

BOYS TOWN NATIONAL RESEARCH HOSPITAL

EMAV: Otoacoustic Emission Averager

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ABSTRACT

EMAV is a program for measuring and displaying otoacoustic emissions. The distortion-product otoacoustic emission (DPOAE) test features artifact rejection, measurement-based stopping rules, and noise estimation at the same frequency as the distortion product. The transient-evoked otoacoustic emission (TEOAE) test features octave-band analysis and nonlinear-transducer-artifact correction. The PROBE test estimates Thévenin-equivalent impedance and pressure of the sound source. EMAV uses high-quality soundcards for stimulus generation and synchronous averaging for noise reduction. Currently supported soundcards include Indigo IO (from Echo Digital Audio) and the Babyface (from RME). An OAE probe microphone, such as the Etymotic ER-10C, is required for OAE measurement. Computer systems without special audio hardware can still view EMAV data files or print data displays to a PostScript or PCL printer. This document describes EMAV version 3.32.

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1. INTRODUCTION

EMAV is a contraction of “emission averager.” The user interface for the DPOAE test in EMAV is similar in style to CUBeDIS, a DPOAE measurement program written by Jont Allen. The methods used by EMAV for TEOAE tests are similar to those developed for the ILO88 OAE averager from Otodynamics. EMAV was designed to provide enough flexibility to allow experimenting with new ideas, while being friendly enough for routine use in a clinical setting.

EMAV was originally designed to use the same hardware as CUBeDIS: an Ariel DSP-16+ board, a pair of ER-2 tube-phones with a 40 dB step-down transformer, and an Etymotic ER-10B with 40 dB of gain. Recent versions of EMAV support any Windows soundcard that has the ability to synchronize input and output. The CardDeluxe (from Digital Audio Labs) is currently the most widely used soundcard at BTNRH for data collection. The Indigo IO (from Echo Digital Audio) and Lynx L22 (from Lynx Studio Technology) are also being used. Custom-made cables allow these soundcards to be used with the ER-10C¹. Under Windows, EMAV may use MME, WDM, or ASIO device drivers; however, synchronous averaging requires that the soundcard synchronize input and output consistently. A Linux version of EMAV is also available and uses ALSA device drivers. See the Appendices for additional information about soundcard installation and calibration.

2. TUTORIAL

A “setup” wizard is available for installing the EMAV program on a Windows system. See Appendix A for software installation details.

The most convenient way to use EMAV is to create a dedicated folder to contain all the protocol and data files associated with EMAV. To designate a working directory before using EMAV in Windows, right-click on the EMAV icon on the Desktop and choose the Properties/Shortcut tab. In the “Start in” text field enter the pathway of the folder containing the necessary protocol (list) files that you will be using and the current configuration file.

Prior to data collection, sound card parameters should be measured and set in the configuration file. These parameters should be set for each unique hardware configuration so that the effects of different loads can be taken into account. Within EMAV the only directly accessible files are those found in the current working directory.

All of the primary testing options in EMAV (DPOAE, TEOAE, Probe, Cavity) have three phases: Check-fit, Calibration and Measurement.

Check-fit

During the *check-fit* phase, both time and frequency representations are displayed and continuously updated. Each new waveform displayed on the screen is superimposed over previously displayed waveforms. The screen is cleared automatically every few seconds or whenever you press *Space*. You can judge the quality of the fit by viewing the recorded waveform and corresponding spectra that are displayed on the screen. If the fit is satisfactory, press *Enter* or select *Continue* to proceed to the calibration phase. Otherwise, press *Escape* to cancel the test and return to the top-level menu selection. The check fit phase is similar in DPOAE, TEOAE and Probe tests.

DPOAE Calibration and Measurement

The *calibration* phase of the DPOAE test presents chirps to each of the two receivers (loudspeakers) in sequence. The program uses the measured response to these chirps to calculate the voltage needed to produce any desired sound pressure level at the microphone. Press *Y* to accept the calibration and proceed to the measurement phase. Otherwise, you can press *N* to repeat the check-fit and calibration. The calibration stimuli may be specified as waveform files or may be generated by EMAV. The calibration response waveforms are saved in MATLAB-compatible (*.CAL) files.

The first step in the *measurement* phase of the DPOAE test is to select the stimulus conditions to be presented to the ear. This may be done by creating list files that contain lists of stimulus conditions (or protocols) ahead of time or these files can be created through EMAV by selecting *File* and then *Create List File*. If you have previously created list files and placed them in the working directory, they will be displayed when you select the *DPOAE* test

¹ The maximum sound output of the ER-10C is limited to about 75 dB SPL unless the Preamp box is modified to bypass the “receiver equalization” that is normally “in series” with the receivers (loudspeakers) and which attenuates the output level at low frequencies by about 20-dB.

option. If you create a new list file it will be saved in the working directory. Select from the list by using the Up/Down cursor keys and pressing **Enter** at the desired protocol file. Or, press **Escape** to remove the list of previously-saved protocols and create a new list of stimulus conditions. Once the measurement is underway, a countdown of the number of sweeps averaged is displayed. The measurement of a stimulus condition is terminated when any of the three specified stopping conditions (time, noise, signal-to-noise ratio) specified in the list file are met. The measured levels of F1, F2, and Fd are displayed on the screen in the upper right box. Ld+ represents the emission level at the 2*F1-F2 frequency and Ld- represents the noise level at the same frequency. The DPOAE measurements are stored in text (DAT) files in a format designed to be usable with the BTNRH PLT program. See Figure 1 for an example of the results of a typical DPOAE test.

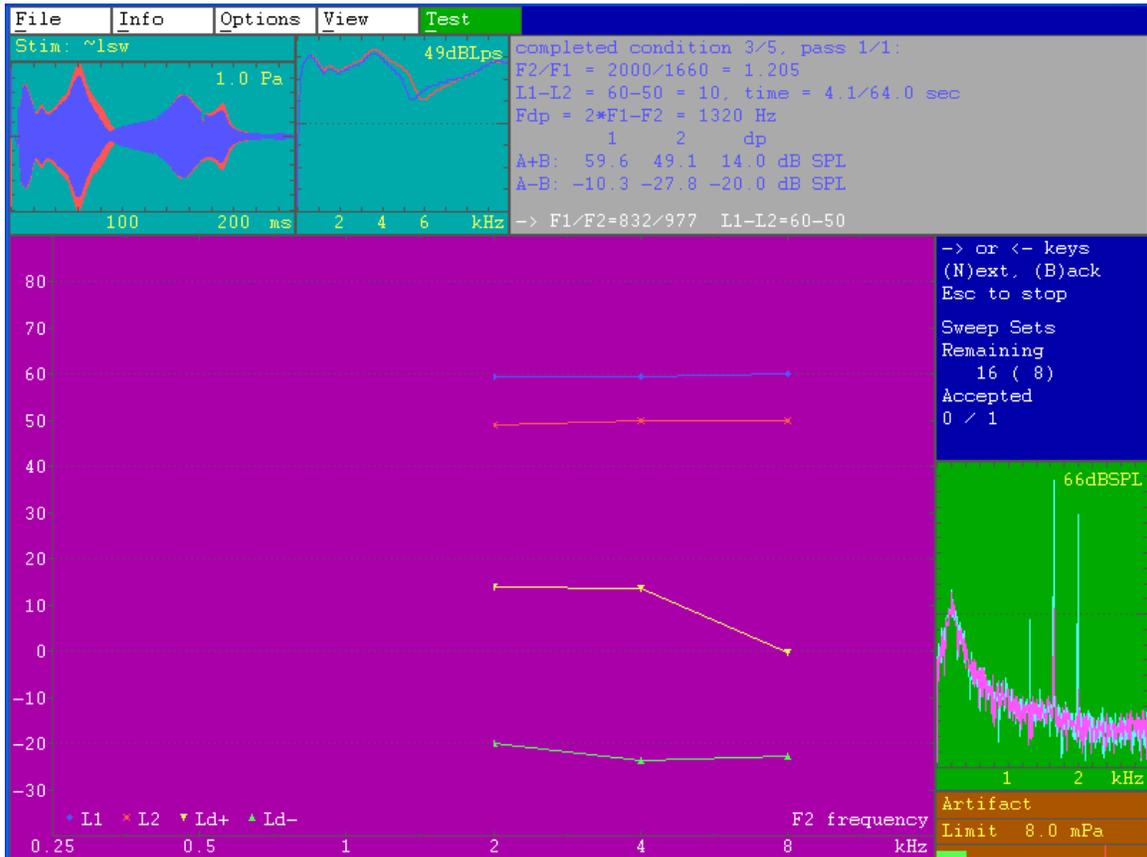


Figure 1. DPOAE test display. The calibration waveform using a frequency sweep stimulus is shown in the upper-left panel of the display and the spectrum of this waveform is shown in the adjacent panel. The upper right panel contains information in blue regarding the most recently completed condition and the print in white at the bottom of the panel is information regarding the current condition. The levels of the two stimulus tones measured at the plane of the probe and the distortion product are shown in the lower-left panel. The frequency of F_2 is indicated on the abscissa and its level (50 dB SPL) indicated by the x symbol. The frequency of F_1 was a fixed ratio of 1.2 below F_2 and its level (60 dB SPL) is indicated by the + symbol. The level of the $2F_1-F_2$ distortion product is indicated by the inverted yellow triangles and the noise level at this frequency is indicated by the upright green triangles. The bottom-right panel is the response window. It shows a spectral display of a running average of the stimuli and response components of ear canal pressure. The A+B average is light blue and the A-B (noise) is in pink. The middle left display contains information regarding the current stimulus condition. The number in parenthesis represents the number of (one-quarter second) sweeps in each sub-average. The upper-right panel contains information regarding the most recently completed condition (in blue) and information regarding the current condition (in white at the bottom of the panel).

TEOAE Calibration and Measurement

The *calibration* phase of the TEOAE test allows automatic adjustment of the signal amplitude for "in the ear" calibration. Within the calibration phase, you can press **A** to toggle the automatic signal amplitude adjustment on and off². The noise-level rejection threshold can also be adjusted during the calibration phase using the Left/Right cursor keys. When calibration is complete, press **Enter** to proceed to the measurement phase. The calibration stimuli may be specified as waveform files or may be generated by EMAY. The calibration response waveforms are saved in MATLAB-compatible (CAL) files.

During the TEOAE measurement, a countdown of number of sweeps is updated continuously. The TEOAE waveform and spectrum display is updated as often as desired by pressing the space bar or will update automatically every 5 seconds. You can also press **Escape** to terminate the test early or press **F9** to suspend (pause) the test (e.g., in case you need to talk to someone in the middle of the test). The TEOAE responses are stored in a special binary (TOK) format. See Figure 2 for an example of the results of a typical TEOAE measurement.

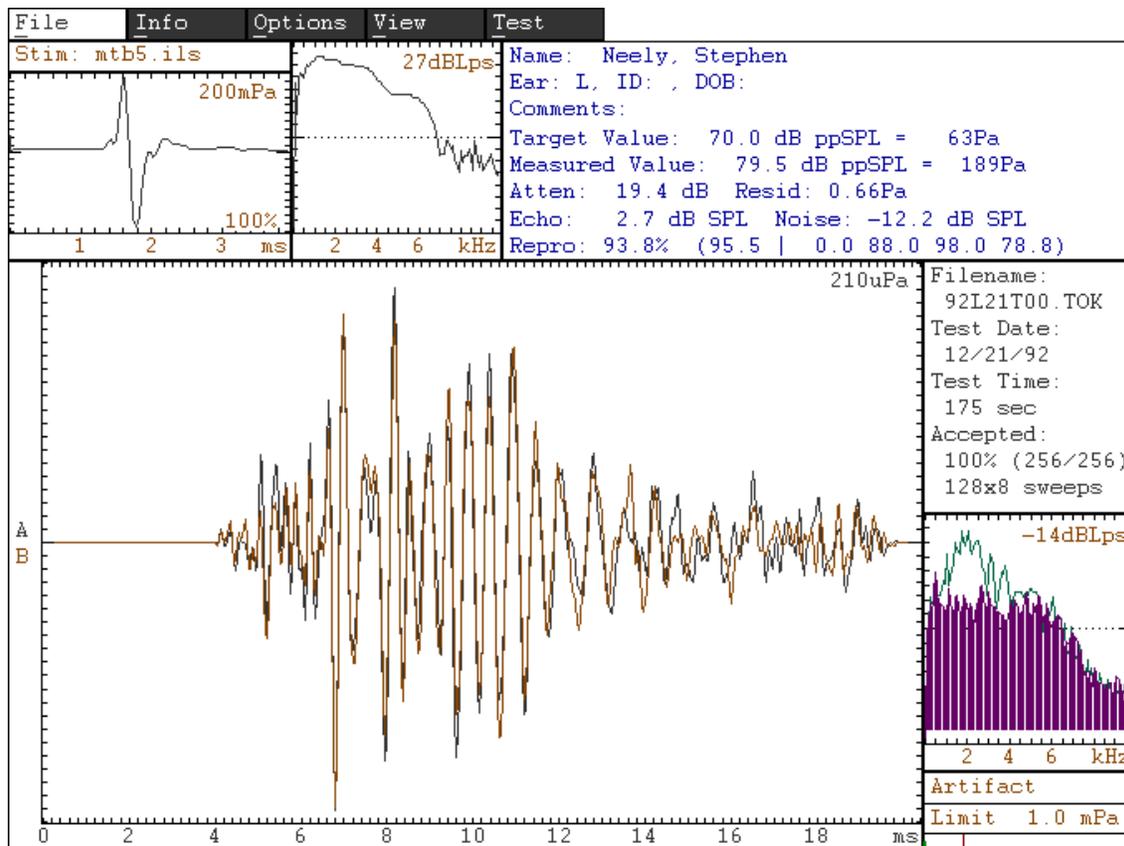


Figure 2. TEOAE test display. The stimulus was a band limited click at a level of 79.5 dB ppSPL. The stimulus waveform and spectrum are shown in the upper left and adjacent panels of the display. The response waveform (echo) is shown in the large panel at the lower-left of the display. The response spectrum (to the right of the waveform) indicates a response peak of -20 dB L_{ps} at about 2 kHz and a noise floor about 20 dB below the response. The overall echo level is 2.7 dB SPL and overall noise level is -12 dB SPL.

² When the adjustment is turned on, you can also try pressing "R" to apply an "error correction" to the last part of the stimulus. This residual error correction (REC) is designed reduce the stimulus artifact due to transducer (i.e. receiver) nonlinearity. The REC is computed by inverse filtering the residual in the stimulus interval of the waveform that remains after adding the four parts of the response. The inverse-filtered residual is added to a stored correction signal that is added to the fourth part of the stimulus waveform. The REC process requires several iterations and is still somewhat experimental. Although REC helps to make the ER-10C usable for TEOAE measurements, a better solution would be to use a more linear sound source.

Probe Calibration and Measurement

The probe option allows EMAV to calibrate stimuli using Forward Pressure Level (FPL) or Sound Intensity Level (SIL) instead of Sound Pressure Level (SPL). Theoretically, SIL and FPL provide measures of stimulus level that better indicate what reaches the eardrum because they are not affected by standing waves. SIL and FPL calibration both require that the acoustic load impedance of the ear canal be measured, which requires that the Thévenin-equivalent impedance and pressure of the sound source be determined from a set of known acoustic loads.

