IMPORTANT CAUTIONS: PLEASE READ!

1) When used also as multimedia soundcards, some of the DSP cards default in their driver software to a setting which connects the Auxiliary Input to play through the DSP card's output at initialization. If that input happens to be sensing the DSP card's output (directly or through an amplifier) via probes or microphone, oscillation could result. If the card is playing through an external amplifier or speakers, this oscillation, if sustained, could conceivably damage the external equipment! The oscillation may be very high in frequency and inaudible, so: **always disconnect (or otherwise disable) the probes and mic from sensing the DSP card output until the Liberty Audiosuite (LAUD) program has been started up, and disconnect or disable them before exiting LAUD.**

2) The 1X Direct Probes usable with the Turtle Beach cards are designed to provide protection of the card's inputs for voltages up to 40V peak (linear measurements to 20V peak; for higher peak voltages use a 10x Direct Probe). The MIC/PROBE preamp is designed to withstand accidental levels of over 100 Volts peak on its probes (NOT at the preamp connector, but through the 47.5Kohm resistor in the probe!). **Do not** use this equipment to measure points with high voltages such as are found on power lines, internal stages in tube amplifiers, electrostatic loudspeaker panels or other high voltage circuitry. **LAUD and associated hardware are intended only for circuitry operating at 40V and below!**

3) LAUD and the applicable DSP card types are capable of generating very high level (loud) sinewaves, squarewaves, pulses, and noise bursts which can damage your hearing as well as your equipment. Use caution and proceed slowly during tests. Liberty Audiosuite is intended for technical users who are familiar with the risks and proper precautions of operating test equipment, and it should not be used as a toy. Anyone lacking such familiarity and responsibility should NOT attempt to use the system.

4) If the DSP card is also being used as a multimedia soundcard, any computer system speakers should normally be disconnected or switched off during use of LAUD, to avoid the high drive and sound levels which could result in damage to hearing or to the speakers.

5) NEVER use LAUD while wearing headphones connected to a DSP card's speaker output jack (if present). Very high sound levels and hearing damage can result!

6) The DSP card inputs are designed to handle signal levels of only a few volts and can be easily damaged by high AC or DC voltage levels. **Because of this sensitivity to inadvertent damaging levels, Liberty Instruments claims no responsibility for equipment damaged when used with LAUD**. We cannot control what is connected to your DSP card, so **implementation of adequate care and protection is the user's responsibility**.

7) While the DSP card provides computer output level control, it will generate transients at turn-on and turn-off times and during initialization; these transients can potentially damage speakers or electronics in some setups. When using LAUD with an external power amplifier, particularly one of large power capability, **a volume or level control must be provided** between the sound card line outputs and the external power amplifier's input to control the gain. This can be accomplished by using a separate control-preamp unit, a passive level control (internal or external to the amplifier), or the level controlling features of an integrated amplifier or receiver.

System Description and Features

Liberty Audiosuite is an integrated collection of high performance audio test instruments implemented in software and making use of the versatile capabilities of the Turtle Beach Pinnacle/Fiji or the "PSA" type DSP soundcards.

TO GET GOING FAST: Read the system requirements (two pages ahead), then perform the hardware and software installations in the next section. You can then use the "Easy Scripts" for many measurements immediately, and come back later to read in the manual if more specialized usage is needed.

The Liberty Audiosuite (or "LAUD") instruments include:

- MLS/FFT: A sophisticated dual channel MLS (pseudorandom noise) or impulse-based network and impedance analyzer, ideal for loudspeaker and room acoustics work. This instrument is similar to Liberty Instruments' popular IMP/M system, but with four times the acquisition length, choice of 14 different sample rates and an improved graphical display.
- SINE: A versatile dual-channel gated or continuous sinewave-based network/impedance analyzer for more conventional audio frequency device measurements and for noise immunity at very low frequencies. With LAUD gated-sine measurements, both magnitude and phase are determined. Measurements can be made at log or linear frequency increments or at user specified frequency points. Both the MLS and the SINE instruments support microphone response correction and system normalization ("Cal").
- SCOPE: Dual Channel triggerable digital oscilloscope with adjustable sinewave/squarewave generator.
- SPEC_AN: Two types of audio-band spectrum analyzers are provided: first, an FFT-based spectrum analyzer with Pink and White noise sources; supports linear or log format frequency displays and power response averaging. Second, a constant percentage bandwidth Real Time Analyzer, providing both 1/3 octave resolution and also 1/6th octave resolution; averages of multiple separate measurements can be accumulated, and Cascade and Normalize math functions are provided, as well as A,B, and C weighting to the displayed data.
- DIST_AN: Harmonic distortion analyzer, with user selectable analysis of harmonics from second through ninth (as well as Total Harmonic Distortion) and display of distortion product percentage as a function of fundamental frequency. Special intelligent routines for sample rate and size selection minimize measurement times while preserving sensitivity to low product levels. Product levels under 0.03% are resolvable using PSA/Echo cards or under 0.005% using the Turtle Beach cards.

Each instrument is equipped with a range of features and options providing great versatility for advanced users. Yet Liberty Audiosuite is easy to operate, allowing new users to achieve useful results in minimal time.

Special built-in "**Easy Scripts**", which are guided and highly automated test procedures for commonly needed measurements, make for extremely simple usage even for users unfamiliar with the operation or theory of the measurement system.

For manufacturing purposes, LAUD now includes powerful Pass/Fail production control features. These use easily generated upper and lower limit curve data to be defined over a desired frequency range which is used to Evaluate a frequency response, impedance or distortion measurement. Limit ranges may also be specified for loudspeaker driver primary parameters for use in incoming inspection and test for loudspeaker manufacturers.

Some advanced functions and features included within various instruments in the package are:

- **Automatic input level adjustments** greatly simplify the initial setup of new measurements for optimum usage of the Analog to Digital converter's dynamic range. Use [Shift-F10].
- Optional user-initiated **automatic vertical display scaling** quickly and easily formats data displays. Use [=].
- A **MERGE facility** for editing frequency response data. Allows response curves (both magnitude and phase) to be pasted together, to be summed or subtracted over specified ranges, or even be graphically "drawn in".
- Ability to make quasi-anechoic loudspeaker measurements without need for an anechoic chamber, using both impulse response (MLS) and gated-sine techniques. LAUD version 2 now adds **selectable "automatic time-marker setting**" to identify the leading edge and first echo in many impulse response measurements, greatly **simplifying the quasi-anechoic measurement process**. Even better, an Easy Script is provided for these measurements, making what was once a complex process into a very simple one.
- A measurement and extraction system for **Thiele and Small (loudspeaker) parameters** (Qe, Qm, Qt, fs, Vas, Re, Efficiency, Bl product, Mass and Compliance) which can operate in both the MLS or SINE instruments. Once measured, these parameters help in accurate design of enclosures. For Vas, the added mass and closed-box methods are supported.
- Full measurement of **both magnitude and phase** in the MLS, SINE, DIST_AN instruments, in frequency response measurements as well as impedance and distortion measurements.
- Extremely versatile and easily programmable **Script processor** for high-level automation of special test sequences, data collection procedures, quality control processes and for simplified operation of many commonly required measurements (via instantly accessible **built-in "Easy Scripts"**).
- A versatile **1/3rd and 1/6th octave real-time analyzer**, with optional on-graph level readouts for each display bar. The 1/6th octave analyzer, rare in even expensive RTAs, is particularly useful for in-room response determination and optimization, providing a more accurate estimate of the spectrum perceived by a listener than is provided by a 1/3rd octave unit.
- **Energy-time curve analysis with RT60** measuring capability in selectable octave or full bands
- Calculation of the **Power Cepstrum (echo analysis)** for investigation of driver character, edge diffraction and baffle reflection sources.
- Powerful **data manipulation and conversion facilities** for frequency response and impedance measurements allowing you to take, for instance, frequency response or impedance data with frequency points at constant spacing (as acquired from the FFT based MLS analyzer instrument) and curve fit it to frequency points at log spacing (constant multiple) or even to a provided list of arbitrary frequency points. Data can be exported in ASCII format at frequencies best suited for design CAD programs.
- **Definable display axis and scaling**, rather than the fixed frequency ranges or magnitude scales of more limited systems.
- **Floating, dual on-trace data markers** for reading of data values in the SINE and MLS instruments and for publication quality printed plots, rather than just a "cross-hair" line.

- A **distortion "Visualizer"** which aids in determination of the cause, rather than just the amount of many types of distortion. The Visualizer allows you to dynamically increase and decrease the relative distortion levels of a displayed sine wave while keeping all non-fundamental components in the same proportions and phase. Using this, can help identify, for instance, distortion which is due to clipping onset or crossover notch in a power amplifier.
- **Cumulative Spectral Decay (Waterfall) plots** for identifying resonances in loudspeaker cones and enclosures. Vertical (dB/division), Horizontal (frequency range) and Depth (time period) are all adjustable for the plot.
- **Software controllable input and output gains**. The input levels can also be automatically adjusted.
- Choice of **14 different sample rates from 5.5125kHz to 48kHz** with **linear phase FIR-based antialiasing filtering**.
- **Dual channel 16-bit** data acquisition.
- **Data sizes and MLS sequence sizes to 16,384 points**, maintained in memory for fast access and processing. FFT sizes at all powers of 2 from 256 points to 16,384 points.
- **Capacitance and inductance measurement**, along with determination of effective series and parallel resistance, provides more useful information than just a simple "LC" meter. With LAUD, acquiring these values requires no special effort-- just place a marker on the impedance curve and read the displayed R-C or R-L equivalents.
- **User selectable time domain data windows** (Bingham. Hamming, Blackman and "half window" versions) are provided in SINE, MLS, SPEC_AN and DIST_AN measurements to optimize spectral containment by minimizing spectral leakage.
- Built-in "**Automeasure**" macro functions allow easy performance of basic level common measurements (MLS, SINE instruments).
- **Graphical interface** with mouse-based control of functions (except for SCOPE instrument)
- **Adjustable delay compensation** for correction or investigation of phase data
- **Built-in HELP**.
- HP LaserJet II, DeskJet, and Epson dot-matrix printer or Windows-compatible "bitmap file" (color or black/white) **graphics output support**.
- "**Cycling**" feature allows repetitive MLS measurements of networks and speakers, so you can watch results change while you work on the system (not just measure between adjustments).

System Requirements

Liberty Audiosuite (LAUD) version 2 requires a 386, 486 or Pentium based computer system with at 8MB of memory, a VGA or SVGA monitor, a mouse, and a loaded DOS mouse driver. If a 386 or 486SX type processor is used, a math coprocessor is also required. The computer must be equipped with a Turtle Beach Pinnacle or Fiji (these types only) or an ECHO DSP (or other Personal Sound Architecture DSP type card). Available Hard Disk drive C: space of 4MEG or greater is recommended.

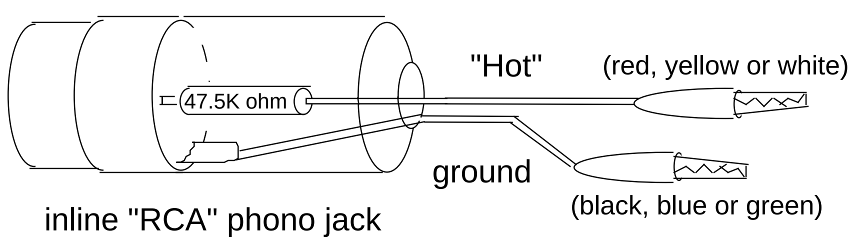
<u>Note that common multimedia FM-synthesis ("blaster" type) soundcards will not function with Liberty Audiosuite ***at all.***</u> This does not necessarily have to do with the quality of the sound cards-- some types of soundcards that are very good (for multimedia purposes) will not work with LAUD. LAUD is based on DSP collection, generation and buffer programs which require a <u>true</u> programmable DSP of the proper type as well as appropriate on-card memory. It is the exceptional versatility of these cards which allows LAUD to utilize them as a reliable, high performance audio test suite.

Many measurements will require the following additional equipment:

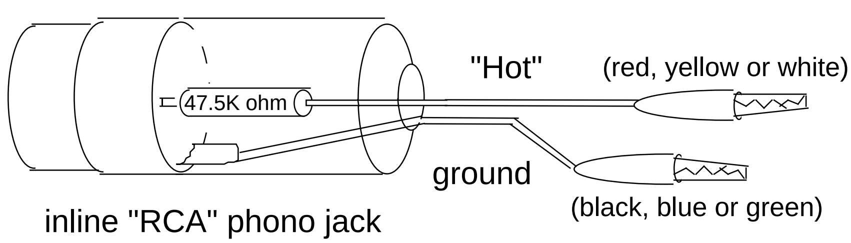
- Measurement quality microphone (for acoustical measurements). The small microphones sold for normal soundcard use are not recommended, as the response flatness is not sufficient for measurement purposes. Good microphone capsules can be obtained inexpensively or housed and calibrated units are available at reasonable cost. A LAUD system based on one of the PSA type soundcards (ECHO, Orchid Soundwave, Cardinal) will also require an external mic preamp (this should be fed into the line inputs of the DSP card, not the mic input)!
- Probes. **It is strongly recommended that the DSP card line or mic inputs not be driven directly from high-level external circuits** such as audio power amplifiers or high output signal generators. DSP card inputs are designed to handle only several volts and can be easily damaged by high AC or DC voltage levels. Because of this sensitivity to inadvertent damaging voltage levels, Liberty Instruments will bear no responsibility for equipment damaged when used with LAUD (even if used with a probe preamp), so implementation of adequate care and protection is the user's responsibility. The probe construction will differ depending on whether a Mic/Probe preamp is in use (required for PSA based systems) or if it is to connect directly to the LINE inputs of a Turtle Beach card (the "Directprobe" configuration).

An IMP probe (used with a mic/probe preamp) is merely a cliplead with a 47.5K ohm (1%) resistor in series, which feeds the center conductor of a connector for a shielded cable (which is normally terminated in RCA phono plugs). This works with a 2.21K shunt resistor which is part of the Mic/Probe Preamp.

A Directprobe (for use with Turtle Beach cards only) uses a 7.68K ohm series resistor and a 2.87K shunt resistor (across the card's line input), for a probing impedance of 10K ohms and cable operational impedance of 2.2K ohms.



IMP Type Probe (for use with Mic/Probe Preamp)



Directprobe (for use with Turtle Beach cards only)

- Power Amplifier (for speaker work). A unit with a low frequency cutoff below 10 Hz is recommended if Thiele-Small measurements will be made. Also, to avoid accidental damage from shorting amplifier outputs and for correct results in impedance measurements, **amplifiers used with LAUD must have their negative output terminal at ground potential. This means that "bridged" amplifiers, "booster" amplifiers and many tube amplifiers are NOT suitable and should not be used as test amplifiers.**

- **To prevent damage due to transients or accidental high-level tones, the power amplifier must have a level control or be connected through a control preamp or another means of manually reducing its input gain during setup and operation.**
- Two **stereo "mini-phone" to RCA plug adapters**, available at most electronic stores, are required to connect to the line inputs and line outputs of the DSP card. A third stereo adapter is required if the microphone is to be connected directly to the mic input of the card (recommended only with the Turtle Beach cards).
- Various long shielded RCA cables for connection between card, preamp (if used), probes, microphone and power amplifier.

Installing and Configuring Liberty Audiosuite

Hardware:

Your DSP card should be installed as directed in the instructions provided by its manufacturer. In general, to minimize noise pickup, it is best (but not essential) to install it in the slot farthest from the computer power supply.

If you already have a “Soundblaster” or equivalent soundcard installed in your system, it is often possible to install the ECHO DSP or Turtle Beach card along with the existing soundcard. (The ECHO DSP cannot, however, coexist with other soundcards which use either AD1848 or AD1846 CODEC chips, or chips which emulate these). In the case of the ECHO, be sure to connect the jumper on the ECHO DSP to the HIGHER PORT ADDRESS (HEX250) and DO NOT install the ECHO DRIVERS. LAUD is able to find the PSA type DSP cards at either of the two port addresses.

For PSA or ECHO cards:

Note that any multimedia software drivers or applications which might be supplied with the card are <u>not</u> needed or used by laud! If you will be using the dsp card only for measurement purposes, it is simpler to not install these drivers.

When using the MIC/PROBE preamp with a LAUD system, connect its outputs to the line (<u>not</u> mic) inputs on the back of the DSP card. This will require adapters along with cables. The DSP cards use miniature stereo phone connectors, while the MIC/PROBE preamp uses RCA connectors (because good quality RCA shielded stereo cables are the easiest and least expensive to obtain). The Channel 1 output of the MIC/PROBE Preamp should connect to the “left channel” input of the DSP card’s input socket (the tip connection of the connector). The IMP or Mitey Mike microphone connects to the CH1 mic input and the two probe cables (with 47.5K ohm resistor probes at the other ends) connect to the provided input jacks. Initially set the gain switches to the 0dB positions.

The DSP card LINE outputs can be connected into an external power amplifier (for speaker tests) or used directly as a source to drive low-level input devices. Although software control of the output level is provided, an external manual volume control for this output level is required for practical application and safety.

For Turtle Beach cards:

For Windows systems, the card’s software installation should be performed first as detailed in your Turtle Beach card’s instruction manual. The installation should be done in **NON-PLUG-N-PLAY mode**, as LAUD must be able to consistently determine the card settings (which could change under PLUG-N-PLAY). In DOS (or Windows 3.1) installations, pay special attention to the instructions for adding or modifying the “EMM386” line in the CONFIG.SYS (or CONFIG.DOS) file.

For Turtle Beach cards, LAUD will look for the file “PINCFG.INI” in the “LAUD” hard disk directory – if the file is found, LAUD will attempt to use the Turtle Beach type card; if the file is not found, LAUD will instead search for a PSA type card.

When using a Turtle Beach type card without a mic/probe preamp, the Directprobes connect via adaptors and any desired extension cables (up to about 30 feet) to the LINE input of the card. The microphone (calibrated electret type) connects through a similar adaptor/extender to the LEFT (white, or tip connection) channel of the MIC input.

The DSP card LINE outputs can be connected into an external power amplifier (for speaker tests) or used directly as a source to drive low-level input devices. Although software control of the output level is provided, an external manual volume control for this output level is required for practical application and safety.

Software Installation:

The following procedures assume that the software is to be installed to drive C: from a floppy disk. If a drive other than C: is desired as the target, the file “INSTALL.BAT” should be edited (using the Windows Notepad or DOS Edit program) to change any references from drive C: to the desired drive before installation. If the floppy disk drive is not A:, substitute the drive letter you are using in the instructions below.

Three sets of install instructions are given below, for installation from DOS, Windows 3.1 or Windows 95:

TO INSTALL LAUD version 2 FROM DOS:

Put the floppy disk in your drive (probably A:) and log to that drive:

A: [enter]

When you get the prompt for that drive, type:

INSTALL [enter]

This will make a directory on your C: drive called "LAUD" and load the program to it.

(If you are using a PSA/ECHO card, skip ahead to the section on running LAUD)

If you are using a Turtle Beach card for your LAUD system and do NOT have Windows 3.1 installed, you will not have been able to install your card’s software or “INI” files by the procedure described in its hardware manual. You must perform these installations manually.

You will first need to add or modify a line in your CONFIG.SYS file similar to:

DEVICE=C:\DOS\EMM386.EXE NOEMS X=D800-DFFF

As described in your card’s manual, the values following the “X=” in the line above reserve upper memory space. That space (which can be changed as described in your card’s manual) must not already be reserved for use by another device. Your system must be rebooted after changing the CONFIG.SYS file for the change to have any effect.

Next, find the file on your Fiji or Pinnacle installation disk called “PINCFG.EXE” and copy it to your new C:\LAUD directory.

You should then find the files on your Fiji or Pinnacle installation disk called “PIN250.INI”, “PIN260.INI” and “PIN270.INI”. Choose the file which has the number matching the jumper setting you have chosen for your card’s NON-PNP address, and copy that file to your C:\LAUD directory.

Then, you must change the name of that file (in the C:\LAUD directory) to “PINCFG.INI” using the DOS “ren” command.

Next, you must edit the file (using DOS EDIT or another simple text editor) so that it indicates the proper reserved memory area (as specified above in your CONFIG.SYS file). The section of the PINCFG.INI file which you must change is the parameter “RAMAddress” under the heading [LogicalDevice0], similar to that shown below:

```
// ld0 DSP
[LogicalDevice0]
Active=1
IOAddress0=290
IOAddress1=000
IRQNumber=11
```

RAMAddress=d800

The RAMAddress value must match the first number after the “X=” in the edited line in CONFIG.SYS.
If the IO address “290” (hex) is already in use by another device, you may have to change the value “IOAddress0” in the same section of the PINCFG.INI file to another unused value (140,150, 160, 210, 230, 280, 290, 360, or 3E0).

Lastly, you must add a line to your AUTOEXEC.BAT file as follows:
C:\LAUD\PINCFG /FC:\LAUD\PINCFG.INI
You must reboot for this change to have effect.

To <u>run</u> Liberty Audiosuite from DOS, first log to that drive and directory:
C: [enter]
CD [enter]
CD LAUD [enter]
Then type:
LAUD [enter]
You may wish to create a batch file called LAUD2.BAT, containing the command lines (given above) for running LAUD and save it to your root directory (or a directory listed in your DOS Path).

<u>TO INSTALL AND RUN LAUD Version 2 from Windows 3.1:</u>
(note: a greater amount of base memory may be required to run under Windows. Minimize the number of TSR -- “terminate and stay resident” -- programs which are being loaded by your AUTOEXEC.BAT file if there are difficulties in running LAUD from within Windows).

Put the floppy in the A: drive and from Program Manager, select RUN and then type:
A:INSTALL [Enter]
If you are using a Turtle Beach card for your LAUD system, you must first copy (not move!) the file “PINCFG.INI” from your C:\PINNACLE or C:\FIJI directory to your C:\LAUD directory (using the Program Manager) before running LAUD.

To run the program, use File Manager to find the LAUD directory on your C: drive. Double click on the LAUD directory and within it find the file LAUD.EXE. Double click on LAUD.EXE to start the program.

You can add LAUD as an iconized program item to your desktop in an existing group (or a new program group) by using the “New” menu option under “File” in the Program Manager. There is a special icon provided in your LAUD directory which can be used to identify the program item. Consult the Windows HELP file or manual if further details are required.

<u>TO INSTALL AND RUN LAUD Version 2 from Windows 95:</u>
Put the floppy in the A: drive and push the START button. Then click on RUN and type:
A:INSTALL
(If you are using a PSA/ECHO card, skip ahead to the section on running LAUD)

If you are using a Turtle Beach card for your LAUD system, you will need to install an INI file into the LAUD directory to inform Liberty Audiosuite of the proper <u>two port addresses</u> and <u>memory range</u> used by the card. This isn’t difficult to accomplish, as detailed below. Be sure to do the following <u>after</u> you have done the Turtle Beach hardware and software installation into Windows95:

The first step is to determine the values of the three parameters. The parameter values are expressed in hexadecimal, that is, they may have the characters A through F, as well as the decimal characters 0 through 9. You need to locate <u>three</u> values.
1) The start of the address range for your card's <u>Control Port</u> (one of: **250, 260, or 270**. This value must match the jumper position you have set on the Fiji or Pinnacle board, as described in your card's manual.)
2) The start of the range for the <u>DSP Port</u> (usually one of the following: **140, 150, 160, 210, 230, 280, 290, 300, 3E0, or 360**.)
3) The start of the range for the <u>DSP Ram Address</u> (one of the following: **C800, D000, D800, E000, or E7FF**.)

To find these values: <u>right</u>-click on the **"My Computer"** icon on your open desktop (using the right mouse button, rather than the usual left mouse button). Then in the displayed menu list, <u>left</u>-click on **"Properties**". Left-click on the "**Device Manager**" tab. Then locate the "**Sound, Video and Game Controllers**" group (you may need to scroll down on the list to find it). To see the specific "**Sound, Video and Game Controllers**", left-click on the (+) sign to the left of the words. You should see an entry for the "**TBS Pro Series Digital Audio**" device. Left-click on this entry to highlight the words, then left-click on the "**Properties**" button and left-click on the "**Resources**" tab. In the displayed window, you should see <u>two</u> ranges for "**Input-Output Range**". The beginning of one of these ranges will be the <u>Control Port</u> value (either 250, 260 or 270; ignore the leading zero in the values). The beginning value of the other range is the <u>DSP Port value</u>. The first value shown in "**Memory Range**" is the <u>DSP Ram Address</u> (do not use the leading three "0"s in the Memory Range value).

To make a PINCFG.INI file with the correct parameters, you will use one of the three files "PIN250.INI", "PIN260.INI" or "PIN270.INI". These can be found on your Pinnacle or Fiji Install floppy disk (or in your hard drive's "PINNACLE" directory, under the "PROGRAM" subdirectory). Use your "My Computer" icon (or Explorer) to locate these files. Choose and double click on the file with the port number (250, 260 or 270) of your card's Control Port. This should bring up the Windows95 **NOTEPAD** editor.

- In this file, the information under the heading "// ld0 DSP" should be edited, if necessary, so that "IOAddress0" equals the number for your DSP port. To edit, just left-click your mouse to position the text cursor just after the equal sign and use the "Delete" key to remove the current value. Then type in the new value.
- Similarly, the value "RAMAddress" under the same "// ld0 DSP" heading should be edited to equal the DSP Ram Address.
- No other values in this file need be changed for LAUD use.
- When the changes have been made, you then need to save this file into your LAUD directory under the name "PINCFG.INI". To do this, click on the menu item **"File**" at the top of Notepad, and then on "**Save <u>A</u>s**". In the "Save As" window, browse to your "LAUD" directory (usually on the "C" drive, unless you installed it elsewhere) and then change the "**File <u>n</u>ame**" at the bottom of the window from "PIN250.INI" (or PIN260 or PIN270) to "PINCFG.INI". Then click the "**Save**" button. Then close the NOTEPAD.

To run the program, use the Explorer to find the LAUD directory on your C: drive. Double click on the LAUD directory and within it find the file LAUD.EXE. Double click on LAUD.EXE to start the program.

<u>Program Configurations</u>:

After installation, the initial program configuration for LAUD can be easily accomplished by use of a built-in Easy Script. To run this, first start up the LAUD program. On first startup and after the title screen (which will stay until a key is pressed or the mouse is clicked), the computer will sound a tone and a message will appear at the top of the screen warning that the "STANDARD" configuration file can not be found. This file, called STANDARD.ICF, is a special file in which LAUD stores parameters about the physical configuration of the system (DSP card type, printer type, system gains, etc.) and your choice of various default start-up display, acquisition and processing parameters. The warning serves to alert you that the file needs to be created. Just press the [Enter] key to continue.

Creating the STANDARD configuration file is easy in LAUD version 2. A special Easy Script is built into the program which will prompt you for some information, direct you through any actions which might be required, and create the STANDARD configuration file (it also provides a good introduction on how to use Easy Scripts).

To start the Easy Scripts, identify the large button near the bottom of your screen. Click your mouse on this button (using the LEFT mouse button), or press the "reverse apostrophe" key (the ` key, usually at the top left corner of most keyboards). This will bring up a pop-up window with a selection of different Easy Scripts. You may make your selection by either pressing the number key (<u>not</u> the function key) associated with the desired choice or by clicking on the line containing the Easy Script selection number. Script number 1 offers brief descriptions of the Easy Scripts. The second to last selection in the first Window is for the Software Installation script which will create your STANDARD configuration file.

The same Easy script can also be used later should you want or need to change some of the primary system characteristics with your installation of LAUD. Of course, the STANDARD configuration file can also be updated without the script by using normal menu options within LAUD. If you are using a Turtle Beach card, an ECHO DSP/mod, or another extended low frequency card, you may skip the next section.

<u>**Creating and Declaring BAL Data**</u>

NOTE: <u>If you are using LAUD with a Turtle Beach card, the Echo DSP/mod</u> or any card which has been modified for extended low frequency performance, <u>the BAL data and process should **NOT** be used</u> -- just skip this section. **Bal** is intended to compensate, in software, for differing rates of low-frequency rolloff which occur within the measurement band in some DSP cards sold for multimedia soundcard use. By not using the BAL facility, you can achieve faster program operation and lower memory utilization.

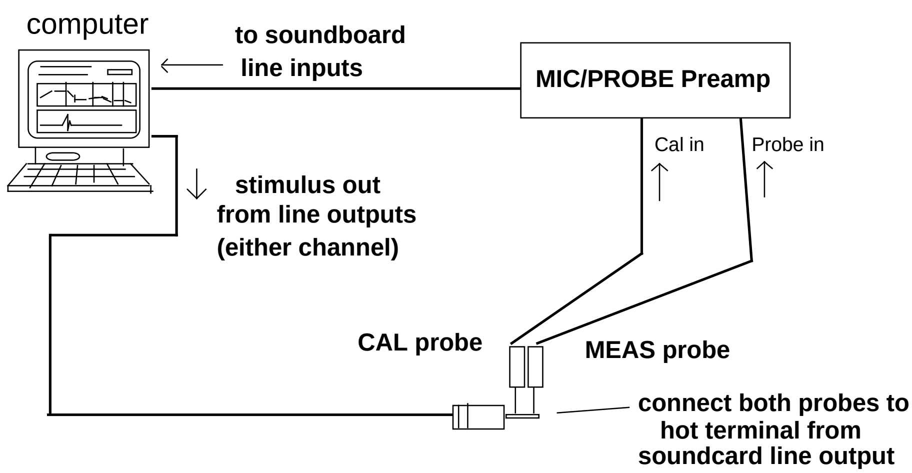
As described elsewhere, Bal (balance) data is a correction file for the frequency domain differences between the two channels at lower frequencies. A Bal data set, if used, should be made when LAUD is initially installed and need only be changed if you change probes, your DSP card or your mic/probe preamp.

To generate this data set, connect both probes to the same output of your DSP card (to the same channel). The Line outputs are preferred. Press the [%] key so that Dual Channel mode is selected. Go to the SINE instrument and select [Acquire Freqs] and then select [Log_f]. Temporarily choose a first frequency of 200 Hz, a last frequency of 400 Hz and Num_points equal to about 100. (NOTE: the menus are navigated by either clicking on the designated menu words in order, or by pressing the key which is capitalized or highlighted in each word. The frequency range parameters will be prompted for -- just type in each number and press [Enter]). See the connection diagram following this text.

Be sure that the word “PROBE” is displayed in the top-of-screen box marked “4:INPUT”. IF not, press [F4] until it is. Also, if the box marked “9:DLAY” does not show “0.000”, use [F9] to set it to zero.

Now go to [* Acquire Set_levs Auto_adj], and select “Both_match”. Now go back to {* Acquire Freqs] and change the first frequency to 2 Hz. Then go to the [* Acquire Modes] menu and set [Resp_mode] to [Optimized] and at 5 cycles. Also ensure that the Gating parameter under [* Acquire Modes Resp_mode] is set to OFF.

Press [* Acquire sWeep Freq_response] and take the time to read through your documentation while Liberty Audiosuite compares your two channels at low frequencies. After the process has completed, go to the [* System Z_set] menu and choose [Set_bal], and answer affirmative to the “are you sure?” question. The data will be automatically saved, ready for future use.



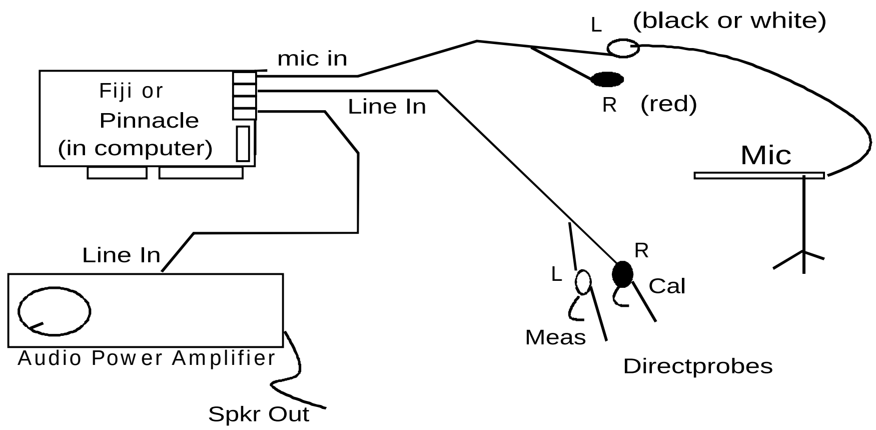
“Bal” file creation equipment setup

(SEE NOTE ABOUT APPLICABLE DSP CARDS)

<u>Using the Turtle Beach Pinnacle™ or Fiji™ without a Mic/Probe Preamp (“Directprobe”):</u>

When LAUD is used with the ECHO or other PSA type soundcards, the Mic/Probe Preamp is needed for most practical measurements. This is because the microphone preamp in those cards does not generally have a sufficiently extended flat response range for measurement purposes. That card design also lacks sufficient gain or adjustment range on the Line inputs to allow resistor probes to adequately handle a wide enough range of input levels with a direct connection. The Mic/Probe Preamp therefore provides the needed flat gain mic stage as well as an interface to allow resistor probes to be used safely and switchable attenuator steps to greatly extend the input range.

When LAUD is used with the Multisound Pinnacle™ or Multisound Fiji™ cards, the system can be operated without a Mic/Probe preamp. This is practical because the mic and line input hardware of these cards have very low noise as well as a wider adjustment range. This type of operation, in which LAUD operates with the Pinnacle™ or Fiji™ cards but without an external Mic/Probe Preamp, is referred to in the documentation and Help files as a **"Directprobe"** configuration. ***Note that much of this manual will describe only (or primarily) the Mic/Probe Preamp-based configurations. If you are using the Directprobe configuration, you may ignore references herein to external Mic/Probe switches and most External Gain or Attenuation settings. Note that the probes used with this Directprobe connection are not the same as those used with a Mic/Probe Preamp based system. Rather than connecting the microphone and probes through the Mic/Probe Preamp, these devices will instead connect directly to the Turtle Beach card as shown below:***



Note: In most cases, the ground leads of the probes (black, green or blue) need not and SHOULD not be connected at all.

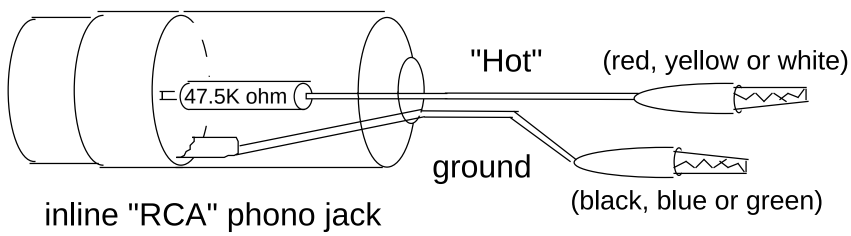
Directprobe operation of the Multisound Pinnacle™ or Multisound Fiji™ cards is generally simpler than is operation with the Mic/Probe Preamp. There are fewer hardware switches and connections to worry about and one less piece of equipment to purchase or maintain. Measured distortion residuals are considerably lower with the Directprobe connection (it avoids the inevitable distortion which would otherwise be added by the Mic/Probe Preamp's circuitry). Also, there is less need for keeping the LAUD software informed about external settings. For example, when using the Mic/Probe Preamp, you must make certain that the "INPUT" ([F4]) button of LAUD matches the switch setting of the Preamp. But with Directprobe operation this occurs automatically because the LAUD software is able to directly switch between the Microphone and the Line (probe) hardware.

There are some special considerations, however, when operating LAUD in the "Directprobe" connection:

- **The microphone correction data** (its "calibration" file) may not be correct for the bias conditions provided with direct mic connection to the Turtle Beach™ cards. The correction data provided for capsules intended for use with the Mic/Probe Preamp assume a source bias of 2.5Volts, supplied via a resistor of 2.2k ohms. The Turtle Beach cards provide a source bias of 5.0Volts, supplied through a resistor of 4.7k ohms. Typically this will result in the reference sensitivity being 3.4dB higher (a voltage ratio of 1.48x) when the microphone is powered directly from the card rather than from a Mic/Probe Preamp. If your microphone's calibration was done previously only for the Mic/Probe Preamp, a good compensation can be provided by changing the first line in its calibration data file so that the number provided there (the "mV/PA" value) is 1.48 times its original value. For example, if the sensitivity value was 6.30 mV/Pa, it should be changed to 9.32 mV/Pa. Alternately, the capsule can often be recalibrated by its supplier for the new conditions at a nominal cost.

- The resistor probes used with the Directprobe connection are of different construction than are the "IMP" style probes used with the Mic/Probe Preamp. The IMP probes used a single 47.5k ohm resistor in series with the hot lead (this worked against the 2.2k load resistance of the Mic/Probe Preamp's input to provide attenuation and voltage protection). This provided protection to terminal voltages up to 100V peak, and linear operation up to 40V peak.

The Directprobe probes, however, contain both a series resistor of 7.68K ohms and a shunt resistor of 2.87k ohm (the "1x" directprobe). This provides DSP card protection for terminal voltages of up to 40Vpeak and linear operation to 20V peak. ***Do NOT use these probes at points which may potentially exhibit voltages above 40V peak. Liberty Instruments, Inc. shall NOT be responsible for damage to equipment due to excessive voltages at probed points .***



Directprobe, 1x type (for use with Turtle Beach cards only)

For higher voltages, a "10x" Directprobe can also be purchased or constructed. This, however, is unlikely to be needed by most users. A 10x Directprobe provides 20dB more attenuation and is usable to levels of approximately **100Vpeak, (use above this level is not recommended for safety reasons)**. A 10x Directprobe uses a series resistor of 9.76k ohms and a shunt resistor of 232 ohms.

<u>Using the Turtle Beach Pinnacle™ or Fiji™ WITH a Mic/Probe Preamp:</u>
Use of LAUD *with* the Turtle Beach DSP cards and *with* the Mic/Probe Preamp is done the same as is done when using the ECHO/PSA cards. The Mic/Probe Preamp's outputs connect to the Line In jack of the DSP card (Channel 1 = left channel = "tip connection" = white or black connector). Use of the Mic/Probe Preamp provides some advantages:
- The microphone input sensitivity is considerably increased
- The input voltage handling range is greater (without changing to different kinds of probes)
- The attenuator (GAIN) switches on the Mic/Probe Preamp provide greater versatility
- Cable lengths can be longer (the low impedances of the DSP card's line output and the Mic/Probe Preamp's output allow cable lengths of up to several hundred feet, if required).

<u>Using the Turtle Beach Pinnacle™ or Fiji™ both ways:</u>
You can easily change back and forth, as desired, between using these cards with or without the Mic/Probe Preamp. This way, you can use the Directprobe connection for best convenience and lowest distortion residuals for most uses, but add the Mic/Probe

Preamp when you need the extra microphone sensitivity or must use longer cable runs between the computer and the measurement area. To do this, use the following process:

1. After hardware and software installation of the card and LAUD software, use the Installation **Easy Script** to configure your system for the Turtle Beach card as connected without the Mic/Probe Preamp. Be sure to have 1x “Directprobes” connected to the card, and to use a microphone correction data set which is appropriate for direct connection to the card (5V bias, 4.7K feed). Complete the setup, all the way through where it modifies your “STANDARD” configuration file. This is very easy and takes only a few minutes.
2. Then, rerun the the Installation **Easy Script** to configure the system for the Turtle Beach card with the Mic/Probe Preamp. Be sure to use the microphone correction data, in this case, which is correct for use with the Mic/Probe Preamp (2.5V bias, 2.2k feed). However, at the last step in the Script, instead of allowing the “STANDARD.ICF” file to be overwritten, instead choose “**NOT ACCEPT and leave the configuration file unchanged**”. Then immediately use the menu selections: [* System Config_file Save MPP] and press the [Enter] key. To do this, as will be explained in the chapter on “Shared System Characteristics”, only the following keys should be pressed, in the following order:

***SCSMPP**

and then press the **[Enter]** key. If you are asked if the existing file “MPP” should be overwritten, press the **y** key (“yes”) in response.

When you start up LAUD, it will normally come up configured for use WITHOUT the Mic/Probe Preamp, as saved in the “STANDARD” configuration file. If you wish to use it on this occasion with the M/PP, just use the menu selections [* System Config_file Use MPP] and press the [Enter] key. This will then restore the configuration (mic correction file, gain settings, etc.) for using LAUD with the M/PP. If you later wish to make changes to this configuration, be sure to save the Config_file under the name “MPP”, rather than “STANDARD” (because STANDARD is being used for the more commonly used “Directprobe” configuration).

Liberty Audiosuite Scripts

An Audiosuite script is a text file program, or list of instructions, which can operate the LAUD system as if an operator were present directing the measurement process. In its simplest form, the script can merely be the keystrokes the expert might press in the course of performing a desired operation. In more complex uses, the script can be programmed to provide text screens for a non-expert to read. The programmer can offer "radio button" check box choices to allow on-demand variations in the measurement process. He can program conditional alternatives which can alter program flow depending on the values of system parameters or of measured data. The script can also cause bitmap images (such as of equipment connections for a desired measurement setup) to pop-up on the screen to give a diagram, company logo, or illustration. Examples of some relatively complex scripts can be found in the built-in Easy Scripts, the files for which can be found in the directory LAUD\SCRIPTS\EASY. All script files (except the Easy Script main menu file "EASY.MNU", in the LAUD base directory) have the filename extension ".SCR".

