vowel.
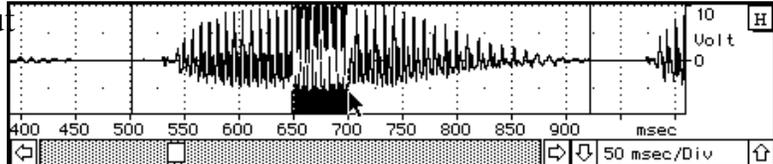
• Choose Sound Statistics ▸ Selected from the Wave menu. The "NAN" value for shimmer indicates that it could not be computed. As noted in Appendix A, shimmer requires a longer time interval for a good calculation.

```
                Sound Statistics:  [ Selected ]

Voiced: 100.000 %              Jitter (RAP): 0.074 % (-62.61 dB)
Unvoiced: 0.000 %              Fo Average: 123.274 Hz
Silent:   0.000 %             Fo Kurtosis: -0.714
Shimmer (APQ): NAN(021) %      Fo Skewness: -1.124
HNR: NAN(021) dB               Fo Range: 0.407 ST (2.884 Hz)
Breathiness: -8.110 dB         Fo Std Dev: 1.249 Hz

                            [ Options... ]  [   OK   ]
```
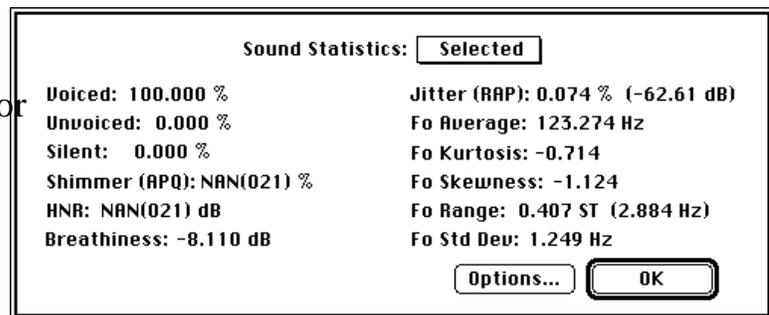
• From the Sound Statistics pop-up menu at the top of the dialog, choose A_S1. After several seconds, you will see the result for the entire segment.

```
                Sound Statistics:  [ A S1 ]

Voiced: 85.714 %               Jitter (RAP): 0.432 % (-47.29 dB)
Unvoiced: 0.000 %              Fo Average: 111.489 Hz
Silent:   14.286 %            Fo Kurtosis: -1.300
Shimmer (APQ): 2.506 % (-32.02 dB)  Fo Skewness: -0.113
HNR: NAN(021) dB               Fo Range: 8.489 ST (51.107 Hz)
Breathiness: -7.496 dB         Fo Std Dev: 14.583 Hz

                            [ Options... ]  [   OK   ]
```
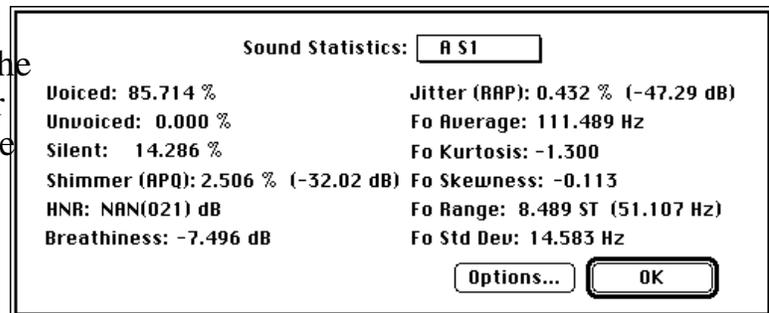
Note that, given the larger amount of time as input, shimmer was successfully computed. For more information on these statistics, see *Chapter 7, Short Menubar* and *Appendix A, Sound Analysis Computations*. You can also compute individual statistics and save the result to a journal or a wave, via task instructions.

FILTERING
Filters attenuate certain frequency components in a sound. This section will explore filtering, and introduce a new software instrument.

- Select Open from the File menu, switch up to the SoundScope folder, and into the Instruments folder, and choose "2 Channel Analyzer".

This instrument is designed to compare two soundwaves. We will synthesize a tone, add noise, analyze the result, and then filter out the noise.

- Choose Synthesize ▸ A from the Wave menu. Enter 5000 into the Length textbox. The default tone (periodic, sine) is fine, so click OK.

- Choose Synthesize ▸ B from the Wave menu. Enter 5000 into the Length textbox. Select Uniform noise, type 1 into the adjacent volts box, and click OK.

- In order to get a good look at the data, change the scales for both time displays to 2 msec/Div.

- Choose Play A and then Play B to hear the data.

- To create a noisy tone, we will add the two sounds (B=A+B). Choose Calculate from the Wave menu. Press on a few pop-ups to familiarize yourself with this dialog. When you're done exploring, select B = A + B from the appropriate pop-ups, as illustrated to the right.
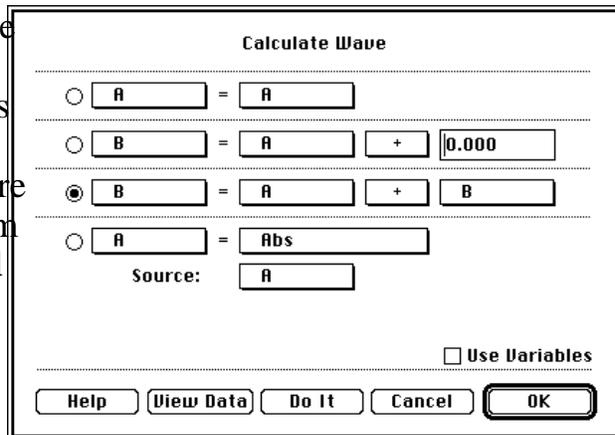
- Click on Do It, and then OK. The noise is very visible - and audible (select Play B). With SoundScope, the noise can also be analyzed.
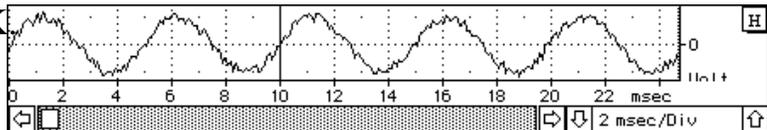
- For an overall picture, click the Calc buttons on both analysis displays. Where the original tone is in a single frequency band, the noisy tone has components all over.

- For an instantaneous picture at a certain point in time, use markers and the snapshot displays. First, move one marker into position by selecting Move ▸ Time_D from the Display menu, unchecking the Display checkbox, checking Marker, selecting To the X secs position and typing .010 (10 msec). Confirm that the settings match those in the figure below, then click OK.
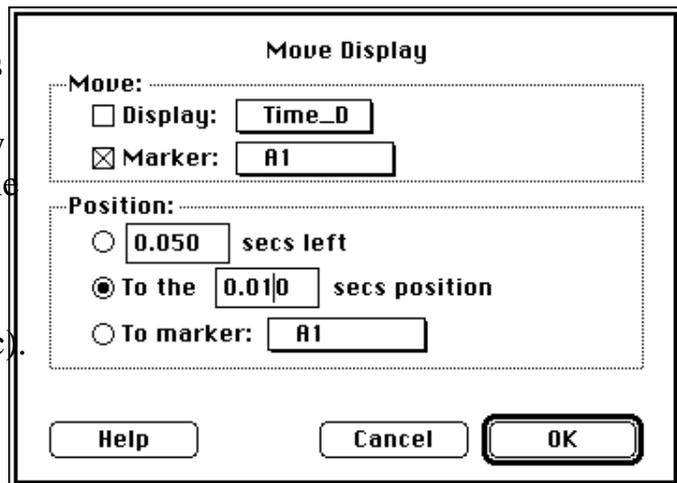
- The procedure for marker B1 is easier, since the dialog uses your last choices as the defaults: select Move ▸ Time2_D from the Display menu, choose B1 from the Marker pop-up, and click OK.

- Make sure that both snapshot displays show Wideband FFT. If not, use the top control to change from another option. In Snap_D, move the marker to the peak of the data, yielding the frequency of the tone. In Snap2_D, move the marker to the second peak, to get an idea of what frequencies we should try to filter out.
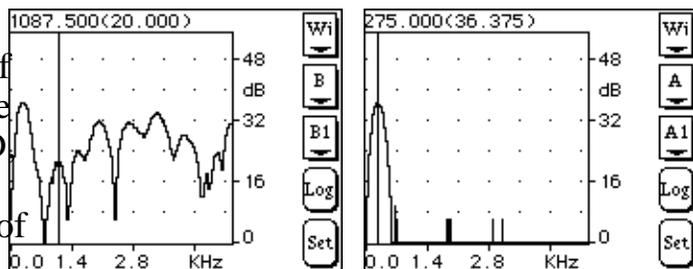
- Click once inside Time2_D (on the bottom) to make it active.

- Using the mouse, select a small portion of the noisy tone that includes the marker, say from 5 to 15 msec.
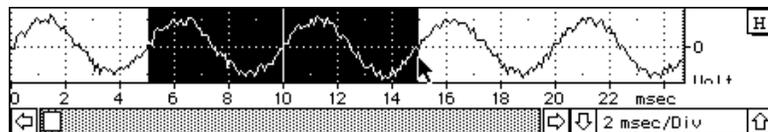
- Choose Filter ▸ Selected from the Wave menu, and change the Frequency Cutoff for the filter to 10% - yielding a number close to the target determined above. Click OK.

Filter: [Selected]

Type: [Smoother (ham)]   [Custom...]
Freq Cutoff: [10%]   (% of sample rate)
Boundaries: [zero pad]

Freq Cutoff: 100.0 Hz   Phase: linear
Stopband: -42 dB   Taps: 20

[Help...]   [Cancel]   [OK]
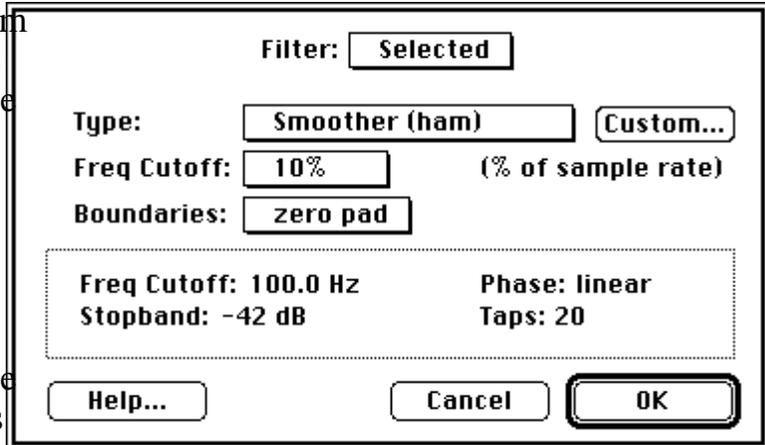
Clearly, the filter worked! Visually, the noise is gone. The snapshot display confirms it, as will a spectrogram. Because we only filtered a small portion, you will not be able to hear the difference. If you like, filter the entire soundwave B, and play the result.

EXPLORING SOUNDSCOPE
Using the techniques that you have learned, please explore SoundScope on your own. Try loading other instruments and waves. Test out any menu item or front panel control. If you get stuck and need some help, please feel free to give us a call.

QUITTING SOUNDSCOPE
The next section covers advanced features of SoundScope. If you do not want to tackle that quite yet, you can quit and explore more at another time.

- If you do not wish to continue, select Quit from the File menu.

PART 4: DESIGNING INSTRUMENTS

Waves, markers, displays, journals, and tasks can be combined like building blocks to create software **instruments**. A few pre-defined instruments have been included with SoundScope. This section describes how to customize SoundScope by modifying an existing software instrument, or creating a new one from scratch. This information is not required for everyday user of SoundScope. You may want to gain experience as an instrument user before learning the advanced features aimed at the instrument designer.

The following pages demonstrate how to change the front panel layout and add a new display. Before starting, reload the instrument used in Part 1.

- Select Open from the File menu, and choose "1 Channel Analyzer".

THE FRONT PANEL

The main SoundScope window is referred to as the **front panel**. It includes all displays and front panel journals. You can change the size and position of each front panel object with Panel Edit Mode. For example, you may want a wider journal to enter more substantial comments. The following steps show you how to modify the standard "1 Channel Analyzer" instrument to achieve this end.

Before   *After*
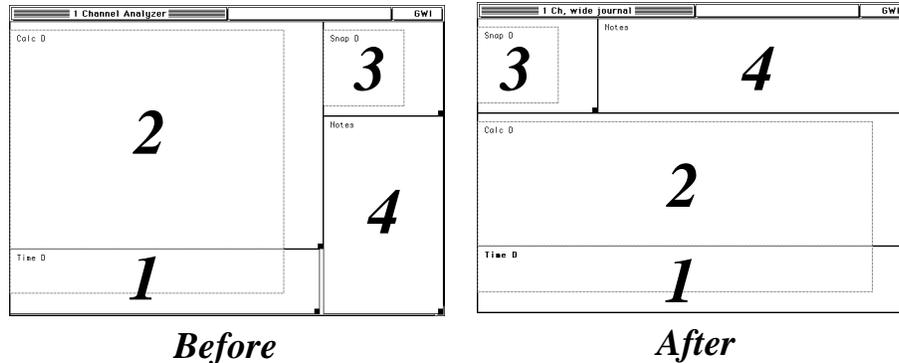
In Panel Edit Mode, all displays, journals, controls, indicators and pictures can be moved, resized, and deleted. When Panel Edit is turned on, a rectangular outline of each front panel object appears, with the object name in the upper left corner. If an object is selected, its name appears bold. To resize, one simply drags the resize box (i.e. little black square at lower right). To reposition, one drags the actual object. To reposition an object one pixel at a time, one selects with the mouse, and then taps on the ← ↓ ↑ → keys. To resize one pixel at a time, one selects with the mouse, holds down the *Option* key, and then taps on the ← ↓ ↑ → keys. This is very similar to working with rectangles in MacDraw.

SoundScope prohibits shrinking displays below a practical minimum size to ensure

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readable content. Individual items within a objects cannot be resized independent of the object. Should displays or journals overlap, a warning is issued (i.e. "Invalid Panel" in status bar) and the user is prohibited from leaving design mode until the overlapping is eliminated.

A PICT file (72 dpi picture file) can be placed onto the front panel by first transferring it to the clipboard (with Cut or Copy) and then moving it to the display with the Paste command, as illustrated to the right. Also, PICTs can be Cut or Copied from the SoundScope front panel to the clipboard. Since front panel displays, journals, and so forth are not PICTs, they cannot be moved to or from the clipboard.
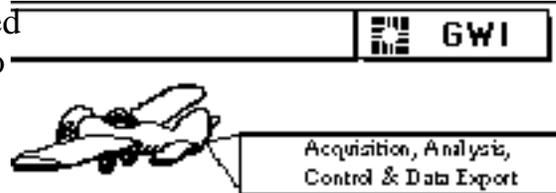
- Select Panel Edit On from the Display menu.

- Resize the Calc_D analysis display and move it down, to make room for rearranging other items.
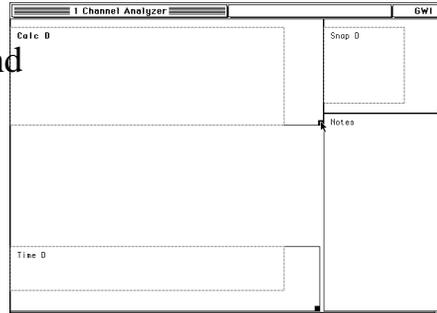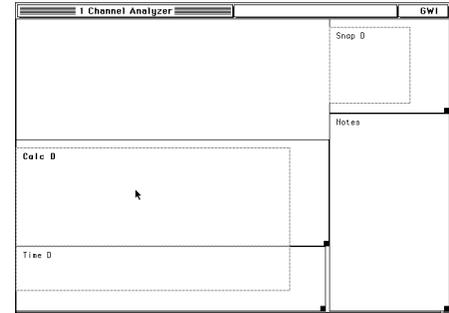
Before   *After*

- Move Snap_D to the left corner. Then resize the Notes journal, in preparation for moving it.
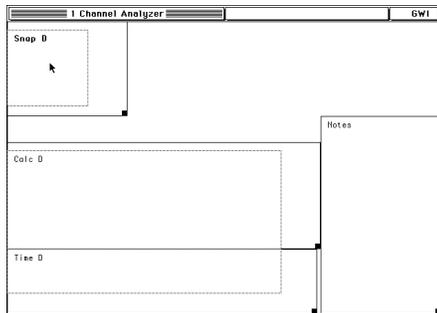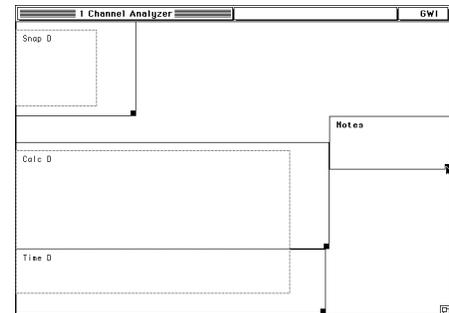
Before   *After*

- Move the Notes journal so it is adjacent to Snap_D. Then move the Calc_D analysis display up until it touches Snap_D.
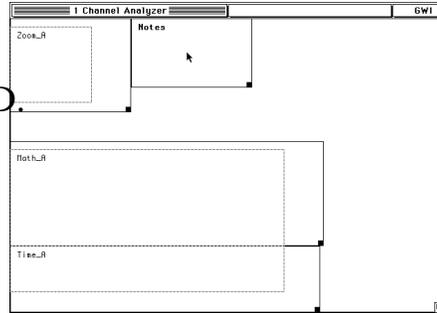
Before
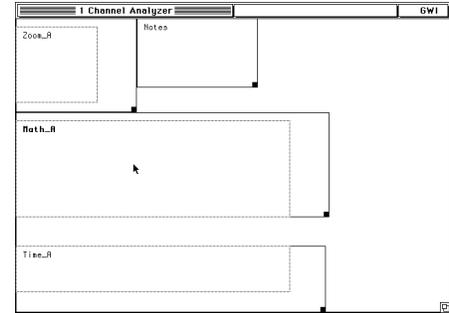
After

- Resize Calc_D to make it taller and occupy the full screen width. Then carefully resize the Time_D wave plot display so that the dotted vertical lines meet. (Tip: hold down the *Option* key and use the right and left arrow keys to resize one pixel at a time.)
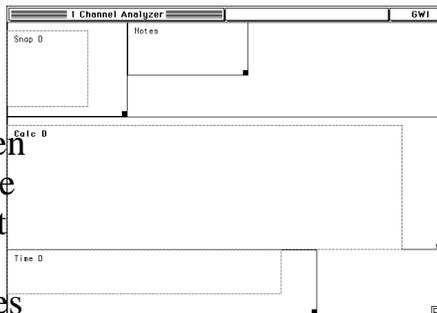
Before
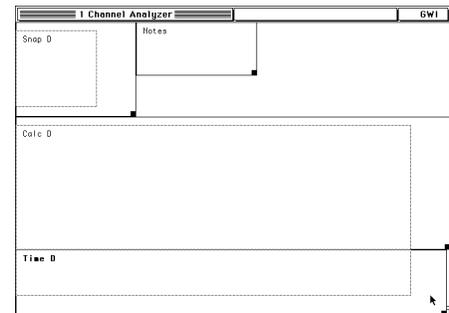
After

- Finally, resize the Notes journal to fill the entire corner.

- Select Panel Edit Off from the Display menu. You can now use the new layout.

- To save this instrument for future use, select Save As from the File menu, type "1 Ch, wide journal" into the standard file dialog, and click the Save button.
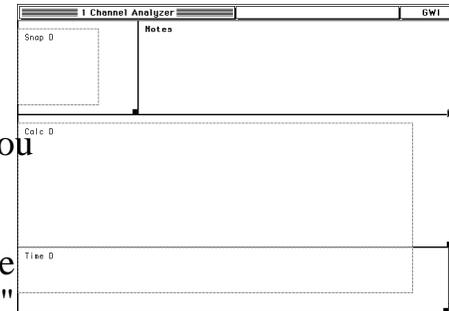
You have now added a new instrument to your portfolio.

ADDING A DISPLAY

In addition to simply rearranging a front panel layout, you may want to create new displays or journals. The following steps show you how to add a second snapshot display to the above instrument.

- Select Choose Menubar ▸ Full from the Edit menu. New items will appear in most of the menus (although you will not notice until you pull them down).

- Snapshot (and analysis) displays require waves to hold the result of their calculations. To create one, select New from the Wave menu.

- Type in the name "Snap2" and click OK.
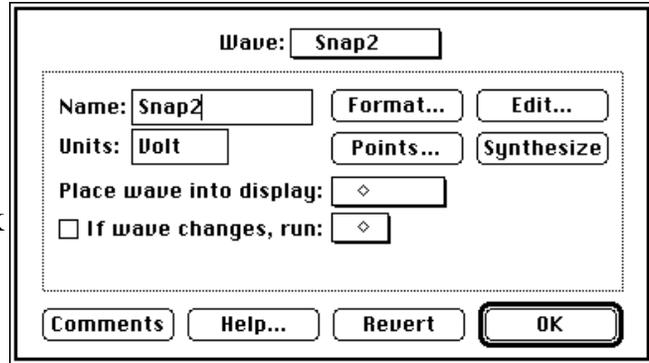
- Select Panel Edit On from the Display menu.

- Resize the Notes journal to make room for a new display.
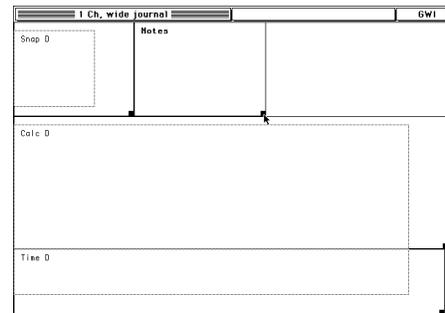
- Select New from the Display menu.

- Type in the name "Snap2_D", and select Snapshot from the pop-up immediately below. In the scrollable list towards the right, click in the gray area (or down arrow) of the scroll bar to scroll down. Drag the Snap2 wave into the top slot on the right, as shown in the diagram.
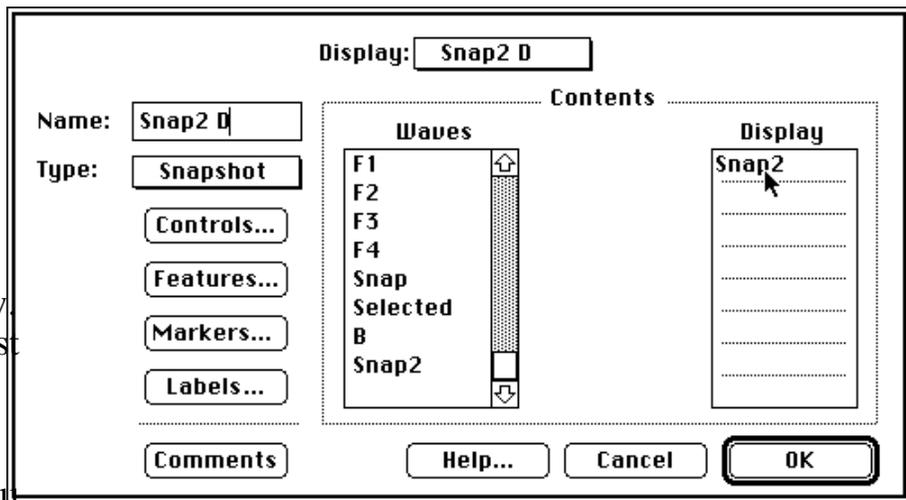
- Click OK to accept the changes and exit the dialog.

- Move and resize Snap2_D until it fills the upper right corner.

Before   *After*

- Select Panel Edit Off from the Display menu. You can now use the new layout.
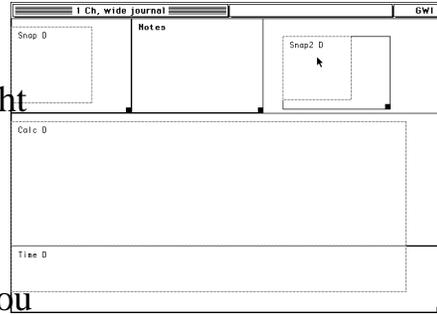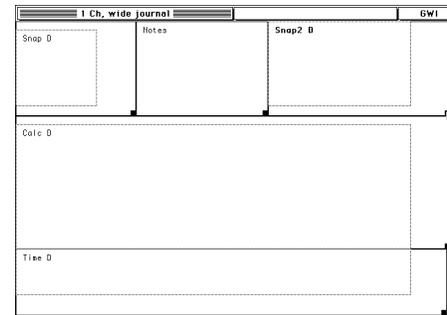
- To save this instrument for future use, select Save As from the File menu, type "1 Ch, 2 snapshot" into the standard file dialog, and click on the Save button.

This new instrument is very similar to the "1 Ch (dual snap)", although it lacks a fourth marker, marker label line and various custom display settings. For more information about these settings, please see the following section, and the SoundScope & SuperScope II Reference Manual.

CREATING A NEW INSTRUMENT

This section shows you how to create a simple instrument with a time display, analysis display, journal, waves, markers and a segment. The result will look like the illustration to the right.

- Select New Instrument from the File menu and click OK in the confirmation dialog. This command deletes all waves, displays, journals, etc., allowing you to proceed with a "clean slate."
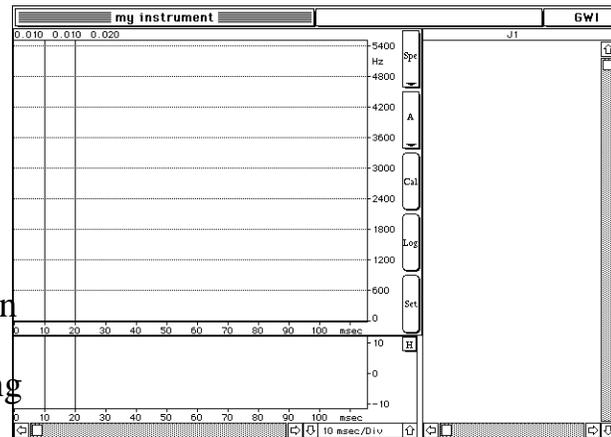
WAVES

This instrument will require two waves, an integer wave for holding the sound, and a floating-point wave to hold the calculation result shown in the analysis display.

- Select New from the Wave menu. The wave options dialog will appear. Type "A" in the Name edit box.

- Sound data is stored as a series of integers. Click the Format button and select the 16-bit Integer option.

  Waveforms recorded with 8bit, 12bit and 16bit digitizers are all stored in 16bit integers with values ranging from -32K to +32K (i.e. the data is packed into the most significant bits in the case of the 8 and 12 bit digitizers).
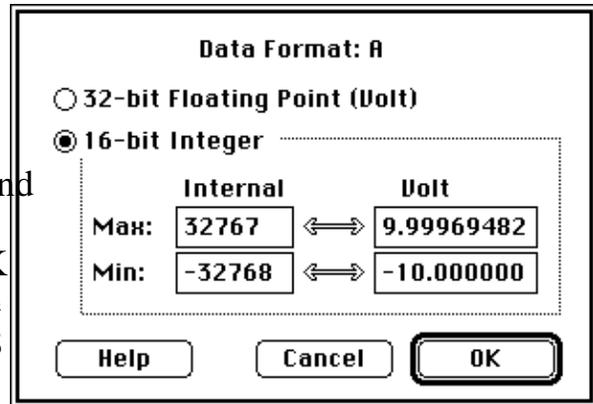
- Click OK to exit the Format dialog.

- From the Wave pop-up menu at the top of the wave options dialog, select New Wave. Type the name "Calc". By default, waves are stored as 32-bit floating point data, as desired for calculation results.

- Click OK to exit the Wave dialog.
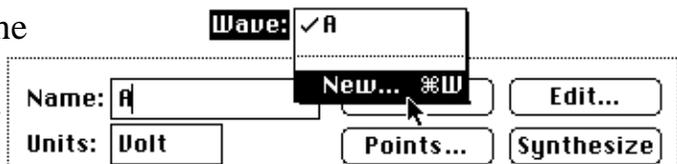
DISPLAYS
Next we will create the displays and arrange them on the screen.

- Select New from the Display menu. Type "Time_D" in the Name edit box.

- Notice the two lists in the Contents area of the dialog. The Waves list on the left shows all available waves and segments; the Display list on the right shows which waves and segments are to be shown in our display. Drag wave A from the Waves list to any slot in the Display list.
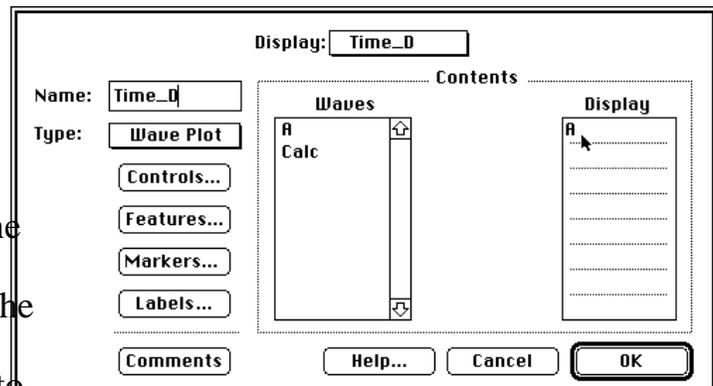
- From the Display pop-up menu at the top of the dialog, select New Display. Type "Calc_D" in the Name edit box, and select Analysis from the Type pop-up menu.
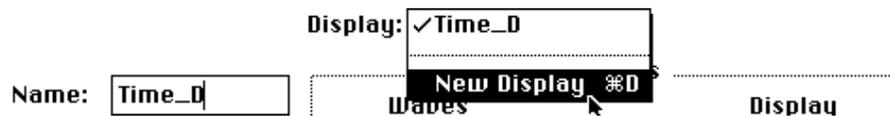
- Drag the result wave Calc from the Waves list to the top slot in the Display list. In analysis and snapshot displays, the calculation result is stored in the wave that is in the top slot. (With the Spectrogram and LPC History analysis options, the next 5 slots indicate the result waves for formant-tracking.).

- Click OK to accept your changes and exit the Display dialog.

The next step is to arrange the displays on the front panel.

- Select Panel Edit On from the Display menu.

- Drag Time_D bottom left corner and resize it so that it occupies about 3/4 of the width of the front panel.
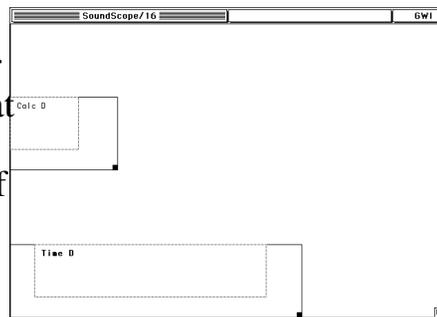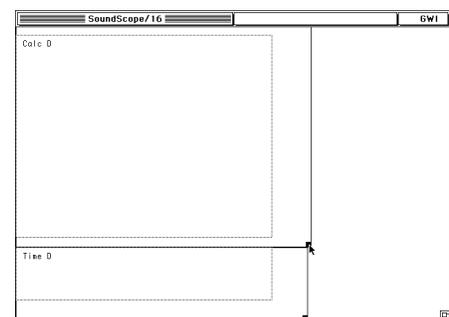
Before   *After*

- Drag Calc_D to the upper left corner and resize it to fill the space above Time_D.

JOURNALS
Next, we will create a journal for storing notes and key values extracted from waves.

- Select New from the Journal menu. Type "Notes" in the Name edit box.

- Click OK to create the journal and exit the dialog.

- Move and resize the journal so that it fills in the gap remaining on the front panel.

• Select Panel Edit Off from the Display menu. You should see something similar to the front panel pictured to the right.
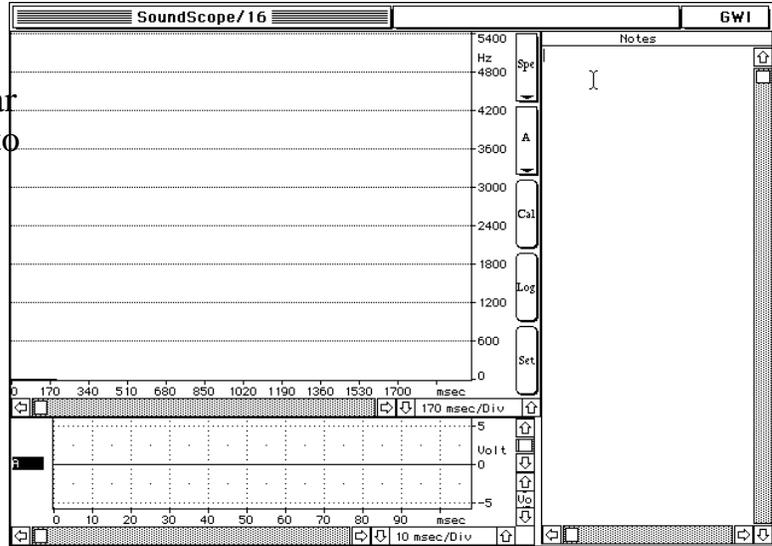
MARKERS & SEGMENTS

Now we will create two markers and a segment.

• Select Markers ▸ Time_D from the Display menu.

Take a moment to survey this powerful dialog. It is divided into three areas. The *Display* area determines which markers are shown in each display. The *Segment* area is used to create, modify and delete segments; and the Marker area is used to create, modify, and delete markers.
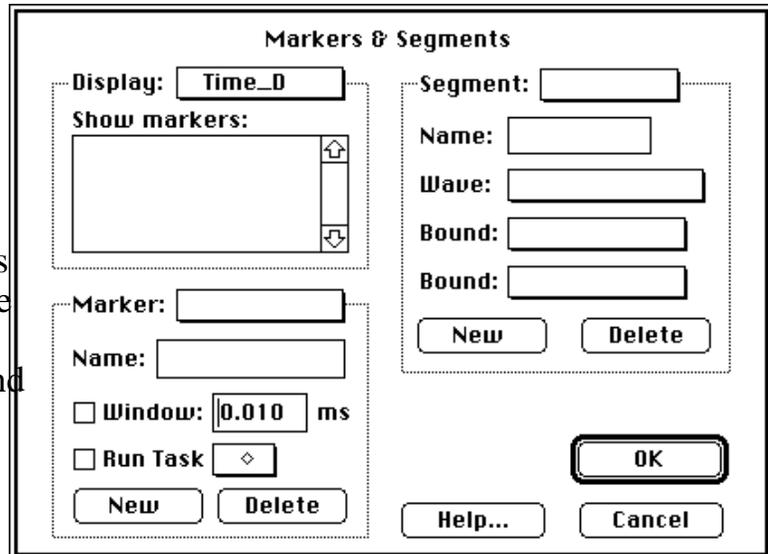
We will use all three parts in this dialog.

• In the marker area of the dialog, click the New button. Type "A1" into the Name edit field.

• In the same way, create a second marker named "A2".

- Both marker names will appear in the Show markers list in the Display region. Click A1 and then A2 to show both markers in Time_D.

Display: [Time_D]

Show markers:
A1
A2

- In the Display pop-up menu, select Calc_D. Again click A1 and then A2 to show both markers in this display.

- In the segment area of the dialog, click the New button. Type "A_S1" into the Name edit field. Note that wave A and the marker boundaries A1 and A2 are automatically selected in the corresponding pop-up menus.

- Click OK to exit the Markers & Segments dialog.
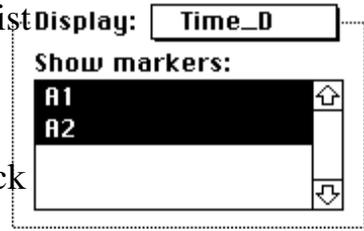
Now markers A1 and A2 are visible in the time and analysis displays.

REFINING THE DISPLAYS

The next step is to improve the appearance of the time and analysis displays.

- Select Controls ▸ Time_D from the Display menu. In Macintosh terminology, buttons and scroll bars are called controls.

This dialog is divided into two main areas that specify controls and default values for horizontal and vertical scale and position.