The first step in FPL or SIL calibration is to estimate the source impedance and pressure. A chirp stimulus is presented in SPL to each of 5 tubes of known length. The length of the tubes were selected so that resonant peaks occur at 2, 3, 4, 6 and 8 kHz. See Appendix E for additional details about this set of cavities. Thévenin source characteristics are obtained by repeatedly solving 5 linear equations in order to minimize the difference between measured and expected pressure responses. An error value less than 1 for each loudspeaker indicates a good calibration, less than 0.1 is very good. See Figure 3 for an example of the results of a typical probe source calibration display.

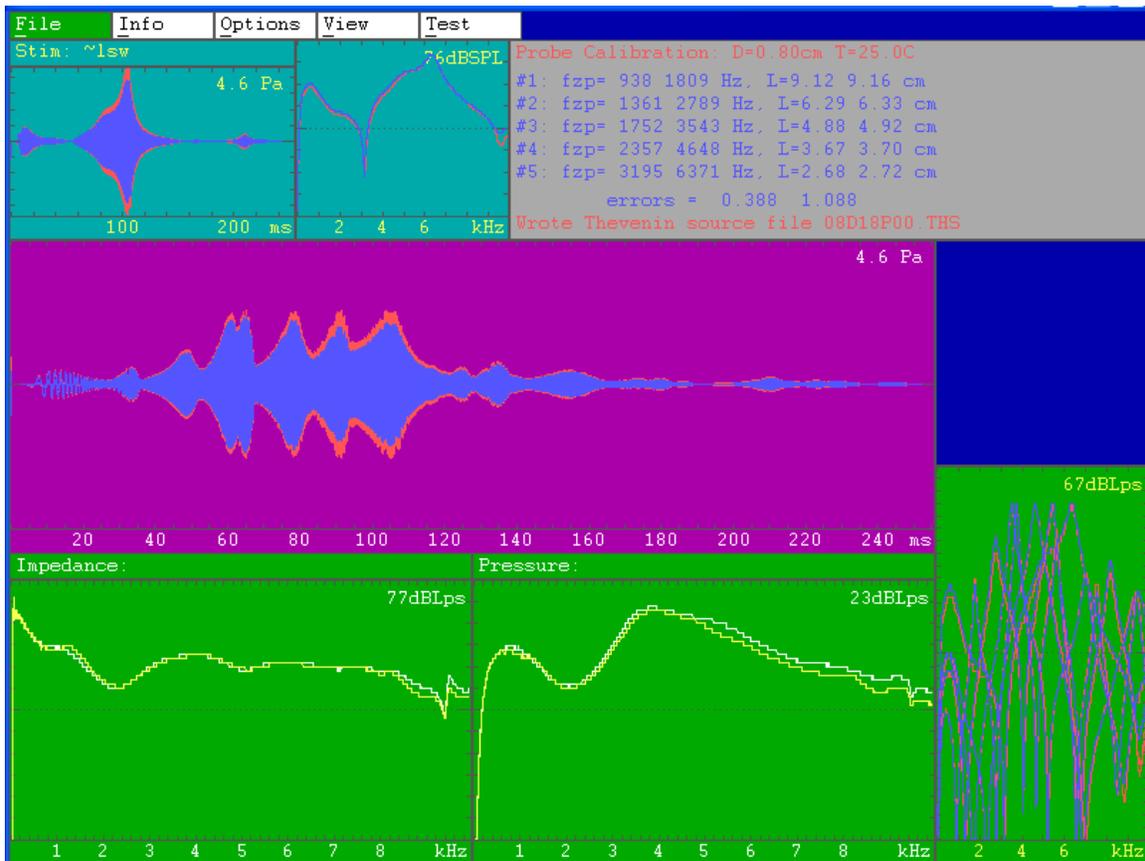


Figure 3. PROBE test calibration display. The calibration pressure waveforms from the last measured cavity are shown in the upper-left panel. As before, the red line is from the left loudspeaker and the blue line is from the right loudspeaker. The spectra of these waveforms are shown in the adjacent panel. The upper right panel reports the frequency of the first notch in the spectrum for each of the two sound sources, followed by the estimated tube lengths, which are calculated from the frequency of the first notch. Results for each cavity are on a separate line. The error term associated with the estimated Thévenin-equivalent source pressure and impedance is reported for each sound source. An error value less than one indicates a good calibration, less than 0.1 is very good. The .THS file named at the bottom of the panel contains the Thévenin-equivalent source impedance and pressure for the two sound sources. The middle panel shows (superimposed) the calibration waveforms for each of the 5 cavities. The bottom-left panel shows the estimated source impedance magnitude for the two sound sources. The bottom-middle panel shows the estimated source pressure for the two sound sources. When the impedance and pressure curves are not as smooth as those shown here, rerunning the PROBE calibration should be considered. The bottom-right panel is the measured spectra of all the cavities in SPL superimposed.

Load impedance, if required, is determined during the in-the-ear calibration phase of the DPOAE test with the probe assembly in the subject's ear canal. Following the check-fit phase, load pressure is measured. Ear canal pressure combined with previously-determined Thévenin source characteristics provides the necessary information to calculate load impedance. Load impedance is required to convert measured pressure in the ear canal to SIL and FPL. See Figure 4 for an example of an FPL/SIL calibration display.

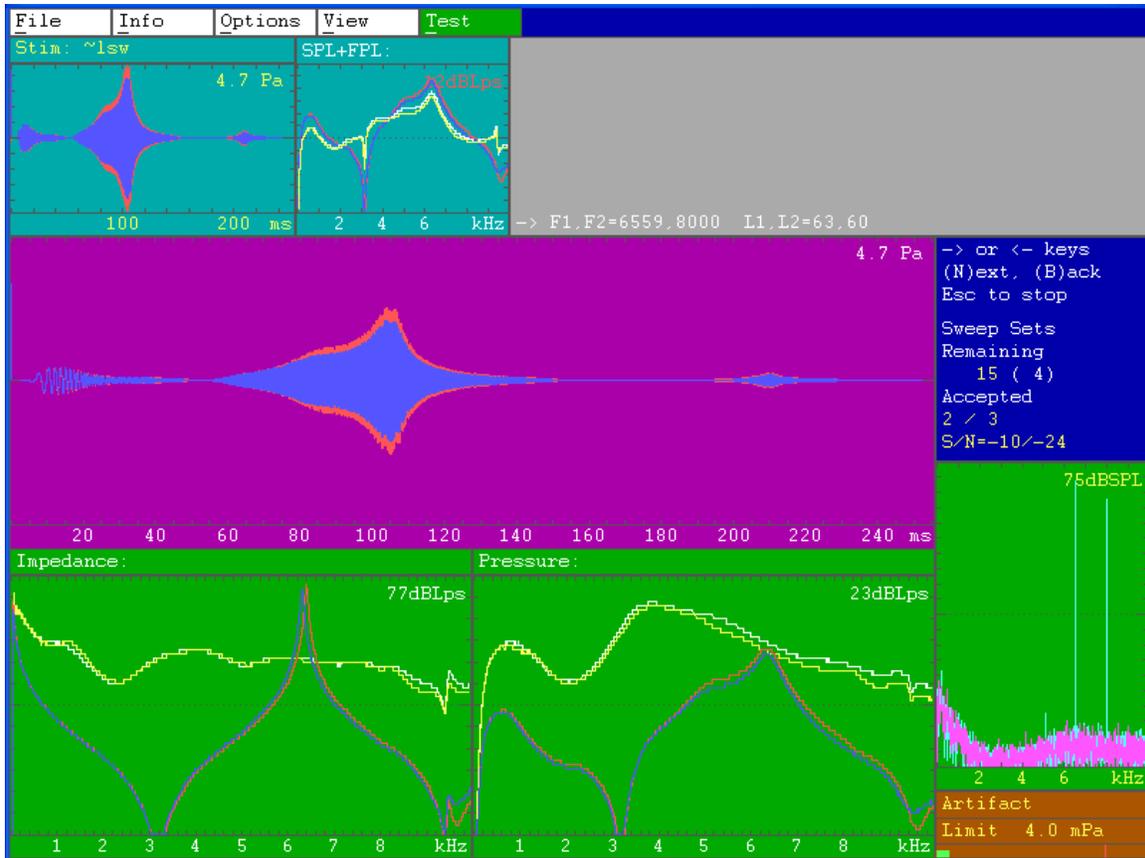


Figure 4. FPL/SIL calibration display. This display is produced after in-the-ear (ITE) calibration during the DPOAE test when stimulus level is specified as SIL or FPL. The conversion from SPL to SIL or FPL requires Thévenin-equivalent source values, which are stored in a .THS file after performing the calibration phase of the PROBE test. The top left panel shows the pressure waveform as measured during ITE calibration. The red and blue lines represent the left and right loudspeakers. The red and blue lines in the adjacent panel show the spectra of these pressure waveforms calibrated in SPL (per unit volt). The yellow and white lines represent the same calibration spectra converted to FPL or SIL (per unit volt). EMVA uses the FPL and SIL calibration spectra (yellow and white lines) to produce stimulus levels for the DPOAE test that are calibrated to these units instead of SPL. The line of text above these spectra indicates whether EMVA will use SIL or FPL calibration. The middle panel contains the same pressure waveforms shown in the upper-left panel. The bottom-left panel shows the magnitude (in dB re 1 cgs acoustic ohm) of the estimated internal source impedance for the two sound sources (yellow and white) with the load impedance of the ear canal (red and blue) superimposed. The bottom-middle panel shows source pressure (yellow and white) for the sound sources and load (in the ear) pressure measured by the sound sources (red and blue). The bottom-right panel is the response window. It shows a spectral display of a running average of the stimuli and response components of ear canal pressure. The A+B average is light blue and the A-B (noise) is in pink. The middle left display contains information regarding the current stimulus condition. The number in parenthesis represents the number of (one-quarter second) sweeps in each sub-average. The upper-right panel contains information regarding the most recently completed condition (in blue) and information regarding the current condition (in white at the bottom of the panel).

3. TESTS

EMAV tests perform synchronous averaging and require access to a soundcard that has the capability to synchronize input and output consistently. EMAV can also retrieve and display the results of previously saved OAE measurements using the File/Open function. This capability does not require a special soundcard; however, only files in the current working directory can be opened.

When displaying the results of a previous measurement, the patient/subject information can be modified, if desired, and the modifications saved to the file using the File/Save function. A hard copy of the EMAV screen can be obtained at any time by pressing Ctrl-P or by using the File/Print function. When printing, the output can be directed to a PostScript file or printed directly to a PostScript (or HP PCL) printer. The printer name and print format are specified in the File/Printer Setup menu.

There are many options that allow variations in the way the OAE tests are performed. For example, the abscissa of the DPOAE display may be changed to L1 or L2 level, to F1, F2 frequency, or to "Trial" to allow successive trials of any stimulus conditions to be displayed sequentially from left to right. EMAV menus are described in more detail in the **Menus** section below.

An EMAV configuration file (emav.ini) allows setting defaults for most of the options that appear on menus and for other options not on the menus. A complete description of the configuration file parameters appears in the **Configuration File** section.

3.1 DPOAE

The DPOAE test is used to measure distortion-product otoacoustic emissions. The probe (i.e., the sound source and microphone assembly) should be placed in the ear prior to starting the DPOAE test. When the test is started, the *check-fit* phase is immediately initiated. During this phase, a calibration stimulus (either a chirp or white noise) is presented to the ear. The measured response to this stimulus is displayed both as a waveform and a spectrum. The fit of the probe in the ear canal should be adjusted to obtain the flattest possible spectrum. A proper fit sometimes requires checking to make sure the holes in the probe tip are unobstructed and that the probe is inserted as deeply as possible in the ear canal.³

When the fit of the probe is satisfactory, press **Enter** to proceed to the *calibration* phase of the test⁴. During this phase, the system response of each channel is measured and displayed on the screen. The responses of the two channels should be similar to each other. If you are satisfied that the calibration was successful, press **Y** or click on Yes to proceed to the measurement phase of the test. Otherwise, you can press **N** or click on No to repeat the check-fit and calibration.

When the calibration waveform has been accepted, EMAV will either (1) begin immediately with the first stimulus condition, (2) present a list of file names for selection, or (3) present a menu of parameters for generation of a new list file, depending on how the list parameter is set (via the configuration file or the Options menu). The stimulus conditions for the DPOAE test are always obtained from a list file. After the list file has been specified, the presentation of stimuli begins. If you have only a single set of stimulus conditions that will be used for every DPOAE test, then the list file that specifies these conditions should be specified before starting the test. If you have several sets of conditions, then the list specification should match all of these files. If you need to generate a new list file, you can do this prior to starting the DPOAE test using the "Create List File" option on the File menu or press **Escape** when presented with a list of filenames during the DPOAE test.

Some aspects of the DPOAE test can still be modified after the presentation of stimuli has begun. For example, pressing the left or right cursor will adjust the threshold for artifact rejection. The artifact rejection threshold is displayed in the lower-right corner of the screen. Pressing **F2** allows recalibration of the stimulus in the middle of the test. Pressing **F9** will suspend the test temporarily. Pressing **Escape** will terminate the test early without discarding any recorded measurements. Pressing **B** will terminate the current stimulus condition and return to the previous stimulus condition. Pressing letter **N** will terminate the current stimulus condition early and proceed to the next stimulus condition.

³ When compressing the foam tip prior to insertion, pulling the foam away from the open end may improve acoustic measurements by creating a flatter surface at the plane of the probe.

⁴ If a calibration has already been performed as part of a previous DPOAE test, then it is possible to skip the calibration phase and proceed directly to the test phase by pressing the letter **T**.

At the end of the test you will be asked whether the recorded data should be saved in a file. Press **Y** or click on Yes to save this test and **N** or No if not. Even if you choose **N**, the test results can still be saved later, as long as these results are still displayed, by selecting the File/Save function.

3.2 TEOAE

The TEOAE test is used to measure transient-evoked otoacoustic emissions in human ears. The probe (i.e., the sound source and microphone assembly) should be placed in the ear prior to starting the TEOAE test. When the test is started, the check-fit phase is immediately initiated. During this phase, a transient stimulus is presented to the ear and the measured response is displayed both as a waveform and a spectrum. The fit of the probe in the ear canal should be adjusted to obtain the flattest possible spectrum. A proper fit sometimes requires checking to make sure the holes in the probe tip are unobstructed and that the probe is inserted as deeply as possible in the ear canal.

When the fit of the probe is satisfactory, press **Enter** to proceed to the calibration phase of the test. During this phase, the program will automatically adjust the output level to achieve a specified “target” level. Further correction of the stimulus waveform in order to reduce the nonlinear stimulus artifact can be initiated by pressing **R**. Both the level adjustment and artifact correction are optional. Other aspects of the stimulus presentation (e.g. number of sweeps and artifact rejection threshold) can be modified during the calibration phase. When the calibration adjustments have been completed, press **Enter** to proceed with the measurement phase of the test.

After the measurement phase of the test has begun, it is still possible to control some aspects of the test. Pressing the left or right cursor will adjust the threshold for artifact rejection. This is displayed in the lower-right corner of the screen. Pressing **F9** will suspend the test temporarily. Pressing **Escape** will terminate the test early without discarding any recorded measurements.

At the end of the test you will be asked whether the recorded data should be saved in a file. Press **Y** or click Yes if you want to save this test and **N** or No if not. Even if you choose **N** you can still save the test later by using the File/Save function, as long as the recorded results are still displayed.