Before attempting to write a script, it is recommended that new users become familiar with the LAUD system menu operations (or at least those which will be used within the script) by manually conducting the desired measurement. Descriptions of the basic operation of the LAUD menu system and special key operations begins in the next section "Shared Instrument Characteristics".

This scripting system is relatively easy both to use and to program. While not a full programming language like C, Pascal or Basic, LAUD Scripts can be very powerful due to their ability to control the LAUD2 Audio Analyzer system. The script can be thought of as a layer of control which exists above the LAUD2 menuing system, and which can serve as an intelligent intermediary between the system and a user. Note that scripts can not be used to control the scope instrument, nor can they be launched from it.

The most obvious use of the Script processor is to run the built-in "Easy Scripts", which can be launched by either pressing the (`) key (which is usually at the top left on most keyboards) or by clicking on the large button which is on the lower portion of the main screen of the MLS, SINE, SPECAN and DISTAN instruments. Easy Scripts are built-in scripts provided with LAUD2 to allow users to make meaningful measurements without large amounts of training or hours of studying the users' manual

The Idea Behind Scripts

Liberty Audiosuite, in its continued development, had begun to become susceptible to a common fault of large programs known as "feature bloat". The desire to make the system as powerful and as versatile as possible, as well as to provide most of the features which users have asked for, tends to make the software more and more difficult for new or occasional users to operate, if only because of the sheer number of choices he or she must make. Some of the modern word processors are subject to this problem: they are capable of doing nearly anything you might want of a word processor, but a new user will often become extremely frustrated at the seemingly crazy things than might happen as he tries to type a simple memo.

Scripts provide the needed link between the very comprehensive LAUD system facilities and the need for simple operation during often-performed processes. With Script processing, Liberty Audiosuite is well suited to beginners and minimally trained personnel as well as to engineers and researchers who need control of system details.

Additionally, scripts provide the needed program flexibility for realistic use in production control applications, enabling the engineer to prepare his test software so it can react to numerous different occurrences while minimizing the training required for the personnel who will assist the software in performance of the production test.

Lastly, a script is a useful medium in which design or research engineers can create, review, and edit complex measurement procedures. The self-documenting nature of the text format used in the scripts and the ability to pop up messages and pictures by which a script can “explain itself” make for a much better controlled and understood interaction between the programmer and the measurement system.

Creating Scripts

There are four main steps in making a script:

1. **Writing the text file**: The file can be created on any editor or word processor capable of saving in ASCII format. Alternately, to save typing, an existing script which shares some similar desired functions to the one you wish to create, can be loaded into your text editor, renamed, and then modified. Be sure to save the script using a name with an extension of “.SCR”. The script should normally be saved in a separate subdirectory (which you should create using normal DOS or Windows commands) under LAUD\SCRIPTS\, but it can also be saved in one of the existing directories under that structure.

A particularly convenient method of developing scripts which allows them to be interactively tested is to operate within Windows, running LAUD at the same time as Windows’ “Notepad” text editor. You can write or edit the script in Notepad, then use the Windows “Alt-Tab” key combination to switch quickly to LAUD in order to test your script as it is being developed. One thing to be careful of, however, is that the script should terminate (or be ended, using the [End] key) before switching back to Notepad with the “Alt-Tab” combination, or Windows will give you a message about “share” errors and will not let you save the next edit you make in Notepad until your return to LAUD and first end the script.

2. **Create any needed Bitmap pictures and/ or LAUD configuration files** : The bitmap pictures can also be made by modifying existing .BMP files in the Easy or other existing script directories. If you are creating them from scratch, be sure that you make and save them as Monochrome (also called “two color” or black-and-white) bitmaps, which is NOT the same a <u>gray-scaled</u> black-and-white bitmap. Within Windows, these bitmaps can be easily created using the built-in Paintbrush utility, which can also be operated concurrently with LAUD and the Notepad during script development.

Configuration files are often loaded at the beginning of a script in order to set various parameters such as sample rate, display frequency limits, scales, etc. to a desired state before further operations. The configuration files must be created from within LAUD. Simply use the standard LAUD menus or buttons to set the system parameters to the state required within your script. Then save the configuration file to the directory of the script you are developing, using commands under the script directory.

Configuration files loaded by scripts will always treated as being the “hardware independent” type, meaning that hardware specific parameters such as DSP card type, preamp gain, mic data file name, and input and output gains will not be altered by loading the configuration file. This allows scripts to be shared among users who have different hardware. Of course, you will have to explicitly set things like output gain and input gain (or use Automatic level setting) within your script to get these to the desired state. Configuration files loaded from within a script must always come from the same directory within which the script resides. This is done so that a script can be moved from one system to another in its entirety by copying the contents of that subdirectory.

3. **Move or save the Script, Bitmap and Configuration files to a Script Directory**: If you will also be using existing bitmap pictures (extension ".BMP") or LAUD configurations (extension ".ICF") associated with an original script you are using as a starting point, be sure to copy them as well from the originating script directory.

4. **Test and develop the Script**: Of course a script, like any program, will probably not work exactly the way you want it to the first time you try it, so the above steps may have to be repeated until the desired result is achieved. For debugging tips, see the discussion under that heading in the file "EXAMPLE.SCR" (the listing for this file is provided for convenience at the end of this section). A special "single-line-step" mode can be coded into your script as an aid in determining what may be happening in difficult cases.

Custom scripts are launched using menu options in the "Cntrl" section of the main menus of the MLS, SINE, SPEC_AN and DIST_AN instruments. First the Script subdirectory must be selected (from the supplied list), and then the script itself is selected, after which it immediately starts. It is perfectly permissible for one script to start another; however the original script is terminated at the beginning of the new one.

If you need to print a line of text out to a printer (such as measured data for research or production control documentation purposes), a special feature is provided by the [Cntrl pRint] option which allows a single text line or linefeed to be printed.

If you are distributing a script to another LAUD version 2 user, most compression programs (such as PKWARE's "PKZIP") are capable of zipping files along with their directory structure, so that they can be installed in the proper locations for immediate execution.

<u>Executing Scripts</u> : Easy Scripts can be launched via the [`] or the large "press for Easy Scripts" button and by using the selection script which is then presented. Custom scripts can be launched using the [Cntrl sCripts Custom] menu, first by selecting the proper script directory and then the desired script.

During execution, a script can be interrupted or terminated by using the **[End]** key (and possibly also the [Esc] key). Scripts stopped in this way can usually be restarted from where it was interrupted by pressing the key combination **[Shift-Tab].**

<u>The Script Language</u>

The best source of specific information on the script language is in the file "LAUD\SCRIPTS\DEMOS\EXAMPLE.SCRIPT", which is also listed at the end of this section.

An important point to keep in mind when writing scripts is that the script processor is <u>case sensitive</u>. This means, for instance, that the label {here} is different from the label {Here} and that the command [j{label}] will behave differently than will [J{label}] (which, in fact, will do nothing).

There are several types of codes which may be included in a script. These are:

-- COMMENTS, which are lines in the script file which begin with the quote (") mark as the first non-blank character. These lines have no effect on the script execution other than to prevent execution of anything further beyond the quote on that line. A comment line is treated the same as a blank line. Comments can also be at the end of most other types of lines (message lines and variable assignment lines excepted), in which case the rest of the line after the quote will be ignored by the script processor. Use comments to make the script more understandable to others or to yourself should you need to modify it at a later date.

-- LABELS. These are lines which begin with the open curly-bracket ({) in the leftmost column. All text up to the close curly-bracket (}) will be taken as the label name, which you can define arbitrarily. For instance, a line which contains only:
{someroutine}
is a label for a section of the script called "someroutine". Labels are used for program control, allowing jump and conditional statements (similar in operation to Basic and Pascal "Goto" statements) to shift control to that section of the script.

-- SPECIAL COMMANDS. These are commands, not part of the normal LAUD menu operation, which are defined particularly for use within scripts. They begin with an open "square bracket":
[and end in a close "square bracket": **]** . In some cases, they are used for key operations used by LAUD which cannot be normally entered on an ASCII text editor. For example, the function keys F1 through F9 are entered by the codes **[F1]** to **[F9]** and the Alt and Shift function keys can be entered via **[A1] to [A9]** and **[S1] to [S9]** respectively. Similarly, the codes **[AM], [AS], [AO], [AP] and [AD]** are used to transfer to the MLS, SINE, OSCOPE/GEN, SPEC_AN and DIST_AN instruments, respectively. In general, it is wise to avoid using the cyclic function buttons which rotate parameter values round-robin style (such as [F2], or "SIZE), because their end result depends on their values before execution -- instead use the more explicit versions provided under the "Cntrl setup2" menu.

Other special commands include:
[j{???}] which is a jump command, transferring script control to the line after the label represented by ??? in the current script. If the label does not exist, the script will terminate, restoring the user to the normal LAUD operation. There should be no comments following these codes, which should be the only code on that line.

[=w=???????] math or assignment operations which assign a numerical or text value to one of the twenty-three variables "a" through "k" and "A" through "L" (represented by the character "w", here. The portion represented by the ??????? characters varies depending on the command, and is detailed in the EXAMPLE.SCR listing given at the end of this section. These types of codes should always be put first on a given line (or be the only code on the line) because they will be executed first before any other commands.

[~w&&&&&&{???}] conditional jumps to the label depending on the numerical relationship between the variable and the characters represented by &&&&&. These should be on lines by themselves, without any comments following.

[@w] variable readout codes which print out the value of the variable represented by the "w" into either the LAUD command line (such as for answering one of LAUD's prompts for a value) or for printing within a "quick message" line (described below).

Other constructs, like **[T??],** which defines the time-out period in seconds to wait for a user response after a message window and **[y??],** which defines how many pixels from the top the next message window should be displayed. For debugging, the **[1]** code turns single-step mode on and **[0]** turns it back off.

-- POP-UP “QUICK MESSAGE” LINES, each of which starts with a vertical line: **|** . After the end of the displayed text, another vertical line should follow. If the text line is to mark a radio button option (i.e., a line which has a number displayed for a user option), the vertical line should be followed by the jump label (within curly brackets) to which control should be changed if this selection is chosen by the user. After the second vertical line or label name, a linefeed should follow (i.e., press [Enter] in your text editor). All the quick message lines in an unbroken sequence (uninterrupted by a non-message line) will be shown together. If none of the lines have a radio button option, all message lines will be centered horizontally; otherwise they will be left justified.

-- POP-UP BITMAP PICTURE FILE DECLARATIONS which must be on their own lines and are composed of:
A period followed by a space, then the bitmap file name (complete with the dot and extension).
If desired, values for a horizontal position (in pixels from the left of the screen) and vertical position (pixels from top) can be specified, preceded by spaces, after the bitmap file name. The picture will be shown along with the next quick message that appears. If you wish it displayed with later messages, you must declare it again before each one. If no position values are given, the script processor will attempt to evenly space the picture and message text, horizontally centered, on the screen.

-- SUBSITUTION CODES, used for non-alphanumeric key inputs which are inconvenient or impossible with a text editor. One example is the #??? type code, which is used for odd character input such as a backspace, an End key, or the like. The pound-sign ‘#’ must be followed by exactly three characters (described in the “example” script).
Two important substitution codes are ‘<‘ and the ‘>‘ characters, which are used in place of the escape key and the enter key, respectively. These two character will have their usual meanings of “greater than” and “less than” when used within a math operation special command.
Laud keystrokes with printable characters, such as the asterisk (used to move to the top-of-menu), use the character itself.

-- LAUD COMMAND CODES. These are the menu words used in normal Laud operation. Lower case letters, the underscore, and spaces are ignored in these, so that the entire keyword can be used to improve readability. However, beware of some LAUD keywords, such as “Dual2” which contain both a capitalized letter and a number-- be sure to include only one of these.

The active characters are sent directly to the LAUD input command stream, so that the commands are executed as if someone were there typing the active characters normally onto the keyboard during LAUD operation. The programmer must be sure that each command occurs in the appropriate context, that is, that the commands are appropriate for the instrument and menu position which are current at the time of execution. It is a good idea to start each new command chain after a label (which might be a target for a jump), or where the current position may be uncertain, with an asterisk (*) to start at a known position at the top-of-menu, followed by all the sequence of menu keywords toward the desired operation.

One potential trouble spot is the type of command which might cause LAUD to issue an “are you sure?” warning. An example is any kind of file save, in which you are normally asked for validation should a file with that name already exist. To avoid this problem, LAUD has been adapted so that this “Are your sure” question will not occur while running a script. While coding your commands, leave the “Yes” out of the coding and be sure you back up or don’t need any data files which may be overwritten by the script!

The following is a listing of the file “Example.script”, (as of May, 1996), which contains an active documentation of the script language in its comments and examples. The script can be run, if you wish to see how the commands behave, although it doesn’t do anything useful other than demonstrate commands. Other script examples to which you may wish to refer can be found in the Easy Scripts, which are in the subdirectory “\LAUD\SCRIPTS\EASY\”, and can be read or printed using any text editor.

```
[AM]
"switch to mls instrument; next is a label:
{beginscr}
"****************INTRODUCTION TO SCRIPTS************************
"This an example script file, with documentation given in the
"  comment lines. Max length of each script line is 72 characters,
"  except comment lines. Scripts are a way to program Liberty
"  Audiosuite for simplified end user operation.
" *** You can program series of menu keywords which will be executed in
"  order.
" *** You can display bitmap pictures and message windows
" *** You can allow the user to select options which will control the
"  flow of the script program.
" *** You can ask the user to enter values or names, and can even
"  do some simple math operations on the values and have program
"  flow be altered by the values.

"*****************CHARACTERS AND THEIR EFFECTS****************
"A comment line is any line beginning with a QUOTE (") in its first
" non-blank character. These lines will be ignored during execution.

"Lower case letters (except in [ ] constructions, labels and variables)
"  and spaces are also ignored -- this means you can type entire keywords
"  in executable lines, but be sure there is only one capital or
"  numerical character per keyword!

"The '>' character is used for "Enter";  the '<' is "Esc"
"  [t] is a Tab,  [d] is the delete/backspace.

"Function keys can be entered by using square brackets, also.
"  For example, [F3] means function key F3 -- be sure the F is
"  capitalized.  Use [S3] for Shift-F3, [C3] for Ctrl-F3, and
"  [A3] for Alt-F3 (i.e., User Macro #3). Also, [AM] is Alt-M,
"  [AS] is Alt-S, [AO] is Alt-O, [AP] is Alt-P and [AD] is Alt-D.
"  (instrument change keys).

"The arrow keys, Home, End, etc., are more obscure.
" They are entered as a '#', followed by three numerical characters.
" (ALL THREE characters must be there!}.
" The codes are as follows:
"  leftarrow is #075, rightarrow is #077, End is #079,
"  ctrl-rightarrow is #116, ctrl-leftarrow is #115, Home is #071.

"*****************EXECUTABLE SCRIPT LINES******************************
```

```
"Below is an executable line; Just keywords and keystrokes used in LAUD,
"  in CAPITALS or numbers, except for the exceptions mentioned.  Note that
"  in executable lines, lower-case letters, blanks, and the _ character
"  do nothing except make it easier to read.  Watch out for some LAUD
"  keywords which have both an upper-case letter and a number: be sure to
"  have only ONE of these in your executable string (for example, use
"  'dual2' or 'Dual', NOT 'Dual2').

File Retrieve Freq_response FOAM> Display Format Scale

"Note that only the characters FRFFOAM>DFS really do anything in
" that line!  The rest are for readability.

"Execution will continue in the succeeding lines.  You must make sure
" that the preceding line left you in the proper menu position for
" your commands!  And beware that errors during execution (like
" trying to read-in a file which doesn't exist) will cause following
" script commands to be out of context.  Use an * if there might be
" an error to restore context to "top of menu".
{dbs}
dBperdiv 10>

dBperdiv 1> B2> B3> B4> B5> B10>
"You can continue with commands appropriate to your current menu position,
" as shown above.

"************************BITMAP PICTURE FILES*******************
"Begin a line with . (period) to define the next bitmap picture file,
"  followed by a space, then by the horzontal pixel position from left,
"  then space, then the vertical pixel position from top (positions
"  are for top left corner of picture).  The bitmap MUST be a two-color
"  (black and white) bitmap.
"For example-- put at horizontal 350 pixels across, vertical 20 pixels down
. tst.bmp 350 20
"The bitmap will not be shown until a "quick message" is encountered in
" the script.  It will then need to be redefined (using the period,
" again) before each "quick message" is popped-up.


"*****************QUICK MESSAGES*************************************
"A "quick message" command lets you display a pop-up message. Just
" begin each text line with a | as the first non-blank character.
" All such lines, up to a 'prompt line' or else the next line which
" doesn't start with |, will be shown together.
"End each message line also with |.  If a bitmap file is defined,
" it will also be shown.

. tst.bmp 0 0
|This is a quick-message!|
|These lines will be grouped together up to the next line which|
|isn't a quick-message type.|

"You can change the timeout given for the user to respond by the
" code [Txx], where xx is the number of seconds to wait; 0 means forever.
[T02]
"Use the code [yxxx] (where xxx is a number) to position the next quick msg
```

```
" someplace other than the centered position on the screen.  The number is
" pixels from the top.  This value will be used only for the next displayed
" quick-message.  If the number is too big, the msg will be at screen bottom.
[y400]
|These lines are in their own window, and lower on the screen|
|The following functions are in slow-demo mode:|

"************SLOW/FAST MODES**********************************************
" [s] is a code for 'slow demo-display': [s] makes each step wait 1 second:
*FRF
[s]
FOAM>

"the code [f] returns you back to fast(normal) execute mode:
[f]
|Now, back to fast normal execution|
*FRF FOAM>

"By the way, if a script is running and you want to abort it, tap the
"  [End] key 2 or 3 times.  That should bring it all to a halt.


"*****************LABELS AND JUMPS****************************************
"You can also execute a "jump" within a script.  A line which starts with
" a { is a label line.  That means the characters within the {} are a label
" which can be used as a jump target during script execution. Labels can be
" 15 characters in length, maximum.
"Farther down this file, you'll find a line which starts with {righthere}.
" That is a 'lable line'.  The line which has [j{righthere}] will cause
" execution to JUMP to the line right after the label {righthere}.
"Lines between will be ignored.  The jump can also be to a previous line
" in the file, so you can make looping scripts.
"Don't put a jump as the last line in a script.  If necessary,
" put a comment line afterwards, to insure proper execution.

[j{righthere}]

{nothere}  " you can put comments after the brackets on label lines...
"This line (which would make a beep) will be skipped:
* pCntrl beeP  "you can also end LAUD menu entry lines with a comment

{righthere}
"This line (which will load an impedance file) will execute:
* File Retrieve Z>

"The usefulness of the labels comes from another variation on the
" quick-messages.  If you put a label name (inside its {curly brackets})
" right after a line in a quick message, a user selection button will be
" displayed with that line.  If the user clicks that button (or presses
" its number key), execution will jump to that label in the script.
" If there are labels tied to quick message lines, the message will NOT
" be centered in the window, as it normally would be. Max number of option
" labels per quick-message is 9.

{thebuttons}
"First, set the timeout to 0 (no timeout: just wait)
```

```
[T00]
"Now the quick menu with selections and jumps:
|        Select one of these options:|
| |
|Go to the label 'label1'|{label1}
|Go to the label 'label2'|{label2}
|Go to the label 'continue', and stop showing this screen!|{continue}

"a timeout, if enabled, will make execution go to this line:
* pCntrl beeP
| Hey! wakeup!!! |
[j{continue}]

{label1}
|You pressed the 1 button|
[j{thebuttons}]

{label2}
|You pressed the 2 button|
[j{thebuttons}]

{continue}
|This is the message right after 'continue'|

"***********VARIABLES*********************************************
"There are 23 'variables' which can be assigned in the script or via
" inputs from the user.  For instance, here's how to prompt for
" variable 'A': make the first character within a quickmessage be '?':
|Enter a value for the variable A (positive integer or text)|
|?A|

"For variable 'a', a "real valued" variable:
|Enter the real-type value 'a':|
|?a|
"The current value of the variable is first displayed, and if the user
" presses [Enter], the value will be left unchanged. Otherwise, the
" user can type in a new value.
"The user gets a second chance to check his entry: if he then presses
" [Enter] without changing the value, the entry is accepted.
"If an invalid input is given by the user, variable 'a' will not be
" changed and the computer will beep.


"Note that there should NOT be labels on ANY of the quickmessage lines
" in a window in which a user is asked for a variable value! The
" variable request will always be taken as the last line of the
" quickmessage window.

"There are three types of variable: real (floating point variables), called
" 'a' through 'k' (lower case), and the integer/text variables 'A' through
" 'K' (Upper Case). An additional "upper case" variable, 'L', is text only.
" The real types can hold positive or negative whole and/or fraction values.
" These can only be properly typed back into the command stream when the
" number of characters in the number representation is 8 or less,
" including a decimal point and any minus sign.  If the magnitude of the
" value is over 32767, then it is divided by 1000 and shown with a k in
```

" the last place to mean "thousands". Note that precision will be lost
" in user data entry and in typing out to the command stream -- these
" variables are not meant to have the power of Pascal or Fortran, only
" to let you vary display scaling and do some conditional operations!

"The data type of the integer/text variables is unusual. The first 11 of
" these variables (A through K) can be up to 15 characters long.
" (The L variable can be up to 50 characters long for use in specifying
" full DOS paths and file names). If an integer/text variable has a
" numerical value, it should be a positive whole number or zero. If an
" integer/text variable is assigned a value from a real variable, the
" number will be rounded first. You may get a negative value into an
" integer text variable, but operations using it will give incorrect
" results (or possible program malfunction). For proper operation, the
" numerical value in an integer/text variable should be less than
" 65,536.
"
"The L variable is text-only. Any assignment from another variable to
" L will cause L to hold the character representation of the assigning
" variable's value, including decimal points from real-type variables.

"You can have the script type out a variable (B, for example) into the
" LAUD input stream by using [@B]. The same format is used for typing out
" real variables, i.e. [@b]. The variables b and B, although cases of the
" same letter, are independent of each other. You must make sure to have
" your script type out ONLY whole number variables when LAUD is expecting
" such (such as for vertical scale "per division" values), or an error will
" result to which your script may not be able to respond!
"Note that all characters of a variable will go into
" the command stream, unlike the executable lines of a script, for which
" lower case letters, spaces, and the '_' are not sent.

"You can also display variable values within a quickmessage. Just put
" the code: [@a] to show the value for real variable 'a', [@A] to
" show the value for integer/string variable 'A', etc. Remember that
" these values may consist of up to 15 characters long: be sure to leave
" room in the line for the value to fit!

"You can assign the variables directly via the script. The following will
" assign the value 'THIS VALUE' to variable E:
[=E=!THIS VALUE]

"The above makes E have a non-numerical value. The following gives it a
" whole number value:
[=E=!375]

"Note that assignments always start with '=', followed by the variable
" name (a letter), then another '='. The exclamation point (!)
" causes the rest of the text (up to the ']') to be taken literally
" (to differentiate the text from a variable name) in an assignment
" or expression. The type of a literal assignment must be proper
" for predictable script operation; be sure a valid real number is being
" expressed if you are assigning to a real variable!

"IMPORTANT!: These assignment codes should NOT follow any non-assignment
" or math operations on the same line, as they may be executed out of order.

```
" For example, the line:
"         * Display Format Scale dBperdiv > [=a=x]
" would do the [=a=x] assignment first, which is not obvious from reading
" the code.  It is safest to put these assignments and math operations on
" separate lines.

"************************MATH OPERATIONS********************
"Some tips about math operations:
"The script math operations, although simple, provide great power to
" the LAUD system, making it extremely customizable.
"In General: Please note that this script language is a rather simple
" system.  There are no error messages and the number handling is
" limited.  If there is an error in your script program, execution
" can give strange results (usually due to unintended characters
" being typed by the script into the command stream).  It is best to
" write your script a little at a time, verifying operation at each
" new step so that errors can be more easily found!
"For integers (capitalized variables):
" Only positive integers and 0 are allowed.
" Dividing by zero will give the result: 999999.  Results larger than
" 65535 will not calculate correctly.  Any negative results from
" subtraction will result in a value of 0.
"For reals (lower-case variables):
" Beware of the display limits mentioned earlier.  Also avoid
" multiplying very large numbers or dividing them by very small
" values, as there is only minimal overflow checking provided.
"Math operations can also involve text values for the variables. The
" first operand can be a text/numerical type which has a number value either
" at its end or before a period (.) within the text (such as a
" DOS file name, for example example: FILE17 or DATA1.DAT).
" This only works for the variables A through K. (Number math can't be done
" with variable L).  When using text variables in a math expression,
" operations + and - can be performed (with the second value being an
" integer) and the result will be the same text string, but with the
" number part altered by the math.  This is provided so a numbered
" sequence of data files can be generated and saved using a script.
" For example: FILE35 + 1 = FILE36. (This would be expressed using
" variables and maybe a '!' immediate, such as [=A=A+!1]).
" Only the FIRST operand of a math expression can be a text value.
"
"Legal Math operations include + - * /.  Two special operations (; and x)
" are also provided for changing between scalar magnitude and dB notation.
" To load variable b with the voltage decibel equivalent of variable
" 'e', use the format:  [=b=e;!20].  This takes the base10 logarithm
" of e, multiplies it by 20 and puts the result in b.  The opposite
" operation is [=b=ex!20], which divides variable e by 20 and takes 10
" to that power, then puts the result in b.  For instance, if c equals
" 2, then after [=c=c;!10], c will equal 3.010... (which means double
" the power when expressed in dB). If c equals -6 then after [=c=cx!20],
" c will be 0.5011..  (-6dB is half voltage or pressure).
"
"The L text variable CAN combine variables or text using the comma (,)
" as an operation.  The format is: [=L=  , followed by a variable name
" (any of a..k, A..K, L, x..z), then the comma, then another variable
" name or an !immediate, then the ].  You can also assign from a variable
" or !immediate directly to L, leaving out the comma and second variable.
```

```
"Some examples:  [=L=!Title for a data plot] [=L=a,! is the real result "a"]
   [=L=A,a] [=L=L,!with some text added] [=L=A,B]
" Remember, though, that the other variables can't be assigned values from
" L directly or via math operations.  L is only for entering text into
" LAUD command streams (usually as a line of text).
"
"There can be at most two operands and one result in each math operation.
" There must not be any extra stray characters or blanks within the
" math expressions.  Using LAUD Titles (like in the example below) is a
" good way to display and check results of variable manipulations while
" you are working on a script.  Alternately, you can add an extra prompt
" for the value, since the prompts always start showing the current value.

{math}
|What is variable K:|
|?K|

|Give a real variable f:|
|?f|

" variable B is here assigned two blanks, as that's a way
"  to get the space character into the input stream in a script...
[=B=!  ]

"Add K and A and put the result in C... Note the two equal signs for
" assignment.  You could also use a constant for the second operand,
" such as: [=C=K+!2] (but don't forget to include the !)
[=C=K+A]
"Note that if either K or A, above, can not be interpreted as a valid
" whole number, variable C (the result) will not be changed from its
" previous value.

"Real values or mixed types are used in math operations using the same
" notation as are the whole numbers:
[=c=f*a]


| Click to do another math operation|{moremath}
| Click to stop this nonsense and go on...|{cont}
{moremath}
|Give a new value for A|
|?A|
|Give a new value for REAL variable 'a'|
|?a|
[j{math}]

{cont}

"IMPORTANT!:  These math operations or assignment codes should NOT follow
" any other types of codes on the same line, as they may be executed out
" of order.  For example, the line:
"        * Display Format Scale dBperdiv > [=a=x]
" would do the [=a=x] assignment first, which is not obvious from reading
" the code.  It is safest to put these assignments and math operations on
" separate lines.
```

```
"*****************CONDITIONAL JUMPS***************
"You can use the results of the math operations or of some measured
" or configured values in LAUD to control the flow of the script
" 'program'.
"Conditional jumps are always contained within square brackets and
" start with the character '~'.  Following the '~' must be a variable
" name ('a' through 'k' or 'A' through 'K'), then one of four comparison
" symbols ('<', '>','=', or '#'), and then either another variable or an
" immediate constant (preceded by a '!').  Following that is the name
" of the label which the script execution should jump to if the
" result of the comparison is true.  For instance:

{AISMORE}
[=a=!6.2] "assign the value 6.2 to real variable 'a'
[=E=!21]  "assign 21 to the integer value E
[~a>E{AISMORE}]

"The last line above would cause execution to jump back to
" the label 'AISMORE' if a were larger; but it isn't, because we just
" assigned it a smaller value than E.  Since the condition is not true,
" the script will proceed to the next executable line.


"The following will keep repeating until you enter a value of 5:
{doitagain}
|Give me 5:|
|?A|
[~A=!5{outahere}]
[j{doitagain}]
{outahere}

"Beware that floating point representation is somewhat indefinite, so
" using '=' or '#' with floating point variables should be avoided.  The
" program may not consider 2.999999 (a real type value) to be equal to 3,
" for example, or may hold more significant digits than it displays to
" you.

[=b=!41.999]
{doitagain2}
|Give me a number 42 or higher:|
|?a|
[~a>b{outahere2}]
[j{doitagain2}]
{outahere2}
[T02]
|Right!...|

"Don't put a conditional jump as the last line in a script.  If necessary,
" put a comment line afterwards, to insure proper execution.

{dataread}
"*************READING DATA AND PARAMETER SETTINGS*************
"The values of certain system parameters (such as display scales
" or delay values) or data values which can be accessed via markers
" (dB, angle, impedance, etc.) can be loaded into the script variables
" for use in math operations or in affecting script program flow.
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```
"When a numerical input is prompted for by LAUD in its operation,
" the last valid value of that parameter is saved in a real-type
" variable 'x'.  For instance, if you are setting the "Ohms/division"
" value for displaying impedance, LAUD asks for your input for that
" value and that scaling value is saved in 'x'.  This is useful when the
" the prompt occurs within a script execution, for it allows your script
" program to determine system parameters.  Note that some parameters
" are entered via menu choices or function keys; these are not loaded
" into 'x'.

"When a marker is used to read time response, frequency response or
" impedance, the variable 'x' as well as two more special variables
" 'y' and 'z' are loaded with data from the marker's last position.
" The assignments are as follows:
"
"  Display          x                      y                   z
"  Time response  index(point number)   time(ms)            value(normalized)
"  Freq response   frequency             Magnitude(dB)      angle(degrees)
"  Impedance       frequency             Magnitude(Ohm)  angle(degrees)
"
"In a pass/fail production control test (with "Eval" turned on, see the
" menus under [Cntrl Setup1]), the x variable will hold the result,
" with 1 meaning "pass" and 0 meaning "fail".
"
"You can use the values in the x, y, and z variables by assigning
" them to normal script variables (a through k and A through K).  This
" is done in a single operand, single result assignment, such as in the
" following examples:
[=a=x]
[=A=y]
"The x,y and z variables can also be loaded or combined into the L
" text variable.

"When a whole number variable, such as A, is loaded from x, y or z, the
" value is rounded-off during the assignment.  The values x, y or z can
" not be assigned values except via normal LAUD menu prompts,
" and their values can not be directly typed into the command stream.
" They can, however, be used as the first operand of a math operation
" such as: [=a=x*!2.05] in assignment of normal variables.

"Here is an example of how to read the values at a frequency point.
" First, we use load a file to make sure a frequency response plot is
" being displayed.  The lower display limit is set to 1000Hz and the
" upper limit is set to 20000Hz to be sure the frequency is on the screen.
" Then the marker is called up using F5, and is moved to a
" frequency point by entering the frequency.  The point is then read
" into the x, y and z variables by again pressing F5.  Finally,
" the values in x, y and z are loaded into a, b, and c and then displayed
" for this demonstration using the Title as before:

* File Retrieve Freqfile FOAM> "FOAM is a .FR2 file which comes with the LAUD disks
[S5] 1000> [S6] 20000> [F5] 2000>
"Now, call the marker back so the values are loaded into x,y,z:
[F5] >
[=a=x]
[=b=y]
```

```
[=c=z]
[=B=!  ] "this is for typing in spaces.

* Display Title Edit #071
"The following is text entry into the title up till the '>' (Enter)
FREQUENCY= [@a]MHZ [@B] DB= [@b] [@B] ANGLE=[@c] >
* Display Title View *

"You can read and change parameters under script control:
{readscale}
[T00]
"Read the dBperdiv scale:
* Display Format Scale dBperdiv > "read and just use Enter
[=A=x] "read the value into A
[y500] "move the quickmessage box to the bottom of the screen
"       so you can see the middle of the screen
|click here to increase the scale by 1|{upone}
|click here to decrease the scale by 1|{downone}
|click here to stop playing with the scale..|{nomore}
[j{nomore}]
{upone}
[=A=A+!1]
* Display Format Scale dBperdiv [@A] > "put in the new value
[j{readscale}]
{downone}
[~A<2{readscale}] "don't reduce it if it's already down to one!
[=A=A-!1]
* Display Format Scale dBperdiv [@A] > "put in the new value
[j{readscale}]
{nomore}

"*************LAUNCHING SCRIPTS FROM SCRIPTS**************
"You can launch a script from within another script, or from a macro.
" The way to do this is to use the menu keystrokes for the
" script launching options under the 'Cntrl sCripts' menu. For instance,
" the following line would, if it weren't commented-out by the quote
" mark at its beginning, start this example script over at its beginning.
" If you remove the quote and rerun this script, you can escape the
" eternal loop by pressing the [End] key repeatedly.

"* Cntrl sCripts Custom scr_Directory DEMOS > run_Script EXAMPLE >

"Be sure to select the proper script directory before running the script!

"**********************WRITING SCRIPTS****************************
"You can write scripts using nearly any text editor which can
" save an ASCII file.  The Edit program of DOS works well.  A
" particularly convenient way to develop scripts is to run LAUD
" from within Windows 3.1.  You can switch out of LAUD and back into
" Windows Program Manager (or any other active Windows application)
" by holding down the [Alt] key and pressing [Tab].
" From there, you can enter the Notepad for writing or editing moderate
" sized scripts.  Save the file from Notepad, press [Alt Tab] again
" till you get back to LAUD, and try the script.  You can then pop
" back and forth between LAUD and the Notepad for testing and debugging.
"Be sure you exit the script in LAUD before trying to edit it in Notepad
```

```
" or you'll get a Windows error message.
" Remember that the script you write and run must be in a subdirectory
" under SCRIPTS, which is under your Base LAUD2 directory.

"******************DEBUGGING SCRIPTS*************************
"The execution of a script or of part of a script can be put into
" 'single-step' mode to help in debugging.  To initiate single-step
" mode, put the code:    [1]   into your script on its own line.
" to exit single-step mode, use:  [0]   , also on its own line.  You
" can go into and out of single step mode as many times as desired
" within a script (but the mode can be changed only by putting the
" codes in the script, not via LAUD keyboard commands).  When in
" single-step, each LINE will be displayed at the bottom of the LAUD
" screen before execution.  You must press the [Enter] key for execution
" of that line to proceed.  Comment lines will not be displayed or
" paused, nor will 'quick-message' lines other than the first in a
" block.
"
"You should switch to single-step mode during development when a script
" is not behaving as intended in a certain region.  You may have to
" break up some lines into separate lines of one or two commands each
" in order to find where things are going wrong. The single-step mode
" will pause only for a new whole line, not for each LAUD command key
" or script code.
"
"Most script programming errors tend to be context problems: LAUD
" command keys getting typed-in by the script when the wrong menu is
" being presented (such as trying to use F for 'Format' while not
" in the main Display menu). Remember to use the '>' (script code
" for the [Enter] key) after having a script respond to a LAUD prompt.
"
"Another common problem is use of incorrect directories
" when switching to another script or when loading a data file.
"
"The most difficult problems to catch are 'grammar' errors in script
" command codes, particularly those involving variables. Remember:
"  *** For math operations or assignments involving variables, only
"      the LAST argument can be a constant. These are legal:
"            [=a=!2.113] [=a=b+!2.113] [=A=!Some Text] [=L=x,!Hz]
"      (Be SURE to use the ! for immediates).  The following are ILLEGAL:
"            [=a=!2.345+b] [=a=20] [=A=Some Text]
"      The code [=A=!2.543+b] is legal, but it will put the character
"       string '2.543+b' into variable A, rather than the result of
"       a math operation involving variable b.
"      Be aware of what the ! does: it takes literally what follows it
"       up to the ']' and uses that for the data.  For instance, if
"       you use [=A=!B], you'll get variable A filled with the text
"       character 'B', and NOT with the value of variable B! But
"       the code [=A=B] will fill variable A with the contents of
"       variable B.  That is why [=a=3] will not work: there is no
"       variable named 3.
" *** The [@ code is for typing variable contents to be used as LAUD
"      program commands or within quickmessages ONLY.
"      It will NOT nest text within other text.
"      For instance, the code
"                      [=A=!Some Text] [=B=![@A]]
```

"   will only put the characters '[@A' into variable B, and will
"   not be seen as a reference to the contents of A.
"   When the script interpreter sees a code like [@C] (not within
"   another code), it types-out the characters corresponding to
"   the value of the variable (C, in this case), into LAUD as commands
"   or into a quickmessage. This is altogether different from
"   your typing of characters into a script, which happens outside LAUD.
"   In general, you cannot nest square-bracketed codes within other
"   codes. To combine text or variable results, use variable 'L' and
"   the comma (,) operation.

[T15]
|This concludes the LAUD SCRIPT Example. We hope you find writing and|
|using these scripts will make LAUD even more powerful and useful for|
|audio system measurement purposes.|
|For more practical examples of script programming, read the text of|
|the various scripts supplied with your LAUD installation disks.|
|Thank You.|

Instrument Shared Characteristics and Terminology

The five main instruments of Liberty Audiosuite are selectable by clicking the mouse on the appropriate "Instrument Button" at the bottom of the screen, or by pressing the underlined letter shown in the Instrument Button while holding down the [Alt] key. For instance, to switch to the DIST_AN instrument, hold down [Alt] and press the "D" key (hereafter designated as "[Alt-D]"). The mouse will not function during use of the SCOPE instrument because of real time interrupt and graphic routine conflicts, so the Alt keys must be used to switch out of the SCOPE instrument.

Mouse clicks (left button) will also activate certain areas of the screen (except in the SCOPE). Right button clicks are usually similar to [Esc].

In the MLS and SINE instruments, clicking to either side of a frequency domain display allows you to easily change the displayed frequency range. Clicking within a plot area which supports markers will cause the last activated marker jump to that horizontal position on the displayed trace, and will cause the data values at that point to be displayed above the plot frames.

Markers (in MLS and SINE) are "temporary" until they are "fixed": a left mouse button click will place a marker temporarily for positioning or readout. A center button click (or the [Enter] key or the [ok] screen button) will fix the marker position. Temporary placement of markers is useful for reading values at a desired frequency point; the value of the data at that point is displayed above the graphs and below the Function Key buttons on the screen. Using [Esc] instead of [Enter] returns the marker back to its previous position.

In a dual channel time domain display, clicking within a channel's display frame will allow placement of a marker for that channel.

Clicking on the [Liberty Audiosuite] logo at the top right of the screen (or pressing the asterisk key, or the shifted "8" key) will bring you to the top level menu in the menu driven instruments.

Clicking [Shift-F1 for Help] will bring up the LAUD HELP screen and system, as will the [Shift-F1] key combination. The HELP facility provides much (but not all) of the content of this manual in condensed form. HELP is keyed to the menu structure so that when HELP is entered the text it presents is related to your present menu position. If the text does not cover the information you require, click on the [INDX] button (in HELP) for a list of topics from which you can select. For more in-depth discussions on a topic, the LAUD manual should be consulted.

Menus are navigated in the same manner as those in many spreadsheet or editor applications. Press the key corresponding to the letter which is capitalized or highlighted to execute the desired operation or to bring up a submenu.

"Domains"

The expression "time domain" refers to a set of data (or a graphical chart or curve) which has as its argument time; for instance, if you ask "what is the pressure on the microphone 3 milliseconds after the impulse is sent?", you are requesting time domain information. Time domain information, as used in Liberty Audiosuite and nature (and as opposed to mathematical theory), is scalar and "real"; it can be positive or negative, but has no “imaginary” part or phase angle. Examples are voltage, current, energy, pressure or velocity vs. time, as seen by the SCOPE or collected by the MLS instrument.

Frequency domain data has as its argument, of course, frequency. If you ask “what is the impedance of the network at a frequency of 200 Hz?”, or “how strong is a speaker’s relative response at 3100 Hz?”, you are asking for frequency domain information. Most frequency domain information in the MLS, SINE or DIST_AN instruments is complex, that is, each value has both a magnitude and a phase (or, expressed in a different format, a real part and an imaginary part). Examples of complex frequency domain data are impedance response (resistance and reactance combined), frequency response or distortion products. If desired, only the magnitudes of these data can be displayed and used.

Frequency domain data is not always complex. An example would be the power response display of the SPEC_AN instrument.

Some data can be dual-domain. The Waterfall plots provided by the MLS instrument, for instance shows a 3-D surface which shows response magnitude information as both a function of time and frequency. There is, however, an inherent uncertainty built into any such measurement which the user should be aware of. Greater resolution in the frequency domain can be obtained only at the expense of poorer resolution in the time domain and vice versa. This limitation is a mathematical constraint (and is in fact an analog of Heisenberg’s Uncertainty Principal in Physics), and is not a system defect. You cannot determine a response at an instant in time and at a single frequency-- how can there be a defined frequency (something that happens per second) at a time instant of zero duration? A narrow band of frequencies implies a wide time span, and a narrow time span implies a wide frequency band.

