

MLSSA

**Maximum-Length Sequence System Analyzer**

Reference Manual

Version 10WI

Rev 8

*DRA Laboratories*

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## 8. General

You are responsible for installation, management and operation of the MLSSA Program.



# 1. Introduction

*MLSSA* (pronounced Melissa) is an audio and acoustics measurement system based on maximum-length sequences.

*MLSSA* can measure and analyze many types of linear systems but its primary application is in the area of audio and acoustics measurements.

*MLSSA* is a single channel analyzer that can do the work of conventional dual-channel analyzers. The result is an effective doubling of useful bandwidth and a significant reduction in cost. Conventional FFT analyzers require two channels (each with its own antialiasing filter) to measure the system transfer function, using white noise as a stimulus. One channel samples the input noise while the other samples the resulting system output. The transfer function is then estimated using the cross-spectra or similar methods.

This dual channel technique can only estimate the transfer function because of the random nature of white noise and because of the windowing required to reduce spectral leakage. In contrast, *MLSSA* employs a special type test signal called a maximum-length sequence (MLS) as the preferred alternative to the conventional white noise stimulus. Unlike white noise, a maximum-length sequence is deterministic and periodic yet still retains most of the desirable characteristics of white noise. The deterministic nature of the MLS means that it can be pre-computed and need not be measured simultaneously with the system response. The periodic nature of an MLS means that a single time-domain circular cross-correlation calculation can recover the complete system impulse response without the need for data tapering windows. The result is zero windowing error as long as the entire period of the sequence is used to make the measurement, which *MLSSA* does automatically.

The MLS technique measures the impulse response — the most fundamental description of any linear system — from which a wide range of important functions are derived through computer-aided post-processing. The transfer function, for instance, is obtained by applying an FFT (Fast Fourier Transform) to a segment of the impulse

response. From the transfer function, *MLSSA* then derives and displays the frequency response, phase response, group delay and other curves.

A common misconception is that MLS methods and *MLSSA* are based on the FFT. An FFT analyzer performs its entire signal processing in the frequency domain to obtain the transfer function. This requires very many FFTs applied to segments of the acquired time data. Furthermore, each data segment must also be multiplied by a data tapering time window prior to performing the FFTs in order to prevent spectral leakage errors. These many windowed FFT results are then averaged together and scaled to recover the system transfer function. To obtain the system impulse response from an FFT analyzer requires an inverse FFT applied to this transfer function. The same holds for TDS analyzers, which also measure the transfer function in the frequency domain.

*MLSSA* takes exactly the opposite approach of an FFT analyzer. *MLSSA* uses cross-correlation to recover the system impulse response directly in the time domain without using any FFTs or inverse FFTs. The fact that *MLSSA* often applies an FFT to the measured impulse response to obtain the system transfer function or frequency response does not make it an FFT analyzer. *MLSSA* merely uses the FFT as post-processing operation to go from the time domain into the frequency domain.

An important advantage of *MLSSA* is its ability to make wideband long-duration impulse response measurements containing up to 65535 points. The time-bandwidth product of a *MLSSA* measurement can exceed 20,000. Conventional dual-channel FFT and chirp-based TDS methods are generally unable to achieve such a large time-bandwidth product in reasonably short measurement times. Consequently, there is an inevitable tradeoff between frequency resolution and frequency range when using these other techniques. This follows because true frequency resolution is given as  $1/T$  Hz where  $T$  is the duration of the measured impulse response in seconds. Dual-channel FFT and TDS analyzers, for instance, can typically measure and display an impulse response only 1024-points in length. Zoom processing in dual-channel instruments can provide very fine frequency resolution but this also limits the frequency range of the measurement to a very narrow band. *MLSSA*, however, easily makes wideband (20 kHz) transfer function measurements with true 1 Hz frequency resolution or, can

measure low bandwidth (1 kHz) transfer functions with 0.065 Hz resolution. Thus, there is no need for zoom processing with *MLSSA*.

Because of its large time-bandwidth product, a single *MLSSA* measurement would require dozens of separate TDS or FFT measurements. In addition, *MLSSA* includes all the post-processing algorithms needed to extract from the measured impulse response all the results obtainable by FFT and TDS measurements.

Furthermore, *MLSSA*'s integrated programmable antialiasing filter allows you to trade frequency resolution for bandwidth and visa versa. This eight-pole filter is under complete software control and allows the measurement bandwidth to be set anywhere from 1 kHz to 40 kHz. Filter gain is also programmable from -14 to +54 dB in 1 dB steps. The software uses the variable gain feature to perform autoranging during signal acquisition. As a result, *MLSSA* automatically accommodates both large and small input signal levels ranging from 10 millivolts to 20 volts RMS with almost no loss in A/D resolution.