3.3 PROBE

The PROBE test is used to calibrate the sound source for FPL or SIL measurements. Estimation of Thévenin-equivalent source characteristics requires measurements in a set of cylindrical cavities. The default number of cavities is five. Prior to initiating the Probe test, the probe (i.e., the sound source and microphone assembly) should be placed in the cavity-set coupler. See Appendix E for instructions regarding construction of the set of cavities. EMAV repeats the check-fit and calibration phase for each of the cavities. After the required number of calibration waveforms have been accepted, EMAV estimates source pressure and source impedance based on the cavity measurements, iterating if necessary. The error associated with the estimation of source characteristics is displayed on the screen. An error value less than one indicates a good calibration.

EMAV uses the acoustic impedance of the ear canal to produce the SIL or FPL that has been specified in the stimulus protocol (or list) file. EMAV reads the desired SIL or FPL in the list file, measures the SPL in the ear canal, and applies the FPL/SIL conversion to determine the required voltage to the loudspeaker to produce the desired stimulus level. There are four steps involved in calibrating using FPL/SIL.

1. The internal impedance and pressure of the sound source is determined.
2. The impedance of the ear canal (load) is determined from an SPL measurement of the ear canal combined with the source impedance and pressure.
3. The load impedance is used to convert SPL to FPL or SIL.
4. The FPL or SIL calibration is used to determine the voltage presented to the loudspeakers during DPOAE measurement.

These steps are described in greater detail below.

Step 1. Determine source impedance and pressure. The source impedance and source pressure are estimated from pressure measurements in a set of cylindrical cavities with known impedance. In practice, the theoretical impedance of a cylindrical cavity is so sensitive to the length of the cavity that cavity lengths are initially assumed to be only approximate and these lengths are iterated to improve agreement with the measurements. The theoretical impedance of a cavity may be calculated when the length, diameter and temperature of the cavity are known. The initial cavity lengths are calculated from the frequency of first spectral notch. The cavity length is assumed to be one-fourth of the wavelength of the notch frequency. $L=(c/f)/4$, where c is the speed of sound. The theoretical cavity impedances and measured pressures allow a least-squares estimate of source impedance and pressure. The error in

this estimate indicates how much the theoretical description of the sound source and cavity impedance deviates from the combined set of measured pressures. By repeating the procedure using small changes in cavity lengths, the error term can be minimized. The optimum source impedance and pressure are associated with the smallest error. The cavity chirp-response pressure waveforms are stored in a .PRB file. The source impedance and pressure are stored in a .THS file.

Step 2. Determine load impedance (or ear-canal acoustic impedance). Load impedance is calculated from (1) load pressure (ITE calibration), (2) source impedance, and (3) source pressure. Source impedance and source pressure are found in the .THS file created at the conclusion of Step 1. The ear-canal chirp-response pressure waveforms (for each sound source) are stored in a .CAL file. The load impedance is stored in a .THL file.

Step 3. Load impedance is used to convert load pressure (SPL) to SIL or FPL. The SIL conversion is based on the real part of the load admittance, which is called conductance. The FPL conversion is proportional to sum of the load impedance and its characteristic (or surge) impedance.

Step 4. EMAV uses the load impedance to determine the desired SIL or FPL in the ear-canal. EMAV reads each stimulus level in the list file and adjusts the voltage to the speaker to produce the desired level in SPL, SIL, or FPL. The stimulus-level unit is specified in the header of the list file.

The files produced by the SIL/FPL calibration procedure are listed below. Unique files names (*) are generated for .PRB, .THS, and .THL file. The .txt files are temporary, are overwritten each time they are produced, and may be deleted whenever desired.

File name	When	Contents
*.PRB	PROBE test	Waveforms in dB SPL for all cavities and both channels.
*.THS	PROBE test	Thévenin-equivalent source impedance and pressure for both channels in 4 Hz steps.
Zcav0.txt	PROBE test	Final length estimate and internal impedance and phase for each cavity for each sound source.
Zcav1.txt	PROBE test	Alternate impedance for programmer only.
Zcav2.txt	PROBE test	Alternate impedance for programmer only.
THS1.txt	PROBE test	.THS file in .txt format-used for plotting-channel 1.
THS2.txt	PROBE test	.THS file in .txt format-used for plotting-channel 2.
*.THL	DPOAE test when level unit=FPL or SIL	Created from .CAL and .THS files and used to convert SPL to FPL or SIL. This file can be recreated from the .CAL and .THS files at any time.

Reflectance: When the PROBE option is used to calculate FPL or SIL, middle-ear reflectance is calculated and reflectance values, in dB, are reported in the .DAT file for F1, F2, and F3.

3.4 CAVITY

The CAVITY test makes the same load-pressure measurement as the PROBE test, but only in a single cavity or other acoustic load. This test may be used to measure acoustic impedance or reflectance in an ear canal.

3.5 TONE

The TONE test is used more for testing the hardware than for testing ears. Presentation of the tone stimulus is initiated by selecting Start Tone. The level and duration of the tone can be specified. Tones can be output on either one of the two channels (A or B) of the soundcard. You can tell EMAV whether or not to remove the “DC” component from the response waveform. You can select whether the tone is presented once or repeated. The recorded response to the tone is displayed along with its calculated P-P and RMS level.

4. MENUS

There are five major-function items listed on the menu bar at the top of the EMVA screen: File, Info, Options, View, and Test. “View” puts EMVA in a special mode for examining the results of an OAE measurement in more detail. The other four major functions produce submenus of additional items.

Most of the EMVA functions are menu-driven. Choose the desired item on the menu bar at the top of the screen by either pressing the designated “short-cut” key or clicking on the menu item with the mouse cursor. The short-cut key is indicated on each menu item as an underlined letter. For example, press **I** to enter patient/subject information. Alternative short-cut keys, if any, appear in parentheses at the right-hand side of a menu item. Within submenus, an ellipsis “...” at the end indicates that another submenu will be displayed, a colon “:” indicates that a text field will be opened for editing, and an equals sign “=” indicates that a multiple-choice item will be “toggled” to the next value. Items within a menu may be selected by using the Up/Down cursor keys and pressing **Enter**. Multiple-choice items are altered by pressing **Enter** to “toggle” their value. Text fields are altered by pressing **Enter** to open the menu item for editing. When editing of a text field is complete, you must press **Enter** again to close and accept the edited text field. Use **Escape** to close and discard any changes to the text. Use **Escape** also to remove any subset of menu items from the display. To initiate an EMVA test, press **T** to open the “Test” menu, then press **D**, for example, to start the DPOAE test or press **T** to begin the TEOAE test. These two OAE tests both consist of three phases: (1) check-fit, (2) in-the-ear calibration, and (3) OAE measurement. If you want to measure DPOAEs in FPL (Forward pressure level) or SIL (sound intensity level) rather than SPL, you must determine the Thévenin characteristics of the probe first using the PROBE option.

4.1 FILE MENU

The File menu lists items that perform functions related to data files, screen printing, and program control.

Open... Allows existing data files to be opened for review and limited editing, for example of patient/subject information. A submenu is presented with three items.

Open File - causes the file specified in the File Name field to be opened, or if the file name contains a wild-card character⁵ it presents a list of matching files for selection.

File Type = toggles between DPOAE, DPSWP, DPCAL, PROBE, and TEOAE. Each file type has a different default filename that matches any file name with the appropriate extension (.DAT, .SWP, .CAL, .PRB, or .TOK).

File Name : specifies the file to be opened or pattern (with one or more wild card characters) of file names to be listed for selection.

New - Erases contents of screen, resets all program variables to their initial state, and re-reads the configuration file (emav.ini) containing default parameter values.

Print Screen - Causes the contents of the screen to be printed as specified in the Printer Setup menu. This command can also be invoked by pressing Ctrl-P (or Ctrl-W).

Printer Setup... Specifies the manner in which the screen is printed. A submenu is presented with three items.

Printer Port : specifies the port (e.g. LPT1) of a directly connected printer, a file to which the printer output will be directed, or the network name of a printer on the local area network (e.g., \\cellinux\lp).

Printer Type = the printer type will toggle between PostScript, Color PostScript, and PCL. The Color PostScript option is probably the best choice when printing to a file, because it can be inserted into Word documents. The PCL printer type is usable with many Hewlett-Packard (DeskJet and LaserJet) printers.

Orientation = the printer orientation will toggle between Portrait and Landscape. The Portrait option produces a smaller screen image at the top of the page. The Landscape option produces a larger, rotated screen image.

⁵ An asterisk “*” matches any number of any characters in a filename, while a question mark “?” matches only a single instance of any character.

File Save - Allows an open file to be written again to the same file name to save any changes made to the patient/subject information. When no file is currently open, the File Save function is disabled.

Create List File - Provides a menu of stimulus parameters to assist with the creation of new stimulus protocol (i.e., list) files for the DPOAE test.

Create Load File - Allows creation of a new Thévenin “load” file (THL) from two existing files: a “calibration” file (CAL) and a “probe” file (PRB).

Extract SFOAE - Interprets a previously opened DPOAE data file (DAT) as containing responses to pairs of stimulus conditions: with a suppressor and without a suppressor. A new DAT file is produced that contains the response with suppressor subtracted from the response without suppressor.

About EMAV - Displays information about the program version and authors of the EMAV program. The currently selected soundcard is also displayed. This function may also be invoked by pressing the F1 function key.

Exit - Terminates the EMAV program. This function can also be invoked by pressing Ctrl-C.

4.2 INFO MENU

The Info menu allows the user to enter the patient/subject information that will be stored in the data file. The patient/subject information in existing data files may be modified by opening the file, editing the information, and then saving the file using the File/Save function. There are seven items on the Info menu: Last Name, First Name, Ear (Left or Right), ID (identification number), Date of Birth, Threshold and Comment.

4.3 OPTIONS MENU

The Options menu contains parameters that affect program operation and test measurement to be viewed and modified. Any changes made in the Options menu will not be retained once EMAV is closed. When EMAV is reopened the parameters will have returned to the default settings specified in the EMAV configuration file (emav.ini). Default settings can be modified in the configuration file itself. There are five items on the Options menu: System, DPOAE, TEOAE, PROBE, and TONE.

System.... Parameters that affect all EMAV measurements.

Display Units (mV or mPa) = specifies whether information is displayed during the tests in units of sound pressure (mPa) or voltage (mV).

I/O Sensitivity... - Displays seven input/output sensitivity parameters:

AD_sensitivity (cnt/V) : the sensitivity of the A/D converter in count per volt.

DA1_sensitivity (cnt/V) : the sensitivity of the first (or Left) D/A converter in count per volt.

DA2_sensitivity (cnt/V) : the sensitivity of the second (or Right) D/A converter in count per volt.

MP_transfer : The name of a file that specifies a frequency-dependent microphone sensitivity. The MP_transfer file may be a text file with one frequency and sensitivity (V/Pa) pair on each line. In this case, the MP_sensitivity parameter is still used to scale waveforms, but not frequency components levels. Alternatively, the MP_transfer file may be a MATLAB-compatible SYSRES data file. In this case, the transfer function contained in the MAT file is multiplied by the MP_sensitivity parameter.

MP_sensitivity (V/Pa) : sensitivity of the microphone in volts per pascal, when it is assumed to be constant for all frequencies. The value of this parameter for the ER-10C is 0.05, 0.5, or 5 depending on whether the GAIN switch is set to 0, +20, or +40 dB.

LS1_sensitivity (V/Pa): sensitivity of the first (Left) receiver (loudspeaker) in volts per pascal.

LS2_sensitivity (V/Pa): sensitivity of the second (Right) receiver (loudspeaker) in volts per pascal.

DPOAE... Parameters that affect the DPOAE test.

Buffer Size : the number of samples in the DPOAE stimulus. The default value is set by the parameter “size” in the configuration file.

Rate of Clock (Hz) : the A/D and D/A sampling rates (samples/second).

Sweeps per Set : This parameter specifies the number of sweeps in each set for the measurement phase of the DPOAE test. For efficiency, the total number of sweeps averaged for each test is divided into sets with a few sweeps in each set.

DP Frequency = the distortion-product frequency to be measured. The value toggles among $2 * F1 - F2$, $3 * F1 - 2 * F2$, $4 * F1 - 3 * F2$, $2 * F2 - F1$, and $F2 - F1$.

Input Mode = controls the reading and writing of sweep (SWP) files. The value toggles among REAL TIME, RECORD, and PLAYBACK. REAL TIME is the normal A/D Input Mode with no sweep file involvement. RECORD mode reads input from the A/D and saves each subset of recorded sweeps to a sweep file. PLAYBACK reads input from a sweep file instead of from the A/D. A sweep file must be opened (using the File/Open function) to allow PLAYBACK mode.

Stimuli... parameters related to the DPOAE stimulus

Checkfit stimulus : the name of the stimulus file to be used during the check-fit phase of the DPOAE test. Special names indicate stimuli generated by EMAV: “~lsw” produces a chirp stimulus and “~bbn” produces a white-noise stimulus.

Calibration stimulus : The name of the stimulus file to be used during the calibrate phase of the DPOAE test. Special names indicate stimuli generated by EMAV: “~lsw” produces a chirp stimulus and “~bbn” produces a white-noise stimulus.

Calibration attenuation : The (internal) attenuation (dB) applied to the stimulus during the calibrate phase of the DPOAE test. The level of the calibration stimulus will be this number of decibels below the maximum possible output.

Suppressor : an additional stimulus that is added to the second (Right) channel. Possible values are: None for no suppressor, Tone for a tonal suppressor consisting of one or two tones, BPN for band-pass noise, and XBPN for band pass noise that excludes DP frequencies. Use F3, F4, L3, and L4 in the list file to specify the frequency and level of the two tones or the edges of the band-pass noise.

Adjust F1 : an option to allow the user to vary the level and phase of the first primary tone during the DPOAE test. Possible values are Yes and No.

Ramp Time (ms) : The duration (in ms) of a linear ramp applied to the beginning and end of all tones used in the DPOAE test. Because the ramp must be applied prior to response averaging, this feature is not enabled unless the number of skipped sweeps (skips) is greater than zero.

CheckFit Time (s) : When this parameter is non-zero, the check-fit portion of a test will time-out after the specified number of seconds and the subsequent in-the-ear calibration will be accepted. This feature is useful when performing self-tests when the probe and computer are not in the same place.

Protocol File : The name of the parameter list file that specifies the stimulus conditions to be presented during the measurement phase of the DPOAE test. If the file name contains a wild card character (such as “*”), then a list of all matching file names will be presented for selection during the DPOAE test.

Level Unit : Tells how EMAV should interpret the levels specifies in list files. The choices are SPL (sound pressure level in dB re 20 micropascal), SL (sensation level dB re threshold), V (volts), FPL (forward pressure level in dB re 20 micropascal), and SIL (sound intensity level in dB re 1 picowatt/meter²).

Display... parameters related to the DPOAE measurement display

Maximum Ordinate Level (dB) : The maximum level (in dB SPL) on the y-axis of the measurement display.

Minimum Ordinate Level (dB) : The minimum level (in dB SPL) on the y-axis of the measurement display.

Maximum Abscissa Level (dB) : The maximum level (in dB SPL) on the x-axis of the measurement display when the abscissa represents levels.

Minimum Abscissa Level (dB) : The minimum level (in dB SPL) on the x-axis of the measurement display when the abscissa represents levels..

Maximum Frequency (oct) : The maximum frequency on the x-axis of the measurement display when the abscissa represents frequencies. The frequency is expressed as the number of octaves relative to 1 kHz.

Minimum Frequency (oct) : The minimum frequency on the x-axis of the measurement display when the abscissa represents frequencies. The frequency is expressed as the number of octaves relative to 1 kHz.