Controls: [Time_D]

Horizontal
Position: [Scrollable]
Scale: [Adjustable]
☐ Add "H" auto adjust button

Vertical
Position: [Scrollable]
Scale: [Adjustable]
☐ Add "V" auto adjust button

☐ Fix sample period to [0.001000] sec/pt

[Help...]   [Cancel]   [OK]
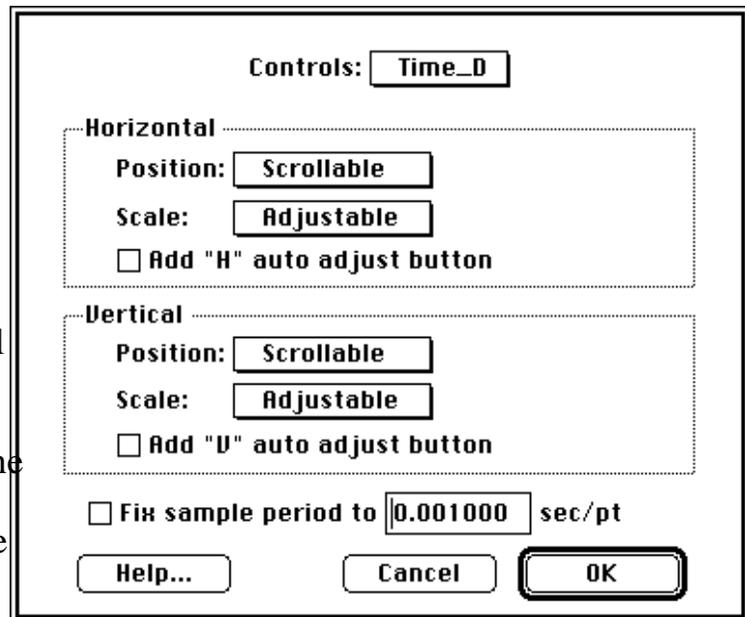
- In the Horizontal region of the dialog, notice that Position is set to Scrollable and the Scale to Adjustable. These settings put a scroll bar control and a scale control in Time_D. Click on the Add "H" auto adjust button checkbox to add an additional control that will automatically change the scale so that the entire sound fits into the display.

Since the amplitude of digitized soundwaves is fairly constant, there is usually no need to be able to change the vertical position or scale directly from the front panel. We will remove these controls to simplify the instrument.

- In the Vertical region of the dialog, select Fixed from the Position menu accepting the default value of 0 that appears to its right. Select Fixed from the Scale menu and type "10" in the edit box that appears, to set the scale to 10 Volts per division. (One division on the screen is 32 pixels, about 1/2".)
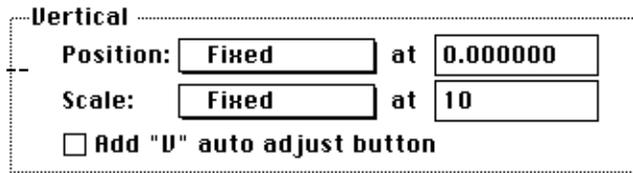
For Calc_D, we would like the horizontal axis to be "linked" to Time_D so that they will both scroll and change scale in unison.

- Select Calc_D from the Controls menu at the top of the dialog. Note that an additional pop-up menu appears at the top, so that each analysis option can have different controls.

- In the Horizontal region of the dialog, select Previous from the Position menu and Previous from the Scale menu. Note that Time_D is the previous one in the list of displays.

- Notice that the vertical Position and Scale are set to Fixed base and Fixed, with values in the edit boxes appropriate for the analysis option indicated at the top of the dialog.

- Select LPC History from the analysis menu in the top right of the dialog, and change the horizontal Position and Scale to Previous. Repeat for HS Slice, skip Avg Spectrum, and repeat for the remaining analysis options. The horizontal axis of Average Spectrum is frequency rather than time, so it does not make sense to link the position and scale to the soundwave.

- Click OK to exit the Controls dialog.

Next we will modify some display features.

- Select Features ▸ Time_D from the Display menu.
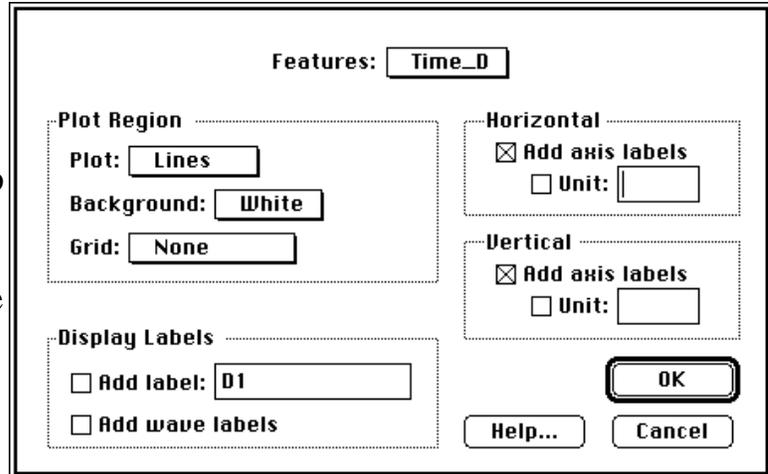
- Under Plot Region, select None from the Grid pop-up menu. This setting eliminates the grid (dotted lines at each division) in the time display.

Features: [Time_D]

Plot Region
Plot: [Lines]
Background: [White]
Grid: [None]

Display Labels
☐ Add label: [D1]
☐ Add wave labels

Horizontal
☒ Add axis labels
☐ Unit: [ ]

Vertical
☒ Add axis labels
☐ Unit: [ ]

[OK]
[Help...] [Cancel]

- In the Display Labels area, click the Add wave labels checkbox to de-select it, removing the wave label region on the left of the time display. (The wave label region is useful when more than one wave is shown in a display.)

- Click OK to exit the Features dialog.

We will also add a marker label line to show the time positions of markers A1 and A2.

- Select Labels ▸ Calc_D from the Display menu.
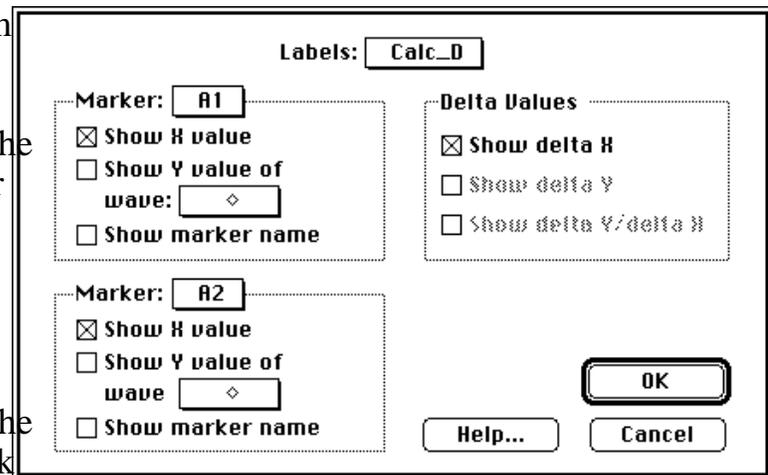
- Note that A1 is selected in the Marker pop-up at the upper left corner of the dialog. Click the Show X value checkbox below it to select this option.

Labels: [Calc_D]

Marker: [A1]
☒ Show X value
☐ Show Y value of wave: [◇]
☐ Show marker name

Marker: [A2]
☒ Show X value
☐ Show Y value of wave [◇]
☐ Show marker name

Delta Values
☒ Show delta X
☐ Show delta Y
☐ Show delta Y/delta X

[OK]
[Help...] [Cancel]

- Note that A2 is selected in the other Marker pop-up. Click the Show X value checkbox below it to select this option. The time values of markers A1 and A2 will be shown in the marker label line.

- In the Delta Values area of the dialog box, click the Show delta X checkbox to select this option. The time difference between markers A1 and A2 will appear on the left edge of the marker label line.

- Click OK to exit the Labels dialog.

SAVING THE INSTRUMENT
Save your new instrument to disk for later use.

- Select Save As from the File menu.

- Type "my instrument" into the standard file dialog, and click on the Save button.

You may want to practice working with this instrument. Afterwards, feel free to make additional changes to it. Also, feel free to copy and modify the instruments included with SoundScope, or swap instruments with friends and colleagues. If you create a special-purpose instrument, please consider sending it to us so that we can share it with others.

WHERE TO GO FOR MORE INFORMATION
Congratulations! You have completed the tour and are well on your way to becoming a SoundScope expert. Chapter 3 is a detailed discussion of a sophisticated software instrument. Chapter 4 covers the building blocks used in instrument design. Chapter 5 describes many of the instruments that are included with SoundScope. Chapter 6 covers the front panel, and Chapter 7 outlines all the menus and dialogs encountered by a typical user. For background information on computer analysis of speech, please refer to the "Manual of Clinical Applications," available free from GW Instruments.

SoundScope is built on top of the SuperScope II engine, and includes all features available in the SuperScope II application program. Advanced features such as instrument design and multi-channel data acquisition are described in the SoundScope & SuperScope II Reference Manual.

Chapter 3
Advanced Tutorial

The dilemma in designing any product involves the tradeoff between the number of features and ease-of-use. Products with a few features tend to be easy to learn and use, and those packed tend to be more challenging. And in many cases, simple applications do not satisfy the user's requirements. SoundScope overcomes the dilemma with several features which include a menubar editing environment, a dialog-oriented command language (referred to as "tasks"), and a powerful set of database export tools. The Menubar Editor enables an instrument designer to create a menubar, and assign tasks to each menu item. The menubar can contain any number of menus and commands; subsequently making it possible to create a very simple instrument, with only a few commands (e.g. Record, Analyze, Print, Save Data, Export To Database). And hence, the user can have it both ways with SoundScope: simple and easy, or featureful and complicated. An example is the Professor who wants a powerful environment for himself/herself, yet who wants a very simple environment for his/her students.

This chapter provides a tour of a simple instrument that records three vowels, analyzes them, and sends the results to a FileMaker database. The name of the instrument is "TriVowelgram™", and it is described in detail in *Chapter 5, Instruments*. The following discussion will show the reader how to create tasks and menubars, and how to interface SoundScope to another application program. It is not necessary to understand these features in order to use SoundScope; they are only used by the Instrument Designer, as opposed to the Instrument User. So, if you want to learn about instrument design, please proceed. Please note that SoundScope/8 does not run Tasks, and therefore will not support several of the instructions in this chapter.

- Please exit SoundScope, find the TriVowelgram™ instrument, select it, and choose Duplicate under File. This will cause the file to be duplicated and renamed to "TriVowelgram™ copy". It is good to work with a copy of an instrument when doing design; so you can return to the starting point, if desired. Also, as you are designing, it is always good to save your instrument every now and then with an different file name (e.g. fred1, fred2, fred3, etc), to protect against the unexpected.

- Double-click on TriVowelgram™ copy; or launch SoundScope, and then choose Open under File to load the instrument.

The TriVowelgram will appear as shown below.
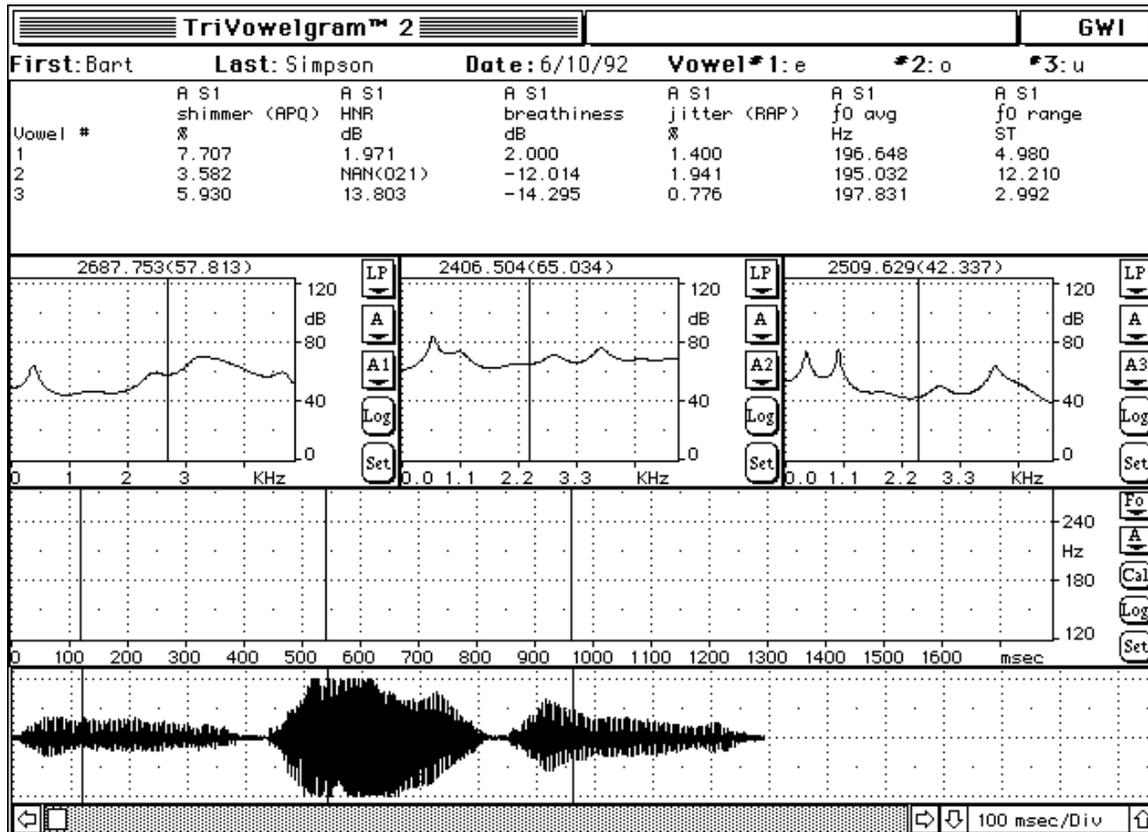
The lower most display shows a timewave which contains three vowels (e, o, u). The display above the timewave is used for pitch tracking, and the three above it show an LPC for each vowel. The main text area shows several analysis parameters calculated for each vowel. The front panel Sizing option has been set to Fixed; therefore this instrument will always open into a 640 x 480 window. If your monitor is smaller, you can either do the tutorial with a portion of the instrument in view, or you can choose Menubar ▶ Full under Edit, choose Panel Options under Display, set Sizing option to Resize Proportional, press OK, choose Save under File to save the instrument to disk, and then choose Open under File to load it. The instrument should reappear with it's window the size as the computer screen. If the front panel is too crowded on the screen (i.e. it gives you a "Front Panel is Invalid" message), you will need to resize several front panel objects to eliminate overlapping, and then choose Panel Edit Off under Display.

• Press the Calc (labeled "Cal") button in the Analysis display, above the timewave, to initiate the Fo calculation.

With the current analysis options, an Fo track may not be feasible (i.e an Fo plot

does not appear).

- Choose Options under Calculate (or press *Option* 'l' {lower-case letter L}) to open the Analysis Options dialog, and then set the acceptable pitch periods to 150Hz to 210Hz. Press Ok to exit the dialog, and then press the Calc button again. A valid pitch track should appear, as shown below.

Pitch tracking is more of an art than a science, and it is often necessary to adjust the pitch track options before proceeding with pitch-based analysis such as HNR, shimmer and jitter. In other words, don't trust the TriVowelgram's data if a good pitch track could not be attained. In the above example, the second vowel is a little clipped at the ±10V bound, causing it's Fo plot to waiver. Consequently, the HNR value calculated from this vowel is NAN(021), which means "not a number" (i.e. it could not be determined). We include this case intentionally to make the reader more aware of issues that appear in the field.
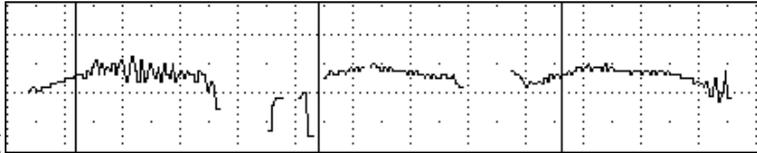
- Choose Complete Analysis in the Calculate menu. This will cause the Analysis Task to run, which does the following:

  a) Clear the main journal, above the 3 LPC displays.

  b) Calculate the voiced intervals of the timewave, and place the result into a wave called "Voiced". The points in this wave are set to 1.0 where the timewave is voiced, and 0.0 elsewhere. It is plotted above the timewave, as shown in the illustration to the right.
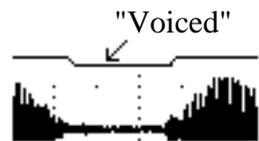
  c) Do the following three times:

     A) Find the first pulse in the Voiced wave. Notice that a pulse in Voiced corresponds to a voiced interval, which should be a vowel. Markers A1 and A2 are moved to the beginning and end, respectively, of the voiced interval. Segment A_S1 is defined in the Marker dialog as the part of the timewave (referred to as "A") between markers A1 and A2, and should therefore contain a vowel.

     B) Calculate sound statistics on segment A_S1 (i.e. the vowel). Shimmer, HNR, breathiness, jitter, Fo average, and Fo range values are sent to journal Notes, and appear as one row in the table.

d)   Move marker A1 to the first vowel, and then move it 0.1 seconds to the right, to place it inside the vowel. The Snap_D display shows an LPC of the timewave about this marker (i.e. it shows a transfer function of the vocal track during the vowel's voicing).

e)   Move marker A2 into the second vowel, to setup it's LPC.

f)   Move marker A3 into the third vowel, to setup it's LPC.

g)   Change the word "Pulse" to "Vowel" in the Notes journal.

h)   Copy the contents of the Notes journal into the Table journal. The Table journal is a repository for data that is later sent to a database.

i)   Beep, to tell the user that the task is done.

- Select Choose Menubar ▶ Full in the Edit menu. The menubar will change to one that includes the standard SoundScope commands.

- Select Edit ▶ Analyze under Task to view the task that we just ran. The Task Editor dialog will appear, as shown to the right. Analyzes instructions are in the upper window, and the Instruction Dictionary is at the lower left. The instructions are executed in order, top to bottom. The Instruction Dictionary contains the building blocks for creating a task. For details on tasks, please see the Task discussion in *The Menubar* chapter of the SoundScope & SuperScope II Reference Manual. For details on each instruction, please see the *Instructions* chapter in the same reference manual.
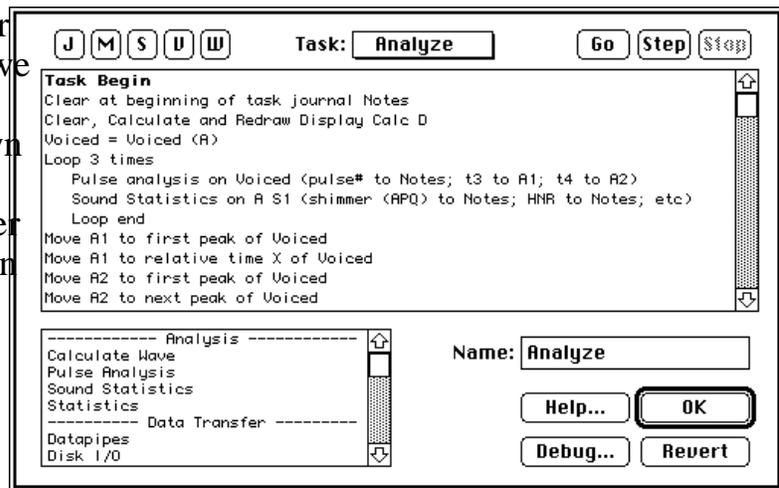
Task Editor dialog:

```
[J][M][S][V][W]    Task: [ Analyze ]        [Go][Step][Stop]

Task Begin
Clear at beginning of task journal Notes
Clear, Calculate and Redraw Display Calc D
Voiced = Voiced (A)
Loop 3 times
    Pulse analysis on Voiced (pulse# to Notes; t3 to A1; t4 to A2)
    Sound Statistics on A S1 (shimmer (APQ) to Notes; HNR to Notes; etc)
    Loop end
Move A1 to first peak of Voiced
Move A1 to relative time X of Voiced
Move A2 to first peak of Voiced
Move A2 to next peak of Voiced

------------ Analysis ------------        Name: [Analyze]
Calculate Wave
Pulse Analysis
Sound Statistics
Statistics                                [ Help... ]  [[ OK ]]
---------- Data Transfer ---------
Datapipes                                 [ Debug... ] [ Revert ]
Disk I/O
```

- Double-click on the Voiced = Voiced(A) line. A Calculate Wave dialog will appear, as shown to the right. This dialog is used to execute one of the 80 functions and operators. For details on each, please see the *Functions and Operators* chapter in the SoundScope & SuperScope II Reference Manual. Pop-up menus and radio buttons are used to setup a calculation. Each pop-up contains a list of waves, a list of functions (e.g. cos(), FFT()), or a list of operators (e.g. +, - ,*). The dialog is currently setup to calculate the Voiced interval of the A wave and place the results into a wave named Voiced.
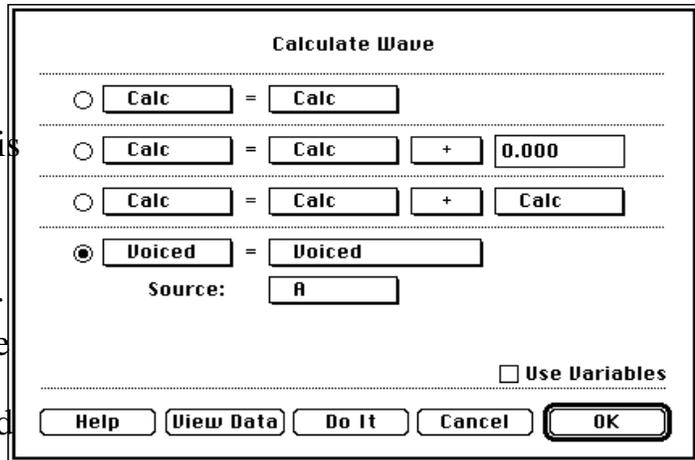
- Press on a few pop-ups to view the available waves, functions and operators.

- Press Cancel to exit the instruction.

- Each instruction can be viewed and modified by double-clicking on the text that appears in the task region. Double-click on a few to see their dialogs, remembering to press Cancel to exit each instruction.

There are over 20 different instructions in the Instruction Dictionary, each with it's own dialog environment. One views or edits an existing instruction by double-clicking on it and one copies an instruction by selecting it and then choosing Copy. To paste a copied instruction into a task, install a task insertion pointer (i.e. a horizontal dotted line) by clicking on the instruction above your target, and then choose Paste. The copied instruction will appear immediately after the selected instruction (which is highlighted). One can Delete an instruction by selecting it and pressing the Delete key, and create a new instruction by dragging it from the Instruction Dictionary into the task region. Tasks can be created, deleted, edited, and run with the New, Delete, Edit, and Run commands in the Task menu. And this is how one "programs" SoundScope.

- To gain a sense of the richness of the SoundScope programming environment, double-click on each instruction in the Instruction Dictionary, pressing Cancel to exit each instruction's dialog. Double-clicking is like dragging the instruction from the Dictionary to the end of the task. Pressing OK causes the instruction to be inserted into the task, Cancel causes the insertion to be aborted.

Recall that the Analyze task bases calculations on the segment of timewave that is

---

considered voiced. Fo tends to vary a little at voicing onset and offset; subsequently, it is desirable to move the two segment-defining markers, A1 and A2, inward, to eliminate the onset-offset effects. We will do this with two Move Marker instructions.

• Scroll the Instruction Dictionary until the Move Marker Instruction is visible, as shown to the right.

```
Journals
Move Marker
Read Wave Internals
```

• Drag the Move Marker instruction from the Instruction Dictionary into the task area and put it under the Pulse analysis instruction, as shown to the right.

```
Pulse analysis on Voiced (pulse# t
Sound Statistics on A S1 (shimmer
```
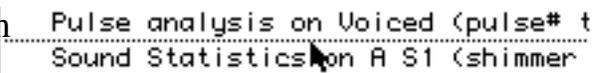
The Move Marker dialog will appear, as shown to the right. This instruction is used to move markers to waveform minimums, maximums, local minimums ("valleys"), local maximums ("peaks"), upstrokes, downstrokes, specified points, and specified wave values.

Move Marker: [A1]

Move to [next maximum] of wave [Calc]

Options

Help

Do It

Cancel

OK
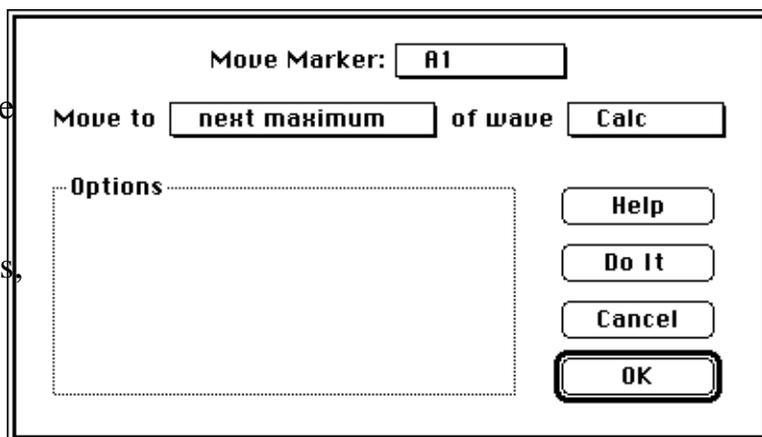
• Set the Move to pop-up to relative time X, set the X offset edit field to +0.050, and press OK. This instruction is now setup to move the A1 marker to the right by 50ms, before analyzing the vowel.

Now lets move the right edge marker to the left.

• Click once on your new Move Marker instruction to select it, choose Copy under Edit to copy it to the clipboard, and then choose Paste under Edit to paste it into the task, at the position of the dotted line, which is conveniently right before the Sound Statistics instruction.

• Double-click on your new instruction, to open it's dialog, and then select marker A2 in the upper-most pop-up, change +0.050 to -0.050, and then press OK. This instruction is now setup to move the A2 marker to the left by 50ms, before analyzing the vowel.

• Press the Go button at the upper right to run the task in debugger mode. A triangle will appear at the left (▶) that shows which instruction is being executed. After a few seconds, press the Step button to temporarily halt the execution.

[Go] [Step] [Stop]

- Now press the W button, at the upper-left, to open the ⌈C⌉ ⌈J⌉ ⌈M⌉ ⌈S⌉ ⌈V⌉ ⌈W⌉
  Wave dialog. From here, one can view a wave's values
  with the Edit button, or the number of points with the Points button, for example.
  To view another wave, simply select it in the upper-most pop-up menu. Press
  OK to exit this dialog.

The C, J, M, S, V and W buttons are used to view and modify controls, journals, markers, strings, variables, and waves; during, before, or after running a task. This is helpful when a "problem" is encountered. These buttons cause a new object to be created if one does not already exist; otherwise they open the dialog for the first item in the corresponding object list.

- Press the Step button to execute another instruction. The (▶) will increment one
  line. Press Step several times, and then press Go to let the task run free. When
  the task is finished, exit the Task Editor by pressing OK.

Another form of debugging is to send information to a journal at run time. Several journal reporting features are available, and fully described in *The Menubar* Chapter of the SoundScope & SuperScope II Reference Manual.

- Choose New under Journal to create a new journal. When the journal dialog
  appears, change the journal's name to Task80; and then set the Position pop-up
  menu to Window. Press OK to open the new window-based journal.

Clicking in the front panel will cause the Task80 journal to become the back-most window. To bring it to the front, choose Show ▶ Task80 under Edit.

- Choose Run ▶ Analyze under Task to run the
  task. Each instruction will be written to the
  Task80 journal, as it is executed; as shown to the
  right. It will beep when finished (this task may
  take a little while).

- When the task is done, click once in the journal
  to select it's window.
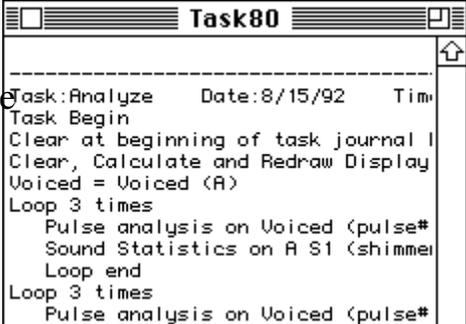
```
▤□▭▭▭▭▭▤ Task80 ▤▭▭▭▭▭▭ ▱▤
                                        ⇧
-------------------------------------
Task:Analyze      Date:8/15/92    Tim
Task Begin
Clear at beginning of task journal |
Clear, Calculate and Redraw Display
Voiced = Voiced (A)
Loop 3 times
    Pulse analysis on Voiced (pulse#
    Sound Statistics on A S1 (shimmel
    Loop end
Loop 3 times
    Pulse analysis on Voiced (pulse#
```

- Choose Cloverleaf 'a' (a) to select the entire contents of the journal, and then
  press the Delete key to erase it.

- Choose Copy Task ▶ Analyze under Edit to copy a textural description of the task
  to the clipboard. Click once in the Task80 journal window and then choose Paste

---

under Edit. A description of the task will be dumped to the journal.

- Now choose Clear ▸ Task80 under Journal to clear the journal of it's contents (notice this is similar to ⌘ a), press *Option* 'a' to copy a textual description of the entire instrument to the clipboard, click once in the Task80 journal, and then choose Paste under Edit. A description of the instrument will be dumped to the journal. Use the scrollbar and resize box to view this description. For a more complete description, press *Option Shift* 'A'.

We will now create a task, create a new menu command, and then setup the menu command to run the task. For details on the Menubar Editor, please see *The Menubar* chapter in the SoundScope & SuperScope II Reference Manual.

- Choose New under Task to create a new task. Name it "myTask", and then drag the Alert, Beep or Delay instruction (near the bottom of the instruction dictionary) into the task region. Select the Display message radio, enter "Hello." into the edit field, and then press OK to install the instruction.

Alert, Beep & Delay

○ Beep
○ Stop task
○ Break out of Trace Loop
○ Wait for mouse down
○ Wait for mouse up
○ Wait for key [\r] press
○ Delay for [2.000] seconds
○ Synchonize to [1.000] second intervals
● Show Alert: [Hello.]

(Help)   (Do It)   (Cancel)   [OK]

- Press Go to see the task run. It should execute the instruction which produces the "Hello," alert. Press Continue to exit this alert. In Go mode, tasks beep upon completion.

- Press OK to exit the Task Editor.

- Choose Edit Menubar ▸ Full under Edit to open the Menubar Editor dialog.

- Choose Task in the Menu pop-up to select this menu.

- Choose New User Item in the Item pop-up to create a new menu item.

- Choose the Show custom text radio under the item pop-up to specify that the text in the adjacent edit field be placed into the menu. Type "Say Hello" into this field.

- Choose myTask in the Run Task pop-up -- to specify that this task be run when Say Hello is selected under Task.
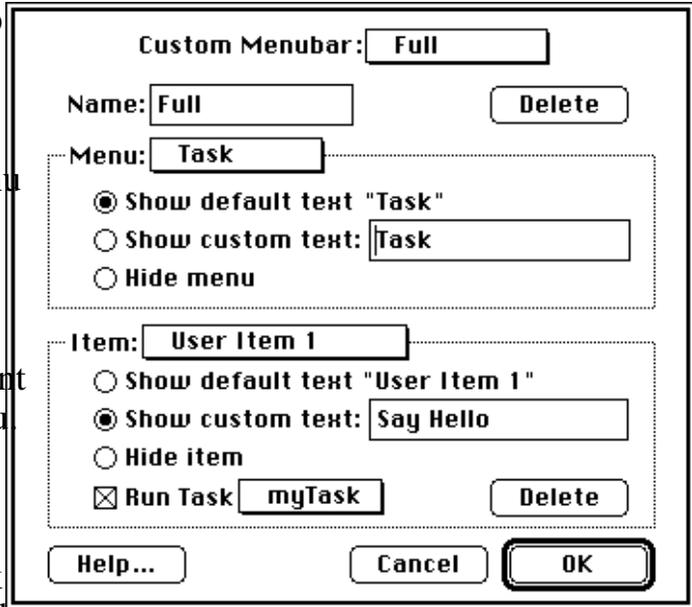
- Press Ok to exit the Menubar Editor.

- Choose Say Hello in the Task menu to view your creation. An alert should appear, as illustrated to the right.
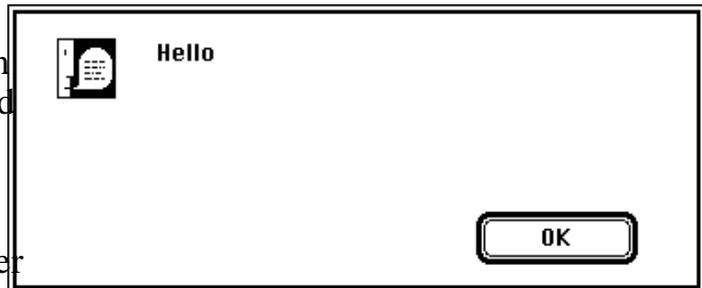
- Now choose Save As under File to save this new instrument under a new name, such as "myTriVowelgram". Consequently, double-clicking on the new file will open this instrument.

Notice that the Menubar and Task Editors enable you to build virtually any menu driven application program, without knowledge of a programming syntax. This is new in the world of computing, and is something that we are very excited about.

This concludes the first round of the Advanced Tutorial. For a tour of QuickTime, HyperCard XCMDs, and database work; please continue.

Instruments   **3 - 9**

QuickTime & HyperCard XCMDs

In the 1970's, the world developed computing tools for handling text. The 1980's was the decade of graphics (e.g. the Macintosh user interface). Many expect the 1990's to be the age of moving graphics (video). It is also theorized that this will be as influential to our society as the revolutions in text and graphics. Apple's technology for recording, playing, editing, compressing and databasing computer videos is commonly referred to as "QuickTime". Many tools that support QuickTime movies are available from Apple and Third Party Developers. SoundScope can play and plot pre-made QuickTime movies via several QuickTime XCMDs.

XCMDs are the programming port-hole through which HyperCard communicates with the world. Physically, they are little functions whose code resides in a file on disk. They are used primarily by HyperCard to control multimedia devices such as tape recorders and framegrabbers, yet can be used to perform virtually any function. XCMDs are callable from SoundScope, via the HyperCard XCMD/XCFN instruction, and consequently add a new dimension to SoundScope programming.

An example of an XCMD is FinderEvents, by Jon Pugh. This XCMD sends messages to the Finder and is distributed non-commercially by user-groups and computer networks. The XCMD is stored in a HyperCard stack file called "FinderEvents", illustrated to the right. The stack contains documentation on how to call the FinderEvents() function. It explains that the XCMD has one parameter that tells the Finder what to do. For example, executing *FinderEvents "AboutMacintosh"* from HyperCard or SoundScope instructs the Finder to show the "About This Macintosh" dialog. To install this XCMD into SoundScope, the user must place a copy of the "FinderEvents" file into the "HyperCard XCMD/XFCN" folder, inside the "Resources" folder, which must be at the same level as SoundScope, and then restart SoundScope. Subsequently, one can create a SoundScope instruction
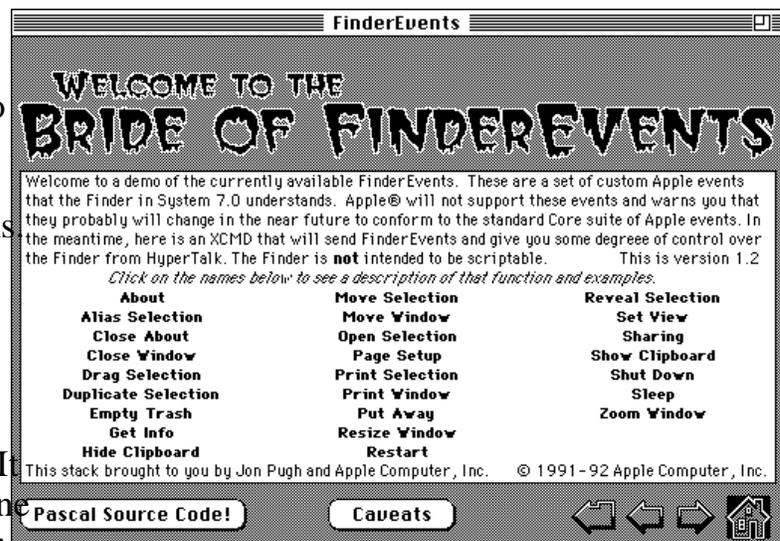
that opens the About This Macintosh
dialog by typing *FinderEvents*
"AboutMacintosh" into the HyperCard
XCMD/XFCN instruction. XFCN are
identical to XCMD, except that they
always return information into a
SoundScope string named "retValue".

```
 About This Macintosh

                          System Software 7.0.1
 [==] Macintosh IIfx       © Apple Computer, Inc. 1983-1991

 Total Memory:   20,480K   Largest Unused Block:   2,739K

 [■]  'SS2.ɲ       3,000K   ████████████████▏
 [■]  'THINK C 5   3,200K   ███████████████▏
 [◉]  HyperCard    4,000K   █████████▏
 [▥]  MacDraw II     801K   ███▏
```
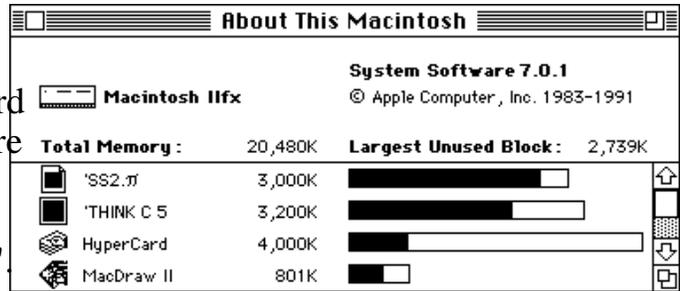
Thousands of XCMDs are available commercially and as shareware. Several Apple
authored XCMDs that play QuickTime movies are included with SoundScope. For
details on XCMDs and the SoundScope-to-XCMD interface, please see the
Instructions chapter in the SoundScope & SuperScope II Reference Manual.