Measurement systems without autoranging must use much higher resolution A/D converters to achieve the same resolution as *MLSSA* when the input signal level is well below maximum. This is because, without autoranging, low signal levels use very few bits of the available A/D resolution. For example, a 16-bit A/D with an input that is 24 dB below maximum will achieve an effective resolution of 12 bits. For an input 48 dB below maximum, a 16-bit A/D will achieve an effective resolution of only 8 bits. While *MLSSA* uses only a 12-bit A/D, autoranging insures that a signal fully 60 dB below maximum will still be digitized with at least 11 bits and usually 12 bits of effective resolution.

Furthermore, *MLSSA*'s 12-bit A/D is a successive approximation type that does not produce any acausal pre-ringing in the measured impulse response. The most commonly used 16-bit A/D converters, in contrast, use a sigma-delta conversion algorithm which causes acausal pre-ringing, that is, ringing which occurs before the first arrival of sound in the measured impulse response of a loudspeaker, for example. Such pre-ringing is undesirable since it complicates determining the exact acoustical

arrival time as well as proper windowing of the impulse response to eliminate room reflections in anechoic loudspeaker measurements.

*MLSSA 2000* cards output both the traditional white-MLS on J3 as well as a quasi pink-filtered MLS on J1. Older *MLSSA* cards output only a white-MLS on J3 and J1 must be left unconnected. If you own an older *MLSSA* card please be sure to leave J1 unconnected at all times even if this manual suggests otherwise. A *MLSSA 2000 Mothercard Upgrade Kit* is available from DRA Labs or authorized *MLSSA* distributors.

The quasi pink-filtered MLS has a signal-to-noise ratio advantage in room measurements because room noise is usually concentrated in the bass region. Since this filter boosts the lower frequencies of the MLS test signal, the signal-to-noise ratio in the bass region is improved as compared to using a white-MLS. The *MLSSA* software contains a precision high-speed digital inverse filter, which corrects both the magnitude and phase response of the on-board quasi pink filter.

## 2. Installation

### 2.1 System Requirements

*MLSSA* requires a PC having a full-length ISA or EISA expansion slot. An 8087, 80287 or 80387 numeric coprocessor is required except in 486DX and Pentium PCs, which include the numeric coprocessor on the main CPU chip.

*MLSSA* will run with full functionality under all MS-DOS versions 5.0 and above, Windows 95, Windows 98, Windows 98ME and Windows 98SE. For best performance, Windows 98SE or DOS 6.22 are suggested. For DOS installations only, a disk cache program such as SMARTDRV.EXE is highly recommended.

When running under Windows 95, 98, 98ME or 98SE, a Pentium PC with at least 24 MB of RAM is recommended. A fast Pentium PC with lots of RAM will allow you to take full advantage of *MLSSA* running in the Windows 98SE multitasking environment. Slower PCs or those with limited amounts of RAM can also be used but may result in frequently aborted measurements when other Windows applications are simultaneously open and active. Although the *MLSSA* software will automatically retry any aborted measurement until successful and will therefore not result in an error you might have to spend more time waiting for your results when running on slower PCs or those with limited amounts of RAM. Furthermore, it may be difficult to reliably run a *MLSSA* macro as a background task on slower PCs as foreground applications may hog all the available CPU time.

**NOTE:**

In order to run at full bandwidth under Windows 95/98 you must have a *MLSSA* card manufactured after 1995. If you purchased the *MLSSA* software as an upgrade and are not sure if your card will support Windows 95/98 remove it from your computer and read the copyright date etched on the backside (circuit side) of the card. A copyright date of 1995 or later indicates compatibility with Windows 95/98. If your *MLSSA* card is copyright 1988 or earlier you can run it at full bandwidth under DOS or under Windows running in MS-DOS compatibility mode. If you do not own a *MLSSA* card with a copyright date of 1995 or later the maximum bandwidth achievable running under Windows 95/98 will be limited to less than 10 kHz typical although the exact limit is PC dependent. You will get a "DMA buffer overflow" error message during data acquisition whenever this limit is exceeded when running under Windows 95/98. *MLSSA* users with older cards can purchase a *MLSSA* 2000 Mothercard Upgrade Kit.

## 2.2 Software Installation

The *MLSSA* installation diskette contains special calibration files for the particular pair of AFM-50 filter modules mounted on your *MLSSA* card. The hyphenated pair of 4-digit numbers (DDDD-DDDD) handwritten on your *MLSSA* installation diskette should agree with the last four digits of the corresponding pair of 8-digit AFM-50 serial numbers. Assuming you have the correct *MLSSA* installation diskette for your *MLSSA* card filters, insert the diskette into drive A and from Windows 95/98 click Start then Run and enter:

```
a:install
```

and click OK.

If you are installing from DOS type:

```
a:install <enter>
```

To avoid conflicts under Windows 95/98 between *MLSSA* and Plug-and-Play devices you must reserve the DMA and I/O resources required by *MLSSA*. It may also be necessary to reserve the interrupt used by *MLSSA*.