Spectrum... parameters related to the DPOAE spectral display (Response Window)

Spec. Freq. Range Type : the range of the frequencies can be set to FIXED or AUTO. When the range is set to AUTO the maximum frequency on the x-axis of the spectrum varies with the F2 frequency.

Spec. Frequency Range (kHz) : the maximum frequency (kHz) on the x-axis of the spectrum when it is FIXED.

Spec. Level Range (dB) : the range of levels (dB) along the y-axis of the spectrum. The range is relative to the highest stimulus level. If this range is too small, the baseline of the spectrum will not be visible.

Data File... parameters related to the DPOAE data file

Data Format Type : The choices are Normal (one distortion frequency only), High Order ($2*F1-F2$, $3*F1-2*F2$, $4*F1-3*F2$, $5*F1-4*F2$), Extended ($2*F1-F2$, $3*F1-2*F2$, $4*F1-3*F2$, $2*F2-F1$), Multi (Extended plus $2*F1-F3$, $2*F3-F1$, $2*F1-F4$, $2*F4-F1$), or SFOAE ($F2-FM$, $F2+FM$, $F1-FM$, $F1+FM$). Only the first character (N, H, E, M, or S) is checked by the program.

Save Response Binary : Specifies whether the response waveforms should be written to a binary data file (Y or N).

DP Octave : Number of octaves of the primary frequencies (F1 & F2) that are included in the stimulus. The value can be 0, 1, or 2. This feature is disabled when suppressor tones are included in the stimulus.

High-Pass Filter... parameters related to the DPOAE high-pass filter. This filter is applied only to the noise measurement used for artifact rejection.

HPF Frequency (Hz) : cutoff frequency (Hz) of the high-pass filter applied to the difference waveform used to determine artifact rejection when HPF_type=0 (fixed)

HPF Type = Selects the type of high-pass filter applied to the difference waveform used to determine artifact rejection. The choices are 0=fixed, 1=auto, and 2=off. The “fixed” setting tells EMAV to always use the cutoff frequency specified by HPF_freq. The “auto” setting tells EMAV to set the cutoff frequency one octave below F2. The “off” setting tells EMAV not to apply a high-pass filter.

Noise... parameters related to the noise level estimate during the measurement phase.

NNSB : The “number of noise side-bands”. This parameter specifies the number of adjacent-frequency components on *each* side of the DP frequency that will be included in the noise level estimate during the measurement phase of the DPOAE test. When the value is positive, the two partial averages (A & B) are added and the noise level is estimated from the combined spectrum (A+B). When the value is zero or negative, the noise level is estimated from the difference spectrum (A-B). When the value is 0, the default value, the noise level estimate is based on only the (A-B) spectral component at the DP frequency.

NFSB : number of noise sidebands included in the detection of noise-floor separation.

NF_separation (dB) : threshold (in dB) for rejecting sub-averages due to noise floor separation.

Reduction Mode = selects an experimental method for noise reduction. Its value toggles among DP+ (no noise reduction), DPnr1 (method 1, removes mean value), and DPnr2 (method 2, removes trend).

Reduction Threshold (dB) : apply noise reduction only when the noise estimate changes by more than this threshold.

TEOAE... parameters that affect the TEOAE test.

Buffer Size : Number of samples in the TEOAE stimulus. The number of samples in each sweep may be up to four times this number, depending on the stimulus presentation mode.

Rate of Clock (Hz) : the A/D and D/A sample rates (samples/second).

Sweeps per Set : number of sweeps in each set for the measurement phase of the TEOAE test. This parameter is called “sweeps” in the configuration file. For efficiency, the total number of sweeps averaged for each test is divided into sets with a few sweeps in each set.

Number of Sweep Sets : maximum number of sweep sets to be measured.

Stimulus presentation mode = stimulus sequence presented for each sweep. Possible values are {+1}, {+1 -1}, 2={+1 +1 -2}, 3={+1 +1 +1 -3}, 4={L R B}, and 5={L R L R B*2}. Numeric patterns are generated only on the first (or left) D/A channel and the number refers to the relative amplitude of each section of the waveform. The last two patterns alternate between the two channels (L &R) until the final section which is applied to both channels (B).

Stimulus File : the name of the stimulus file used for the measurement phase of the TEOAE test. The default name “~ssw” tells EMAV to generate a short-sweep stimulus.

Checkfit Stimulus : the name of the stimulus file used during the check-fit phase of the TEOAE test. When the name is “~ssw”, EMAV generates a short-sweep stimulus.

Artifact Rejection and Response Filter... parameters related to artifact rejection and response filter.

Hi Pass Filter Frequency (Hz) : The cutoff frequency (Hz) of a high-pass, 12-dB/octave filter applied to the response waveform to improve the signal to noise ratio. The signal pass-band is between F1 and F2.

Lo Pass Filter Frequency (Hz) : The cutoff frequency (Hz) of a low-pass, 12-dB/octave filter applied to response waveform to improve signal to noise ratio. The signal pass-band is between F1 and F2.

Rejection Threshold (mPa) : Initial threshold (in millipascal) for artifact rejection during the measurement phase of the TEOAE test. The threshold value may be modified during the measurement phase by using the left or right cursor keys

Start Time (ms) : The beginning of the response (or echo) interval of the recorded waveform specified in milliseconds from the onset of the stimulus. The recorded waveform is ramped on 1 msec prior to Time1 and is set to zero prior to this 1 msec ramp.

End Time (ms) : The end of the response (or echo) interval of the recorded waveform specified in milliseconds from the onset of the stimulus. The recorded waveform is ramped off at Time2 and is set to zero after this 1 msec ramp.

Ramp Time (ms) : duration of a linear ramp at the beginning and end of the window used to select the TEOAE from the measured waveform.

Spectrum Frequency Range : Horizontal full-scale range (in kHz) of the TEOAE spectrum display..

Spectrum Level Range : Vertical full-scale range (in dB) of the TEOAE spectrum display.

PROBE... parameters that affect the PROBE test. For consistency, the first two items share internal variables with the DPOAE test.

Buffer Size : the number of samples in the DPOAE stimulus.

Rate of Clock (Hz) : the A/D and D/A sampling rates (samples/second).

Spectrum Frequency Range : Horizontal full-scale range (in kHz) of the DPOAE spectrum display.

Spectrum Level Range : Vertical full-scale range (in dB) of the DPOAE spectrum display.

Calibration... parameters that describe the calibration procedure. For consistency, the first four items share internal variables with the DPOAE test.

Sweeps per Set : number of sweeps in each set for the measurement phase of the DPOAE test.

Number of Sweep Sets : maximum number of sweep sets to be measured.

Stimulus File: The name of the stimulus file to be used during the calibrate phase of the DPOAE test. Special names indicate stimuli generated by EMAV: “~lsw” produces a chirp stimulus and “~bbn” produces a white-noise stimulus.

Attenuation : Number of db that the calibration stimulus is attenuated relative to maximum output.

Number of cavities to test : Number of test cavities used during the probe calibration phase.

Number of sources to test : Number of sound sources. Setting nsrc=1 changes the PROBE test to allow load calibration in addition to source calibration.

Thévenin Computation... parameters affecting computation of Thévenin source characteristics.

Compute Crosstalk = Yes or No. Modifies the equations solved for source characteristics to include an additional variable representing crosstalk.

Iterate Length = Tells EMAV whether to perform iterative improvement of its initial estimates of cavity lengths. The default value is “Y: for “yes”. Alternate values are “N” for “no” and “M” for “maybe,” which prompts the user for a yes or no decision after the initial cavity length estimates are displayed.

Iterate T & D = Yes or No. Tells EMAV whether to include cavity diameter and temperature when iterating. The default value is N for no. The alternate value is Y for yes.

Smooth = Setting “smooth” to a number greater than zero tells EMAV to smooth its estimates of Thévenin source characteristics across that number of adjacent frequencies.

f1_erf : Specifies the lower frequency limit of the range of frequencies over which EMAV calculates an error value based on deviation of estimated source characteristics from ideal values.

f2_erf : Specifies the upper frequency limit of the range of frequencies over which EMAV calculates an error value based on deviation of estimated source characteristics from ideal values.

Temperature : Calibration cavity temperature (°C) used in the calculation of ideal cavity impedance. Default value is 25° (Room Temperature). If cavities are measured at body temperature, 37° should be used.

Diameter : Internal cavity diameter (in centimeters) used in the calculation of ideal cavity impedance.

Min. Res. : If the estimated load impedance has a real part less than this value (cgs acoustic ohms), then it will be set equal to this value.

Z_cav = 0 or 1. Selects between different methods of calculating the theoretical-ideal cavity impedance. The default value is 0.

Length Decrease = Yes or No. For initial estimates of cavity length, these lengths may be assumed to be in decreasing order.

TONE... parameters that affect the TONE test.

Buffer Size : the number of samples in the stimulus buffer.

Rate of Clock (Hz) : the A/D and D/A sample rates (samples/second).

Ramp Time (ms) : the duration (in msec) of a linear ramp applied to all tones. No ramp is applied unless the number of skipped sweeps is greater than zero.

Skipped Sweeps : number of sweeps skipped before response averaging begins. Skipping sweeps avoids transient responses due to stimulus onset.

Spectrum Frequency Range (kHz) : maximum frequency on the x-axis of the spectrum.

Spectrum Level Range (dB) : range of levels along the y-axis of the spectrum.

4.5 TEST MENU

There are four tests (DPOAE, TEOAE, PROBE, and TONE) on the test menu. Pressing the designated letter (**D**, **T**, **P** or **E**) causes the test to begin.

5. Function Keys

5.1 ABOUT DIALOG

Pressing **F1** is a shortcut for displaying the About dialog. Besides showing version number for EMAN and the ARSC function library, the About dialog also shows the name of the soundcard that has been selected.

5.2 DEVICE LIST DIALOG

Pressing **F2** displays a list of available devices. The selected soundcard is indicated by a ">" symbol. While the Device List is displayed, pressing a number selects the corresponding soundcard on the list. Alternatively, the desired soundcard may be specified as the default by adding a line saying "DSP_CODE=number" to emav.ini.

5.3 VIEW MODE

When results of an OAE test are displayed, pressing V (for View) puts EMAN into a special View mode that allows the test results to be examined in more detail.

When DPOAE results are displayed in View mode, a small, vertical-line cursor appears on the results of a selected stimulus condition. Pressing +, **Tab**, or right clicking the mouse advances the cursor to the next stimulus condition, while -, **Shift-Tab** or left clicking the mouse backs up to the previous stimulus condition. The stimulus and response information in the upper-right panel changes according to the selected condition. If binary data has been saved for the DPOAE results, then the signal and noise spectra that correspond to the selected condition will be displayed. If the binary data has not been saved in the Start In directory, the Response Window will be blank.

When TEOAE results are displayed, pressing **F6** separates the two waveforms. Pressing + expands the waveform and - contracts the waveform along the time axis. When the waveform is expanded, pressing **Tab** advances the display along the time axis and **Shift-Tab** shifts the display in the reverse direction.

For either type of data, pressing **F7** exchanges the position of the measurement results panel and spectral display panel, to allow for closer inspection of the spectra. Pressing **Escape** exits from View mode.

6. FILE TYPES

6.1 CONFIGURATION FILE

The configuration (initialization) file is called "emav.ini" by default and can be located either in (1) the current directory, (2) the directory containing the program (emav.exe), or (3) any directory listed in the PATH environment variable. The configuration file is divided into six sections SYSTEM, PRINTER, TEOAE, DPOAE, PROBE, and TONE. Each section begins with a line containing the name of the section enclosed in brackets. The parameters that can be given values are described below for each section. The default value for each parameter is indicated. Lines that begin with a semicolon in the configuration file are ignored and can be used for comments or to temporarily disable a parameter setting. The letters in the parameter name are not case-sensitive. Many of the parameters may also be set through the Options menu in EMAV. Changing a parameter value within EMAV will not change the default value specified in the configuration file.

Refer to the Parameters table for a listing of parameters that can be included in a configuration file. A majority of these parameters can also be defined through the Options menu within EMAV. The descriptions of these parameters are located in the Menus section of this manual. All remaining configuration file parameters not defined elsewhere are included below.

SYSTEM Parameters

The SYSTEM section begins with a line containing the word SYSTEM in brackets like this: [SYSTEM]. Parameters in this section apply to all tests. These parameters should be set for your soundcard using the instructions given in Appendix C. They are specific to the sound card and the load characteristics of your hardware, therefore separate configuration files may need to be required for different equipment configurations. The following list of SYSTEM parameters includes default values.

AD_type = 0

This parameter is obsolete. Previously, if this parameter were set to 1, the program would skip clock synchronization checks for the Ariel DSP-16+.

DSP = W

Type of soundcard: W=Windows.

DSP_code =

Name of the file containing DSP code that needs to be uploaded to the soundcard.

GREG = 16

This parameter is obsolete. Previously, this parameter modified the allocation of data memory. The current version always allocates 16K words of for data, which allows DPOAE size up to 2048, and TEOAE size up to 1024.

refresh = 5

Number of seconds between automatic screen refreshes. If refresh is set to zero, automatic refresh is disabled.

AD_sensitivity = 0

Sensitivity of the A/D converter in count per volt. When set to zero, the value is calculated from ARSC registry volts-full-scale value.

DA1_sensitivity = 0

Sensitivity of the first (or Left) D/A converter in count per volt. When set to zero, the value is calculated from ARSC registry volts-full-scale value.

DA2_sensitivity = 0

Sensitivity of the second (or Right) D/A converter in count per volt. When set to zero, the value is calculated from ARSC registry volts-full-scale value.

MP_transfer =

Name of a file that specifies a frequency-dependent microphone sensitivity. The MP_transfer file may be a text file with one frequency and sensitivity (V/Pa) pair on each line. In this case, the MP_sensitivity parameter is still used to scale waveforms, but not frequency components levels. Alternatively, the MP_transfer file may be a MATLAB-compatible SYSRES data file. In this case, the transfer function contained in the MAT file is multiplied by the MP_sensitivity parameter.

MP_sensitivity=0.05

Sensitivity of the microphone in volts per pascal, when it is assumed to be constant for all frequencies. The value of this parameter for the ER-10C is 0.05, 0.5, or 5 depending on whether the GAIN switch is set to 0, +20, or +40 dB.

LS1_sensitivity = 5

Sensitivity of the first (Left) receiver (loudspeaker) in volts per pascal.

LS2_sensitivity = 5

Sensitivity of the second (Right) receiver (loudspeaker) in volts per pascal.

PRINTER Parameters

The PRINTER section begins with a line containing the word PRINTER in brackets like this: [PRINTER]. Parameters in this section relate to printing the contents of the screen. The following list of PRINTER parameters includes default values.

Label = Boys Town National Research Hospital

This causes the menu bar to be replaced by the specified label during printing. Replacement can be eliminated by an empty parameter setting "Label=".

Orient = Landscape

Printing can be formatted in one of two possible orientations: Landscape or Portrait.

Port = screen.ps

The printer port, for a directly connected printer, may be specified by name, such as LPT1. Network printers may be specified by their network name (e.g., \\cellinux\lp). Instead of a port name or network printer name, this parameter may be set to a file name and screen images will be written to the specified file.

Type = PostScript

You can choose one of three printer types: PostScript, Color PostScript, or PCL. The PostScript output is suitable for including in other documents as an Encapsulated PostScript file. The PCL output can be used with HP LaserJet or HP DeskJet printers.