- If you did not install the *QuickTime*, *XLink Extensions*, *XLink Manager*
  Extensions; and the *XLink Setup* Control Panels, please do so now. Playing
  QuickTime movies within SoundScope requires that the *QuickTime* extension be
  installed in your System folder; however, this extension is not included with
  SoundScope. Therefore, you must acquire this from your dealer if you desire
  this feature (it's not needed in most applications). Installing extensions requires
  exiting SoundScope and installing several files into your System Folder, as
  described in the System Extensions Installation discussion at the beginning of this
  manual. These extensions and QuickTime itself consumes some memory;
  therefore, if your memory is already low, don't be surprised if you hit a memory
  error. Make sure you Restart your computer and then launch your
  TriVowelgram after installing these extensions.

- Select Choose Menubar ▶ Real Short in the Edit menu, if it is not already selected.

- Choose Open under QuickTime. A file open dialog will appear. Select a random
  quicktime movie file that is on your computer (several are included with
  QuickTime).

- Now choose Play under QuickTime to play the movie in the left-most LPC
  display. QuickTime movies are beautiful when run on ≥8bit color graphics
  systems. Conversely, they tend lack appeal when run in black-and-white
  environments. Additionally, QuickTime recordings sound better on faster
  machines (≥ 25MHz 68030 processor).

  If you do not see a movie and want to debug the situation: select Choose Menubar
  ▶ Full under Edit, choose New under Journal, select Window in the Position pop-
  up menu, set the journal name to "TaskView", press OK, and then choose Run ▶
  PlayMvie under Task. This will cause the task to run while debugging
  information is sent to the new journal. The right most column in the printout

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will show the value of the retValue string after each instruction. This string is loaded with error information by the QTMovie() XCMD, and may provide a clue to the problem.

- Now choose Plot under QuickTime to plot the movie frames across the screen.

Now lets take a look at the tasks that perform this magic.

- Select Choose Menubar ▸ Full in the Edit menu to enable the complete menubar.

- Choose Edit ▸ OpenMvie under Task to view the task which determined the file pathname to the QuickTime movie. The Open command under QuickTime in the Real Short menubar invokes this task.

The first line in this task calls the FileName() XFCN which opens the standard file open dialog. This function has one argument, a four character string

```
Task Begin
FileName("MooV")
SetGlobal "fileName", retValue
```

that specifies which type of file is shown in the file dialog. QuickTime files are of type "MooV", text files are of type "TEXT". XFCNs always return their results into a string named "retValue", and in the case of FileName(), a file pathname is copied into this string. For example, if the "Resources" folder resides at the top level of a hard disk labeled "myHD", the pathname could be "myHD:Resources:QuickTime Movies:Glottis". The second line in the task calls another XCMD that creates a string named "fileName" (if it does not already exists), and then copies the contents of the retValue string into fileName. We copy the contents since another XCFN may overwrite the retValue string and we need to hold onto the pathname.

- Choose PlayMvie in the upper-most pop-up menu to view the task which plays the QuickTime movie. The Play command under the QuickTime menu in the Real Short menubar invokes this task.

The first line in this task calls the QTMovie() XCMD, which is included in the HyperCard Development Kit and the QuickTime Developer Kit.

```
Task Begin
QTMovie "OpenMovie", Direct, fileName, L_Snap_D, FastIdle, loop
SetGlobal "movieID", retValue
QTMovie Direct, movieID, Idle
QTMovie Direct, movieID, Dispose
```

- Double-click on the first QTMovie() line to open the HyperCard instruction dialog. This instruction executes XCMD and XFCN placed into the "HyperCard XCMD/XFCN" folder (inside the "Resources" folder) via the *XLink Extensions* Extension, *XLink Manager Extensions* Extension and the *XLink Setup* Control Panel.

HyperCard XCMD/XFCN

QTMovie "OpenMovie", Direct, fileName, L_Snap_D, FastIdle,loop

Available Objects

(D)isplays:	Time_D	6, 386, 635, 459
(J)ournals:	Date	6/10/92ªÅ
(M)arkers:	A1	0.120058
(S)trings:	message	
(V)ariables:	error	0.000000
(W)aves:	Calc	
XCMD/FCN:	PostTempo	

Help   Do It   Cancel   OK
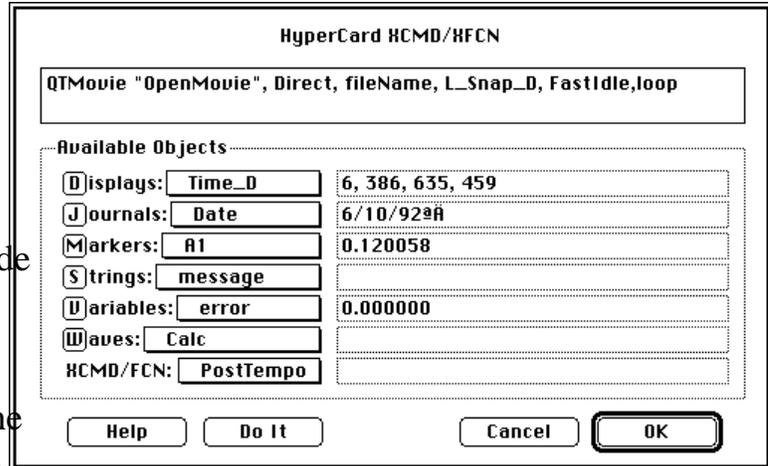
- Press on the lower-most pop-up menu to view which XCMDs are installed in the "HyperCard XCMD/XFCN" folder. Each XCMD is stored in a resource, and many resources can be placed into one file.

The user types a line of text into the upper-most edit field, which is executed at run time. SoundScope objects (e.g. displays, journals, markers, strings, variables) can be passed as parameters to XCMDs. Since HyperCard is string based, these objects correspond to a line of text, displayed to the right of each object's pop-up. For example, in the above illustration, display Time_D is shown as representing a string set to "6, 306, 635, 459". Subsequently, passing Time_D as a parameter is like passing "6, 306, 635, 459" as a string. The object pop-up menus select which object is represented in the text region to the right of each pop-up. The object buttons (e.g. D, J, M, S) open the corresponding object dialogs.

The first QTMovie() XCMD opens a movie file at the fileName location and readies it for play. A movie identification number is returned in retValue, which is subsequently copied into movieID with a SetGlobal XCMD. The third instruction tells the movie to play until the mouse button is pressed, and the forth instruction releases memory used by the movie.

- Press Cancel to exit the HyperCard instruction, and then press OK to exit the Task Editor.

This concludes our tour of QuickTime and the XCMD interface. For more details on this topic, please refer to the HyperCard XCMD/XFCN discussion in the *Instructions* chapter of the SoundScope & SuperScope II Reference Manual.

Database Access

In many cases, it is desirable to save the results of an experiment into a database, spreadsheet, or word processor; for later viewing, analysis, and comparison. A FileMaker database that does this is included with the TriVowelgram instrument. One record is created for each three vowel measurement; subsequently they can easily be viewed, printed, and sorted. This system gathers and stores data in the same way for each trial, and can therefore be used to monitor a subject over time, or to compare multiple subjects. For an example of a printout from this database, please see the TriVowelgram description in *Chapter 5, Instruments*.

- Choose Run ▸ Analyze under Task to execute the three vowel analysis. It will beep when finished (this task may take a little while).

- Choose Show ▸ Table under Edit and then choose Show ▸ Export under Edit to open the Table and Export window-based journals.

- Choose Run ▶ Export under Task, and when the file dialog appears (it might take a little while), save the text under the name "Export". This task composes a one line string with data from the various journals and then saves it to disk. The string contains the following fields separated by commas: firstName, lastName, examDate, vowel1, vowel2, vowel3, shimmer1, HNR1, breathiness1, jitter1, FoAvg1, FoRange1, shimmer2, HNR2, breathiness2, jitter2, FoAvg2, FoRange2, shimmer3, HNR3, breathiness3, jitter3, FoAvg3, FoRange3.

	Table				
	A S1	A S1	A S1	A S1	A S1
	shimmer (APQ)	HNR	breathiness	jitter (RAP)	f0 avg
Vowel #	%	dB	dB	%	Hz
1	4.995	1.920	-6.220	1.309	199.879
2	4.160	NAN(021)	-19.992	0.904	200.041
3	4.535	14.876	-21.906	0.699	200.105

Export

Bart,Simpson,6/10/92,a,e,o,4.995,1.920,-6.220,1.309,199.879,2.500,4.160,NAN(021),-19.9

- Choose Edit ▶ Export under Task to view the task that implemented the export. To gain a sense of the string handling capabilities in SoundScope, double-click on a few instructions, pressing Cancel to exit each instruction's dialog. Press OK to exit the Task Editor.

If FileMaker Pro is installed on your hard disk and you want to see database import at work, please continue; otherwise, we hope you enjoyed the show and request that you phone us at 617/625-4096 if you have any questions. If you are currently running the Demonstration version of SoundScope and want more information, please contact us and request a FREE copy of the SoundScope & SuperScope II Reference Manual.

- Double-click on the TriVowelgram™ Database file; or launch FileMaker Pro, and then choose Open under File to load the database.

- Choose Import under File, select the Export text file (i.e. the one that we just saved), and then press OK when the Specify Field Order dialog appears, as illustrated to the right. A new record will automatically be created, and its fields will automatically be loaded with the data in the text file.

Specify field order for import

Data In: "Export"		Fields In: "TriVowelgram™ D..."
Bart	→	First Name
Simpson	→	Last Name
6/10/92	→	Exam Date
a	→	vowel.1
e	→	vowel.2
o	→	vowel.3
4.995	→	shimmer.1
1.920	→	hnr.1
-6.220	→	breathiness.1
1.309	→	jitter.1
199.879	→	fo avg.1
2.500	→	fo range.1
4.160	→	shimmer.2
NAN(021)	→	hnr.2

[<<] Scan Data [>>]   Record 1   [Cancel]

◉ Add new records   [OK]
○ Replace data in current found set

	A	B	C	D	E	F	G	H	I
1	Bart	Simpson	6/10/92	a	e	o	4.995	1.92	-6.22

Text is the universal language with computers. It is easily transferred between SoundScope, word processors, HyperCard, and spreadsheets. Opening our Export text file with Excel (i.e. a popular spreadsheet authored by Microsoft Corp), with commas separating each cell, causes our data to be placed into one row, as illustrated above. When loading or saving spreadsheet text, each column is separated by a tab or comma, and each row is separated by a carriage return.

To copy a display from SoundScope to a FileMaker picture field, one would choose Copy Display under Edit in SoundScope (to copy the display to the clipboard), click once in the FileMaker window, place the insertion point into the picture field, and then choose Paste. FileMaker Pro, a Claris Corporation product, is probably the best flat file database in the world. One neat feature is its support of simultaneous access over a network from both PC's and Macintosh computers.

Building a dedicated SoundScope instrument and FileMaker database is not a particularly easy task. Learning the SoundScope design features could take 1 to 3 days, learning FileMaker could take 1 to 2 days, building a SoundScope instrument could take 1 to 10hrs, and building a FileMaker database could take 1 to 8hrs; assuming there are no major "snags". Our commitment to database access is strong, and we will continue to develop new and improved methods for communicating with other applications, in our quest to create the world's most advanced recording tool.

Chapter 4
Instrument Design

It is said that, in the future, programmers will not program (in the traditional sense) due to the development of sophisticated software that facilitates programming by the end user, "programming" being defined as the act of building an application program. SoundScope is another milestone in this evolution since it provides an environment where the end user can design their own menu driven application program. We refer to these applications as SoundScope "instruments", and have included several with the product.

SOUNDSCOPE OBJECTS

The SoundScope user creates and customizes objects with mouse driven pull down menus and dialog boxes -- no previous programming experience is necessary. There are several types of objects, as shown below. One can create as many of each type as needed to build the application of their dreams. Each type has it's own dialog where one can view and edit attributes, such as the object's name. This chapter summarizes these powerful building blocks.
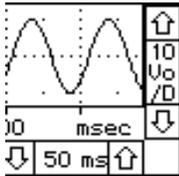
Waves	Displays	Markers
Journals	Controls	Menubars
Strings	Variables	Tasks
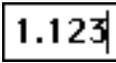
WAVES

Waves are used to represent real world continuous data as a list of values, which when plotted, produce a waveform. Waves can be digitized, synthesized, analyzed, edited, viewed, used to hold the results of analysis, loaded from disk, saved to disk, sent to the clipboard as a column of numbers in text format, and sent to the clipboard as a graphical image. Typical instruments have 3 to 10 (or more) waves and wherever you see a squiggle in SoundScope, you are looking at a Wave object. Waves are stored in memory as a list of 16-bit integer (±32,768) or 32-bit floating point values and their maximum lengths are limited by memory (each point consumes two or four bytes depending on the storage format).
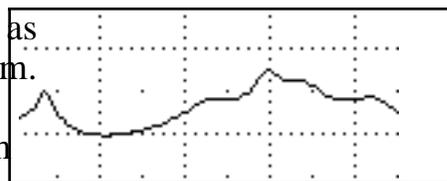
Waves are created by choosing New under Wave. The wave dialog appears and it is from here that one can view and edit the number of points (via the Points button), view and edit the internal storage format (via the Format button), view and edit the individual wave values (via the Edit button), or synthesize.

The **Value Editor** is a spreadsheet-like environment that shows the individual points with 5 points displayed in each row. Cells can be selected, cut, copied and pasted with the mouse. Additionally, selected cells can be edited numerically with the keyboard. Wave segments on the clipboard are stored as a column of numbers in a text format and can easily be transferred to a graphics, spreadsheet or word processing program.

The **Wave Synthesizer** loads a wave with a constant value, ramp, sine wave, triangle wave, square wave, uniform noise (flat histogram), or gaussian noise (gaussian distribution).

One can easily **edit a wave** by moving wave segments to and from the clipboard via the standard Macintosh Cut, Copy and Paste commands. The user identifies a segment by dragging the mouse over the region of interest. The insertion point is positioned by clicking the mouse once. Cut, Copy and Paste commands correspond to the selected wave (a wave is selected by clicking on its wave label; or, if there is only one wave in the display, clicking in the display).
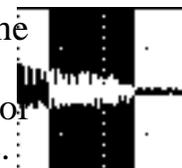
DISPLAYS

Displays reside on the front panel and are used
to view waveforms and perform calculations.
They are extremely versatile with many
customizable attributes such as
horizontal/vertical scroll/position controls,
labels, waves, markers and much more.
Displays can be positioned on the front panel in
any pattern and in any number, space
permitting. Each display can contain up to 8
waves and supports mouse-driven cut/copy/paste of waveform segments, drawing of
waves, and logging wave values to a journal.
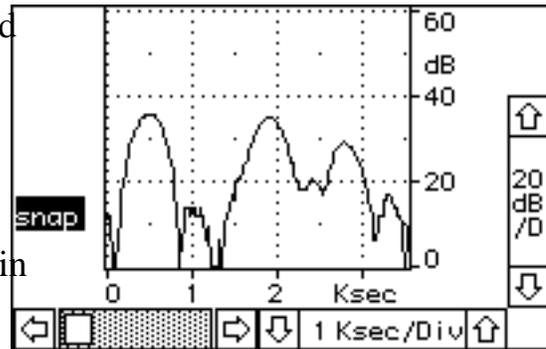
To create a display, one must choose New under Display. Subsequently, the display
dialog appears and it is here that one specifies which waves are to be displayed (this
is done by dragging wave names from the waves list box to the display contents
box). There are four display types: **Wave Plot**, **XY Plot**, **Analysis** and
Snapshot. Wave Plot is used to plot 1 to 8 waves across the screen, XY plots 1 to 8
wave pairs, Snapshot analyzes (e.g. frequency spectrum) a timewave at a marker
position, and Analysis analyzes a segment of a timewave. Customizable labels,
controls, axes, grids, plot graphics and markers are easily adjusted with dialogs
accessed by choosing Features, Labels, Controls, and Markers under Display.

The user can easily **redraw** a waveform segment by placing SoundScope
in Draw mode (choose Mouse ▶ Draw under Display) and then drawing in
a Display with the mouse.

The user can **vertically re-position** a waveform by placing
SoundScope in Vertical Adjust mode (choose Mouse ▶ Vertical
Adjust under Display) and then dragging the selected wave up or
down in a display. A double headed arrow is appended to the wave name of a
vertically adjusted wave to indicate that it no longer corresponds to the vertical scale
at the right of the display region. Clicking this double arrow once will cause the
vertically adjusted wave to snap back into its original position.
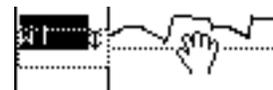
JOURNALS

Journals are text regions that are used to enter, view and edit text in a manner similar to that done with a word processor. With commands in the menubar, the user can Clear, Save, View, Save to Clipboard, Print, Delete and Create Journals. The contents of Journals can easily be saved to disk and then loaded by a word processor, graphics, or spreadsheet program. Journal windows can be resized and positioned on the front panel in any pattern and in any number, space permitting. Many task instructions transfer text to and from journals.
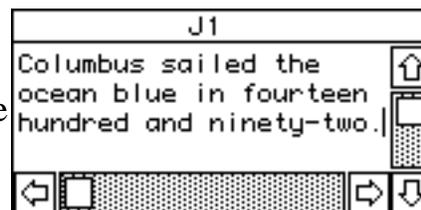
To create a Journal, one chooses New from the Journal menu. Subsequently, the Journal dialog opens, and it is from here that one specifies several Journal options such as journal name, the number of characters between each tab, and whether or not the journal's contents are stored with the instrument file.

Each journal can operate in one of two modes: *Word Processor* or *Spreadsheet*. The Word Processor mode works with a fixed page width (e.g. 7 inches) and supports automatic wrap-around. This mode is useful when one wants to send the journal contents to a word processor or print the journal with a limited width format. The spreadsheet mode is more like a traditional spreadsheet with an unlimited width, no wrap-around, and horizontal scrolling that will go as far right as necessary. This is useful when it is necessary to send the journal's contents to a spreadsheet. Tabular numeric data in SoundScope journals transfer nicely to spreadsheets, with each value being placed in its own spreadsheet cell. This is made possible with a TAB delimiter between each column and a CARRIAGE RETURN character between each row. This is a wonderful standard for text files since it insures compatibility between journals, word processors and spreadsheet application programs!

The user can **log points to a journal** by placing SoundScope in Log Coordinates Mouse mode (choose Mouse ▸ Log Coords under Display) and then clicking the mouse on a selected wave (recall that you select a wave by clicking on its label; or, if there is only one wave in the display, clicking in the display).

CONTROLS & INDICATORS

Front panel Controls and Indicators, shown below, allow the adjustment of, Boolean true/false values, scalars, lists, and text. These objects appear in a variety of styles, sizes, fonts, and colors; and their states are easily read and updated with tasks.

Lights indicate the state of a Boolean true/false value.	**Buttons** often initiate the execution of a task.	**Switches** specify a true/false Boolean condition.
Sliders control a scalar quantity.	**Dials** control a continuous or quantized scalar quantity.	**Meters** indicate a scalar quantity.
Numeric Fields control a scalar quantity or a text string.	**Static Text** shows a scalar quantity or a text string.	
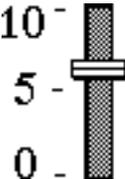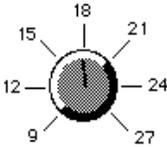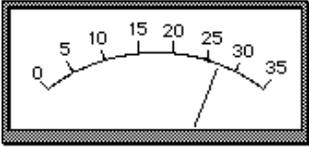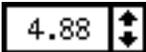
MARKERS

Markers are typically used to mark a time in a wave or display. The user can create as many markers as he/she desires and can place any marker in any display (via Markers under Display). In displays, they appear as vertical lines that can be moved left and right with the mouse. Additionally, the display can be set up to show marker labels at the top of the display (choose Label under Display). A neat trick is to use two markers and one wave to describe another wave, called a "segment", which is defined as the section of the original wave between any two markers (via Markers under Display). The segment and the source wave share the same data.

The **cursor window** is used to numerically view specific points along a waveform. The contents of the cursor window corresponds to the location of the mouse. This is opened by choosing Show ▸ Cursor under Edit.

Wave	Horizontal	Vertical
mouse	3162.500 dB	22.34375 dB
snap	3162.500 dB	9.540 dB

MENUBARS

Menubars are a powerful tool for modifying the current menubar, or creating your own. In some cases, it is desirable to have a very simple menubar to limit a user's options. Menubars are easily created, renamed, and deleted. Each menubar consist of a set of menus and a set of items for each menu. Each menu and item can be renamed (e.g. change "Open..." to "Open Sesame"), hidden, or set up to run a Task when chosen.

Custom Menubar: MyMenu

Name: MyMenu   Delete

Menu: File
  ⦿ Show default text "File"
  ◯ Show custom text: File
  ◯ Hide menu

Item: User Item 1
  ◯ Show default text "User Item 1"
  ⦿ Show custom text: Open Movie...
  ◯ Hide item
  ☒ Run Task OpenMovi   Delete

Help...   Cancel   OK

TASKS

Tasks are little program sequences that perform a series of operations when run. For example, one could write a task to record data, analyze, update the screen, and then print the results. Tasks are easily created, viewed, edited, and debugged; and can be set up to run when a marker moves, when a wave changes, when a

```
Task Begin
Clear at beginning of task journal Data
Clear, Calculate and Redraw Display Analysis
Voiced = Voiced (time)
Move M1 to absolute time X of time
Move M2 to absolute time X of time
Loop 3 times
    Pulse analysis on Voiced (pulse# to Data; t3
    Sound Statistics on Segment (shimmer (APQ) to
    Loop end
Move M1 to first peak of Voiced
Move M1 to relative time X of Voiced
```

specific menu item is chosen or when the user chooses Run Task. One "programs" tasks using a powerful mouse/dialog user interface. The neat thing about programming SoundScope is you do not need to know any syntax -- the mouse-driven dialog boxes take care of you.

INSTRUCTIONS

Instructions are the building blocks used to create tasks. There are over 30 kinds of instructions, each of which is used to perform a specific function (e.g. save a wave to disk, move a marker, choose a menu item, calculate the derivative of a wave, etc.). Each task contains a list of

```
----------- Analysis -----------
Calculate Wave
Pulse Analysis
Sound Statistics
Statistics
---------- Data Transfer ----------
Datapipes
Disk I/O
```

instructions that are executed in the order they appear in the list; and each instruction can be viewed, edited, cut, copied, and pasted.
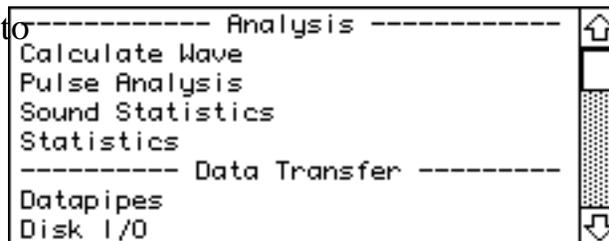
DATAPIPES

A datapipe is a reference to a folder on a hard disk or floppy disk. Technically, a datapipe is a pathname to a folder. Think of it as a pipe, through which you squirt data between SuperScope II and a folder on disk. Datapipes are often used to access folders from within tasks, where the name of the folder may change with each experiment. The Datapipe instruction can be used to create a new data folder, and to attach a datapipe to that folder. The Disk I/O instruction could then be used to push data into the datapipe, and ultimately into the new folder. Datapipes, like other SuperScope II objects, are created, deleted, and named. To view or modify a datapipe's folder, choose Datapipe Folder ▶ under File.

VARIABLES

Variables are used to hold one 32-bit floating point value (e.g. 16, 2.3, 1.34e6). They are easily created, renamed, and deleted; and their values are easily viewed and edited. Many task instructions transfer values to and from variables.
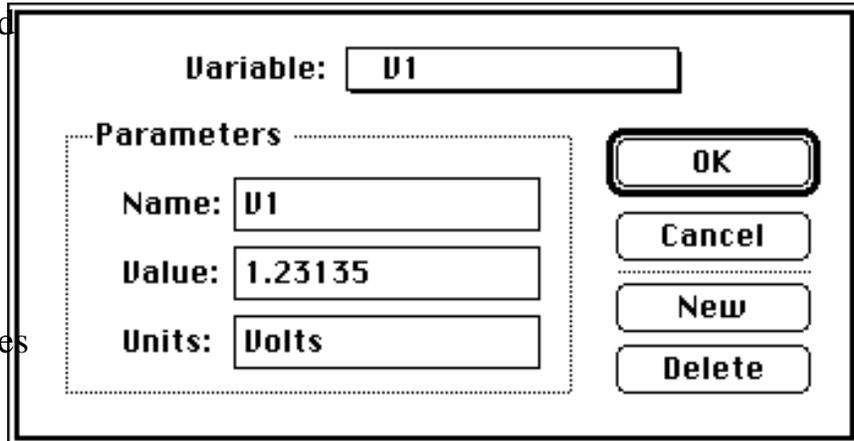
Variable: V1

Parameters

Name: V1

Value: 1.23135

Units: Volts

OK

Cancel

New

Delete

STRINGS

Strings are used to hold a series of characters of any length, memory permitting (e.g. "hi", "1.2"). They are easily created, renamed, and deleted; and their text is easily viewed and edited. Many task instructions transfer text to and from strings.
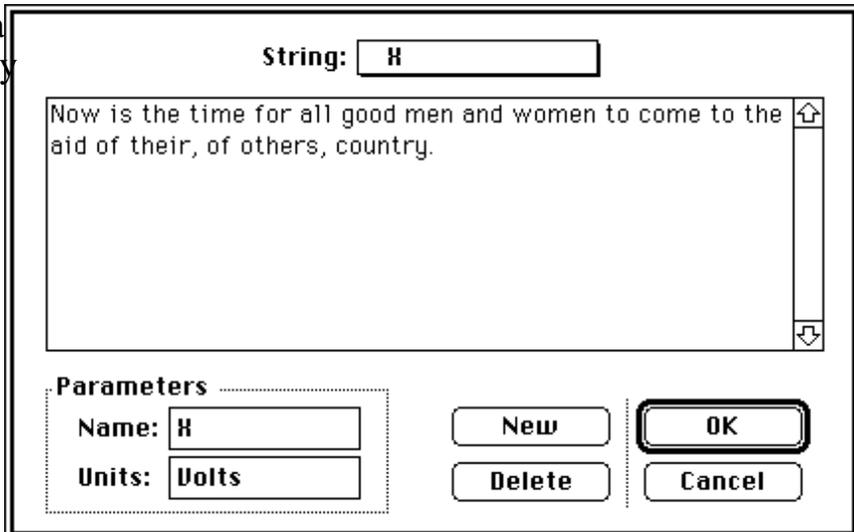
String: X

Now is the time for all good men and women to come to the aid of their, of others, country.

Parameters

Name: X

Units: Volts

New

Delete

OK

Cancel

TASKS AND INSTRUCTIONS

SoundScope provides an easy to use environment by which one explicitly defines acquisition, analysis, archival and presentation features. From this environment, one defines little programs called Tasks, each of which can be created, edited, viewed and run. These little programs consist of a list of instructions which are executed in a top to bottom sequence. Each instruction is not typed however, as done in traditional textual programming environments; instead, each instruction has an associated dialog box from which the user specifies, in detail, the instruction's functionality -- with easy to use pop-up menus, check boxes, radio buttons and edit fields. These dialogs appear when one initially adds an instruction to a task, and when one edits an instruction by double clicking on it's text. Instructions are added to a task by dragging one from the Instruction Dictionary; deleted from a task by being selected and then removed with the Delete key; moved from one position in a task to another via the standard Cut, Copy & Paste clipboard features; and modified by double-clicking with the mouse. An example instruction is the Calculate Wave instruction which enables the user to place mathematical expressions into a task (e.g. W1 = AutoCorrelation(W2)). The instructions are summarized below:

ALERT, BEEP, OR DELAY is used to sound a beep, show a message in an alert, stop the task, wait until the mouse button is pressed, wait until the mouse button is released, wait for a specific key press, freeze for a specified duration, or synchronize to a specified duration.

Alert, Beep & Delay

○ Beep
○ Stop task
○ Break out of Trace Loop
○ Wait for mouse down
○ Wait for mouse up
○ Wait for key │\r│ press
○ Delay for │2.000│ seconds
○ Synchonize to │1.000│ second intervals
◉ Show Alert: │Max FFT Low!│

[Help] [Do It] [Cancel] [OK]
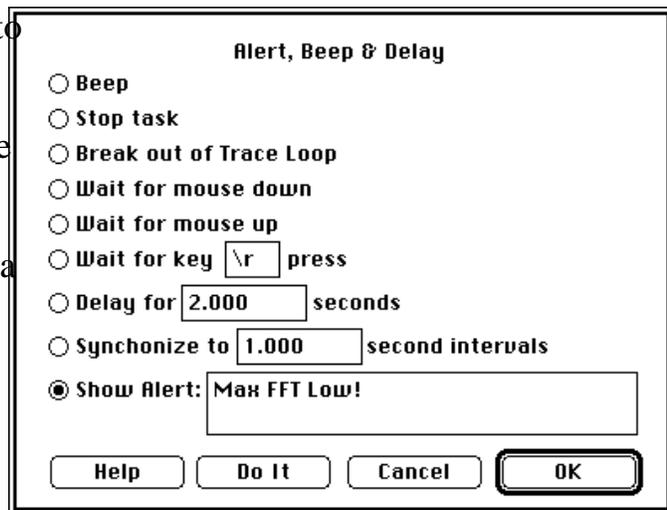
ARITHMETIC is used to arithmetically combine two objects (e.g. +, -, ÷, *) and transfer the result to a third. Objects include variables, strings, waves, markers, journals and controls.

Scalar Arithmetic

│ Journal │   ⇐   │ Marker │   │ + │   │ Wave │
│ view   │   │ M1   │   │ W1   │
[Options...]   [Options...]   [Options...]
0.00000   0.01000   0.00000 Volt

[Help...] [Do it] [Cancel] [OK]
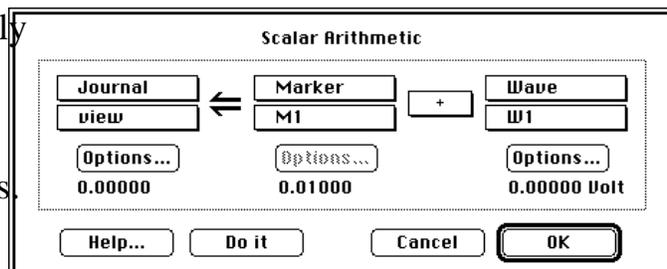
ASSIGNMENT is used to copy the scalar value of one object into another object. A marker's value is derived from it's position; a control's value is derived from it's setting; a string or journal's value is derived from it's text; and a wave's scalar value is derived from one

Scalar Assignment

Variable	⇐	Value
error		0.000

Options...   Options...

1.00000 Volts   0.00000

Help...   Do it   Cancel   OK
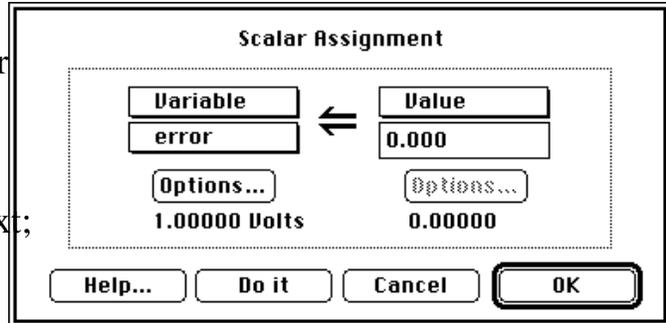
ANALOG & DIGITAL I/O is used to read and write analog and digital values to and from MacADIOS hardware.

Scalar Assignment

Variable	⇐	Value
error		0.000

Options...   Options...

1.00000 Volts   0.00000

Help...   Do it   Cancel   OK
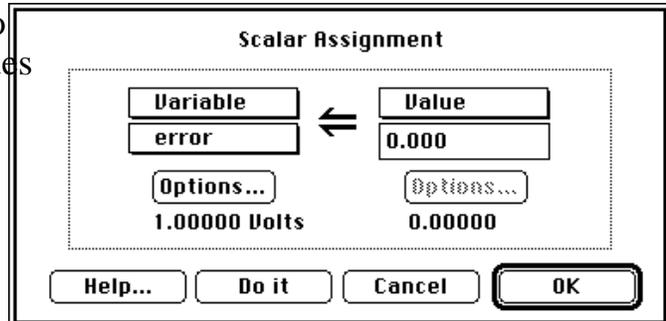
CALCULATE WAVE is used to perform waveform calculations; such as FFT, cos, +, or *. Over 80 functions and operators are supported.

Calculate Wave

○ Ain0 = Ain0

○ Ain0 = Ain0 + 0.000

○ Ain0 = Ain0 + Ain0

◉ W3 = Histogram

     Source: Ain0   Bins: 100.000

     Max: 10.000

     Min: -10.000   ☐ Use Variables

Help   View Data   Do It   Cancel   OK
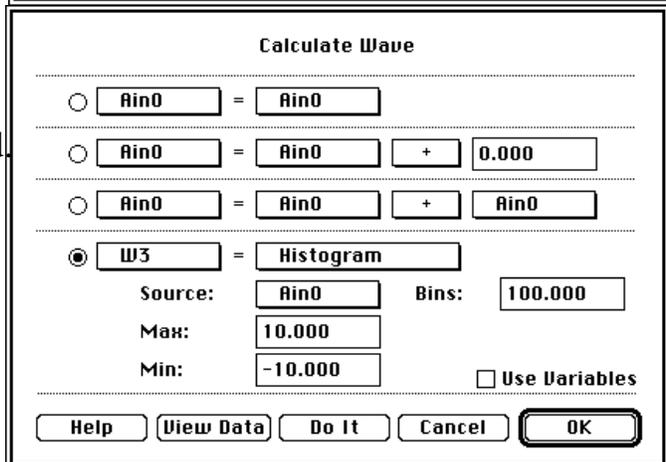
CHOOSE MENU is used to select a menubar command or press a key (e.g. choose Print under File, press *Option* 'e', etc). This is very powerful since is brings all menubar and keyboard commands within the scope of task instructions.