Setting Levels

Note that both output channels of the DSP card will be driven in common when using LAUD.

When making measurements which assume a linear system is being tested, care must be taken to assure that the dynamic ranges of the UUT (Unit Under Test) and the measuring instrumentation are respected. Measuring responses with LAUD is no exception. If a power amplifier (PA) is used as part of a setup, the output level of LAUD (or the input gain to the amp) must be set to avoid clipping the PA.

In particular, care must be taken that the input levels to LAUD are below the overdrive levels. Each of the instruments in LAUD has a menu selection (usually under [Acquire]) or a function key button which allows setting of input and output gains. In all but the SCOPE instrument, these can be easily reached by pressing the “Levels” button on the lower right of the screen (just above the DIST_AN instrument button). The “Levels” button also displays the dB gain settings of the Main or Measure input (“M”), the Cal input (“C”), and the Output (“O”). The minus key ([-]) can also be used to bring up this menu.

These levels can also be automatically set in most situations by using the [Auto_adjust] option under the [Aquire Setlevels] menu. You must specify whether both inputs (Ch1 “MEAS” and Ch2 “Cal”) should be automatically adjusted or only the Ch1 input, and if so, whether they should be adjusted for matched gains (which might be important if you are measuring absolute signal loss in a crossover) or if these two inputs should be adjusted independently. This selection of adjustment type will be used in [auto-Measure] operations (simple canned measurement operations) in which the auto-level setting is enabled. An exception is for [auto-Measure Impedance], in which the setting will be changed to both channels matched. The last used Auto-level_adjust type selection can also be executed by using the key combination [Shift-F10].

In most cases during level adjustment, when the input levels are overdriven such that the Analog-to-Digital Converter (ADC) can no longer follow and convert the input voltage, LAUD will make an error “beep” noise, telling you to turn it down. This will not always occur during actual acquisition (LAUD in some cases skips the level checking to achieve maximum speed), so be sure to have the input levels properly set before beginning a measurement. The input levels can be reduced by decreasing the drive to the UUT (adjusting LAUD’s output level or an external amplifier’s input gain, if adjustable), changing the setting on the switchable attenuators on the external mic/probe preamp, or by adjusting the LAUD channel 1 and channel 2 input gains using the menus or buttons.

There is 22.5 dB of adjustment available via software for the channel 1 and channel 2 input gains. Channel 2 is normally the CAL probe input. Note that the Automatic Level Adjustment is effective only over this range -- if this is inadequate to bring the input signal within usable ranges, an error message will result (for overload) and the preamp attenuation switches (labeled “GAIN”) must be manually changed. If you are running a test for which absolute gain (as opposed to merely response shape) is important, **be sure to inform LAUD of the position of these attenuator switches. The proper menu for achieving this can be reached by using the plus ([+]) key.** The values entered should normally be positive values (i.e., attenuation, rather than gain as is displayed on the mic/probe preamp..... gain = -attenuation).

If the right channel “Cal” (#2) input is not being used (single channel mode, SPEC_AN or DIST_AN modes), it should ideally be disconnected from any signal sources or greatly attenuated by an external preamp gain setting of -40dB. This will avoid overdriving the unused input and degrading the channel in use.

Display Aliasing

At times when you are looking at a time domain sine waveform (via the Scope, the MLS instrument or the Level setting facilities of any of the instruments), the display may appear to get somewhat messy, with strange envelope patterns or just a “grassy” appearance. This is due to the phenomenon of display aliasing. Basically, the computer can display only discrete pixels in making up a trace. If, for instance, you tried to show 10 cycles of a sinewave on a display screen which had only 5 pixels’ width, you could not see your ten cycles--only five points could be drawn. When the computer tries to show many cycles on its 640 pixel wide VGA screen, smooth sine waves cannot always be drawn, at least at the intended frequency; in fact, the apparently drawn frequency may be lower (i.e., still sinusoidal, but fewer displayed cycles) due to the resulting aliasing effect.

A further display effect may occur due to the sampled nature of the data. As the frequency of the sampled tone approaches half of the sampling rate, each cycle of the tone will be represented by fewer and fewer points. Sampling theorem tells us that a low--pass filter could correctly redraw the path between the points, but the LAUD computer display is not a low-pass filter; if it has spare pixel width between measured points, it will draw a line between the known points. Hence, at high frequencies or low sample rates, a sine wave may appear more jagged on the time domain display than you would expect. This is a display phenomenon and will not affect any calculated frequency domain data.

Effects of Delays and FIR Antialiasing Filters

The antialiasing filtering provided in the PSA DSP cards which LAUD uses is accomplished primarily using FIR filtering. This means that the input flatness of the acquisition system remains essentially ruler flat to near the highest usable frequency

("Nyquist" frequency, or half the sample rate) and additionally, the relative phase error of the response is within a few degrees in the pass band, as well. This is generally very good news, a flat filter with an exceptionally sharp cutoff and a perfectly linear phase having been somewhat of a Holy Grail for audio systems through the years. Here, it means that electrical (probe) measurements could be made without a Cal, yet still have little error in the frequency response measurement or time domain audio-band waveform.

But use of a linear phase filter can cause some confusion in a measuring system, for subtle reasons. Along with the linear phase characteristic comes a considerable frequency independent delay. Whereas the IMP system (which used conventional analog filtering) had minimal delay evident in its acquired impulse response and when hooked up to view its own pulse would show it near the very first acquired point, with LAUD the pulse will appear a small number of points later. Hence, if the entire time data for this acquisition of an unmodified test pulse (or MLS result) is transformed via FFT to the frequency domain, the phase curve will show the results of this FIR filter delay. This will not actually cause any problems if measured data is normalized ("Cal'd"), as the delay will also be corrected by the normalization process. Fortunately, using a Cal is easy with LAUD, as it is a dual channel system and can acquire the CAL signal simultaneously with the normal signal -- but that means that the comments in the previous paragraph about a Cal not being needed for a response measurement will apply only when absolute phase or delay data is not needed.

It might seem that it would be an easy matter to have the software just automatically delete the data points corresponding to the delay, but that would unfortunately sacrifice accuracy. The pulse shape which results from Liberty Audiosuite is in fact more similar to a "(sin x)/x" shape than a simple pulse (see figure 1). This is what gives it its flat spectral characteristics. There are slight wiggles before the large pulse, which, if deleted, would alter the resulting response near the antialiasing filter cutoff frequency. Deleting the first few points would remove the leading wiggles from the Cal signal, but not from a measurement signal (such as from a microphone) which is subject to further delay.

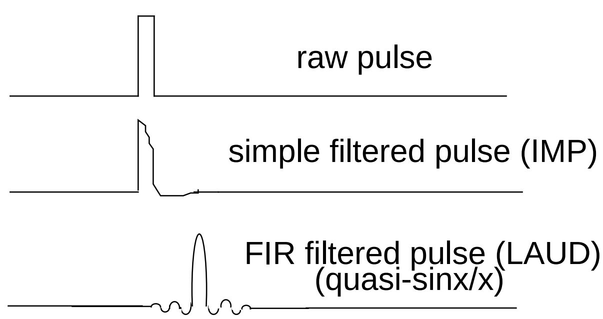


Figure 1: Filtering effects on test pulse used in impulse or MLS testing

DSP card Drivers:

LAUD does not use the driver software supplied with your DSP card, but it can disable any which has been loaded (if the host computer is also using the card as a multimedia soundcard). If a driver is loaded in your AUTOEXEC.BAT file or via CONFIG.SYS, it will need to be reloaded after running LAUD, as LAUD will overwrite the DSP programs which the drivers install onto the DSP card. Your DSP card manual may describe how to reload these programs; otherwise, you may wish to reboot your computer before using software which uses the DSP card for multimedia purposes.

IMPORTANT KEYSTROKES TO KNOW:

The keys to be discussed are summarized in a table in the Reference chapter at the end of the manual. You may wish to keep that page at hand until you become familiar with some of their functions.

Special Keys:

There are a number of special keys endowed with certain powers in Liberty Audiosuite, which are usable with some or all of the included instruments. They are as follows:

[Esc]: The escape key is used to back up one menu selection or as a "never mind" when you are being prompted for values. If you get in trouble, hitting [Esc] will usually get you out safely. If the mouse is enabled, the right mouse button also serves as the [Esc] key for many purposes.

[*] : The asterisk is used to return you immediately to the highest level menu, as an alternative to backing out via the [Esc] key. The backslash [\] will also perform this function except when entering file names. Should you get lost in the menu structure, just press [*] or [\] and you'll be back at the start. When using the mouse, click on the "Liberty Audiosuite " logo at the top right of the screen to perform the [*] function. Note that the SCOPE instrument has no standard menu structure, and will not respond to the asterisk key.

[!]: The exclamation mark key is for resolving DSP or ADC crises. The DSP processor, the Analog to Digital Converter on the DSP card, the main computer CPU, and all the interfacing which goes between them make up a staggeringly complex system. Occasionally something in the handshaking could get out of synch and cause the ADC to no longer provide samples to the system. Rather than require the user to restart the system should this happen, the [!] has been provided with the function of reinitializing the DSP and the ADC, just in case of such emergencies. Should the inputs suddenly seem to go dead, try pressing the [!] key.

[%]: The percent key will change the operating status from single channel mode to dual channel acquisition mode (or the reverse), if applicable. With a mouse, this function can be performed by clicking on the “Channels” button at the lower left of the screen (above the MLS button). When single channel mode is selected, one high-pitched “beep” will be heard; when dual channel mode is selected, two “beeps” will sound. There is also a non-toggling menu option to change this in the MLS or SINE “System” menu and in the [Cntrl setup2] menu. To perform this toggle in the SCOPE instrument, use the Display function key instead. The SPEC_AN and DIST_AN use channel 1 only.

[+]: The plus sign will bring up the menu for informing LAUD of your attenuator (GAIN) switch positions on the Mic/Probe Preamp. Remember to remove the negative sign displayed on the Mic/Probe Preamp front panel.

[-]: The minus sign brings up the manual level setting menu (same as pressing the on-screen Levels button at bottom right).

[=]: Auto scales the frequency response, cepstrum, and impedance graphs (single, not overlaid) to position the curve within the graph. In frequency response graphs, this will not adjust the dB/division vertical scaling (use [Alt-F8] for that), only the display gain or SPL reference.

[`]: Launches the Easy Scripts. This can also be done using the large button (marked with an appropriate description) at the bottom of the screen.

[Shift-Tab]: When a script's execution has not reached completion, but a running script has been interrupted by use of the [End] key, [Shift-Tab] will attempt to restart the script from the point where it was interrupted. The success of this action will often depend on how much the system parameters or menu position has changed since the script was interrupted.

[backspace]: This will work in prompt situations, much as you'd expect. If you give an empty string or invalid input when being prompted, the value being prompted for will usually be left unchanged from its value before the prompt.

[left arrow]: With most displays using markers, this will tentatively move the last changed data marker on a plot one data point (frequency or time) to the left (if there is anywhere to go). If the [Ctrl] key is held down while the left arrow is pressed, the marker will tentatively go ten points to the left. There will usually be some information about the point to which the marker is directed displayed near the top of the screen. Data about finalized marker locations is displayed at the bottom of the screen in most displays. The marker position will not be finalized (or have its data read out at the bottom of the screen) unless the [enter] key is pressed. If [Esc] or [*] are used instead, the marker will revert to its old position (and therefore can be used to just read off data points without permanently disturbing anything). The mouse can also be used to quickly select or position the markers: see "Mouse operations".

[right arrow]: This will behave like the [left arrow], with the expected difference.

[Up or Down arrow]: These are used in some cases (mostly in the SCOPE, which cannot respond to the mouse) for increasing or decreasing a parameter value. Alternately, the desired value can be entered manually. When setting input or output levels using the menu under [Aquire Setlevels] (or the [-] key or on-screen "Levels" button), only the up/down arrows can be used. If you wish to enter a specific value (or when programming a script), use the explicit setting menu under [Cntrl setup2].

[space bar]: If a script is NOT running but a menu is being displayed, this key toggles (enables or disables) display of the display title (a description which you can define and save with data files). If no title is defined, the computer "beeps".

The [space bar] is also used to advance script execution when a script is running. Alternately, a mouse click can be used for this purpose.

[Tab]: Allows you to edit the display title. When doing this, the [Home] key will clear the old title, the [Esc] key will restore the old title and the [Enter] key will replace the old with the new.

[End]: When recording a User Macro, this key ends the recording and assigns it to the appropriate [Alt-F?] key. **WHILE A SCRIPT IS RUNNING**, ends the script (if used while a LAUD prompt is being executed, the [Esc] key may need to be used, first). Scripts ended this way may be restarted (if the current menu position and system parameter mix is compatible) by later using the [Shift-Tab] key combination.

[Ctrl-End]: When recording a User Macro, aborts the operation and restores the previously defined macro.

Single Function Keys:

Some function keys numbers and their assigned names are displayed on buttons at the top of the screen in various instruments. The parameter values or selections associated with these, if active and applicable, are displayed directly below them in the second line.

The function keys can be pressed almost any time to allow manipulation of the related features. All function keys are not usable in all instruments and the function numbers are mostly different for the SCOPE, but are self explanatory.

The following is a summary of what the main function keys do:

[F1]: This key is not displayed at screen top, as it performs a simple but essential function. In the MLS or SINE instruments, pressing this key will cause the most recent time response plots to be redrawn, useful for updating the display after changing SIZE or to allow readjustment of time marker positions for echo elimination after frequency domain data has been displayed.

[F2]: the SIZE is the size of the current Time Response block which will be processed in an FFT (or a non-gated DFT filtering operation in SINE). Minimum is 256 points, maximum is 16,384, and any power of 2 in-between is allowed. The size goes up each time the key is pressed, except 16,384 will change to 256 on its turn. An explicit setting of this value (useful when creating scripts) may be achieved by using an option in the [Cntrl setup2] menu.

In the MLS instrument, when a Time Response is transformed to Frequency Response, the Frequency Response will be of the same size (but half the points are for negative frequencies, and for our purposes, will be ignored). When a time file is saved to disk, SIZE is also the number of points which will be saved. If SIZE is larger than the number of valid data points as might happen when a disk file has been retrieved, the rest of the file up to point number SIZE will be padded out with the voltage value corresponding to the right marker position (during the generating FFT). The bigger SIZE is, the finer will be the frequency resolution of a resulting MLS-generated Frequency Response, but the longer will be the calculation times and the saved files. Be aware that the time data display will not be updated to the new value of SIZE until a marker has been moved or the plot is redrawn (quickly done by typing [F1]). In DIST_AN displays, this button area will show the number of cycles used per each measurement (but the value is changed via the Acquire menu).

[F3]: RATE is the sample rate at which data will be collected (expect reasonably accurate frequency domain results only up to a frequency of about 0.45 times the sample RATE). RATE will cycle through its values, increasing when F3 is pressed, and decreasing when [Shift-F3] is pressed. It will be set to match disk data when it is read in. F3 should generally only be used when you want to collect data at a different RATE than shown. An explicit setting of this value (useful when creating scripts) may be achieved by using an option in the [Cntrl setup2] menu.
For highest frequency resolution in FFT or MLS derive data, use the lowest sample rate needed to achieve your highest required measurement frequency (for example, for woofer impedance measurements of data up to under 2000 Hz, use 5.5125kHz rather than 48kHz).

[F4]: INPUT determines whether data to be acquired into channel 1 will be from the microphone or a measurement probe (channel 2 data is always from the Cal probe). The selection will alternate each time F4 is pressed. An explicit setting of this parameter (useful when creating scripts) may be achieved by using an option in the [Cntrl setup2] menu.

[F5]: MKR1 is the left marker. The number shown is the data point index number. You can move the marker around by pressing F5 and then either using the arrow keys or by typing in numbers. If the plot shown is Time Response and the readouts at the bottom refer to "msec", the numerical data you type will be interpreted as index number (from 1 to SIZE). If Frequency data is shown, numbers will be interpreted as frequency in Hz, with "k" meaning thousands. Remember to hit [Enter] if you want to make the move final, or [Esc] if you were only tentatively moving the marker around to read values. Separate sets of markers are maintained for time responses, frequency responses and impedance data; the set which is controlled at any time is determined by the currently displayed parameter.

[F6]: MKR2 is the right marker. Operation is similar to F5.

[F7]: Changes the data WINDOW which will be used in the next applicable conversion to frequency domain. Windows are a complicated subject, and are also discussed in the “Echo Removal” section for the MLS/SINE instrument. This term does not refer to the Microsoft Windows operating environment, but to a mathematical operation to which you can subject your Time Response when transforming to the frequency domain. Basically, it involves tapering the data at either one or both ends of the response plot smoothly to zero to avoid
extraneous responses due to sharp edges formed when cutting out a section with the markers. The choices are: NONE (no windowing), BLACKMANN (much), HAMMING (moderate), and BINGHAM (mild), and the "one-sided" versions, those designated as .BLACKMAN, .HAMMING, and .BINGHAM. The one-sided windows taper only the right (later) end of the data, and are usable with impulse response traces. Double-sided windows are primarily for use with continuous, periodic data, such as sine waves, when making distortion measurements or wave analysis. An explicit setting of this value (useful when creating scripts) may be achieved by using an option in the [Cntrl setup2] menu.

[F8]: GAIN is used to scale either Time or Frequency responses. If the lower-screen plot is Time Response (again, markers will be in msec) then the GAIN is the factor that each value in the lower plot will be multiplied by for display in the upper plot, for saving to disk and for FFTing. Gain can also be negative to invert the Time Response, if desired.

Note that this gain, controlled via [F8], is a “display gain” only, not an input or output gain. It affects only software interpretation of the signal being analyzed, and has no effect whatever on dynamic range (noise or input overload). To adjust the levels into the ADC or out of the DAC, use the level settings available under [Acquire] or on appropriate function keys of the SCOPE.

When Frequency Response data is on the screen, the GAIN key will be used to set a dB offset for display or for saved ASCII format files. The value will be shown as "dB" in the lower part of the button. Waterfall plots must be restarted to show the effect of changes in GAIN.

In absolute SPL sensitivity mode (MLS or SINE instruments), the GAIN button can be used for adjusting the SPL reference line value.

[F9]: This key can be used to adjust the time delay associated with the current frequency response data display. The effect of this will show up in phase plots as well as IFFTs (inverse FFTs).

[F10]: This key, not shown on-screen, will rapidly move you to the Display/Show section of the menu structure.

Shift and Alt Function Keys:

[SHIFT F1]: This puts you into the Help system. You can also click the mouse left button on the button marked "SHIFT-F1" under the logo at the top of the screen.

[SHIFT F2]: Immediately prints a copy of the plot to the printer (or to a color or monochrome Bitmap file), as configured under [Display Printplot]. If set for a printer, be sure that the configuration is set for your printer type, or gibberish will likely result.

[SHIFT F3]: Decreases the sample rate. See [F3].

[SHIFT F4]: Brings you to the SPL settings (same as selecting [* Display Format Scale Spl On]) when in the SINE or MLS instruments.

[SHIFT F5] and **[SHIFT F6]:** these will allow you to make rapid adjustments to the displayed frequency range of frequency data, rather than going through the menu sequences. Get to know these: they are very useful. If you have a mouse, you can also activate these functions in the MLS or SINE instruments by clicking to either the left or right side, respectively, of a frequency domain (response or impedance) plot.

[SHIFT F7]: In MLS or SINE, changes the display to the plot of frequency response, if a valid response is in memory.

[SHIFT F8]: allows you to quickly change the displayed dB-per-division (scale) of frequency response plots or ohms-per-division of impedance plots (depending on which type is currently being displayed). If you wish to preset either (in the MLS or SINE instruments) when that type of data is not being displayed, use options under [Display Format Scale].

[SHIFT F9]: This option inverts the currently displayed data (multiplies by -1 in time domain, and shifts by 180 degrees in the frequency domain).

[SHIFT F10]: Performs an immediate automatic input level adjustment appropriate to the current instrument, using the type of adjustment parameters last activated.

[ALT F2] through **[ALT F5]:** executes the predefined user auto_measure (macro) operation. See [auto_Measure User] in the instrument descriptions. These macros are useful for defining simple often-repeated key sequences for easy execution. For more complex user-programmed measurement sequences or for measurement procedures which can be published or shared with other LAUD users, a Script can be written to perform a similar (but much more highly developed) function..

[ALT-M]: changes to the MLS instrument, if not there already.

[ALT-S]: changes to the SINE instrument.

[ALT-O]: changes to the SCOPE instrument.

[ALT-P]: changes to the SPEC_AN (spectrum analyzer) instrument.

[ALT-D]: changes to the DIST_AN (distortion analyzer) instrument.

[ALT-Q]: when in the SCOPE, allows you to Quit directly.

[ALT-C]: automatically resets the screen colors to the default colors.

(reserved for picture “MLS SINE SCREEN”)

The MLS/IMP/FFT and the SINE Instruments

Because of the similar operation and functions of the MLS/IMP/FFT and the SINE instruments, they will be discussed in this section together. Menu options which differ between the two instruments will be indicated in the text.

The instrument selected by the leftmost Instrument Button is the MLS/IMP/FFT (or “MLS”) device. This instrument might be considered a greatly advanced version of Liberty Instruments’ popular IMP system, implemented to use a DSP card rather than the external IMP module.

The second Instrument Button from the left selects the SINE instrument. This instrument is capable of performing most of the same basic functions as the MLS instrument, but uses as its stimulus steady state or gated sinewaves.

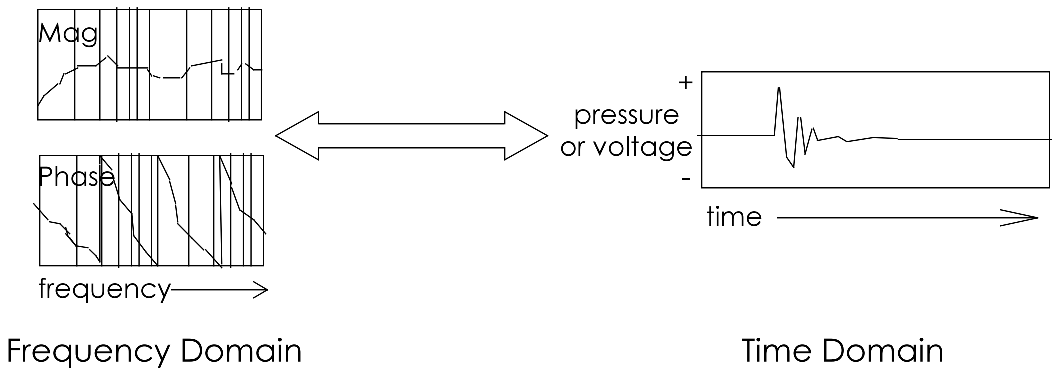
The data collected and generated by the MLS and SINE instruments can each be transformed or interpolated into the frequency point format of the other or onto other arbitrary sets of frequency points. For instance, the MLS instrument inherently generates frequency domain data which will lie on evenly spaced frequency points (f, 2f, 3f, 4f, 5f, 6f, ...). The SINE instrument can directly measure at any spacing of frequency points, but is usually used at “log” spacing of frequency points (f, kf, k^2f, k^3f, ...). There is a Convert function available within either instrument (under the Transform main menu), capable of converting frequency domain data into other formats by use of third order curve fitting.

For those unfamiliar with IMP or similar systems, a brief discussion of the technique is in order. MLS testing is a sophisticated variation on impulse response testing. Impulse response methods rely on the equivalence of a time domain “impulse response” and a frequency domain “frequency response”.

The SINE instrument measures the response to one frequency at a time; the MLS instrument first measures the time response to a broadband stimulus and then transforms it into the equivalent response over a wide range of frequencies. For measurement of frequency responses, both of these instruments require that the stimulus be applied by LAUD simultaneously with the measurement of the result; in cases where this is not feasible (such as when measuring tape recorders or some automotive-based sound systems), a non-synchronous technique such as the RTA provided in the SPEC_AN instrument should be used along with a recorded Pink noise sample.

RESPONSE MEASUREMENT THEORY

When an audio device (such as a loudspeaker or an equalizer) is measured to see how it changes signals of different frequencies which are applied to its input, there are two basic ways to express the result. One is by a frequency response graph, showing magnitude (strength) and phase (or time shift) at each frequency; the other is by an impulse response graph, showing the pressure (or voltage) at each time instant resulting if all frequencies are applied in phase at once. The first is called “frequency domain” data and the second is called “time domain” data. Both graphs contain the exact same information and can, by painstaking calculation or use of a computer, be converted (transformed) into each other.



When an engineer first thinks about frequency response measurement, very often the image that comes to mind is of a system which injects sinewaves of various frequencies into the Unit Under Test (hereafter called the "UUT") and simultaneously measures the size and maybe also the phase of the output from the device. When the measurement result is plotted versus the frequency of the applied sinewave, a frequency response graph results. Such a system (which is implemented, albeit with much greater capability and sophistication, in the SINE instrument) is conceptually simple and might be implemented using an audio sinewave generator and meters as shown in Figure 1.

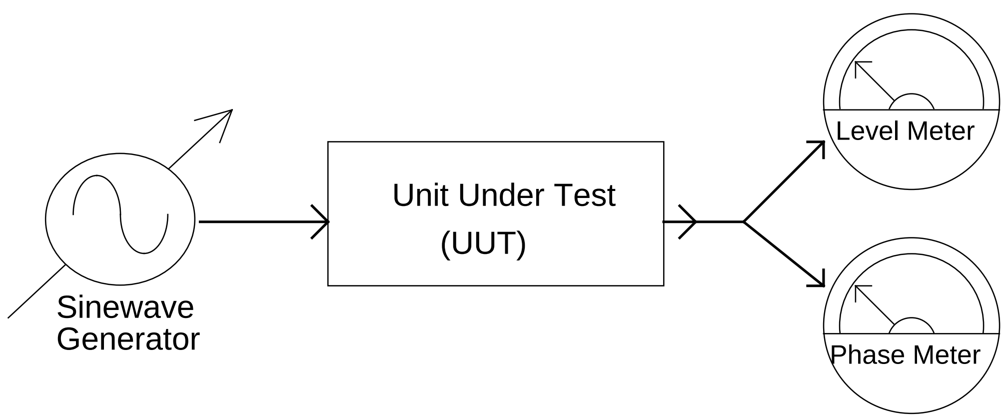


Figure 1: A conventional SINE-based frequency response measurement setup

But in many circumstances, sinewave based system may be less than optimum to the engineer. One immediately obvious disadvantage is the time duration required to perform the test. Each frequency must be applied, the UUT and measuring meter must stabilize and then the measurement can be recorded; this process is repeated at many frequencies to generate data for a plot.

This duration may be tolerable in many cases. Sinewave testing is a standard technique for well behaved systems which are not subject to significant time-delayed responses or reflections. It has the advantage of being usable at an arbitrary frequency (just set the generator to the desired frequency and measure) and has excellent immunity to noise. The UUT is subjected to a relatively narrow frequency band at each stimulus (which is sometimes preferable for devices which have poor linearity) and, if a filter is used in the metering (as is done in the LAUD SINE instrument), the SINE technique has excellent immunity to distortion in the UUT.

ACOUSTIC TESTING (MICROPHONE INPUT)

But a greater problem comes about when the measurement being made is of a loudspeaker's acoustical frequency response, a commonly required application of a response measurement system. Because the loudspeaker radiates into the space around it, the room and the loudspeaker's placement within the room have a strong effect on the measurement. The total energy reaching the microphone after reflection off of walls and other objects in the room can be and usually is stronger than that reaching the microphone directly from the speaker. The designer generally wants to know the response of the loudspeaker without that particular room's effects. Such a measurement is called an anechoic frequency response, and has classically been performed using sine wave techniques in a special room called an anechoic chamber (a chamber which ideally has no echoes). Such a chamber is large, expensive and generally only within reach of large loudspeaker design companies, universities, and national testing laboratories.

This problem can be solved in sinewave-based testing using techniques such as Time Domain Spectrometry (TDS, which equates time delay with a difference between the instantaneous stimulus and measurement frequency, using a chirp stimulus) or by gating (as provided in the Liberty Audiosuite SINE instrument) to measure primarily the signal content in the “first arrival” of the signal from the direct path while rejecting the signal content arriving after the echoes reach the microphone. Time Delay Spectrometry is well suited to the characteristics of primarily analog-based measurement systems, due to the relative ease (although relatively high cost) in generating frequency mixers and detectors. In DSP-based systems such as LAUD, TDS becomes very computation intensive and is much less well suited than gated-sine or MLS-based systems which can perform the same measurements.

The MLS instrument can perform testing using different levels of sophistication ranging from a simple impulse response test through a 16,383 point maximum length sequence correlation based test. Data over a wide range of frequencies can be measured and transformed in a few seconds. The effects of echoes are easily and intuitively removed. In general, if measurement speed is most important or extensive postprocessing, such as calculation of Cumulative Spectral Decay or Cepstral response is needed, MLS is the method of choice.

As shown in Figure 2, an impulse-based measurement begins with a determination of time domain behavior. For the example of a loudspeaker measurement, the signal used to stimulate the loudspeaker (or other UUT) is impulsive and broadband in nature, and echoes can be easily and intuitively separated from the direct response if desired by the user. Because Liberty Audiosuite works with a personal computer, editing and sophisticated display functions are inexpensively provided. The time response, or only a selected part of it, can then be converted by the software running in the computer into the frequency domain to form a frequency response plot. All of the frequencies in a wide band can be measured using a single stimulus. This basic procedure can be modified to provide accurate and easy impedance measurements, to allow microphone and stimulus response corrections, and for extensive analysis and conversion of the acquired data.

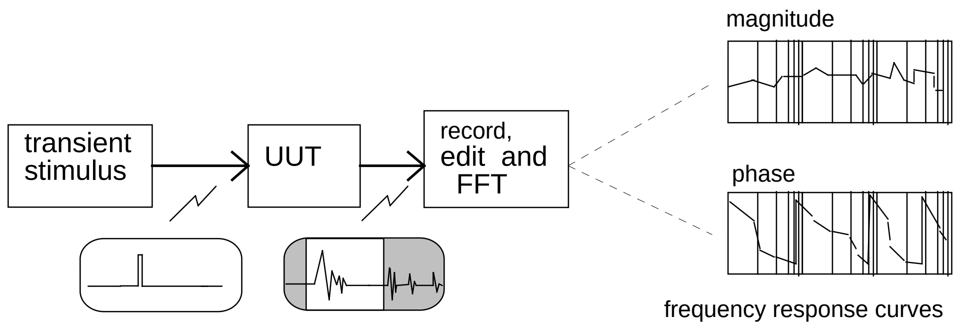


Figure 2: Conceptual diagram of impulse based frequency response measurement

ECHO REMOVAL IN LOUDSPEAKER MEASUREMENTS

In a speaker measurement, the signal picked up by the microphone contains the direct signal from the loudspeaker as well as signals reflected from other objects or surfaces in the room. The echoes, however, will appear later in time than the direct pulse, as shown in Figure 3. You can choose to include the echoes or truncate the response, depending on whether you wish to perform full in-room response measurement or an anechoic measurement.

LAUD has data “windows” available to process time domain data before conversion to frequency domain data. These windows are mathematical operation used to smoothly begin and end time data and to avoid abrupt discontinuities due to editing or finite acquisition size. When the impulse response is to be truncated to remove echoes, it is usually a good idea to use a “half window” to gradually decrease the time domain data near the truncation point to 0, rather than dealing with the sudden step to zero which might otherwise result. Sudden response steps tend to cause “spectral leakage”, or spurious frequency content in FFT’d responses. The half windows in LAUD are designated by a dot as the first character. For example, the window “BLACKMAN” is the full Blackman type of high-performance window in which both the leading and trailing ends of the target time domain response are tapered to zero. The window “.BLACKMAN” (note the dot at the beginning) is the same window, but applied only to the trailing end of the time domain data (the leading end of the data is not tapered).

A well implemented quasi-anechoic measurement (using *any* technique) will be configured when a maximum time period exists between the direct signal and the first reflection. The loudspeaker being tested and the microphone should be situated in the room so as to best achieve this goal. This arrangement often places the center of the speaker as well as the microphone at equal distances from floor and ceiling, and at least that far from other room surfaces or objects (an alternate placement which often provides even lower measurement limits is to place the loudspeaker on the floor facing upwards with the microphone positioned above it. This placement, however, may change the loading on the woofer at lower frequencies to the confinement to half-space radiation). The microphone is usually placed as close to the speaker system as possible, consistent with it receiving a properly integrated acoustic signal from all the drivers and the baffle of the system (the baffle will often have significant reflected output energy). Obviously, the larger the room and the higher the ceiling, the longer the quasi-anechoic sample which can be collected.

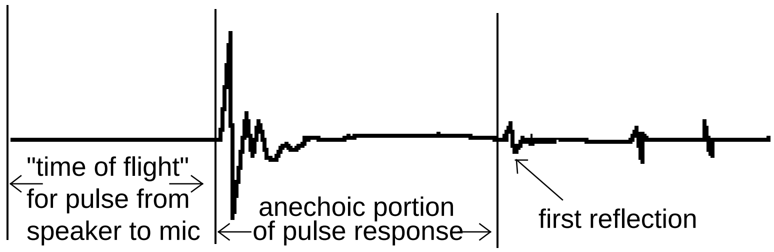


Figure 3: The direct pulse and the following reflections from a loudspeaker in a room

Editing or truncating an impulse response to remove echoes does, however, limit the lower frequency extreme which can be measured. For example, if only 1/200th of a second out of an impulse response measurement is left after truncation, no reliable frequency information below 200 Hz can possibly be obtained from the data; the data must span at least one cycle (and should span several cycles at least) of any frequency desired in order to provide any reasonable measure of it. This limitation, incidentally, is not unique to impulse-based testing. Gated sinewave testing and TDS, as well as all other quasi-anechoic techniques are subject to a frequency limitation imposed by the time gap before the first received reflection. Only extremely large anechoic chambers, of which there are very few (including the largest, known as “outdoors”) are immune to this effect for the lowest frequencies. The effect is not a defect of the measuring systems; it is, rather, a consequence of the fact that **any reflecting object within a wavelength of a loudspeaker surface is essentially inseparable from and should be considered to be part of that loudspeaker at that frequency**.

However, low frequency anechoic-equivalent data can be obtained with all of the anechoic techniques, including the MLS or SINE instruments, by use of near-field (very close) mic placement, which swamps out the effects of echoes while accurately capturing the room-independent low frequency response of a loudspeaker’s radiating elements. This technique provides the low frequency anechoic equivalent response assuming half-space radiation; for full-space, a drop in response of approximately 6dB occurs below a frequency which is determined by the effective baffle dimensions. An Easy-script is provided in LAUD which makes a near-field bass measurement and provides the result in effective SPL sensitivity (for 2.83V at 1 meter).

The technique for avoiding echoes with the SINE instrument involves “gating” the sine wave bursts, so that only the region between the first arrival of a sinusoidal signal at the microphone and the arrival of the first echo is analyzed for content at the measurement frequency. This region is best carved out using a full (NOT half or dotted) window on the data, tapering at the appropriate both ends of the portion of data which is undisturbed by echoes. Finding these proper points would normally be somewhat involved, as visual inspection of the microphone’s time domain output with a sinewave stimulus will not usually provide distinct clues as to where the limits of the uncorrupted direct signal may lie. But, as LAUD also includes the MLS instrument, and as the markers between the SINE and MLS instruments are linked, you can use the MLS instrument to mark off the anechoic region of the time domain impulse response, and then change to the SINE instrument for the measurement. The SINE instrument should be set for Gated, non-optimized mode in the Acquire menu for this operation, and the SIZE ([F2]) should be set to at least 1024 points. If the sample rate is not changed and the microphone or speaker not moved, the markers will then be in the proper positions for gating the sinusoidal burst. The time relationships of time domain responses to impulse and gated sine measurements are shown in Figure 4. Note that windowing is applied during the FFT or filtering process, and will not be visible when inspecting time domain data as is displayed after you use the [F1] key.

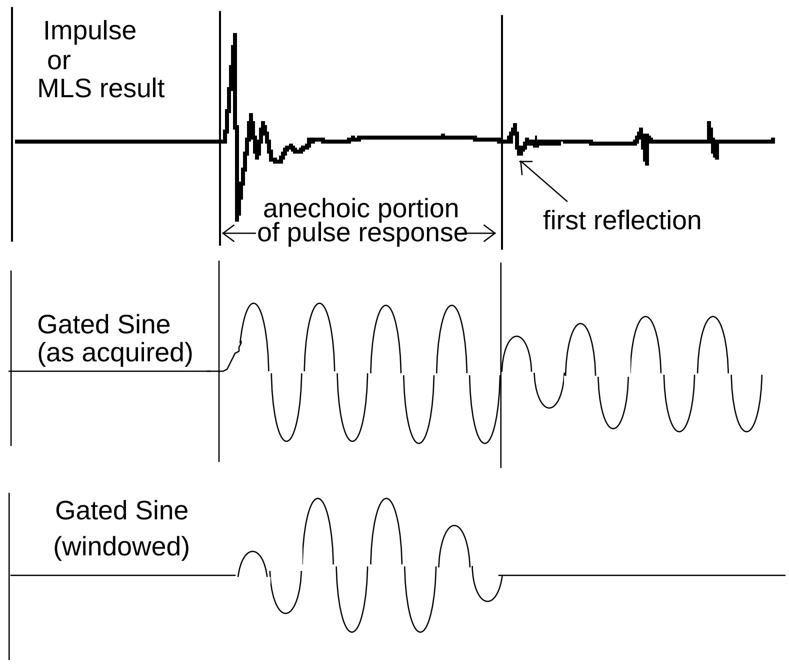


Figure 4: Time relationships between impulse and gated sine responses, and the time domain effect of a full window on the gated sine

If gating is not used in the SINE instrument and the optimized mode is not selected, the entire time domain data will be used in the measurement up to the analysis limit called “SIZE” (controlled by the [F2] button). Optimized mode will measure the ungated sine acquisition (as might be obtained when measuring a device not subject to echoes, such as a filter or amplifier) over a selectable fixed number of cycles for a “constant -Q” type of measurement. To minimize the effects of discontinuities at the analysis limits, full windowing should be used for all SINE measurements.

TIME, FREQUENCY, DURATION AND RATE

There are relationships which exist between the sample rate, the sampling duration, the upper frequency limit of a measurement and the measurement frequency resolution (data point density).

The upper frequency limit is mathematically restrained to be always below one half the sample rate (sample_rate/2, also known as the “Nyquist Frequency”). This limit is never reached, but is closely approached (0.4588*sample_rate on the PSA DSP cards), by use of sophisticated digital antialiasing filters.

The lowest measurable frequency is 1/(sampling_duration), (or for N samples at a sample rate of R samples per second: R/N). This applies for sine-based or FFT-based measurements. In simple terms, you need at least a cycle’s worth of samples to look for frequency content; in units of time, this duration will vary with measurement frequency.

For FFT-based measurements, such as are made with the LAUD MLS instrument, frequency data points will be inherently provided at multiples of (sample_rate) divided by (FFT_data_size). If all the points in the data are valid measured data (no truncation) this will also be the minimum frequency and the resolution. In general, if you have only N data points of valid data after truncation or windowing, there is no real advantage (other than smoother curves) to using an FFT size much larger than N. Using too large a SIZE has the disadvantage of greatly increasing the processing, data, display and storage requirements and slowing down the measurement operation. **It is a good practice with the MLS instrument to use only the SIZE required to accommodate your acquired data length.**

The SINE and the DIST_AN instruments have provisions for optimizing the sample rate and sample duration dynamically for best processing and measuring speed. In these cases, the “number of cycles” analyzed is constant and can be specified, resulting in a “constant-Q” type of measurement in which the resolution at each point is a fixed fraction of the test frequency. The optimized SINE measurement technique cannot be used (the setting will be ignored), however, when performing anechoic gated sinewave measurements, for which the duration is relatively short, fixed and held constant.

MICROPHONE RESPONSE CORRECTION AND THE CAL PROCESS

The processes described previously work well for many non-critical measurements and can be performed using LAUD, but several things are still being left to chance. First, of course, the microphone's frequency response will affect the measurement. Also, the control/power amplifier may have a non-flat frequency response, which will lead to measurement inaccuracies. Less obviously, the pulse or the MLS stimulus from Liberty Audiosuite will also have some finite variations in its frequency content. But these can all be dealt with in simple and inexpensive, yet effective, ways.

The microphone error can be fixed using various approaches. One obvious way is to use an ultra-expensive microphone carefully crafted and adjusted to yield a ruler-flat response over the audio frequency range. This approach represents the “golden sledgehammer” approach, and if an alternative exists, is a case of poor engineering economy. A better way is to measure or otherwise acquire correction data detailing an inexpensive microphone's response curve and then allow the computer to use this data to correct for the effects in its calculations. Liberty Audiosuite is equipped to utilize such microphone correction files for high sample rate measurements, yielding expensive microphone results at a small fraction of the cost. For many purposes, a "typical" curve for a given microphone type will yield adequate results; if full precision (better than a few dB) is required, calibration of an existing microphone or purchase of an inexpensive calibrated microphone should be considered.

The correction of channel response variations can be accomplished in a much more immediate fashion. In practice, we need only to determine what the signal at the UUT's output is compared to the signal at the UUT's input; if the test signal path isn't flat, it matters little? We can simply measure what its response is, and again let the computer divide the error out.