TEOAE Parameters

The TEOAE section begins with a line containing the word TEOAE in brackets like this: [TEOAE]. Parameters in this section relate to the TEOAE test. The following list of TEOAE parameters includes default values.

adjust = M

The type of the attenuator adjustment used during the calibration phase of the TEOAE test: A=auto or M>manual.

chk_atten = 10

Attenuation applied to the stimulus during the check-fit phase of the TEOAE test.

chk_swps = 25

Number of sweeps to be averaged before updating the display during the check-fit phase of the TEOAE test.

HPF1 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 1 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

HPF2 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 2 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

FFTref = 0

Reference level used for the TEOAE spectrum display. A value of 0 specifies a “level per cycle” reference, which is the equivalent SPL level in a 1 Hz bandwidth. A value of 1 species an SPL level *re* 20 micropascal rms.

target = 70

The initial target for the calibration phase of the TEOAE test specified in dB ppSPL or decibels *re* 20 micropascal peak-to-peak.

tolerance = 1

Specifies how close (in decibels) the calibrated stimulus should be to the target level. If the difference between the measured stimulus level and the target level exceeds this tolerance, then a warning message will be displayed and the user will be required to press **Y** to continue.

Trefresh = 0

Number of seconds between automatic screen refresh during the TEOAE measurement phase. A value of zero (the default) turns off the automatic refresh.

DPOAE Parameters

The DPOAE section begins with a line containing the word DPOAE in brackets like this: [DPOAE]. Parameters in this section relate to the DPOAE test.

abscissa = F2

Specifies the stimulus variable used for the abscissa of the data display during the measurement phase of the DPOAE test. The value may be set to F1, F2, Fd, L1, L2, L3, or Trials.

atten = 20

Attenuation (dB) applied to the stimulus during the calibration phase of the DPOAE test.

CalibratePhase = Y

Tells EMAV whether to include phase compensation when calibrating the stimulus. The default value is “Y” for “yes”. The alternate value is “N” for “no.”

cal_swps = 16

The number of sweeps per set averaged during the calibrate phase of the DPOAE test. The program always collects two sets of sweeps from each input channel, so the actual number of sweeps averaged per channel during calibration will be twice the specified number.

calibrate = ~lsw

The stimulus file to be used during the calibration phase of the DPOAE test. Special names indicate stimuli generated by EMAV: “~lsw” produces a chirp stimulus and “~bbn” produces a white-noise stimulus.

chk_atten = 10

Attenuation (dB) applied to the stimulus during the check-fit phase of the DPOAE test.

chk_swps = 4

Number of sweeps averaged before updating the display during the check-fit phase of the DPOAE test.

Count = 0

The starting point when EMAV generates a sequence number for a DAT file name. The default value tells EMAV to generate file names starting with the sequence number 0. Setting count=100 causes EMAV to start generating file names with the sequence number 100, which is encoded in the last two characters of the filename as "AA".

DP_freq = 2*F1-F2

Relation of the distortion frequency to the primary frequencies. Choices are: 2*F1-F2, 3*F1-2*F2, 4*F1-3*F2, 2*F2-F1, and F2-F1.

DataFmt = Normal

Selects the DAT file format. The choices are Normal (one distortion frequency only), High Order (2*F1-F2, 3*F1-2*F2, 4*F1-3*F2, 5*F1-4*F2), Extended (2*F1-F2, 3*F1-2*F2, 4*F1-3*F2, 2*F2-F1), Multi (Extended plus 2*F1-F3, 2*F3-F1, 2*F1-F4, 2*F4-F1), or SFOAE (F2-FM, F2+FM, F1-FM, F1+FM). Only the first character (N, H, E, M, or S) is checked by the program.

FFTDB = 80

The vertical range (in dB) of the DPOAE data display.

FFTkHz = 8

The horizontal range (in kHz) of the DPOAE data display when SpecFreqRange=FIXED.

FFTref = 1

The reference level used for the DPOAE spectrum display. A value of 1 specifies a "level per cycle" reference, which is the equivalent SPL level in a 1 Hz bandwidth. A value of 0 species an SPL level *re* 20 micropascal rms.

HPF1 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 1 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

HPF2 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 2 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

HPF_freq = 200

When HPF_type is FIXED the high-pass cutoff frequency for artifact rejection is specified by HPF_freq (in Hz).

HPF_type = AUTO

Selects the type of high-pass filter applied to the A-B response difference only for the purpose of determining artifact rejection. The choices are: FIXED, AUTO, or OFF. When FIXED the high-pass cutoff frequency is specified by HPF_freq. When AUTO the high-pass cutoff frequency is one octave below F2.

Level_Unit = SPL

Determines the interpretation of stimulus levels in list files. Level_Unit may be set to SPL, SL, V, FPL, or SIL. SPL (dB) indicates stimulus level *re* 20 μ Pa rms. SL (dB) indicates stimulus level relative to subject threshold, which must be specified in the Info menu. V indicates peak voltage. FPL and SIL (dB) indicate forward-pressure level and sound-intensity level, respectively, and require the presence of a THL file.

list = param.lst

The name of the parameter list file that specifies the stimulus conditions to be presented during the measurement phase of the DPOAE test. If the file name contains a wild card character (such as “*”), then a list of all matching file names will be presented for selection during the DPOAE test.

Limit = 2

Initial threshold (in mPa) for artifact rejection during the measurement phase of the DPOAE test. The threshold value may be modified during the measurement phase by using the left or right cursor keys.

MaxFrqOct = 4

Maximum Abscissa (x axis) Frequency (in Octaves re 1 kHz).

MaxLevAbs = 100

Maximum Abscissa (x axis) Level (in dB SPL).

MaxLevOrd = 80

Maximum Ordinate (y axis) Level (in dB SPL).

MinFrqOct = -2

Minimum Abscissa (x axis) Frequency (in Octaves re 1 kHz).

MinLevAbs = 10

Minimum Abscissa (x axis) Level (in dB SPL).

MinLevOrd = -40

Minimum Ordinate (y axis) Level (in dB SPL).

NIC = 1

The “number of input channels” written to BIN files. Setting “**nic=2**” tells EMAV to save the response from the second A/D channel.

nnsb = 1

The “number of noise side-bands” specifies the number of adjacent-frequency components on *each* side of the DP frequency that will be included in the noise level estimate during the measurement phase of the DPOAE test. When the value is positive, the two partial averages (A & B) are added and the noise level is estimated from the combined spectrum (A+B). When the value is zero or negative, the noise level is estimated from the difference spectrum (A-B). When the value is 0, the default value, the noise level estimate is based on only the (A-B) spectral component at the DP frequency.

Octave = 0

Number of octaves above the primary frequencies (F1 & F2) that are included in the stimulus. This allows testing of multiple DPs simultaneously. The value can be 0, 1, or 2. This feature is disabled when suppressor tones are included in the stimulus.

ramp = 10

The duration (in ms) of a linear ramp applied to the beginning and end of all tones used in the DPOAE test. Because the ramp must be applied prior to response averaging, this feature is not enabled unless the number of skipped sweeps (skips) is greater than zero.

rate = 32000

The A/D and D/A sample rates (samples/second) for the DPOAE test.

SaveBin = No

When SaveBin is set to Yes, the DPOAE test will also create a binary file with a .BIN extension that contains the response waveforms for each stimulus condition.

scope = 8192

Allows the waveform display to show less than the entire stimulus interval during the calibrate phase of the DPOAE test.

seed = 68131

Random number generator seed.

sets = 2

For efficiency, the total number of sweeps averaged for each test is divided into sets with a few sweeps in each set. This parameter specifies the number of sets for the measurement phase of the DPOAE test.

signal = DP+

Method used to estimate the signal during the measurement phase of the DPOAE test. The default method (DP+) is to take the sum of two partial averages. Alternate methods (DPnr1 and DPnr2) only apply when an experimental method of noise reduction is activated.

size = 2048

Stimulus and response buffer size (i.e., number of samples per channel in one sweep).

skips = 1

Number of sweeps to skip before averaging to avoid recording the transient response at the beginning the stimulus.

SpecFreqRange = Fixed

The range of the frequencies can be set to FIXED or AUTO. When the range is set to FIXED the maximum frequency is set by FFTkHz. When the range is set to AUTO the maximum frequency on the x-axis of the spectrum varies with the F2 frequency.

sweeps = 32

The number of sweeps per set averaged during the DPOAE test. The program always collects two sets of sweeps from each input channel, so the number of sweeps averaged will always be a multiple of twice the specified number.

thsf =

Thévenin source file name. A Thévenin source file is required for SIL or FPL calibration. Setting "ths_file=*.ths" causes EMVA to prompt for the file name when it is needed.

PROBE Parameters

The PROBE section begins with a line containing the word PROBE in brackets like this: [PROBE]. Parameters in this section relate to the PROBE test. Several parameters from the DPOAE section influence the PROBE test: rate, sets, size, sweeps, mode, stimulus, chk_atten, chk_swps, calibrate, atten, cal_swps, and scope. To ensure consistency, these parameters cannot be set in the PROBE section.

Count = 0

The starting point when EMVA generates a sequence number for a PRB file name. The default value tells EMVA to generate file names starting with the sequence number 0. Setting count=100 causes EMVA to start generating file names with the sequence number 100, which is encoded in the last two characters of the filename as "AA".

Crosstalk = Yes

Estimate cross-talk between receivers and microphone when determining Thévenin-equivalent source characteristics.

dec_length = No

Indicates that cavity measurements are always in order of decreasing length.

diacav = 8

Specifies diameter (mm) of calibration cavities.

f1_erf = 500

Lower limit (Hz) of frequency range over which the length-iteration error is calculated.

f2_erf = 8000

Upper limit (Hz) of frequency range over which the length-iteration error is calculated.

Ext_Iter =

Extends iteration when determining Thévenin-equivalent source characteristics by included cavity temperature and diameter, when Ext_Iter=Temp, or cavity termination reflection coefficient, when Ext_Iter=Refl.

FFTdB = 40

The vertical range (in dB) of the PROBE data display.

FFTkHz = 10

The horizontal range (in kHz) of the PROBE data display.

FFTref = 1

The reference level used for the PROBE spectrum display. A value of 0 specifies a “level per cycle” reference, which is the equivalent SPL level in a 1 Hz bandwidth. A value of 1 species an SPL level *re* 20 micropascal rms.

Iterate = 1

Iterate lengths to reduce error when determining Thévenin-equivalent source characteristics.

Max_Len = 12

Tells EMAV that no cavity length will exceed this maximum value (cm).

minres = 0

Minimum (cgs acoustic ohms) of the real part of the estimated load impedance.

Ncav = 5

Number of calibration cavities, which must be more than 1.

Niter = 100

Number of iterations when improving cavity length estimates.

Nsrc = 2

Number of sound sources.

scope = 8192

Allows the waveform display to show less than the entire stimulus interval during the calibrate phase of the DPOAE test.

smooth = 0

Number of points included when smoothing Thévenin impedance and pressure.

surge = 8

Number of iterations when determining “surge gain”, which is an estimate of the ratio of calibration-cavity diameter to load-cavity diameter. When surge=0, the ratio the diameters are assumed to be equal.

temp = 25

Specifies temperature (degrees Celsius) of calibration cavities.

write_txt = Y

Tells EMAV to write estimated Thévenin source and load characteristics (both left and right channels) to text files (ths1, ths2, thl1, and thl2) suitable for plotting. The default value is “Y: for “yes”. The alternate values is “N” for “no”.

TONE Parameters

The TONE section begins with a line containing the word TONE in brackets like this: [TONE]. Parameters in this section relate to the TONE test.

duration = 1

Specifies the duration of the tone in seconds.

frequency = 1000

Specifies the frequency of the tone in Hz.

HPF1 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 1 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

HPF2 = 0

The cutoff frequency (Hz) of a 12-dB/octave, high-pass filter on A/D channel 2 for the purpose of blocking very-low-frequency noise. The default value is zero, which disables this feature.

level = 65

Specifies the level of the tone in dB SPL. The level calibration is based on the sensitivity of the speakers and not on microphone measurement.

6.2 LIST FILE

A list file is an ASCII file that specifies the stimulus conditions that are to be presented during the measurement phase of the DPOAE test. The list file contains two sections. The header contains lines beginning with a semicolon that specify aspects of the DPOAE test other than the stimulus conditions. Comments may also be included in this section if preceded by a semicolon. Following the list file header is a section that contains one line for each stimulus condition.

Refer to the Parameters table for a listing of parameters that can be included in the header of a list file. A majority of these parameters can also be defined through the Options menu within EMAV. The descriptions of these parameters are located in the Menus section of this manual. All remaining list file parameters are defined below.

Items = 1

Maximum number of items for Trials axis.

Repeat = 1

Number of times that the stimulus conditions will be repeated

If the default values are satisfactory, then it is not necessary for a list file to contain any of the above header lines.

Each stimulus condition line should contain at least seven numbers:

1. Frequency of the F2 tone in Hz (F2)
2. Frequency of the F1 tone in Hz (F1)
3. Level of the F2 tone in dB SPL (L2)
4. Level of the F1 tone in dB SPL (L1)
5. Duration of averaging time (not including portions rejected due to artifacts) in seconds (T)
6. Noise level stopping criterion in dB SPL (Noise)
7. Signal-to-noise ratio (SNR) stopping criterion (at the DP frequency) in dB (SNR)

Other columns can be added to specify the frequency and level of suppressor tones, the phase of the F1 tone, and the attenuation on the soundcard input.

8. Frequency of the F3 tone in Hz (F3)
9. Level of the F3 tone in dB SPL (L3)
10. Frequency of the F4 tone in Hz (F4)
11. Level of the F4 tone in dB SPL (L4)
12. Phase of the F1 tone in degrees (<dp)
13. Attenuation on the soundcard input (dB)

With 24-bit soundcards, no attenuation hardware is needed, so the equivalent input attenuation is implemented in the software prior to averaging.

It is easier to edit an existing list file than to create a new file in a text editor. New files can be generated from within EMAV by selecting "Create List File" in the File menu or by not selecting any existing list files in the measurement phase of the DPOAE test.

6.3 STIMULUS FILES

Although EMAV generates chirps and white noise stimuli internally, it may use also external stimulus files for the check-fit and calibration phases of both the TEOAE and DPOAE tests. The TEOAE stimulus used for calibration is always the same as the stimulus used for measurement. Tone stimuli are always generated internally.

External stimulus files have a binary format which is essentially the same as an ILS sampled-data file. These files have a 512-byte "header" followed by the stimulus waveform. The header contains information about the sample rate and length of the stimulus. The waveform is written as 16-bit integers, two bytes per sample. The EMAV program is currently set up to use two stimulus files: mtb5.ils and sweep32.ils. The sampling rate of the stimulus waveform should match the sampling rate to be used for the DPOAE or TEOAE test.

EMAV stimulus files can be created from other waveform file formats by using the BTNRH program SDPC, which runs under DOS. For example, to create a stimulus file from a WAV file, type "sdcpc file.wav file.ils"

6.4 DATA FILES

The TEOAE test generates a binary data file with a .TOK extension that contains three waveforms: the recorded stimulus waveform and two separate measurements of the response (echo) waveform. Patient/subject information and stimulus conditions are also saved in this file. The file format is a BTNRH tokenfile. This file format is similar to an ILS sampled-data file, with additional information appended. The waveforms in the TEOAE data file may be viewed with the BTNRH WavEd program and spectrographs can be generated with the BTNRH SPECTO program.

The TOK file may be converted to a MATLAB compatible file using the TOKMAT program (described in Appendix D).