From Windows double click My Computer then Control Panel then System. Click on the Device Manager tab then double click on Computer, which is the first device listed. Click on the View Resources tab and click the interrupt request (IRQ) button. Note if any other device is using IRQ 3 or whatever the interrupt setting is for your *MLSSA* card. If a serial port either COM1 or COM2 is using that interrupt then there should be no conflict and you should not reserve the interrupt for *MLSSA* since doing so will prevent the serial port from also using it. If another device is using the *MLSSA* interrupt, click on the Reserve Resources tab and click Add. Enter the interrupt number *MLSSA* is configured to use (default = 3) and click OK to reserve it. If Windows issues an error message indicating a conflict, restart your computer and see if the conflict goes away. When restarted, Windows will try to reconfigure Plug and Play devices to use another interrupt thus resolving the conflict. To check this return to the View Resources tab and look to see if there is still a conflict.

There are no known standard devices able to share a DMA channel with *MLSSA*. Therefore, you should always reserve a DMA channel for *MLSSA*. From Windows double click My Computer then Control Panel then System. Click on the Device Manager tab then double click on Computer, which is the first device listed. Click the Reserve Resources tab and then on the Direct Memory Access (DMA) button. Click Add and enter the DMA channel *MLSSA* is configured to use (default = 1) and click OK. If there is a conflict, you will have to change the *MLSSA* DMA channel or change or disable the conflicting device's DMA.

Click on the Input/Output (I/O) button under Reserve Resources, click Add and enter the I/O port range used by *MLSSA*. The default here is 03e0 hex through 03ef hex. Click OK to reserve the I/O range. Click OK to save all the entries and be sure to restart your computer in order for the changes to take effect.

To change the interrupt, DMA or I/O resources used by *MLSSA* please refer to section 2.8 below titled "Resolving Bus Conflicts".

Create a shortcut to *MLSSA* on your Windows 95/98 desktop. To create the shortcut click Start then Find then Files-and-folders. Enter *MLSSA.EXE* in the Named-field and the full DOS path (drive and directory) into the Look-in field then click Find-now. Point

to the [MLssa] program icon and while holding down the left mouse button drag it to the Windows desktop then release to create the shortcut. Left click the shortcut name and edit it to read "MLSSA" then click the desktop to finish.

Right click the shortcut icon then click Properties. Click the Program tab and edit the "Cmd line:" field by appending a "?" after the program name so that Windows will prompt you for command line parameters before starting *MLSSA*. Click the Misc tab and move the Idle-sensitivity slider to the low position. Click Apply then click OK to finish.

Low idle sensitivity tells Windows to keep running *MLSSA* even when there is little keyboard input thus insuring that Windows will not suspend a *MLSSA* macro running as a background task.

## 2.3 Hardware Installation

**CAUTION:**

Never work on the inside of your computer while it is connected to the AC power lines.

**RECOMMENDATION:**

The *MLSSA* card consists of the A2D-160 sampling digitizer card with two AFM-50 antialiasing filter modules installed. If you intend to install the *MLSSA* card into a portable computer that will be moved about or, into a tower-style computer in which the card will be mounted upside down, there is a risk that the AFM-50 filter modules may eventually work themselves loose and cause a malfunction. To avoid this risk it is recommended in such cases that both AFM-50 filter modules be secured to the A2D-160 mothercard using plastic cable ties or thick rubber bands wrapped lengthwise over the top of each module and under the A2D-160 mothercard. Two cables ties in series are usually required for each module.

Disconnect your PC from the AC power line and remove the cover from your computer in accordance with the manufacturer's recommendations. Locate an unoccupied full-

length ISA or EISA expansion slot. The card is compatible with both AT style (16 bit) or PC style (8 bit) ISA slots as well as EISA slots.

Remove the slot's cover plate located at the rear of your machine by removing the single screw retaining it. Install the *MLSSA* card into the expansion slot and secure the card's rear bracket with the screw you removed from the cover plate.

Now is a good time to check that your computer contains an 8087, 80287 or 80387 numeric coprocessor. The *MLSSA* software will not run if a coprocessor is not present. Note that the 80486DX and Pentium processors do not require a separate coprocessor chip while the 80486SX does.

Before reassembling your computer, proceed to the checkout section below.

## **2.4 System Checkout**

On the rear panel bracket of the *MLSSA* card locate three RCA phono jacks and a single 9-pin "D" connector arranged in a vertical row. These have the following functions.

<i>Designation-location</i>	<i>Function</i>
J1 top phono jack	Pink-filtered Analog Output
J2 middle phono jack	Analog input
J3 bottom phono jack	Normal Analog Output
J4 (9 pin "D" connector)	Various digital TTL I/O signals

See Appendix E for more details about these connectors including impedances, maximum safe operating voltage limits and the functions of the digital I/O signals available on J4.

**CAUTION:**

If your *MLSSA* card was made before the year 2000, phono jack J1 is not used and must be left disconnected at all times. Only *MLSSA 2000* cards output a pink-filtered MLS on J1. *MLSSA 2000* cards have a copyright date of 2000 etched on the backside.