Choose Menu

◉ Menu: Wave

   Command: Save As...   abs

○ Key press: ☐

○ ⌘ Key: ☐

○ Open instruction number 1 in task TakeData

☐ Automatically press encountered OK buttons

Help...   Do It   Cancel   OK
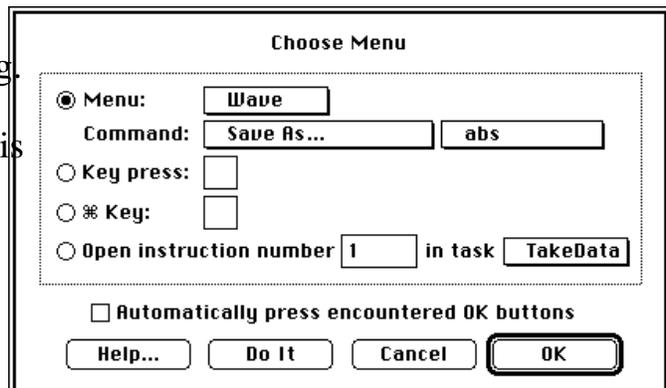
CURVE FITTING fits a wave to a lorentzian, sin, gaussian, polynomial, exponential or linear expression.

```
                    Curve Fit:  Ain0

  ┌─Method──────────────────────────────────────────┐
  │                                                   │
  │   Model:  Polynomial        # of terms:  1       │
  │                                                   │
  │   a0 + a1ʜ + a2ʜ†2 + ...                          │
  │                                                   │
  └───────────────────────────────────────────────────┘

  ┌─Results─────────────────────────────────────────┐
  │                                                   │
  │   Transfers    ☐ Replace source wave with result  │
  │                                                   │
  └───────────────────────────────────────────────────┘

      Help          Do It        Cancel         OK
```

C PROGRAMMERS can create and debug their own instructions with SoundScope Open running under Think C. For more details on this, please request the SoundScope Open Programmer's Documentation.

```
Example of an instruction written by the End User in C    More Info...

 ┌─At execute time:─────────────────────────┐        dotted
 │                                           │        rectangle
 │  • Write Hi!          to journal  Data  . │
 │                                           │       ⊠ Check Box 1
 │  • Move marker  M1    0.100  sec right.   │       ☐ Check Box 2
 │                                           │        static text 1
 │  • Add variable  error  value to wave  Fo │        peace
 │    point at the marker  M1    position.   │
 └───────────────────────────────────────────┘        Button 1

 MENU rsrc popup:  medium   line: ─────                Button 2

                              Edit Field (long):   1    ○ Radio 1
   Press To Play with Wave                              ○ Radio 2
 wave popup:  Fo             Edit field (double): 2.000 ⦿ Radio 3

                             Edit Field (text):  Hello!     OK

 Please see 'developer examples.c' and
 'Examples.rsrc' files for details on how this is done.    Cancel
```

DATAPIPES is used to show a datapipe folder dialog, create a new folder for data, or create a new folder and prompt for a folder name.

```
                    Datapipe:  DP1

  ⦿ Show datapipe's Folder Dialog
  ○ Create new folder and prompt for folder name
  ○ Create new folder with default name, don't prompt

    Help...        Do It        Cancel          OK
```

DISK I/O is used to transfer a wave or journal to or from disk, delete a file, transfer wave data to a journal, or transfer journal data to a wave.

```
                    Data Transfer

  ○ Save   wave       Fo         to disk
  ⦿ Load   journal    Data       from disk
  ○ Transfer wave   Fo      to journal   Data
  ○ Transfer journal  Data    to wave    Fo
  ○ Delete last saved file
  ○ Delete last loaded file          Do It      OK

    File...   Options...          Help    Cancel
```

DISPLAYS is used to clear, calculate, or redraw a display. This is important since updating a display takes processor time, and therefore should not be done redundantly or too seldom.

Display: time

◉ Clear display
○ Clear, Calculate and Redraw display
○ Calculate and Redraw display; don't Clear
○ Clear, Calculate and Redraw all displays which are not current with the latest data

| Help... | Do It | Cancel | OK |
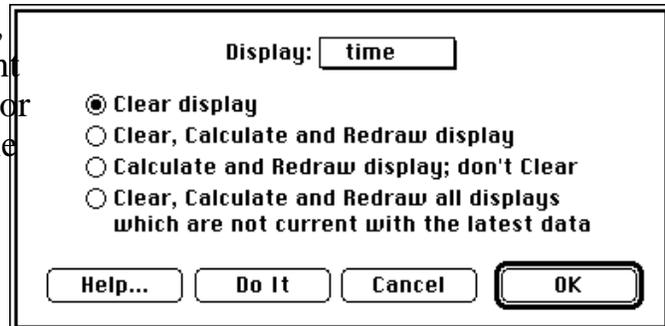
EXTERNAL INSTRUMENT is used to communicate with IEEE-488 hardware devices without programming at the 488 command level.

Instrument: Keithley 2001 DVM

Options
Function: DC Volts
Speed: Normal
Filter: Off
Resolution: Automatic
Send to: retValue
488 Address: 16

| Help | Do It | Cancel | OK |
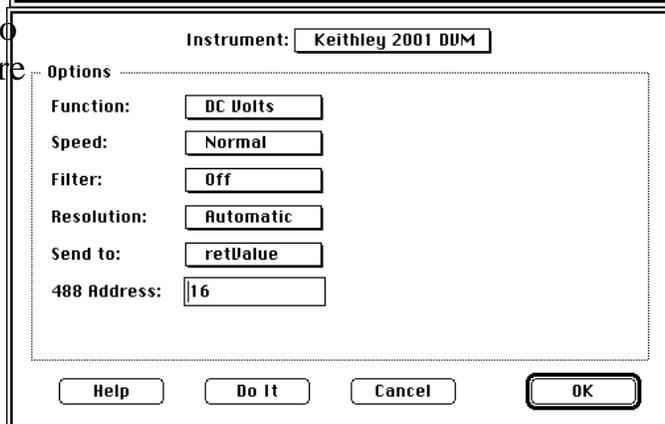
FILTER is used to run a low-pass, or high-pass FIR filter on a waveform. Alternatively, you can provide your own FIR filter coefficients to implement a custom filter.

Filter: Fo

Type: Low pass (FIR) [Custom...]
FreqCutoff: 25% (% of sample rate)
Boundries: zero pad

Freq Cutoff: 125.0 Hz   Phase: linear
Stopband: -98 dB   Taps: 99

| Help... | Cancel | OK |
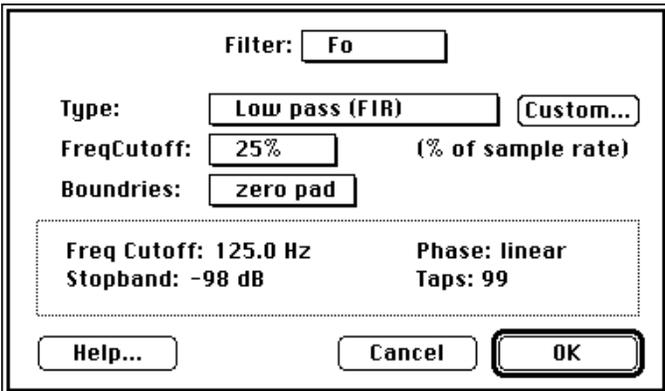
FUNCTION GENERATOR is used to control the MacADIOS fg function generator hardware. This device outputs a signal with a programmable waveform shape, frequency, amplitude, and offset.

MacADIOS fg Function Generator

Set frequency ...   With...
frequency
amplitude
offset
shape
aux D/A Voltage

☐ Read: Value 0.00000000
Current value: 0.00000 [Options...]

[W] [V] [J] [S] [C] [M]

| Options... | Do It | Clear | Cancel | OK |
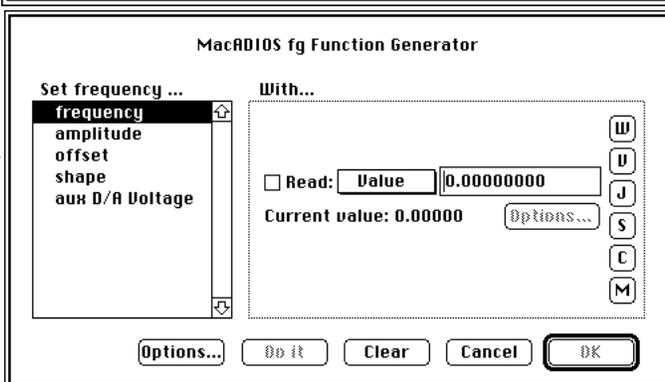
GET TIME obtains immediate time information for purposes of absolute time, and duration measurements. For example, one could use this instruction to read the time since the beginning of the task, twice, accurate to ±20μs, and then subtract the two values to produce a duration measurement accurate to ±20μs.
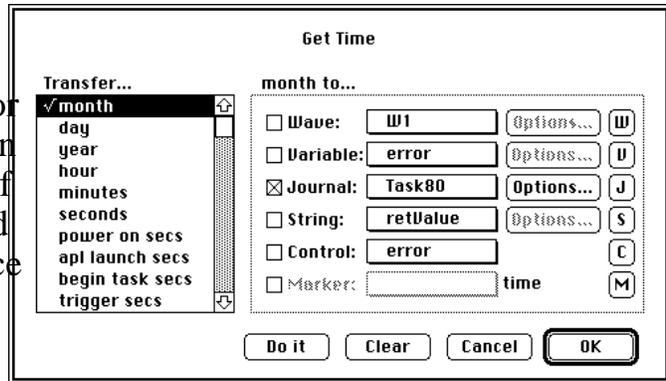
HYPERCARD XCMD/XFCN is used to call HyperCard 1.x XCMD/XFCN functions. This provides access to multimedia devices such as tape recorders and frame grabbers; yet more exciting are the thousands of public domain XCMD/XFCN routines.
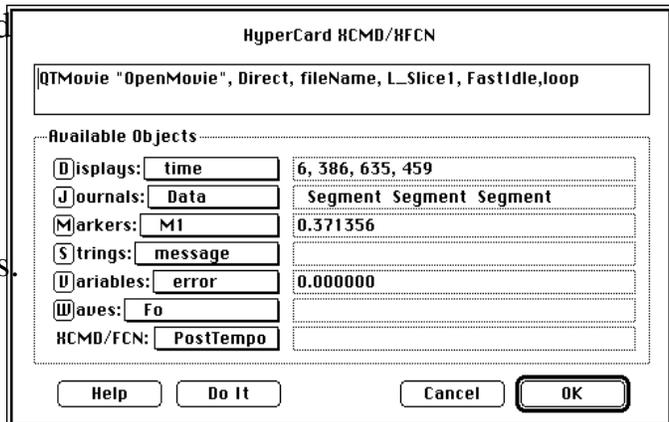
IEEE-488 is used to communicate with IEEE-488 hardware devices via MacADIOS 488s or MacADIOS 488n hardware, available from GW Instruments.
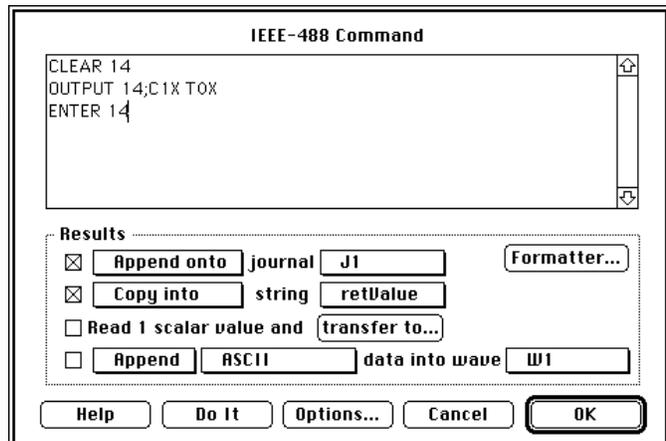
JOURNALS & STRINGS is used to clear, insert text, append, copy, and search-replace text within a journal or string.

```
Journal or String:  Task80

○ Clear                           ○ Insert wave definitions
◉ Clear at beginning of task      ○ Insert today's date
○ Copy to clipboard               ○ Insert today's time
○ Copy into:      Task80          ○ Insert trace number
○ Append onto:    Task80          ○ Update all journals
○ Remove row number        1      (top row is #1)
○ Remove column number     1      (first column is #1)
○ Change  \t               to  ,
○ Insert:  Hello.

   Help     Do It              Cancel    OK
```
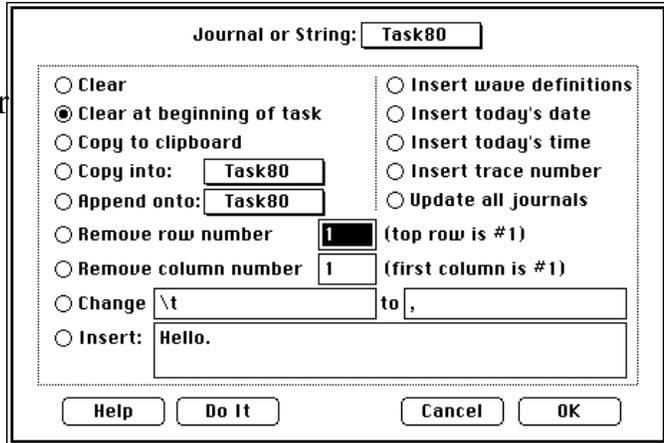
LOG MARKER is used to send the marker position, or wave value at a marker position, to journals, waves, controls, strings, variables, and markers.

```
        Marker  M1   and Wave  W1   Intersection

Transfer...              value to...
√value            ⬆      ☐ Wave:     W1      Options... W
 point#                  ☒ Variable: U1      Options... U
 time                    ☐ Journal:  Task80  Options... J
 daytime                 ☐ String:   retValue Options... S
                         ☐ Control:  error            C
                  ⬇      ☐ Marker:   M1       time     M

 Help...              Do it   Clear   Cancel    OK
```
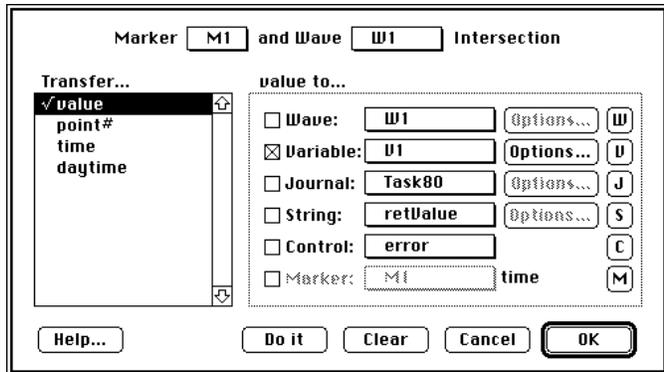
MOVE MARKER is used to move markers to waveform minimums, maximums, local minimums ("valleys"), local maximums ("peaks"), upstrokes, downstrokes, specified points, and specified wave values. This instruction supports seamless traces; subsequently, trace breaks have no effect (e.g. ten seamless traces would be viewed as one long trace).

```
        Move Marker:  M1

Move to   first downstroke   of wave   Fo

 Options                              Help
  Hysteresis:   0.010   dB
                                      Do It
  Threshold:    2.000   dB
                                      Cancel

                                      OK
```

Instruments   **4 - 15**
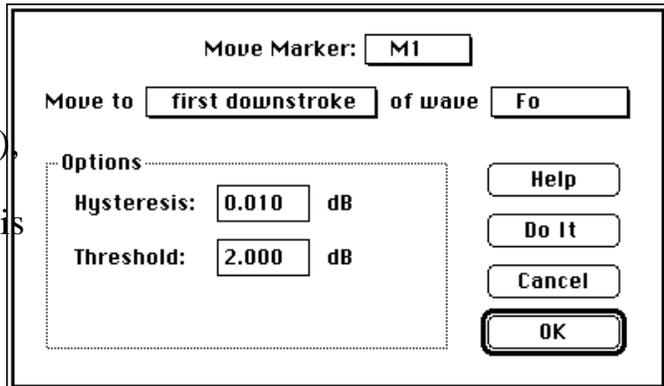
PROGRAMMING supports common programming features such as If...Then, Goto, While Loop, Loop N Times, Define Subroutine, Jump To Subroutine, Insert Comment, and Insert Label.

```
Select instruction:              Add Subroutine
Add Subroutine
Jump to Subroutine
Add Label                        Enter Subroutine Name:
Jump
Add Comment                      mySubroutine
Do Every X Traces
Do Every Y Points
Loop
If-Then
While Loop

     Help                        Cancel      OK
```
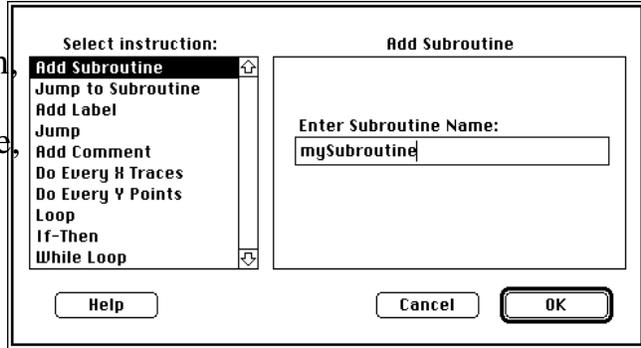
PULSE ANALYSIS is used to analyze pulses and send up to 24 characteristics (e.g. min, max, rise time, fall time, etc.) of each pulse to journals, waves, controls, strings, variables, and markers.

```
Pulse Analysis on:  W1

Send to journal:  Data              ⊠ Log pulse number
```

```
Transfers...                Do It     OK
Detection...                Help    Cancel
```
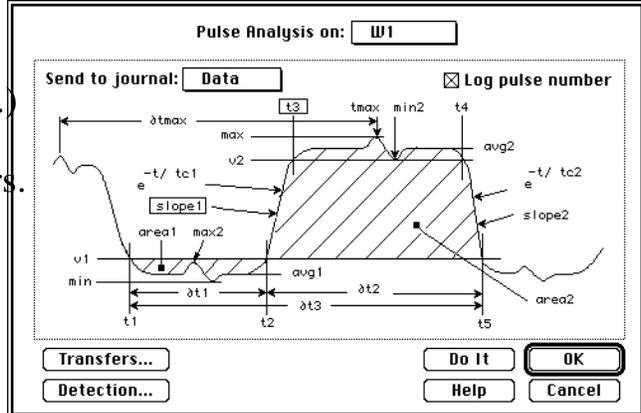
READ WAVE INTERNALS is used to extract wave characteristics, such as sample rate and # of points, and transfer them to journals, waves, controls, strings, variables, and markers.

```
Read Wave Internals:  W1

Transfer...          start time to...
√start time          ☐ Wave:      W1      Options...  W
 sample time         ⊠ Variable:  error   Options...  V
 #valid pre pts      ☐ Journal:   view    Options...  J
 #valid data pts     ☐ String:    retValue Options... S
 pre size            ☐ Control:   error               C
 data size           ☐ Marker:    M1      time        M
 post size
 min IU code
 max IU code
 min EU value

 Help...          Do it    Clear    Cancel    OK
```
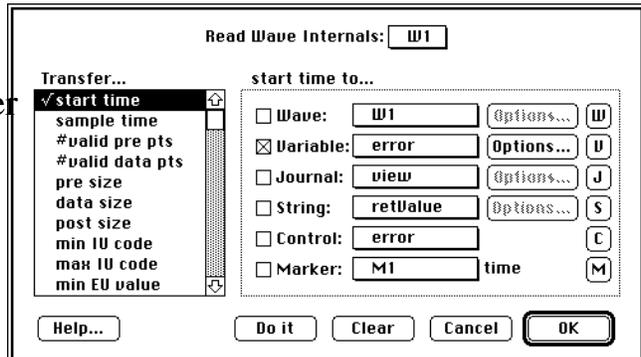
RS-232 is used to communicate with RS-232 hardware devices via the computer's MODEM and PRINTER ports. These ports support serial communication at rates up to 57.6K BAUD via the Macintosh Serial Port Driver. This instruction can be used to send text, receive text, copy a received string into a journal, and interpret a received string.

```
RS-232 Command Sequence

D

Results
⊠  Copy into    journal  J1         Formatter...
☐  Copy into    string   retValue
☐ Read 1 scalar value and  transfer to...
☐  Append    ASCII    data into wave  W1

  Help    Do It   Options...  Cancel    OK
```
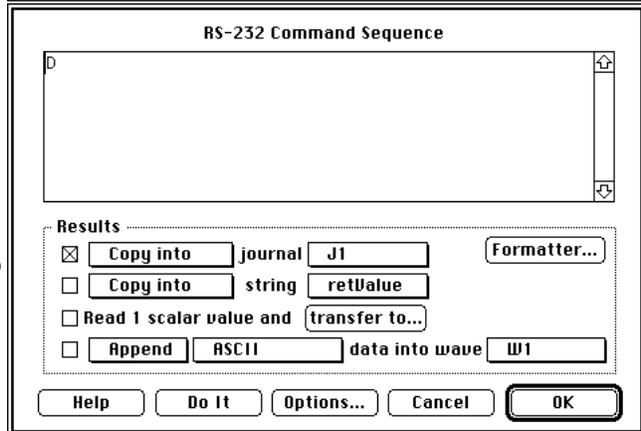
SET WAVE INTERNALS is used to set wave characteristics, such as sample rate and # of points. This is done with values from journals, waves, controls, strings, variables, and markers.

Set Wave Internals: `W1`

Set sample time ... | With...
- start time
- √ sample time
- #valid pre pts
- #valid data pts
- pre size
- data size
- post size
- min IU code
- max IU code
- min EU value

⊠ Read: `Variable` `X`

Current value: 0.00000 `Options...`

[W] [V] [J] [S] [C] [M]

`Do it` `Clear` `Cancel` `OK`
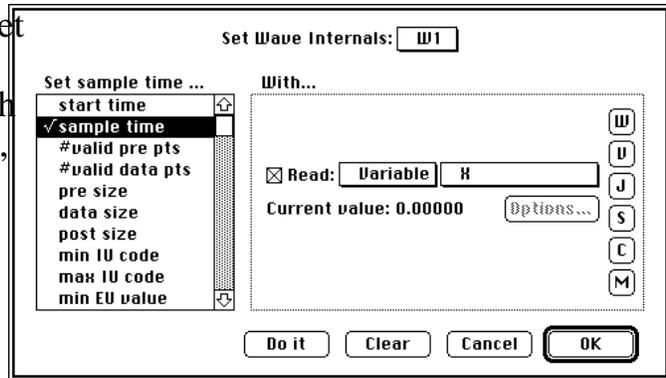
SOUND STATISTICS is used to calculate the following statistics on a speech wave: % voiced, % unvoiced, % silent, shimmer, jitter, HNR, breathiness, Fo average, Fo kurtosis, Fo skewness, Fo range and Fo std dev.

Sound Statistics on: `W2`

Transfer... | silence to...
- voiced
- unvoiced
- √ silence
- shimmer (APQ)
- HNR
- breathiness
- jitter (RAP)
- f0 avg
- f0 kurtosis
- f0 skewness

☐ Wave: `W1` `Options...` [W]
⊠ Variable: `error` `Options...` [V]
☐ Journal: `view` `Options...` [J]
☐ String: `retValue` `Options...` [S]
☐ Control: `error` [C]
☐ Marker: `MI` time [M]

`Help...` `Options...` `Do it` `Clear` `Cancel` `OK`
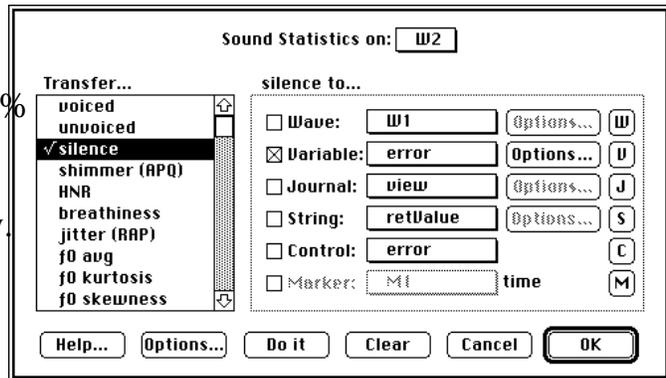
STATISTICS is used to calculate the following statistics on a wave: avg, min, max, time of min, time of max, std dev, rms and area. Results are easily sent to Journals, Waves, Variables, Strings, Controls and Markers.

Statistics on: `Ain0`

Transfer... | std dev to...
- √ avg
- min
- max
- X at min
- X at max
- area
- √ std dev
- rms
- duration
- sum of points

☐ Wave: `deriv` `Options...` [W]
☐ Variable: `error` `Options...` [V]
☐ Journal: `Notes` `Options...` [J]
⊠ String: `retValue` `Options...` [S]
☐ Control: `◇` [C]
☐ Marker: time [M]

`Help...` `Do it` `Clear` `Cancel` `OK`
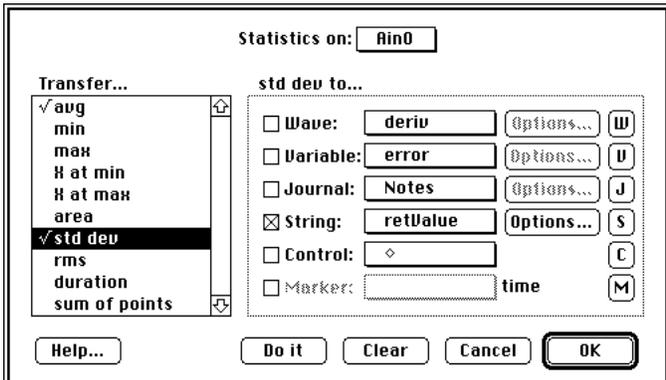
SYNTHESIZE is used to load a wave with a constant value, ramp, sine wave, triangle wave, square wave, gaussian noise (gaussian distribution), or uniform noise (flat histogram).

Synthesize: W1

Length: **200** points

◉ Periodic: 52 points/cycle
10.000 code amplitude
◉ Sine ○ Square ○ Triangle

○ Ramp: 0.000 to 5.000 code
○ Constant: 0 code
○ Guassian: 5.000 code rms noise
○ Uniform: 10.000 code random noise (±)

[Help] [Do It] [Cancel] [OK]

TRANSCENDENTAL is used to calculate a transcendental function (e.g. sin, cos, abs, ln) on one object, and transfers the result into another object. A marker's value is derived from it's position; a control's value is derived from it's setting; a string or journal's value is derived from it's text; and a wave's scalar value is derived from one point.

Scalar Transcendental

| String | ⇐ | Cos | | Marker |
| Y | | | | M1 |

[Options...] [Options...]
0.00000 Volts 0.01000

[Help...] [Do it] [Cancel] [OK]

TURBODRIVERS is used to control MacADIOS Nubus and SCSI data acquisition hardware. For details on these A/D and D/A products, please request the GW Instruments Data Acquisition catalog.

TurboDrivers™ Library

Select function: Enter parameters:

CkControl
CkSetup
Cread
Cwrite
Digitize
Fastio
FqControl
FqSetup
Inout
PdControl

Waves
Points: **1000**   0: Voiced   4: ◇
                   1: time   5: ◇
Bank: 0   2: ◇   6: ◇
Period: 20   3: ◇   7: ◇
Trigger: None

[Device...] [Help...] [Do It] [Cancel] [OK]

WAVEFORM GENERATOR is used to continuously output a waveform in the background, while the computer operates normally in the foreground. Once the interrupt process is installed, it runs smoothly until finished, even during file I/O or switching to another application.

Interrupt-Driven Waveform Generator

Continuously output wave [W1] to board port [AoutO]

Number of points: [200]
Number of passes: [100]
Time between points: [500]   µs

Command: [error]
0 = Install
1 = Deinstall
2 = Deinstall All

(Help) (Device...)   (Do It)   (Cancel)   [OK]
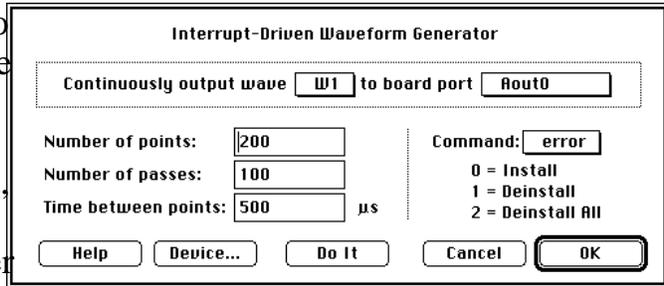
USER INTERFACE is used to monitor the X-Y position of the mouse, the status of the mouse button (up or down) and the status of the keyboard (i.e. the last keypress). Results are sent to journals, waves, variables, strings, markers, and controls.

User Interface

Transfer...
√mouse X
  mouse Y
  mouse status
  key press

mouse X to...
☐ Wave: [W1]   (Options...) (W)
☒ Variable: [V2]   (Options...) (V)
☐ Journal: [Data]   (Options...) (J)
☐ String: [retValue] (Options...) (S)
☐ Control: [error]   (C)
☐ Marker: [M1]   time   (M)

(Do it) (Clear) (Cancel) [OK]
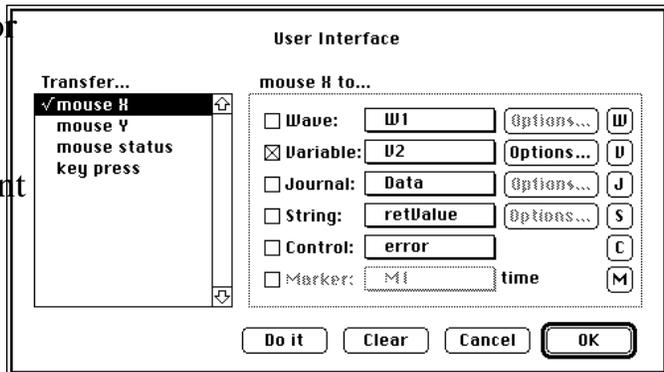
USER PROMPT is used to show a custom alert, with an optional response field and an optional Cancel button. For example, the instruction to the right results in the alert pictured below.

User Prompt

Message
┌─────────────────────────────┐
│ Do you cook with gas?   │
│   │
│   │
└─────────────────────────────┘

☒ Response:
┌─────────────────────────────┐
│ No, I cook electric.   │
└─────────────────────────────┘

Right Button: [Yep]
☒ Left Button: [Nope]

Variable "error" is loaded with 1 if Right button is pressed; 2 otherwise. String "retValue" is loaded with response text.

(Help) (Do It)   (Cancel)   [OK]

Do you cook with gas?

┌─────────────────────────────┐
│ No, I cook electric.   │
└─────────────────────────────┘

(Nope)   [Yep]

Instruments   **4 - 19**
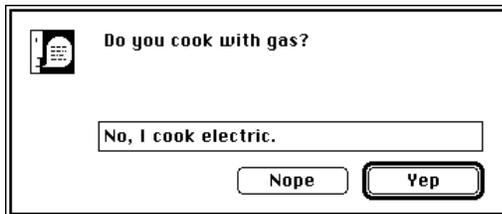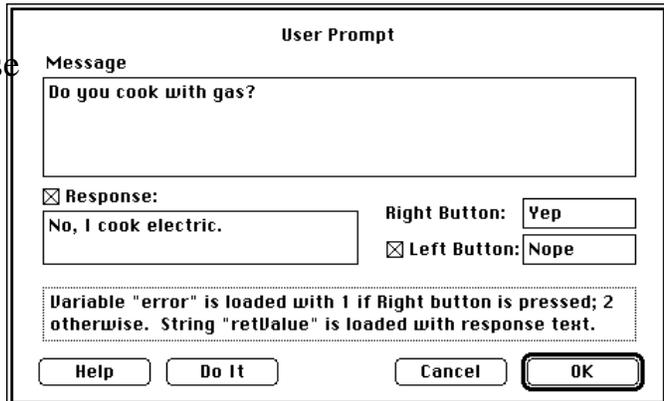
FUNCTIONS

The Calculate Wave instruction allows one to define waves in terms of other waves (e.g. "W1 = deriv(W2)"). From within the Calculate Wave dialog, one can use any of 12 operators and any of over 65 functions (e.g. FFT, cos, signal average), as summarized in the table below.

Waves of type 16-bit integer and 32-bit floating point can be intermixed within a function (e.g. "int16Wave = Abs(flt32Wave)"). Integer waves are always analyzed with respect to their engineering units. For example; if wave *Result* and *Source* are both 16-bit integer waves with ±10 Volts mapped to ±32,768 (e.g. 5V corresponds to 16384 internally, etc.), *Source* is loaded with 1.0V (3,277 internally) and one does "*Result* = cos(*Source*)", then *Result* would be loaded with 0.54 (1770 internally). The neat thing about integer waves is that the user, the displays and the tasks only see Engineering Units (e.g. Volts); subsequently, one does not need to keep track of the internal values. Digitized data (e.g. from an analog-to-digital or digital-to-analog converter) is always contained in 16-bit integer waves, since this is the native format for I/O hardware. To see internal data, choose Edit Values under Wave and then select Internal.

Function	Description	Example
+	add	W1 = W2 + W3
-	subtract	W1 = W2 - W3
*	multiply	W1 = W2 * W3
/	divide	W1 = W2 / W3
AND	bitwise AND	W1 = W2 & W3
OR	bitwise OR	W1 = W2 | W3
<	1 if less then; 0 otherwise	W1 = W2 < W3
>	1 if greater then; 0 otherwise	W1 = W2 > W3
≤	1 if < or equal; 0 otherwise	W1 = W2 ≤ W3
≥	1 if > or equal; 0 otherwise	W1 = W2 ≥ W3
==	1 if equal; 0 otherwise	W1 = W2 == W3
!=	1 if not equal; 0 otherwise	W1 = W2 == W3
Abs	absolute value	W1 = Abs(W2),
Alarm	beep if value is out of bounds	W1 = Alarm(W2, 5, 0)
Append	append one wave to another	W1 = Append(W2, W3)
Arccos	inverse cosine	W1 = Arccos(W2)
Arcsin	inverse sine	W1 = Arcsin(W2)
Arctan	inverse tangent	W1 = Arctan(W2)
AutoCorrelation	autocorrelation	W1 = AutoC(W2)
AvgToDate	average value to date	W1 = AvgToD(W2)
Blackman	generate a Blackman window	W1 = Blackman(256)
Compress	decrease sample rate	W1 = Compress(W2, 2)
Convolve	convolution	W1 = Convolve(W2, W3)
CopyTiming	copy sample period & start time	W1 = CopyTiming(W2)
Cos	cosine	W1 = Cos(W2)
CrossCorrelation	cross correlation	W1 = CrossC(W2, W3)
CrossPower	cross power	W1 = CrossP(W2, W3)
DeConvolution	de-convolution	W1 = DeConv(W2, W3)
Delete	delete a waveform segment	W1 = Delete(W2, 50, 100)
Deriv	derivative	W1 = Deriv(W2)
DerivFivePt	5 pt. Lagrange derivative	W1 = DerivFivePt(W2)
Exp	exponential	W1 = Exp(W2)
Expand	increase sample rate	W1 = Expand(W2, 2)
FFT	fast Fourier transform, reals	W1 = FFT(W2, 256)
Hamm	generate a Hamming window	W1 = Hamm(256)
Hann	generate a Hanning window	W1 = Hann(256)
Histo	histogram	W1 = Histo(W2, 0, 50,10)
IIR	IIR filter	W1 = IIR(W2, W3, W3)
Imag	convert complex imaginary	W1 = Imag(W2)
IndexSort	sort given indices	W1 = IndexSort(W2, W3)

Function	Description	Example
Insert	insert a segment into a wave	W1 = Insert(W2, W3, 50)
Int	convert to closest integer	W1 = Int(W2)
Integ	integrate	W1 = Integ(W2)
IntegAV	integrate, reset when area = A	W1 = IntegAV(W2, 10)
IntegPT	integrate, reset when time = T	W1 = IntegPT(W2, 0.1)
IntegTL	integrate, reset at list of times	W1 = IntegTL(W2, W3)
IntegTV	integrate, reset when wave > V	W1 = IntegTV(W2, 5)
InvFFT	inverse fast Fourier transform	W1 = InvFFT(W2, 256)
Kaiser	generate a Kaiser window	W1 = Kaiser(W2, 256)
Last	return data from last trace	W1 = Last(W2)
Limit	apply high & low bounds	W1 = Limit(W2, 5, 0)
Ln	natural logarithm	W1 = Ln(W2)
Log10	logarithm with base 10	W1 = Log10(W2)
Mag	convert complex to magnitudes	W1 = Mag(W2)
MakeComplex	return complex given real	W1 = MakeC(W2)
MakeIndex	return indices for a sort	W1 = MakeI(W2)
Maximum	maximum value	W1 = Maximum (W2)
MaxToDate	maximum value to date	W1 = MaxToDate(W2)
Minimum	minimum value	W1 = Minimum (W2)
MinToDate	minimum value to date	W1 = MinToDate(W2)
Mod	modulo	W1 = Mod(W2)
MvFFT	return magnitude FFT	W1 = MvFFT(W2, 256)
OnOff	on/off feedback control loop	W1 = OnOff(W2, 5, 0)
Peak	find peaks	W1 = Peak(W2, 2.5)
PID	PID feedback loop	W1 = PID (W2, W3, 1...
PulseEndTimes	return list of pulse end times	W1 = PulseEndTimes(W..
PulseMaxTimes	return list of pulse max times	W1 = PulseMaxTimes(W..
PulseStartTimes	return list of pulse start times	W1 = PulseStartTimes(W..
Phase	return phase given complex	W1 = Phase(W2)
Real	return real given complex	W1 = Real(W2)
Reverse	reverse the order of elements	W1 = Reverse(W2)
Reciprocal	return reciprocal of source	W1 = Reciprocal(W2)
Shift	shift wave horizontally	W1 = Shift(W2, 50)
SignalAvg	determine waveform average	W1 = SignalAvg(W2)
Silent	determine silent region	W1 = Silent(W2)
Sin	sine	W1 = Sin(W2)
Smooth	smooth waveform by n points	W1 = Smooth(W2, 12)
Sort	sort wave elements	W1 = Sort(W2)
Spectrum	calculate frequency spectrum	W1 = Spectrum(W2, 256)
Sqrt	square root	W1 = Sqrt(W2)

Function	Description	Example
Tan	tangent	W1 = Tan(W2)
TC	find °C given thermocouple V	W1 = TC(W2, 1)
TimeHisto	time histogram	W1 = TimeH(W2, 2, 100)
TimeValues	returns values given times	W1 = TimeV(W2, W3)
UnVoiced	determine unvoiced region	W1 = UnVoiced(W2)
Voiced	determine voiced region	W1 = Voiced(W2, W3)

RECORD EMG, EEG, PRESSURES, AND SOUND SIMULTANEOUSLY

With Nubus and SCSI based data acquisition products from GW Instruments, one can digitize up to 112 channels (e.g. pressure, EMG, EGG, sound, etc.) at sample rates as high as 833K samples/second. SoundScope supports the hardware devices listed in the table below. For details on these products, please contact GW Instruments and request The Macintosh Data Acquisition Catalog.