The DPOAE test generates an ASCII data file with a .DAT extension containing the measurements of the stimulus and distortion product levels. The data file has several lines of “header” information followed by one data line for each stimulus condition. The header lines all begin with a semi-colon to distinguish them from the data lines. The data lines have one of several possible formats: Normal, Extended, High Order, Multi, or SFOAE. The formats are described below.

The DPOAE test also creates a binary file with a .CAL extension containing the system calibration waveforms (BTNHR token-file format) for the two channels. When the SaveBin parameter is set to Yes, the DPOAE test will also create a binary file with a .BIN extension that contains the response waveforms for each stimulus condition. The .CAL and .BIN files may be deleted to save disk space if the information that they contain is not needed.

The names of .TOK and .DAT files are generated automatically. The .CAL and .BIN files are given the same name as the corresponding .DAT file. The file names are based on the current date and a sequence number. The file name begins with the last two digits of the current year, followed by a single letter corresponding to the month, (A=January, B=February, etc.), then the day within the month and, finally, the sequence number. There are 777 possible sequence numbers: 00, 01, ... 99, AA, AB, ... ZZ, and ##. The current sequence numbers for both .TOK and .DAT files are stored in a file called “emav.cnt”. A new file name is first compared with files in the current directory and the sequence number incremented until no existing file is found with the same name.

Normal Format

In the Normal data format, each data line contains 10 numbers. The header line above the data line header indicates the DP component. When the calibration is performed in FPL or SIL and Normal Format is selected, FPL and SIL voltages are not reported in the .DAT file.

1. F2 frequency (Hz) (F2)
2. F1 frequency (Hz) (F1)
3. Level measured at the F2 frequency (dB SPL) (L2)
4. Level measured at the F1 frequency (dB SPL) (L1)
5. Total duration of sweeps accepted for averaging (sec) (T)
6. Signal level measured at the DP frequency (dB SPL) (Ld)
7. Noise level measured at the DP frequency (dB SPL) (Ndp)
8. Reproducibility, defined as the cross-correlation between partial averages (A & B) (Rep)
9. Phase of the DP frequency component in degrees relative to onset of the stimulus (deg) (<dp)
10. Total averaging time including rejected sweeps and pauses (sec) (AvT)

Extended Format

In the Extended data format, each data line contains 24 numbers when there are no suppressor tones or 32 numbers when suppressor tones have been enabled. The first 5 numbers are the same as in the Normal format.

1. F2 frequency (Hz) (F1)
2. F1 frequency (Hz) (F2)
3. Measured level at the F2 frequency (dB SPL) (L2)
4. Measured level at the F1 frequency (dB SPL) (L2)
5. Total duration of sweeps accepted for averaging (sec). (T)

The next 15 numbers describe the signal level, noise level and phase at several distortion product frequencies. These numbers are grouped into 5 sets of three numbers

- 6 – 8. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \cdot F1 - F2$ frequency (Ld, Ndp, <dp)
- 9 – 11. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $3 \cdot F1 - 2 \cdot F2$ frequency (Ld, Ndp, <dp)
- 12 – 14. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $4 \cdot F1 - 3 \cdot F2$ frequency (Ld, Ndp, <dp)

- 15 – 17. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \times F_2 - F_1$ frequency (Ld, Ndp, <dp)
- 18 – 20. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $F_2 - F_1$ frequency (Ld, Ndp, <dp)

The next four numbers describe the noise level and phase at the stimulus frequencies (F_1 & F_2).

- 21 – 22. Noise (dB SPL) and phase (deg) at the F_1 frequency (N_1 , <1)
- 23 – 24. Noise (dB SPL) and phase (deg) at the F_2 frequency (N_2 , <2)

The next eight numbers describe the frequency, signal level, noise level, and phase at the suppressor frequencies (F_3 & F_4). These lines are only present when suppressor tones have been enabled.

- 25 – 28. Frequency (Hz), signal (dB SPL), noise (dB SPL), and phase (deg) of F_3 (F_3 , L_3 , N_3 , <3)
- 29 – 32. Frequency (Hz), signal (dB SPL), noise (dB SPL), and phase (deg) of F_4 (F_4 , L_4 , N_4 , <4)

The next two numbers are the L_1 and L_2 that were specified in the list file. If a suppressor was presented, the next two numbers are the L_3 and L_4 that were specified in the list file. The next two numbers are the output voltages V_1 and V_2 (in dB re 1 mV). If a suppressor was presented, the next two numbers are the output voltages V_3 and V_4 (in dB re 1 mV).

- 33 - 36. L_1 - L_4 that was specified in the list file. L_3 and L_4 will only be present if suppressors were presented.
- 37 - 40. Output voltages (in dB re 1 mV) for L_1 - L_4 calculated using the calibration procedure specified in the list file (SPL, FPL or SIL). V_3 and V_4 will only be present if suppressors were presented.

When measurements are made in FPL or SIL, the following columns are added to the .DAT file.

- 41 - 44. FPL (L_1 - L_4). The level required to produce the desired level at the eardrum when FPL is used to calibrate. L_3 and L_4 will only be present if suppressors were presented.
- 45 - 48. SIL (L_1 - L_4). The level required to produce the desired level at the eardrum when SIL is used to calibrate. L_3 and L_4 will only be present if suppressors were presented.
- 49 - 52. RFL (R_1 - R_4). Middle-ear reflectance (dB) calculated from measurements made during the sound source calibration for FPL. R_3 and R_4 will only be present if suppressors were presented.

High Order Format

The High Order data format is identical to the Extended data format except that $2 \times F_2 - F_1$ distortion component is replaced by the $5 \times F_1 - 4 \times F_2$ component.

- 15 – 17. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $5 \times F_1 - 4 \times F_2$ frequency (Ld, Ndp, <dp)

Multi Format

The Multi data format is similar to the Extended data format with additional distortion frequencies related to suppressor tone interaction. It was designed to explore the placement of F_3 and F_4 immediately adjacent to F_2 . Assuming that “suppressor” tones are enabled, there are 44 numbers on each line in the Multi format. The first twenty numbers are the same as in the Extended format. Here is a description of numbers 21 - 44.

- 21 – 23. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \times F_1 - F_3$ frequency (Ld, Ndp, <dp)
- 24 – 26. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \times F_3 - F_1$ frequency (Ld, Ndp, <dp)
- 27 – 29. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \times F_1 - F_4$ frequency (Ld, Ndp, <dp)
- 30 – 32. Signal (dB SPL), noise (dB SPL), and phase (deg) at the $2 \times F_4 - F_1$ frequency (Ld, Ndp, <dp)
- 33 – 34. Noise (dB SPL) and phase (deg) at the F_1 frequency
- 35 – 36. Noise (dB SPL) and phase (deg) at the F_2 frequency
- 37 – 40. Frequency (Hz), signal (dB SPL), noise (dB SPL), and phase (deg) of F_3 component
- 41 – 44. Frequency (Hz), signal (dB SPL), noise (dB SPL), and phase (deg) of F_4 component

The next two numbers are the L1 and L2 that were specified in the list file. If a suppressor was presented, the next two numbers are the L3 and L4 that were specified in the list file. The next two numbers are the output voltages V1 and V2 (in dB re 1 mV). If a suppressor was presented, the next two numbers are the output voltages V3 and V4 (in dB re 1 mV).

45-48. L1-L4 that was specified in the list file. L3 and L4 will only be present if suppressors were presented.

49-52. Output voltages (in dB re 1 mV) for L1-L4 calculated using the calibration procedure specified in the list file (SPL, FPL or SIL). V3 and V4 will only be present if suppressors were presented.

When measurements are made in FPL or SIL, the following columns are added to the .DAT file.

53-56. FPL (L1-L4). The level required to produce the desired level at the eardrum when FPL is used to calibrate. L3 and L4 will only be present if suppressors were presented.

57-60. SIL (L1-L4). The level required to produce the desired level at the eardrum when SIL is used to calibrate. L3 and L4 will only be present if suppressors were presented.

61-64. RFL (R1-R4). Middle-ear reflectance (dB) calculated from measurements made during the sound source calibration for FPL. R3 and R4 will only be present if suppressors were presented.

SFOAE Format

In the SFOAE data format, each data line contains the following numbers. Fm refers to the frequency of the modulation of F1.

1. F2 frequency (Hz)
2. F1 frequency (Hz)
3. Level measured at the F2 frequency (dB SPL)
4. Level measured at the F1 frequency (dB SPL)
5. Total duration of sweeps accepted for averaging (sec)
6. Signal level at the F2-Fm frequency (dB SPL)
7. Signal level at the F2+Fm frequency (dB SPL)
8. Signal level at the F1-Fm frequency (dB SPL)
9. Signal level at the F1+Fm frequency (dB SPL)
10. Signal level measured at the DP frequency (dB SPL)
11. Noise level measured at the DP frequency (dB SPL)
12. Phase of the DP frequency component in degrees relative to onset of the stimulus (deg)

If a suppressor was presented, the next eight numbers are F3, L3, N3, <3, F4, L4, N4, and <4, where < denotes the phase of that component. The next two numbers are the L1 and L2 that were specified in the list file. If a suppressor was presented, the next two numbers are the L3 and L4 that were specified in the list file. The next two numbers are the output voltages V1 and V2 (in dB re 1 mV). If a suppressor was presented, the next two numbers are the output voltages V3 and V4 (in dB re 1 mV).

7. EMAV PARAMETERS

The following table contains parameters that can be used in list and configuration files. The default values specified in the EMAV manual are set by a function within EMAV that is called when the program starts. The configuration file is also read when the program starts, so it can change any of the default values. The list file is read whenever the DPOAE test starts. The parameter codes are not case-sensitive.

Parameter Name	Menu Name (Options Tab)	Page in Manual	Default Value	Range of Values	List File	Config File Section
abscissa		21	F2	F1, F2, Fd, L1, L2, L3, Trials	X	DPOAE
AD_sensitivity	AD_sensitivity	13	3276.7			SYSTEM
AD_type		19	0	0,1		SYSTEM
adjust		20	M	A, M		TEOAE
atten	Calibration attenuation	14	20			DPOAE
cal_swps		21	16			DPOAE
calibrate	Calibration stimulus	14	~lsw	~lsw, ~bbn		DPOAE
calibrate_phase		21	y	y, n		DPOAE
checkfit	Checkfit stimulus	16	~ssw			TEOAE
checkfit	Checkfit stimulus	14	~lsw	~lsw, ~bbn		DPOAE
chk_atten		20	10			TEOAE
chk_atten		21	10			DPOAE
chk_swps		21	25			TEOAE
chk_swps		22	4			DPOAE
count		22	0			DPOAE
crosstalk	Compute Crosstalk	17	Y	N, Y		PROBE
DA1_sensitivity	DA1_sensitivity	13	3276.7			SYSTEM
DA2_sensitivity	DA2_sensitivity	13	3276.7			SYSTEM
DataFmt	Data Format Type	15	Normal	Normal, High Order, Extended, Multi, SFOAE	X	DPOAE
dec_len	Length Decrease	18	N	N, Y		PROBE
diacav	Diameter	17	0.8			PROBE
DP_freq	DP Frequency	14	2*F1-F2	F2-F1, 2*F1-F2, 3*F1-2*F2, 4*F1-3*F2	X	DPOAE
DSP		19	A	A, W		SYSTEM
DSP_code		19				SYSTEM
duration		26	1			TONE
ext_iter	Iterate T & D	17	N	N, Y		PROBE
F1	Hi Pass Filter Frequency	16	2000			TEOAE
F1_adjust	Adjust F1	14		Yes, No	X	none
f1_erf	f1_erf	17	500			PROBE
F2	Lo Pass Filter Frequency	16	5656.9			TEOAE
f2_erf	f2_erf	17	8000			PROBE
FFTdB	Spectrum Level Range	16	40			TEOAE
FFTdB	Spectrum Level Range	15	80			DPOAE

Parameter Name	Menu Name (Options Tab)	Page in Manual	Default Value	Range of Values	List File	Config File Section
FFTdB	Spectrum Level Range	17	80			PROBE
FFtkHz	Spectrum Frequency Range	16	8			TEOAE
FFtkHz	Spectrum Frequency Range	15	8			DPOAE
FFtkHz	Spectrum Frequency Range	17	8			PROBE
FFTmax		22	80			DPOAE
FFTmin		22	-60			DPOAE
FFTref		21	0	0, 1		TEOAE
FFTref		22	1	0, 1		DPOAE
FFTref		24	1			PROBE
frequency		26	1000			TONE
GREG		19	16			SYSTEM
HPF_freq	HPF Frequency	15	200			DPOAE
HPF_type	HPF Type	15	1	0, 1, 2		DPOAE
HPF1		21	0			TEOAE
HPF1		22	0			DPOAE
HPF1		26	0			TONE
HPF2		21	0			TEOAE
HPF2		22	0			DPOAE
HPF2		26	0			TONE
items		27	1		X	DPOAE
iterate	Iterate Length	17	Y	Y, N, M		PROBE
Label		20	Boys Town National Research Hospital			PRINTER
level		26	65			TONE
level_unit	Level Unit	14	SPL	SPL, SL, FPL, SIL	X	DPOAE
limit	Rejection Threshold	16	1		X	TEOAE
limit		22	2		X	DPOAE
list	Protocol File	14	*.lst			DPOAE
LS1_sensitivity	LS1_sensitivity	13	20			SYSTEM
LS2_sensitivity	LS2_sensitivity	13	20			SYSTEM
max_len		25	9			PROBE
MaxFrqOct	Maximum Frequency	15	4		X	DPOAE
MaxLevAbs	Maximum Abscissa Level	15	100		X	DPOAE
MaxLevOrd	Maximum Ordinate Level	14	80		X	DPOAE
MinFrqOct	Minimum Frequency	15	-2		X	DPOAE
MinLevAbs	Minimum Abscissa Level	15	10		X	DPOAE
MinLevOrd	Minimum Ordinate Level	15	-60		X	DPOAE
minres	Min. Res.	17	0			PROBE
mode	Stimulus Presentation Mode	16	3	0, 1, 2, 3, 4, 5		TEOAE
MP_sensitivity	MP_sensitivity	13	5	.05, .5, 5		SYSTEM
MP_transfer	MP_transfer	13				SYSTEM

Parameter Name	Menu Name (Options Tab)	Page in Manual	Default Value	Range of Values	List File	Config File Section
ncav	Number of cavities to test	17	5			PROBE
nic		22	1	1, 2	X	DPOAE
niter		25	100			PROBE
nnsb	NNSB	15	0		X	DPOAE
nsrc	Number of sources to test	17	2			PROBE
octave	DP Octave	15	0	0, 1, 2	X	DPOAE
Orient		20	Landscape	Landscape, Portrait		PRINTER
Port		20	screen.ps			PRINTER
ramp	Ramp Time	16				TEOAE
ramp	Ramp Time	14	10			DPOAE
ramp	Ramp Time	18	10			TONE
rate	Rate of Clock	16	32000			TEOAE
rate	Rate of Clock	14	32000			DPOAE
rate	Rate of Clock	18	32000			TONE
RedThr	Reduction Threshold	16	0		X	DPOAE
refresh		19	5			SYSTEM
repeat		27	1		X	DPOAE
SaveBin	Save Response Binary	15	No	Yes, No	X	DPOAE
scope		23	8192			DPOAE
sets	Number of Sweep Sets	16	128			TEOAE
sets		23	2			DPOAE
signal		24	DP+	DP+, DP-	X	DPOAE
size	Buffer Size	16	512			TEOAE
size	Buffer Size	14	2048			DPOAE
size	Buffer Size	18	2048			TONE
skips		24	1			DPOAE
smooth	Smooth	17	0			PROBE
SpecFreqRange	Spec. Freq. Range Type	15	Fixed	Fixed, Automatic		DPOAE
stimulus	Stimulus File	16	~ssw			TEOAE
suppressor	Suppressor	14		None, Tone, BPN, XBPN	X	none
sweeps	Sweeps per Set	16	8		X	TEOAE
sweeps	Sweeps per Set	14	32		X	DPOAE
target		21	70			TEOAE
temp	Temperature	17	25			PROBE
thsf	THS File	24	*.THS			DPOAE
time1	Start Time	16	6			TEOAE
time2	End Time	16	16			TEOAE
tolerance		21	1			TEOAE
trefresh		21	0			TEOAE
Type		20	PostScript	Color PostScript, PostScript, PCL		PRINTER
write_txt		25	Y	Y, N		PROBE

ACKNOWLEDGEMENTS

Extensive testing of the EMAV program would not have been possible without the collaboration of Michael Gorga and other colleagues at BTNRH. Jon Siegel has also done extensive testing of EMAV and has provided many excellent suggestions for new program features. The DPOAE test was designed to be similar to the CUBeDIS program written by Jont Allen. The TEOAE test was designed to be similar to the ILO88 program distributed by Otodynamics. Tom Creutz designed the original token-file format on which the TEOAE data file format was based. Judy Kopun and Sarah Michael revised and expanded the 2008 version of the User's Manual. Since 1994, commercial rights to the EMAV program have been licensed by BTNRH to the Bio-logic Systems Corporation; however, the EMAV program may still be redistributed for non-profit use in scientific and clinical research.