LAUD accomplishes this by providing for a CAL probe which can be connected to the UUT's input side (usually at the power amplifier's output, in the case of loudspeaker tests). In Audiosuite this measurement can be made simultaneously with the normal response measurement. In IMP or Liberty Audiosuite jargon, this probe input and the related normalization process is called a "Cal".

The easiest way to use Cal in the MLS and SINE instruments of LAUD is to operate in dual channel mode with the “cal” (channel 2 or right) probe sampling the stimulus. IN

DUAL CHANNEL MODES, THE CAL OPERATION IS DONE AUTOMATICALLY, HOWEVER THE CAL DATA IS NOT SAVED AFTER USE. BE SURE THAT THE CAL PROBE IS SENSING THE SIGNAL YOU WISH TO NORMALIZE WITH, OR NO USABLE FREQUENCY RESPONSE DATA WILL RESULT ! That is because you would be normalizing (dividing) each data point by a somewhat random near-zero value, giving a very large and inconsistent result. When in dual channel mode in the MLS and SINE instruments, be sure to use a 'cal' probe connection!

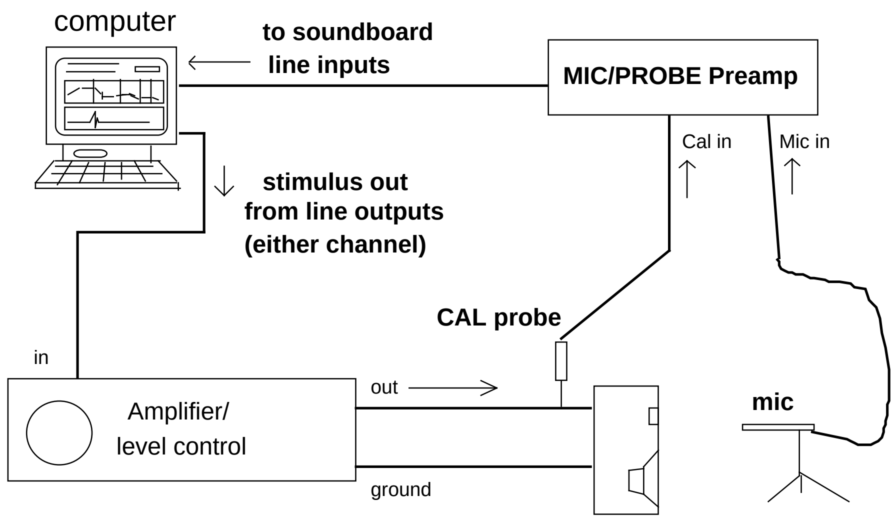


Figure 5: Setup for Acoustical Measurement of Loudspeaker, including CAL

When using probes, be very aware of the following:

CAUTION! In order to avoid damage to the computer DSP card or the user's power amplifier, several things should be kept in mind:
1). The DSP card has a very limited input voltage range, and exceeding this range can permanently damage the DSP card! It is strongly recommended that, unless the AC or DC input voltage from the unit which you are testing is guaranteed to never exceed 2 volts peak, you should use a mic/probe preamp to pre-scale and/or limit the peak voltage levels. If using the Liberty Instruments design for a mic/probe preamp, use IMP-type probes and never feed the probe inputs of the mic/probe preamp except via such a probe having an appropriate series resistor (Figure 6). The standard IMP-type probe used with LAUD has a 47.5K ohm series resistor in the hot lead at the sense end, and with the mic/probe preamp, provides protection for up to 140 volt signal peaks. These probes cannot be used connected directly to the DSP card, but require a Mic/Probe Preamp to restore gain and maintain bandwidth.
2). The ground lead of the probes must be connected ONLY to a grounded circuit node (i.e., one which is electrically identical to the IMP test output shield ground). In nearly all cases, the probe ground lead need not and should not be connected at all.

When using a Cal, the spectrum is measured before and after the tested device in the test configuration. The output spectrum referenced to (divided or "normalized" by) the UUT's input spectrum gives the UUT's frequency response, while all of the filter responses, pulse shape, and amplifier responses are removed mathematically from the measurement. Precision is achieved by technique rather than by high cost.

When using the MLS or SINE instruments **in dual channel mode**, the Cal process operates transparently (other than the need for setting its acquisition levels manually or automatically) and requires no user direction.

In **single channel mode**, a response curve must be measured and declared as the “Cal” data, the data it is to be applied to must be acquired or loaded, and then the Cal specifically applied, using the “Cal” command. This procedure may be useful in cases where you wish to normalize the results of one response measurement with the results of another, such as during investigation of small changes which might occur from modifications to your UUT. In that case, the curve you obtain is the change in response from one state to another, rather than a measure of the response shape.

ELECTRICAL MEASUREMENTS

The MLS or SINE instruments can also be used for non-acoustic measurements. To make such operations easy to perform, a second probe is used. As before, the Cal probe samples the UUT's input (typically LAUD’s stimulus output or a power amplifier's output). The channel 1 probe (used in place of the microphone) samples the UUT's electrical output (such as after a crossover network or equalizer circuit) and LAUD calculates and accurately displays the response change due to the tested device.

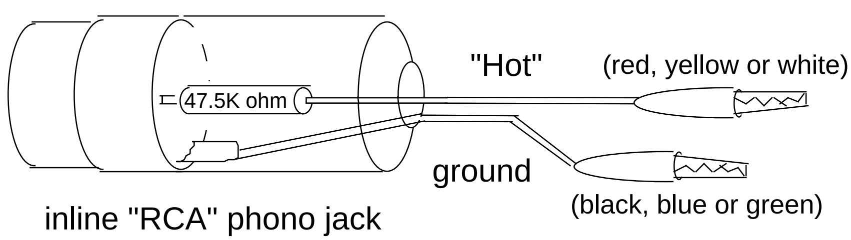


Figure 6: Required resistor within Probe circuit (at sensing end)

The ability to measure the frequency response of electrical devices can be extended to allow measurement of network impedances. This is accomplished by configuring the UUT within a voltage divider as the "bottom leg" with a known test resistor as the "top leg" (Figure 7). The MLS or SINE instrument then measures the signal content (magnitude and phase) at the top of the divider (the power amplifier/resistor connection) and at the center of the divider (the resistor/UUT connection). These two measurements and the value of the test resistor can then be used to quickly determine the complex impedance of the UUT at all measured frequencies. This function is useful not only for looking at speaker impedance and determining Thiele and Small parameters but also for measuring inductors and capacitors.

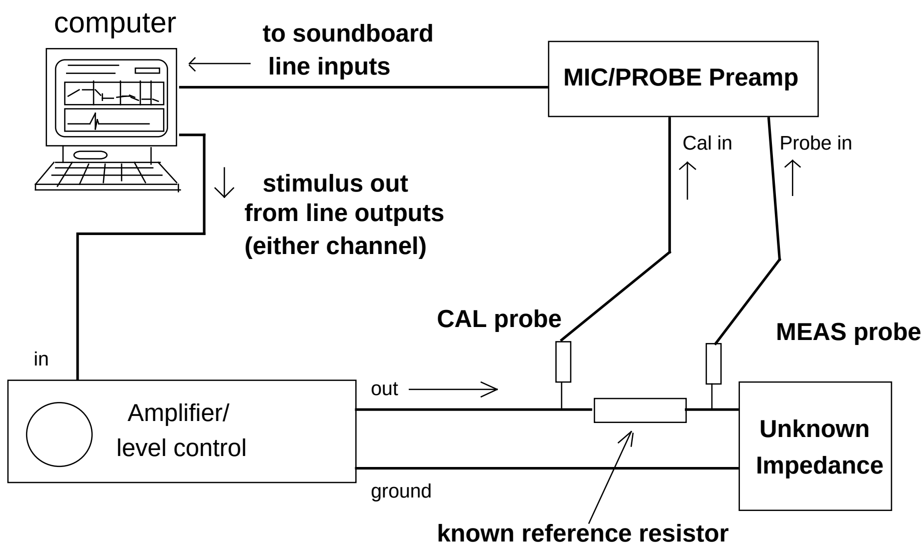


Figure 7: Impedance measurement configuration

the BAL PROCESS

Note: “BAL” is intended to correct for the effects of differing audio band low frequency rolloffs (occurring as high as 80 Hz) in some DSP cards. With cards such as the **ECHO DSP/mod** or other types which have been modified for extended low frequency response, **BAL is NOT NEEDED,** and for optimum speed and minimum complexity, should NOT be used.

In addition to the ability to use a set of “Cal” data, Liberty Audiosuite also can create and use a data set called “Bal”. Bal is meant to correct for slight channel response imbalances in the DSP card and preamp, and is intended for low frequency impedance measurements.

Bal data is similar to Cal data in that it is used to correct responses for errors in the equipment. Cal data is a correction file for an individual measurement’s common frequency domain response errors, and is (for best accuracy) refreshed with each measurement. Bal (balance) data, on the other hand is a correction file for the frequency domain differences between the two channels at lower frequencies. A Bal data set, if used, should be made when LAUD is initially installed and need only be changed if you change your DSP card. For frequency response measurements, Bal data is not used, as the correction is quite small and affects only very low frequencies. Small channel differences, however, can significantly affect infrasonic impedance measurements, which can be important in accurately determining Thiele and Small parameters.

NOISE, AVERAGING, AND MLS OPERATION

One disadvantage of pure impulse measurement as compared to sinewave-based measurements is that it is more easily corrupted by noise, both electrical and acoustical. The pulse energy is quite small; the pulse is narrow, but the system must measure the UUT's response for a long time period in order to provide low-frequency information. Noise has the whole time span to affect the measurement, but the test pulse has only its pulse width. The low frequencies are particularly subject to this noise contamination.

A common solution to this problem for dual-channel measurements is to use a filter to increase the low frequency energy relative to the high frequency energy in the test pulse. Because that filter's effect is seen by both the Measure probe (or mic) and the Cal probe, and because dual channel Cal'd techniques measure just the difference between input and output of a UUT, the filter's effects on the measured spectrum are automatically eliminated. However, the signal to noise ratio is improved in the frequency range which is emphasized. For specific measurements for which increased low frequency noise immunity is desired, this filter can be a simple one-pole RC filter consisting of a series resistor between the DSP card output and the UUT (or power amplifier input), with a shunt capacitor between the UUT input (or power amp input) and ground. **CAUTION: Do NOT connect capacitors directly between a power amplifier output and ground!** The corner frequency of the filter, $f= 1/(2\pi RC)$, is not critical, but should be selected near the highest desired frequency for the measurement. For general purpose measurement over various frequency ranges, a "Pink" filter can be used. A Pink filter provides an overall minus 3dB/octave slope to a response and can be obtained from several sources.

Another solution to the low frequency noise problem is based on the fact that LAUD has a computer at its disposal. This provides a simple means of improving noise rejection by averaging. The pulse test can be repeated many times under automatic computer control with the time response results of the tests being averaged. The pulse response will be identical for each test and will not change with averaging. The noise will be different each time, and will be reduced upon averaging. Each time the number of tests is doubled, the noise can be reduced by 3 dB. Each input (the mic, measurement probe or CAL probe) can be set up to take a user selectable number of test/average sequences whenever that input is used.

This averaging technique can give very good results within reasonable test times, but loses much of the speed advantage over SINE-based testing. When in noisy environments, when the device being measured is to be tested at low signal levels, or when very low frequency data is required, 20 dB or more of noise floor reduction may be required. This would require over 100 tests, and could take quite a while.

The **MLS technique** is a very sophisticated solution to this requirement. Basically, MLS mathematically squeezes the equivalent of thousands of test-and-average sequences into one acquisition. The test signal sounds like a hissing noise, rather than the tick or pop of the pulse. The UUT's response to this noise is rapidly converted to an equivalent pulse response, which is then treated just the same as if it had come from a pulse test. Noise immunity is greatly improved or, alternately, test time is drastically reduced. If still greater immunity is needed, the MLS tests themselves can be filtered, an/or repeated and averaged.

While basic impulse testing is provided for completeness in the LAUD MLS/IMP/FFT instrument, there is usually no reason to use it in lieu of MLS. MLS will, in nearly all cases, be the technique of choice for response measurements rather than impulse or even SINE measurements because of its excellent noise immunity and speed.

In MLS testing, a burst of pseudorandom noise is applied to the UUT, rather than a simple pulse. The UUT's response to this noise is processed by a very fast routine called a "Fast Hadamard Transform" or "FHT". The FHT quickly converts the UUT's response to the MLS noise burst into the UUT's impulse response. From that point forward, the time domain data looks like and is treated exactly the same as if it were a directly acquired impulse response. The only real difference is the extra noise immunity achieved.

SINE testing

LAUD supports both SINE and MLS synchronous response measuring techniques, allowing maximum versatility. Some advantages that the SINE technique does have over MLS are that the exact test frequencies can be specified and that the test operates in a manner which is more comprehensible to many users, making those unfamiliar with (or in some cases, unconvinced of) MLS techniques more comfortable with the measurement. One situation in which the SINE technique shines is when very low frequency impedance measurements are made; in this case, the SINE technique is capable of greater noise immunity (at the cost of measurement time), and is therefore the preferred means of collecting impedance data for Thiele and Small parameter determination of woofers with very low resonance frequencies.

If a loudspeaker measurement is being witnessed by a non-technical or semi-technical customer, the SINE based response measurement will be more easy for him to understand. The discrete frequencies can be heard as they are stepped through, and the corresponding data points are plotted. But the test will take much longer than the corresponding MLS measurement, for essentially the same result.

The advantage of being able to choose the specific result test frequencies is in most cases nullified by Liberty Audiosuite's very effective curve fitting data conversion capabilities (called "conVert" , under the Transform menus). These capabilities allow the MLS (FFT) acquired data to be remapped onto test points selected by the user. But in cases where results at only a few test frequencies are required, the SINE instrument can be used with greater efficiency.

Additionally, greater noise immunity is possible with SINE in situations which are not hampered by echoes (but are hampered by noise) and distortion immunity can in troublesome cases be better using SINE. High non-linear distortion in an MLS test can manifest itself as an apparently increased noise level.

MLS and SINE Menu Descriptions

These menus options will be listed primarily in order of occurrence from left to right, with each submenu discussed in order under the higher level keyword. The commas preceding a leading keyword indicate the number of selections from the [*] top of menu required to show the option.

The top level can be reached at nearly any time within these instruments, unless you are being prompted for information or when you are in Help, by pressing [*] or clicking on the [Liberty Audiosuite] button/logo.

Display: Provides access to display formatting such as graph scales or smoothing; allows entry of reference display or manual override of use of microphone response correction data; data titles can be edited and displayed on-screen and included in saved data; screen contents can be printed on a printer or to a bitmap file; additionally, the view and selection of data for display are controlled from this submenu. An "Overlay" submenu is also provided to allow up to five traces (1 from memory and four from disk files) to be on the same plot.

,Format: Allows you to change the display presentation by adjusting the start and stop frequencies, the vertical scale of the plots, select SPL sensitivity display for acoustic frequency response (if you have a calibrated microphone), select the resistor value used to measure impedances, add or subtract delay to the phase plots, select octave or sub-octave smoothing or select whether Phase will be displayed on frequency domain plots.

,,freq_range: Allows you to select lower or upper frequency range display limits for frequency domain data. These will also be the range of Frequency response and Impedance data stored during "ASCII" (not "full") save operations under the [File] submenu. You can set the low limit more easily via [Shift-F5], or the upper limit via [Shift-F6], whenever a menu line is displayed, or click the mouse at the left or right of a frequency response or impedance graph.

,,Scale: Select scale values for dB/division (frequency response plots), Ohms/division (impedance plots), Reference Resistor (which you supply and use when measuring impedance), and the dB/division setting for frequency response plots. You can also choose SPL sensitivity mode for display of acoustic frequency responses (provided a valid mic ".DAT" file is loaded and an external mic/probe preamp is used).

,,,Spl: This submenu determines the presentation of Frequency Response data which is generated. It gives you a choice of “on” or “off”. “Off” is the normal dB-relative mode. “On” is SPL sensitivity mode. All calculations assume that the value for the microphone preamplifier gain (relative to the gain from a probe tip) has been correctly set. This value may be set by selecting [mic_Preamp]. For the Liberty Instruments design mic/probe preamp, which provides 54 dB reference gain to the mic and 0dB reference gain to the probe tip (with a 47.5k resistor probe), the mic_Preamp parameter should be set to 54.

Any non-permanent external attenuation applied (such as the attenuator switch settings of a Mic/Probe preamp) should be entered into the Main or Cal “_ext_atten” parameters. These values should be entered as positive values, i.e., a switch setting of -20dB should be entered as “20”. You can use the [+] to reach the menu for entering the attenuator values.

Note that the SPL sensitivity measurement is a relative gain measurement of a loudspeaker’s sensitivity, yielding the SPL level which would result if 2.83V rms were applied. This does not mean this SPL level is produced during the test! To measure actual SPL levels, use the SPEC_AN instrument.

,,,,oN: chooses SPL sensitivity display mode for frequency responses and provides a further submenu to set parameters related to it. This can be reached more quickly using [Shift-F4].

,,,,,Nominal: allows you to input the SPL sensitivity level which will be shown at the line that is normally 0 dB on frequency response plots.

,,,,,Main_ext_atten: This value should be set to the dB value of any fixed attenuators which might be used with channel 1(using the microphone). Use positive numbers for attenuation, i.e., a "-20dB" switch setting on an external mic/probe preamp should be registered as "20". This parameter can be quickly set from other menu positions using [+].

,,,,,Cal_ext_atten: for entering the attenuation value of the Cal (channel 2) probe input (see "Main_ext_atten"). This parameter can be quickly set from other menu positions using [+].

,,,,,Ok: completes attenuation settings and redraws the display. Use this rather than the [Esc] key so that your SPL parameter changes are effected immediately.

,,sMoothing: Lets you choose the fraction of an octave over which frequency response data will be averaged. Choices are None, Octave, Half (one half octave), Third (one third octave), Sixth octave, and tWelfth (one twelfth octave). Each point represents the average of the dB values for all other points present within a span of the specified fraction of an octave. Having this set to a value other than "none" will slow down displays noticeably unless a fast computer is used. Smoothing is a display function. Full frequency responses saved to disk (.FR2 files) with smoothing ON will also be displayed after recall with smoothing ON, but the smoothing can still be removed as it affects only display, not the actual data. Saved ASCII files (.FRD) will have the effect of any smoothing which was active when the file is saved. Smoothing affects only frequency response magnitude, not phase or impedance displays. When a Hilbert transform is performed with smoothing on, the smoothing will be applied before the transformation (and should be turned off afterwards, so that it is not viewed doubly-applied).

When smoothing is on, the characters "sm/?" will be displayed at the left side of frequency response magnitude plots, where "?" is 2 for Half, 3 for Third, etc.

,,Delay: Select the delay which will be applied to the phase plots (and used during IFFT or ASCII frequency response save operations). You can also do this quickly by using [F9]. The number corresponds to the amount of delay you wish to REMOVE, not add! The value, if non-zero, will appear on phase plots, along with the amount of delay contributed by the offset in time marker #1 when the time data was originally transformed. To invert phase (not change delay, but offset all frequencies by 180 degrees), use [Shift-F9].

,,shoPhase?: Select whether phase plots should be shown with frequency response graphs. Phase is always shown (within the same frame) on impedance graphs. If phase display is not selected, the frequency response magnitude will be shown in a large frame; if phase is shown, magnitude and phase will be shown in separate half-sized frames.

,show: Select which data set and format to display; Get here fast by pressing [F10].

,,Impuls_resp: A display of time domain data. If dual channel mode is on, the bottom display is for channel 1 and the top is for channel 2. If dual channel mode is off, the top display is channel 1 from sample points 1 to SIZE and the bottom display is the part of the top display which is between the two time markers, and modified by the time domain gain (IMP users note that the top/bottom displays are reversed from their usage in IMP). Use [F1] to quickly get this display or to update the display.

,,Freq_resp: Shows magnitude or magnitude and phase, depending on whether dual channel mode is on. Phase plots will be modified by the Delay parameter ([F9]). You need to FFT acquired data (in MLS instrument), make a response measurement (SINE instrument) or load a frequency response file before displaying this. Use [Shift-F7] to show this quickly.

,,Time+freq: shows both time response of channel 1 and the frequency response on screen.

,,Z_plot: Display of complex impedance versus frequency. The impedance magnitude value depends on the setting of the value "Ref_resistor", under the [Display Format Scale] menu; be sure that this value corresponds to the resistor value used when the curve was measured!

,,Cal: Shows the current CAL data (NOT including the mic correction data), if any. The markers will not work with this display.

,,>More: Other ‘show’ options:

,,,dRiver_data: plot of both frequency response magnitude and device complex impedance on the same screen. Both impedance and frequency response data must be in memory.

,,,Distance: (FFT, or linear frequency, formatted frequency response data only) calculates the acoustic distance from the microphone to the driver which is producing output in the frequency range lying between the frequency response display markers. The distance shown is that which the output from a minimum-phase driver with the same magnitude response as the given frequency response data must travel through to best approximate the phase response of the given data. The minimum-phase equivalent is calculated here using a Hilbert Transform on the data currently in the frequency response block. The effects of the time response "offset", if any, are not included in the distance calculation. Use this to determine the acoustic positions of drivers on a baffle in multi-way systems, for time alignment, or for adjustment of delay networks.

,,,Mic_data: Shows the curve currently in the mic correction data set.

,,,Step: calculates and displays the step response equivalent to the current impulse response in the channel 1 time domain data. This is meaningful only if channel 1 time domain data is an impulse response! It will work best if the source was an MLS measurement, and may require some optimization of SIZE to make up for sloped sections of the curve due to offsets. Step response at any point is a summation of impulse response time domain values up to that point. Offsets in the acquired data can therefore cause a sloping in the displayed response. If a printout of the step response is required, it must be performed immediately after display, as other actions will cause display to revert to impulse response.

,Printplot: Make a graphic hardcopy of the screen and the current active title. Supported devices are Epson/IBM compatible matrix printers and HPLaserJet compatible printers including the Deskjet 500, 500C, 550, or 520. If your printer is not compatible with one of these types and you have Windows or a DOS graphics editor which can work with bitmap files, you can save screen graphics also to a color or black-and-white bitmap file. For the Epson/IBM type printers, choose one of modes 1-13 for best presentation (experiment to see which is best with your printer). Use "Go" to print from this menu or use [shift-F2].

Note: for printed hardcopy which shows the LAUD screens in their entirety (buttons, menus, etc.), for purposes of training manuals or promotion, there is a special “promo mode” available. To enter or exit promo mode, use “togglePromo” from this menu. Promo mode remains in effect for all the Liberty Audiosuite instruments until exited.

,Title: View or Edit the current title. The space bar can also be used to toggle the title display and the Tab key can let you quickly edit it without coming to this menu location.

,,View: shows the title on the screen.

,,Hide: use it to minimize screen clutter while working. The [space] key will toggle between view and hide whenever a menu line is present.

,,Edit: lets you change the Title. The [Tab] key also performs this function whenever a menu line is present! Editing is done from the right side of the line -- to change a part of the title that is not at the end of the title line, you must backspace (delete) all which follows it. To restore the title as it was before the current edit, press [Esc]. To clear the entire title, press [Home]. To save your changes (make them permanent and beyond instant restoration), press enter.

,Mic_correc: LAUD can use a microphone response data file to correct measured response curves along with the electrical CAL data when a "cal" is performed. The file should be in ASCII, have the suffix ".DAT", and be copied to the “base” (\LAUD) directory of your hard disk. The file is activated or a different correction file is selected via [File Micdat]. If a micfile is present and entered into LAUD, a small "m" will appear to the left of the "Liberty Audiosuite" title at the top of the screen. Data plots to which the micfile correction has been applied will have the "m" to the right of the LAUD logo. You can also use this menu to display the micfile, if present.

This submenu allows you to view the current microphone correction data plot in frequency response format (Show_mic_data) and to dictate to LAUD whether it should or should not apply microphone correction data to the current Time_data or Freq_data when LAUD performs a cal operation (remember that “cal” occurs automatically when a dual-channel mode measurement is made). LAUD normally notes whether the data came from the microphone input (per the user setting the INPUT button to match the input before an acquisition is made) and uses the correction data accordingly. However, you can override LAUD’s determination (or correct an error due to incorrect setting of the INPUT parameter) from this submenu.

Note that the correction will not be applied until a cal is performed (except for frequency response data which has already been cal'd -- mic correction will then be applied immediately). In dual channel mode, cal is applied automatically.

The “Micsens” parameter shows the reference sensitivity for the current mic data curve. The value can not be changed; this selection is provided only to allow the sensitivity to be read from within a Script (by loading the “x” variable when “Micsens” is used).

(reserved for overlay picture)

,Overlay: This submenu allows multiple trace plots to be composed using data in memory and up to four additional sets of data from files. The plots must be all of the same type: time response, frequency response or impedance. Files to be included in the plot must be of the "compact" or "full" types (.IM2, .FR2 or .ZM2, not ASCII types) and in the currently selected data directory. The trace from memory will always be trace #1, and the other four can be selected from a presented list.

,,Type: Choose the type of data (time, frequency or impedance). The time response plots will be shown in order from top to bottom and vertically separated and scaled to fill the window. Frequency Response and Impedance plots will be overlaid on the same reference line. For Frequency Response or Impedance types, you will be offered the selection of phase angle or magnitude display. The gain settings of the memory trace will be taken from the current settings; for the file traces, they will be as they were when each file was saved.

The user must assure that display modes are compatible: for instance, if the memory trace is showing absolute SPL data, all will be displayed in that format and any included file data must also be for a valid SPL measurement in order to get an accurate display.

,,Curves: Allows you to select whether to display each curve and the source file from which it is to come**.** [Add] includes a new trace, [Remove] deletes one and [Back] moves you back to the previous submenu.

,,Symbols: Lets you add symbols to each trace (Frequency Response and Impedance only) to aid in differentiating the curves on black and white printouts and select whether the symbols should be [Sparse]-ly or [Dense]-ly applied.

,,Legends: Lets you include the titles from the current memory trace on the display and the title that corresponds to each included file, as an explanation of what each curve is. If 'oFf' is selected, the plot size is increased to better utilize the available screen.

,,Go: ...Starts drawing the overlaid (multiple trace) plot.

File: Menu section for disk I/O operations, including output data storage retrieval and deletion (time, frequency, impedance, driver and cal data) and access to microphone data files. (NOTE: LAUD configuration files are accessed under the [Setup] menu.) LAUD data files have different extensions depending on what kind and format of data is involved. File types ending in ".IMP" and ".IM2" are time domain data. ".IMP" is in ASCII format (readable by other programs and into spreadsheets as a "printfile"). ".IM2" is in a special compact format readable, only by Liberty Audiosuite and IMP. ".FRD" is frequency domain ASCII data, for export only. ".FR2" is a frequency domain compact format which can be read back into LAUD. ".ZMA" is ASCII formatted impedance data for export and ".ZM2" is its compact format version for retrieval into LAUD.

Files for Production Control limits (Pass/Fail testing) must be in ASCII format. It is advisable that when these are generated from a measured curve that a compact version of the file also be saved so it can, if necessary be retrieved by LAUD for modification (gain settings or "MERGE" editing).

,Retrieve: Read in time domain, frequency response, impedance ("Zdata") or Cal data. For convenience, some environment conditions are usually changed to match those existing when the file was saved.

Select which type and the name of the file you wish to retrieve. You will be given a list of available files in the default directory. For particularly easy operation, use the mouse to double-click the desired file name. If you type the name in, do not type the .??? suffix. The default directory and drive can be changed by the [Change_dir] option; it can also be over-ridden on a single instance by specifying a complete path in the filename (the second character must be a ":").

The only ASCII files which can be retrieved into LAUD (other than as Pass/Fail limits files) are for time domain data. LAUD can retrieve “compact” (or “full”) format data for frequency response, impedance, time domain and cal.

,Save: Write to disk the time domain, frequency response, impedance or cal data, in Ascii or Compact/Full format. Except for time domain data, Ascii data cannot be read back into LAUD except for use as a Pass/Fail limit file. Compact or Full formats can all be read back in to the system. Various LAUD software settings and the title will be saved with the files for later retrieval.

The default drive and directory to which the files will be saved can be changed by the [Change_dir] option; it can also be over- ridden for the instance by specifying a complete path in the filename (the second character must be a ":").

Select the data type (time, frequency response, cal, or impedance) to save and the name under which to save it. Except for time data, ASCII data CANNOT be read back in to LAUD-- this format is provided primarily to make LAUD MLS and SINE processed data available to simulation, display and analysis programs. LAUD can, however, read in the "compact" or "full" data formats available in the save submenu.

Saved full frequency responses contain all the effects of calibration and microphone correction, but not of smoothing. The smoothing parameter value will be saved with the file and will be applied to it upon retrieval and display, but may still be switched off using [Display Format sMoothing].

,,Timeresp: Time data can be written out in two equivalent formats. Both contain the same information. Ascii mode is readable by other programs and spreadsheets. Compact mode requires about half the disk storage. The ASCII mode is a compatible superset of the original IMP and the IMP/M time data format, and is readable by previous software versions of IMP. Liberty Audiosuite can also read in .IMP files from the basic IMP software package; in that case, some system switches, such as whether the data was acquired from a microphone, may require resetting by the user.

,Delete: Enables you to delete unwanted ASCII or compact (including .IM2, .FR2, .ZF2, .CAL and configuration .ICF) files, to retrieve disk space and for data directory management. Files in the Script directories can not be deleted by this option, but should be done via DOS or Windows commands.

,Change _dir: You can specify the default directory to be used from now on. This directory will be only for acquired data files, such as time, frequency response, impedance or cal. Mic correction files, and configuration files will still be saved to and retrieved from the directory in which the LAUD program is resident.

,Micdat: You can specify the microphone correction data file to be used with mic-input data. Specify "NONE" if you want to disable microphone correction. The name of the Micdat file in use (if any) will be saved (to be restored later) with any configurations which are saved (see [Setup Config_file]).

Transform: Operations which convert data from one form to another, such as from time domain to frequency domain and back (only the MLS instrument converts back to time domain); manual "cal" (measurement system electrical response correction) data declaration for single-channel operation; Waterfall plot and Energy-Time Curve generation (MLS instrument only), formatting and display; Hilbert (minimum phase) transform; Cepstral response processing; swapping of time domain data between channel 1 and channel 2; display of the products (cascaded results) of transfer functions; data format transformations between the “FFT” formats (those usually generated by an FFT, with an implied sample rate and with data at evenly spaced frequency points) and arbitrary or log formatted data. Also included under the Transform menu are the MERGE and DRAW frequency response data editing facilities.

,Set_cal: This will not be used if you normally operate the SINE and MLS instruments in the recommended dual-channel mode. Set_cal declares the current frequency response data to be the Cal data for use in subsequent data correction. This is meant for use in <u>single channel mode</u>, during which the Cal time data and the measured data are acquired at separate acquisitions, as in the IMP system. IN dual channel mode, the normalization is an automatic operation. To change to and from dual channel mode, use the “%” key.

Set_cal is usually done when frequency response data is the response at the input side to the speaker or other unit under test (acquired with a probe into the "Cal" input). Cal data can also be loaded in from disk via selections in the [File] menu. As opposed to size and rate restrictions in IMP and IMP/M, Audiosuite will curve-fit the Cal data to match the frequency points of the data it is to be applied toward. Best accuracy, will, however, result when the size and rate of the original Cal data matches the measured data.

,Cal: Applies the Cal data declared with [Set_cal]. For use in single channel mode. Data at each point in the frequency response block is complex divided (i.e., normalized) by the corresponding data in the Cal data block (after curve-fitting, if required). The result is put back into the frequency response data block. The data in the Cal block is not affected.

If the frequency response data is from a microphone measurement (or if you have convinced the computer that it was by using [* Display Mic_correction]), the mic correction data, if present, is also applied to correct the response file.

,Xfer and **,conVert:** These operations are available in both the MLS and SINE instruments. In the SINE instrument, they are at this menu level; in the MLS instrument they are under the “moRe “ heading. Descriptions of Xfer and conVert are given below, under “moRe”.

,Fft: (MLS instrument only) The time domain data between the time markers is transformed into frequency response data in the frequency response block. If dual channel mode is active (as recommended), channel 2 will also be transformed, with channel 2’s result used as ‘Cal’ to normalize channel 1’s frequency domain result. If you use dual channel mode with an FFT, <u>be sure to connect the Cal probe and adjust the channel 2 level</u> during the relevant data acquisition, or nonsense will result!

If you are doing an FFT of time domain data retrieved from a file, you should switch to single channel mode before using FFT (or it may be divided by channel 2 derived data, which may sometimes be useful if used in conjunction with “Swap_channls”, in normalizing one frequency response by another for comparative purposes).

After an FFT, the original time data in the time domain is unchanged, but any data which was formerly in the frequency response block is overwritten. Note that, while channel 2’s data in dual channel mode is transformed and used to normalize channel 1’s result, it is never put into the Cal data block and declared for usage later; the separate “setcal” and application of “cal” procedure is meant for a single channel mode of usage.

When transforming data obtained from the MLS instrument, a “half-Blackman” (.BLACKMAN) window is advised to minimize truncation-induced effects.

,Ifft: (MLS instrument only) Can be used only with “Linear “ formatted frequency response data, obtainable by an FFT (MLS) based measurement or by **conVert**ing other-formatted data to linear type. The current frequency response, modified by the dB gain or delay parameters or by cal normalization or microphone correction processes, is converted back into time domain data (into channel 1’s data). Channel 2’s data is loaded with an impulse. In LAUD2, the frequency response data is not destroyed in this process, as it was in previous versions.

Very strange dense and periodic IFFTs can result if the frequency response data contains unnaturally abrupt discontinuities, as can sometimes occur when “conVert” is used to transform data from a format which contains insufficient information at the frequency extremes or when a MERGE operation leaves artifacts in the magnitude or phase response. If such strange results occur, look at the source frequency response from within MERGE (set to a moderate sized frequency set of Linear spaced frequencies) to detect and if necessary edit out (draw over) any such aberrations.

(reserved for waterfall plot picture)

,Waterfal: (MLS instrument only) A cumulative spectral decay plot of the current time data block can be configured and displayed here. A waterfall plot is an approach toward showing both time and frequency-domain behavior on a single plot. These are used primarily for loudspeaker or loudspeaker driver characterization.

You can change the nominal length (time period over which the time variation is analyzed) and number of steps that are used via provided menu options. Your choices will be treated as "nominal", and will be adjusted to give even spacing of curves (i.e., to give a good presentation within the limits of the available computer screen dot pitch) and sufficient coverage. You can also decide here whether the curves should be cal'd (and mic-corrected, if appropriate).

Waterfalls are fascinating plots to make. They can provide very useful information about cabinet resonances and reflections, tweeter ringing, and other interesting phenomena. If configured carelessly, they can, however, make very impressive plots that mean nothing whatsoever.

OK. What is a "waterfall plot"? Basically, you start with a normal Frequency Response plot at the back of the graph, showing the spectrum (frequency output) the speaker would produce with a pure impulse input (truncated to avoid echoes, of course). Then you move the starting point of the time response to be transformed forward a bit in time and calculate the spectrum of this shorter time trace and plot it a little "closer" on the graph. The time axis is the one coming toward you on the plot. In effect you only include frequency content that is still present after the time as indicated on the time axis.

Now, if the transducer or speaker is perfect, its acoustical output, in response to an impulse input, is also a pure impulse. If the starting point of the trace to be FFT'd is then moved beyond the non-zero part of the impulse, all that's left is nothing, which has a spectrum of nothing. The plots from forward of this point then show no energy at all, which to avoid confusion (and to make the plot look cleaner), is represented as a straight line at the "floor" or minimum point of the graph (all values below the floor level are displayed at floor level). The spectrum decays in one steep drop from all to nothing.

Less perfect speakers will not have all their spectrum-producing time data concentrated in one instant -- so the response will decay more gradually. And if any components such as cabinets, cones, or dustcaps are causing ringing or poorly damped output, a decreasing ridge will appear at the resonating frequency (note: long NON-decreasing ridges can mean stray pick up of spurious signals, computer-generated EMI, or 60/120/180/etc. Hz hum).

Be aware that any delayed responses such as echoes or excess noise will have frequency content that occurs very late and will show as a raised floor in the measurement. When configuring a Waterfall plot, be sure to adjust the time domain markers properly for anechoic response: marker 1 set to just before the first arrival of the impulse response at the mic, and marker 2 set to just before the first significant echo pulse.

,Etc: (MLS instrument only). Energy-Time Curves (ETCs) show what is essentially a time domain analog of a decibel scaled frequency domain plot: the energy, both transferred and stored, shown as a function of time. The curve is generated from full-room impulse response data, which must be present in channel 1 before using ETC. ETCs can be used to investigate reflections such as can occur within a horn or in a room. ETC can be calculated for full band (up to as high a frequency as is passed at the sample rate) or for an octave band centered at a chosen frequency. For longest displayed decay curves and for lowest noise floors, choose a sample rate, if possible, just slightly above twice the maximum frequency of interest for the measurement (in other words, don't use a 48kHz sample rate if you are looking at octave band ETCs under 10kHz). To improve the noise immunity of the test, you should configure the impulse response measurement for an average of at least 10 acquisitions (this can be configured from the [System Avg] menu). A basic ETC measurement process is provided in the Easy Scripts.

The slope (decibel decrease per second) of an ETC display of an impulse response from a room excited by a loudspeaker is indicative of the reverberation time of the room (RT60, or the time taken for it to drop 60 dB). The impulse response should preferably be made using MLS at 16,383 points (be sure to set Acq_size, under the System menu to 16k), and far from the speaker. To use Audiosuite to read RT60 from such a plot, click the mouse twice within the ETC display frame. A line will appear connecting the large plotted peak (the direct pulse) to the mouse position, and LAUD will display a value of the RT60 which is equivalent to that slope. Move and click the mouse so that the line is parallel to most of the peaks in the portion of the curve which you are analyzing to read RT60.

Note that ETCs are "non-causal". In other words, there *will* be activity in the plot just to the left of the main "initial arrival" peak. This is not an error, but the reasons for it are not easily explained within this manual and will not be attempted.

Some of the activity at the very beginning of the plot (if it is about the same level as at the very end of the plot when fully displayed) may be due to the time-periodic nature of the MLS process. Such an effect *is* due to an error, and is typically indicative of reverberation time greatly in access of the acquisition duration (size/sample rate); usually RT60 determination will still be possible, but if you desire, a longer duration can be achieved by decreasing sample rate and re-acquiring the data.

,cePstr: A Power Cepstral response facility for loudspeaker echo and driver character investigations. The power cepstrum is calculated from the current <u>frequency response</u>, rather than from the impulse response as in the case of the Energy Time Curve.

This MLS instrument utility displays the Power Cepstrum of an acoustical frequency response curve. The Cepstrum (an anagram of the word "spectrum") is used to detect the presence of echoes or reflections by analyzing the patterns of frequency response variations. Peaks in the Cepstrum will correspond to late arrivals or reflections in a loudspeaker's radiation. These late arrivals could be due to edge diffraction from the cabinet, grilles or baffle hardware, from reflections in waves carried by a driver cone, from reflections in a horn, etc. There is evidence that the internal and nearby reflections in drivers contribute significantly to the audible differences reported in listening tests performed between different drivers, and may play a key role in explaining origins of "horn" sound, "electrostatic" sound, "cone midrange" sound, etc.

To use this, first have the frequency response data set in memory. The frequency response must be in FFT format, i.e., linear frequency format, as is obtained from an FFT or MLS instrument measurement.

Then adjust the display gain (use [F8]) so that the top mostly-flat region of the response curve is centered on the 0dB line. You must NOT be in SPL display mode to be able to do this. Then from the Cepstr menu, select analyze. The power cepstrum will be computed and displayed. Any Impedance data which is in memory will we overwritten by this operation and will be lost if it is not first saved. You can adjust the span of the resulting display (the x-axis or "quefrency", in units of time) and modify the relative vertical display power gain. A "hpFilter" option can be selected to remove much of the effect of a drivers' general response shape from masking low-level details in the Cepstrum at the low-quefrency (leftmost) part of the plot. Additionally, you can select whether the “negative frequency” part (normally neglected) of the source frequency response data is to be used in calculating the cepstrum via a “Use_negf” option-- results are often more comprehensible when this is set to “No”.

Frequency response data originating from SIZE values of over 2048 points will be preprocessed to provide cepstrum results equivalent to that from a 2048 point file. This is done to avoid the very long processing times which might otherwise result from the rather complex cepstrum calculations on very large data files.

The original reference on Cepstral analysis is:
B.P. Bogert et al., “The Quefrency Alanysis of Time Series for Echoes: Cepstrum, Psuedo-Autocovariance, Cross-cepstrum and Saphe Cracking”, Proceedings of the Symposium on Time Series Analysis, M. Rosenblatt, Editor, Wiley, NY, 1963, pp 209-243.

Other essential reading:
Holland, Keith R., “The Use of Cepstral Analysis in the Interpretation of Loudspeaker Frequency Response Measurements”, Proceedings of the Institute of Acoustics, Volume 15, Part 7, 1993, pp 65-71.

Also see:
Newell, Phillip and Keith Holland, “Round the Horn”, in Speaker Builder Magazine, 8/94, pp 24-36.

...and an upcoming article in Speaker Builder magazine about Cepstral Analysis and LAUD.

,moRe: (MLS instrument only) Provides other choices under the Transform menu.

,,Swap_channels: This option allows you to swap the time domain data in channel 1 for that in channel 2 and vice-versa. This might be useful when using LAUD in non-standard ways, such as getting acoustic data acquired from the mic into channel 2 for normalization of other data. It is provided for general versatility.