Device	Description
MacADIOS II	Nubus board with sixteen (expandable to 112) 12-bit analog inputs, two 12-bit analog outputs, 8 digital I/O and a counter/timer.
MacADIOS II/16	Nubus board with eight (expandable to 56) 16-bit analog inputs, two 12-bit analog outputs, 8 digital I/O and a counter timer.
MacADIOS II Jr	Nubus board with sixteen 12-bit analog inputs, two 12-bit analog outputs, 8 digital I/O and a counter timer.
MacADIOS 8ain	SCSI peripheral with eight 12-bit analog inputs.
MacADIOS adio	SCSI peripheral with two 12-bit analog inputs, 1 12-bit analog output, and several digital I/O.
MacADIOS 8dio	SCSI peripheral with eight digital I/O.
MacADIOS fg	SCSI peripheral that outputs a signal with a programmable frequency, amplitude, and waveform shape (triangle, square or sine).

QUICKTIME

SoundScope supports the playing and plotting of QuickTime movies via the QTMovie() XCMD, which is included with SoundScope. SoundScope only supports "Direct" QuickTime features; for the more extensive "XWindow" based features, we recommend a dedicated QuickTime applications such as MediaGrabber or MediaTools.

GLOTTAL VIDEO

We are very excited about several emerging technologies that facilitate the digitization, viewing, editing and storing of QuickTime movies (i.e. computer videos). With the MediaTime Nubus board from RasterOps, one can view live video on the computer screen from a camera or VCR. And at any time, one can record to disk live video and 2 channels of 16-bit audio. Once on disk, you can easily view, edit, compare, and save using one of the many QuickTime applications programs such as Adobe Premiere™. SoundScope/16 fully supports the MediaTime board's record and play features (i.e. the SoundScope/16 software thinks the MediaTime board is a SoundScope/16 16bit Digitizer board). For details on the MediaTime board and QuickTime software, contact:

RasterOps Corp.
2500 Walsh Ave.
Santa Clara, CA 95051
800/729-2656 (tel.)
408/562-4065 (fax)
•

MediaTime™ Nubus Board
• 24-Bit video circuit
• real-time frame grabber
• 2 channels 16-bit record
• 2 channels 16-bit play
$2000 list price

Adobe Systems, Inc.
1585 Charleston Rd.
Mountain View, CA 94043
800/833-6687 (sales)
415/961-4400 (tel.)
415/961-3769 (fax)
•

Premiere™
• QuickTime Processing Software
• $495 list price

Photoshop™
• Image Processing Software
$895 list price

DiVA Corporation
222 Third St., Suite 1320
Cambridge, MA 02142
800/FYI-DIVA (394-3482)
617/491-4147 (tel.)
617/491-2010 (fax)

VideoShop™
• QuickTime Processing Software
• $595 list price

Chapter 5
Instruments

This chapter describes several software instruments that are included with SoundScope. A SoundScope instrument consists of waves, markers, displays, journals, datapipes, and tasks that have been assembled like building blocks to create a software environment that is appropriate for specific work. You can use these instruments as is, customize them to your specific needs, or build new instruments from scratch. *More instruments may have been added to the disk since this chapter was printed. Most object names are consistent across instruments, so the extra ones should be easy to understand.*

USING AN INSTRUMENT

To begin using an instrument, either double-click its icon from the Finder, or launch SoundScope and then use Open Instrument from the File menu. It is usually a good idea to verify the hardware setup, and save the instrument under a different name. For example:

- From the Finder, double-click on the "2 Channel Analyzer" file in the Instruments folder.

- Select Setup from the Sound menu. Under the title Hardware, make sure all four pop-up menus (Rec A, Rec B, Play A, Play B) show the name of your hardware (e.g. 16-Bit Digitizer), and the correct slot or port if applicable.

- Press the Rec A button and speak into the microphone for two seconds. Then press Play A. If you could clearly hear the sound that you recorded, simply click the OK button to exit the dialog box. If not, verify that your hardware is correctly installed. (See *Chapter 1, Installation*.)

- Select Save As from the File menu. Move up one level to the "SoundScope Application" folder, as shown in the illustration.

Change the filename to "my 2 Channel" and click the Save button.
You may want to repeat a similar procedure each time you open a new instrument for the first time.
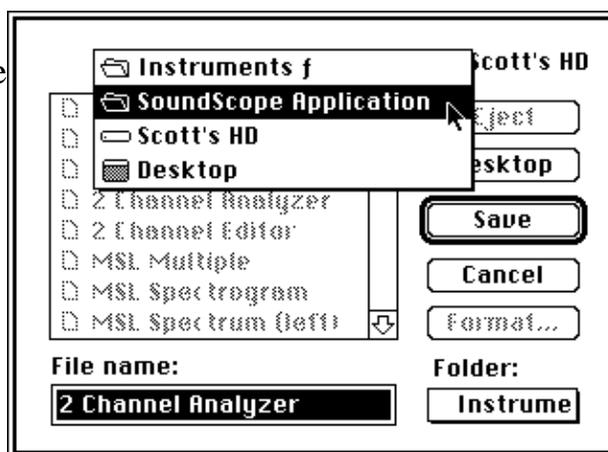
OBJECT NAMING CONVENTIONS

Nearly all SoundScope instruments from GW Instruments follow a single naming convention. This consistency makes it easy to use the instrument most appropriate to a specific task, without having to relearn the basics for each. It also allows tasks to be created on one instrument and used in any instrument that contains similar objects. These names strike a balance between simplicity and flexibility. You may alter any object name, or use any names when creating your own instrument. However, doing so will make it more difficult to exchange tasks and instruments with colleagues.

WAVES

Soundwaves are named with letters of the alphabet, to match the names for SoundScope's two record/play channels. Waves which store the result of a calculation specified in a display are named with the display type, abbreviated to 4 characters (Calc, Snap). If the instrument includes more than one, each subsequent wave is numbered (e.g. Calc2, Snap2). Formants are a special case of a result wave. For the first (or only) soundwave, the Formants are labelled F1, F2 ... F5. For additional soundwaves, the formants are prefixed by the associated Calc wave (e.g. Calc2_F1).

SEGMENTS

Segment names are simply the wave name, plus "S" followed by the segment number (e.g. A_S1).

DISPLAYS

Displays are named for their type, followed by "D" to distinguish the display from the result wave that certain display types produce. The Wave Plot display can store a family of types, including time (Time_D), frequency (Freq_D), and other data (Plot_D).

MARKERS

Marker names reflect the wave with which they are associated. For soundwaves, the marker number is simply appended to the wave name (e.g. A1). For other waves, the letter "M" and the marker number are appended to the wave name (e.g. Snap_M1).

JOURNALS

Journal names are simple, as there is less need to convey specific information about the type or contents. Journals are named "Notes", followed by the number if there is more than one. Journals that contain instrument documentation are named "ReadMe". The contents of the ReadMe file is copied to the clipboard, along with other things, when the user presses *Option Shift* 'A'.

SOUNDSCOPE INSTRUMENTS

Each instrument is documented in the following pages.

Chapter 6
Front Panel

This chapter provides a detailed description of all Front Panel features that are relevant to instrument users, who interact with SoundScope using pre-defined software instruments. You may use the standard instruments provided by GW Instruments, or custom instruments created by a colleague. The SoundScope & SuperScope II Reference Manual describes customization features for instrument designers.

There are three different kinds of displays. A **wave plot** display shows a plot of a sound waveform. It allows you to scroll through the wave and change the scale. An **analysis** display shows the result of a sound analysis calculation, such as a spectrogram, Fo plot or energy envelope. Finally, a **snapshot** display shows information on a small portion of a wave (e.g. 20 msec) centered about a marker. A commonly used option in the snapshot display is FFT (power spectrum). Both analysis and snapshot displays have controls on the right-hand side for setting various options.
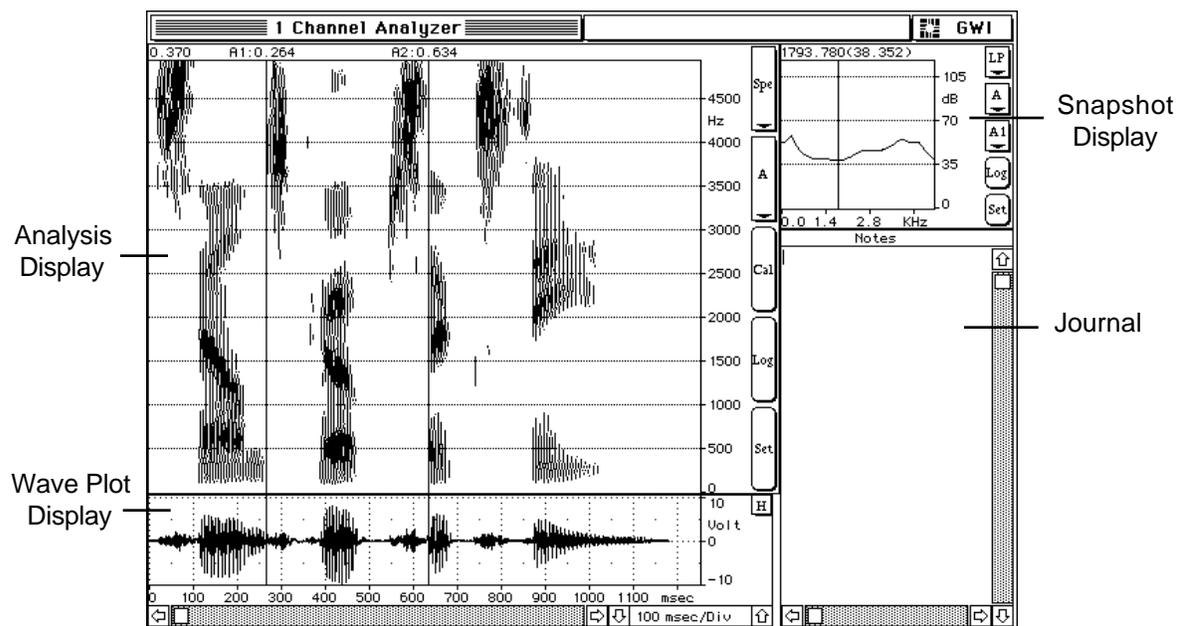
Three Display Types

WAVE PLOT DISPLAYS

A wave plot display shows a plot of a sound waveform. An instrument designer can select specific features and controls, so not all wave plot displays look exactly like the one shown above. If the duration of the wave exceeds the time range of the display, you can scroll the display back and forth to see different parts of the wave. As with any Macintosh application, you can drag the square **scroll box** to a new position, click within the gray area on either side of the box, or click on the left and right **scroll arrows**.

To see more of a wave, or expand the view for a small portion, click on the horizontal and vertical **scale arrows**. Much like an oscilloscope, the scale is set in terms of units per division. The horizontal axis of a wave plot display is usually time, typically in units of msec/div (milliseconds per division). Each division is approximately 1 cm wide, and is labelled with the time. The total time shown in the display is simply the time per division multiplied by the number of divisions in the display. On a standard Macintosh II screen, for example, a wave plot display the full width of the screen contains about 18 divisions. At 100 msec per division, the total time displayed is 1800 msec (1.8 sec), as shown in the screen shot.
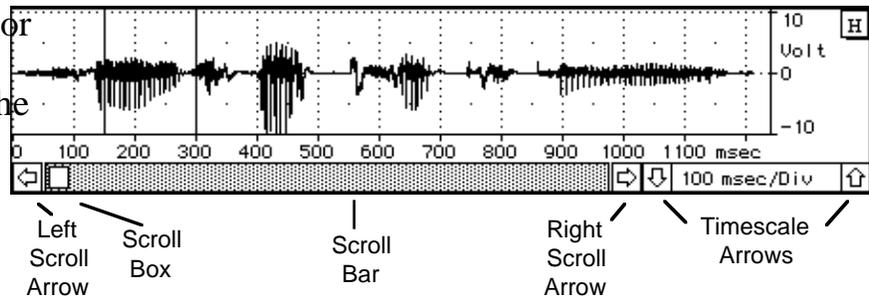
The vertical scale is wave amplitude, measured in volts/div. Digitized waves span from -10 volts to +10 volts. (Note that many instruments omit the vertical scale arrows.)

ANALYSIS DISPLAYS

An analysis display shows the result of a sound analysis calculation. There are five controls on the right-hand side of the display, the analysis option pop-up menu, source wave pop-up menu, Calc button, Log button, and Setup button. The display takes the source wave (or segment of a wave) as input, computes the specified analysis option, produces a result wave and displays that wave. The Calc button recalculates the specified analysis. Pressing the *Command* and period keys simultaneously (.) halts most calculations. The Log button performs actions such as logging data to a journal, or executing a task. The Setup button lets you alter the parameters of each analysis option, and select the events invoked by the Log button.

Each analysis option and its associated parameters is described in the following pages. Although the analysis for a whole wave or segment is displayed, most calculations are performed one frame at a time, on a sequence of frames spanning the source wave. (Recall that a frame is a small interval of a wave, with a frame length typically on the order of 10 to 50 ms.) Consecutive frames, separated by an amount of time referred to as the frame advance, are used to build the result wave from left to right. Since the frame length typically exceeds the frame advance, successive frames will overlap. Reducing the frame advance increases the overlap, thus increasing the resolution. For further technical information, see *Appendix A, Sound Analysis Computations.*

✓ Spectrogram	
LPC History	Analysis Options
HS Slice	
Aug Spectrum	
Fo Plot	Source Wave
Jitter	
Shimmer	
HNR	Calc Button
Envelope	
Energy	
Zero Crossing	Log Button
Spline	
LPC Residual	Setup Button

✓ A
A S1
Selected
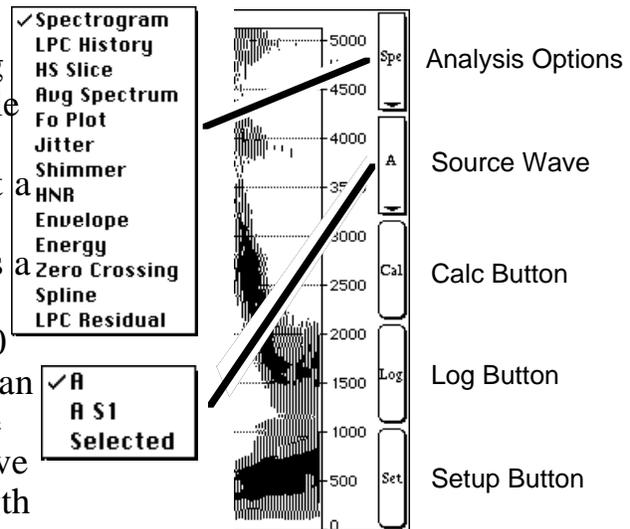
SPECTROGRAM

A Spectrogram shows the frequency components of a soundwave, with the relative intensity (indicated by darkness or color) at each frequency (vertical axis) plotted as a function of time (horizontal axis).

Spectrogram Setup

Analysis

FFT Points: [256]

Filter: [300 Hz (5ms)]

Darkness: [Medium] Max: 47 dB Min: 20 dB

Display range: [0.000] to [5.000] kHz

Frame advance: [3.000] ms (0.1...50)

☒ Use 6 dB pre-emphasis

☐ After calculating, run task: [◇]

Log

☐ Run task: [◇]

[Help...] [Color...] [Options...] [Cancel] [OK]

```
 32
 64
 128
 256
✓512
1024
2048
4096
8192
16384
```

The FFT points pop-up, at the left, sets the number of data points used by the FFT algorithm.

The Filter pop-up, illustrated to the right, selects the filter bandwidth. This selection imposes a lower limit on the possible number of FFT points (i.e. smaller bandwidths require more FFT points), so some of the point values may be grayed out. If you select a filter bandwidth that cannot be computed with the specified number of FFT points, SoundScope automatically chooses the minimum number of points and updates that pop-up. Note that a 59 Hz filter corresponds to narrowband analysis and 184 Hz to wideband analysis.

```
   45 Hz (33ms)
   59 Hz (25ms)
✓ 150 Hz (10ms)
  184 Hz (8ms)
  300 Hz (5ms)
  450 Hz (3.2ms)
  600 Hz (2.5ms)
```

The Darkness pop-up controls the appearance of the spectrogram by changing the range of intensity values that are plotted. Four options are available: three pre-set levels (Dark, Medium and Light) and a Custom setting for specifying the exact Max and Min thresholds. Energy levels in the spectrogram range from 0 through 16000. Energies greater than or equal to the Max value are represented by the Max color (which is black in a gray-scale spectrogram); energies less than or equal to the Min value are represented by the Min color (which is white in a gray-scale spectrogram). Everything between the minimum and maximum energy levels appears as an intermediate color or shade of gray between the Min and Max colors.

The "optimal" setting depends on the nature of the data, type of analysis being performed, and personal preference. A narrow range in energy levels yields a dark, high contrast plot. However, if the range is so narrow that most of the energy values fall either above or below the thresholds, the spectrogram is not very informative.

The `Display range` specifies the lower and upper frequency display limits of the spectrogram (but does not affect the calculation). The `Frame advance` specifies the distance that the frame is shifted right between consecutive FFT calculations.

If the `Use 6 dB pre-emphasis` option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

The only Log option is Run task, which will run the specified task each time you click on the **Log button** that appears with the other display controls.
Note that spectrogram analysis ordinarily produces no result wave, since the result of the analysis is a two-dimensional image rather than a one-dimensional graph.

SPECTROGRAM OPTIONS
The Options button in the Spectrogram Setup dialog box allows you to add formant-tracking data to the plot. Formants are tracked using the LPC method, based on the parameters set in the LPC History dialog. The Calculate check boxes (Spectrogram, 1st formant, 2nd formant, etc.) allow you to specify
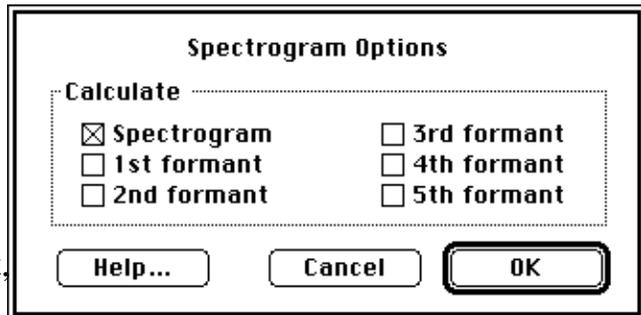
Spectrogram Options
┌Calculate────────────────────────
☒ Spectrogram   ☐ 3rd formant
☐ 1st formant   ☐ 4th formant
☐ 2nd formant   ☐ 5th formant
[Help...] [Cancel] [OK]

which values to plot. Some formant options may be grayed out if there are not enough result waves in the display to hold the formant calculation results. (If necessary, create additional waves via the New command in the Wave menu. Then add the waves to the display list via the Options command in the Display menu, starting with F1 in the second slot, F2 in the third, etc. See *Chapter 2, Tutorial* and the appropriate sections of the SoundScope & SuperScope II Reference Manual for more information.)

SPECTROGRAM COLOR
The Color button in the Spectrogram Setup dialog box allows you to change the colors (or shades of gray) associated with the Max and Min thresholds. The Screen pop-up menu selects between Color and Gray (i.e. gray-scale) spectrograms. This setting takes effect the next time the Calc button is clicked. The range of colors plotted in the spectrogram is displayed as a spectrum between the Picker buttons, ranging from the Min color on the left to the Max color on the right. If Gray is selected in the Screen pop-up, the range consists of all the available levels of gray (typically 4, 16 or 256 shades, depending on the monitor and the setting in the Monitors Control Panel). Selecting Color enables the Picker buttons for altering the Min and Max colors with Apple's standard color picker. The Color option is only available on color monitors, and only works when the monitor is set to operate in color mode, as specified in the Monitors dialog box in the Control Panels under the Apple () menu.
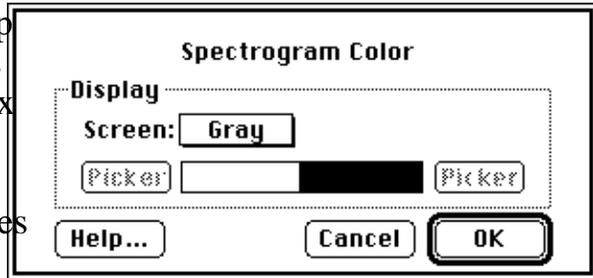
To select a new color, click on the desired shade directly in the color wheel. The upper half of the color sample box indicates your new selection, the bottom half your previous selection. You can make the color brighter or darker by scrolling the brightness scroll bar to the right of the color wheel. You can also directly edit the HSB or RGB values.
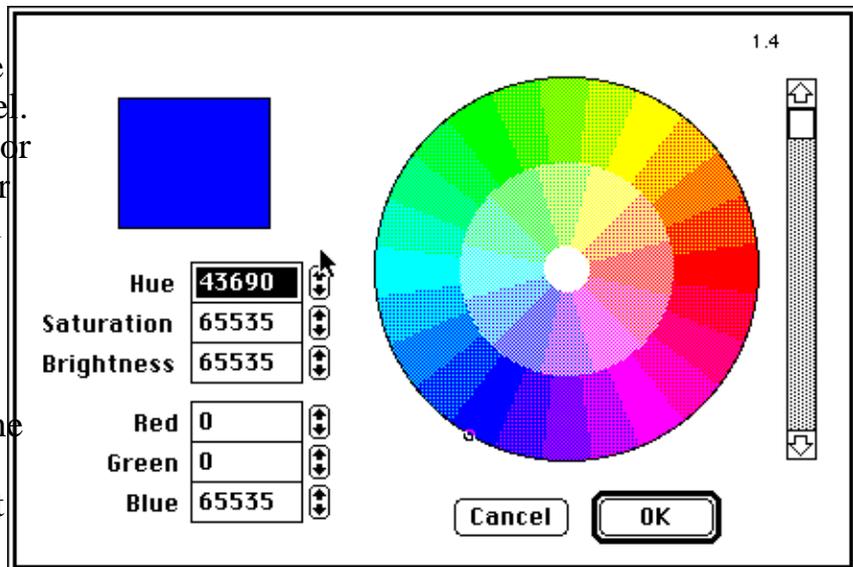
LPC History

LPC (Linear Predictive Coding) History shows the estimated vocal tract transfer function over time. The data is a series of vertical lines that represent formants, showing the peaks and bandwidth of the LPC at each frame. One can think of the LPC History as an alternative view of the same information contained in a spectrogram.

Note that spectrogram analysis ordinarily produces no result wave, since the result of the analysis is a two-dimensional image rather than a one-dimensional graph.

LPC History Setup

Analysis

LPC coefficients: `14` (0...257) ☒ Default

Frame length: `20` ms (5...30)

Frame advance: `5.000` ms (1...30)

FFT Points: `256` (Hz resolution)

Display range: `0.000` to `5` kHz

☒ 6 dB pre-emphasis

☐ After calculating, run task: `◇`

Log

☐ Run task: `◇`

[Help...] [Options...] [Cancel] [OK]

The LPC coefficients textbox lets you adjust the number of coefficients used in the calculation. Increasing the value helps to resolve formant peaks that are of similar frequency, but can also introduce extraneous peaks. Clicking the Default checkbox sets the value that is appropriate for the current sample rate. The Frame length textbox sets the size of the frame on which LPC coefficients are based, and the Frame advance textbox sets the distance the frame moves between consecutive calculations. The FFT points pop-up sets the number of data points used by one step in the LPC algorithm. A higher value will help to resolve peaks, at the expense of computation time. The Display range specifies the lower and upper frequency display limits of the LPC History (but does not affect the calculation).

If the Use 6 dB pre-emphasis option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

The only Log option is Run task, which will run the specified task each time you click on the **Log button** that appears with the other display controls.

LPC OPTIONS

The Options button in the LPC History Setup dialog allows you to add formant-tracking data to the plot. The Calculate check boxes (LPC history, 1st formant, 2nd formant, etc.) allow you to specify which values to plot. Some formant options may be grayed out if there are not enough result waves in the display to hold the formant calculation results. (If necessary, create additional waves via the New command in the Wave menu. Then add the waves to the display list via the Options command in the Display menu, starting with F1 in the second slot, F2 in the third, etc. See *Chapter 2, Tutorial* and the appropriate sections of the SoundScope & SuperScope II Reference Manual for more information.)

LPC History Options

Calculate:
⊠ LPC history   ☐ 3rd formant
☐ 1st formant   ☐ 4th formant
☐ 2nd formant   ☐ 5th formant

Help...   Cancel   OK
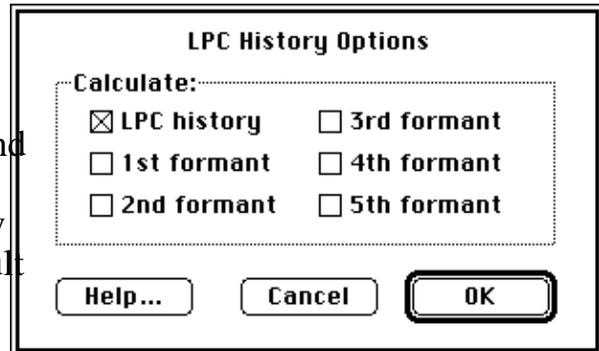
HS SLICE

HS (Horizontal Spectral) Slice takes a specific frequency of the soundwave and displays the intensity level (in dB) as a function of time. The name derives from viewing this computation as a horizontal "slice" across a spectrogram at a single frequency value (actually a specified bandwidth around that frequency). This analysis is the only one that focuses on a single frequency.
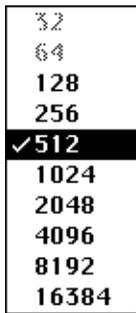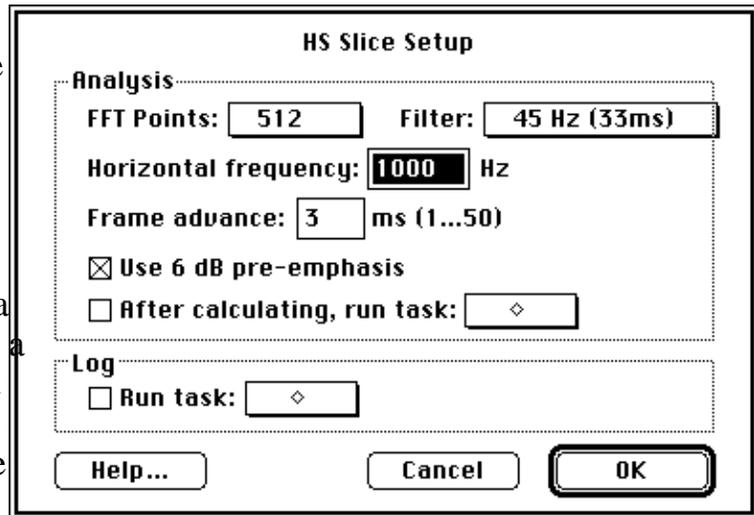
HS Slice Setup

Analysis
FFT Points: [512]   Filter: [45 Hz (33ms)]

Horizontal frequency: [1000] Hz

Frame advance: [3] ms (1...50)

☒ Use 6 dB pre-emphasis
☐ After calculating, run task: [◇]

Log
☐ Run task: [◇]

[Help...]   [Cancel]   [OK]

The `FFT points` pop-up, illustrated to the left, sets the number of data points used by the FFT algorithm.

32
64
128
256
✓512
1024
2048
4096
8192
16384

The `Filter` pop-up, illustrated to the right, selects the filter bandwidth. This selection imposes a lower limit on the possible number of FFT points (i.e. smaller bandwidths require more FFT points), so some of the point values may be grayed out. If you select a filter bandwidth that cannot be computed with the specified number of

45 Hz (33ms)
59 Hz (25ms)
✓150 Hz (10ms)
184 Hz (8ms)
300 Hz (5ms)
450 Hz (3.2ms)
600 Hz (2.5ms)

FFT points, SoundScope automatically chooses the minimum number of points and updates that pop-up. Note that a 59 Hz filter corresponds to narrowband analysis and 184 Hz to wideband analysis.

The `Horizontal frequency` textbox specifies the single frequency of interest (the value at which the "slice" occurs). The Frame advance textbox sets the distance the frame moves between consecutive calculations.

If the `Use 6 dB pre-emphasis` option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

The only Log option is Run task, which will run the specified task each time you click on the **Log button** that appears with the other display controls.

AVERAGE SPECTRUM

The Average Spectrum shows the frequency distribution for the entire wave or segment, with intensity (in dB) on the vertical axis and frequency on the horizontal axis. It is often referred to as a power spectrum. This analysis is the only one in this pop-up menu that does not have time on the horizontal axis.

Average Spectrum

Analysis
- FFT Points: 256   Smoothing: None
- Filter: 300 Hz (5ms)
- Display range: 0 to 4 kHz
- Frame advance: 5 ms (5...30)
- ☒ Use 6 dB pre-emphasis
- ☐ After calculating, run task: ◇

Log
- ☐ Send up to 5 peaks to journal: Notes
- ☐ Transfer spectrum to: Default folder
- ☐ Run task: ◇

[Help...]   [Cancel]   [OK]

```
 32
 64
 128
 256
✓512
 1024
 2048
 4096
 8192
 16384
```

The FFT points pop-up, illustrate to the left, sets the number of data points used by the FFT algorithm.

```
 45 Hz (33ms)
 59 Hz (25ms)
✓150 Hz (10ms)
 184 Hz (8ms)
 300 Hz (5ms)
 450 Hz (3.2ms)
 600 Hz (2.5ms)
```

The Filter pop-up, illustrated to the right, selects the filter bandwidth. This selection imposes a lower limit on the possible number of FFT points (i.e. smaller bandwidths require more FFT points), so some of the point values may be grayed out. If you select a filter bandwidth that cannot be computed with the specified number of FFT points, SoundScope automatically chooses the minimum number of points and updates that pop-up. Note that a 59 Hz filter corresponds to narrowband analysis and 184 Hz to wideband analysis.

```
✓None
 Low
 Medium
 High
```

The Smoothing pop-up, illustrated to the left, allows you to select the degree of smoothing to apply to the FFT. For example, smoothing plots that contain many jagged peaks will make broad trends more visible. The Display range specifies the lower and upper frequency limits to display. (This range affects only the display, not the actual calculation.) The Frame advance textbox sets the distance the frame moves between consecutive calculations.

If the Use 6 dB pre-emphasis option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up

menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer up to *X* peaks to journal option finds the first *X* peaks and sends their frequency and intensity values to the designated journal. The Transfer spectrum to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

FUNDAMENTAL FREQUENCY

The fundamental frequency (Fo) corresponds to the rate of vocal cord vibration. An Fo plot is also known as a pitch plot. It displays estimated glottal frequency (in Hertz) on the vertical axis and time on the horizontal axis. Note that unvoiced portions of a soundwave and other portions that have no discernable fundamental frequency appear as gaps in the Fo plot.

Fo Plot Setup

Analysis

Method: ⊙ Peak-Picking
○ Autocorrelation ☐ Faster

Display range: 50 to 190 Hz

☐ After calculating, run task: ◇

Log

☐ Transfer average f0 to journal: Notes
☐ Send period peaks to journal: Notes
☐ Transfer f0 wave to: Default folder
☐ Run task: ◇

Help... Options... Cancel OK

Method specifies the type of pitch tracker, either Peak-Picking or Autocorrelation. With the latter, the Faster checkbox speeds up the calculation (and reduces the accuracy). Peak-picking is generally more reliable over a wider pitch range than autocorrelation. The Display range specifies the lower and upper frequency limits to show on the screen, but has no affect on the actual calculation (see Options, below).

Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer average Fo to journal option computes the average and sends it to the designated journal. The Send period peaks to journal option sends the raw peak data to the designated journal. The Transfer Fo wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

To change other
significant parameters of
the Fo calculation, click
the Options button (this
dialog can also be accessed
by pressing *Option* 'l'
(lower-case L)).
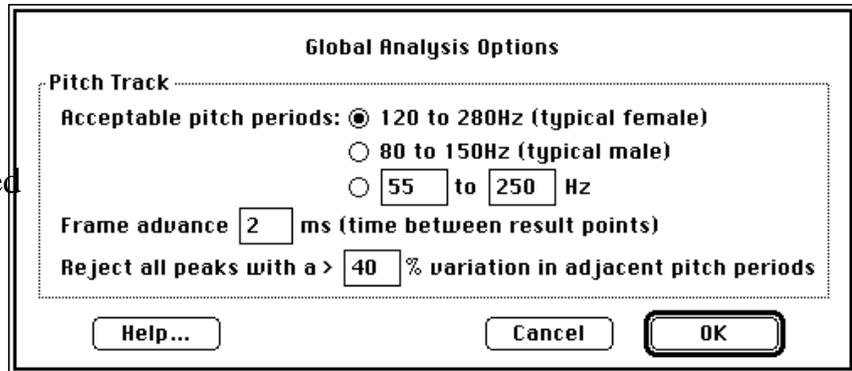
```
┌─────────────────────────────────────────────────────────┐
│                 Global Analysis Options                   │
│ ┌─Pitch Track─────────────────────────────────────────┐  │
│ │ Acceptable pitch periods: ◉ 120 to 280Hz (typical female) │
│ │                           ○ 80 to 150Hz (typical male)    │
│ │                           ○ [55  ] to [250 ] Hz           │
│ │ Frame advance [2  ] ms (time between result points)       │
│ │ Reject all peaks with a > [40 ] % variation in adjacent pitch periods │
│ └──────────────────────────────────────────────────────┘  │
│  ( Help... )              ( Cancel )   ▐  OK  ▌            │
└─────────────────────────────────────────────────────────┘
```

Acceptable pitch
periods sets the bounds
of fundamental frequencies for the pitch tracking algorithm to seek. The Female
and Male radio buttons give suggested pitch ranges for adults. You may also specify
your own exact pitch range. This setting has a dramatic effect on the accuracy of the
Fo calculation, and all calculations that depend on Fo (Jitter, Shimmer, HNR & many
statistics).

The Frame advance textbox sets the distance the frame moves between
consecutive calculations. Reject all peaks with a > X% variation in
adjacent pitch periods specifies the maximum increase or decrease over the
previous pitch period allowed in the search for the next consecutive pitch period.