Use an audio patch cable to connect J2 to J3 thus completing a loopback connection from the analog output to the analog input.

Power up your computer.

If your computer is running Windows 95/98, start *MLSSA* by double-clicking its shortcut icon. Click OK when prompted for command line arguments.

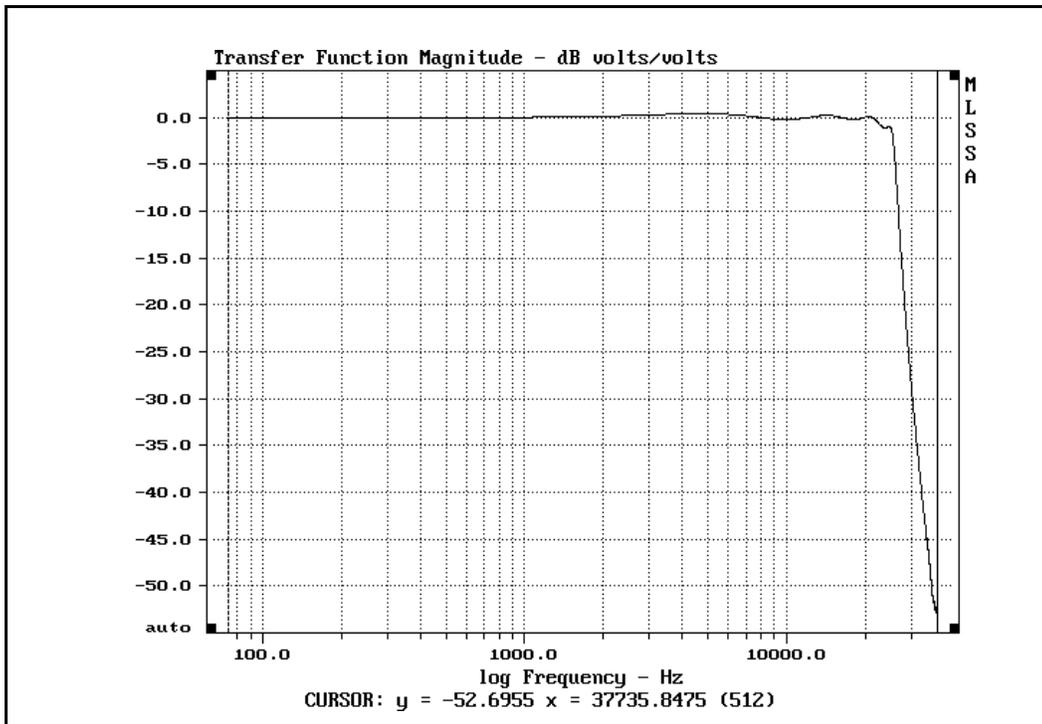
If your computer is running DOS, change to the *MLSSA* directory and then start *MLSSA* from the DOS command line. For example, if the *MLSSA* directory is "MLS" type:

```
CD\MLS <Enter>
```

```
MLSSA <Enter>
```

After *MLSSA* is initialized, type the letters "g" (for Go) followed by "o" (for Once) to initiate a measurement sequence. *MLSSA* should acquire 65,535 samples and compute the first 16,384 points of the impulse response. The result is the impulse response of *MLSSA*'s antialiasing filter.

The filter's transfer function is obtained by computing an FFT on the measured impulse response. To do this, type the letters "f" (for FFT) followed by "e" (for Execute). *MLSSA* should now perform a 1024-point FFT, compute the magnitude of the filter's transfer function and enter the frequency domain. The displayed magnitude response should be reasonably flat up to about 25 kHz and then sharply roll-off above 25 kHz (see figure below). This curve is the magnitude response of the Chebyshev antialiasing filter.



**Loopback transfer function**

The Chebyshev filter's transfer function will not be perfectly flat however. A perfect Chebyshev filter will show 0.5 dB of ripple. In addition, production tolerances will sometimes result in larger deviations from flatness up to 1 dB from 0 to 20 kHz. Furthermore, the response may be down as much as 3 dB at the 25 kHz cut-off frequency. Such small deviations from flatness are normal and do not necessarily indicate a hardware malfunction. Besides, any residual antialiasing filter errors can be corrected, if desired, through the Reference Store command in the frequency domain. Press "rs" for Reference Store and observe that the response is now perfectly flat.

Press "e" for EXIT to return to the time domain and then "q" to quit *MLSSA*.

Unless you have encountered a problem, power down and reassemble your computer. Otherwise, see the "Troubleshooting" section below.