,,Hilbert: Use only on clean “FFT” format data! It operates on the response magnitude in the frequency response block using the Hilbert Transform to provide an estimate of the minimum-phase response which corresponds. If display smoothing is active, it will be applied (permanently) to the response before transformation. The frequency response phase data is replaced with the minimum phase version, so you may want to save it first. You can IFFT the result, if you want to see what the minimum phase version of the pulse would look like -- some investigators feel that performing this process before making a waterfall plot may provide a more intuitive picture relating to subjective sound quality. Note that single drivers are almost always minimum phase, while multi-way speaker systems are almost never. You can find the effective distance from a driver to a microphone (and hence, the driver's acoustical position) by finding the delay which must be subtracted from a frequency response delay plot to get it to match the Hilbert Transform of the same data; or you can use [Display Show More> Distance] in the MLS instrument to do it for you.

A discussion of the Hilbert Transform in any detail is beyond the scope of this guide, but information can be found in many digital signal processing textbooks such as Discrete-Time Signal Processing by Oppenheim and Schafer.

,,Xfer: Full name is "Transfer Function Product": this feature displays the result obtained by multiplying (rather than dividing) the frequency response by the cal. This is primarily intended as an aid in adjusting parametric equalizers to help deal with listening room problems. It will operate only in single channel mode and can process only data which is in FFT (linear frequency spacing) format. In practical use, both the Cal and the Measured frequency response data should be obtained consecutively with the same sample rate and size settings.

This process may now be more easily accomplished in most cases by using the 1/6th Octave RTA and Pink noise source of the Spec_an instrument.

The way this is meant to be used is as follows: using single channel mode (use [%] if necessary), first MLS acquire and FFT the response, complete with reflections, of each speaker as heard by the mic from the listening position. Declare this data as the Cal. Now configure LAUD (MLS instrument) to make a measurement of electrical frequency response of your equalizer using the probes. DON'T use a power amplifier with this part of the test! Select [* Transform moRe Xfer] and you will see the curve of the raw system (data in Cal block), the curve of the equalizer (Frequency response block) and the product curve you would get with the equalizer connected into your system. The Xfer Product curve is each point of the Measured frequency response plot times the corresponding Cal point. Using this, you can adjust the equalizer easily and quietly without making zillions of acoustic measurements during the process. When you've made the best adjustment you can, put the equalizer into the system and verify the performance with a LAUD MLS dual channel measurement.

,,conVert: This is the data conversion facility for frequency response or impedance data. Its function is to allow you to convert data which currently exists in one format (values given at a certain set of frequency points) into equivalent data in another format (i.e., at another set of points). A major use for this is in conversion of data generated by an FFT (which always produces constantly spaced frequency points) into data at “log spaced” points at frequencies each of which are at a constant multiple of the preceding point. This feature effectively negates the much-touted advantage of sinewave-based measuring systems in providing the more efficient “log” spacing of data -- LAUD provides the fast, accurate results obtainable with MLS along with the capability of reporting the results in log, linear or arbitrary frequency format. Of course, LAUD also provides the SINE based instrument as well, and can further convert the SINE obtained data into a format which can be IFFT’d to generate the time impulse response or Cepstral response.

The conversion is most easily understood if you consider the true data as consisting of a continuous curve (or set of curves, for complex data). At any frequency which the curve covers, a value exists (ohms, dB, degrees). The computer cannot, of course, store the data of a continuous curve, as a curve includes an infinite number of points. Neither can LAUD measure data at such an infinite number of points. But discrete data will be smooth if viewed in enough detail, and if the values of a limited set of data points are known, others nearby can be determined with reasonable accuracy.

Liberty Audiosuite uses a set of known data from the four closest frequency points around the frequency point of the unknown value to generate a third order curve fit. From the cubic equation for this curve, the data value of this unknown point is calculated. If the frequency of the point lies within the frequencies of the known points, the process is known as an “interpolation”. If it lies outside the four points (this will happen only when the unknown point is at a frequency higher or lower than all known points), it is called “extrapolation”. Obviously, interpolation is a more accurate and reliable operation than extrapolation.

The curve fitting routines in LAUD have been found to be very reliable with real data, provided some precautions are heeded:

1) Do not extrapolate far beyond known data, as LAUD will have only vaguely related data on which to base its calculation; the result will then be little more than a guess and may show abrupt aberrations.
2) Avoid multiple conversions on the same data set (repeatedly converting the results from previous conversions). Each conversion will introduce some error which, although usually small, will successively increase.
3) Be sure that the original data is adequately dense for a valid conversion. Clearly, an interpolation cannot uncover a series of peaks and dips, for example, if it is given only

one data point on either side of the area in question. Typically, this will be more of a problem when going from “log” formatted data to a “linear” (FFT type) set.

,,,Freq_response: chooses the conversion to operate on the current frequency response data (if any). See submenu selections below “Impedance”, below.

,,,Impedance: chooses conversion of impedance data (if any is in memory after a measurement or disk file read).

There are three possible sources from which to indicate the new target data point frequencies, data for Log frequency format, data for Lin (FFT) format, and a Freq_list (which is an ASCII disk file of target frequencies).

,,,,Log: chooses the current data to be converted to log formatted data (constant frequency multiple) format, such as 100 Hz, 110 Hz, 121 Hz, 133.1 Hz (100, 100*1.1, 100*1.1*1.1, 100*1.1*1.1*1.1,.....). The following submenu allows you to define the first frequency in the set, the last frequency, and the number of data points in the set; use the [Ok] option to back up a menu.

,,,,liN: chooses linear, or FFT format. The number of data points will be taken from the “SIZE” parameter (changeable via the [F2] key), and the effective sample rate will come from either the system “Rate” ([F3]), or a sPecified value entered by the user. The points will be at frequencies of 0 Hz (not readable on screen), sample_rate/size, 2*sample_rate/size, 3*sample_rate/size, etc. Data resulting from this conversion can be IFFT’d using that option in the Transform menu of the MLS instrument.

,,,,Freq_list: indicates to LAUD that you wish the target frequencies to be read from a disk file. The file must be in ASCII format, and the first string on each line must be a desired target frequency. The frequencies MUST be in increasing order, the first frequency being the lowest. Exceptions are any line in which an exclamation point (!) appears, which line will be ignored. Further numbers or characters on each line after the frequency will have no effect and need not be removed; hence “.FRD” or “.ZMA” data files written by LAUD or IMP or “.DAT” microphone correction data can be read in for use as Freq_list points. These lists can also be easily generated on most spreadsheets or text editors to match desired values for speaker design software packages.

,,Merge: (two menus from top in MLS; one menu from top in SINE). This facility allows you to combine, sum, subtract, multiply (cascade), divide (normalize), paste together or redraw sections of frequency response data into a new composite response data set. This facility is usable only for editing frequency response curves, not impedance curves. If any Impedance data which you wish to preserve is in memory, you should first save it, for the Merge operation will overwrite it in memory.

For a discussion on the concepts and operation of Merge, refer to Chapter XI “Merge Operations”.

,,,Paste: brings you to a menu which displays the three curves (as discussed in the Chapter XI coverage of Merge) and allows you to paste part or all of the current frequency response data set into the Merge space. Any previous merge data which existed in the frequencies pasted will be replaced with the data from the frequency response data.

,,,Sum: lets you combine the arithmetic sum of a selection portion of the frequency response curve with that in the same frequencies of the Merge curve. This will have the same result as might occur from the output of two loudspeaker drivers summing acoustically.

,,,Diff: similar to Sum, but the selected portion of the frequency response curve is first shifted by 180 degrees before the summation.

,,,Mult: in the selected range, the result would be like that of two cascaded filter stages. This would be equivalent to <u>adding</u> the dB values.

,,,diV: The result is that of normalization by the frequency response data. Like Mult, but equivalent to subtracting the dB values of the selected portion of the frequency response curve from those of the existing Merge curve.

,,,Clear: clear the Merge space, in order to start from an empty data set.

,,,draW: puts you into drawing mode.

,,,Freqs: this is used to choose the target frequencies upon which all Merge operations will be conducted and on which the final resulting data set will lie.

Acquire: These are the operations which control and activate direct collection of data from the DSP card into computer memory for display and/or further processing. These menus differ considerably between the MLS instrument and the SINE instrument, and with the exception of the Set_levels and Look options, will be covered separately below. When transforming data obtained from the MLS instrument, a “half-Blackman” (.BLACKMAN) window is advised. When acquiring data from the SINE instrument, in which a transformation automatically occurs, use of the full Blackman or Hamming window is advised.

,Set_levels: This allows the input and output gains of the DSP card to be manually adjusted for optimum data acquisition. You can get to this menu quickly using the **[-]** key, or by clicking on the button at the lower right of the screen which displays current values for the Main input channel (“M”), the Cal input channel (“C”) and the Output level (“O”) setting.

These adjustments directly affect the resolution, dynamic range and distortion levels of the data conversions. The input gains for both channels should be set optimally for within about 6dB of full scale (i.e., so that the input does not exceed the framed display area in the time domain plot with gain=1, or of the display given by the “Look” option). If the gains cannot be so adjusted below clipping, acceptable levels can be usually obtained by using the attenuator setting switches on the “Mic/Probe Preamp”. The output levels can in most cases be set more freely (but should be kept below a value of minus 6dB to avoid clipping in the DSP card output circuitry). “Look” will do a test run of the data acquisition while showing the acquired data in the time domain to allow you to check for proper level adjustment. **Note that for impedance measurements, the input levels “Main” and “Cal” (and their corresponding coarse attenuator settings on the MIC/PROBE preamp) must be set to the same values, the levels being set to accommodate the stronger of the two input signals. If this is not the case, the Levels button near the lower right part of the screen will be colored red instead of the normal gray-blue (assuming default colors).**

Set_levels also now contains an **automatic input gain adjustment** (“Auto_adj”) facility, which can be used to more quickly and easily set these values. This can be quickly activated by the [Shift-F10] key combination.

,Look: Used to check the input gain level settings (see discussion under "Set_levels").

,Collect: (MLS only) Causes an impulse or MLS data acquisition to commence. If MLS is active, the "Fast Hadamard Transform" will be performed to convert the result to an impulse response; in either case, the end result will be a time-domain impulse response data set ready for FFTing, or other processing. If dual channel mode is on, two impulse responses will be obtained, one from each channel. The levels should be properly adjusted before using this to avoid measurement error.

IMP users should note that the old "Repeat" function is now called "Look".

In manually conducted measurements, "Collect" should be used initially before using the Auto_Measure functions in order to obtain a plot on which to set the time domain markers. If you are making quasi-anechoic response measurements of a loudspeaker, you can often use the "Auto_tmkrs" (automatic time marker setting) function to set the time domain markers to the positions before and after the impulse response. Once you have set the levels properly and have set the time domain markers as desired (or enabled automatic adjustment of these) and also chosen the appropriate single or dual-channel mode, you can then use the Auto_Measure functions to easily and automatically perform all the acquisition and conversion steps for successive similar measurements.

,Mls: (MLS only) Chooses the stimulus type for this instrument: a Sixteen-k (16383 point) Maximum Length Sequence, a Four-k MLS or Off, which is a simple semi-sinx/x impulse stimulus.

,Freqs: (SINE only) Provides submenus by which you can indicate to the SINE instrument at what frequencies you wish it to collect data.

There are three possible sources from which to indicate the data point frequencies, Log frequency format, Lin (FFT) format, and a Freq_list (which is an ASCII disk file of target frequencies).

,,Log: chooses log formatted data (constant frequency multiple) format, such as 100 Hz, 110 Hz, 121 Hz, 133.1 Hz (100, 100*1.1, 100*1.1*1.1, 100*1.1*1.1*1.1,.....). The following submenu allows you to define the first frequency in the set, the last frequency, and the number of data points in the set; use the [Ok] option to back up a menu.

,,liN: chooses linear, or FFT format. The number of data points will be taken from the "SIZE" parameter (changeable via the [F2] key), and the effective sample rate will come from either the system "Rate" ([F3]), or a sPecified value entered by the user. The points will be at frequencies of: {sample_rate/size, 2*sample_rate/size, 3*sample_rate/size, etc.}. Data resulting from this data type can be IFFT'd using the IFFT option in the Transform menu of the MLS instrument.

,,Freq_list: indicates to LAUD that you wish the target frequencies to be read from a disk file. The file must be in ASCII format, and the first string on each line must be a desired target frequency. The frequencies MUST be in increasing order, the first frequency being the lowest. Exceptions are any line in which an exclamation point (!) appears, which line will be ignored. Further numbers or characters on each line after the frequency will have no effect and need not be removed; hence ".FRD" or ".ZMA" data files written by LAUD or IMP or ".DAT" microphone correction data can be read in for use as Freq_list points. These lists can also be easily generated on most spreadsheets or text editors to match desired values for speaker design software packages.

,Modes: (SINE only) This allows you to choose between two different SINE measurement modes for impedance or frequency response; there are also options to enable use of gating for frequency response measurements. The commands to bring up the mode change menus are “Zmode” for impedance and “Resp_mode” for frequency response.

,,Gating: (SINE only) This parameter affects only SINE instrument frequency response measurements. When “On”, gating will acquire and measure only the response which occurs between the two time markers (viewable by selecting the time-domain display via [F2]). This is generally advised only for acoustic measurements when quasi-anechoic results are desired. As discussed in the introductory sections, the easiest way to set the time markers properly (to mark off the anechoic portion) is to first obtain the time domain impulse response via the MLS instrument, mark off the range from just before the pulse response begins to just before the first visible reflection, and then switch back to the SINE instrument for the measurement. If the marked range is shorter than one cycle of a frequency to be measured, the computer will “beep” during the measurement, and the program will advance to the first practical requested frequency. An Easy Script performing this process is built into LAUD version 2, which can be used as-is or followed as an example.

When gating is “Off”, SINE instrument acquisitions will be of ‘SIZE’ ([F2]) time domain data points in length. If the period SIZE/Sample_rate is shorter than one cycle of a frequency to be measured, the computer will “beep” during the measurement to indicate inadequate measurement time for the test frequency, and the program will advance to the first practical requested frequency in the set. Gating will override the Resp_mode setting.

,,Zmode and **,,Resp_mode**: The “Use_size/rate” setting will perform the measurement using the same data size and sample rates as are displayed on the [F2] and [F3] buttons at the top of the screen. **If “Gated” is on, the Resp_mode setting will have no effect.**

The “Optimized” mode chooses the effective SIZE and the sample rate for each data point so as to minimize measurement time without compromising accuracy; this is further optimized by processing a fixed number of cycles of the test frequency (i.e., a “constant-Q” type of measurement). This number of cycles can be set by the user when the optimized modes are enabled. In nearly all cases, the **“Optimized” mode is recommended over the Use_size/rate mode** for best and fastest results.

The optimized mode should however be used only in dual channel mode (with a Cal probe). This is because the uncal’d phase data obtained may become non-continuous and invalid for the single channel, optimized sample rate, case.

,Sweep: (SINE only) This is the command which causes the indicated SINE measurement to begin. The result will be frequency domain transfer function response or impedance data. If dual channel mode is On (use the [%] key or click on the lower left “channels” button to change), each data point acquired by the main channel will be normalized with the corresponding point from the CAL channel *as each measurement is made* (but the microphone correction, if used, will be applied overall after the sweep is complete). If the length of each data acquisition is shorter than one cycle of a frequency to be measured, the computer will “beep” during the measurement, and the program will advance to the first practical requested frequency in the set. Similarly, if a requested data frequency is above approximately 0.45 times the indicated sample rate, an error “beep” will be sounded.

,Auto_Tmkrs: (MLS only) This is used to instruct LAUD2 to locate the leading edge and/or the echo edge in an impulse response which is acquired using the MLS instrument. Such setting of the time domain markers selects the portion within which is to be transformed in an FFT. This is useful for automating measurement of the anechoic response pof a loudspeaker by selecting only the anechoic portion (that which exists before the first echo arrives). The effectiveness of this automated action depends in large part on how well separated in time are the main activity of the impulse response and the first echo. If the speaker is still ringing or generating significant pressure variations when the echo arrives, the Auto_tmkrs option may not be able to separate them. Be sure the microphone and speaker placement is well arranged to provide as much echo-free acquisition time as possible.

Removing the time of flight of the pulse (via the “First_edge” option, which moves marker 1 to the leading edge of the impulse response) can be used to remove noise which is picked up by the mic before the pulse (or MLS burst) arrives. It will also remove the delay (phase roll) in the measured response; however, because of filter delays in the DSP card channels which become mismatched when all delay in the Measurement channel (Ch1) is removed (but not in the Cal channel), the resulting phase curve will often show negative delay (phase rising with frequency). To approximately compensate for this, you can then apply a delay correction (via [F9]) of <u>minus</u> (61000/sample_rate) {milliseconds}.

If “Echo_edge” is selected, the leading edge of the first echo will be located (marker 2 will be moved to that point). If this option is selected, the delay parameter will be set to compensate for the time of flight delay (with Marker 1 remaining at its leftmost position). The “Echo_edge” option is used to eliminate the effects of echoes from the measured response.

“Both_edges” is used to mark both the leading edge (as in “First_edge”) and the Echo edge. This is the best setting to use when making waterfall plots, as it eliminates both the time of flight (which gives no information in the waterfall) and the echoes (which limit the “floor” in the waterfall plots).

The setting can be made to apply Now or on the next impulse response acquisition. If this feature is not desired (such as for full-room measurements or for frequency response measurements of an electronic device) it must must be disabled by selecting “Off”. Using the “Both_edges” option is particularly useful when using [* auto_Measure Freqresponse] in the MLS instrument with “cYcling” enabled, to work on loudspeaker response adjustments.

System: Allows the user to save or restore a system configuration file, along with any user-defined macro operations; port designations for the printer and settings for the DSP card; selection of number of stimuli to average for each input (MLS only); selection of the color palette used in the display; choice between single and dual channel mode; the desired data acquisition size; the time gap between successive measurements; and entry of probe resistance values for optimum impedance measurement accuracy and of test resistor values (duplicating the option provided in the [Display Format Scale] and [auto_Measure T/s Setup] menus).

,Cfg_file: Here you can save or restore a Liberty Audiosuite configuration file for the MLS and SINE instruments. The configuration file contains information about the current display settings, operation modes, screen colors, and any defined user macros, along with the name of the Mic_dat file in use. This allows you to create for later reuse custom test settings for different functions. For instance, you may want to save a special configuration for measuring crossover curves, a different one for tuning enclosure ports, one for working with diffraction, one for setting equalizers, etc.

The data in the Time, Frequency or Cal memory is NOT changed or saved by these operations. The configuration file "STANDARD.ICF", if present in the same directory from which LAUD is run, will be automatically loaded at initialization. If you want to have

LAUD come up in the present settings, just save the configuration under the name "STANDARD" (any .ICF file named "standard" will always be automatically stored in the same directory as the LAUD .EXE file to simplify the process).

You can start LAUD from DOS to use a predefined configuration file (in the /LAUD directory) by following the command word LAUD with the name of the configuration file, such as:
C:\LAUD\: LAUD MYCONFIG

You may also select the configuration files being saved or loaded to be of the **"Hardware Independent"** types. These type configuration files (or normal files loaded as such) are arranged so as not to alter certain system characteristics such as the DSP card type, Mic correction file name, preamp gains, etc. This makes these configuration files more portable to another LAUD system which may be using different hardware.

You may also elect to save to or load from configuration files which are **in_Script_directories.** (Configuration files loaded from Scripts must always be in the directory of the calling script and are loaded using the normal "Use" command. Configurations loaded from a Script are also always loaded as "Hardware Independent" types, so that the scripts can be shared or published for use by other LAUD users.

,Ports: Allows you to select the printer (LPT# parallel only) port for printing of plots, to read the version of the ASIC chip on your sound card or check for its presence and proper installation, and set (if necessary) the address which will be used to address the CODEC (the combined stereo analog-to-digital and digital-to-analog chip). The CODEC address is normally that used by the "Windows Sound System", and is typically set to 530 (hexadecimal).

In addition, you can here tell LAUD which type of (PSA compatible) DSP card you are using. The Orchid types differ from the Cardinal and ECHO types in that the Orchid boards use the "LINE" input to the A/D and the others use the "AUX" input (although on the back of the card, they are labeled "Line In"). If the wrong type is selected, you will be able to make sounds with LAUD, but not make any readings.

,Avg {MLS only}: Select how many acquisitions of time data are to be averaged together each instance [Acquire Collect] or one of the auto_Measure operations is summoned using the MLS instrument.

,coLor: Lets you choose the colors which LAUD will use for the screen graphics. After choosing which of text, curves, background or fill to change, you can use the up or down arrows to cycle through the colors and immediately see the effect (note that the descriptions refer to the most common usage in the MLS and SINE instruments, and all colors may not be present at the same time with any particular screen). You can make the text or trace colors the same as the background colors, making the screen totally unreadable --if you get in that kind of trouble, hit [Esc] and then [Alt-C] to restore the default screen colors. The new colors can be saved with your configuration file using the [Setup Config] selection.

This sub-menu also allows the user to select monochrome (B/w) rather than color display for use with other screen-to-printer copy routines than the ones included in LAUD's [Display Printplot] menu, or for use with computers with monochrome screens.

,Z_set: Provides several options concerning impedance measurements.

,,probZ: Enter the value of the series resistor which is inside the probes (in the “hot” lead). This will normally be 47.5K ohms for use with the recommended MIC/PROBE preamp.

The value probeZ is used by the software in calculation of measured impedance values to take into account the effects of probe loading, and will typically have little effect when using an impedance reference resistor (the one connected in line with the device under test, not the probe resistor) of under a few hundred ohms. It is essential, however, to enter the correct value for probeZ if high (kilohm) impedances are being measured using a high value of reference resistor.

,,Set_bal: This is where you can declare your “Bal” data, which will be copied from the current frequency response data (if it exists and is cal’d). **This process is NOT recommended in systems which use the ECHO DSP/mod or other DSP card modified for extended low-frequency response.** Bal data is similar to Cal data in that it is used to correct responses for errors in the equipment. Cal data is a correction file for an individual measurement’s frequency domain response errors, and is (for best accuracy) refreshed with each measurement. Bal (balance) data, on the other hand is a correction file for the frequency domain differences between the two channels at lower frequencies. If needed, Bal data set should be made when LAUD is initially installed and need only be changed if you change probes, your DSP card or your mic/probe preamp. For frequency response measurements, Bal data is not used, as the correction is quite small and affects only sub-audible frequencies. Small channel differences, however, can significantly affect infrasonic impedance measurements, which can be important in accurately determining Thiele and Small parameters.

The Bal data you are setting should be the result of a two-channel frequency response measurement made with both probes measuring the same signal (usually from a power amp output). The measurement is best made in SINE, optimized mode (for low frequency noise immunity), and should contain low frequencies down to 4 Hz. **Bal is meant only for use with cards which roll off the low frequencies above the normal 20Hz audio band cutoff.**

,,Use_bal: Here you inform LAUD whether to use the Bal data to correct impedance measurements. Using the Bal data will make the measurement take slightly longer. It is usually only needed when using impedance measurements to determine low frequency (<60 Hz) Thiele and Small parameters and with cards which have compromised low frequency response, and can otherwise be disabled.

,Inpts: lets you select single or dual channel mode (as an alternative to using the [%] key or the channels button) or select the nominal data acquisition size. The nominal data acquisition size is relevant primarily for the MLS instrument, and should be set to Four-k (as a time saver) unless large 16k data will be utilized, such as for ETCs.

In addition, this is where you inform LAUD whether you will be using an external MIC preamp (feeding into the LINE/AUX input to the DSP card), or the DSP card’s MIC preamp. An external preamp (such as the Liberty Instruments designed “MIC/PROBE PREAMP”) is recommended for most accurate results.

,timeGap: this is a parameter which allows you to insert a delay between data acquisitions, to allow echoes to die down in a room, or just to slow things down in general. If this is not needed (as for impedance or non-acoustic data), set this to zero for quickest operation. The timegap parameter has units of time, but the delay will vary with the speed of the computer; therefore the timegap value must be set empirically. Typical values will be within 0 to 400.

auto_Measure: The Auto__measure routines provide a means by which relatively complicated measurement procedures may be performed with minimal operations by the user. In addition to built-in routines, there is also provision for the user to create and store his own “User Auto-measure” procedures (macros) for simple and efficient repeated measurement processes. For more complex operations, a Script can instead be programmed for much greater versatility and ease of use.

Facilities are provided for automatic extraction of Thiele and Small parameters from impedance data, including fully automated data acquisition and extraction capability. A particularly useful variant which is provided for some of the built-in Automeasure functions is “cycling”, which repeatedly makes certain frequency response or impedance measurements, showing only the final results; with MLS operation and well-chosen SIZE and RATE, this can be somewhat like watching frequency response or impedance in real-time, while tuning is performed. Some examples of this are given in the Easy Scripts (such as the Loudspeaker MLS Anechoic Response Measurement script).

,Setup: Not to be confused with the main Setup menu directly under the main heading! Choose windowing or cal operations to be performed during auto_Measurements. Also can select the "cycling" option, which gives continuously updated display of measured frequency or impedance data ("acous", "elect" and "impedance. You may also choose whether to enable **automatic input level setting** for the measurements. (Note: these settings do not apply to User defined auto_Measurements.)

(MLS only:) Note that Auto_time_markers, if enabled in the Acquire menu will also apply during auto_Measure Freq_response operation. You should check this setting before proceeding.

,,Windows: choose whether one of the six windows should be used when time domain data is transformed to frequency domain data. The Blackman window is particularly good for SINE based measurements. For pulse or MLS testing, NO window (other than the inherent rectangular one) or one of the "one-sided" windows, such as ".BLACKMAN", is advised.

,,Cal_source {MLS, single channel ONLY}: choose whether cal should be used, and if so, whether it should be re-acquired each time or just reused from the previous data. This makes sense only for single channel mode (mode may be changed using [%]);

,,cYcling: when set on, the auto_measure acous, electric and impedance functions will continuously repeat and update when used, which is very useful for real-time system adjustments! Only the final display of each auto_measure operation will be updated on the screen, to give an animated effect. Hold down [Esc] to break the cycle. Be sure to turn cycling off when you are finished with it. For fastest update, it is best to keep SIZE low and smoothing off when using this in MLS mode.

,,Auto_in_levs: enables automatic input level adjustment before each auto_Measure operation. This makes operation of the system much more convenient and foolproof.

,User: This submenu allows you to create your own macros which can later be executed by pressing the **F2** through **F5** keys while holding down the [Alt] key. A Macro is a collection of keystrokes (up to approximately 100, roughly twice as long as in LAUD version 1) which can be “played back” using a key combination. The macros, once created, are be saved and restored along with any configuration file using the [Setup Config_file] submenu. Each macro can have a title or description which is entered after the macro is recorded and which can be seen by using [List].

,,Record: Records user-defined auto_Measure or "macro" operations. When record is enabled and the macro number is selected, LAUD will record your keystrokes for later reuse. Select the number of the macro to record by pressing the 2, 3, 4 or 5 key (NOT the function keys!). "1" selects "Boot", which defines any operations which you wish to have occur when the LAUD program is first brought up each time (with the macro defined in the configuration file “STANDARD.CFG”. The macro can be executed later by pressing 'Alt' and [F2], [F3], [F4] or [F5] respectively (DO use the function keys when executing!). Certain operations are disallowed (or won't record) when making a user macro:

- Calling HELP
- Quit
- Starting another macro record
- Operations resulting in error messages

When macros are executed, they will cease on any errors encountered. Certain operations within a macro which respond to the user pressing a key, will also wait for a key when the macro is "played back". If the recorded macro is too long, or if any disallowed operations occur, three beeps will be heard, macro recording will cease, and the macro keystrokes will be discarded (any old macro which was being overwritten will be restored).

To end macro recording, press [End] and enter the description. To abort macro recording (and not save it), press [Ctrl-End].

,,List: Gives a list of defined user auto_measure macros, and their descriptions. Use this to see what user macros you may have defined. The descriptions may be edited by pressing the appropriate number.

,Freq_response: Makes a frequency response measurement. Be sure to set gains, levels and time-domain marker settings (for anechoic or gated operation) beforehand, or enable the automatic setting facilities, for the proper results.

,Impedance: performs an impedance measurement. Impedance is always a dual-channel operation.

,Impulse: (MLS only) Makes a time-domain impulse response measurement. This is provided to allow cYcling (repeated automatic measurements for “live” adjustments) in impulse response measurement, which is useful for distance aligning multiple-driver loudspeaker arrays.

(reserved for T/s picture)

,T/s: This submenu is for loudspeaker work. It extracts the Thiele and Small parameters of a driver from a series of two impedance curves.

The Thiele-and-Small parameter extraction facility allows you to use LAUD to measure the parameters Qe, Qm, Qt, and fs quickly and easily. The Vas parameter can be measured using a series of two extractions, via either the closed- box method or the delta-Mass method. From these, the parameters Bl (the "B-L product" in Tesla-meters), Cms, Mms, and reference efficiency are calculated and displayed. For detailed descriptions of these methods and T/S measurements in general, see The Loudspeaker Design Cookbook by Vane Dickason or the original JAES paper by Thiele. Either the SINE or the MLS instruments may be used for T/S extraction.

Because of the limited very-low frequency response of some of the DSP cards, there may be difficulties in measuring T/S parameters of woofers with resonances below 20 Hz. In those cases, the SINE instrument along with restricting measured frequency points to only those above 10-15 Hz (such as: Log_f type, 50 points from 15 Hz to 100 Hz) will generally give better results.

,,vas_Method: Select the Vas determination method you wish to use. If you choose the box method, you must enter the box volume; the box must be totally enclosed and must have no lining or stuffing. Because of likely degrading effects on the measurement of any air leaks in either the box or in the driver itself, the added mass method generally is easier to perform and provides better results. When using the mass method, you must specify the mass which you are adding to the cone (usually done with a ring of modeling clay placed on the cone, which is positioned face-upward).

If the delta-mass method is used, the percent shift in the resonant frequency will be reported; the clay mass used should be such that this value is between 15% and 35%, for best accuracy.

After Vas is determined when using the box method, a value will be reported for the mass ratio in air vs. in the box. According to Thiele, this value should be between 0.8 and 0.95 if the measurement was good; otherwise suspect air leaks, too large a box, or improper drive levels.

,,Extract: Used to extract T/S parameters from impedance curves which have already been measured (use "full_Auto" for measuring impedance and extracting all in one operation). Use [new_Driver] to clear out any existing curve-fit data when you begin with a driver. [Normal] is for impedance curves measured with the driver in free-air; [Loaded] is for the driver loaded by the mass or the test box, depending on the Vas method being used. Both extractions must be performed WITHOUT doing [new_Driver] between them, in order to get a value for Vas. LAUD will show the curve of the model which it fits to your impedance data, so that you can evaluate the precision of the fit. If the curves are not well matched in the region of the reported resonance, do not trust the reported values but look for problems in the measurement or setup.

,,full_Auto: Used to measure impedance curves and extract the T/S parameters of drivers with a minimum of keystrokes. Use [new_Driver] to clear out any existing curve-fit data when you begin with a driver. [Normal] is for the configuration with the driver in free-air; [Loaded] is for the driver loaded by the mass or the test box, depending on the Vas method being used. Both extractions must be performed WITHOUT doing [new_Driver] between them, in order to get a value for Vas. LAUD will show the curve of the model which it fits to your impedance data, so that you can evaluate the precision of the fit.

,,Setup: Use this submenu to define your driver diameter in inches or centimeters and the measured DC resistance of the driver (these values are required for the parameter determinations). Alternately, you can request LAUD to estimate (extract) the DC resistance from the impedance curve.

The two values "**Ref_resistor**" (the series resistor used between the power amplifier and a device for which the impedance is to be measured), and "**Ohms/div**" (the scale factor used for impedance curve displays) for convenience have also been made changeable from this setup submenu, as well as from the [Display Format] submenu.

Ctrl: These are menus for use in writing or running custom scripts or for production control purposes. They also include alternate ways to change certain system parameters which are normally controlled in a cyclic fashion (such as the SIZE [F2] button).

,sCripts: For launching Easy or Custom scripts.

,Lim_fil: Allows you to declare an upper and/or lower limit file to be used in making Pass/Fail evaluations for production control. For limit files to be used in evaluating Thiele/Small parameter measurements, a separate selection is provided for loading the necessary combined upper/lower ".LTS" file type. You may also select here the Script directory in which the files may be found.

,sHow: This shows the measured data of the type selected by "tst_Mode" in the [Ctrl setup1] submenu along with the limit curves from the limit files, and **if setEval is enabled, performs a Pass/Fail evaluation**.

,beeP: Inserts a "Beep" tone, which can be used to get the user's attention.

,Ext: Allows you to control or wait for external hardware (such as a rotation table), usually from within a script. Communication occurs through a parallel (printer) port.

,,select_Port: Choose the port through which external communications will be effected. It is strongly recommended that this NOT be the same port to which your printer is connected. Extra printer port cards for your computer can be obtained a very low cost from most computer shops.

,,Signal_out: Lets you send an 8-bit data word out to the printer port (using its designated "data" , rather than control bits).

,,wait_for_Hi: used to configure the computer to wait until a logic High is detected on data bit 0 of the designated parallel port. The wait can be immediate or to occur just before the next acquisition.

,,wait_for Lo: similar to the above description, but substitute "logic Low".

,setup1: allows you to cause a **Pause** of desired duration or to **pRint** a text line or a formfeed to the current printer.

,,Tst_Mode: from this menu, you may select the type of test which will be performed when **sHow** is executed. Choices are Frequency Response (**Freq_resp**), Impedance, and Thiele/Small parameters (**T&s**).

,,setEval: Choose if a Pass/Fail evaluation should be performed when [Cntrl sHow] is executed. Enter 1 for evaluation=on, 0 for evaluation=off.

,,setAdj: Choose whether **in Frequency Response evaluations only**, LAUD will be allowed to try to adjust the vertical display gain to try to fit the measured curve between the two limit files. This is for use when only frequency response shape (or perhaps, flatness) are being tested rather than sensitivity. To be meaningful, the setEval parameter must also be enabled (=1) and both an upper and a lower frequency response limit file must be declared.

,setup2: alternate methods of explicitly adjusting various system parameters, such as SIZE, RATE, INPUT source, data WINDOW, input gains, output gain, and number of channels.

Quit: You are asked if you are sure, and if you answer yes, you are returned to DOS.

VII. OSCOPE/GEN (SCOPE) OPERATION

(reserved for picture “oscope/gen screen”)

OSCOPE/GEN OPERATION

The oscilloscope/generator (or OSCOPE/GEN, or just "SCOPE") is a real-time, digital, dual-channel triggered oscilloscope and sinewave/squarewave generator. Both of these functions operate simultaneously so that the generator can be used to drive a circuit or device while the oscilloscope is used to monitor the response. All OSCOPE/GEN settings are displayed on the SCOPE screen and on printed plots.

As an oscilloscope, SCOPE provides a flexible scaleable display with user controllable Horizontal scale (the Hor sweep rate, in units of time per division), vertical gain scales (in units of Volts per division for each channel), and vertical position (expressed relative to midscale and as a percentage of screen height). Triggering can be disabled, single-shot, or continuous, can occur on a rising or falling edge crossing a user determined level, and can respond to either channel1 or channel2.

Limitations

As a generator, SCOPE generates low distortion sinewaves at user defined frequencies (which can also be easily stepped up or down on a musical scale ratio) and at adjustable levels for output from the DSP card's line and power amplifier outputs. Additionally, squarewaves can be selected as the output waveform, or the generator can be turned off. Because of the use of a audio DSP card for the generation, all output signals are bandlimited to under 22 kHz; in practical terms, this means that any squarewave above approximately 1/3 of the current sample rate will essentially contain only the fundamental, i.e., a sinewave. Also, all outputs are AC coupled, that is, are symmetrical in voltage with respect to ground. Hence, the squarewave output is good for stimulating an audio circuit, but not really for, say, driving a digital logic circuit.

Similarly, as an oscilloscope this instrument is also AC coupled, and cannot be used to measure DC voltage levels. The input spectrum displayed by the oscilloscope is bandlimited to approximately 0.45 times the current sample rate (about 22 kHz for the highest 48kHz rate). As opposed to a traditional analogue 'scope for which the cutoff frequency is usually a -3dB sensitivity point above which signals can still be seen (although attenuated and phase shifted), the LAUD SCOPE will show essentially no usable information above its cutoff point. The frequency limit, although linear in phase, is absolute in amplitude. This is a result of the sampled nature of DSP card inputs and their use of digital antialiasing filters. An additional artifact of the sampled input format is that the display is in a "connect the dots" pattern. For instance, if you used a sample rate of 16kHz (one sample every 62.5 usec) with a display horizontal scale of 50 usec/div, you would have many fewer actual measured points horizontally than would make up the screen width; Audiosuite will draw straight lines between the measured points that are available. Care should be exercised in interpreting such displays-- sharp edges in the trace may be artifacts of the sampled input or the sampled display (because the screen can also only display on exact pixels, not between them) and not in the actual data.

Operation

The SCOPE instrument of Liberty Audiosuite is different than the other instruments in that it operates essentially continuously. While active, it is always acquiring data and, if triggered, redrawing the display. This necessitates some changes in the way the instrument is operated.

The most significant difference is that the SCOPE instrument does not respond to the mouse or display the mouse cursor. This is because of conflicts that would occur between the mouse interrupts and cursor and the data acquisition timing and graphic updates. User control of the SCOPE is instead accomplished via the function keys.

When a control is selected with a function key, the related parameter can be typed in on the keyboard or the up/down arrows can be used to change the parameter in convenient preset increments. In some cases, submenus are given to allow control of further parameters. For example, the [F10] key brings up the generator control parameters and allows the waveform, level or frequency of the signal generator to be set. To complete a parameter change (and make the change, if typing in a value), press [Enter]. If you are again changing the most recently controlled parameter, pressing the up or down keys will bring it up for readjustment (i.e., the function key sequence need not be repeated).

If a "menu" is shown, or if you are being prompted for a value, pressing a function key or an [Alt-_] key combination will not immediately actuate the desired new function: in this case, press it twice, or use [Esc] first, followed by the desired key or combination. Additionally, each key press will not register until the end of a display sweep. For very long sweeps, this can be somewhat disconcerting if you are not expecting it; for rapid trace sweeps, it will not generally be noticeable.

To change from the SCOPE instrument to another instrument, use the [Alt] key and the appropriate letter for the instrument (underlined on the buttons at the bottom of the screen). To quit Audiosuite from the SCOPE, use [Alt-Q].

Adjustments and Parameter Settings

The settings of the SCOPE (and other LAUD instruments), as will be described below in the Function Keys section, will be saved in any configuration file written from the SINE or MLS instruments' [System Config_file] commands. If the configuration file is named "STANDARD" before being saved, the settings and calibrations of the SCOPE will be automatically loaded when Liberty Audiosuite is started.

Function and Control Keys

The active function keys include normal function keys ([F1] through [F10]), and several [Shift]ed or [Alt]ed keys.

[Esc]: This is used to interrupt operations or abort a parameter change which is not yet completed. In some cases, this must be pressed twice.

[Ins]: This key will toggle on/off the signal generator output, if the generator waveform is set to sine or square.

[F1] or [Shift F1]: Help. This brings up the help screen (the information you are now reading) for the SCOPE instrument.

[F2]: Hor (-izontal sweep rate). This allows you to set the sweep rate in units of milliseconds per division. The value can be entered in by hand or can be incremented by using the [Up arrow] or [Down arrow] keys.

[F3]: Rate. Control of the acquisition sample rate. You may increment/decrement the rate by use of the [Up arrow] or [Down arrow] keys over the fourteen provided values (5.12kHz to 48kHz). A slower sample rate will result in more frequent screen updates and minimum delay between data sampling and display. A faster rate will provide better trace resolution and allow viewing of traces with higher frequency content. The rate can not be changed while the scope is in "hold" mode (from a triggered single sweep); in this case, change to continuous mode before changing the rate.

[F4]: Trig(ger). Provides a menu of choices concerning which channel, if any, will be used as the trigger source, what voltage value will be considered the triggering level and whether the triggering will occur on the rising or falling slope passing that value, as well as whether the triggering will occur continuously or only on a “single-shot” basis. When in single-shot mode, another triggered trace can be “armed” by again selecting [Sing].

[F5]: Vrt1. This is the control for the channel 1 vertical gain, in units of volts per division. Numerical values can be typed in or the [Up arrow] or [Down arrow] keys can be used for convenient scale changes. The accuracy of this setting will depend on proper values being supplied for the external attenuators (or attenuator switch settings on the Mic/Probe preamp), the preamp or DSP card effective “In-gain”, and possibly gain fine correction factors. For information on these adjustments, see the explanations for [Alt-F5], [Shift-F5] and the [In-gain] parameter in the [F9] menu.

[F6]: Vrt2. This is the control for the channel 2 vertical gain, in units of volts per division. Numerical values can be typed in or the [Up arrow] or [Down arrow] keys can be used for convenient scale changes. The accuracy of this setting will depend on proper values being supplied for the external attenuators (or attenuator switch settings on the Mic/Probe preamp), the preamp or DSP card effective “In-gain”, and possibly gain fine correction factors. For information on these adjustments, see the explanations for [Alt-F6], [Shift-F6] and the [In-gain] parameter in the [F9] menu.