JITTER

Jitter is a measure of the pitch perturbation of the fundamental frequency, with percent on the vertical axis and time on the horizontal axis.
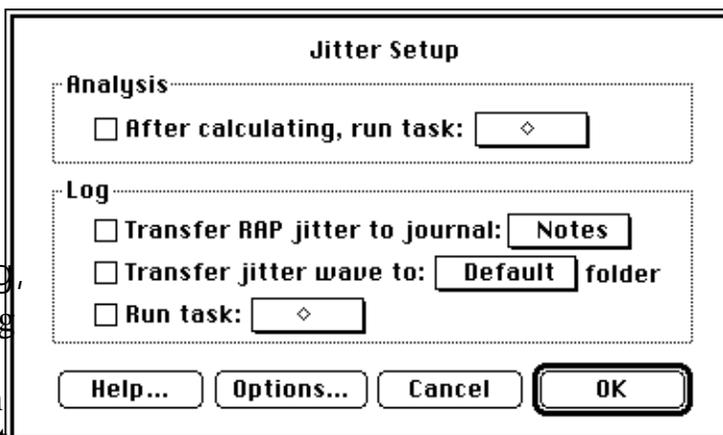
Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation. Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer RAP jitter to journal option computes the average and sends it to the designated journal. The Transfer jitter wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

The significant parameters of jitter are the same as for the Fo calculation, so Options leads to the same dialog (this dialog can also be accessed by pressing *Option* 'l' (lower-case L)). It is generally easier to adjust the parameters to obtain a satisfactory Fo, and not make further changes here.

Acceptable pitch periods sets the bounds of fundamental frequencies for the pitch tracking algorithm to seek. The Female and Male radio buttons give suggested pitch ranges for adults. You may also specify your own exact pitch range.

The Frame advance textbox sets the distance the frame moves between consecutive calculations. Reject all peaks with a > X% variation in adjacent pitch periods specifies the maximum increase or decrease over the previous pitch period allowed in the search for the next consecutive pitch period.

SHIMMER

Shimmer is a measure of the amplitude perturbation of the fundamental frequency, with intensity (in dB) on the vertical axis and time on the horizontal axis.

After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation. Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer APQ shimmer to journal option computes the average and sends it to the designated journal. The Transfer shimmer wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.
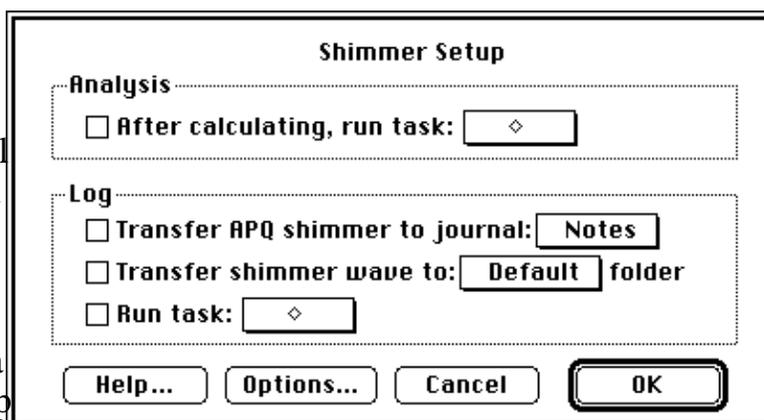
The significant parameters of shimmer are the same as for the Fo calculation, so Options leads to the same dialog (this dialog can also be accessed by pressing *Option* 'l' (lower-case L)). It is generally easier to adjust the parameters to obtain a satisfactory Fo, and not make further changes here.
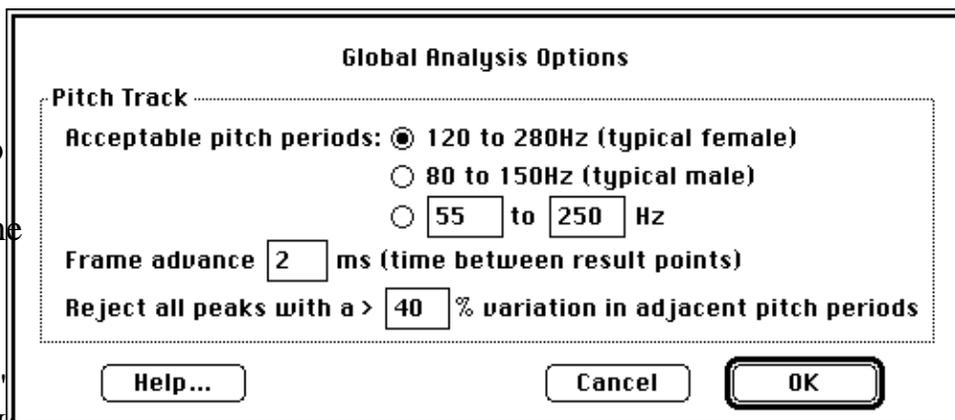
Acceptable pitch periods sets the bounds of fundamental frequencies for the pitch tracking algorithm to seek. The Female and Male radio buttons give suggested pitch ranges for adults. You may also specify your own pitch range.

The Frame advance textbox sets the distance the frame moves between consecutive calculations. Reject all peaks with a > X% variation in adjacent pitch periods specifies the maximum increase or decrease over the previous pitch period allowed in the search for the next consecutive pitch period.

HNR

Harmonic-to-noise ratio (HNR) is a common measure of noise, and is often used to indicate the hoarseness of a voice. The plot displays the HNR (in dB) on the vertical axis and time on the horizontal axis.

The `Average over X pitch periods` textbox specifies the number of pitch periods averaged for each HNR calculation. Increasing this value improves accuracy at the expense of computation speed. Enabling the `After calculating, run task` checkbox and choosing a task from the associated pop-up menu will run that task after each calculation. Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Send average HNR to journal option computes the average and sends it to the designated journal. The Transfer HNR wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.
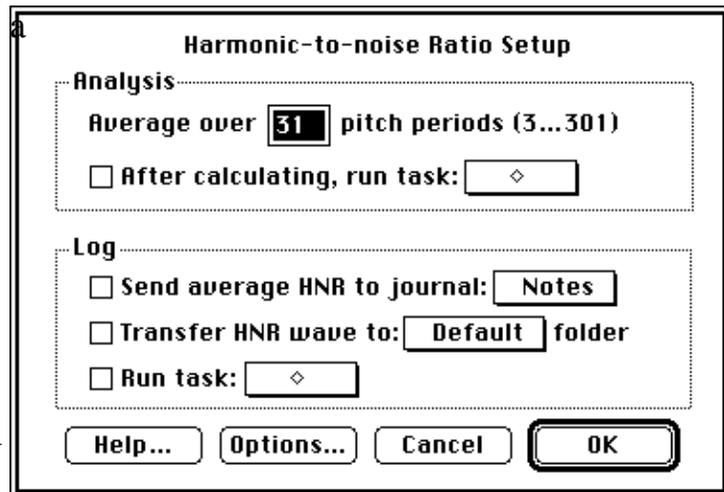
The other significant parameters of HNR are the same as for the Fo calculation, so Options leads to the same dialog (this dialog can also be accessed by pressing *Option* 'l' (lower-case L)).. It is generally easier to adjust the parameters to obtain a satisfactory Fo, and not make further changes here.

`Acceptable pitch periods` sets the bounds of fundamental frequencies for the pitch tracking algorithm to seek. The Female and Male radio buttons give suggested pitch ranges for adults. You may also specify your own exact pitch range. The `Frame advance` textbox sets the distance the frame moves between consecutive calculations. `Reject all peaks with a > X% variation in adjacent pitch periods` specifies the maximum increase or decrease over the previous pitch

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period allowed in the search for the next consecutive pitch period.

ENVELOPE

The envelope of a soundwave represents the outer bounds of its amplitude (in volts), plotted as a function of time. It is the curve created by connecting the peaks.

The Frame advance textbox sets the distance the frame moves between consecutive calculations. Unlike most other sound calculations, increasing the frame advance results in a smoother envelope plot. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Envelope Setup

Analysis
Frame advance: `10` ms (1...50)
☐ After calculating, run task: ◇

Log
☐ Transfer avg amplitude to journal: `Notes`
☐ Transfer min amplitude to journal: `Notes`
☐ Transfer max amplitude to journal: `Notes`
☐ Transfer envelope wave to: `Default` folder
☐ Run task: ◇

`Help...`   `Cancel`   `OK`
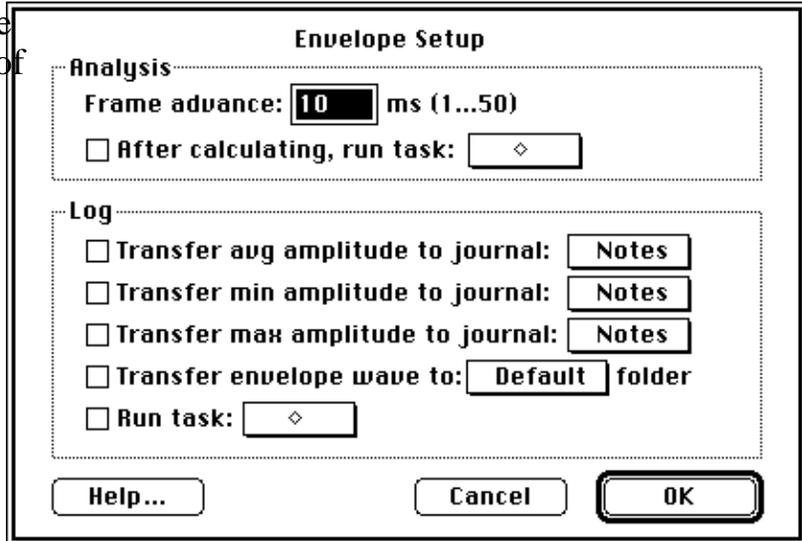
Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer avg, min & max amplitude to journal options compute the parameter and send it to the designated journal. The Transfer envelope wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

ENERGY

The energy function plots the energy envelope, i.e. the outer bounds of sound intensity (in volts squared) as a function of time. It is created by computing intensity and connecting the peaks.

The Frame advance textbox sets the distance the frame moves between consecutive calculations. Unlike most other sound calculations, increasing the frame advance results in a smoother envelope plot. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Energy Setup

Analysis

Frame advance: `10` ms (1...50)

☐ After calculating, run task: `◇`

Log

☐ Transfer avg energy to journal: `Notes`

☐ Transfer min energy to journal: `Notes`

☐ Transfer max energy to journal: `Notes`

☐ Transfer energy wave to: `Default` folder

☐ Run task: `◇`

`Help...` `Cancel` `OK`

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer avg, min & max energy to journal options compute the parameter and send it to the designated journal. The Transfer energy wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

ZERO CROSSING

A Zero Crossing plot displays the rate (in Hertz) at which the soundwave crosses through zero volts, as a function of time. It is often used to identify when the sound is voiced, unvoiced or silent.

The Frame length textbox sets the size of the frame over which zero crossings are counted. A longer frame length produces a smoother plot. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.
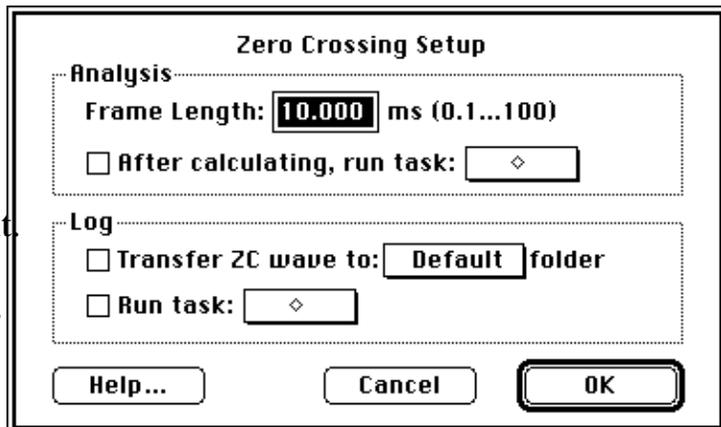
Zero Crossing Setup

Analysis
Frame Length: `10.000` ms (0.1...100)
☐ After calculating, run task: `◇`

Log
☐ Transfer ZC wave to: `Default` folder
☐ Run task: `◇`

[Help...]   [Cancel]   [OK]

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer ZC wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

SPLINE

The Spline plot displays a very smooth version of the soundwave, derived from a cubic polynomial curve fit.

The Knot width specifies the number of data points between adjacent knots. A smaller knot width produces a better fit, but takes longer to compute. The Show radio buttons (Splined signal, 1st derivative, 2nd derivative) selects which wave to compute and display.
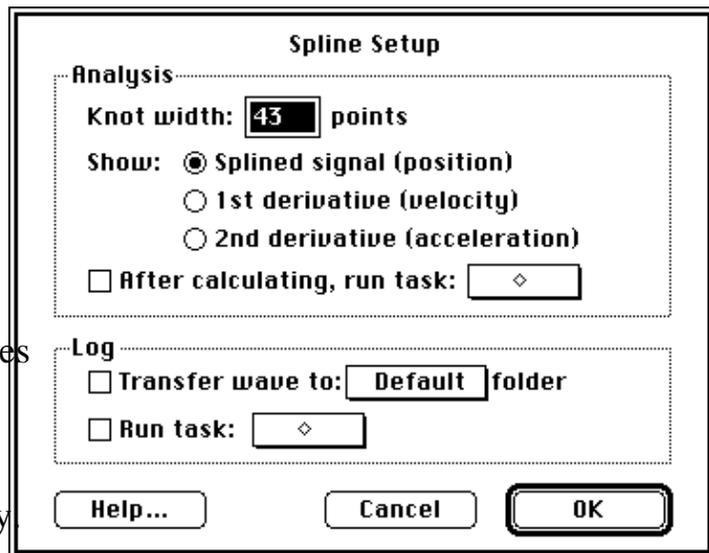
Spline Setup

Analysis

Knot width: [43] points

Show: ● Splined signal (position)
       ○ 1st derivative (velocity)
       ○ 2nd derivative (acceleration)
☐ After calculating, run task: [◇]

Log
☐ Transfer wave to: [Default] folder
☐ Run task: [◇]

[Help...]   [Cancel]   [OK]

Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.
Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

LPC RESIDUAL

The LPC (Linear Predictive Coding) Residual is the difference between the estimated vocal tract transfer function and the actual soundwave.

The LPC coefficients textbox lets you adjust the number of coefficients used in the calculation. Clicking the Default checkbox sets a value that is appropriate for the current sample rate. The Frame length textbox sets the size of the frame on which LPC coefficients are based. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer wave to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

SNAPSHOT DISPLAYS

A snapshot display is used for viewing information about a small time interval (called a frame) of a wave, centered about a designated marker.

There are five controls on the right-hand side of the display, the snapshot option pop-up menu, source wave pop-up menu, source marker pop-up menu, Log button, and Setup button. The display takes the source wave as input, computes the specified snapshot option for the frame centered about the source marker, produces a result wave and displays that wave. Each time one of these items changes, the display automatically recalculates and displays the new result. The Setup button lets you alter the parameters of each snapshot option, as shown in the dialogs below. Finally, the Log button performs the actions that were selected in the Setup dialog box, such as logging data to a journal, or executing a task. Each snapshot option and its associated parameters is described below. For further details on specific algorithms, see *Appendix A, Sound Analysis Computations*.
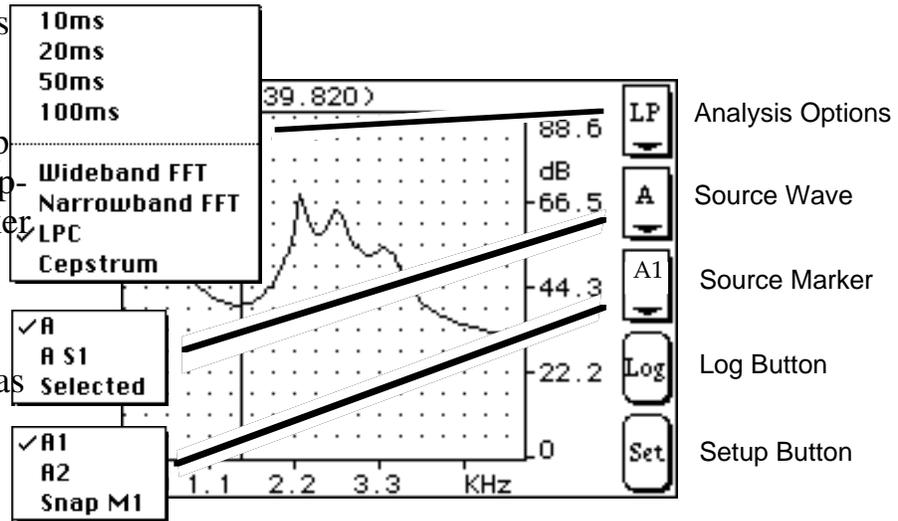
10ms	
20ms	
50ms	
100ms	
Wideband FFT	
Narrowband FFT	
LPC	
Cepstrum	

Analysis Options

Source Wave

Source Marker

Log Button

Setup Button

TIME INTERVALS

As show above, the snapshot option
menu includes selections for 10 ms,
20 ms, 50 ms, and 100 ms.
These commands zoom in on the frame
of data around the marker. Because

This form of analysis does not support the log button.

OK

they do not involve any computation, the Setup and Log buttons do not apply. If you
press those buttons, you will get the dialogs shown above.
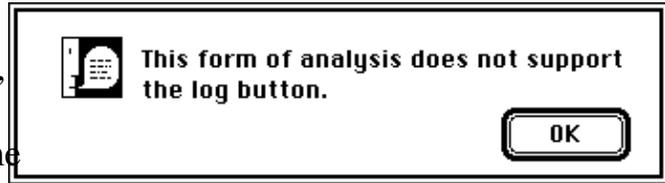
WIDEBAND AND NARROWBAND FFT

An FFT (Fast Fourier Transform) shows the frequency energy of a waveform (vertical axis) as a function of frequency (horizontal axis). The result is often referred to as a power spectrum.

The only difference between Wideband FFT and Narrowband FFT is the default Filter and number of FFT points. The Wideband default Filter is 184Hz (5ms); whereas the Narrowband default Filter is 59Hz (25ms).
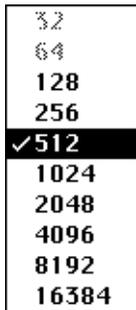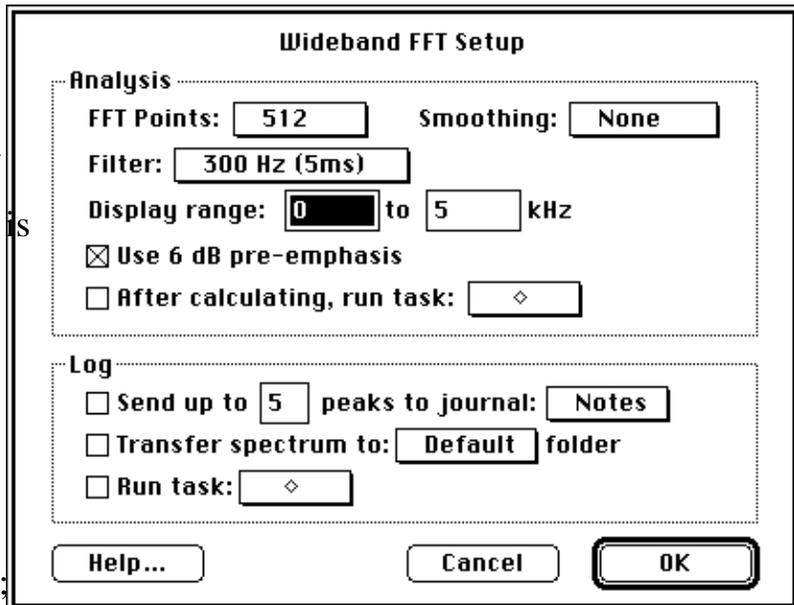
Wideband FFT Setup

Analysis

FFT Points: [512]   Smoothing: [None]

Filter: [300 Hz (5ms)]

Display range: [0] to [5] kHz

☒ Use 6 dB pre-emphasis

☐ After calculating, run task: [◇]

Log

☐ Send up to [5] peaks to journal: [Notes]

☐ Transfer spectrum to: [Default] folder

☐ Run task: [◇]

[Help...]   [Cancel]   [OK]

The FFT points pop-up, illustrated to the left, sets the number of data points used by the FFT algorithm.

```
  32
  64
 128
 256
✓512
1024
2048
4096
8192
16384
```
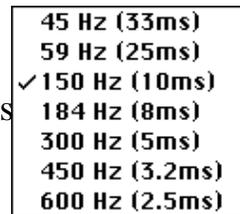
The Filter pop-up, illustrated to the right, selects the filter bandwidth. A wideband filter (184 Hz) shows broad trends, while a narrowband filter (59 Hz) reveals sharp details. This selection imposes a lower limit on the possible number of FFT points (i.e. smaller

```
  45 Hz (33ms)
  59 Hz (25ms)
✓150 Hz (10ms)
 184 Hz (8ms)
 300 Hz (5ms)
 450 Hz (3.2ms)
 600 Hz (2.5ms)
```

bandwidths require more FFT points), so some of the point values may be grayed out. If you select a filter bandwidth that cannot be computed with the specified number of FFT points, SoundScope automatically chooses the minimum number of points and updates that pop-up.

```
✓None
 Low
 Medium
 High
```

The Smoothing pop-up menu allows you to select the degree of smoothing to apply to the FFT. For example, smoothing plots that contain many jagged peaks will make broad trends more visible. The Display range specifies the lower and upper frequency limits to display. (This range affects only the display, not the actual calculation.)

If the Use 6 dB pre-emphasis option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the

---

After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer up to n peaks to journal option finds the first <u>n</u> peaks and sends their frequency and intensity values to the designated journal. The Transfer spectrum to folder option stores the calculation result in the folder designated by the specified datapipe. Finally, the Run task option will run the specified task.

LPC

Linear Predictive Coding (LPC) provides an estimate of the vocal tract transfer function, and the residual signal is the difference between these predicted values and the actual soundwave. One can think of an LPC as a different way to look at the frequency spectrum of a soundwave. As with an FFT, frequency is displayed on the horizontal axis, and intensity (in dB) on the vertical axis.

```
┌─────────────────────────────────────────────┐
│              LPC Setup                        │
│ ┌─Analysis──────────────────────────────────┐ │
│ │  LPC coefficients:  [14] (0...257)  ☒ Default│
│ │                                            │ │
│ │  LPC frame length: [20] ms (5...30)        │ │
│ │  FFT Points:       [  256  ]  (Hz resolution)│
│ │  Display range:    [0] to [5]  kHz         │ │
│ │  ☒ Use 6 dB pre-emphasis                    │ │
│ │  ☐ After calculating, run task: [  ◇  ]    │ │
│ └────────────────────────────────────────────┘ │
│ ┌─Log────────────────────────────────────────┐ │
│ │  ☐ Transfer [5]  peaks to journal: [ Notes ]│
│ │  ☐ Run task: [  ◇  ]                        │ │
│ └────────────────────────────────────────────┘ │
│   ( Help... )      ( Cancel )   (( OK ))        │
└─────────────────────────────────────────────┘
```

The Show LPC and Show Residual Signal radio buttons select which analysis to display. In either case, calculations are performed using the number of coefficients specified in the LPC coefficients box. Use the Default checkbox to let SoundScope select a value appropriate for the current sample rate. The value specified in the LPC frame length box sets the length of the frame (in milliseconds) on which the calculations of the LPC coefficients are based. The FFT points pop-up sets the number of data points used by one step in the LPC algorithm. A higher value will help to resolve peaks, at the expense of computation time. The Display range specifies the lower and upper frequency limits to display. (This range affects only the LPC plot display, not the actual calculation.)

If the Use 6 dB pre-emphasis option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer up to *X* peaks to journal option finds the first *X* peaks and sends their frequency and intensity values (as well as the LPC coefficients) to the designated journal. Finally, the Run task option will run the specified task.

CEPSTRUM

Cepstrum is a computation derived from the excitation-modulation model of speech. It attempts to separate the peaks due to speech from those due to glottal excitation. Time is displayed on the horizontal axis and intensity (in dB) on the vertical axis. (It is interesting to note that the name cepstrum was chosen to mirror "spectrum", from which the calculation derives. Hence, the time axis is usually referred to as quefrency, to mirror the frequency associated with a spectrum.)
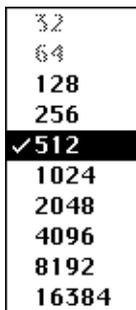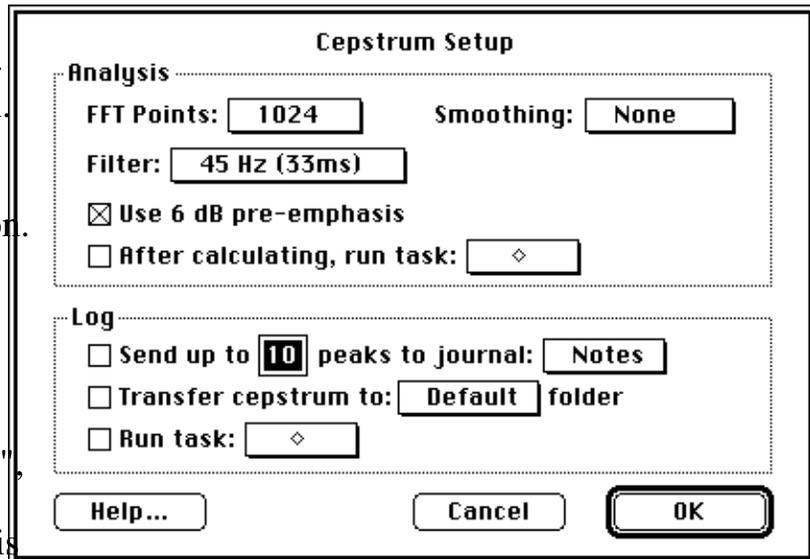
Cepstrum Setup

Analysis

FFT Points: 1024   Smoothing: None

Filter: 45 Hz (33ms)

☒ Use 6 dB pre-emphasis

☐ After calculating, run task: ◇

Log

☐ Send up to **10** peaks to journal: Notes

☐ Transfer cepstrum to: Default folder

☐ Run task: ◇

Help...   Cancel   OK

The FFT points pop-up, illustrated to the left, sets the number of data points used by the FFT algorithm.

The Filter pop-up, illustrated to the right, selects the filter bandwidth. This selection imposes a lower limit on the possible number of FFT points (i.e. smaller bandwidths require more FFT points), so some of the point values may be grayed out. If you select a filter bandwidth that cannot be computed with the specified number of FFT points, SoundScope automatically chooses the minimum number of points and updates that pop-up. Note that a 59 Hz filter corresponds to narrowband analysis and 184 Hz to wideband analysis.
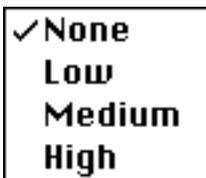
The Smoothing pop-up menu allows you to select the degree of smoothing to apply to the cepstrum. For example, smoothing plots that contain many jagged peaks will make broad trends more visible.

If the Use 6 dB pre-emphasis option is checked, the high frequencies of the source wave are emphasized by 6 dB per octave before processing. Enabling the After calculating, run task checkbox and choosing a task from the associated pop-up menu will run that task after each calculation.

Any Log options that are checked will be executed each time you click on the **Log button** that appears with the other controls. The Transfer up to *X* peaks to journal

option finds the first *X* peaks and sends their time and intensity values to the designated journal. The Transfer cepstrum to folder option stores the calculation result in the folder designated by the specified datapipe Finally, the Run task option will run the specified task.

Chapter 7
The Short Menubar

SoundScope has a set of short menus designed for the instrument user. Advanced features that are only needed by instrument designers are described in the SoundScope & SuperScope II Reference Manual.

Like most Macintosh applications, SoundScope includes an About command under the Apple menu. The menubar then has the standard File menu for loading and saving instruments. The Edit menu features the standard Cut, Copy, and Paste for the current selection, plus additional commands to copy any SoundScope object to the Macintosh Clipboard. The Sound menu covers recording, playing and hardware setup. The Wave menu lets you load, save and manipulate sound waves. The Display menu contains various options for interacting with the display, such as defining mouse behavior. The Journal menu is for loading, saving and printing textual notes and values. And the
Task menu allows you to run a pre-defined sequence of events.

To access a more complete menubar, choose Menubar ▸ Full under Edit. To enable the multi-channel data acquisition features in SoundScope (e.g. record 112 analog input channels via a MacADIOS board), choose Edit Menubar ▸ Full under Edit, select Hardware in the Menu pop-up, press "Show Default Text" (instead of "Hide Menu"), and then press OK. A new menu will appear, illustrated to the right, and it is from here that one sets up a MacADIOS Digitizer. After setting up a digitizer, one can create a task that uses it by choosing New Digitizer under Task. For more details on multi-channel recording and the Full menubar, please see the GW Instruments Data Acquisition Catalog and the SoundScope & SuperScope II Reference Manual.

Hardware	
New...	⌘N
Channel...	▸
Delete	▸
Timebase...	⌘M
Trigger...	
Trigger Modes...	
Digitizer	▸
Digitizer Setup...	
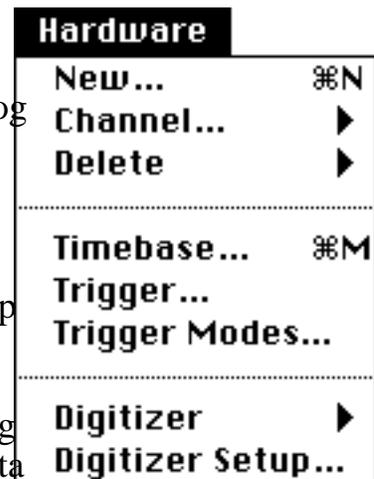
FILE MENU

In many Macintosh applications, the File menu contains all commands that interact with the disk (e.g. Open, Save) and that deal with the entire file or document (e.g. Page Setup, Print). But typical applications only deal with one kind of document or object. In contrast, SoundScope includes many different objects, such as displays, waves, journals and tasks. Each of these objects has its own menu for loading, saving and printing (if applicable). Together, they comprise a software instrument. The SoundScope File menu deals with the entire instrument.

```
┌─────────────────────────────┐
│ File                        │
├─────────────────────────────┤
│ New Instrument...           │
│ Open...                 ⌘O  │
│ Save                    ⌘S  │
│ Save As...                  │
├─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┤
│ Datapipe Folder...       ▶  │
├─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┤
│ Page Setup...               │
│ Print Setup...              │
│ Print...                ⌘P  │
├─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ─ ┤
│ Quit                    ⌘Q  │
└─────────────────────────────┘
```
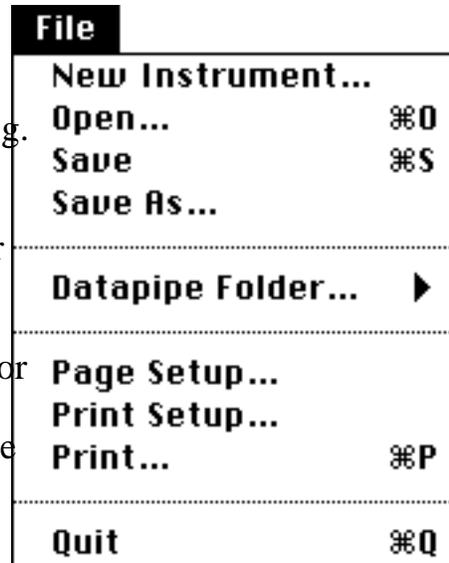
New Instrument clears all displays, markers, waves, journals, tasks and datapipes from memory; and lets you create your own instrument from scratch. SoundScope can only have one instrument active at a time.

Open... (O) lets you select a pre-defined instrument file from disk (using the standard File dialog) to replace the current instrument. SoundScope can only have one instrument active at a time. Unless you press Cancel, all displays, markers, waves, journals, tasks and datapipes will be cleared from memory, and replaced by new ones. All changes that have not been saved will be lost. Be careful, there is no warning prompt.

Save (S) writes the instrument configuration to disk using the current instrument name (shown in the title bar), effectively deleting the previous version. An Instrument Designer has some control over which information is saved with the instrument. The name and attributes of all objects are always saved. To make sure their contents (data) are saved, verify that the appropriate Save contents with instrument file checkbox is selected. (In the Full Menubar, select the Options command from the Wave menu, and click the Points button. Also, select the Options command from the Journal menu. For more information, see the SoundScope & SuperScope II Reference Manual.)

Saving wave and journal contents with the instrument is a convenient way of letting you start up exactly where you left off. Note that this additional information can increase the instrument size greatly.

Save As… prompts you for a filename and location with the standard File dialog, and then saves the instrument to disk.

Datapipe Folder… lets you change the folder associated with a datapipe. A datapipe is a pointer to a folder on a hard disk or floppy disk. It provides a shortcut method for loading and saving files repeatedly to or from the same folder, saving you the trouble of having to find this folder each time you need to access it. Datapipes are also useful for accessing folders from within tasks, where the name of the folder may change with each experiment.

Page Setup… presents the standard dialog for the current printer (specified from the Chooser in the Apple menu). For example, the LaserWriter dialog is shown above. Refer to your printer documentation for detailed explanations of the available settings.

```
LaserWriter Page Setup                              7.0      OK

Paper: ⦿ US Letter   ○ A4 Letter
       ○ US Legal    ○ B5 Letter    ○ [ Tabloid    ▼ ]   Cancel

Reduce or  [100] %         Printer Effects:              Options
Enlarge:                   ⊠ Font Substitution?
                           ⊠ Text Smoothing?
Orientation                ⊠ Graphics Smoothing?
                           ⊠ Faster Bitmap Printing?
```
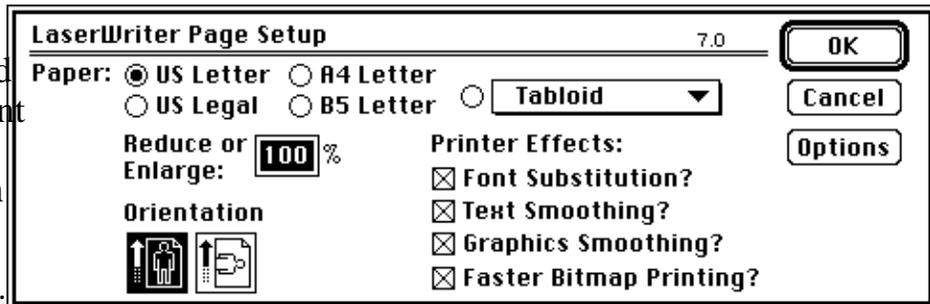
Print Setup… lets you select the objects to print. You may print the entire Front panel or one or more Displays, using horizontal and vertical scales according to the Front panel setting or Fit to page. Printout shows you the Current length, and lets you set a maximum number of pages to print.

```
Print Display Control

┌Print──────────────────┐   ┌Horizontal Scale:─────┐
│ ○ Front panel         │   │ ⦿ Front panel setting │
│ ⦿ Displays: Time D  ⇧ │   │ ○ Fit to page         │
│             Calc D    │   └───────────────────────┘
│             Snap D    │
│                       │   ┌Vertical Scale:───────┐
│                    ⇩  │   │ ⦿ Front panel setting │
└───────────────────────┘   │ ○ Fit to page         │
                            └───────────────────────┘
┌Printout───────────────┐
│ [100]    pages max    │              OK
│ Current length: 2 pages│
└───────────────────────┘    Help      Cancel
```
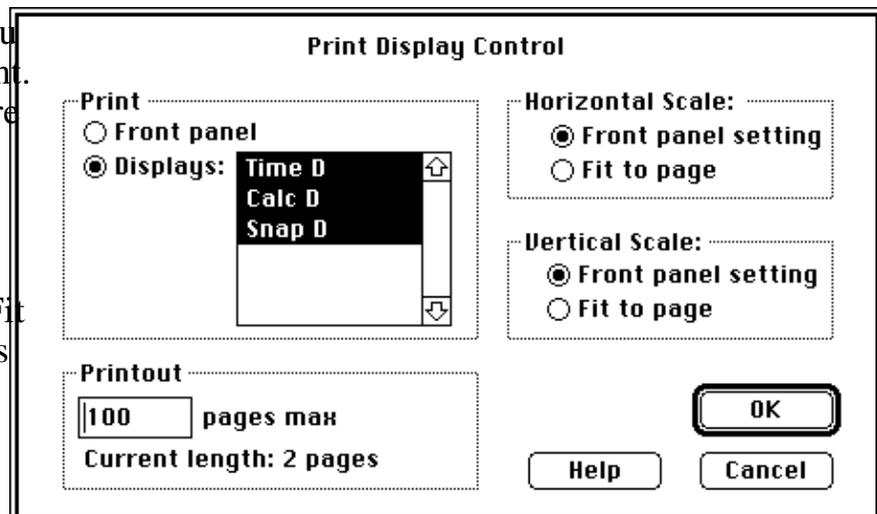
Print... (⌘P)
presents the standard
Macintosh print
dialog, similar to the
one shown to the
right. You may
simply click OK or

LaserWriter "LaserWriter II NT" 7.0	Print
Copies: 1 Pages: ⦿ All ○ From: To:	Cancel
Cover Page: ⦿ No ○ First Page ○ Last Page	
Paper Source: ⦿ Paper Cassette ○ Manual Feed	
Print: ⦿ Black & White ○ Color/Grayscale	
Destination: ⦿ Printer ○ PostScript® File	

hit the Return key to print. Refer to your printer documentation for detailed
explanations of the available settings.
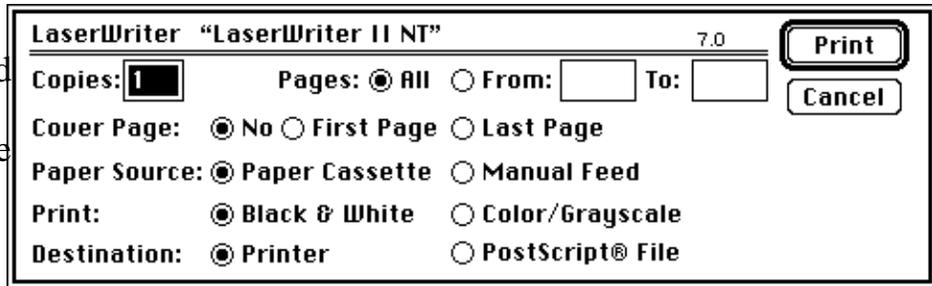
Quit (⌘Q) leaves SoundScope, returning you to the Finder (the desktop). All
changes that have not been saved will be lost. Be careful, there is no warning
prompt.