APPENDIX A: EMAV INSTALLATION

Windows. The most recent version of EMAV may be downloaded at <http://audres.org/downloads/emav-setup.zip>. Inside this zip file is an installation wizard (setup.exe) that will copy program files to your computer and place a shortcut on the Desktop.

It is recommended that you create a dedicated folder for EMAV data files. Edit the “Properties” of the EMAV shortcut on the Desktop to the “Start in” the folder that you’ve created for data files. Into this folder, copy emav.ini from “C:\Program Files\BTNRH\EMAV\” and any protocol (*.lst) files that you will use with EMAV. Edit your copy of emav.ini to specify the correct soundcard, adjust the sensitivity parameters (see Appendix C), and modify any default values for other EMAV parameters. Having your local changes to emav.ini in the data folder prevents them from being overwritten when a new version of EMAV is installed.

The User’s Manual is online at <http://audres.org/downloads/emavtm.pdf>. Examples of data and list files, along with a few related DOS programs are at <http://audres.org/downloads/emav.zip> (See Appendix D). Several files that may facilitate the use of EMAV with MATLAB are at <http://audres.org/downloads/emav-mat.zip>.

Linux. The Linux version of EMAV is available at <http://audres.org/emav-linux.zip>.

The executable program file “emav” should be placed in /usr/local/bin. The configuration file “emav.ini” should be edited to have the desired default parameter values and be placed in a subdirectory created for EMAV data files.

EMAV uses the X window system to provide a graphical user interface under Linux and an ALSA driver is assumed to be available for the soundcard to be used with EMAV. The Linux version of EMAV has been tested with Mandriva Linux 2008 and the Indigo IO soundcard.

APPENDIX B: SOUND CARD INSTALLATION

CardDeluxe

The CardDeluxe is manufactured by Digital Audio Labs. <http://www.digitalaudio.com/>

Connection of the CardDeluxe to the ER-10C requires three cables that each have a mono ¼" phone plug (TS) at one end and a BNC connector at the other end.

Windows. The CardDeluxe can perform synchronous averaging using its original MME Windows drivers.¹ In the device properties for this soundcard, the following configuration changes should be made.

Under the Configuration tab, set the "Record mode" to "lock to playback" and check to see that the "24-bit mode" is set to "left".

Under the Monitor tab, select only "Analog Output" in the Analog Output mixer section.

In the file EMAV.INI, set "DSP=W" for specify use of the Windows driver for this soundcard, set "DSP_code=", and set the soundcard sensitivity parameters according to the procedure in Appendix C.

Linux. The CardDeluxe is not usable under Linux because no ALSA drivers are available.

Indigo IO

The Indigo IO is manufactured by Echo Digital Audio. <http://www.echoaudio.com/>

Connection of the Indigo IO to the ER-10C requires two cables that each have a stereo 1/8" phone plug (TRS) at one end and two BNC connectors at the other end.

Windows. The Indigo IO has acceptable performance with the most recent WDM driver. The I/O synchronization is consistent more than 99% of the time with the WDM driver. The ASIO driver provides better I/O synchronization that is consistent 100% of the time. EMAV can use either the WDM or ASIO driver with the Indigo IO, but prefers to use the ASIO driver when it is installed.

Indigo IO Console (software):

1. Open the Console and Click on "Settings" Button
 - a. Driver Tab:
 - i. Select "The sample rate is unlocked"
 - ii. Check "Synchronize wave devices"
 - b. Console Tab:
 - i. Check the following (these are the defaults and are not critical to device function; "i" maintains the settings chosen in this setup.
 1. "Load last session when console starts"
 2. "Auto-save session when console exits"
 3. "Load wave device configuration from sessions"
 - c. GSIF Tab (default setting, not important for EMAV, SYSRES operation)
 - i. Select 128 samples
 - d. About Tab
 - i. EMAV and SYSRES function has been verified for driver and console versions 6.14 downloaded from www.echoaudio.com
2. On the console itself (the part that looks like a graphic equalizer)
 - a. Unmute (m button looks raised):
 - i. Virtual out channels 1 and 2
 - ii. Analog out channels 1 and 2
 - b. All other inputs and outputs are muted

¹ As of February 2008, we continue to use the (now obsolete) MME drivers because, in our tests, the newer WDM drivers would not synchronously record and playback consistently. The most recent WDM driver available on the manufacturer's website (5.10.3517) has reportedly reacquired consistent I/O synchronization, but we have not yet been verified this claim.

Windows Settings (software)

1. Open Control Panel and Click on “Sounds, Speech, and Audio Devices”; Under “Sounds and Audio Devices” do the following:
 - a. Audio Tab:
 - i. All default devices should be set at something other than the Indigo IO
 - ii. Check “Use only default devices”
 - b. Hardware Tab: open properties for the Echo Indigo io device
 - i. General Tab: Device usage should be set to: “Use this device (enable)”
 - ii. Properties Tab: Click on “Indigo io wave driver” under “Audio Devices” and then click on “Properties” button
 1. Select “Use audio features on this device”
 2. Check “Do not map through this device”
2. All other settings left at default or as configured for other devices

EMAV Config File settings (Software control file):

Need to choose the Indigo (as compared to the CardDeluxe) in the configuration file settings as shown below:

```
; Indigo [23-Oct-03]
;
; DSP=W
; DSP_CODE=Indigo
; AD_sensitivity=3566000 ; cnt/V
; DA1_sensitivity=15000 ; cnt/V
; DA2_sensitivity=15000 ; cnt/V
;
; CardDeluxe [CEL, 24-Oct-03]
;
; DSP=W
; DSP_CODE=CardDeluxe
; AD_sensitivity=856907 ; cnt/V (+4dB)
; DA1_sensitivity=6922 ; cnt/V (+4db)
; DA2_sensitivity=6548 ; cnt/V (+4dB)
```

Indigo IO Soundcard Setting (Hardware)

1. Volume control should be at max out (fully rotated away from the computer)

Verification

With the software and hardware configured as above the following results should be obtained when running the following programs:

TSTLAT: (A DOS program run from the command prompt)

“Outstanding” latency consistency (all 100 clicks have a latency of 1.875ms) – slight deviation from this is acceptable (e.g., a single click out of 100 with a different latency), deviations more than this suggest a problem with the setup/drivers

SYSRES (run tstlat.lst in SYSRES)

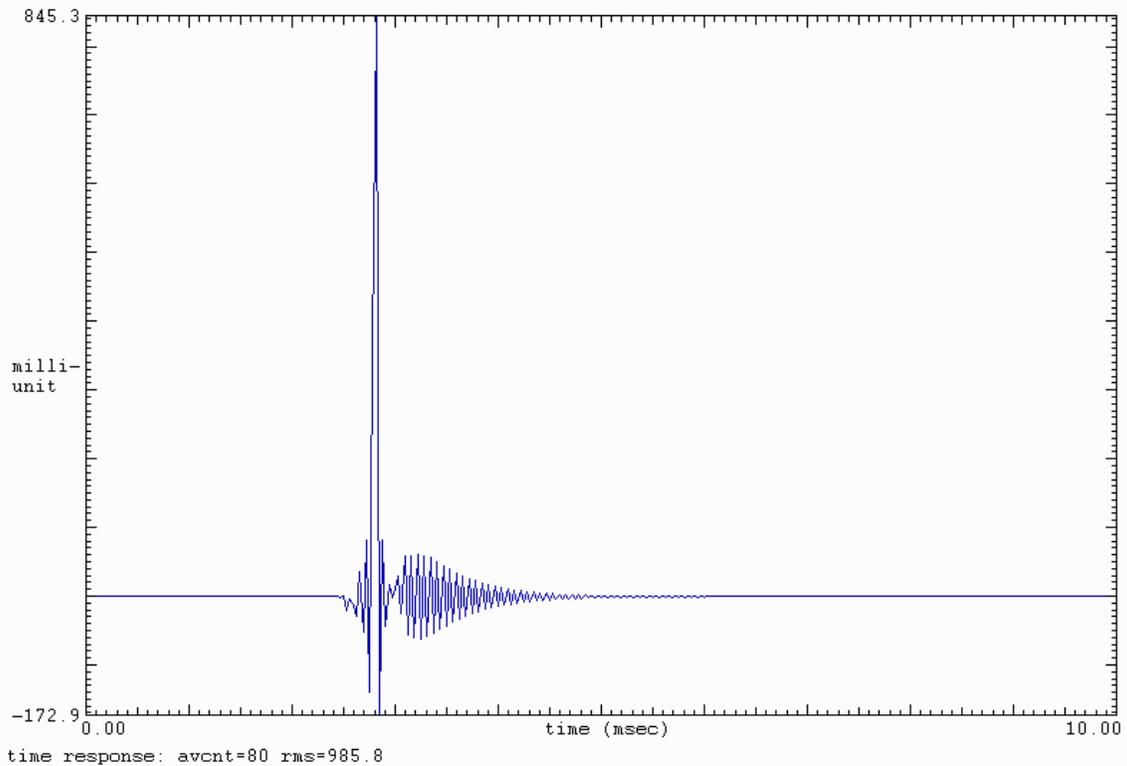
tstlat.lst contains the following code:

```
-----
; tstlat.lst - test stability of latency
;
st=1
sr=32000
ga=0.1
zt2=0.01
na=80
go
```

Should get the following result:

amplitude=1.000V d/a_mode=L a/d_mode=L stimulus=impulse
#averages=80 #samples=1024 rate=32kHz gap=100ms

SysRes
09/13/2006
16:09:36



EMAV: (use a loopback configuration – output routed directly to the input)

Run TEOAE check fit protocol. Should observe a single click with no “small” amplitude click waveforms at other latencies

Linux. The Indigo IO uses the ALSA driver under Linux and synchronizes I/O correctly in our tests.

APPENDIX C: SOUND CARD AND PROBE CALIBRATION

These procedures may be used to determine values for the soundcard and probe sensitivity parameters in the configuration file (emav.ini). New sensitivity values need to be added to the *.ini file and this file needs to be saved. The configuration file that comes with the installation of a new version of EMAV will need to be edited to contain soundcard and probe sensitivity parameters developed for your hardware. Alternatively, if soundcard sensitivity parameters have been previously specified for the ARSC library, then setting these parameters to zero in *.ini tells EMAV to use the ARSC values.

Soundcard Sensitivity Parameters

1. To make the soundcard sensitivity measurements you will need a multi-meter, double banana to BNC connector, BNC to BNC cable, and two BNC T-connectors.



Double Banana to BNC connector



BNC T-connector

2. Plug the double banana-BNC-connector into the COM and V/ Ω jacks on the multi-meter. Make sure the function selector is set to AC \square .
3. Connect the BNC T-connector to the multi-meter via the double banana-BNC connector. Connect a second BNC-T connector to the first BNC-T connector.
4. Connect the REC1 cable from the computer (output of sound card) to one side of the BNC T-connector. Connect the other side of the BNC T-connector, via the BNC to BNC cable, to the REC1 input of the ER10C.
5. Set LS1_sensitivity=5 (Options→System→I/O Sensitivity→LS1) so that the program will think that 1 volt (rms) generates 0.2 pascal (80 dB SPL) of sound pressure. Use the "Tone" test (Test→Tone) to tell the program to generate an 80 dB SPL test tone on "channel A". Set the duration parameter so that you have enough time to read the multi-meter. The voltage displayed on the multi-meter is the voltage being delivered to "receiver A". If this voltage is not exactly 1 volt rms, then the DA1_sensitivity needs to be changed. Divide the current DA1_sensitivity (Options→System→I/O Sensitivity→DA1) by the voltage just measured to obtain a new value for DA1_sensitivity. Enter the new DA1 setting and repeat step 5 making sure that 1.0 volt is measured.
6. To determine DA2 sensitivity, follow the procedure outlined in step 5. Disconnect the REC1 cable from the BNC_T connector and connect the REC2 cable in its place. Connect the other side of the BNC-T connector to the REC2 input of the ER10C. Set LS2_sensitivity=5, output an 80 dB SPL test tone to "channel B", and measure the voltage being delivered to "receiver B" to determine the new value for DA2_sensitivity.
7. Leave the equipment set up as in step 6. In addition, disconnect the microphone cable from the ER10C and connect it to BNC-T. The cord now goes from the multimeter to the input of the sound card. Keep LS1_sensitivity=5, set MP_sensitivity=5, feed a 1 volt rms signal from the first DAC into the first ADC by telling the program to output an 80 dB tone on channel B. If the "RMS pressure" displayed on the screen is not "80 dB SPL", then the value of the AD_sensitivity parameter needs to be adjusted. Take the current AD_sensitivity value (Options →System→I/O Sensitivity→AD_Sensitivity) times the "P-P" pressure (mPa) displayed on the screen and divide by 565.69 to obtain a new value for AD_sensitivity.

Probe Sensitivity Parameters

1. Set MP_sensitivity (Options →System→I/O Sensitivity→MP_sensitivity) to the value appropriate for the preamp switch setting. Etymotic microphones are calibrated so that 1 microvolt (rms) is generated by a tone at

20 micropascal. If the microphone preamp GAIN=0dB, then MP_sensitivity=0.05 (V/Pa). If GAIN=20dB, then MP_sensitivity=0.5. If GAIN=40dB, then MP_sensitivity=5.

2. Set the ER10C up as if running subjects. With the soundcard and microphone sensitivity parameters set to their proper values, place the probe in a "standard" acoustic cavity and tell the program to generate an 80 dB SPL tone on channel A. The "RMS pressure" displayed on the screen should be "80 dBSPL". If not, then adjust the LS1_sensitivity value until it is. (Don't worry about being too accurate; the correct value varies considerably with frequency and choice of acoustic cavity.)
3. Repeat step 2 on channel B to adjust LS2_sensitivity.

Reminder: All new sensitivity values need to be saved in the .ini file. Any values changed through the menu in EMAV will revert to the default values in the .ini file when EMAV is reopened.