[F7]: Vpos1. Vertical position for the channel 1 trace. This can be entered numerically or altered by the [Up arrow] or [Down arrow] keys. The value is expressed as percent of display frame vertical size. In other words, 0% is in the center of the frame, +50% is the top of the frame, and -50% is the bottom. Vpos1 is normally set at +25% for dual channel operation and 0% for single channel operation.

[F8]: Vpos2. Vertical position for the channel 2 trace. This can be entered numerically or altered by the [Up arrow] or [Down arrow] keys. The value is expressed as percent of display frame vertical size. In other words, 0% is in the center of the frame, +50% is the top of the frame, and -50% is the bottom. Vpos2 is normally set at -25% for dual channel operation and 0% for single channel operation.

[F9]: Disp[lay]. Provides a means to control whether you wish to display channel 1 alone, channel 2 alone or both channels.

Additionally includes a setting [In-gain] for you to adjust accuracy for variations in general input sensitivity. This difference may be present due to differences in DSP card brands, or in external preamplification/input protection devices. The value entered is scalar, and is best adjusted experimentally for a proper trace amplitude reading when sampling an external known source signal with one of the probes.

[**F10]: Gen(erator).** Includes controls for the built-in sinewave/squarewave generator. These are provided on a menu as follows:

[Freq]: The fundamental frequency of the desired signal. Remember that the sample rate must be at least 1/0.45 times any generated frequency. For convenience (and maybe even for tuning guitars!) the frequency can be changed in musical scale increments by use of the [Up arrow] or [Down arrow] keys. Alternately, the frequency value can of course be typed.

[Line_level]: Allows setting of the peak signal level from the line output. To be numerically accurate, the parameter [Gain_l] may require adjustment for your particular sound card. Note also that the frequency response variations of the DSP card will reduce this level slightly (usually below about 80 Hz). Changing the [Line_level] will also change the [Amp_level].

[Amp_level]: Similar to the [Line_level] setting, but for the built-in power amplifier supplied on some cards. May require adjustment of the [gain_A] parameter. This setting interacts with the Line_level setting.

[Waveshape]: Choose from Sine, sQuare or None.

[(Gain_l)] or [(gain_A)]: adjustment to correct for different output gains (line and power amp, respectively) possible with different DSP cards. The values are scalar, and can be set by trial and error, if output accuracy is required, using external equipment or the LAUD scope (assuming its [In-gain] and or [alt-F5/6] adjustments have been properly set).

[Shift-F2]: Prints the screen to the printer. The configuration for the printer should be set up first, using the applicable menu in the SINE or MLS instruments.

[Alt-F5], [Alt-F6]: Provides fine (scalar) gain adjustments for the channel1 and channel2 input gains. These normally need not be set to any value other than 1.0, but are provided to allow correction for channel imbalances or for very critical measurements.

[Shift-F5], [Shift-F6]: This is where you can instruct the LAUD SCOPE to allow for external attenuators or attenuator settings on an external preamp. The value to be entered is in positive decibels (dB) of attenuation being used. For example, if the recommended MIC/PROBE preamp is being used, you would enter a value of “20” if the -20dB switch position is being used with the appropriate probe, or “40” for the -40dB position.

[Shift-F10]: This parameter can be set to indicate the maximum output voltage of which the internal DSP card power amp is capable. This is used only to alarm the user if he tries to set it beyond the amplifier’s capability. If you prefer to defeat this alarm, set the value to 30 Volts. Most cards, such as the Turtle Beach Fiji/Pinnacle or the Echo DSP, do not have on-board power amplifiers, so this will not apply when these cards are used.

<u>(rerved for specan screen shot)</u>

The SPEC_AN (Spectrum Analyzer) Instrument

The Liberty Instruments SPEC_AN instrument contains two separate types of spectrum analyzer instruments. They differ primarily in the way they divide and measure the frequency spectrum, and in the ways in which they are used.

FFT ANALYZER

The first type is an FFT-based spectrum analyzer and noise generator, useful for monitoring of the audio spectrum in signals sensed by the Main probe or microphone. This type of analyzer displays frequency data in bands which of constant width measured in Hz. The spectrum is reported at frequencies which are multiples of the value (sample_rate)/Size. Best resolution (but slowest operation) will result with a large SIZE parameter and the lowest usable sample rate, and leakage into other nearby frequencies (called “bins” in FFT jargon) can be minimized by use of the full BLACKMAN window.

White noise will show a flat response on this type of analyzer. The display can be set to user-determined frequency ranges and can create constant frequency increment formatted or log-frequency scaled plots. Measurements made from the probe (as determined by the “4:Input” button) are displayed in units of dBmV (decibels referred to 1mV). Those made from the microphone are displayed in dB SPL (no weighting). For the levels to be accurate, a valid Mic file must be loaded and the DSP card/preamp gain multiplier must be properly adjusted.

Note that as SIZE is increased, finer resolution bandwidth will result (acquisition size is limited to 8192 points). This will influence the way noise-like signals (like that from the built-in white noise generator) appear on the display. Finer resolution means each FFT bin is narrower, so that a noise signal will have less total energy in each bin (i.e., the total energy is evenly distributed over more bins). As SIZE is increased, the display level from a noise-sourced signal will drop. While watching a tone-based signal (periodics such as sine, square, sawtooth, for example) however, the displayed levels will not change appreciably as SIZE is changed. This is because a pure tone introduces constant energy into a bin in which it is the sole contributor, regardless of the bin bandwidth.

*******Caution! The white noise test signal contains considerable high frequency energy. Subjecting full-range loudspeakers to extended periods of high level white noise could cause heating and damage to drivers -- use caution and moderate levels with this stimulus. ********

A useful feature is the ability to perform power averaging of the spectrum, which shows the average power detected over the selected number of acquisitions. This can be used with varying microphone positions to make a determination of the average or typical response of an audio system within a room (although, in general, this is more easily accomplished using the RTA instrument and its “progressive averaging” capability). In order to aid in microphone positioning, a “beep” tone will sound right after each acquisition (if the [System Inputs Time_gap] parameter is greater than 0) to indicate when a position change can be made. The time gap delay between acquisitions is made quite a bit longer to provide more time for the user to reach the next position.

Measurements made with the SPEC-AN cannot be saved as data files (if a data file of a spectrum analysis is required, it could be achieved by using the MLS instrument with MLS=OFF and FFTing the result of an acquisition). Therefore, any graph which you wish to save should be printed before continuing.

RTA ANALYZER

The second LAUD spectrum analyzer is what is commonly called an “RTA” or “real-time analyzer”. The RTA name is not very descriptive of the distinction between this type of analyzer and the FFT type, but is traditionally used to indicate a spectrum analyzer which measures the spectrum in sections which are of constant *percentage* (or fractional octave) bandwidth. The RTA type of analyzers show a flat response when Pink noise is measured. The LAUD RTA can be configured as either the ubiquitous 1/3rd octave analyzer or as a higher resolution 1/6th octave analyzer.

The data processing (filtering) used in the LAUD RTA is a fast DFT (Discrete Fourier Transform) operation, rather than an average of FFT bins as is often used in DSP realized RTAs. The DFT technique provides a true constant percentage bandwidth characteristic to the measurement over all frequency bands. The processing rate of the RTA will be approximately proportional to the square of the <u>lowest</u> frequency band calculated and displayed; hence a display with a minimum frequency of 100Hz will update (on less speedy computers) about four times as fast as one which uses a minimum frequency of 50Hz. Similarly, use of 1/6th octave resolution requires approximately four times the processing time per update as when using 1/3rd octave resolution. These factors will probably not be of much concern, however, when using a fast Pentium based computer.

A,B, and C weighting functions are provided for display, along with a Max-hold function and support for curve math between currently measured data and a reference pre-declared “2nd curve”. The data can be displayed as a running average of up to the last 500 points (in 1/3rd octave mode), to smooth out the variations caused by using a noise stimulus yet still provide a constantly updating display. A Pink noise or white noise source can be selected for use as a stimulus. The use of pink noise as a stimulus is advantageous because the sound of Pink noise is much less bothersome than is white noise, and is also less stressful on devices being tested (particularly tweeters). The constant percentage bandwidth characteristic provides more consistent display resolution in a spectrum plot formatted in log-frequency format.

The RTA instrument can save and retrieve measured data to and from disk storage. The file data format is ASCII, allowing for simple export to other packages or spreadsheets. Data curve printouts made using the RTA can be configured to display at the top of the graph, the final measured value at each bar.   The center frequency of each bar is displayed at the bottom of the graph.

You can make an average of multiple measurements (including those made using running average mode), combining each into an on-going “progressive average” data set. This feature makes it very easy to measure, for example, an in-room response using multiple microphone placements, yielding an overall “average” response. The progressive average operation can be performed as each measurement is taken, or as a post-processing operation performed using retrieved data files.

SPEC_AN Applicable Function Keys : FFT Type

F2: Size. Select the size of the time data set which will be analyzed in each transformation operation. The higher the size is, the higher the resolution but the longer the time between screen updates. A very fast computer will make this less of a consideration.

F3: Rate. Increases the sample rate for the analysis. Higher rates mean more extended high frequency analysis, but less dense low frequency data (unless the size of the data set is made larger). [Shift-F3] will decrease the sample rate.

F4: Input. This selects either the channel 1 probe or the channel 1 mic (the SPEC_AN is a single channel instrument). This will also change the display mode between dBmV (decibels referenced to 1 millivolt) for the probe and dB SPL (sound pressure level) for the mic.

F7: Window. Selects the windowing function which will be applied to the input signal. The "**.**????" half windows are not practical here, so use the full Blackman, Hamming or Bingham windows.

F8: Gain. In the SPEC_AN, this button will allow setting of the top-of-screen display level, in dBmV (decibels referenced to 1 millivolt) for the probe and dB SPL (sound pressure level) for the mic. This assumes a valid mic data file has been loaded in (via the MLS or SINE instruments).

Shift-F2: Permits quick hardcopy printing of the display; use only if the printer has been properly set-up beforehand.

Shift-F3: Decreases the sample rate. See [F3].

Shift-F5, SHift-F6: Changes displayed frequency limits. No effect will be seen until "Acquire Continuous" is again executed.

Shift-F8: Allows you to change the display scale (dB/division). Will apply only to data acquired after the gain change.

Shift-F10: Performs automatic input level adjustment (a time domain display will then be displayed until another acquisition is performed).

<u>SPEC_AN Applicable Function Keys : RTA Type</u>

note: the SIZE and RATE keys are changeable within the RTA, but have no effect on the RTA measurement.

F4: Input. This selects either the channel 1 probe or the channel 1 mic (the SPEC_AN is a single channel instrument). This will also change the display mode between dBmV (decibels referenced to 1 millivolt) for the probe and dB SPL (sound pressure level) for the mic.

F8: Gain. In the SPEC_AN, this button will allow setting of the top-of-screen display level, in dBmV (decibels referenced to 1 millivolt) for the probe and dB SPL (sound pressure level) for the mic. This assumes a valid mic data file has been loaded in (via the MLS or SINE instruments).

Shift-F2: Permits quick hardcopy printing of the display; use only if the printer has been properly set-up beforehand.

Shift-F5, Shift-F6: Changes frequency limits for processing and display. For fastest update rate, select the lower limit no lower than required for each particular measurement

Shift-F8: Allows you to change the display scale (dB/division).

Shift-F10: Performs automatic input level adjustment (a time domain display will then be displayed until another acquisition is performed).

SPEC_AN Menu Descriptions (FFT type)
In these discussions, the number of commas preceding a menu option indicates how many levels down the option occurs from the top-of-menu [*] position. For example, " **,,View** " indicates that this choice is two selections or keystrokes from the menu top. **The menu descriptions for the RTA type follow afterwards.**

Type: used for changing between the FFT and the RTA spectrum analyzer.

Display: options for controlling the display format, hardcopy printing, and definition and display of an on-screen data title.

,Format: settings for the display frequency range limits, the scale (dB/division and top-of-screen levels), whether they should be in log-frequency or linear frequency format, and whether to display full bars for each frequency bin or just the tops of the bars (which makes printing less cumbersome in plots with dense data).

,Printplot: Make a graphic hardcopy of the screen and the current active title. Supported devices are Epson/IBM compatible matrix printers and HPLaserjet compatible printers including the Deskjet 500, 500C, 550, or 520. Additionally, "printing" to a Windows compatible bitmap file is supported. For the Epson/IBM types, choose one of modes 1-13 for best presentation (experiment to see which is best with your printer). Use "Go" to print from this menu or use [shift-F2]. The "Promo" mode causes printouts to be made which are the same as the screen display, useful for documentation and instruction figures.

,Title: controls a user defined on-screen title to identify displayed or printed plots.

,,View: shows the title on the screen. This also can be quickly accomplished by using the [space] bar.

,,Hide: use it to minimize screen clutter while working. The [space] key will toggle between view and hide whenever a menu line is present.

,,Edit: lets you change the Title. The [Tab] key also performs this function whenever a LAUD menu line is displayed. Editing is done from the right side of the line -- to change a part of the title that is not at the end of the title line, you must backspace (delete) all which follows it. To restore the title as it was before the current edit, press [Esc]. To clear the entire title, press [Home]. To save your changes (make them permanent and beyond instant restoration), press [Enter].

Acquire: controls the acquisition of data, whether it is power-averaged or continuous (each acquisition done fresh), the levels into the analog-digital converter and out of the digital-analog converter, and whether a pink or white noise source is enabled.

,Average: switches to power-averaging mode, and allows definition of the number of samples that will be averaged. After the number has been acquired and averaged (which begins by selecting "Go"), the analyzer will halt and hold the display for printing or for restarting. If the number of averages is not 1, the delay between acquisitions (provided so changes can be made between acquisitions) can be removed by setting the [* System timeGap] parameter to zero.

,Continuous: Each acquisition and analysis will be independent of the previous one, and there will be no extra delay injected between acquisitions. This will continue until interrupted the user interrupts with the [Esc] key.

,Set_levels: This allows the input and output gains of the DSP card to be adjusted for optimum data acquisition. The input gain (designated as “Main_in”) should be set to be sufficiently high without overdriving the input (i.e., does not exceed the framed display area in the time domain plot with gain=1, or of the display given by the “Look” option in the Set_levels submenu). If the gain cannot be so adjusted, acceptable levels can be usually obtained by using the attenuator setting switches on the “Mic/Probe Preamp”. The output levels can in most cases be set more freely (but should normally be kept below -6dB). “/Auto_adj” will automatically adjust the input gain of the DSP card. “/Look” will do a test run of the data acquisition while showing the acquired data in the time domain to allow you to check for proper level adjustment.

,Noise: Allows you to select whether the internal White or Pink (or oFf --none) noise generator is activated. This occurs immediately upon selection. An “Ok” option is also provided to backup one menu.

For a constant bandwidth analyzer, like the SPEC_AN’s FFT-Type, constant energy per band occurs for a signal which has a flat spectrum; hence the use of white noise is appropriate. A “flat” system will show a generally flat characteristic on the LAUD FFT SPEC_AN when reproducing white noise. However, be cautious of subjecting tweeters to continuous uninterrupted high level white noise, as it contains considerable high frequency energy.

,Look: Allows you to manually adjust levels for best use of the DSP card dynamic range by showing the acquisition in the time domain. In most cases, it is more convenient to use the Automatic Adjustment feature, which can be quickly performed using [Shift-F10].

System: settings for system parameters and for saving or loading a configuration file.

,Cfg_file: provides facilities for saving or loading a configuration file, similar to that in the MLS and SINE instruments. You may also select to load or save a configuration from one of the Script directories (rather than LAUD’s base directory), or to designate the save or load as being “hardware independent” -- meaning that parameters to a specific hardware installation such as mic correction file, DSP card type, etc., are not changed when the configuration is loaded. The “Init “ option allows you to quickly restore the STANDARD start-up configuration.

,Port: Allows you to select the printer (LPT# parallel only) port for printing of plots, to read the version of the ASIC chip on your sound card or check for its presence and proper installation, and set the address which will be used to address the CODEC (the combined stereo analog-to-digital and digital-to-analog chip). The CODEC address is normally that used by the “WINDOWS SOUND SYSTEM”, and is typically set to 530 (hexadecimal).

In addition, you can here tell LAUD which type of (PSA compatible) DSP card you are using, if not previously configured. The Orchid types differ from the Cardinal and Echo types in that the Orchid boards use the “LINE” input to the A/D and the others use the “AUX” input (although on the back of the card they are labeled “Line In”). If the wrong type is selected, you may be able to make sounds with LAUD, but not make any readings.

,Input_gain: This setting can by used to adjust for the DSP card and the external mic/probe preamp gain. The settings will track for both mic and probe if a Liberty

Instruments Mic/Probe preamp design is used (Mic gain = 54 dB relative to the probe level). The “Card&preamp” setting is best set as conducted in the software installation Easy Script.

,Ext_atten: This allows you to indicate to the SPEC_AN that extra attenuation has been introduced into the channel 1 input (the attenuation switch is off of the 0 dB position, using the Liberty Instruments Mic/Probe preamp design). Be sure to enter a <u>positive</u> number for a
negative gain (gain is the negative of attenuation). These settings can also be made by pressing the [+] special key.

,coLor: Lets you choose the colors which LAUD will use for the screen graphics. After choosing which of text, curves, background or fill to change, you can use the up or down arrows to cycle through the colors and immediately see the effect (note that the descriptions refer to the most common usage in the MLS and SINE instruments, and all colors may not be present at the same time with any particular screen). You can make the text or trace colors the same as the background colors, making the screen totally unreadable --if you get into that kind of trouble, hit [Esc] and then [Alt-C] to restore the default screen colors.

This sub-menu also allows the user to select monochrome (B/w) rather than color display for use with other screen-to-printer copy routines than the ones included in LAUD’s [Display Printplot] menu, or for use with computers with monochrome screens.

,timeGap: this is a parameter which allows you to insert a delay between data acquisitions, to allow echoes to die down in a room, or just to slow things down in general. In the SPEC_AN instrument, the resulting delay is made much larger than for the other instruments so that the user can move the microphone between acquisitions, if desired; when power averaging is “ON” and “Time_Gap” is non-zero, the microphone movement should be made right after the tone is heard from the computer (the tone signals that an acquisition has just completed and a pause is beginning). The timegap parameter has units of time, but the delay will vary with the speed of the computer; therefore the timegap value must be set empirically. Typical values will be within 20 to 400.

auto_Meas:
,Go: Restarts the acquisitions, if stopped. The mode (Average or Continuous) will be the last selected.

,User: Allows you to record a macro command or list the descriptions of the currently defined macro. Macros will begin by changing to the instrument in which they were generated, and are saved along with Configuration files.

Cntrl: These are menus for use in writing or running custom scripts. They include alternate ways to change certain system parameters which are normally controlled in a cyclic fashion (such as the SIZE [F2] button).

,sCripts: For launching Easy or Custom scripts.

,beeP: Inserts a “Beep” tone, which can be used to get the user’s attention.

,Ext: Allows you to control or wait for external hardware (such as a rotation table), usually from within a script. Communication occurs through a parallel (printer) port.

,,select_Port: Choose the port through which external communications will be effected. It is strongly recommended that this NOT be the same port to which your printer is connected.

Extra printer port cards for your computer can be obtained a very low cost from most computer shops.

,,Signal_out: Lets you send an 8-bit data word out to the printer port (using its designated "data" , rather than control bits).

,,wait_for_Hi: used to configure the computer to wait until a logic High is detected on data bit 0 of the designated parallel port. The wait can be immediate or to occur just before the next acquisition.

,,wait_for Lo: similar to the above description, but substitute "logic Low".

,setup1: in this instrument and type, allows you to cause a pause of desired duration or to print a text line or a formfeed to the current printer.

,setup2: alternate methods of explicitly adjusting various system parameters, such as SIZE, RATE, INPUT source, data WINDOW, input gains, output gain, and number of channels.

SPEC_AN Menu Descriptions (RTA)

In these discussions, the number of commas preceding a menu option indicates how many levels down the option occurs from the top-of-menu [*] position. For example, " **,,View** " indicates that this choice is two selections or keystrokes from the menu top.

Type: used for changing between the FFT and the RTA spectrum analyzer.

Display: options for controlling the display format, hardcopy printing, and definition and display of an on-screen data title.

,Format: settings for the display frequency range limits, the scale (dB/division and top-of-screen levels), whether weighting should be applied to the display, whether final data point readout text should be displayed for each bar, and whether the 2nd "reference" curve should be displayed along with the acquired data.

,Printplot: Make a graphic hardcopy of the screen and the current active title. Supported devices are Epson/IBM compatible matrix printers and HPLaserjet compatible printers including the Deskjet 500, 500C, 550, or 520. Additionally, "printing" to a Windows compatible bitmap file is supported. For the Epson/IBM types, choose one of modes 1-13 for best presentation (experiment to see which is best with your printer). Use "Go" to print from this menu or use [shift-F2]. The "Promo" mode causes printouts to be made which are the same as the screen display, useful for documentation and instruction figures.

,Title: controls a user defined on-screen title to identify displayed or printed plots.

,,View: shows the title on the screen. This also can be quickly accomplished by using the [space] bar.

,,Hide: use it to minimize screen clutter while working. The [space] key will toggle between view and hide whenever a menu line is present.

,,Edit: lets you change the Title. The [Tab] key also performs this function whenever a LAUD menu line is displayed. Editing is done from the right side of the line -- to change a part of the title that is not at the end of the title line, you must backspace (delete) all which
follows it. To restore the title as it was before the current edit, press [Esc]. To clear the entire title, press [Home]. To save your changes (make them permanent and beyond instant restoration), press [Enter].

,pRogressive_avg: controls for the progressive averaging process.

,,Clear: clears the current progressive average set.

,,Show_prog_av: displays the current contents of the progressive average set (caution: replaces the current acquired data set).

,,Include_in_av: includes the currently acquired data set into the progressive average. If there is currently no data in the progressive average set, initializes the set with the current acquired data.

File: options for saving or retrieving the current data set to disk, deleting disk files, changing the default directory (to which data will subsequently be saved or retrieved), and for loading a different set of microphone correction data.

Acq: controls the acquisition of data, the selection of RTA resolution, the Averaging mode settings, the levels into the analog-digital converter and out of the digital-analog converter, whether a pink or white noise source is enabled, and processing options (i.e., math operations using a separate “2nd” curve).

,Go: Begins RTA acquisition and display. Display will continue until interrupted by a keystroke (normally, the [Esc] key).

,Avg_mode: Note that this is different than the averaging options and technique used in the FFT type spectrum analyzer, and also different than the “progressive averaging” functions in this RTA.

,,Number: Choose how many power responses to average in each set (whether the set is a running average for a continuous display, or a fixed set of acquisitions which will cause acquisition to cease when achieved).

,,Running: Chooses an averaging mode which keeps a running average for continuous display. This smoothes out the random variations which come from sensing a noise signal. The **Number** parameter, when used in running average mode, effectively determines the attack and decay time of the RTA display. This can be plainly seen if you configure for a running average of number=10, look at the spectrum from a microphone source, and whistle near the microphone. There will be a delay before the peak due to the whistle appears, and it will take some time for it to disappear. However, with number=1 (no averaging), the effects will be more immediate, but the noise background will show greater random variations.

,,Fixed: Chooses an averaging mode which collects and averages the specified number of responses and then stops. This is most useful within scripts, where the fixed option will minimize the need for operator intervention.

,,Max_hld: The max-hold option, when enabled, keeps track of the largest value seen in each band and makes a composite curve of these values. During acquisition, both the max-hold curve and the currently measured response sample (or running average) are displayed. When the display updates cease (via user interruption in running mode, or completion in fixed mode), only the max hold curve will be displayed. If running average mode is on, the max hold will not begin to be evaluated until the parameter Number has been achieved (until that many curves have been collected).

You should keep in mind that the LAUD RTAs do not continually monitor the input signal, but look at it in sections. It is entirely possible for a large but very brief transient to occur but not be detected by the RTA, if the transient should occur at a time in which the input is not being sampled. This RTA is designed to be used for analysis of noise or responses using noise as a stimulus, but not to be used as a gunshot detector!

,Set_levels: This allows the input and output gains of the DSP card to be adjusted for optimum data acquisition. The input gain (designated as “Main_in”) should be set to be sufficiently high without overdriving the input (i.e., does not exceed the framed display area in the time domain plot with gain=1, or of the display given by the “Look” option in the Set_levels submenu). If the gain cannot be so adjusted, acceptable levels can be usually obtained by using the attenuator setting switches on the “Mic/Probe Preamp”. The output levels can in most cases be set more freely (but should normally be kept below -6dB). “/Auto_adj” will automatically adjust the input gain of the DSP card. “/Look” will do a test run of the data acquisition while showing the acquired data in the time domain to allow you to check for proper level adjustment.

,Noise: Allows you to select whether the internal White or Pink (or oFf --none) noise generator is activated. This occurs immediately upon selection. An “Ok” option is also provided to backup one menu.

For a fractional octave analyzer, like the SPEC_AN’s RTA-Type, constant energy per band occurs for a “pink” spectrum characteristic. A “flat” system will show a generally flat characteristic on the LAUD FFT SPEC_AN when reproducing pink noise. This is generally a more benign and pleasant stimulus than is white noise.

,Proc: The process option lets you declare (or set) the current RTA curve as the “2nd curve” (usable somewhat similarly to the Cal curve of the MLS and SINE instruments); to choose whether to Normalize measured curves with the second curve (subtract the value, in dBs, of the second curve from each); to Cascade measured curves with the second curve (add the value, in dBs, of the second curve to each); to turn such processing Off; and to choose whether to do the processing selected now, rather than on the next acquisition.

The user should be wary that Cascade and Normalize will usually cause a great change in the display characteristics. Normalizing a curve which is around 90dB SPL with one at 93dB SPL will result in one at -3dB, probably totally off the screen! You may need to use the auto-scale [=] key to bring the resulting curve back into display.

,Look: Allows you to manually adjust levels for best use of the DSP card dynamic range by showing the acquisition in the time domain. In most cases, it is more convenient to use the Automatic Adjustment feature, which can be quickly performed using [Shift-F10].

Sys: settings for system parameters and for saving or loading a configuration file.

,Cfg_file: provides facilities for saving or loading a configuration file, similar to that in the MLS and SINE instruments. You may also select to load or save a configuration from one of the Script directories (rather than LAUD’s base directory), or to designate the save or load as being “hardware independent” -- meaning that parameters to a specific hardware installation such as mic correction file, DSP card type, etc., are not changed when the configuration is loaded. The “Init “ option allows you to quickly restore the STANDARD start-up configuration.

,Port: Allows you to select the printer (LPT# parallel only) port for printing of plots, to read the version of the ASIC chip on your sound card or check for its presence and proper installation, and set the address which will be used to address the CODEC (the combined stereo analog-to-digital and digital-to-analog chip). The CODEC address is normally that used by the “WINDOWS SOUND SYSTEM”, and is typically set to 530 (hexadecimal).

In addition, you can here tell LAUD which type of (PSA compatible) DSP card you are using, if not previously configured. The Orchid types differ from the Cardinal and Echo types in that the Orchid boards use the “LINE” input to the A/D and the others use the “AUX” input (although on the back of the card they are labeled “Line In”). If the wrong type is selected, you may be able to make sounds with LAUD, but not make any readings.

,Input_gain: This setting can by used to adjust for the DSP card and the external mic/probe preamp gain. The settings will track for both mic and probe if a Liberty Instruments Mic/Probe preamp design is used (Mic gain = 54 dB relative to the probe level). The “Card&preamp” setting is best set as conducted in the software installation Easy Script.

,Ext_atten: This allows you to indicate to the SPEC_AN that extra attenuation has been introduced into the channel 1 input (the attenuation switch is off of the 0 dB position, using the Liberty Instruments Mic/Probe preamp design). Be sure to enter a <u>positive</u> number for a negative gain (gain is the negative of attenuation). These settings can also be made by pressing the [+] special key.

,coLor: Lets you choose the colors which LAUD will use for the screen graphics. After choosing which of text, curves, background or fill to change, you can use the up or down arrows to cycle through the colors and immediately see the effect (note that the descriptions refer to the most common usage in the MLS and SINE instruments, and all colors may not be present at the same time with any particular screen). You can make the text or trace colors the same as the background colors, making the screen totally unreadable --if you get into that kind of trouble, hit [Esc] and then [Alt-C] to restore the default screen colors.

This sub-menu also allows the user to select monochrome (B/w) rather than color display for use with other screen-to-printer copy routines than the ones included in LAUD’s [Display Printplot] menu, or for use with computers with monochrome screens.

,timeGap: this is a parameter which allows you to insert a delay between data acquisitions, to allow echoes to die down in a room, or just to slow things down in general. In the SPEC_AN instrument, the resulting delay is made much larger than for the other instruments so that the user can move the microphone between acquisitions, if desired; when power averaging is “ON” and “Time_Gap” is non-zero, the microphone movement should be made right after the tone is heard from the computer (the tone signals that an acquisition has just completed and a pause is beginning). The timegap parameter has units of time, but the

delay will vary with the speed of the computer; therefore the timegap value must be set empirically. Typical values will be within 20 to 400.

auto_Meas:
,Go: Restarts the acquisitions, if stopped. The mode (Average or Continuous) will be the last selected.

,User: Allows you to record a macro command or list the descriptions of the currently defined macro. Macros will begin by changing to the instrument in which they were generated, and are saved along with Configuration files.

Cntrl: These are menus for use in writing or running custom scripts. They include alternate ways to change certain system parameters which are normally controlled in a cyclic fashion (such as the SIZE [F2] button).

,sCripts: For launching Easy or Custom scripts.

,beeP: Inserts a “Beep” tone, which can be used to get the user’s attention.

,Ext: Allows you to control or wait for external hardware (such as a rotation table), usually from within a script. Communication occurs through a parallel (printer) port.

,,select_Port: Choose the port through which external communications will be effected. It is strongly recommended that this NOT be the same port to which your printer is connected. Extra printer port cards for your computer can be obtained a very low cost from most computer shops.

,,Signal_out: Lets you send an 8-bit data word out to the printer port (using its designated “data” , rather than control bits).

,,wait_for_Hi: used to configure the computer to wait until a logic High is detected on data bit 0 of the designated parallel port. The wait can be immediate or to occur just before the next acquisition.

,,wait_for Lo: similar to the above description, but substitute “logic Low”.

,setup1: in this instrument and type, allows you to cause a pause of desired duration or to print a text line or a formfeed to the current printer.

,setup2: alternate methods of explicitly adjusting various system parameters, such as SIZE, RATE, INPUT source, data WINDOW, input gains, output gain, and number of channels.

IX. THE DIST_AN INSTRUMENT

(reserved for dist_an screen)

The DIST_AN (Distortion Analyzer) Instrument

The Liberty Audiosuite DIST_AN is a versatile harmonic distortion analysis system, featuring both a low distortion sinewave source and a coherent wave analyzer. The measurement floor is normally limited by the DSP card circuitry to approximately 0.04%THD residual (lower for individual products). For minimum distortion residual, keep the output level of the DSP card to under approximately -7.5dB, as higher levels can cause excess distortion in driving the analog output amplifiers on the DSP card. You may need to experiment to find the optimum DSP card output level for your particular board.

In normal function, the output of the sinewave source (i.e., the DSP card line output) is applied to the unit being tested and the output of the tested device is routed back in to the DSP card's line input for analysis. The unit under test could be an amplifier (or other electronic processor) or a loudspeaker. If a loudspeaker is being tested, the DSP card line output is boosted by a power amplifier and the loudspeaker's output is sensed by a microphone. Under LAUD's control, the fundamental test frequency is stepped through the test frequencies specified by the operator and the output of the unit under test is analyzed for content at the harmonics of the applied fundamental. These harmonic levels (either individually or Total) are expressed as percent of the fundamental frequency level or as absolute levels.

Rather than the usual simple display of a single number for the total harmonic distortion for a given frequency, the LAUD DIST_AN plots curves of individual harmonics versus a range of fundamental frequencies. Total Harmonic distortion (all products within the audio band up to 10^{th} harmonic) can also be calculated and displayed as a function of the stimulus test frequency.

The display can be formatted so that the harmonic content is displayed as either a percentage of the fundamental level, or as an absolute level itself in dB volts or dB SPL.

This version also adds the ability to measure loudspeaker distortion under quasi-anechoic "gated" conditions. This technique, like the gated sinewave tests of the SINE instrument, allows echoes to be windowed out of a distortion measurement. When using the Gated mode of the DIST_AN, the gating window is determined by first setting the time domain markers of the MLS instrument. These markers are easily configured manually or automatically via the impulse response which is obtained in an MLS "Collect" operation.

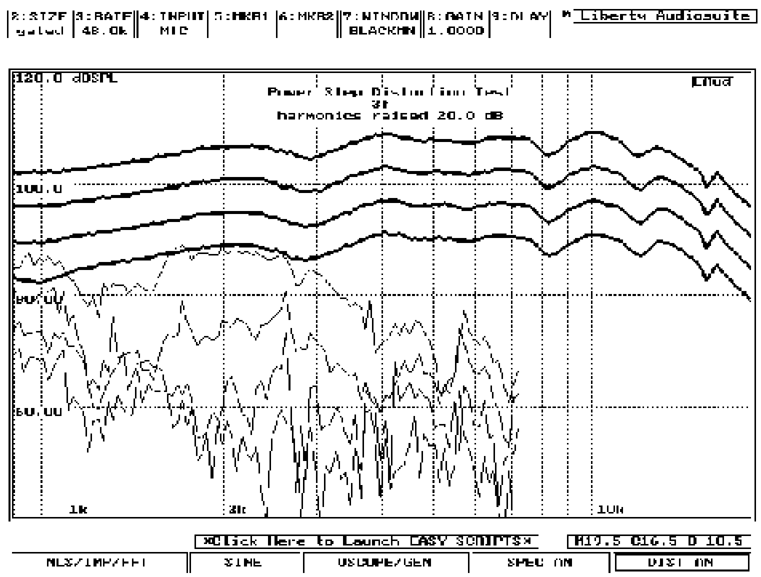
Not only the magnitude of the distortion products (expressed as percent of fundamental level) but also the relative phase of each product can be easily measured and displayed by the DIST_AN. The phases of the harmonics are expressed in degrees relative to the fundamental's phase; in other words, if the fundamental can be expressed as a waveform:

$A*\cos(2*\pi *f*t0)$

it will reach its positive peak when t0 is an integer multiple of the fundamental's period. Then a harmonic component with zero degrees shift will also be at its positive peak at these same instants in time; a harmonic with 180 degrees shift will be at its most negative peak at these time instants and so forth.

These curves, while being far more informative than the rudimentary results obtainable from older style (but much more expensive) instruments, still can cause difficulty in developing get intuition about the character of distortion types. A new tool, the **Distortion Visualizer**, has been developed and included within LAUD. The Visualizer first measures the THD (Total Harmonic Distortion) at the specified frequency and then displays this value along with a plot of the sinusoidal waveform. With most common distortion levels, the waveform is indistinguishable to the eye from a pure sine wave. The Visualizer lets you visually “turn up” the distortion (maintaining relative phases and levels between the products) to make the products’ effects on the waveshape visible and to get a feel for the character of the nonlinearities.

If a distortion test using an external sinewave source is required (at a single frequency), the THD analyzer of the Visualizer can be used to determine the total harmonic levels, provided the fundamental test frequency of the Visualizer is carefully set to match the external source.



A new addition within the auto_Measure menu of the DIST_AN is “Power Stepping”. This facility allows LAUD to generate a family of curves representing fundamental and/or harmonic levels as a function of both power level and frequencies, collected and displayed in a simple operation. Power stepped response or distortion measurements are useful in showing driver compression or distortion onset effects of loudspeakers. A high SPL-handling microphone may be needed to reveal the compression levels of many loudspeaker drivers.

(reserved for visualizer picture)

A Note About Levels:

LAUD is a digital audio based system, and therefore optimum distortion performance will depend on the utilization of the dynamic range of the analog/digital system. In order to get the best (lowest) residual distortion levels, the input sensitivities should be set so that that the acquired sine wave peaks are near full scale (as seen over the entire sweep using the "Look" option of the Set_levels menu), without clipping (or the computer giving an error beep while using the "Look" option). Typically, the automatic level setting feature can be used to quickly provide an input level very near the optimum.

Please see the note at the beginning of this chapter regarding optimum setting of the DSP card output level.

Measuring Loudspeaker Distortion:

When measuring the distortion of a loudspeaker, remember that the speed of sound is finite. When using a non-gated ("optimized") acquisition mode, you should therefore measure distortion using a relatively close-mic technique, as otherwise the unaccounted for "dead" time in the time-of flight from loudspeaker to microphone, if excessive, may affect the measurement. In "Gated" mode, the dead time as well as the arrival of echoes are removed by the setting of the time domain markers (from within the MLS instrument).

Additionally, consider the effects of echoes in the measurement: reflections cause ripples in the fundamental frequency response curve. Echoes also cause variations in the levels of the harmonics sensed by the microphone, but not in the same places or character as those in the fundamental. Such variations can have a strong effect on measured distortion curves for loudspeakers. Care should be taken to minimize echoes in the measurement, using near-field microphone placements, a low reflection measurement environment, or by use of Gating (for frequencies above approximately 1kHz);

Making Distortion Measurements

When using the Dist_an, you should first choose the measurement frequencies, which can be selected using the [Acq Freqs] menu. Usually Log-frequency spacing will be desired; if so, be sure to set the desired Number of frequency points to achieve the desired resolution without making the measurement process overly time consuming. Notice that because LAUD can measure frequencies only up to approximately 21kHz, there can be no distortion product measurements for fundamental (test) frequencies much above 10kHz (when the second harmonic goes outside the audio range). Similarly, measurements of third harmonic can not be made above a test frequency of approximately 7kHz.

When using "Optimized" acquisition mode, be sure that the numCycles parameter (value displayed on your F2 screen button) is set to the desired value; usually values greater than 25 yield no further improvement, except when measuring a device with a very irregular frequency response. When using "Gated" mode., be sure to first set the time markers to frame the anechoic portion of the impulse response from within the MLS instrument.

For electronic devices which you are sure will not put out large voltages (over about 5V peak), a lower distortion residual can sometimes be achieved by running the Device Under Test's output directly to the DSP card's input. But do this at your own risk: large voltage levels into the card can damage its acquisition circuitry.

Before making a measurement, you should first set your levels. These include:
-- the output level from the DSP card. A value of -9 dB is usually a good choice. Avoid using greater values (such as 0dB), as they can cause the cards output circuitry to clip. The DSP card output level and its input gains may be set manually via the [Acquire Set_levels] menu, or more quickly via using the [-] key or clicking on the blue "levels button" near the bottom right of the screen.
-- the output level of the unit being tested. For a power amplifier or a loudspeaker, this is done using an external volume control or low-distortion control preamp (if the amp has none). You can use [Acq Look] to view the output level (in volts, or in Pascals for microphone input), which will be displayed at the top of screen. However, this may require adequate preadjustment of the input gains to assure linear measurement, as described next. For reference and assuming a sinewave signal: 1Pascal = 94dBSPL, 0.5 Pascals=88dBSPL . To get from pressure in Pascals rms to dBSPL, take 20*log(pressure) and add 94.

- the input gains, including external attenuator settings. For high-level measurements of a power amplifier, you will need to switch your mic/probe preamp's attenuator (GAIN) switches to 40(-40dB gain) or 20(-20dB gain). **If the GAIN switches are changed, be sure to inform LAUD of their settings** (for proper level readouts); the menu for doing this is in [System Ext_attenuators] or can be quickly reached by using the [+] key. The value entered is an attenuation value, i.e., if the Mic/Probe preamp GAIN setting is "-20dB", enter the POSITIVE value "20".

Similarly LAUD should be informed of which Ch1 input is being used, the PROBE or the MIC; the setting may be toggled using the F4 key, or set from a script using the option in the [Cntrl setup2] menu.

Use [Acquire Set_levels Auto_adj] (or if desired, perform the adjustment manually) as required to be certain that the curve seen using [Acq Look] shows unclipped sinewaves. You may have to adjust input levels and output drive a few times iteratively in the case of large signals.

You may want to adjust the vertical display limits before the measurement ([Display Format Dist_range)], and the Level_scale (if the fundamental level will be displayed on the plot). You should definitely check, and if necessary adjust, the displayed frequency range (use [Shift-F5] and [Shift-F6]), as it is rather unsatisfying to not be able to see part of a measurement while it is being made.

After all the settings are made, use **sWeep** (in the top level menu or in the Acquire menu) to make the measurement.

DIST_AN Function Keys

F2: Size. This button cannot be used to change system parameters in this instrument, however it will display the currently selected number of cycles which will be used for each measurement. This value corresponds to the Q (selectivity quality factor) of the measurement. Fastest measurement times result for the lowest value of this parameter (which may be selected via the Acquire menu), but a minimum number is required to achieve a required distortion residual. Generally, there is no advantage to using much greater values than needed for a particular measurement, and the test time will suffer.

F4: Input. This selects either the channel 1 probe or the channel 1 mic (the DIST_AN is a single channel instrument).

F7: Window. Selects the windowing function which will be applied to the input signal. Either the Blackman, Hamming or Bingham window may be selected. In most cases, the Blackman window will be the best choice.

Shift-F5, Shift-F6: Changes upper and lower display limits.

Distortion Analyzer Menu Descriptions

Display: Selection of which harmonic levels will be analyzed, the display format, hardcopy generation and provisions for an on-screen data title.