EDIT MENU

As with most Macintosh applications, the Edit menu contains commands to edit information and place data on the Clipboard for exchange with other software. It also lets you show or hide the Clipboard and other objects. Since waves are represented as a column of numbers separated by carriage returns, one can easily move wave data between SoundScope, spreadsheets, word processors and graphics programs. To select part of a wave with the mouse, make sure that the Mouse mode (under the Display menu) has been set to Edit. Then, click once to select the display that contains the wave of interest. Finally, drag over any part of the wave, highlighting your selection. To select text in a journal, click once on the journal then double-click on a word or drag over any amount of text.

```
Edit
Undo                    ⌘Z
..............................
Cut                     ⌘X
Copy                    ⌘C
Paste                   ⌘U
Clear
Select All              ⌘A
..............................
Show                    ▶
Copy Task               ▶
Copy Display            ▶
Copy Wave Text          ▶
Copy Wave Graph         ▶
..............................
Choose Menubar          ▶
```

Undo (Z) is used by desk accessories to undo the last command.

Cut (X) removes the selected wave data or text, placing it on the Clipboard.

Copy (C) places the selected data on the Clipboard.

Paste (P) is used in a number of ways. In a wave, if the cursor is a cross-hair (i.e. Mouse ▶ Edit is selected under Display), Paste inserts the wave data stored in the Clipboard at the cursor position. (It inserts nothing if the Clipboard contains non-numerical text.) If part of the wave is selected, the Clipboard data overwrites the selection. (If the Clipboard contains non-numerical text, Paste has the same effect as Clear, i.e. it deletes the selection without inserting any data in its place.). In a journal, Paste inserts the text stored in the Clipboard at the cursor position. (It inserts a the wave values if the Clipboard contains wave data.) If some text is already selected, the Clipboard data overwrites the selection.

Clear removes the selected wave data or text, without altering the contents of the Clipboard.

Select All (A) selects all of the data in the active display or journal.

Show will bring a different window to the front, creating it if necessary. As with most Macintosh applications, Show ‣ Clipboard opens a window onto the Clipboard data. Show ‣ Cursor presents a

Cursor ☐		
Wave	**Horizontal**	**Vertical**
mouse	0.079 Sec	14.453 Volt
Ain0	0.079 Sec	-0.967 Volt
W1	0.079 Sec	0.981 Volt

window, shown to the left, that tracks the wave values for the current cursor location. The Show command will also bring any window-based journal to the front. This is useful since the Front Panel often occupies the entire screen.

Copy Task places a textual copy of a task onto the Clipboard. To save or print, simply paste the data into a SoundScope journal or another application. To copy all tasks to the clipboard, press *Option Shift* 'O'. To copy a description of the instrument to the clipboard, press *Option* 'a'. To copy a description of the instrument, including the menubars, press *Option Shift* 'A'.

Copy Display places a visual copy of a display (in PICT format) onto the Clipboard. To create a report or presentation, simply paste the data into another application.

Copy Wave Text places wave values (in text format) onto the Clipboard. To save, print, plot or analyze the data, simply paste into a SoundScope journal or another application.

Copy Wave Graph places a visual copy of a wave (in PICT format) onto the Clipboard. To create a report or presentation, simply paste the data into another application.

Choose Menubar lets you select SoundScope's Full menubar, or any other menubar created by an Instrument Designer. For information on Full menubars, see the SoundScope & SuperScope II Reference Manual. To copy a description of the current menubar to the clipboard, press *Option* 'p'. To copy all menubars, press *Option Shift* 'P'.

SOUND MENU

SoundScope is designed to record, play and analyze two channels of data, referred to as Channel A and Channel B (or simply A and B). You can specify the hardware source and destination for each channel. Depending on your hardware, SoundScope may record (and/or play) these two channels simultaneously (although it cannot both record and play at the same time). However, you can still analyze two channels of sound by recording and playing them one at a time.

Sound	
Play A	⌘6
Play B	⌘7
Play Stereo	⌘8
Play Selected	⌘9
Record A	⌘0
Record B	⌘-
Record Stereo	⌘=
Test...	
Controls...	⌘B
Setup...	

Play A (⌘6) outputs the Channel A wave through the Channel A hardware.

Play B (⌘7) outputs the Channel A wave through the Channel A hardware. (See the Setup command, below.)

Play Stereo (⌘8) outputs both the Channel A sound and the Channel B sound at the same time. At present, only the SoundScope/16 NuBus card supports this feature, so this command will be grayed out for other hardware (the Macintosh internal speaker and MacSpeech Lab).

Play Selected (⌘9) outputs the currently selected sound (i.e. the highlighted portion of a wave) through the Channel A hardware. It will do nothing if there is nothing selected. (Remember that the portion between two markers is a segment not a selection.)

Record A (⌘0) digitizes sound for a specific duration (set with the Controls command, below) from the Channel A hardware and places the data into the Channel A wave. Recording begins as soon as you release the mouse button. There is no way to stop recording in the middle.

Record B (⌘-) digitizes sound for a specific duration (set with the Controls command, below) from the Channel A hardware and places the data into the Channel A wave. Recording begins as soon as you release the mouse button. There is no way to stop recording in the middle.

Record Stereo (⌘=) digitizes both the Channel A sound and the Channel B sound at the same time. Recording begins as soon as you release the mouse button. There is no way to stop recording in the middle. This feature is only available with the SoundScope/16 NuBus card or two MacRecorders, so this command will be grayed

---

out for other hardware (the Macintosh internal microphone and MacSpeech Lab).

Test… presents a series of dialog boxes to guide you through record/play hardware diagnostics. Each dialog corresponds to a different test, which is described in the dialog. The first test plays a tone when the Play button is pressed, the second records when the Record button is pressed, the third plays your recording when the Play button is pressed, and the forth lets the user record and play at will. In each dialog, the Continue button advances to the next test, and the Stop button aborts the test sequence.
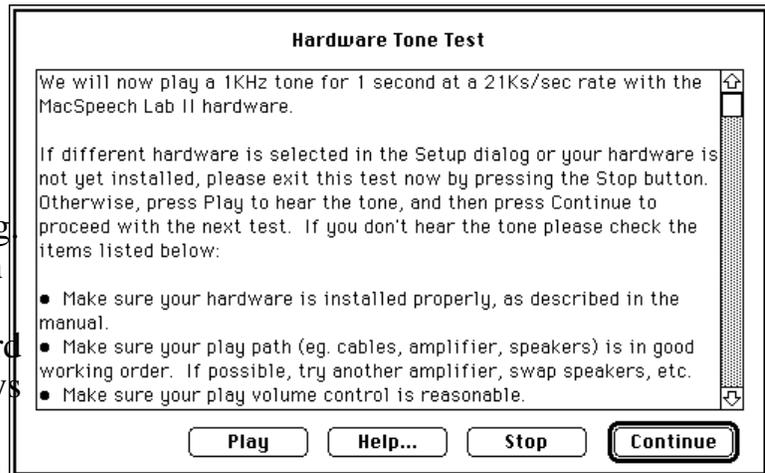
Hardware Tone Test

We will now play a 1KHz tone for 1 second at a 21Ks/sec rate with the MacSpeech Lab II hardware.

If different hardware is selected in the Setup dialog or your hardware is not yet installed, please exit this test now by pressing the Stop button. Otherwise, press Play to hear the tone, and then press Continue to proceed with the next test. If you don't hear the tone please check the items listed below:

● Make sure your hardware is installed properly, as described in the manual.
● Make sure your play path (eg. cables, amplifier, speakers) is in good working order. If possible, try another amplifier, swap speakers, etc.
● Make sure your play volume control is reasonable.

[Play] [Help…] [Stop] [Continue]

Controls… (B) opens a dialog, illustrated to the right, that affects various record & play parameters. Before activating this dialog, SoundScope performs a quick test to make sure the hardware is connected and working properly. If the hardware check is unsuccessful, you will get an error dialog and then the Setup dialog.

Record/Play Control

Record

Duration: 2.000 secs

Input level: 3

☐ Auto normalize

VU Meter

-12-
-10-
- 8 -
- 6 -
- 4 -
- 2 -
- 0 -
L   R

Play

Volume: 3   Repetitions: 1

[Setup…]   [Cancel] [OK]
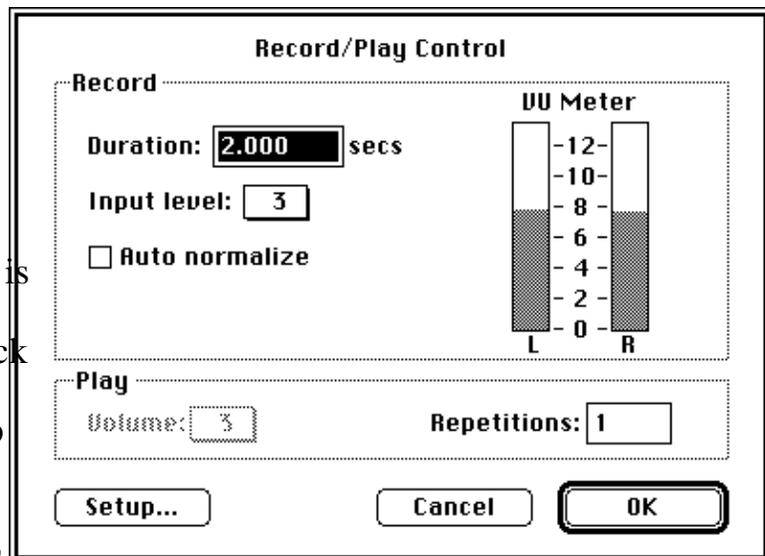
The Duration textbox defines the amount of time to record. SoundScope records continuously during this period, even during moments of silence. The Input level pop-up menu controls the input volume of the 16-bit digitizer, with 0 denoting minimum sensitivity, and 5 denoting maximum sensitivity. This item will be grayed out for other sound hardware. Selecting the Auto normalize checkbox normalizes the entire digitized wave immediately after recording, digitally amplifying as high as possible without clipping. (You can manually normalize the wave at any time with the Normalize command in the Wave menu.)

The VU Meter shows the sound volume as you speak into the microphone, to help you find the optimal distance between the microphone and your mouth. If the level

is too high, you risk signal clipping; recording at too low a level reduces signal resolution. Level 8 is a good balance.

In the Play region, you can set the Volume for the internal Macintosh speaker. This item will be grayed out for other sound hardware. The number of Repetitions specifies how many times the selected channel is repeated when you select Play from the menu. Clicking the Setup button at the bottom of the dialog has the same effect as selecting Setup from the menu.

Setup… brings up a dialog to specify your hardware configuration. SoundScope supports two record channels and two play channels, which can each be set independently. Although the A and B channels are typically set up to play the waves named A and B (if the latter exists), any wave in integer format can be played. Note that most analysis calculations yield a floating point result wave. Certain hardware configurations can record (one AudioMedia board, two MacRecorders) or play (AudioMedia board) both channels simultaneously.

Each hardware supports different Sample Rates. The sample rate corresponds to the number of times per second that the analog sound signal is digitized. Higher sample rates are able to capture higher frequency components but require more memory, disk storage and computation time.

If you want to record from the microphone selected in the *Sound* Control Panel (e.g. internal microphone, MediaTime digitizer, AudioMedia digitizer), select Apple with the Internal Option, and use the Options button in the Sound Control Panel to set the input options.

SoundScope/16
SoundScope/LC
A 16bit AudioMedia I, AudioMedia II, or MediaTime digitizer board is setup, as illustrated to the right. The default sample rate is 22.05 kHz, half of the CD standard. This rate captures most of the "interesting" frequencies in

Hardware Setup

	Wave	Hardware	Option	Sample Rate
Rec A:	A	16-Bit Digitizer	Left+Mic	22.05 kHz
Rec B:	B	16-Bit Digitizer	Right+Mic	22.05 kHz
Play A:	A	16-Bit Digitizer		
Play B:	B	16-Bit Digitizer		

Rec A | Rec B | Rec A & B | Test... | OK
Play A | Play B | Play A & B | Help... | Cancel
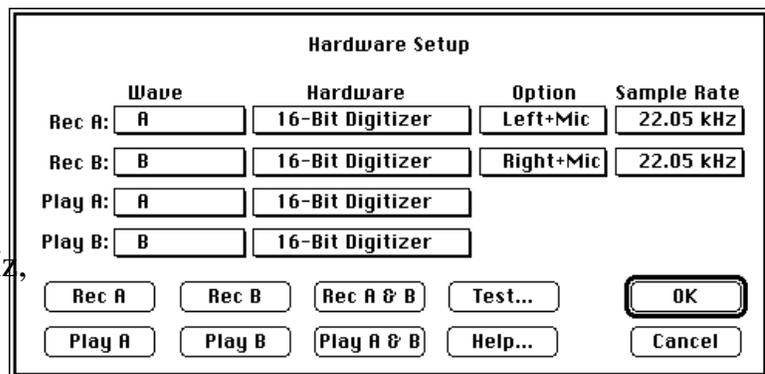
speech, and speeds up all analysis by a factor of two (since the computer processes half the samples that it would with 44.1KHz data). The AudioMedia I & II boards can only directly record and play at the 44.1 kHz rate. Data at lower sample rates is filtered after recording and before playing, resulting in a noticeable delay. The MediaTime board (available from RasterOps Corporation for ~$2000) includes a 24bit video circuit, a real time frame grabber, two 16bit record channels, and two

16bit play channels -- it is truly amazing.

SoundScope/12 &
MacSpeech Lab I
A MacSpeech Lab I or
SoundScope/12 digitizer is setup
as illustrated to the right. The
default sample rate is 10.4 kHz.

```
                     Hardware Setup

              Waue        Hardware      Option    Sample Rate
     Rec A: [  A  ]  [ MacSpeech Lab I ]  [  ◇  ]  [ 10.4 kHz ]
     Rec B: [  B  ]  [ MacSpeech Lab I ]  [  ◇  ]  [ 10.4 kHz ]
     Play A:[  A  ]  [ MacSpeech Lab I ]
     Play B:[  B  ]  [ MacSpeech Lab I ]

     ( Rec A )   ( Rec B )   (Rec A & B)  ( Test... )   (( OK ))
     ( Play A )  ( Play B )  (Play A & B) ( Help... )   ( Cancel )
```

SoundScope/8 &
Sound Control Panel
The Apple Record setting refers
either to the MacRecorder, or
the current device specified in
the *Sound* Control Panel. The
Option pop-up specifies whether
to use the modem port, printer
port, or internal microphone (on
machines such as the Macintosh

```
                     Hardware Setup

              Waue        Hardware        Option    Sample Rate
     Rec A: [  A  ]  [    Apple     ]  [ Modem ]  [ 22.2 kHz ]
     Rec B: [  B  ]  [    Apple     ]  [ Modem ]  [ 22.2 kHz ]
     Play A:[  A  ]  [ Internal Speaker ]
     Play B:[  B  ]  [ Internal Speaker ]

     ( Rec A )   ( Rec B )   (Rec A & B)  ( Test... )   (( OK ))
     ( Play A )  ( Play B )  (Play A & B) ( Help... )   ( Cancel )
```

LC and IIsi). The internal microphone is that which is selected in the *Sound* Control
Panel, and can therefore be used to access any standard sound input device (e.g.
VoiceLink digitizers, RasterOps MediaTime board, etc).   Note that you can record
in stereo with two 8-bit digitizers, one connected to the modem port, the other to the
printer port.

MacSpeech Lab II
For the MacSpeech Lab II
hardware, a Slot pop-up will
appear above the OK button.
SoundScope can usually
determine which slot contains
the board, although it cannot
distinguish between an MSL
board and a standard GWI
MacADIOS data acquisition

```
                     Hardware Setup

              Waue        Hardware        Option    Sample Rate
     Rec A: [  A  ]  [ MacSpeech Lab II ]  [  ◇  ]  [ 10.4 kHz ]
     Rec B: [  B  ]  [ MacSpeech Lab II ]  [  ◇  ]  [ 10.4 kHz ]
     Play A:[  A  ]  [ MacSpeech Lab II ]
     Play B:[  B  ]  [ MacSpeech Lab II ]          Slot: [ 4 ]

     ( Rec A )   ( Rec B )   (Rec A & B)  ( Test... )   (( OK ))
     ( Play A )  ( Play B )  (Play A & B) ( Help... )   ( Cancel )
```

board. Note that the Macintosh IIcx and IIci number their slots beginning with 4.
See *Chapter 1, Installation*, for details.

WAVE MENU

The Wave menu groups together various actions that you can perform on waves. Most items are hierarchical menus with which you can choose an entire wave (by name), a named segment (defined as the portion between two markers) or the currently selected portion (if something has been selected). Note that both the recorded sounds and the results of most calculations are stored as waves, and certain functions only apply to the waves associated with the two channels of sound data.

Wave	
Save As...	▶
Load Data...	▶
Edit Values...	▶
Synthesize...	▶
Zero...	▶
Normalize...	▶
Amplify...	▶
Calculate...	
Filter...	▶
Wave Statistics...	▶
Sound Statistics...	▶
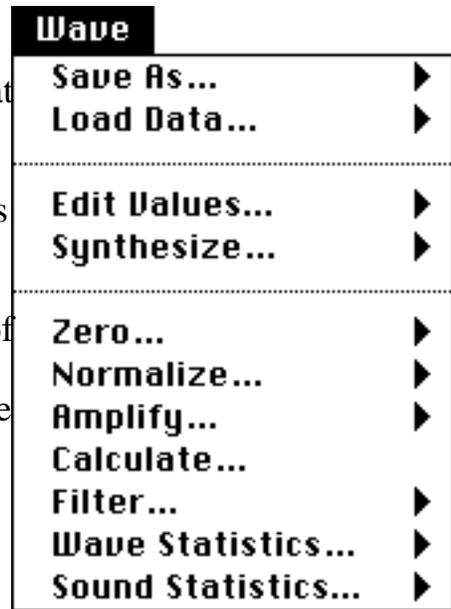
Save As… writes to disk the data in the wave that you select, prompting you for a filename and location (folder), as illustrated to the right. Specify the name of the file in the File name textbox. For soundwaves, the default name is the channel (A or B), so you will usually want to enter a more descriptive name. Like all standard File dialogs, the pop-up menu at the top lets you move "up" in the folder hierarchy. As an alternative, the Folder pop-up menu in the bottom right corner lets you select a datapipe as the destination folder. If no datapipes exist, it will simply show the current folder.
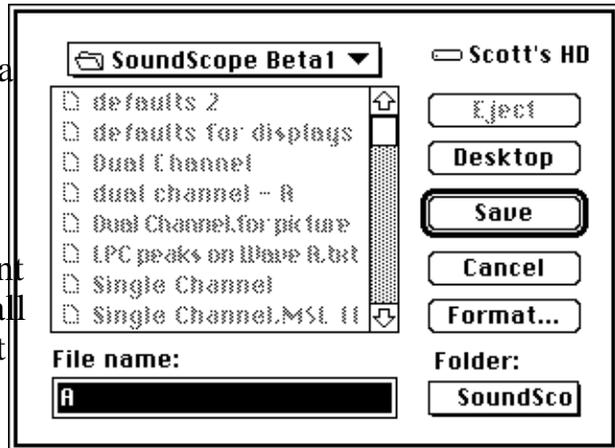
The Format button lets you change the file format and other parameters. The default format is 16-bit integer or 32-bit floating point, depending on the internal data type. (To check or change the type, switch to the Full Menubar, select the Options command from the Wave menu, and click on the Points button.) The data can be saved as Text for use in another application. The Audio IFF (AIFF) format is an emerging industry standard for sound. (See *Appendix B, File Formats* of the SoundScope & SuperScope II Reference Manual for additional information.)
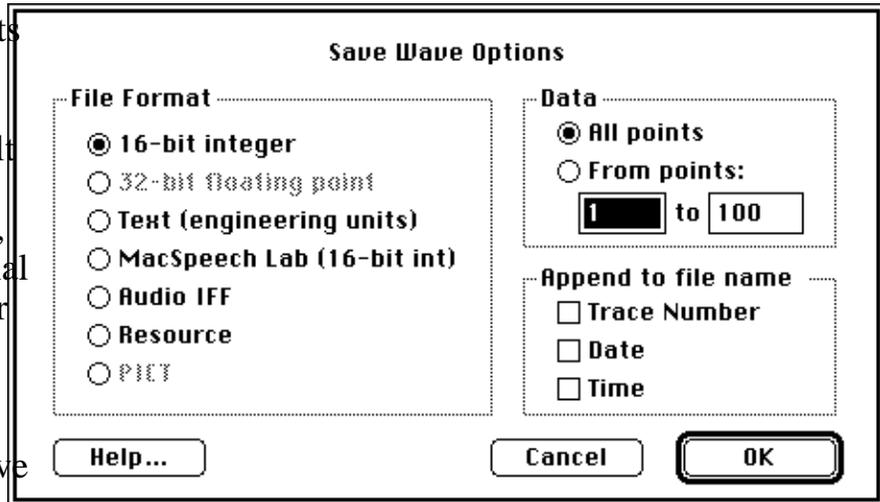
The Data options let you save only a subset of the wave, and the Append choices will annotate the filename with useful information. As noted earlier, wave data can also be saved with the instrument.

`Load Data…` lets you select a file, and then loads its contents into an existing wave, typically A or B (the waves associated with record/play channels). The new data will overwrite information currently in the wave.

`Edit Values…` organizes wave values in a table for simple viewing and editing, as illustrated to the right.

The leftmost column in the table is an index which corresponds to the point in the adjacent column (e.g. in the dialog to the right, the value at .00048sec is .044Volts). This index is shown in units of time (e.g. seconds) or point

Edit Values: A					
sec					
0.000	-0.039	0.029	0.020	0.034	0.049
4.80e-4	0.044	0.034	0.020	9.77e-3	9.77e-3
9.60e-4	9.77e-3	0.020	0.039	0.059	0.049
1.44e-3	0.020	9.77e-3	0.015	0.015	-9.77e-3
1.92e-3	-0.044	-0.229	-0.200	-0.176	-0.161
2.40e-3	-0.146	-0.142	-0.107	-0.088	-0.068
2.88e-3	-0.044	-0.039	-0.029	-0.039	-0.068
3.36e-3	-0.059	-0.020	0.024	0.039	9.77e-3
3.84e-3	-9.77e-3	-0.020	-0.020	-9.77e-3	-9.77e-3
4.32e-3	-0.020	-0.024	-0.020	-0.020	-0.020

Options

Index	Format	Units	Typing	
● Time	● Dec	● Volt	● Overwrite	**OK**
○ Point	○ Hex	○ Internal	○ Insert	**Help**

number (i.e. an incrementing integer, starting with 1), depending on the setting of the Index radio buttons. The five right-most columns hold the wave data, proceeding left to right then top to bottom.

There is no way to cancel or undo changes made in this dialog. To change the value at any point, just click on its cell and type a new number. Use the arrow keys to move the cursor up, down, right and left between cells. Depending on the Typing option, both the Tab and Return keys will either move the cursor to the right (Overwrite mode) or insert a new cell (Insert mode). To select a range of values, simply drag the mouse over any series of cells. Although the File menu is disabled, the keyboard equivalents for Cut (X), Copy (C), and Paste (V) work as expected. The Delete key has the same effect as Cut, and the Enter key will accept changes and exit the dialog.

Several display options are available. The two Format radios determines whether points are displayed in Decimal or Hexadecimal form (e.g. decimal 25 in hexadecimal form is 0x019). Hexadecimal can only be selected when displaying 16bit integer Internal data. The Units option specifies whether values are displayed in engineering units (e.g. -10 to +10Volts values) or Internal units (e.g. -32768 to +32767 values from a 16bit A/D converter). Internal units only apply to 16-bit integer waveforms. Independent of the display options, data is always Cut, Copied, or Pasted in engineering units.

The Typing option controls the behavior of the Tab and Return keys. In Overwrite mode, pressing Tab or Return moves the cursor one cell to the right. In Insert mode, Tab or Return inserts a new cell, increasing the total length of the wave by one point. Finally, the upper-most Wave pop-up menu allows you to select another wave, segment, or selection.

Caution: Only 3 digits after the decimal are shown in the Wave Editor, yet more precision is stored internally. 16bit integer waves often have a least-significant-bit value of 0.000305 (i.e. 20/65536) and 32bit floating point waves offer 7 significant digits of scientific notation accuracy (i.e. $\pm$X.XXXXXX e $\pm$YY). Copy Wave Text in the Wave menu is a more accurate way to copy wave data to the clipboard. Pasting into the Wave Editor is very accurate and does not involve unnecessary truncating. If you select a cell and move to another cell, it will read the text if it differs by more that .001 from the value stored internally and update the internal value to that (truncated) viewed value (careful!).

Synthesize... allows you to load waves with internally-generated (i.e. synthesized) data. Waves can be loaded with a sine wave, square wave, triangle wave, ramp, constant value, guassian noise, or uniform noise. Sine, Triangle and Square waves are defined with the Points-per-cycle and Peak-to-Peak Amplitude edit fields. The Ramp is define with two edit fields that specify the value at the two end points. Gaussian noise has a gaussian distribution (i.e. a histogram of the wave data is a gaussian curve centered about 0) with the specified RMS (Root Mean Square).   RMS is the same as standard deviation is this case. For example, 10Vrms gaussian noise will have 60.6% of it's values between -10 and +10V. Uniform noise is evenly distributed about the specified bound. A histogram of uniform noise shows values evenly distributed between -Bound and +Bound. For example, each point of $\pm$10V uniform noise has an equal probability of appearing anywhere between -10V and +10V. The Length edit field is used to specify the # of points in the synthesized wave, the maximum being that permitted by available memory. The wave is loaded with synthesized data when the user presses OK or moves to another wave via the pop-up at the top of the dialog. If you want to use the pop-up and do not want synthesis to occur, select the Don't

---

Synthesize radio before hitting the pop-up menu.

Zero… sets every point in the designated wave, segment or selection to 0 Volts; effectively replacing sound with absolute silence.

Normalize… amplifies the designated wave, segment, or selection so that its extreme value (maximum or minimum) equals the upper or lower bound (±10 Volts).

Amplify… increases or attenuates the amplitude (voltage) of the designated wave, segment or selection by a specified percentage, clipping any values that exceed the ±10 Volt bounds. For example, an "amplification" factor of 50% will multiply by one-half; a factor of 200% will multiply by two. To invert the wave, use a negative number (e.g. -100%).
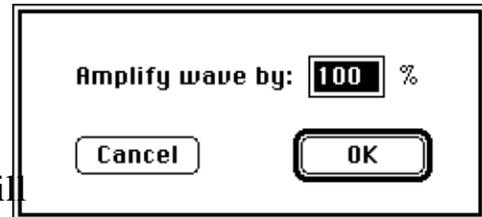
Calculate… is used to perform waveform calculations. Over 80 functions and operators (e.g. fft, cos, +, *) are supported. <u>The calculation takes place <i>only</i> when the Do It button is pressed.</u> For details on this dialog box, please consult the Calculate Wave instruction (which is identical to the Calculate command under the Wave menu) in the *Instructions* chapter of the SoundScope & SuperScope II Reference Manual. For details on each function, please refer to the *Functions & Operators* chapter in the same reference manual.
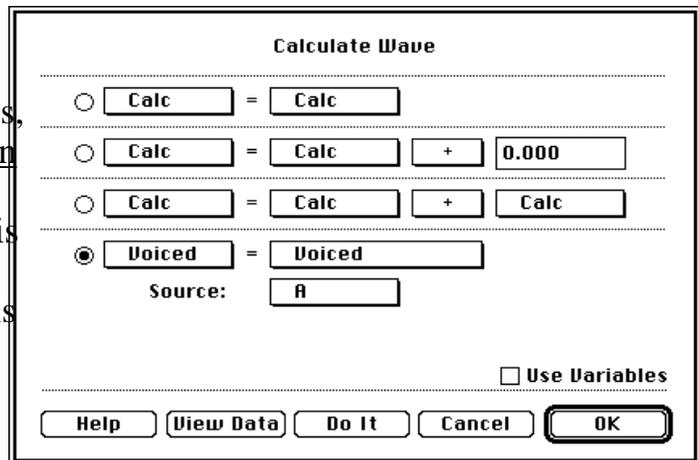
Filter... is used to run a low-pass, high-pass, band-stop, band-pass or user defined FIR filter on a wave, segment or selection. For details on this dialog box, please consult the Filter instruction (which is identical to the Filter command under the Wave menu) in the *Instructions* chapter of the SoundScope & SuperScope II Reference Manual.

```
                    Filter: [ Selected ]

Type:          [ Smoother (ham) ]   [Custom...]
Freq Cutoff:   [  10%  ]       (% of sample rate)
Boundaries:    [ zero pad ]

  Freq Cutoff: 100.0 Hz        Phase: linear
  Stopband: -42 dB             Taps: 20

[ Help... ]              [ Cancel ]  [  OK  ]
```

Wave Statistics... provides general information on the designated wave, segment or selection. Depending on the size of the wave or segment, you may have to wait several seconds for the data to appear. The Wave pop-up menu at the top of the dialog box allows you to view statistics for a different wave, segment or selection.

```
        Wave: [ Selected ] Statistics

Number of points in memory:  504
Data sampled at:  10.417 points / msec
Minimum:  -6.733 Volt (at 145.4 msec)
Maximum:  9.985 Volt (at 145.9 msec)
Mean:  0.040 Volt
Standard deviation:  2.902 Volt
Area under wave:  1.93e-3 Volt - sec
Root mean square:  2.902 Volt

                          [  OK  ]
```

Sound
Statistics...
provides a variety
of calculations for
the designated
wave, segment or
selection. Many of
the calculations
depend on a valid
Fo, based on the

Sound Statistics: [Selected]

Voiced: 100.000 %	Jitter (RAP): 0.326 % (-49.73 dB)
Unvoiced: 0.000 %	Fo Average: 122.911 Hz
Silent: 0.000 %	Fo Kurtosis: -0.860
Shimmer (APQ): NAN(021) %	Fo Skewness: 0.716
HNR: NAN(021) dB	Fo Range: 0.615 ST (4.378 Hz)
Breathiness: -3.977 dB	Fo Std Dev: 1.594 Hz
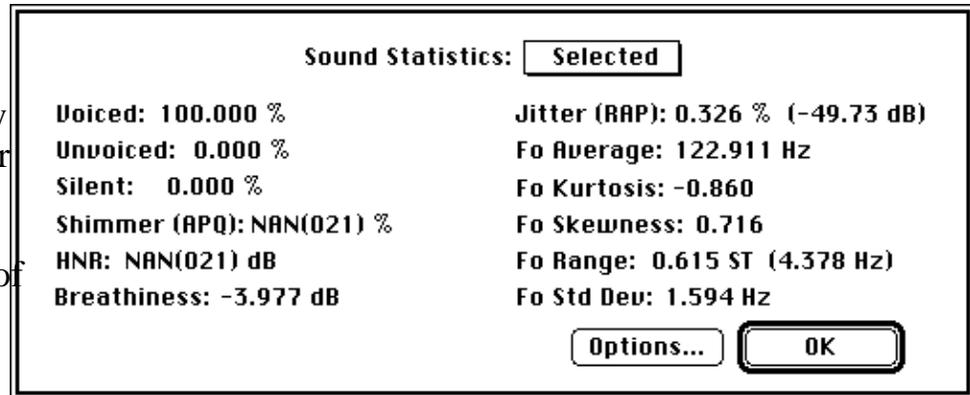
[Options...] [OK]

settings in the Fo options dialog. The Fo Options dialog is opened by pressing the
Options... button, or by pressing *Option Shift* 'l' (lower-case letter 'L') when the
front panel is visible. For long utterances (i.e. several seconds), you may have to
wait several seconds for the data to appear. Depending on your machine, it could
take from one to over ten seconds to analyze a single second of data.

The Wave pop-up menu at the top of the dialog box allows you to view statistics for
a different wave, segment or selection. Note that it is not always possible for a
statistic to be computed, yielding "NAN" (not a number). For technical information
on these statistics, see *Appendix A, Sound Analysis Computations*.

DISPLAY MENU

The Display menu controls your interaction with the SoundScope front panel. It includes commands to control marker location and various choices for mouse behavior.
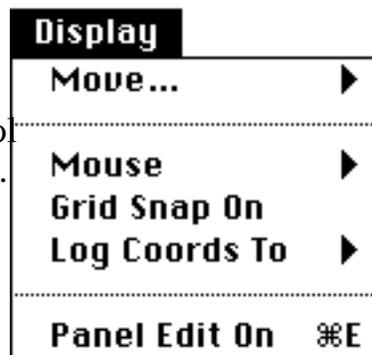
Move… shifts a display, moves a marker or both. For displays, the first Position option shifts the contents by the specified amount. To move left, enter a positive number; to move right, enter a negative number. Note that "moving" the display contents left is equivalent to pushing an imaginary piece of graph paper to the left. An unfortunate (but arguably unavoidable) subtlety of the Macintosh user interface is that "moving" left is equivalent to "scrolling" right. The second option sets the left edge of the display to the specified value, and the third sets the left edge to the marker position. For markers, the first Position option moves the marker by the specified amount. To move left, enter a positive number; to move right, enter a negative number. The second option moves the marker to the specified value, and the third to the position of another marker.
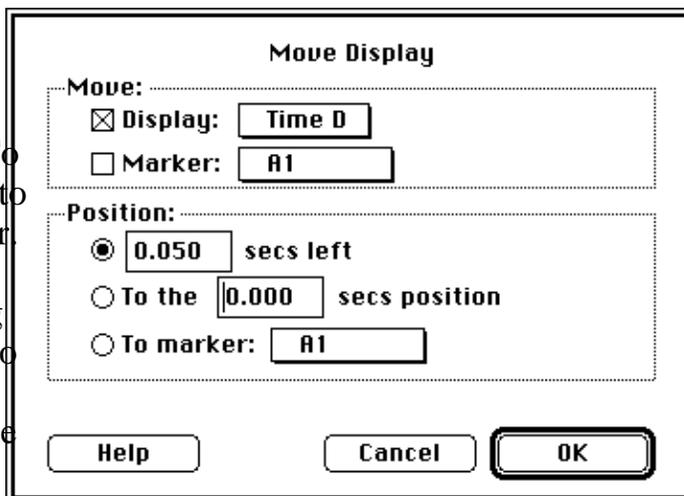
Mouse… is used to select one of five mouse modes: Edit, Draw, Move Marker, Log Coords, or Vertical Adjust. In each mode, the operation of the mouse relates to a single wave in a display that has been selected. If a display includes more than one wave and you need to operate on a wave other than the first (as viewed in the Contents list of the Display Options dialog), wave labels must be visible. To enable wave labels, select Features from the Display menu (which is available in the Full Menubar), and click on the Add Wave Labels checkbox. The first four letters of each wave name are shown in the wave label area, and clicking on a wave label selects that wave for mouse work.
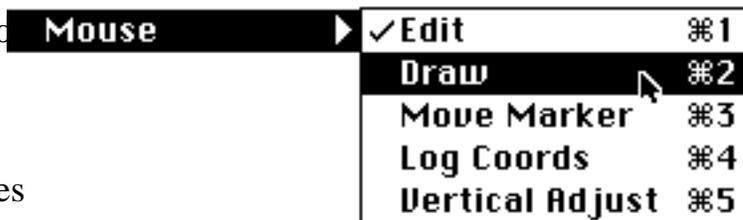
Edit (1) is the most commonly used mode and is automatically selected when you first launch SoundScope. In this mode, you can Cut Copy and Paste sections of a wave much like one would edit text in a word processor (i.e. by dragging the mouse over the area of interest). The selected portion is highlighted and magically becomes a waveform of its own called "Selected". The Selected wave appears in wave submenus, shares data with the host wave, and is viewed as an ordinary waveform by the analysis and presentation tools. Only the user can change the length of the Selected wave (i.e. with the mouse).
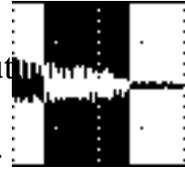
Draw (2) changes the cursor to a pencil and allows you to edit the wave data graphically. When the mouse button is held down, the pencil leaves a mark everywhere it is dragged. When the mouse button is released, the wave is updated with the drawing.
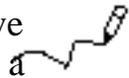
Move Marker (3) changes the cursor to a "⬌", which is used to move markers. If Move Marker mode is not selected, you can still move markers by holding down the *Option* key while dragging the mouse.