## 2.5 System Optimization

To maximize the amount of available conventional memory it is recommended that you install EMM386.EXE and specify the NOEMS option in your CONFIG.SYS file. This configuration maximizes the amount of conventional memory available to *MLSSA* when running under Windows 98SE. If they are not already present in your CONFIG.SYS file, you should add the following lines to it using a text editor and then restart Windows:

```
DEVICE=C:\WINDOWS\HIMEM.SYS
DEVICE=C:\WINDOWS\EMM386.EXE NOEMS
DOS=HIGH,UMB
FILES= 30
BUFFERS=17
```

If you run *MLSSA* under MS-DOS version 5.0 or above, *MLSSA* can access up to 160K or more bytes of UMB memory. Note that Windows 95/98 normally hogs all available UMB memory and therefore UMB memory is not directly available to *MLSSA* when running under Windows 95/98.

*MLSSA* temporarily buffers overlays and certain other data to your hard disk. By default, *MLSSA* buffers the data to its own directory but you can override this default and buffer data to another drive or directory by setting the TMP environment variable in your AUTOEXEC.BAT file. For example, to buffer data to the directory D:\DATA just add the line:

```
SET TMP=D:\DATA
```

to your AUTOEXEC.BAT file.

## 2.6 A Quick Start

Now that you have installed and checked out *MLSSA*, it's important to complete the tutorial given in Section 4.0. The knowledge covered by that section is indispensable if you want to quickly learn to use the program effectively. After completing the tutorial, you can then proceed to Section 6.0, which covers specific applications of *MLSSA*. Later, when you have the time, read the other sections of this manual.

## 2.7 Troubleshooting

If one of the system checkout steps fails, this section may help you determine the cause and take corrective action.

Symptom 1: After starting *MLSSA* the message "Board not installed at base address: 3e0" is displayed. After the main *MLSSA* menu appears and a Go Once command is issued the message "Board not installed" is displayed.

Probable Cause: Another card installed in your computer such as a serial port adaptor may be in conflict with the *MLSSA* card's required I/O space. *MLSSA* requires 16 contiguous I/O ports, beginning with the I/O port base address. For example, for an I/O port base address setting of 3e0 hexadecimal, I/O ports from 3e0 through 3ef hexadecimal inclusive must not be in use by any other device in your PC. A conflict with any one of the I/O ports in this range can result in this error message. See section 2.8 below titled "Resolving bus conflicts".

Possible Cause: You changed DIP-switch SW1 on the *MLSSA* card to an I/O port base address other than 3e0 but did not edit *MLSSA.INI* to reflect the change. Use a text editor such as Notepad under Windows or EDIT under DOS to change the I/O port base address line of *MLSSA.INI*. This file is located in the *MLSSA* directory.

Possible Cause: If you have a *MLSSA* card made before 1995, the bus speed of your computer may not exceed 8.25 MHz. *MLSSA* cards made after 1995 including *MLSSA 2000* and *MLSSA 95* cards will work with ISA bus speeds as high as 16 MHz. To determine the age of your card, power down your computer and remove the card. Look for the copyright date etched on the back of the card. A copyright date prior to 1995 means you cannot run at ISA bus speeds over 8.25 MHz. Note that ISA bus speed refers to the rate at which data is read or written to ISA expansion bus cards whereas the CPU speed refers to the rate at which your computer executes instructions and is usually much higher than the ISA bus speed. Check your computer documentation or contact the computer manufacturer to determine the ISA bus speed for your PC. An ISA bus speed of 8 MHz is the de facto standard but some recent PCs may exceed 8 MHz. In some cases, it's possible to change the ISA bus speed via a setup utility that

came with your PC. You should run the setup utility that came with your computer and select a lower bus speed or a longer bus cycle time and retry. When in doubt about the bus speed of your computer, check with the computer manufacturer.

Symptom 2: After starting *MLSSA* the message "MLSSA card trigger failure —possible ISA bus conflict" or "MLSSA card malfunction —possible ISA bus conflict" is displayed.

Probable Cause: Another card installed in your computer such as a serial port adaptor may be in conflict with the *MLSSA* card's DMA channel or interrupt. See section 2.8 below titled "Resolving bus conflicts".

Symptom 3: *MLSSA* starts normally but does not respond to commands and the computer cannot be rebooted by pressing the Ctrl-Alt-Del key combination.

Probable Cause: Your keyboard may not be IBM compatible. *MLSSA* normally tries to speed up the keyboard typematic rate to make the cursor move faster. Typematic rate refers to the repeat rate generated by the keyboard when a particular key is held down. Edit your *MLSSA.INI* file using a standard text editor such as Notepad under Windows or EDIT under DOS and change the "Keyboard typematic" field from "speed up" (1) to "leave alone" (0). Exit the editor with changes saved and restart *MLSSA*.

Symptom 4: *MLSSA* starts normally but the Go Once command results in the message "Input Exceeds 25 Volts!" when clearly this is not the case.

Probable Cause: One of the two AFM-50 antialiasing filter modules installed on the *MLSSA* card is loose and needs to be resealed. These modules sometimes become loose during shipping. Power down your computer and remove the card. Place it component side up on a flat surface. Firmly press down on each filter module to insure each is completely seated. Note that if you are installing *MLSSA* into a portable computer that will be taken on the road, it is recommended you secure each filter module to the main card using plastic cable ties or heavy rubber bands wrapped around each module. Two cable ties can usually be connected in series to surround each module thus mechanically securing it to the main card.