APPENDIX D: AUXILIARY FILES

Examples of data and list files, along with a few related DOS programs, are at <http://audres.org/downloads/emav.zip>. Here's a list of the files included.

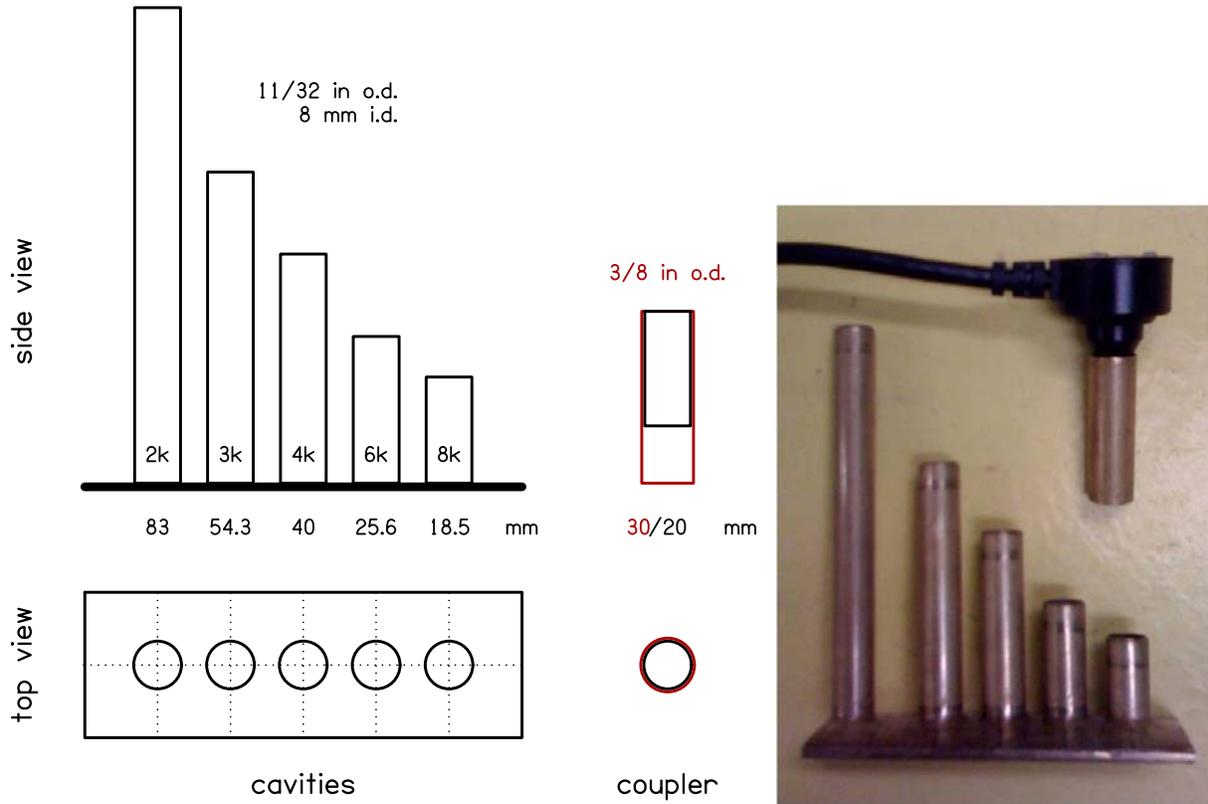
- 92L21D00.CAL - In-the-ear chirp response for 92L21D00.DAT.
- 92L21D00.DAT - DP-gram with five F2 frequencies in a normal-hearing ear from 1992.
- 92L21D01.CAL - Cavity chirp response for 92L21D01.DAT.
- 92L21D01.DAT - Same stimuli as used for the DP-gram in 92L21D00.DAT, but measured in a cavity.
- 92L21D03.CAL - In-the-ear chirp response for 92L21D03.DAT.
- 92L21D03.DAT - DP-filter function for $2 * F_1 - F_2$ with $F_2 = 4000$ Hz.
- 92L21D04.CAL - In-the-ear chirp response for 92L21D04.DAT.
- 92L21D04.DAT - DP-filter function for $3 * F_1 - 2 * F_2$ with $F_2 = 4000$ Hz.
- 93C25D00.CAL - In-the-ear chirp response for 93C25D00.DAT.
- 93C25D00.DAT - DP-filter functions for $2 * F_1 - F_2$ with $F_2 = 500, 1000, 2000, 4000,$ and 8000 Hz.
- calmat.exe - A DOS program that transforms an old-format DPOAE calibration file (.CAL) into a MATLAB data file. The current CAL format is already MATLAB compatible.
- emav.exe - A DOS version of EMAV that can display EMAV data files, but no longer has the capability to access the soundcard, so does not perform any synchronous averaging functions.
- emav.ico - The graphic used as an icon for EMAV.
- emav.ini - Example of an EMAV configuration file.
- fdpswp.lst - A list of stimulus conditions for measuring a DP-filter function for $2 * F_1 - F_2$ at $F_2 = 8000$.
- oba2.exe - A DOS program that performs an octave-band analysis of the response waveform stored in a TEOAE data file.
- std.lst - A list of stimulus conditions for measuring a DP-gram with five F2 frequencies.
- suppr.lst - A list of stimulus conditions that include a third tone as a suppressor.
- calmat.exe - A DOS program that transforms an TEOAE data file (.TOK) into a MATLAB data file.

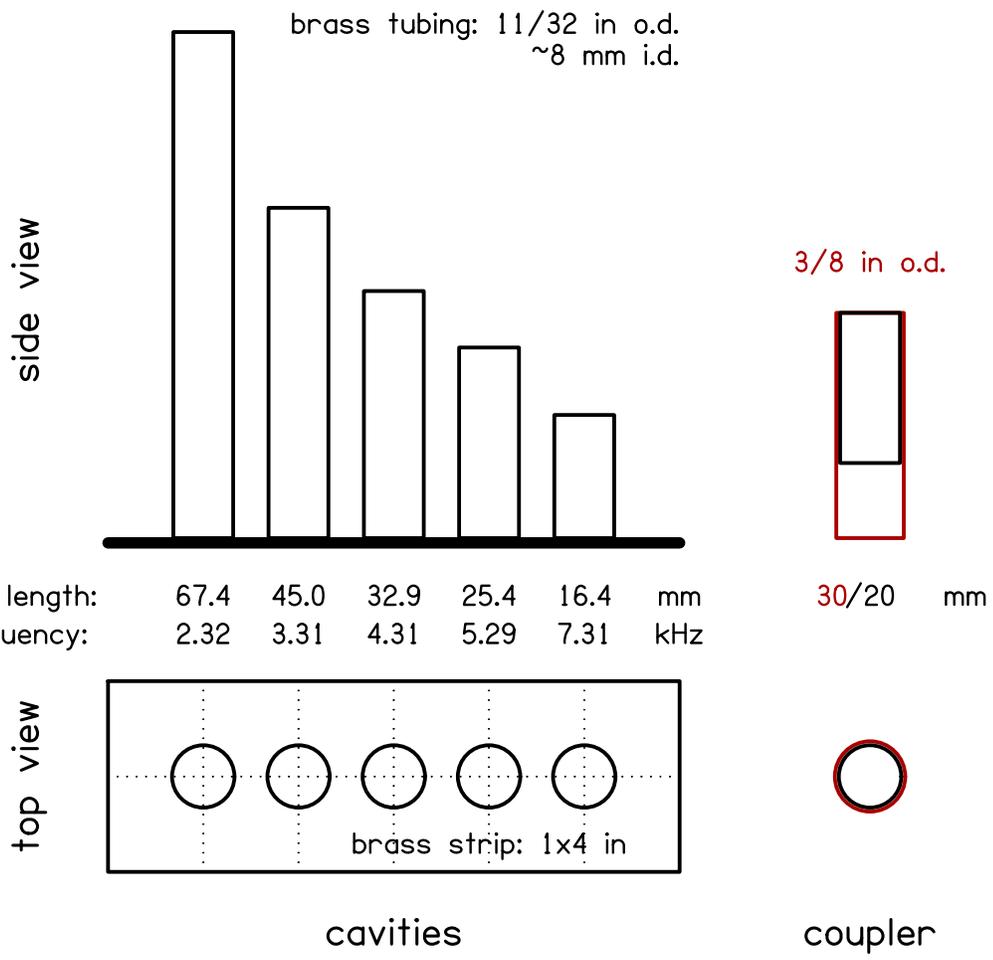
Several files that may facilitate the use of EMAV with MATLAB are at <http://audres.org/downloads/emav-mat.zip>. Here's a list of the files included.

- binmat.c - C source code that transforms a DPOAE binary file (.BIN) into a MATLAB data file. The binary file contains the response waveform for a single DPOAE stimulus condition.
- binmat.exe - A DOS program that transforms a DPOAE binary file (.BIN) into a MATLAB data file.
- calmat.c - C source code that transforms an old-format DPOAE calibration file (.CAL) into a MATLAB data file. The current CAL format is already MATLAB compatible.
- calmat.exe - A DOS program that transforms an old-format DPOAE calibration file (.CAL) into a MATLAB data file.
- calmat.m - A MATLAB script that displays the contents of MATLAB data file created by calmat.
- tokmat.c - C source code that transforms a TEOAE data file (.TOK) into a MATLAB data file.
- tokmat.exe - A DOS program that transforms a TEOAE data file (.TOK) into a MATLAB data file.
- tokmat.m - A MATLAB script that displays the contents of MATLAB data file created by tokmat.

APPENDIX E: CAVITY CONSTRUCTION

The probe calibration cavities are constructed from brass tubing mounted on a brass plate. A tube cutter is used to cut the brass tubes in the correct lengths. Super glue is used to attach the brass tubes to the base plate. Once the superglue is dry, epoxy resin is applied around the bottom of each cavity on the base plate in order to permanently adhere the cavities to the base plate. Super glue is used in the construction of the coupler. The original dimensions are given below. Alternative dimensions, which may be advantageous for frequencies above 10 kHz are given on the next page.





APPENDIX F: CALIBRATION FILES

When EmAv performs in-the-ear calibration, the stimulus waveform and two microphone responses (one for each loudspeaker) are stored in a CAL file. When stimulus levels are specified in SIL or FPL, EmAv requires additional calibration files with extensions PRB, THS, and THL. All EmAv calibration file types have formats compatible with MATLAB (version 4) data files. Within the MATLAB program, use the option “-V4” to load MAT (version 4) files.

CAL

CAL files contain the responses for a single acoustic cavity (typically an ear) to each of the two sound sources in a DPOAE probe.

CAL file variables			
name	length	type	Description
EMAV_CAL	1	short	number of cavity response waveforms
rate	1	float	sampling rate (Hz)
npts	1	short	number of points in each waveform
reps	1	short	sweeps per set
atten	1	float	stimulus attenuation re maximum amplitude
adsn	1	float	ADC sensitivity (32-bit count/volt)
da1sn	1	float	DAC1 sensitivity (16-bit count/volt)
da2sn	1	float	DAC2 sensitivity (16-bit count/volt)
mpsn	1	float	microphone sensitivity (V/Pa)
Scale	1	float	scale factor that converts waveform to pascal
stimfn	nchars	text	stimulus filename
stim	npts	short	stimulus waveform (16-bit version)
ncav	1	short	number of cavities (1)
cal01	npts	long	sum of 2*reps response waveforms (DAC1)
cal02	npts	long	sum of 2*reps response waveforms (DAC2)
sets	1	short	number of sets of sweeps (added in 3.19)

PRB

PRB files contain the responses for more than one acoustic cavity to each of the sound sources in a DPOAE probe. These cavity responses are typically used to determine Thévenin-equivalent source characteristics. The PRB file format is identical to the CAL file format except that there are two response waveforms for each cavity with variable names calNN where NN is a sequentially incremented two-digit number.

THS

THS files contain Thévenin-equivalent source characteristics derived from a PRB file. Crosstalk pressure is optional in the computation of source characteristics and set to zero when not computed.

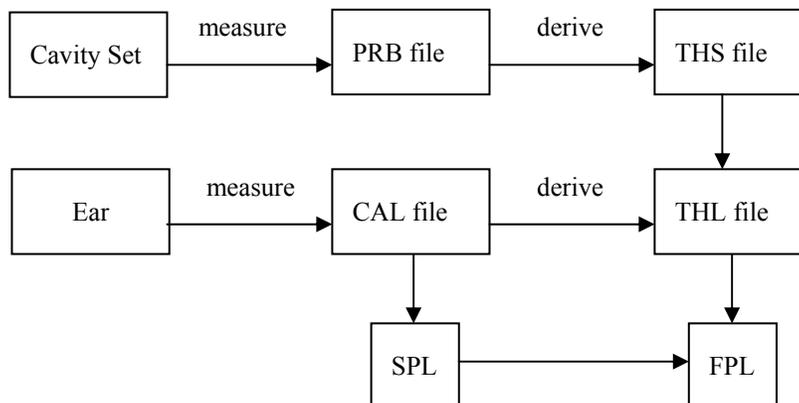
THS file variables			
name	length	type	description
SRC_THV	1	short	number of frequency components (nf)
rate	1	double	sampling rate (Hz)
df	1	double	frequency bin (Hz)
dc	1	double	cavity diameter (cm)
tc	1	double	cavity temperature (°C)
zs1	nf	complex double	DAC1 source impedance (cgs acoustic ohms)
zs2	nf	complex double	DAC2 source impedance (cgs acoustic ohms)
ps1	nf	complex double	DAC1 source pressure (Pa/V)
ps2	nf	complex double	DAC2 source pressure (Pa/V)
px1	nf	complex double	DAC1 crosstalk pressure (Pa/V)
px2	nf	complex double	DAC2 crosstalk pressure (Pa/V)
lc1	ncav	float	DAC1 estimated cavities lengths (cm)
lc2	ncav	float	DAC1 estimated cavities lengths (cm)
atten	1	float	stimulus attenuation (dB <i>re</i> maximum amplitude) (added in v3.16)
sets	1	short	number of sets of sweeps (added in v3.19)

THL

THL files contain Thévenin-equivalent load characteristics derived from a CAL file and a specified THS file. Characteristic impedance has the same value at all frequencies. Load pressure represents desired pressure/volt for SPL calibration. Relative pressure represents desired pressure for SIL or FPL calibration divided by desired pressure for SPL calibration. Thus, relative pressure provides the scale factor required to convert an SPL calibration into an SIL or FPL calibration.

THL file variables			
name	length	type	description
LD_THV	1	short	number of frequency components (nf)
rate	1	double	sampling rate (Hz)
df	1	double	frequency bin (Hz)
z0	nf	complex double	characteristic impedance (cgs acoustic ohms)
z1	nf	complex double	DAC1 load impedance (cgs acoustic ohms)
pl1	nf	complex double	DAC1 load pressure (Pa/V)
pr1	nf	complex double	DAC1 relative pressure for SIL or FPL
z2	nf	complex double	DAC2 load impedance (cgs acoustic ohms)
pl2	nf	complex double	DAC2 load pressure (Pa/V)
pr2	nf	complex double	DAC2 relative pressure for SIL or FPL
attn	1	double	stimulus attenuation (dB <i>re</i> maximum amplitude) (added in v3.19)
sets	1	short	number of sets of sweeps (added in v3.19)

The figure below illustrates how EMAV calibration files are related to each other.



INDEX

.CAL	24	End Time	15
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checkfit	11	HPF Type	14
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