,select_Harmonics: The display will show up to two relative harmonic levels at a time. The display is limited to two curves to avoid confusion, as the traces inevitably overlap each other; further individual harmonics can be analyzed on successive sweeps if desired. Harmonic numbers from second through 99th, or a single curve showing Total Harmonic Distortion, can be selected for the A and B curves.

In “Optimized” mode, measurement of higher level harmonics will usually take longer, as the sample rate will be chosen by LAUD to try to accomodate the measurement . Remember that no frequency can be measured which is above 0.45*sample_rate. Higher sample rates will require that more points be analyzed for the fundamental. For most devices, the second and third harmonics will dominate the distortion spectrum.

,Format:

,,freq_Range : Allows you to select the desired frequency range for the display (not for the measurement --the measurement frequency range is selected under the Acquire menu).

,,stYle : Choose between display of harmonics as a percentage of fundamental (first harmonic) level or as absolute levels (dB Volts or SPL). In “percentage” style displays, the test signal fundamental level measured curve can be displayed on the graph.

,,shoPhase : select whether the phase of the distortion products (relative to that of the fundamental) should be displayed.

,,Dist_range :

,,,Lower%, Upper% : For percentage style display, the upper and lower percent value for the display graph are selectable.

,,,dbOffset : For level style display, this selects the amount, in decibels, which the distortion products will be increased on the graph. This allows the distortion curves to be shown nearer to the fundamental response curve without requiring large dB/division scale values.

,,,dB/div : For level style displays, selects the vertical scale factor.

,,,Numdivs : For level style displays, selects the number of vertical divisions to be displayed (up to 10).

,,Level_scale : Selects the value of the top of the graph in Volts or in dB SPL, depending on the source (mic or probe) of the current distortion data, if any.

,Printplot: Make a graphic hardcopy of the screen and the current active title. Supported devices are Epson/IBM compatible matrix printers and HPLaserJet compatible printers including the DeskJet 500, 500C, 550, or 520. You may also choose to “print” to a Windows-compatible color or black-and-white bitmap file, should you be using a different type printer and have a graphic editor program available. For the Epson/IBM type printers, choose one of modes 1-13 for best presentation (experiment to see which is best with your printer). Use "Go" to print from this menu or use [shift-F2].

,Title: controls a user defined on-screen title to identify displayed or printed plots.

,,View: shows the title on the screen.

,,Hide: use it to minimize screen clutter while working. The [space] key will toggle between view and hide whenever a menu line is present.

,,Edit: lets you change the Title. The [Tab] key also performs this function whenever a menu line is present! Editing is done from the right side of the line -- to change a part of the title that is not at the end of the title line, you must backspace (delete) all which follows it. To restore the title as it was before the current edit, press [Esc]. To clear the entire title, press [Home]. To save your changes (make them permanent and beyond instant restoration), press enter.

,show_Level: used to select whether the voltage or SPL level of the fundamental (test) frequency is to be displayed on the “percent” style distortion sweep plots. If set to “oN”, the parameter “Level_scale” in the [Display Format] menu should be set to some value just above the largest expected fundamental level. In percent style display, the display of fundamental level will always be scaled at 20dB (10x voltage) per major division, the major divisions being those labeled for percent distortion. This option can also be used to select the value (dB volts or SPL) for the top line of the graph in “level” style distortion display.

Acq: options for level settings of the DSP card input and output; for the fundamental frequencies over which the distortion products will be measured; for the number of cycles at each frequency which will be analyzed; for the Mode (Optimized or Gated) of acquisition; a command to begin the swept measurement; and an option (“Look”) to view the time domain output signal to be analyzed, so that the desired drive level may be set.

,Set_levels: This allows the input and output gains of the DSP card to be adjusted for optimum data acquisition. These adjustments directly affect the resolution, dynamic range and distortion levels of the data conversions. The “/Auto_adj” option will automatically set the input levels; the success of this may depend on proper setting of the Mic/Probe Preamp’s attenuator settings. “/Look” will do a test run of the data acquisition while showing the acquired data in the time domain to allow you to check for proper level adjustment.

,Freqs: provides submenus by which you can indicate to the DIST_AN instrument at what frequencies you wish it to collect data.

There are three possible sources from which to obtain the data point frequencies, Log frequency format, Lin (FFT) format, and a Freq_list (which is an ASCII disk file of arbitrarily increasing target frequencies).

,,Log: chooses log formatted data (constant frequency multiple) format, such as 100 Hz, 110 Hz, 121 Hz, 133.1 Hz (100, 100*1.1, 100*1.1*1.1, 100*1.1*1.1*1.1,...). The following submenu allows you to define the first frequency in the set, the last frequency, and the number of data points in the set; use the [Ok] option to back up a menu.

,,liN: chooses linear, or FFT format. The number of data points will be taken from the "SIZE" parameter (changeable via the [F2] key), and the effective sample rate will come from either the system "Rate" ([F3]), or a sPecified value entered by the user. The points will be at frequencies of: sample_rate/size, 2*sample_rate/size, 3*sample_rate/size, etc.

,,Freq_list: indicates to LAUD that you wish the target frequencies to be read from a disk file. The file must be in ASCII format, and the first string on each line must be a desired target frequency. The frequencies MUST be in increasing order, the first frequency being the lowest. Exceptions are any line in which an exclamation point (!) appears in any position, in which case the entire line will be ignored. Further numbers or characters on each line after the frequency will have no effect and need not be removed; hence ".FRD" or ".ZMA" data files written by LAUD or IMP, or ".DAT" microphone correction data, can be read in for use as Freq_list points.

,Mode : selects whether the distortion measurement should be gated (for making quasi-anechoic measurements of a loudspeaker from a microphone); or Optimized (non-gated, which also allows the sample size and rate to be optimized for each test frequency).

For Gated mode, the time marker positions (of the MLS instrument) should first be set to frame the anechoic portion of the loudspeaker impulse response. See the discussions in the MLS and SINE instrument sections of this manual. In Gated mode, there will be a low frequency limit imposed on the measurement due to finite quasi-anechoic data length. In addition, there will also be distortion degradation of the acquired signal at low frequencies due to the windowing and limited data size. A practical minimum test frequency for Gated distortion measurements is approximately 1kHz.

,numCycles: numCycles: input the (whole) number of fundamental frequency cycles which should be acquired and analyzed for the frequency test point (normal distortion or visualizer measurement). If set too low, greater residual distortion levels will result; if set too high, the measurement will take considerably longer. In most cases, 5 to 30 cycles will be sufficient. The greater this number is, the more selective the measurement filters for each harmonic will be.

,sWeep: begins the measurement sweep. You can interrupt via the [Esc] key, if necessary.

,Look: Allows you to pre-check the drive level at the output of the Unit Under Test (as sensed by the probe or microphone) for adjusting input gains or for setting the UUT to a target level. The Peak-to-Peak voltage or pressure is displayed just above the top graph window.

System: settings for system parameters.

,Ports: Allows you to select the printer (LPT# parallel only) port for printing of plots, to read the version of the ASIC chip on your sound card or check for its presence and proper installation, and set the address which will be used to address the CODEC (the combined stereo analog-to-digital and digital-to-analog chip). The CODEC address is normally that used by the "WINDOWS SOUND SYSTEM", and is typically set to 530 (hexadecimal).

,coLor: Lets you choose the colors which LAUD will use for the screen graphics. After choosing which of text, curves, background or fill to change, you can use the up or down arrows to cycle through the colors and immediately see the effect (the descriptions given refer to the most common usage in the MLS and SINE instruments, and all colors may not be present at the same time with any particular screen). You can make the text or trace colors the same as the background colors, making the screen totally unreadable --if you get in that kind of trouble, hit [Esc] and then [Alt-C] to restore the default screen colors.

This sub-menu also allows the user to select monochrome (B/w) rather than color display for use with other screen-to-printer copy routines than the ones included in LAUD's [Display Printplot] menu, or for use with computers with monochrome screens (which is not generally recommended, but may be needed in some cases).

,timeGap: these are parameters which allow you to insert a delay between data acquisitions, to allow echoes to die down in a room, or just to slow things down in general. The timegap parameters have units of time, but the delay will vary with the speed of the computer; therefore the timegap values should be set empirically. Typical values will be within 0 to 400. In most cases, for distortion measurements (other than of loudspeakers at a distance) they should be set to zero. The **quietGap** parameter determines a delay between one measurement before the next stimulus is given, to allow ringing or reverberation to die down (this is seldom if ever used for distortion measurements). The **measDelay** parameter determines how long after the stimulus is applied before the measurement sensing will begin; this is provided to account for delays in the UUT or setup (such as time-of-flight, or of an FIR filter).

sWeep: begins a measurement sweep. Identical to the option of the same name under the [Acquire] menu.

auto_Meas :

,User : Allows generation of User macros.

,Go(regular_sweep) : same as sWeep in the main menu.

,Power_step : This facility allows LAUD to generate a family of curves representing fundamental and/or harmonic levels as a function of both power level and frequencies. This allows investigation of driver response or distortion changes as they relate to signal or power level. After the curves are generated, the plot graphics should be printed to paper or to a disk file. The data set can not be saved in this form to a retrievable data file, and only the last trace generated will remain in memory after the plot is finished.

For each curve generated, the output signal from the DSP card is reduced by the selected step size and a distortion measurement is performed over the set of test frequencies. If possible, the input gain of the card is also increased at the same time to optimize dynamic range; however, the gain can not be increased to more than the 22.5dB limit of the CODEC. The test should be begun with the output gain (and external volume controls) set for the maximum desired test level.

If possible, the input gain settings are best set to low values (but of 6 dB, minimum); but the "Look" facility of the Acquire menu must, as always, be used to assure that the DSP card is being driven within its dynamic range. Also provide for adequate headroom in the input gain setting should the gain of the device being tested increase (come out of compression) at lower drive levels during the stepping process.

Power Stepping while viewing harmonic products is usually practical only with the “level” display style (rather than the “percent” style). This is because the curves in the final plot become hard to distinguish or to relate to the proper drive level.

Often, the distortion levels will not drop evenly as the signal level is stepped downward, but will appear to cluster noisily around the same zone after the first few steps. This is due to the product levels approaching the resolution of the digitizing circuitry of the DSP card or related equipment, and to the increased influence of noise at the successively lower test levels. For this reason, stepping with distortion products displayed (rather than just showing the fundamental) is usually best done with only a few moderate-sized gain steps measured near the compression limit of the device being tested.

,,Step_size : Selects the dB step size by which the test level will be changed for each curve generated in the family.

,,Num_steps : Selects the number of steps to be generated (maximum 10).

,,Harmonic_num : Selects which harmonic will be displayed. “2” indicates second harmonic, “3” means third, etc. “1” indicates that the fundamental (only) should be displayed, and “0” indicates that Total Harmonic Distortion (THD) should be displayed.

,,show_f1? : Selects whether the fundamental level should be displayed. This is automatically selected when “Harmonic_num” is chosen to be “1”.

,,Go : Begin the power sweep. You can interrupt the process, if necessary, by pressing [Escape].

Visualize: the Distortion Visualizer. The Visualizer measures the THD (Total Harmonic Distortion) at the specified frequency and then displays this value along with a plot of the sinusoidal waveform. The test frequency should ideally be below 2kHz so that LAUD can measure an ample set of harmonics. With most common distortion levels, the waveform is indistinguishable to the eye from a pure sine wave. The Visualizer lets you visually “turn up” the distortion (maintaining relative phases and levels between the products) to make the products’ effects on the waveshape visible and to get a feel for the character of the nonlinearities. To repeat the process at a different frequency, merely change the [Frequency] option, select [Go] and analysis will automatically begin.

Before starting the Visualizer with [Go], you should check your levels using the special version of [Look] provided in this menu.

The Visualizer will show a picture of the sine waveform, a readout of the fundamental frequency, the measured Total Harmonic Distortion at that frequency, and the current value of the distortion multiplier. If the multiplier is 1, the displayed waveform is as it was acquired (up to the tenth harmonic, provided that harmonic frequency is below 21kHz). To change the multiplier by 1 dB, use the up or down arrow keys. To input a specific multiplier value, use the [dist_Multiplier] option. The displayed trace will change immediately. The displayed waveform amplitude will be adjusted for constant displayed height.

,Frequency: the desired fundamental frequency.

,dist_Multiplier: the value by which all the distortion products will be multiplied on the waveform display.

,Look: allows you to check the input level for the Visualizer instrument at the test frequency.

,Go: Begins a new analysis.

Ctrl: These are menus for use in writing or running custom scripts or for production control purposes. They also include alternate ways to change certain system parameters which are normally controlled in a cyclic fashion (such as the INPUT [F4] button).

,sCripts: For launching Easy or Custom scripts.

,Lim_fil: Allows you to declare an upper and/or lower limit file to be used in making Pass/Fail evaluations for production control. You may also select here the Script directory in which the files may be found.

,sHow: This shows the measured data along with the curves from the limit files, and **if setEval is enabled, performs a Pass/Fail evaluation**.

,beeP: Inserts a "Beep" tone, which can be used to get the user's attention.

,Ext: Allows you to control or wait for external hardware (such as a rotation table), usually from within a script. Communication occurs through a parallel (printer) port.

,,select_Port: Choose the port through which external communications will be effected. It is strongly recommended that this NOT be the same port to which your printer is connected. Extra printer port cards for your computer can be obtained a very low cost from most computer shops.

,,Signal_out: Lets you send an 8-bit data word out to the printer port (using its designated "data" , rather than control bits).

,,wait_for_Hi: used to configure the computer to wait until a logic High is detected on data bit 0 of the designated parallel port. The wait can be immediate or to occur just before the next acquisition.

,,wait_for Lo: similar to the above description, but substitute "logic Low".

,setup1: allows you to cause a **Pause** of desired duration or to **pRint** a text line or a formfeed to the current printer.

,,setEval: Choose if a Pass/Fail evaluation should be performed when [Cntrl sHow] is executed. Enter 1 for evaluation=on, 0 for evaluation=off.

,setup2: alternate methods of explicitly adjusting various system parameters, such as SIZE, RATE, INPUT source, data WINDOW, input gains, output gain, and number of channels.

Quit: exits Liberty Audiosuite and returns to DOS.

Production Control (and Pass/Fail) Testing

LAUD2 provides capability for production testing of magnitude of frequency response, impedance and distortion and for receiving inspection tests of the Thiele/Small parameters Qe, Qm, fs and Vas. Automated Pass/Fail tests can be configured by writing custom scripts to run the measurements and by providing upper/lower limit data files to be used for evaluating the result as a pass (within limits) or fail (out of limits). Such tests can also be conducted manually (without a script), although such an approach usually requires a higher level of production test operator training.

The writing of custom scripts is described in the chapter on Scripts. After the script makes a measurement on the device or circuit, the script should use the Cntrl menu of the appropriate instrument to:
-- define the test Mode (if in SINE or MLS instrument , where there are several options). This is done in the [* Cntrl setup1] menu. The test mode defines what type of data will be evaluated.
-- activate the Evaluation operation, if Pass/Fail operation is required, by setting the "setEval" parameter (in the [* Cntrl setup1] menu) to "1". Additionally, if the test mode is Frequency Response and only response shape (rather than absolute gain) is to be evaluated, you may want to also activate the "setAdj" parameter (set it to "1") which allows LAUD2 to attempt to fit a measured frequency response curve within the limit lines by moving it up or down vertically on the display.
-- tell LAUD2 the names of the limit files which are to be used for the evaluation. The upper limit file defines the highest the measured data can reach at each frequency in a given range and still be accepted (Passed), and the lower limit file defines the lowest it can reach to still be accepted. This is done using [* Cntrl Lim_fil]. The Thiele/Small parameter limit files contain both the upper and lower limits in a single file.
-- use the [Cntrl sHow] option to cause the measured data (of the chosen mode) to be displayed, along with the limit curves. If "setEval" is active, LAUD will also make a pass/fail judgment and display an "accepted" or "out of limits" message, along with a corresponding indicator sound from the computer speaker.

Creating Limit Files

All of the limit file types use ASCII format. These files can be created using any text editor (such as Windows 3.1 Notepad or DOS's 'Edit'), or in some cases can be generated by LAUD2's ASCII [File Save ...Ascii] options. The required extensions for the file names of the limit files are as given below; for example an upper distortion limit text file might be named **MAXDIST.LDS**

Limit File Type	File Extension Name
Frequency Response	**.FRD**
Impedance	**.ZMA**
Distortion	**.LDS**
Thiele/Small Parameters	**.LTS**

IMPORTANT! When using limit files from within a Script, the limit files must be in the test script's subdirectory (such as in "C:\LAUD2\SCRIPTS\TESTS"). After creating the limit files using the text editor or via LAUD measurement, be sure to move the files to the script's directory. This must be done using DOS commands at the DOS prompt or via the Windows 3.1 File Manager or Windows 95's Explorer. The limit files can still be tested in an evaluation performed in manual (non-script) mode by selecting the script directory of the limit file when the limit files are chosen from the [Cntrl] menus.

Frequency Response and Impedance limit files:

For frequency response and impedance, the file formats are similar. Two files can be supplied for each test, one for the Upper Limit curve and one for the Lower Limit curve. The first text line of each file is generally ignored and can therefore be used to provide a very brief comment or description; you should begin the comment line with a quote (“) mark. Successive lines should contain first a frequency in Hertz (beginning in the first column), followed by one or two spaces, then the magnitude, then optionally another space and then the angle in degrees. The angle is not used in pass/fail evaluations, but may be present so that normal LAUD2 ASCII saved files (which generate this data) can be used as limit files. For frequency response limit files, the magnitude is expressed in dB. For impedance files, magnitude is specified in ohms.

The frequencies can be any value in the audio range and need not be at any particular points or spacing. The frequencies must, however, start with the lowest frequency and be arranged in increasing frequency order. The frequency ranges given will be interpreted as connected and data at frequencies lower than the first given frequency and higher than the last frequency will not be evaluated. If there are “don’t care” ranges within other ranges you do wish to have evaluated, set the upper limit file values in that range to very high values and in lower limit files to very low values (such as very negative for frequency response, or zero for impedance). You should have one “dummy” line below the last desired data line (otherwise the last point might not be used in the evaluation). The “dummy” line can be nothing more than a carriage return or some text such as “End of data”.

As a simple example, it you wish to test a device’s impedance to verify that it never falls below 4 ohms, the Lower Limit file could be nothing more than the following:
“a comment line like this one; start it with a comma; then the data lines below it:
20.000 4
20000 4
End of data

Alternately, a device having an overall “worst case” impedance could be measured using LAUD, and the file saved as ASCII for use as a lower or upper Limit file. You may, in that case, still wish to edit extremes of the frequency range out of the limit file (using a text editor) to avoid evaluation at “don’t care” or in ranges of very noisy data.

Frequency responses are often tested to be within a certain tolerance (plus or minus so many dB). Limit files for such tests can be generated by measuring a reference device (if it is a measurement from a microphone, use “relative” or non-SPL mode), then adjusting the GAIN parameter ([F8]) up or down the desired amount for each limit file. Save each adjusted frequency response to disk in ASCII format. If more complex modification of the curve is desired, you can also use the MERGE facility to alter or redraw part of the curve before saving it out as an ASCII file (see the discussion on [* Transform moRe Merge] in the MLS/SINE menu description file).

If the “Adjust” option is used when the pass/fail evaluation of frequency response is made, LAUD2 will pass a tested curve if it can be adjusted by a single gain value to fit between the two Limit file curves. In such a case, the dB difference between the two limit curves is what is important, rather than the absolute gain. In other words, there would be no difference between [upper=+3dB, lower=-3dB] and [upper=+6dB, lower=0dB] when evaluating a frequency response curve with “setEval”=1 and “setAdj”=1.

Distortion Limit Files:

Distortion limit files are created using a text editor. The format is similar to that for Impedance and Frequency Response limit files: First there is one comment line. Then lines of limit data expressed as: frequency in Hertz, then a space or two, then the distortion magnitude limit value for that frequency. The frequencies must appear in ascending order. The distortion magnitude is expressed in percent of fundamental level. The values will be treated as if they were connected via straight lines on the log-frequency distortion plot, so that data at any frequency point falling between those specified will be evaluated using the interpolated value. There should be a final “dummy” line after the desired data lines.

The interpretation of the distortion value (as total harmonic, or as one of the individual harmonics) is determined by the type of distortion measurement. If the measurement is for Total Harmonic Distortion, that is what is evaluated per the limit files. Otherwise, the values of the harmonic which is specified as the “A” harmonic in the DIST_AN will be evaluated.

The following is an example of a simple distortion limit file which begins evaluation at 100Hz and has a limit of 0.2% between 100Hz and 300Hz. The limit curve then ramps down to 0.05% at 400Hz and then is a line at that value between 400Hz and 3000Hz. It then goes back up to 0.1% at 3100Hz and stays there all the way up to the 10000Hz limit.

Distortion Limit File
100 0.2
300 0.2
400 0.05
3000 0.05
3100 0.1
10000 0.1
end of data

There is usually no need for a lower distortion limit file, as distortion is typically to be minimized . LAUD2 is able to use a lower limit file, however, for use by makers of electronic “effect” boxes or euphonically modified electronics who may want to control the amount of a specific harmonic which is generated.

Thiele/Small Parameter Limit Files:

Thiele/Small parameters are evaluated using a single limit file which encompasses both the maximum and minimum values. The parameters Re, fs, Qe, Qm, Qt and optionally Vas may be evaluated. When the evaluation is performed, a series of horizontal “bars” is displayed on-screen. These bars represent the range of acceptable values (between the lower and upper limit) for each parameter. A red marker is shown on each bar to indicate the position of the measured value within (or outside of) each range. If “setEval” is set to “1”, LAUD2 will make a pass/fail decision based on the parameters and the limit values.

The format of the Thiele/Small parameter limit file differs from that for Frequency Response, Impedance and Distortion. The first line in the T/S limit file must start with the number 0, followed by one or more spaces, then an optional comment. Comments can be on all lines after the given value, as shown in the example below. The second line is the Upper limit for the DC resistance Re in ohms; the third is the Lower limit for Re. Fourth is the Upper limit for Qe. The succeeding lines are as described in the example. A dummy line should be placed at the end. If Vas is not to be evaluated, those values may be left out of the limit file.

```
0    You can type comments after the number on each line;first line is 0
10  This is the upper limit for the Re
8   Lower limit for Re
53   Upper Limit for fs
45   Lower Limit for fs
0.85 Upper Limit for Qe
0.55 Lower Limit for Qe
7    Upper Limit for Qm
2.5  Lower Limit for Qm
0.65  Upper Limit for Qt
0.53 Lower Limit for Qt
20   Upper Limit for Vas
15  Lower Limit for Vas
end of data (dummy line)
```

Using the MERGE facility

Merge is an option under the Transform menu in the MLS and SINE instruments. MERGE allows you to combine, sum, subtract, multiply (cascade), divide (normalize), paste together or redraw sections of complex frequency response data into a new composite response data set. This facility is usable only for editing frequency response curves, not impedance curves. If any Impedance data which you wish to preserve is currently in memory you should first save it to disk for the Merge operation will overwrite it in memory.

Merge is useful for patching together separate sections of response measured by different techniques such as near-file and 1-meter quasi-anechoic. It can also be used to combine the responses from separate measurements such as for elements in an array or driver and port combinations.

Before Merge can be used initially, a frequency response data set should be present in memory. After you activate Merge initially during a session, **the first thing you should do is select the Merge option "Freqs" and define the set of frequency points which will be used** for the end result as well as onto which all merge operations will be performed. Any time the set of points is redefined or re-entered using the "Freqs" option, the merge space will be cleared. Hence, you need to set Freqs to the desired points at the outset of each Merge session. In later re-entries into Merge in which you wish to work with the existing merge data curve, you should NOT use the "Freqs" option. Additionally, do not retrieve or measure any impedance curves if you have a curve in Merge with which you still wish to work. Loading of an impedance curve will clear the Merge data.

When the "Freqs" are defined, the current frequency response data will then be mapped to the selected frequency points, ready to be combined in whole or in part into the Merge data space. Subsequent combinations of other frequency response curves (if any) into Merge will always be made from the frequency response data in memory. These might come from a saved response file or from a new measurement. That data will also be mapped onto the preselected set of frequency points when Merge is re-entered (the frequency point definition should not be restated or changed if you are building a composite response curve in Merge, as that would Clear all merge data). If the response is to be postprocessed by operations which require Linear (FFT) format data such as Cepstrum, Hilbert, Xfer or IFFT, you can choose a Linear formatted set of frequency points; otherwise a Log-spaced set usually makes for easier merging.

The Merge Process:

The best way to visualize MERGE operation is to imagine it as occurring on a large bulletin board. The board is initially empty (or can be made empty via the "Clear" option). After determining which frequency points will be used, you take a response curve (or just a piece of it) and Paste it onto the appropriate part of the bulletin board which is designated for the matching frequency points. Then you get another frequency response curve (by a new data acquisition or a disk file retrieval) and take it, or a part of it, and put that onto the bulletin board also. It can either cover up parts of the previous response curve (Paste) or can be mathematically combined (Summed, Differenced, Multiplied or Divided) with that portion of preexisting response data with which it shares some frequency points. You can continue getting and combining data onto the board (Merge set) until you are satisfied with the result. If you want, you can take a pen and draw in an arbitrary phase and/or magnitude curve over parts of the composite curve which you are developing.

While working with Merge, you will often **see three sets of data curves** displayed (described here using the default system colors): In **thick white** will be the current merge data set. In **thin yellow** will be the current frequency response data which is in memory. In **thin dotted red** will be the preview of the resulting Merge curve you would get if you were to then select “Ok” from the Merge menu. Some of the curves may not be visible at times if they are undefined or off the screen.

Remember that you are editing complex data. In other words, you are working with data which has both a magnitude and a phase associated with it. You must take into account both relative magnitudes and phases when combining responses. For instance, if you were to Sum two responses of the same magnitude, the result could be anywhere between a 6dB increase (if the phases match) and a 70dB or greater decrease (if the phases differ by 180 degrees); you may get one result at one frequency and another at a different frequency if the phase delays do not match. If you combine data measured with different effective delays, such as might be caused by different distances from the measurement microphone or as might occur from removing time of flight from impulse response data, you will normally get severe response aberrations. It is usually most expedient to adjust the delay of the frequency responses being combined into merge using [F9] visually so that the slopes of the phase curves of the current frequency response data and the current merge data match. This could also be accomplished by careful numerical accounting for differences of sound travel time to adjust the relative delay (and avoidance of time-of-flight removal before any FFT transformations); but generally good results can be much more easily accomplished doing the correction “by eye” with each Merge combination operation.

Combining/Pasting:

At each step during combining or pasting, you may try different operations which can combine some of the frequency response (yellow) data into the merge data (white). You may select to combine or paste **All** of the frequency response data into the Merge data, only the region between (“**Btwn_mkrs**”) the two frequency markers (which you can set normally with the mouse), the region below the #1 marker (“**<=mkr1**”), or above the number #2 marker (“**>=mkr2**”). You get a preview of the result of your action in the dotted red curve before you finalize it and actually modify the Merge data set. If it all looks as you want it, you then approve (finalize) the change with the “**Ok**” selection. You are then asked if you want to copy the composite curve from the Merge set into the frequency response set. If you answer “No”, then you are returned to Merge for more operations, if desired. If you answer “Yes”, the frequency response curve is replaced with the curve from Merge and you are returned to the top [*] level menu from which you can then save or process the data.

During Merge paste or combining operations (and before choosing “Ok”), you can change the gain of the current frequency response curve to move it vertically on the plot by using [F8] as usual. You can also change the Delay of the frequency response ([F9]) to adjust the “roll” of the phase curve. These changes affect how the current frequency response data curve is to be included into the merge data set. You may also change the dB/division ([Shift-F8]) or frequency display limits ([Shift-F5] and [Shift-F6]) for all the curves as required.

If you wish to change the Gain or the Delay of the Merge data, however, you must first copy it to frequency response (you can use “None” to bring this about without changing anything), then go back into Merge and paste the entire curve back into the Merge space (overwriting itself there); but before choosing “Ok”, first change the Gain or Delay as desired.

Drawing:

Another option in Merge is “Draw”. In this portion of merge, you at first see only the thick white curve of the data currently in the Merge set (if there is not data in the merge set, you should probably first Paste in “All” of a frequency response curve so you can see where you are working). You then position your mouse at a frequency and data value (in either the magnitude or the phase display frame) at which you wish to draw onto the curve. Left click the mouse once and a red line will appear, which will be your preview curve. The red line will be thin where it merely copies the pre-existing data; it will be thick where the old data will be replaced with new data as directed from you mouse clicks. If you click too high or too low, just move the mouse cursor vertically at the same horizontal position, click again, and the line should jump to the corrected point.

All the data points which you click-draw between the time an instance of “draW” is started and “Ok” or “Cancel” is selected will be considered as connected. For example, if you were to click once toward the left side of the graph and then once toward the right side of the graph, all the points of the original data in-between will be replaced by values which fall on a straight line between your two drawn points. If you wish to draw separate unconnected sections of a curve, you must do it in two selections of “draW”. The only exception to this is if the two sections are in different display frames (one in the phase frame and one in the magnitude frame).

Should you not like the result of your drawing (and if you haven’t selected “Ok”, yet), you can exit by selecting “Ok”, which will leave the Merge set unchanged. Or less drastically, you can instead use your right mouse button to remove the more recent points you have drawn in most-recent-first order. Each right click will remove the last drawn point which has not yet been deleted. You can switch back and forth between left and right mouse buttons to draw and delete points as needed. When the results are satisfactory, choose “Ok” and you will be offered the option of copying the new Merge result to frequency response, as for the Paste/Combine operations described earlier.

TROUBLESHOOTING

The following is a brief list of problems and likely causes/solutions:

**** It just doesn't work!**
-- wrong type DSP card (only the PSA DSP-based types can be used with Liberty Audiosuite)
-- DSP card not installed correctly

**** Scrambled video or other general system misbehavior when DSP card is installed**
-- DSP card conflicting with some other card in the system. We have had several reported cases of some of the cards having trouble with some video cards. This seems to also depend on the motherboard being used. A swap of video cards may be in order. Alternately, contact the card manufacturer or dealer.
-- Be sure the computer power supply is not being overloaded. Try removing any unneeded cards to see if proper operation then results; if so, the difficulty may result from an inadequate power supply for all the hardware being used.

**** Trouble with multimedia functions of DSP card** (note: We at Liberty Instruments are not multimedia system experts. If you are using your DSP card also for multimedia "computer sound" and are having difficulties, you should consult the card or software manufacturer).
-- IN WINDOWS95: The Windows 3.1 drivers or install programs provided with most of the PSA DSP card types have shown compatibility problems with WIN95 (or rather, WIN95, *which came later*, has shown itself to have compatibility problems with many sound card drivers). The difficulties seem to vary from machine to machine, some working fine, others not working at all. For most of the cards, the manufacturer has revised installation instructions available for using the Windows 3.1 drivers or the generic drivers built into WIN95 and should be consulted.

For the ECHO DSP/mod card, Liberty Instruments can provide revised installation instructions which allow WIN95 to use its own built-in 32-bit drivers to operate the card. We advise that the installation program supplied with the card NOT be used with WIN95. Some of the files on the disk, however will be needed to be loaded in the revised installation.

**** I get clicking sounds out of the DSP card at startup and at times during operation.**
-- The DSP card will emit brief clicks when it changes sample rate or is initialized. It is a good idea to turn down the volume level of any external power amplifier connected to the DSP card during computer or program startup or shutdown.

**** Irregular hesitation and "popping" sound when doing SINE or DISTAN frequency sweeps.**
-- this is occurring when LAUD changes the sample rate during optimization of the sample time/rate and is normal. The effect can be minimized by operating the DSP card output level as high as possible before distorting (turn it up from the "levels" button in most of the instruments) and turning down the external power amplifier gain.
-- clicking sounds will also result when the sinewave generator in LAUD turns off between sweep steps or changes frequency; while the sinewaves generated always start at a zero-crossing, they may not end at one, and this will result in an audible click when the wave stops.

**** I get no visible frequency response results in MLS/SIN, although my Main input levels look fine.**
-- if you are in dual channel mode, be sure that the Cal probe is connected to the reference source signal.. If not, the measured response will be divided by nearly zero... which can make the result rather large and off the graph.

**** When I measure impedance, the phase appears to be about 180 degrees out from what makes sense.**
-- you probably have the Cal and Meas probes (or the mic/probe preamp outputs) switched.
-- your power amplifier may not be of a ground referenced type. See the discussion in the chapter System Requirements.

**** My MLS or pulse generated frequency responses suddenly look weak or irregular! This seems to happen just after I've been using one of the other instruments.**
-- check your "Window" parameter (controlled by [F7]). Frequency responses generated using MLS or impulse should use either a "dot-window" (Half-window) or no window at all ("None"). A full window will taper both ends of the time record to zero, and the important part of an impulse response is at the starting end!

**** My unCAl'd responses show a distinct rolloff at the low frequencies (<70 Hz).**
-- this is often a result of DSP card characteristics, notably the size of the coupling capacitors used. Some manufacturers evidently feel that very low bass will waste the already minuscule power of the on-card power amplifier, and they therefore roll off the low end. Fortunately, this will be corrected by the Cal process for most situations. Some cards, such as the modified Echo DSP card marketed by Liberty Instruments, feature an extended low end for improved performance in measurement applications. This type of card is recommended for critical low frequency measurements.

**** My impedance measurements are more noisy at very low frequencies.**
--see previous question.
--this may be due to the falling stimulus level at lower frequencies. This can be overcome by using the SINE instrument for measuring impedances at very low frequencies, or by averaging or filtering techniques as discussed in the chapter "The MLS/IMP/FFT and SINE INSTRUMENTS", under the heading "Noise, Averaging, and MLS Operation".

**** My distortion measurements are often below 0.03% at mid and higher frequencies, but as high as 0.1% at low frequencies.**
--see previous two questions. This happens due to falling test level making less effective use of the input dynamic range. It can be avoided by measuring the lower frequencies in a separate sweep, with the MAIN input level adjusted for the lower test levels of the low frequencies.

**** My impedance measurements show absolute nonsense!**
-- are both probes connected, and properly?
-- are both input gains (Main and Cal) set to the SAME level (they must be)
-- are both attenuator switches of the Mic/Probe Preamp at the same position?
-- Is the test resistor used for the measurement within 20%
to 500% of the impedance values to be measured? (preferably should be)

**** Program and computer will occasionally lock during an acquisition.**
-- this can occur if a glitch between the computer and DSP card breaks communications when LAUD is attempting a high-speed data acquisition. Because LAUD operates using

very tight processing loops, these loops provide no way to break out on a hardware error. If the problem occurs frequently, there may be a problem with the card or the computer.

****High frequency sine waves look messy on time domain ([F1]) displays, particularly at lower sample rates.**
-- this is due to Display Aliasing, which is described under that heading, as can be found in the on-line help index.

****I get "noise spikes" on impulse responses measured using MLS.**
-- This may be due to severe nonlinearities somewhere in the measurement chain. Possibilities include: output level from DSP card too high (try reducing using Levels button), external amplifiers equipment being overdriven, Mic/Probe preamp overdriven (use moderate to high DSP card input gains, and attenuate input levels if necessary using gain switches on external Mic/Probe preamp), microphone overload, or measurement of a non-linear device (blown speaker or amplifier).
-- In acoustical measurements, the spikes may be acoustical reflections in the test area.
-- Small pulses may result in single channel measurements if the unused Cal probe is left connected to a high level source. When not using an input to the DSP card or Mic/Probe preamp, it is good practice to disconnect it.

****Phase measurements and distance measurements give strange results, such as negative delay.**
-- In single channel modes, the phase will not be normalized by a Cal operation. With the MLS instrument, this will still give comprehensible results but the zero-delay phase plane will be shifted due to DSP card digital filter delays in both the stimulus output and the measure input. In the SINE instrument single channel measurements will not typically provide usable phase data, particularly if an "optimized" rate mode is used -- use dual channel mode with a Cal normalization if phase data is required.

-- In dual channel modes, the zero phase reference plane will be correct if you set the number 1 (left) time marker of channel 1 to point #1. The digital delays will then be the same for both channels and will be "Cal'd"-out. If you move marker 1 to the first edge of the impulse response or sine burst arrival, the resulting phase plot will, however, show a negative delay (phase rising with frequency). This negative delay results from normalization with the positive delay present in the Cal channel. The resulting delay value will be approximately minus (-) 61,000/sample_rate [milliseconds], and can if desired be removed by entering that value as the delay parameter using [F9].

**** I'm operating dual channel MLS and I'd like to save the time domain data sets from both channels. But [File Save Time_resp] saves only channel 1.**
-- Generally there is no need to save both of the raw time domain files. If you save a Cal'd frequency response file, you can use IFFT to transform it into a single cal'd (normalized) time domain file if required. However....
-- If you do, for whatever reason, need to save or restore a channel 2 time domain file from the MLS instrument, you can get there by using the [Transform More Swap_channels] option to exchange the two channel data sets. Then save the Time domain data, which will be what used to be in Channel 2.

**** I can't seem to get any low frequency data when I'm measuring frequency response (with the MLS or SINE instruments).**
-- If you are making a quasi-anechoic acoustic measurement (gated type SINE, or MLS with time domain truncation), you will be unable to resolve below the frequency 1/(measurement time period). You need at the very least acquired data of a full cycle's time (and more practically, two or three) in order to detect any given frequency component. To get lower frequency data, you may be able to compromise your echo elimination somewhat (move marker number 2 to the right and tolerate some echo

contamination). Otherwise, try reconfiguring your microphone and UUT to maximize direct to reflected (echo) arrival time and to minimize reflection intensity (sometimes use of foam sheets, blankets or pillows around the measurement area can be used to advantage). A closer microphone position will always help, also, by increasing the intensity of the direct component relative to that of the reflected ones. But close-micing may cause irregularities at higher frequencies due to different path lengths from various portions of the radiator to the microphone.

-- In the MLS instrument, if you are making a measurement which is not subject to echoes, put the time domain markers at MKR1=1 and MKR2=SIZE and **set auto_tmkrs to Off.**
-- In the SINE instrument, do not use the gated mode except for quasi-anechoic acoustic measurements, and in that case be sure to set the time domain markers properly (preferably by setting them within the MLS instrument via the impulse response). The optimized modes are recommended for measurements other than quasi-anechoic acoustic measurements or for SINE impedance measurements.

****When I am using the SCOPE, the mouse doesn't work and I can't click on the buttons.**
-The mouse is not usable with the SCOPE instrument. This is because the display routines which are constantly busy updating the scope display clash with the mouse cursor drawing procedures. While the routines could be rewritten to accommodate simultaneous mouse use, the reduction in speed was deemed too high a price for the small benefit of a mouse- operated oscilloscope. The buttons are operated by using the function keys for which the numbers are displayed within the buttons.

****I can't get LAUD to run from WINDOWS, although it runs fine in DOS.**
-This is often caused by insufficient base (first 640K) memory. This is the memory referred to in the Windows 3.1 PIF Editor's "Memory Required" checkbox. You probably have too many "stay-resident" programs or drivers loaded into memory by your "autoexec.bat" or "config.sys" files. Try taking a look at your autoexec file and removing things which may no longer be in use.
-You can also try lowering the "KB Required" and "KB Desired" settings in the LAUD.PIF file (in your LAUD directory), as the ones given in the default LAUD.PIF file have a little spare.

****LAUD runs from WINDOWS, but the mouse won't work in DOS.**
-You must have a DOS mouse driver loaded to run LAUD in DOS. A driver disk should have been provided with your mouse. This driver is normally loaded in your "autoexec.bat" or "confiq.sys" files.

****I get Windows error messages about Multimedia problems after running LAUD.**
(or) **My DSP card won't work as a soundcard after running LAUD.
- LAUD reprograms the DSP card with measurement code, and removes the soundcard emulation code in the process. You may need to reboot your machine to restore the emulation code; alternately, you may be able to use DOS commands or a batch file to load and reinitialize the souncard code after using LAUD --- see your DSP card manual.

Basic Operational Tutorial

This section provides a step-by-step example of simple measurement procedures using Liberty Audiosuite. Most will be based on the MLS instrument, as MLS and SINE are the more complex devices within LAUD and are well suited to showing how the system is configured and used. Most of the settings shown here are not, however, applicable to the SCOPE instrument; the chapter on the SCOPE should be consulted directly.

These examples are basic and are certainly not intended to be comprehensive, as a wide variety of measurements can be configured using LAUD. But by working through these examples and by referring to the individual instrument documentation in this manual, new users can become familiar with the controls and operation of the instruments and can develop the concepts needed to configure special purpose measurements.

If your immediate need is for a basic measurement rather than instruction, please refer to the chapter on Scripts and to the built-in Easy Scripts contained in the LAUD software which provide guided and automated procedures for many of the most commonly needed measurements.