Possible Cause: This problem can also result from an I/O port conflict. *MLSSA* requires 16 contiguous free I/O ports starting at the I/O port base address. See section 2.8 titled "Resolving Bus Conflicts" below.

## 2.8 Resolving Bus Conflicts

*MLSSA* requires one interrupt (IRQ), one DMA channel and 16 contiguous I/O ports to communicate with the *MLSSA* card. If another device in your computer tries to use the same interrupt, DMA channel or I/O port selections the resulting conflict may cause a malfunction, which may occur only intermittently. Usually, the conflict causes *MLSSA* to display an error message such as "Board not installed at base address: 3e0" if it is an I/O port conflict or "A2D-160 trigger failure" if it is a DMA (Direct Memory Access) or interrupt conflict. The following table shows the ISA expansion bus selections normally used by *MLSSA*.

### *Default ISA Bus Selections*

I/O ports	3E0 through 3EF hex
DMA channel	#1
Interrupt (IRQ)	#3

If your PC has the ability to boot into Windows, you should check for a conflict from inside Windows. From Windows double click My Computer then Control Panel then System. Click on the Device Manager tab then double click on Computer, which is the first device listed. Click on the View Resources tab and click the Interrupt request (IRQ) button. Note if any other device is using IRQ 3 or whatever the interrupt setting is for your *MLSSA* card. If a serial port either COM1 or COM2 is using that interrupt then there should be no conflict and you should not reserve the interrupt for *MLSSA* since doing so will prevent the serial port from also using it. If the words "System reserved" are displayed next to the IRQ, it means you had previously reserved it. If any other device name is associated with it, you have a conflict.

Click on the Direct Memory Access (DMA) button. Note if any other device is using DMA channel 1 or whatever the DMA setting is for your *MLSSA* card. If the words "System reserved" are displayed next to the selected DMA channel it means you had

previously reserved it. If any other device name is associated with the DMA channel in question then there is a DMA conflict.

Click on the Input/output (I/O) button. Note if any device is using I/O ports 3e0 through 3ef or whatever the I/O port range setting is for your *MLSSA* card. If the words "System reserved" are displayed next to the I/O ports in question, it means you had previously reserved them. If any other device name is associated with the I/O ports in question, you have a conflict.

If you cannot boot into Windows, check with the computer manufacturer to determine which interrupts (IRQs), DMA channels and I/O ports are free and which are already in use by other devices. This will enable you to quickly determine the source of the conflict and what corrective action is required to resolve it.

Most computers have many devices built into their motherboards that also require expansion bus resources including interrupts, DMA channels and I/O ports. You can usually change the selections for these devices using a special setup program supplied with the computer or from inside Windows. This is often easier than changing the bus selections used by *MLSSA* to resolve the conflict.

If there is no conflict with the standard devices in your computer and *MLSSA* still does not start correctly, check to see if there are any non-standard adaptor cards installed in your computer. If so, try removing them one at a time to determine which one is causing the conflict. If you find that removing a particular card solves the problem refer to the documentation for that card to determine what interrupt, DMA channel or I/O port is in conflict with *MLSSA*. Either you must then change the other card's or *MLSSA*'s default selections to resolve the conflict. To change the other card's selections refer to the documentation covering the other card.

You can also resolve a bus conflict by changing the DMA channel, interrupt number or I/O port base address used by *MLSSA*. You can choose interrupt numbers 3 or 5. Although interrupt number 2 can also be selected, its use is not recommended because interrupt number 2 is the cascade interrupt on most computers and its use can cause other problems. You can choose DMA channels 1, 2, or 3 and any I/O port base

address between 210 and 3F0 hex ending in a hex 0. Remember that *MLSSA* needs 16 contiguous ports beginning with the base address you select. For example, base address 3F0 causes *MLSSA* to use ports 3F0 through 3FF inclusive.

Refer to Appendix A to change the jumper and switch settings to re-configure your *MLSSA* card for a different I/O base address, a different DMA channel or, a different interrupt. After changing any jumpers or switch settings, you must edit the text file *MLSSA.INI*, located in the *MLSSA* directory, to reflect any changes you made. The first three lines of this file tell the software what I/O port base address, DMA channel and interrupt (IRQ) are used by the *MLSSA* card. Use a text editor such as Notepad from Windows or EDIT from DOS to edit *MLSSA.INI*. You can also use a standard word processor provided you save your changes in plain text format.

**WARNING:**

When re-configuring the *MLSSA* card jumpers and switches to resolve a bus conflict you must edit the text file *MLSSA.INI* to reflect any changes you made. The first 3 lines of this initialization file tell the software what I/O port base address, DMA channel and interrupt (IRQ) are used by the *MLSSA* card.