Log Coords (4) changes the cursor to a cross-hair, which is used to log wave coordinates to a journal. Each time the mouse is clicked on a wave, it's position (usually time) and value are sent to a journal. The coordinates are preceded with a header on the first click, as shown in the above illustration (e.g. "X:Hz   Y:dB"). The Log Coords To submenu specifies which journal receives the text.

Vertical Adjust (5) changes the cursor to a hand, which is used to move the selected wave up and down in its display. Moving a wave is purely a graphical operation and does not affect the wave data. Clicking once on the "⬍" symbol next to the wave name will cause that wave to snap back to it's non vertically adjusted position.
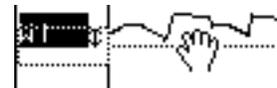
Grid Snap On/Off is used by the Draw mode, described above, to force the cursor to specific grid intervals while drawing.

Log Coords To is used by the Log Coords mode, described above, to specify which journal receives coordinate data.

JOURNAL MENU

A SoundScope journal is a text editor for entering notes, logging values and recording any information pertinent to your analysis. Text may be transferred between journals and the Clipboard using Cut, Copy and Paste. Journal files that have been saved to disk can be read directly from most spreadsheets and word processors.

Journal
Clear ▶
Save As... ▶
Load Text... ▶
Page Setup... ▶
Print... ▶
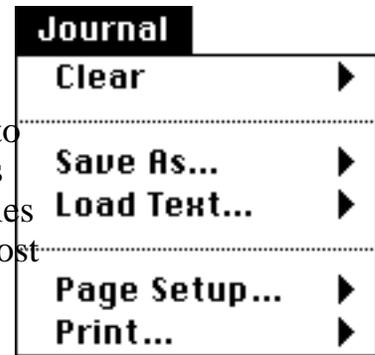
Clear... erases the contents of the specified journal. All changes that have not been saved will be lost.

Save As... writes the specified journal to disk, prompting you for a filename and location (folder) with the standard File dialog.

Load Text... reads from disk the contents of a text file that you select with the standard File dialog, placing the data in the specified journal.

Page Setup... presents the standard dialog for the current printer.

Print... presents the standard dialog and then prints the journal contents.

TASK MENU

SoundScope tasks let you set up sequences of events such as sound acquisition, analysis, playback, printing and file output.

Open… reads in a task from disk.

Run executes the task that you select from the hierarchical menu.

Stop halts the task that you select from the hierarchical menu.

CONTINUE EXECUTES THE TASK THAT YOU SELECT FROM THE HIERARCHICAL MENU, BEGINNING WHERE IT WAS PREVIOUSLY STOPPED.

Appendix A
Sound Analysis Computations

This appendix explains in detail how each SoundScope analysis function is computed. For information on how to perform the analysis, please see *Chapter 6, Front Panel*.

THE FOURIER TRANSFORM

This section introduces you to the basic tools and terminology of Fourier analysis, and describes how it is used in SoundScope. For a more detailed and comprehensive introduction, you may want to read a book on speech analysis such as Thomas Parsons' <u>Voice and Speech Processing</u>.

INTRODUCTION

Soundwaves may be described as fluctuations of sound pressure over a period of time. The most direct way of recording these fluctuations is to use a microphone to translate the sound pressure into a time-varying voltage. The resulting timewave may be plotted on a graph, typically with time on the horizontal axis and voltage on the vertical axis.

Information about the various frequency components contained in a timewave can be derived using Fourier analysis. Fourier analysis is based on the principle of Fourier's Theorem, which states that any timewave can be expressed as a sum of sinusoidal waves. In general, the more complex the timewave, the more frequency components enter into this sum. Each frequency component contributes a certain magnitude and phase. (The phase describes how the frequency component is shifted in time, e.g. to differentiate between a sine wave and a cosine wave.) The magnitude and phase values may be expressed as a function of frequency; this function is called the Fourier transform of the timewave. You can think of the Fourier transform as a "frequency wave," an alternate representation of the timewave. It contains all of the information of the original timewave, and in fact the timewave can be recovered by applying an inverse Fourier transform to the "frequency wave."

If you ignore the phase information of the Fourier transform and graph only the magnitude as a function of frequency, you obtain an estimate of the power spectrum, also called a frequency spectrum or energy spectrum. (The magnitude is usually expressed in dB, and the frequency in Hz.) The peaks in the power spectrum identify the strongest frequency components in the signal. For example, the power spectrum of the sound of a soloist tuning her instrument to middle A (440 Hz) would

have a sharp peak centered about 440 Hz. Other frequency components would be present as well, because of overtones created by the instrument, the presence of noise coming from the audience, etc. As another example, the power spectrum of a vowel usually contains peaks at the formant frequencies.

DIGITIZATION AND THE FFT

In order to understand how SoundScope uses Fourier techniques to analyze sounds, you should have a basic understanding of how sound is represented in the computer's memory. When the computer records a sound, a special piece of hardware called an analog-to-digital (A/D) converter translates the continuously-varying analog signal from the microphone into a sequence of numbers. In this digitization procedure, the signal is sampled at a certain sample rate (typically tens of thousands of times a second), and the digitizer assigns an integer value to each sample. Each integer value corresponds to a voltage; together, the set of all allowed integer values maps out the entire voltage range used by the digitizer. These coded values can later be converted back into sound — that is, played back through a loudspeaker — via a digital-to-analog (D/A) converter.

The sequence of numbers (digital data) captured by the A/D converter forms a wave, which is so named to suggest its correspondence to the original continuous analog waveforms. Just as a mathematician applies the Fourier transform to a continuous waveform, the computer applies a specially adapted version called the Discrete Fourier Transform to a wave. A short-cut method called the Fast Fourier Transform (FFT) lets a computer calculate Fourier transforms in a reasonable amount of time, but restricts the length of waves it can analyze to a power of two, like 16, 256 or 1024 points. (This restriction can be circumvented by a technique known as zero padding, discussed later in this chapter.)

The Fourier transform accepts a wave containing N samples that are equally spaced in time and returns magnitude and phase values for N/2 frequencies. The frequency components returned by an FFT are equally spaced in frequency from 0 to half the sampling frequency. Increasing the sampling rate improves the fidelity with which signals are captured by making the digitizer receptive to higher frequencies, but increases the memory required to store digitized sounds.

FRAMES, FILTERS, AND THE SHORT-DURATION FFT

Although it is possible to perform an FFT on waves of unlimited length, most kinds of frequency analysis look at relatively small portions of larger waves, called **frames**. (A typical frame has a duration of about 10 to 50 ms.) It is often useful to compile FFT results taken from a number of frames spaced evenly throughout an utterance. This procedure yields different results than performing one FFT on the entire utterance. It is important to consider how the size of an FFT frame affects calculation results.

A basic principle of Fourier analysis holds that time information and frequency information are complementary — the more you know about one, the less you know about the other. In theory, to know the exact frequency composition of a signal, you have to evaluate the Fourier transform from t = -infinity to t = +infinity. However, this precise spectrum contains no information about when the various frequency components appeared and disappeared. On the other hand, if you transform a finite segment of that signal, you obtain time information at the expense of some frequency resolution. The frequency resolution (also called bandwidth) describes how close together two frequency components can come and still be distinguished as separate peaks in the FFT.

In addition to frame duration, you may need to know how the frame was isolated from the rest of the signal. One method of isolating the frame is to multiply the entire signal by a wave called a rectangular window that has a value of 0 everywhere outside the frame and 1 inside. In fact, non-rectangular window shapes are used more often. The choice of window shape affects the frequency spectrum obtained by the FFT, and different window shapes are appropriate for different applications. The most common window used in speech analysis is the Hamming window. This window attains a maximum in the middle and tapers off close to zero at the edges. (The formula for the shape is given later in this chapter) The Hamming window has the advantage of introducing fewer spurious peaks in the frequency spectrum, with the disadvantage that its bandwidth is greater than that of a rectangular window of the same duration.

In some forms of analysis, many short-duration FFT's are performed on an entire sound utterance, each applied to a different section of the timewave. A prime example of this technique is found in the spectrogram, which shows the progression of the frequency spectrum over a period of time. Each FFT is applied to a Hamming-weighted frame. The position of the frame is advanced between consecutive FFT calculations by an amount of time called the frame advance. You can adjust the length (amount of time) of the frame by specifying the filter frequency, a parameter originally used to describe the frequency resolution of traditional analog spectrograph machines. A low filter frequency, such as 59 Hz, specifies a narrow bandwidth, which means that the spectrogram plot can resolve frequency components as small as 59 Hz apart. A high filter frequency, such as 184 Hz, specifies a wide bandwidth, indicating a coarser resolution. The advantage of the wide bandwidth spectrogram is that it is more sensitive to the timing of spectral events (e.g. the onset of a vowel) than the narrow bandwidth spectrogram.
Since multiplying a time waveform by a window corresponds to convolving the timewave's frequency spectrum with the FFT of that window (signal processing theory says this), there exists a relationship between the window width (ms), and the width of it's associated FFT (Hz), as noted in the table below:

Window Width (ms)	Filter Width (Hz)
33	45
25	59
10	150
8	184
5	300
3.2	450
2.5	600

MacSpeech Lab I and II refer to an 8ms window as a "300Hz filter", and refer to a 25ms window as a "45Hz filter"; whereas SoundScope refers to an 8ms window as a 184Hz filter, and a 25ms window as a "59Hz filter". Subsequently, MacSpeech Lab I & II's 45Hz filter is referred to as 59Hz in SoundScope, and
MacSpeech Lab I & II's 300Hz filter is referred to as 184Hz in SoundScope. The change is attributed to a more popular method for interpreting filter width (i.e. the newer method is used by other manufacturers). We apologize for any confusion that this may have caused.

BASIC DEFINITIONS

This section describes some computations that are common to many SoundScope analyses. The following section *Algorithms and Formulas*, assumes that you understand the material presented here.

THE FFT

An N-point FFT of the wave composed of discrete points A_0 A_1 ..., A_{N-1} where N is some integer power of two, is defined as:

$$F_k = \sum_{n=0}^{N-1} A_n \, e^{(-2\pi i) nk/N}$$

for k=0 through N-1. The result is a sequence of N complex numbers, each number F_k representing the magnitude and phase at a frequency equal to k/N times the sampling rate of wave A_n. Since the SoundScope FFT routine always expects wave A_n to be composed entirely of real numbers, the output points possess a complex conjugate symmetry: $F_K = F^*_{k-N/2}$ where $N/2 \leq k \leq N-1$. As a result, wave F_k only needs to be specified for values of k from 0 through N/2, corresponding to frequencies ranging from 0 to the Nyquist frequency (half the sampling rate).

Note that, although the formula is valid for all positive integers N, the FFT algorithm used to compute the formula only works when N is an integer power of two.

THE INVERSE FFT

An N-point inverse FFT of the sequence F_0, F_1, ..., F_{N-1} is defined as:

$$A_n = \frac{1}{N} \sum_{k=0}^{N-1} F_k \, e^{(2\pi i) nk/N}$$

for n=0 through N-1. This equation is the dual of the FFT equation above. In general, the result A_n is a sequence of N complex numbers equally spaced in time. If the sequence F_k possesses conjugate symmetry (see the FFT definition), then each number A_n must be real.

Note that, although the formula is valid for all positive integers N, the inverse FFT algorithm used to compute the formula only works when N is an integer power of two.

THE HAMMING WINDOW
The Hamming window is an N-point wave defined by the formula

$$\text{Hamm}_n = 0.54 - 0.46 \cos (2\pi\, n/(N-1))$$

where n ranges from 0 through N-1. The window attains a minimum of 0.08 at both edges and peaks at a value of 1 in the middle. The bandwidth ("filter frequency") of the spectrum resulting from an FFT of a frame of data which has been multiplied point-by-point by a Hamming window is given by:

$$\text{Bandwidth (in Hz)} = 1475 / \text{duration of frame (in ms)}.$$

ZERO PADDING
The number of samples included in a frame for an FFT calculation must equal an even power of two. To guarantee this, SoundScope sets the frame size to a number 2^N that equals or exceeds the size of the Hamming-weighted frame to be processed. Any points beyond the actual data are padded with zeroes.

6 dB PRE-EMPHASIS
A 6 dB pre-emphasis option is available in many SoundScope analyses. When this option is enabled, the wave to be analyzed is processed so as to emphasize the higher frequencies before the analysis is performed. The transformation is achieved by "differentiating" the wave to be analyzed, i.e. by generating a new wave whose points equal the differences of consecutive points in the original wave. (Note that the 6 dB pre-emphasis generates a new wave that is used internally in the analysis calculations; the original wave being analyzed is unaffected.) The procedure boosts higher frequencies at the rate of 6 dB per octave. This option is particularly useful for speech analysis because it counteracts the natural high frequency roll-off of speech, yielding less downward-skewed graphs for those plots that display frequency on the horizontal axis.

Note about Decibel Values: Many SoundScope analyses such as the FFT and LPC return results in decibels (dB). It is important to realize that the decibel values have not been calibrated to any absolute scale, so they should be interpreted as relative values. If you need to calibrate to an absolute scale, you will have to digitize some reference signals and use the decibel readings you obtain as reference dB levels. When you are making recordings with a microphone, you should keep in mind that the distance between the microphone and the speaker's mouth affects dB levels.

ALGORITHMS AND FORMULAS

This section describes the computational details behind each form of analysis. It explains how the result wave of each analysis (e.g. average spectrum) is calculated from the source wave (e.g. human speech). Familiarity with the previous section, *Basic Definitions*, is assumed.

AVERAGE SPECTRUM

An Average Spectrum plot displays the average power spectrum of an entire segment or wave. The spectrum is graphed in terms of dB versus frequency. The following algorithm is used:

1. For each frame:
   - Apply optional 6dB pre-emphasis.
   - Hamming-weight the frame.
   - Take FFT (magnitude values only), after zero padding, if necessary.
   - Take $20*\log_{10}$.
   - Apply optional smoothing.
2. Average the results for each frame to obtain result wave.

CEPSTRUM

The cepstrum of a speech waveform is the Fourier transform of the logarithm of its power spectrum. The motivation for defining this waveform arises from the excitation-modulation model of speech, which hypothesizes a linear relationship between speech and glottal excitation: $X(f) = G(f) H(f)$, where X and G are the Fourier transforms of the speech and glottal excitation waveforms, respectively, and H is the vocal tract transfer function. The argument f denotes frequency. The cepstrum $C(q)$ is given by:

$$C(q) = \text{FFT}\{\log |X(f)|^2\}$$
$$= \text{FFT}\{2 \log (|G(f)| |H(f)|)\} = 2\, \text{FFT}\{\log |G(f)|\} + 2\, \text{FFT}\{\log |H(f)|\}\ .$$

The argument q has the unit of time; it has been dubbed "quefrency" to suggest its duality with frequency. (Similarly, the name "cepstrum" was chosen to mirror the spectrum from which it derives.) The logarithm serves the important function of converting the product of G and H into a sum, so that the peaks due to G will be visible in $C(q)$ separately from those due to H. If $X(f)$ is the Fourier transform of a vowel sound, then the peak due to H is found near q=0, whereas the one due to G is found at a quefrency corresponding to the fundamental period of the glottal frequency. Peaks at multiples of the fundamental quefrency may also be visible.

The excitation-modulation model is idealized and must be modified to account for the presence of noise that is inherent to all speech: $X(f) = G(f) H(f) + \text{NOISE}(f)$.

The expression for C(q) no longer breaks down neatly into a sum of G and H components, but the main features of the graph remain visible provided the noise is not too heavy. The following algorithm is used:

1. Take frame of speech about the marker.
2. Apply optional 6dB pre-emphasis.
3. Take FFT (magnitude values only), after zero padding, if necessary.
4. Take $\log_{10}$, then take FFT (magnitude values only) again.

ENERGY
An energy plot describes the shape of the amplitude squared of a wave's peaks. The following algorithm is used:

1. For each frame, find the maximum of the square of all samples in the frame.
2. Divide by 32768 to convert to energy, in units of volts squared.

ENVELOPE
The envelope of a wave is the shape that describes the amplitude of its peaks. The envelope is related to the energy of the wave; see also the *Energy* entry above. The following algorithm is used:

1. For each frame, find the maximum of the absolute value of all samples in the frame.
2. The sequence of maxima (one obtained for each frame) forms a wave.

FFT
An FFT plot shows the power spectrum of a frame of timewave data centered about the marker. The spectrum is graphed in terms of dB versus frequency.
The following algorithm is used:

1. Apply optional 6dB pre-emphasis.
2. Hamming-weight the frame.
3. Take FFT (magnitude values only), after zero padding, if necessary.
4. Take $20*\log_{10}$.
3. Apply optional smoothing.

FUNDAMENTAL FREQUENCY (F_0)

Fundamental frequency (F_0) corresponds to the rate of the speaker's vocal cord vibration. This vibration appears as a periodic component in a speech waveform whose frequency varies over time. Naturally, the fundamental frequency can only be evaluated during voiced intervals. Fundamental frequency is determined using either an auto-correlation technique or a peak-picking algorithm, as specified in the

F_0 Setup dialog box.

The auto-correlation technique is a standard one (see Rabiner and Schafer in <u>Digital Processing of Speech Signals</u>, pp. 141-144 and 150-154 for details), and uses a frame size equal to the minimum acceptable period specified in the F_0 Analysis Options dialog. Calculation speed is directly proportional to the Acceptable pitch periods range, specified in the same dialog. If the Fast option is selected, a center clipper is applied to the source wave, thereby speeding up the auto-correlation calculation (at the expense of some accuracy). This clipper converts all samples into either 0, 1 or -1 according to the following rule: samples near 0 Volts are converted to 0, samples above a certain threshold are converted to 1, and samples below a certain threshold are converted to -1.

The peak-picking algorithm identifies the individual cycles in the source wave in order to calculate the fundamental frequency. The technique used is described in Gold, "Computer Program for Pitch Extraction." JASA, 34(7). The peak-picking algorithm produces a sequence of pitch values, one value for each pitch period. Variations in pitch cause this sequence of numbers to be spaced somewhat irregularly in time.

Both pitch-tracking techniques limit their search for pitch values to the pitch range specified in the F_0 Options dialog. Gaps appear in the F_0 plot in those regions in the source wave where no pitch has been detected in the pitch range.

HARMONIC-TO-NOISE RATIO (HNR)
Vowel sound waveforms may be regarded as a sum of two components: a harmonic (i.e. periodic) component, which characterizes the fundamental frequency F_0, and a noise component. The ratio of the energy of the harmonic component to that of the noise component is called the harmonic-to-noise ratio (HNR). An HNR plot shows the ratio (in dB) over time. This ratio has been used as an index of the degree of hoarseness of speech, hoarse speech having more noise and a weaker harmonic component than normal speech.

SoundScope uses the algorithm described by Yumoto, Gould, and Baer in "Harmonic-to-noise ratio as an index of the degree of hoarseness" JASA 71(6). A brief explanation follows. A peak-picking algorithm (see *Fundamental Frequency* above) identifies the cycles (pitch periods, from one peak to the next) in the wave. (The autocorrelation method is not used in computing HNR.) For each pitch period P, the following calculations are performed: To cancel out noise, the speech wave is averaged over *n* consecutive pitch periods centered about P, resulting in a wave one pitch period long that is almost entirely harmonic. (The number *n* of consecutive pitch periods is specified in the HNR Setup dialog. For the statistics calculation, *n*

is fixed at 31. The *n* pitch periods must be similar.) This harmonic wave is subtracted from the one pitch period centered about P, yielding a noise wave. HNR is the ratio of the two. The following calculation is used:

M = number of points in one pitch period
N = number of pitch periods averaged to find harmonic energy

$$\text{Wave}_{\text{AVGi}}(n) = \frac{1}{N} \sum_{i=N/2}^{N/2-1} \text{Wave}_{\text{Time}}\left((iM+N) \bmod M\right)$$

$$\text{Harmonic Energy} = H_i = \sum_{n=1}^{M} \text{Wave}_{\text{AVGi}}(n)^2$$

$$\text{Noise Energy} = N_i = \sum_{n=1}^{M} \left(\text{Wave}_{\text{AVGi}}(n) - \text{Wave}_{\text{Time}}(n)\right)^2$$

$$\text{HNR (dB)}_i = 10 \log_{10}\left(\frac{H_i}{N_i}\right)$$

Note that HNR can only be computed on a voiced interval that has a valid F_0 calculation over regions with similar pitch periods (e.g. vowels).

HORIZONTAL SPECTRAL SLICE (HS SLICE)
A horizontal spectral slice plot shows a horizontal "slice" of a spectrogram — a graph of intensity in dB versus time at a specified constant frequency.
The following algorithm is used:

1. For each frame:
   * Apply optional 6dB pre-emphasis.
   * Hamming-weight the frame.
   * Take FFT (magnitude values only), after zero padding, if necessary.
   * Take $20*\log_{10}$ of the magnitude at the specified frequency.
   * Apply optional smoothing.
2. The sequence of dB levels over time produces the result wave.

JITTER
Jitter is a time-varying parameter that measures pitch perturbation, i.e. how quickly the pitch period changes in a vowel utterance over time. A jitter plot is formed by tracking the pitch period over 3 successive pitch cycles, using the peak-peaking algorithm described in *Fundamental Frequency*, and the duration of zero crossings preceding each peak. (The autocorrelation pitch-tracking method is not used in

computing jitter.) The following algorithm is used:

$$\text{Period}_{\text{AVG}_i} = \frac{P_{i-1} + P_i + P_{i+1}}{3}$$

$$\text{Period}_{\text{Difference}_i} = \text{Period}_{\text{AVG}_i} - \text{Period}_i$$

$$\text{Jitter}_i \ (\%) = 100 \ \frac{\text{Period}_{\text{Difference}_i}}{\text{Period}_{\text{AVG}_i}}$$

where P_{i-1}, P_i, P_{i+1} are consecutive pitch periods. Note that jitter only makes sense on sustained vowels with a valid F_0. The jitter statistic (F_0 *Perturbation*) is described later in this appendix. Shimmer also measures perturbation, but of amplitude rather than pitch.

LINEAR PREDICTIVE CODING (LPC)
An LPC plot shows an estimate of the vocal tract transfer function, graphed in terms of dB versus frequency. The peaks in the LPC plot of a vowel sound occur at formant frequencies. The plot is derived from a set of **LPC coefficients** which are calculated using an autocorrelation method (with rectangular window) known as Durbin's recursive procedure (see Rabiner and Schafer, p. 411); the length of the frame used in the procedure is specified in the LPC Setup dialog. The estimate of the vocal tract transfer function is based on the assumption that the transfer function H(z) is an all-pole function of the following form:

$$H(z) = \frac{\text{GAIN}}{(1 + a1z\text{-}1 + a2z\text{-}2 + ... + anz\text{-}p)}$$

where a_1 through a_p are the LPC coefficients. The **LPC plot** displays the function $10 \log_{10} |H|^2$ on the vertical axis (in dB) versus frequency on the horizontal axis (in Hz). Alternatively, the LPC Setup dialog allows you to display the **Residual Signal**, explained below, in place of the LPC plot.

The reason for the name "linear predictive coding" is that the coefficients appearing in the transfer function serve a <u>predictive</u> function: given the p most recent data points y_{n-p}, y_{n-p+1}, ..., y_{n-1} of a speech wave, the estimated value y'_n of the next point y_n may be calculated as follows:

$$y'_n = \text{-}a_1 y_{n\text{-}1} - a_2 y_{n\text{-}2} - ... - a_n y_{n\text{-}p}$$

i.e., y'_n is a linear combination of the p points that precede it. The difference between the actual and predicted values forms a prediction error sequence e_n, defined as $e_n = y_n - y'_n$, also known as the **residual signal**. Besides measuring the error of the prediction calculation, the sequence e_n may also be regarded as the voice source that drives the vocal tract and which transforms directly into speech via the transfer function H(z).

The optimal choice of the number of LPC coefficients depends mainly on the sample rate of the speech wave being analyzed. All LPC-related analyses in SoundScope allow you to choose a default number of LPC coefficients given by the following formula (see Rabiner and Schafer, p. 419):

> Number of coefficients = (Sample rate in kHz) + 3
>
> e.g. the default # of coefficients for 10 Ksamples/sec is 13.

However, the number of coefficients may be changed in some analyses. In general, increasing the number helps to resolve formant peaks that are close together in frequency but may also introduce extraneous features into the LPC plot. The default values are optimal for male speech; women or children may require analysis with more coefficients.

LPC HISTORY

The LPC History plot presents a spectrogram-like "history" of the LPC transfer function, with time on the horizontal axis and frequency in Hz on the vertical axis. A point is shaded in on the graph only if the transfer function H(z), calculated at the time and frequency specified by the point, exceeds the **energy cutoff** (measured in dB) specified by the user for the graph. The LPC History plot resembles a very high-contrast spectrogram. The LPC History Setup dialog gives you the option to perform formant tracking, and plot one or more formant lines on top of (or in place of) the LPC History plot. The following algorithm is used:

1. For each frame:
   - Apply optional smoothing to source wave.
   - Compute LPC coefficients and transfer function H(z).
   - If the `LPC history option` has been enabled, plot a vertical stripe of the LPC History, shading in only those frequencies whose energy level exceeds the specified energy cutoff.
   - Advance to the next frame.
2. If any of the `formant options` have been enabled, compute the formant waves and plot them.

LPC RESIDUAL

An LPC Residual plot shows an estimate of the voice source that drives the production of speech according to the linear predictive coding model. This estimate, called the **predictive error** or **residual wave**, corresponds roughly to the vibration of the vocal cords. The following algorithm is used:

1. For each frame:
   - Compute LPC coefficients.
   - Generate a residual signal for the length of the frame.
   - Advance to the next frame.
2. Concatenate (i.e. string together) the residual signal of each frame to generate the result wave.

SHIMMER

Shimmer is a time-varying parameter that measures amplitude perturbation, i.e. how quickly the amplitude changes in a vowel utterance over time. A shimmer plot is formed by tracking the pitch period over 11 successive pitch cycles, using the peak-peaking algorithm described in *Fundamental Frequency*, where the period is from one peak to the next. (The autocorrelation pitch-tracking method is not used in computing shimmer.)
The following algorithm is used:

$$A_{AVGi} = \frac{A_{i-5} + A_{i-4} + \ldots + A_{i+5}}{11}$$

$$A_{Difference_i} = A_{AVGi} - A_i$$

$$Shimmer_i (\%) = 100 \frac{A_{Difference_i}}{A_{AVGi}}$$

where A_i and A_{i+1} are the peak amplitudes of consecutive pitch periods. Note that shimmer only makes sense on sustained vowels with a valid F_0. The shimmer statistic is described later in this appendix. Jitter also measures perturbation, but of pitch rather than amplitude.

SPECTROGRAM

A sound spectrogram (also called a "sonogram") is a visual display of the frequency analysis of a time waveform. The horizontal axis is time; the vertical axis shows frequency. The relative darkness at any point represents the relative energy level at that frequency and time. Traditionally, analog sound spectrographs produced

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spectrograms by converting the energy output of an analyzing filter (184 Hz for wideband and 59 Hz for narrowband) into a variable degree of burn on thermally sensitive paper.

The Spectrogram Setup dialog gives you the option to perform formant tracking, and plot one or more formant lines on top of (or in place of) the spectrogram plot. The following algorithm is used:

1. For each frame:
   • Apply optional 6dB pre-emphasis.
   • Hamming-weight the frame.
   • Take FFT (magnitude values only), after zero padding, if necessary.
   • If the `Spectrogram option` has been enabled, plot a vertical stripe of the Spectrogram. The spectrogram is white everywhere the energy level is less than or equal to the specified minimum energy level; energy levels equal to or greater than the specified maximum energy level are shaded black; and energy levels between the minimum and the maximum assume intermediate levels of shading. (See the note below on color spectrograms.)
   • Advance to the next frame.
2. If any of the `formant options` have been enabled, compute the formant waves and plot them.

Note about color spectrograms: If the color option in the Spectrogram Setup dialog box has been selected, then the shading used falls along a spectrum of shaded tones. This spectrum is bounded by two colors that have been selected in the Color Picker dialog box, one designating the color of the minimum energy level, the other the color of the maximum.

SPLINE
Spline-fitting is a technique for fitting cubic (third-degree) polynomials as closely as possible to waves. A spline plot shows a very smoothed-out version of the source wave.

To produce a spline, the source wave or segment is smoothed out by averaging adjacent points to eliminate noise and is partitioned into smaller segments. The points are averaged together in each small segment to generate a data point called a "knot"; the sequence of knots approximates the entire wave within the current segment. The spline-fitting procedure generates cubic polynomials to connect the knots, requiring that both the first and second derivatives of adjacent polynomials match at the knot between them. A wave based on these polynomials is then generated that passes through all the knots.

The knot width is specified in the Spline Setup dialog, which gives the choice of displaying the spline, as well as its first and second derivatives.

ZERO CROSSING

The zero crossing function counts how often a sound wave crosses the time axis — i.e., how often the signal swings from positive to negative or negative to positive voltage. A zero crossing plot shows the rate of zero crossings (in Hz) over time.

The zero crossing function is computed on a frame-by-frame basis. The zero crossings of the source wave are counted in each non-overlapping frame. The count in each frame is divided by the length of the frame, specified in the Zero Crossing Setup dialog. These frequency values are then linearly interpolated to yield the result wave.

Note that the zero crossing rate grows fairly large during periods of silence. This behavior arises from the fact that even an apparently "silent" signal hovers very close to zero volts and randomly crosses the zero line fairly often. (Of course, a perfectly silent signal — one which remains exactly at zero volts — has a zero crossing rate of zero.)

SOUND STATISTICS

In the explanation of each statistic below, the wave or segment on which the statistics are calculated is referred to as the **source wave**. Many of the values are highly dependent on a valid F_0, and the acceptable F_0 range in the options dialog.

BREATHINESS

This parameter measures the high-frequency content of the source wave. The formula for breathiness (in dB) is given by:

$$\text{breathiness} = 10 \log_{10} \frac{\text{energy (2nd Back Difference)}}{\text{energy (Source)}}$$

Source is the source wave, the energy of a wave is the sum of the square of each point in the wave, and the nth point of 2nd Back Difference is defined as:

$$\text{2nd Back Difference}_n = \text{Source}_n - 2 \text{ Source}_{n-1} + \text{Source}_{n-2}$$

The 2nd Back Difference wave is the discrete version of the 2nd derivative of the Source wave, which is equivalent to the Source wave transformed by a high-frequency emphasis, applied at the rate of 12 dB per octave. Dividing the energies of the 2nd Back Difference and the Source waves produces a number indicating how much of the energy of the Source resides in the high-frequency range.

F_0 AVERAGE

The average F_0 of a speech wave is defined to be the arithmetic mean of the fundamental frequency measured by the peak-picking algorithm.

F_0 KURTOSIS

This parameter measures the fourth moment of the pitch values found in a speech wave and is defined by the formula:

$$\text{kurtosis} = \left[\frac{1}{N} \sum_{i=1}^{N} \left(\frac{f_i - f_{avg}}{\sigma} \right)^4 \right] - 3$$

where f_1, f_2, ..., f_N are the consecutive fundamental frequencies measured by the peak-picking algorithm and f_{avg} is the average frequency of the source wave. S is the standard deviation of the frequency. Kurtosis measures the peakedness or flatness of the frequency distribution relative to a normal (gaussian) distribution; the -3 term in the formula makes the kurtosis zero for a normal frequency distribution, positive for a leptokurtic distribution, and negative for a platykurtic distribution. For more information, see pp. 474-475 of Numerical Recipes in C.

F₀ PERTURBATION

The F_0 perturbation (also known as Jitter or relative average perturbation, "RAP") of the source wave is defined by the formula:

$$RAP = \frac{\sum_{i=1}^{N} |Period_{Difference_i}|}{\sum_{i=1}^{N} Period_i}$$

See *Jitter* (which is discussed earlier in this chapter) for the definition of Period $_{Difference}$. For more information, see <u>Clinical Measurements of Speech and Voice</u>, p. 178.

F₀ Range

The F_0 range of a speech wave measures the difference between the highest and lowest fundamental frequencies tracked by the peak-picking algorithm. This difference, measured in semitones (ST), is calculated using the following formula:

$$f0\ range = 12 \left[\frac{\log \left(\frac{f_{max}}{f_{min}} \right)}{\log 2} \right]$$

where f_{max} and f_{min} are the highest and lowest frequencies, respectively, of the source wave. The value is also displayed in Hz.

F₀ SKEWNESS

This parameter measures the third moment of the pitch values found in a speech wave and is defined by the formula:

$$skewness = \left[\frac{1}{N} \sum_{i=1}^{N} \left(\frac{f_i - f_{avg}}{\sigma} \right)^3 \right]$$

where f_1, f_2, ..., f_N are the consecutive fundamental frequencies measured by the peak-picking algorithm and f_{avg} is the average frequency of the source wave. S is the standard deviation of the frequency. Skewness measures the asymmetry of the frequency distribution relative to its mean; a normal (gaussian) frequency distribution has zero skewness, a distribution with an elongated "tail" extending out to positive frequencies has positive skewness, and a distribution with a tail going out to negative frequencies has negative skewness. For more information, see pp. 474-475 of <u>Numerical Recipes in C</u>.

F$_0$ Standard Deviation

The F$_0$ standard deviation of a speech wave is defined by the formula:

$$\text{f0 std. dev.} = \sqrt{\frac{1}{N-1}\sum_{i=1}^{N}\left(f_i - f_{avg}\right)^2}$$

where f_1, f_2, ..., f_N are the consecutive fundamental frequencies obtained by the peak-picking algorithm and f_{avg} is the average frequency of the source wave.

Harmonic-to-Noise Ratio (HNR)

The HNR parameter in the Sound Statistics dialog equals the arithmetic average of the HNR values computed over time. (See *Harmonic-to-Noise Ratio (HNR)*, discussed earlier in this appendix, for details.)

Shimmer

The shimmer (also known as amplitude perturbation quotient, "APQ") of the source wave is defined by the formula:

$$APQ = \frac{\sum_{i=1}^{N}\left|A_{\text{Difference}_i}\right|}{\sum_{i=1}^{N}A_i}$$

See *Shimmer*, discussed earlier in this appendix, for the definition of $A_{\text{Difference}}$.

Voiced, Unvoiced and Silent

These three parameters measure what portion of the source wave is voiced, unvoiced and silent. All sounds are assumed to fall into one of these three categories. These statistics are computed with the algorithm described in <u>Digital Processing of Speech Signals</u>, pp. 132-134.

Appendix B
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Appendix C
Tips

This appendix contains several Tips on a variety of issues.