Menus and Syntax:

The examples which follow will be presented using some abbreviated directions. Keystrokes will be shown within square brackets. For example, to indicate that the "plus" key should be pressed, we would write: **[+].** To indicate the Enter key should be pressed, we would write: **[Enter].**

The **menu text line** is printed just below the row of buttons which is across the top of the screen. When controlling the LAUD system via the menus, you normally start at the top-level menu, choosing an option from each successive menu until the desired operation or control is effected. For example to change the decibels per division vertical scale parameter for frequency response measurements in the MLS/IMP/FFT instrument (also called the "MLS" instrument), the following sequence might be followed:
{if you are not in the MLS instrument, hold down the [Alt] key and then press the [M] key to get there}
1) Go to the **top-level menu** (if not there, already). You can do this by pressing the asterisk [*] key, by clicking the mouse on the "Liberty Audiosuite" button at the top right of the screen, or by repeatedly pressing the [Esc] key, which backs upward through any chosen menu options. In these tutorials, the top-level menu will be **designated using the asterisk [*].**
2) Select the **"Display"** option. You do this by pressing the [D] key--In general, you can select a menu option by pressing the key corresponding to the highlighted capital letter (or number) in the desired option word of the currently presented menu. You can also select the word by clicking on the desired option word with your left mouse button. Notice that after you make this selection a new menu is presented.
3) Select the **"Format"** option from the new menu (press the [F] key or click on the word "Format"). Again you proceed to another menu below the previous one.
4) Select **"Scale"** ([S] key).
5) Select **"dB/div"** ([B] key).
6) Now, rather than being offered another menu, you are prompted for your desired value. As this parameter is a whole number value, you can either type the value, or click on the up/down arrows shown to the right of the prompt line. Type in, for example, **[5]** and press **[Enter].** You go back, in this case, to the menu from which you selected "dB/div". You could then select other options from this menu and change other scale parameters. Most of the time in this tutorial, however, instruction for each operation will

be given starting from the top level menu, so that it will not be dependent on your current position within the menu structure.

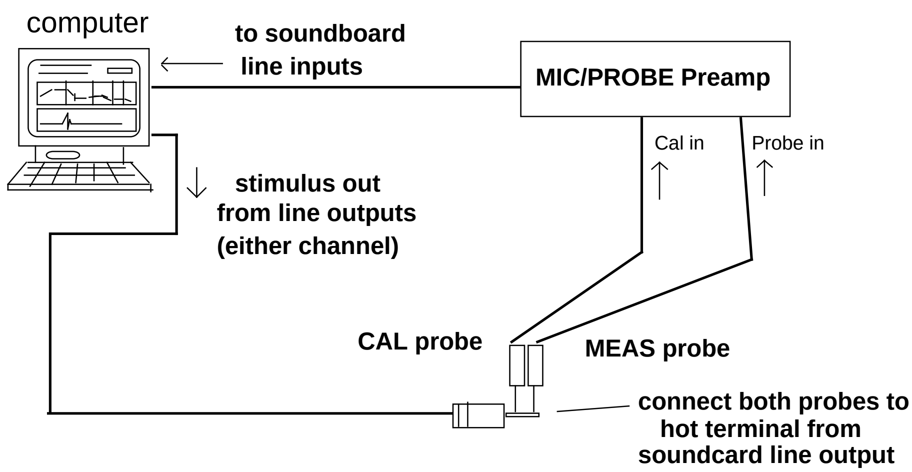
The above rather lengthy discussion provided instruction for making, in essence, only seven keystrokes (or mouse clicks). LAUD controls and commands can normally be made very quickly and efficiently provided you know where you want to go. To indicate the above key sequence in abbreviated form, we instead write:
[* Display Format Scale dB/div], type “5”, [Enter].
The keystroke menu options are given in order of selection within a common set of square brackets, normally starting at the top level [*]. The entire menu keywords are given for clarity, but remember that only the capitalized letter (or numerical digit) of the keyword is to be typed. Responses to prompts are given separately from the menu option, and the [Enter] keystroke is given separately. Note that “[Enter]” indicates the Enter key itself, not the “E” key of a menu option.

Many of the more commonly needed operations can be also performed using special keystrokes. Many, but not all, of these involve the Function keys (the F_ keys which are usually along the tops of most keyboards). For example, the “dB/division” scale factor of the above example can also be set using the [Shift-F8] key combination. This combination is entered by holding the [Shift] key and then pressing the [F8] key. To indicate setting this parameter using the much more convenient function key instead, we would write:
[Shift-F8], type “5”, [Enter]
These special keys are described in the chapters on each instrument and in the reference section of this manual.

Some of the Function keys change parameters “round-robin” style: when the key is pressed, the value increases unless the maximum value is reached. In that case, the value reverts to the minimum value. For example, the [F2] changes the SIZE parameter (displayed in the top-leftmost button on the MLS screen) which is the number of data points to be displayed in time domain plots and which will be transformed in any FFT operations. If you press [F2], the size value will increase unless the value is currently 16384, in which case it will change to 256 again. To indicate that this key should be pressed repeatedly until a target value of 1024 is set, we would use the syntax:
[F2] until SIZE=“1024”
Similar syntax will be used for other keys, such as [F3], [F4] and [F7], which operate in this manner.

<u>Tutorial</u>

First, you will acquire the MLS stimulus as it comes from the Line Out jack of the DSP card. The MLS signal, which looks and sounds like white noise, will be transformed into an equivalent impulse response, which can itself be transformed into a frequency response curve.



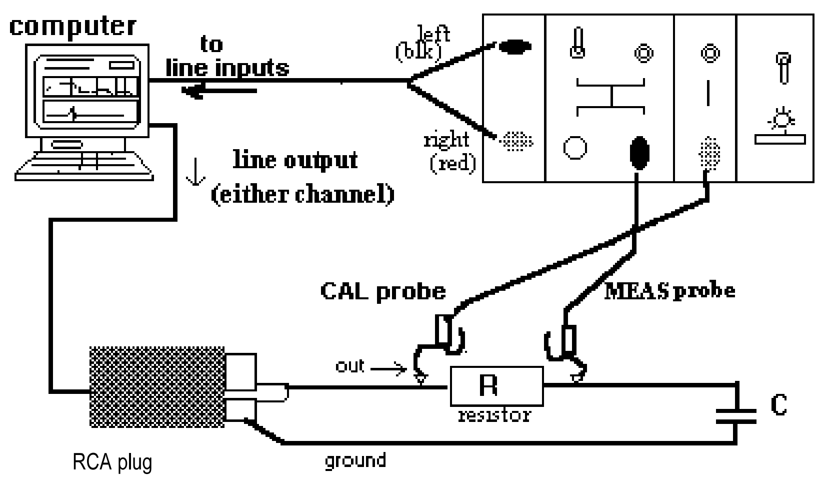
First, set up the equipment as shown. This example will not require use of a power amplifier. The probe ground leads (green, black or blue clips) can be left disconnected or connect to the cable shield for this test. Set the Mic/Probe Preamp switches as follows:

Ch1 Probe/Mic switch:	up (Probe position)
Ch1 Gain switch:	center (0dB)
Ch2 Gain switch:	center (0dB)
Power Switch:	down (On)

1) Start up LAUD and go to the MLS instrument:
[Alt-M]
2) Configure the system:
[F2] until SIZE="1024"
[F3] until RATE="48.0k"
[F4] until INPUT="PROBE"
[F7] until WINDOW=".BLACKM"
(selects the half-blackman data window)
[F8], type "1", [Enter]
[* Acquir Mls Four-k]
(selects a 4k MLS sequence)
[%] until "Dual channel" mode
(this is indicated on the button near the bottom left of the screen; the [%] key is usually [Shift-5])
[* Acquir Auto_tmkrs Off]
(Auto_tmkrs is used to automatically remove echoes from a measurement. Not wanted in this measurement, as there are no echoes)
[* Acquir Set_levels Out], use [Up-arrow] or [Down-arrow] key until out level="-9.00dB"
(these "arrow keys" are the up/down cursor keys of your keyboard)
These system configurations can be saved and/or reloaded in normal use by using options in the [* System Config_file] menu. Doing this can save all the effort of setting each of these parameters each time you need to do a measurement.
3) Adjust the input levels. LAUD has provisions for doing this automatically, but for this example it will be done manually so you can become familiar with the functions:
[* Acquir Set_levels Main_in], use arrow keys until main_in level="7.5dB", [Enter]
[* Acquir Set_levels Cal_in], use arrow keys until cal_in level="19.5dB", [Enter]
The Ch1 ("Main") level is set intentionally low for this situation, so you can see the effects. Now take a "Look" acquisition to check the acquisition level:

[* Acquir Set_levels Look]
The screen will show a repeated acquisition of two traces, the top from the Ch1 "MEAS" probe and the bottom from the Ch2 "Cal" probe. The top trace will be notably smaller in amplitude than the lower trace. The waveform displayed is that of the MLS test signal, and should look like broadband noise. You normally want the amplitude of this trace to be, if possible, between about one fourth and three fourths of the frame height. Most importantly, it must not be above the frame height (the computer will "beep" if it is).
[Esc] *(to break the "Look" cycle)*
Now adjust the levels, checking the result using the "Look" option:
Use [* Acquir Set_levels Main-in] and [.... Cal_in] to adjust the input levels so that the waveform fills about half the screen frames when viewed using [* Acquire Set_levels Look]
(this will probably be when the values of each are at about 18dB)
4) Make the acquisition. When using "Look", you saw the waveform as it actually occurred in the time domain, so that peak levels could be checked. Data acquisition for measurement uses the "Collect" option, which gets the raw data and transforms it into equivalent impulse response data.
[* Acquir Collect]
You should now see two identical impulse responses in the two frames. They are identical because both probes are sensing the same signal.
5) Transform the data to frequency response. First, lets see how a transformation of one channel alone looks. For this, change to Single Channel mode by pressing [%] once:
[%] until "Single channel" mode
Now transform the channel 1 data to frequency response using the Fast Fourier Transform:
[* Transfrm Fft]
Set the vertical scaling for 5dB per division, the vertical dB gain to 0,and set the frequency display range:
[Shift-F8], type "5", [Enter]
[* Display Format Scale Gain_db_fr], type "0", [Enter]
[Shift-F5], type "100", [Enter]
[Shift-F6], type "20000", [Enter]
You should see a moderately flat curve near the top of the magnitude plot. If you had the levels in "Look" set for about half screen, the curve will be above the 0dB line. Enable display of the phase curve (if not already enabled):
[* Display Format shoPhase? Yes]
The zigzag phase curve shown at top is due to filter delays in the DSP card. The downward slopes are indicative of the amount of delay; the upward vertical rises are where the phase periodicity rolls to equivalent values (remember that -190 degrees is the same as +170 degrees, etc.). You can compensate the phase display for this or other delay:
[F9], type "1.17", [Enter]
You can use values other "1.17" (which is the delay removed, in milliseconds), to see the effects on the phase curve. For the ECHO DSP card at this sample rate, 1.17 milliseconds is the approximate value which should yield a relatively flat phase curve. After you are finished experimenting with this parameter, set it back to zero delay compensation:
[F9], type "0", [Enter], [Esc]
6). The slight magnitude variations and phase delay of the single channel measurement/ transformation would normally contribute error to a measurement. A LAUD dual channel measurement, therefore, consists of two acquisitions, one of the stimulus signal itself and one of the signal as it is changed by the device being measured. The final measured result shows only the change, so that measurement equipment gains and responses can be eliminated. In our example, there is no change between the signal measured by the probes (they measure the same point), so a normalized or "cal'd" result should show a line at 0dB, and 0 degrees phase throughout. Switch back to Dual Channel Mode:

[%] until “Dual channel” mode
And now do another FFT. This time, the time domain data in both channels will be transformed to frequency response, and the Ch1(Meas) result will be automatically divided (normalized or “cal'd”) by the Ch2(Cal) result before display.
[* Transfrm Fft]
You should now see normalized data very close to 0dB gain and 0 degrees phase.
7). So far, we aren't really measuring anything other than the LAUD system itself. Next, we'll make a simple low-pass filter and use LAUD to measure its characteristics. In many cases, you would normally have a power amplifier between the DSP card output signal and a network being tested. But if the network is high enough in impedance to not overload the DSP card output, it can be connected directly. Get a resistor (any wattage) of between 1000 and 47,000 Ohms and a capacitor in the range 1000pF to 0.022uF and configure them as shown. The connections can be made using several sets of clipleads. Electrically, the Cal (Ch2) probe remains at the same point; the Meas probe moves to the junction between the resistor and the capacitor so that it senses the rolled-off response caused by the resistor-capacitor filter.
8). Now repeat the measurement, as above. First “Collect” the data:
[* Acquir Collect]
Then Transform (and normalize, as you are in Dual Channel mode):
[* Transform Fft]
You should now see a phase response approaching -90 degrees at the high frequencies, accompanied by a rolloff in the response magnitude. The low frequency level will also be reduced somewhat, due to the loading of the MEAS probe (the probe impedances are 50k ohms, which is relatively high, but will still have an effect on higher impedance networks. For low impedances like dynamic loudspeakers, the loading is insignificant).



You can adjust the vertical position (dB display gain) of the curve by use of the F8 key when a frequency response curve is being displayed. Try this:
[F8], type “4”, [Enter]
The magnitude curve is raised on the graph, but the shape is unchanged. You can also have LAUD try to center the response curve around the zero dB line via display gain:
[=]
Before going on, restore the display gain to 0dB (to show actual gain):
[F8], type “0”, [Enter]

9) In the MLS instrument (and the SINE instrument, when gating is used), the time domain markers can be used to select only portions of the time domain data for transformation to frequency response. (LAUD keeps separate markers for time domain, frequency response and impedance). Such editing or truncating of the time domain data

can be used to remove or isolate echoes in an acoustical measurement. Press [F1] to redisplay the time domain data:

[F1]

Currently in the Channel 1 display, Marker1 should be at the far left and Marker2 should be at the far right. The region between includes all the time domain data within the 1024 point SIZE; in particular, it includes the rather wide and low impulse response of your RC lowpass filter at the left side of the plot. We'll see how selecting parts of this time domain data affects the response.

Select and move marker1:

[F5], type "49", [Enter]

The first rising edge of the response will be slightly after time domain data point #49. The impulse response is still between the two markers.

[* Transfrm Fft]

Note that the magnitude curve doesn't change at all (although the phase does, as delay is in effect removed by moving the starting point of the FFT to the right). Now move marker1 to the right of the main portion of the impulse response:

[F1], [F5], type "170", [Enter]

[* Transfrm Fft]

Now, the magnitude curve is below and off the screen, while the phase curve shows very much "negative delay". The main portion of the impulse is no longer contained between the time domain markers, and is then not included in the FFT.

Now, move marker1 back to point #1, and put Marker #2 at point # 170:

[F1], [F5], type "1", [Enter],[F6], type "170", [Enter]

[* Transfrm Fft]

Now, the impulse response is again between the two markers, but much of the rightmost time domain data is not being included. No results at the lower frequencies can be obtained from this short section of time domain data. But at higher frequencies, the result is essentially unchanged from using the full time span. If the impulse response had echo responses after the direct response, they could be removed by this technique, at sacrifice of low frequency measurement capability.

10) There is also a set of "auto_Measure" options, which combine data collection and transformation into one step, and can also provide other options such as "cycling". Before using auto_Measure, it is a good idea to set all the parameters in the [* auto_Measure Setup] menu:

[* auto_Measure Setup Windows .bLackman]

[* auto_Measure Setup cYcle oFf]

(cycling is used to have a measurement automatically repeat continuously, so the results of adjustments can be made "live")

[* auto_Measure Aut_in_levs oN]

Since we are going to use Automatic input levels ("Aut_in_levs"), we should first set the proper mode. This can be done through the [* Acquir Set_levels] menu or via the [-] shortcut key:

[* - Auto_adj Both_match]

"Both_match" is used so that relative gain will not change. Setting this will cause an automatic level adjustment to be made immediately.

Now, to make the "auto_Measure" measurement:

[* auto_Measure Freq_resp]

The entire cycle of level adjustment, data acquisition and transformation will occur, and the result (the same one as before) will be displayed.

10). Now, we'll switch to the SINE instrument to see how the same measurement would be accomplished there:

[Alt-S]

The screen remains essentially unchanged, except that the "SINE" button at the bottom of the screen is now highlighted, rather than the "MLS" button. In the SINE instrument

(and the DIST_AN instrument and the SINE/MLS MERGE facilities) you must first tell LAUD what frequencies points you wish to use for the measurement/result:
[* Acquir Freqs Log_f First_freq], type "100", [Enter]
Note that the "Acquir" keyword brings up a different submenu than it did under the MLS instrument!
[* Acquir Freqs Log_f Last_freq], type "20000", [Enter]
[* Acquir Freqs Log_f Num_points], type "70", [Enter], [Ok]
This selects a measurement over 70 frequency points evenly log spaced, in the frequency range 100Hz to 20,000Hz.

Now, we should select a SINE measurement mode (there are several available). We'll choose an "optimized" mode, which is usually a good choice if the measurement does not require echo removal:
[* Acquir Modes Resp_mode Optimized], type "10", [Enter]
[* Acquir Modes Gating oFf]
Set levels:
[* Acquir Set_levels Auto_adj Both_match]
Now make the measurement:
[* Acquir sWeep Freq_resp]
The phase will not be displayed until the measurement is complete (which will take considerably longer than the MLS measurement). The result should be the same as for the MLS measurement.

11). The current setup is also usable for measuring impedances (of moderately high impedance networks). In the given setup, the capacitor can be measured. First, you must tell LAUD what value of resistor you are using:
[* Display Format Scale Ref_resistor]; then type the value, in ohms, of the resistor you used; [Enter]
Now, use sWeep again to measure the impedance:
[* Acquir sWeep Impedance]
You may at first see nothing but the dotted phase curve, depending on the scaling. Use [=] to automatically scale the plot:
[=]
The capacitor impedance should show a drop with frequency, and a phase of near -90 degrees throughout. If you click your mouse on the magnitude curve, a marker should jump to near that place, and the capacitor value can be read in the series ("+") and parallel ("||") RC equivalent values displayed above the plot. In impedance measurements, LAUD is able to compensate for loading by the probes. As the impedance being measured is only a (presumable high-Q) capacitor, the resistance values shown will not usually be meaningful (may be "oooooooM", which means out-of-range, or even slightly negative due to measurement tolerance).

This simple tutorial has shown how to use menus, function keys, markers and the basic processes of data acquisition, transformation and display presentation and scaling in the MLS and SINE instruments. The operation of the SPECAN and DISTAN instruments uses many of these same features and procedures. For more in-depth discussion of the features and operation of the LAUD system, please see the chapters on the individual instruments.

Reference Section

These reference pages include a Function and Special Key guide and listings of the menu structures for each instrument. These are provided for quick reference in locating submenus and in determining the active letter or number to be capitalized for each option when writing Scripts. Descriptions and explanations of the functions of the various keystrokes and menu words are given in the instrument chapters in the manual and in the on-line HELP facility.

<u>ASCII File Formats:</u>
<u>NOTE:</u>
The .FRD and the .ZMA files written by the MLS and SINE instruments can be written with a "header line" at the top of the file. When you are saving data in these formats, you will be prompted to edit this line; edit as you would a normal LAUD "title". If you delete the header entirely (press [Home], then [Enter]), no header line will be included. Special headers are needed for some simulation programs which may be using the data; other programs need the header to not exist. Separate headers are maintained in LAUD for .FRD and for .ZMA file types. After changing these headers, you can make them the default headers by saving your Standard Configuration File (STANDARD.ICF, using [* System Config_file Save, "STANDARD"]).

.FRD Files: each line:
Frequency(Hz) Magnitude(dB) Angle(degrees)

.ZMA Files: each line:
Frequency(Hz) Magnitude(Ohms) Angle(degrees)

.IMP Files: first line: SIZE parameter
second line: Number of actual points (which determine resolution)
third line: Sample rate
fourth line: is data cal'd? (0=no)
lines 5 to (4+SIZE): the time domain data values
other lines: internal system parameters

.DST Files: first line: number of data points per data block
second line: the "A" harmonic (ignore if THD)
third line: "B" harmonic (ignore if THD)
forth line: 1 if measured from probe (in volts) or 0 if Mic (Pascals)
fifth line: the word "Overtones" if individual harmonics, else "THD"
For THD (one block): next comes a header line, then the data lines in format:
fundamental_frequency fundamental_level %THD
For individual harmonics (three blocks): a header line, then fundamental data:
fundamental_frequency fundamental_level
then another header, then "A" harmonic data:
fundamental_frequency %A_harmonic A_harmonic_phase
then another header, then "B" harmonic data:
fundamental_frequency %B_harmonic B_harmonic_phase

.RTA Files: first line: resolution (3 for 1/3rd octave, 6 for 1/6th octave)
second line: units (dBmV or dBSPL)
third line: value of top-of-scale line
remaining: data points of measured bands in format:
center_frequency dBvalue

1

Function and Special Keys
(general uses: not active in all instruments)
(see separate listing for OSCOPE/GEN)

Function Key	Normal	Shift	Alt
F1	Redraw Time domain response (MLS/SINE)	Enter HELP facility	
F2	Size (of display or FFT set)	Copy screen to printer	execute macro #2
F3	Sample Rate increase	Sample Rate decrease	execute macro #3
F4	Input source (Mic or Probe) info; set to match Mic/Probe Preamp	SPL sensitivity mode settings (MLS/SINE)	execute macro #4
F5	Marker #1 set/read (MLS/SINE)	Set lower frequency limit of display	execute macro #5
F6	Marker #2 set/read (MLS/SINE)	Set upper frequency limit of display	
F7	Choose data window for Time domain data before transformation	Show Frequency Response data (if valid; MLS/SINE)	
F8	Gain adjust (scalar if time domain, dB display gain if frequency response)	Vertical Scaling (dB/division if frequency response, Ohms/division if impedance)	Auto-scale adjust (dB/division and Gain for frequency response; Ohms/div for impedance)
F9	Set delay (modifies phase responses; MLS/SINE)	Inverts phase (time domain or frequency response)	
F10	"Display Show" options of MLS/SINE menus	Automatic input level adjustment (per settings in the Set_levels menus)	

Key	Purpose
*****	Move to top level of current menu structure
Esc	Back up one menu step, or cancel entry
=	Perform automatic display scaling
+	To adjust readings for Mic/Probe preamp attenuator (Gain) settings
-	Provides menu for manually adjusting input levels
`	Start Easy Scripts
%	Toggle Single/Dual channel mode
!	Re-initialize the DSP code
left arrow	move marker to left one point (with Ctrl: 10 points)
right arrow	move marker to right one point (with Ctrl: 10 points)
Space bar	enables or disables display of plot title, if defined
Tab	Edit plot title
Alt-C	reset screen colors
Alt-D	switch to DISTORTION ANALYZER Instrument
Alt-M	switch to MLS instrument
Alt-O	switch to OSCOPE/GEN
Alt-P	switch to SPECTRUM ANALYZER instrument
Alt-S	switch to SINE instrument

Reference Literature on Loudspeaker Testing

The following are some recommended reading relevant to LAUD, mostly related to loudspeaker testing. The list isn't even vaguely comprehensive, but should serve as a good start in those wishing to pursue the topic further.

Both volumes of Loudspeakers, An Anthology, published by the AES (Publications Office, Audio Engineering Society, Inc., 60 East 42nd Street, New York, New York, 10165; also available through Old Colony Sound Lab, FAX 603-924-9467) should be considered essential to any library of loudspeaker literature. Included are the classic papers by Olson, Villchur, Klipsch, Thiele, Small and others too numerous to list.

Berman, J.M. and L.R. Fincham, "The Application of Digital Techniques to the Measurement of Loudspeakers", J. Audio Eng. Soc., vol 25, 6/77, p. 370. Also in Loudspeakers, An Anthology, Vol.1, published by the AES.
This is the groudbreaking article which set the stage for modern FFT based measurement systems. It remains an excellent reference for these kinds of measurments.

Rife, Douglas D. and John Vanderkooy, "Transfer-Function Measurement with Maximum-Length Sequences", J. Audio Eng. Soc., vol. 37, 6/89, p.419. Also in Loudspeakers, An Anthology, Vol.2, published by the AES.

Frederic J. Harris, "On the Use of Windows for Harmonic Analysis with the Discrete Fourier Transform", Proceedings of the IEEE, vol. 66, No. 1, January 1978, pp 51-83.

Struck, Christopher J. and Temme, Steve F., "Simulated Free Field Measurements", J. Audio Eng. Soc., Vol 42, No. 6, 6/94, pp467-482. ---Highly recommended.

Keith R. Holland, "The Use of Cepstral Analysis in the Interpretation of Loudspeaker Frequency Response Measurements", Proc I.O.A Vol 15, Part 7 (1993), pp65-72

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MLS INSTRUMENT MENU STRUCTURE

- Display
  - Format: Delay shoPhase?
    - freq_Range: Lower Upper
    - Scale: dB/div Ohms/div Ref_resistor Gain_db_fr
      - Spl: oFf
        - oN: Nominal Main_ext_atten Cal_ext_atten mic_Preamp Ok
    - sMoothing: None Octave Half Third Sixth tWelfth
  - Show: Impuls_resp Freq_resp Time+freq Z_plot Cal
    - >More: dRiver_data Distance Mic_data Step
  - Prntplot: togglePromo Go
    - Configure:Ibm/epson Hp_jet to_File
  - Title: View Edit Hide
  - Mic_correc: Time_data Freq_data Showmicdata Micsens
  - Ovrlay
    - Type: Time Freq_resp Impedance
    - Curves: Add Remove reDraw goBack
    - Symbols: Sparse Dense oFf
    - Legends: oN oFf
- File: Micdat
  - Retrieve: Time_file Freq_file Zdata fCal_file
  - Save: fCal
    - Timeresp: Ascii Compact
    - Freqresp: Ascii ascii_mic_.Dat Full
    - Zdata: Ascii Full
  - Delete
    - Ascii: .Imp .Frd .Zfr .Dat .dSt .Rta
    - Compact: iM Fr Zf Cal Icf
  - Change_dir: Base Select
- Transform: Cal Fft Ifft
  - Set_cal: from_Freq_resp from_chan2_time_resp
  - Waterfal: Length Steps Usecal? Go
  - Etc: Cenerfreq dBperdiv Gain Ok
  - cePstr: Gain timeSpan hpFilt Use_negf Analyze
  - moRe: Swap_channls Hilbert
    - conVert: chk_freQ chk_impeD
      - Freq_resp: Freq_list Go
        - Log: First_freq Last_freq Num_points Ok
        - liN: System_rate sPecified_rate Ok
      - Impedance: Freq_list Go
        - Log: First_freq Last_freq Num_points Ok
        - liN: System_rate sPecified_rate Ok
    - Merge: Clear
      - Paste: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
      - Sum: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
      - Diff: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
      - Mult: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
      - diV: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
      - draW: Ok Cancel
      - Freqs: Freq_list Ok
        - Log: First_freq Last_freq Num_points Ok
        - liN: System_rate sPecified_rate Ok
- Acquire: Collect Look

```
        Set_levels: Main_in(one) Cal_in(two)   Out   /Look
                /Auto_adj:  Main_only  both_Independently  Both_matched
        Mls:   Sixteen-k   Four-k   Off(impulse)
        Auto_tmkrs:  Off  Echo_edge  First_edge  Both_edges  Now
Systm:  timeGap
        Cfg_file:  Save  Use  Init
                Hardware_indep:  Save  Use
                in_sCript_dir:   Save   Use   Dir
        Port:  Printer  read_Card_version
                coDec:  A  B  C  D
                Type_card:  Orchid_sw  orchid_Gw+  Cardinal  Wearnes  Echo  Paradise  Adaptec
        Avg:  Mic_inpt  Probe
        coLor: Color B&w
                Palette:  Grid  Background  Reset
                        Lettering: l1  l2  l3
                        Curve: c1  c2  c3  c4  c5
                        Fill:  f1  f2
        Z_set:  Probe_z  Set_bal   Ref_resistor
                        Use_bal?:  Yes No
        Inpts
                Channels: Single Dual
                Mic_input: Line(external_preamp)  Mic(internal_preamp)
auto_Meas:  Imped  Freq_resp  imPulse  justCal
        Setup:  Cal_source
                Windows: None Blackman Hamming bIngham .bLackman .hAmming .binGham
                cYcle: oN  oFf
                Aut_in_levs:  oN  oFf
        User:  List
                Record: Boot alt-f2  alt-f3  alt-f4  alt-f5
        T/s
                vas_Method
                        closed_Box: cubicFt  Litres
                        added_Mass:  Grams  Ounces
                Extract: Normal  Loaded  newDriver
                full_Auto:  Normal  Loaded  newDriver
                Setup: Ref_resistor  Ohms/div
                        Dc_resistance:  Define Extract
                        Effect_dia:  Inches  Cm
Cntrl:  sHow  beeP
        sCripts:  Easy
                Custom: Directory  runScript
        Lim_fil: Upper(max)  Lower(min)  T/s_params  scriptDir
        Ext:  Signal_out
                select_Port: None lpt1 lpt2 lpt3
                wait_for_Hi: wait_Now  before_Acquire
                wait_for_Lo: wait_Now before_Acquire
        setup1: Pause  setEval  setAdj
                pRint: Line Formfeed
                tst_Mode:  Freq_resp  Impedance  T&s
        setup2: Main_in   Cal_in   Out
                Siz: A B C D E F G  {256, 512, 1024, 2048, 4096, 8192, 16384}
                Rate: A B C D E F G H I J K L M N {5.125kHz...48kHz}
                Inpt:  Mic probe1
```

Windw: None Blackman Hamming bIngham .bLackman .hAmming .binGham
cHann: single1 dual2

SINE INSTRUMENT MENU STRUCTURE

- Display
  - Format: Delay shoPhase?
    - freq_Range: Lower Upper
    - Scale: dB/div Ohms/div Ref_resistor Gain_db_fr
      - Spl: oFf
        - oN: Nominal Main_ext_atten Cal_ext_atten mic_Preamp Ok
    - sMoothing: None Octave Half Third Sixth tWelfth
  - Show: Impuls_resp Freq_resp Time+freq Z_plot Cal
    - >More: dRiver_data Distance Mic_data Step
  - Prntplot: togglePromo Go
    - Configure:Ibm/epson Hp_jet to_File
  - Title: View Edit Hide
  - Mic_correc: Time_data Freq_data Showmicdata Micsens
  - Ovrlay
    - Type: Time Freq_resp Impedance
    - Curves: Add Remove reDraw goBack
    - Symbols: Sparse Dense oFf
    - Legends: oN oFf
- File: Micdat
  - Retrieve: Time_file Freq_file Zdata fCal_file
  - Save: fCal
    - Timeresp: Ascii Compact
    - Freqresp: Ascii ascii_mic_.Dat Full
    - Zdata: Ascii Full
  - Delete
    - Ascii: .Imp .Frd .Zfr .Dat .dSt .Rta
    - Compact: iM Fr Zf Cal Icf
  - Change_dir: Base Select
- Transform: Cal
  - Set_cal: from_Freq_resp from_chan2_time_resp
  - conVert: chk_freQ chk_impeD
    - Freq_resp: Freq_list Go
      - Log: First_freq Last_freq Num_points Ok
      - liN: System_rate sPecified_rate Ok
    - Impedance: Freq_list Go
      - Log: First_freq Last_freq Num_points Ok
      - liN: System_rate sPecified_rate Ok
  - Merge: Clear
    - Paste: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
    - Sum: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
    - Diff: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
    - Mult: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
    - diV: All Btwn_mkrs <=mkr1 >=mkr2 None Ok Cancel
    - draW: Ok Cancel
    - Freqs: Freq_list Ok
      - Log: First_freq Last_freq Num_points Ok
      - liN: System_rate sPecified_rate Ok
- Acquire: Look
  - Set_levels: Main_in(one) Cal_in(two) Out /Look
    - /Auto_adj: Main_only both_Independently Both_matched
  - Freqs: Freq_list
    - Log: First_freq Last_freq Num_points Ok

liN: System_rate sPecified_rate Ok
Modes
Zmode: Use_size/rate Optimized
Resp_mode: Use_size/rate Optimized
Gating: oN oFf
sWeep: Freq_resp Impedance
Systm: timeGap
Cfg_file: Save Use Init
Hardware_indep: Save Use
in_sCript_dir: Save Use Dir
Port: Printer read_Card_version
coDec: A B C D
Type_card: Orchid_sw orchid_Gw+ Cardinal Wearnes Echo Paradise Adaptec
coLor: Color B&w
Palette: Grid Background Reset
Lettering: l1 l2 l3
Curve: c1 c2 c3 c4 c5
Fill: f1 f2
Z_set: Probe_z Set_bal Ref_resistor
Use_bal?: Yes No
Inpts
Channels: Single Dual
Mic_input: Line(external_preamp) Mic(internal_preamp)
auto_Meas: Impedance Freq_resp
Setup:
Windows: None Blackman Hamming bIngham .bLackman .hAmming .binGham
cYcle: oN oFf
Aut_in_levs: oN oFf
User: List
Record: Boot alt-f2 alt-f3 alt-f4 alt-f5
T/s
vas_Method
closed_Box: cubicFt Litres
added_Mass: Grams Ounces
Extract: Normal Loaded newDriver
full_Auto: Normal Loaded newDriver
Setup: Ref_resistor Ohms/div
Dc_resistance: Define Extract
Effect_dia: Inches Cm
Cntrl: sHow beeP
sCripts: Easy
Custom: Directory runScript
Lim_fil: Upper(max) Lower(min) T/s_params scriptDir
Ext: Signal_out
select_Port: None lpt1 lpt2 lpt3
wait_for_Hi: wait_Now before_Acquire
wait_for_Lo: wait_Now before_Acquire
setup1: Pause setEval setAdj
pRint: Line Formfeed
tst_Mode: Freq_resp Impedance T&s
setup2: Main_in Cal_in Out
Siz: A B C D E F G {256, 512, 1024, 2048, 4096, 8192, 16384}
Rate: A B C D E F G H I J K L M N {5.125kHz...48kHz}

Inpt: Mic probe1
Windw: None Blackman Hamming bIngham .bLackman .hAmming .binGham
cHann: single1 dual2

OSCOPE CONTROL STRUCTURE

The Oscope/Gen instrument is controlled primarily via the function keys, as summarized below:

[F1], [Shift-F1]: HELP
[F2]: Set Horizontal Sweep
[Shift-F2]: Copy Plot to Printer
[F3]: Set Sample Rate
[F4]: (trigger **menu**) from:[ch1 ch2 None] edge:[Rising Falling] sweep:[Continuous Single] trigLevel
[F5]: Vertical Scale, channel 1 volts/division
[Shift-F5]: external attenuator setting for Channel 1
[F6]: Vertical Scale, channel 2 volts/division
[Shift-F6]: External attenuator setting for Channel 2
[F7]: Vertical Position, Channel 1, % of full-scale
[F8]: Vertical Position, Channel 2, % of full-scale
[F9]: (display **menu**) ch1_only ch2_only Dual_channel Input_gain_multiplier
[F10]: (generator **menu**) Frequency Line_level Amp_level (Gain_line) (gaiN_amp)
Wav: Sine sQuare Off
[Shift-F10]: Peak output alarm level (for use with power amplifier)
[=] : automatic display scaling
[Alt-M,S,P,D]: exit to one of the other instruments
[Alt-Q]: quit LAUD

SPECAN (FFT TYPE) MENU STRUCTURE

```
Type:  Fft(constant_width)  Rta(constant_q)
Display
        Format:
                freq_Range: Lower Upper
                Scale:  dB/div
                        Topval: Mic  Probe
                Log/lin:  loG  liN
                Graph: Lines  Tops
        Prntplot: togglePromo  Go
                Configure:Ibm/epson  Hp_jet  to_File
        Title: View Edit Hide
Acquire:  Continuous  Look
        Average:  Number  Go
        Set_levels: Main_in(one)   Out  /Auto_adj   /Look
        Noise:  White  Pink  oFf  Ok
Systm:  Ext_atten  timeGap
        Cfg_file:  Save  Use  Init
                Hardware_indep:  Save  Use
                in_sCript_dir:   Save   Use   Dir
        Port:  Printer  read_Card_version
                coDec:  A  B  C  D
                Type_card:  Orchid_sw  orchid_Gw+  Cardinal  Wearnes  Echo  Paradise
Adaptec
        Input_gain:  Card&preamp  Mic_pre
        coLor: Color B&w
                Palette:  Grid  Background  Reset
                        Lettering: l1  l2  l3
                        Curve: c1  c2  c3  c4  c5
                        Fill:  f1  f2
auto_Meas: Go
        User:  List
                Record: Boot alt-f2  alt-f3  alt-f4  alt-f5
Cntrl:   beeP
        sCripts:  Easy
                Custom: Directory  runScript
        Ext:  Signal_out
                select_Port: None lpt1 lpt2 lpt3
                wait_for_Hi: wait_Now  before_Acquire
                wait_for_Lo: wait_Now before_Acquire
        setup1: Pause
                pRint: Line Formfeed
        setup2: Main_in   Cal_in   Out
                Siz: A B C D E F G  {256, 512, 1024, 2048, 4096, 8192, 16384}
                Rate: A B C D E F G H I J K L M N {5.125kHz...48kHz}
                Inpt:  Mic probe1
                Windw: None Blackman Hamming bIngham .bLackman .hAmming .binGham
                cHann:  single1  dual2
```

SPEC_AN (RTA TYPE) MENU STRUCTURE

Type: Fft(constant_width) **Rta(constant_q)**
Display
    Format:
        freq_Range: Lower Upper
        Scale: dB/div
            Topval: Mic Probe
        Weights: A B C None
        Data_text: oN oFf
        show2nd: oN oFf
    Prntplot: togglePromo Go
        Configure:Ibm/epson Hp_jet to_File
    Title: View Edit Hide
    pRogressive_avg: Clear Show_prog_av Include_in_av
File: Save Retrieve Micdat
    Delete
        Ascii: .Imp .Frd .Zfr .Dat .dSt .Rta
        Compact: iM Fr Zf Cal Icf
    Change_dir: Base Select
Acquire: Go Look
    Avg_mode: Number Running Fixed
        Max_hld: oN oFf
    Set_levels: Main_in(one) Out /Auto_adj /Look
    Noise: White Pink oFf Ok
    Proc: Nomalize Cascade set2nd_curve Off (Do_now)
Systm: Ext_atten
    Cfg_file: Save Use Init
        Hardware_indep: Save Use
        in_sCript_dir: Save Use Dir
    Port: Printer read_Card_version
        coDec: A B C D
        Type_card: Orchid_sw orchid_Gw+ Cardinal Wearnes Echo Paradise
Adaptec
    Input_gain: Card&preamp Mic_pre
    coLor: Color B&w
        Palette: Grid Background Reset
            Lettering: l1 l2 l3
            Curve: c1 c2 c3 c4 c5
            Fill: f1 f2
auto_Meas: Go
    User: List
        Record: Boot alt-f2 alt-f3 alt-f4 alt-f5
Cntrl: beeP
    sCripts: Easy
        Custom: Directory runScript
    Ext: Signal_out
        select_Port: None lpt1 lpt2 lpt3
        wait_for_Hi: wait_Now before_Acquire
        wait_for_Lo: wait_Now before_Acquire
    setup1: Pause
        pRint: Line Formfeed
    setup2: Main_in Cal_in Out
        Siz: A B C D E F G {256, 512, 1024, 2048, 4096, 8192, 16384}

Rate: A B C D E F G H I J K L M N {5.125kHz...48kHz}
Inpt: Mic probe1
Windw: None Blackman Hamming bIngham .bLackman .hAmming .binGham
cHann: single1 dual2

DIST_AN MENU STRUCTURE

Display
    selectHarmonics: A B Thd
    Format: shoPhase? Level_scale
        stYle: Percent Level
        freq_Range: Lower Upper
        Dist_range: Lower% Upper% dbOffset dB/div Numdivs
    Prntplot: togglePromo Go
        Configure:Ibm/epson Hp_jet to_File
    Title: View Edit Hide
    show_Level: oN oFf
File: Save Retrieve
    Delete
        Ascii: .Imp .Frd .Zfr .Dat .dSt .Rta
        Compact: iM Fr Zf Cal Icf
    Change_dir: Base Select
Acquire: numCycles sWeep Look
    Mode: Gated Optimized
    Set_levels: Main_in(one) Out /Auto_adj /Look
    Freqs: Freq_list
        Log: First_freq Last_freq Num_points Ok
        liN: System_rate sPecified_rate Ok
Sys: Ext_atten
    Cfg_file: Save Use Init
        Hardware_indep: Save Use
        in_sCript_dir: Save Use Dir
    Port: Printer read_Card_version
        coDec: A B C D
        Type_card: Orchid_sw orchid_Gw+ Cardinal Wearnes Echo Paradise
Adaptec
    Input_gain: Card&preamp Mic_pre
    coLor: Color B&w
        Palette: Grid Background Reset
            Lettering: l1 l2 l3
            Curve: c1 c2 c3 c4 c5
            Fill: f1 f2
    timeGaps: quietGap measDelay
sWeep
auto_Meas: Go(regular_sweep)
    User: List
        Record: Boot alt-f2 alt-f3 alt-f4 alt-f5
    Power_step: Step_size Num_steps Harmonic_num shof1? Go
Visualize: Frequency dist_Multiplier Look Go
Cntrl: beeP
    sCripts: Easy
        Custom: Directory runScript
    Lim_fil: Upper(max) Lower(min) scriptDir
    Ext: Signal_out
        select_Port: None lpt1 lpt2 lpt3
        wait_for_Hi: wait_Now before_Acquire
        wait_for_Lo: wait_Now before_Acquire
    setup1: Pause setEval
        pRint: Line Formfeed

setup2: Main_in   Cal_in   Out
Siz: A B C D E F G {256, 512, 1024, 2048, 4096, 8192, 16384}
Rate: A B C D E F G H I J K L M N {5.125kHz...48kHz}
Inpt: Mic probe1
Windw: None Blackman Hamming bIngham .bLackman .hAmming .binGham
cHann: single1 dual2