## 2.9 Upgrading PCs and Swapping Cards

If, after the initial installation described above, you decide to install the *MLSSA* card into a new computer you should *not* re-install the *MLSSA* software from the original distribution disks or from any *MLSSA* software upgrade disks but rather copy the entire *MLSSA* directory from the old PC to the new PC. This method assures that any special setup files you created and any important data files you saved will be transferred along with the software. You can copy over the old directory using several methods. One method is to make a backup of the old directory on floppy disks or tape followed by a restore operation onto the new computer. Another method is to purchase commonly available programs such as LapLink, which are able to copy files and entire directories from one PC to another using the parallel or serial port interfaces.

Note also that there are special calibration files *MLSSA* uses to correctly program the AFM-50 antialiasing filter modules. These files are unique to each *MLSSA* card so you should never swap cards (assuming you own more than one) without also swapping these special files.

These special files are:

BUTTER.FCF  
BESSEL.FCF  
CHEBY.FCF  
AF?????.CAL  
AF?????.CAL

These files reside in the *MLSSA* directory. The four question marks in the two CAL files refer to the last four digits of the serial numbers of the two AFM-50 antialiasing filter modules installed on your particular *MLSSA* card. These CAL files contain calibration data specific to each AFM-50 filter module.

These calibration files are also always on the initial distribution disk, that is, the disk that came with the card when you purchased it. Note that they are not present on any *MLSSA* software upgrade disks so it is wise to back up your original distribution disks and secure them in a safe place in case these special files ever are deleted or lost.

Note that if you own *MLSSA SPO* you must also swap file COMPARMS.OVL in addition to the files listed above in order for *MLSSA SPO* to work correctly when you swap *MLSSA* cards.

## 3. Reference

This section gives a summary of *MLSSA* functions and features. If you are a new user, we recommend that you complete the *MLSSA* commands tutorial given in Section 4 of this manual. Detailed descriptions of each command are provided by the on-line help feature of *MLSSA*.

### 3.1 Windows Considerations

Windows 98SE is recommended as the best operating system for *MLSSA* but it will also run with full functionality under Windows 95, 98 and ME, as well as under all DOS versions 5.0 and above. *MLSSA* will also run under Windows XP, 2000 and NT but you cannot perform actual measurements under these Windows versions. You can however load and analyze TIM and FRQ data files stored on disk. Whenever *MLSSA* detects it is running under Windows XP, 2000 or NT it automatically enters analysis mode. In addition, under these Windows versions, printing is significantly slower than under Windows 98SE, both when printing direct from *MLSSA* as well as when printing through the *Pizazz 5* printer driver.

Because *MLSSA* is a DOS-based application that normally runs in a graphics video mode, Windows 98SE will normally force it to run as a full-screen application. Most of the time this is desirable but in some cases it's useful to run *MLSSA* in a window or as a background task. There are two ways to run *MLSSA* in a window or in the background under Windows 98SE. One way is to run *MLSSA* in CGA graphics video mode. Switching to CGA graphics can be accomplished by pressing Alt=;. Pressing Alt= again switches back to VGA graphics mode. Note here that under Windows XP, 2000 and NT, *MLSSA* can only be run in full screen mode or idled in the task bar but cannot be run in a window.

*MLSSA* will run in a window when you switch to CGA graphics but Windows 98SE will initially display *MLSSA* full-screen. You switch from running full-screen CGA to running in a window by pressing Alt-Enter. To smoothly and easily switch from running full screen in VGA graphics mode to running in a window in CGA graphics mode, hold

down the Alt key then press the = key, release the = key, wait a moment, then press the Enter key. To resume full screen operation in VGA mode press Alt+= again.

If you press Alt-Enter while *MLSSA* is running full screen in EGA or VGA mode Windows 98SE will suspend execution of *MLSSA* even though it will still be placed in a window. In such cases *MLSSA* can only be resumed by going full-screen again either by clicking on the full-screen icon in the window toolbar or by pressing Alt-Enter again.

Pressing Alt-9 causes *MLSSA* to enter Character mode, which switches to a 25-line text video mode. In Character mode, *MLSSA* will also run in a window or as a background task. Obviously, it is not possible to display *MLSSA* graphs in Character mode but in many cases, there is no need to see the graphs until the measurements are completed. You resume graphics mode by pressing Alt-9 again.

For example, you may want to run the POLAR.MAC macro to make polar loudspeaker measurements but you don't need to see the results until the macro is completed. To do this you set up *MLSSA* to run the macro, press Alt-9 to enter Character mode, start the macro and then click the minimize window icon [-] to run *MLSSA* as a background task. You can then open other Windows applications and run them while *MLSSA* is executing the macro in the background. When you want to see the results you simply click on *MLSSA* in the task bar to open it and switch back to graphics mode by pressing Alt-9 again. You could also do this by first switching *MLSSA* to CGA graphics using the Alt+= command, pressing Alt-Enter to run *MLSSA* in a window, start the macro and click the minimize icon to run *MLSSA* in the background.

In Character mode, all of *MLSSA*'s commands, including the graphics commands, function normally — only the graphs themselves are not displayed and the mouse is ignored. This is to allow any macro to run in a window or as a background task without modification. The mouse is disabled in Character mode to prevent inadvertent *MLSSA* cursor movements.

Windows 98SE is a multitasking operating system. This means that Windows allocates CPU time and other system resources such the DMA and interrupt controllers between *MLSSA* and other programs. It is therefore possible for Windows 98SE to occasionally

cause an acquisition error such as DMA overflow. This can occur when Windows is working intensely on another task such downloading a file or printing a document. It can also occur during rapid mouse movements. *MLSSA* automatically retries any aborted acquisition sequence after a 4-second pause. To stop *MLSSA* from retrying an aborted acquisition sequence, hold down both the left and right Shift keys simultaneously.

The Acquisition Retry command lets you disable or re-enable the acquisition retry feature. Note however that when running under Windows 98SE in either Character mode or in CGA graphics mode, *MLSSA* will always retry an aborted acquisition even when retry is disabled by this command. This is done to insure that a macro running as a background task will not stop prematurely due to an aborted acquisition sequence.

Concurrent pre-averaging mode will also work under Windows 98SE as long as other active Windows programs are not running simultaneously. Otherwise, *MLSSA* may not be allocated enough CPU time to keep up with the data being acquired, especially at high sampling rates. For this reason it is advisable that you use Concurrent pre-averaging only when running *MLSSA* full-screen and only after first closing all other active Windows programs. You can also chose Alternate pre-averaging mode through the Acquisition Pre-average Mode command. With Alternate pre-averaging, you can run other active Windows programs simultaneously with impunity.

When running in character mode or CGA graphics mode *MLSSA* automatically switches to Alternate pre-averaging even when Concurrent pre-averaging is selected. This is to insure that a *MLSSA* macro running as a background task will not retry data acquisition indefinitely.

### 3.2 *MLSSA* Help

F1	General and context sensitive help
Alt-h	Statistics commands help

*MLSSA* incorporates extensive on-line help screens. For the menu commands, context sensitive help is available at any time by pressing function key F1. At the top menu

level, pressing F1 displays general help and help for all graphics and utilities commands. Pressing Alt-h displays help for the statistics commands. *MLSSA* help screens are an important source of information regarding *MLSSA* commands. You should read the help screens associated with each command until you become familiar with the program. Once you have learned the commands and no longer need to read the help screens, you will be able to select and execute commands very rapidly using only a few keystrokes.

### **3.3 Graphics Commands**

Graphics control commands are executed using the cursor keypad, which includes the Home, End, PgUp PgDn and arrow keys. In addition, the function keys are also used for graphics functions. Zooming, scrolling, panning, cursor movements, cursor style are all controlled through these keys. The Left and Right arrow keys and the mouse, if installed, move the graphic cursor (solid vertical line) left and right. Pressing F9 or the right mouse button moves the marker (dashed vertical line) to the current cursor position. Positioning the marker and cursor is critical to many other *MLSSA* commands. For example, the FFT Execute command normally operates on data points framed by the marker and cursor. Similarly, the Alt-key statistics commands and the zoom command (F10) both operate on data points lying between and including the marker and cursor positions. For a summary of the graphics commands, press F1 then PgDn and read the help screen provided. After a little practice, you'll find it easy to manipulate the graphics display.

The graphics commands are covered in detail by the tutorial of section 4 of this manual and a summary of the graphics commands is given below.

*** Function keys ***		*** With Shift or [Ctrl] key ***	
F1	.... General and contextual HELP	.....	Expand Graphics Display
F2	.... Move cursor to index	.....	Move cursor to X value [Y value]
F3	.... Plot lines/points	.....	Erase display [Blank display]
F4	.... Linear/log/IEC horizontal scale	..	Autoscale [overplot] on/off
F5	.... Cursor position display on/off	...	Cross-hair [Blank LF region]
F6	.... Expand vertical	.....	Compress vertical [Set scale]
F7	.... Scroll left	.....	Default width
F8	.... Scroll right	.....	Draw grid [grid always on/off]
F9	.... Move marker to cursor	.....	Cursor to marker [w/ value offset]
F10	... Zoom from marker to cursor	.....	Unzoom [move marker with cursor]
*** Cursor keys ***		*** with Ctrl key ***	
Right	. Cursor right	.....	Cursor right by large increment
Left	.. Cursor left	.....	Cursor left by large increment
Up	.... Scroll up 1 division	.....	Not used
Down	.. Scroll down 1 division	.....	Not used
PgDn	.. Scroll down 3 divisions	.....	Resize display
PgUp	.. Scroll up 3 divisions	.....	Resize display w/o moving marker
Home	. Cursor to left edge	.....	Cursor to signal start
End	... Cursor to right edge	.....	Cursor to signal end

### Graphics commands

The overplot feature allows you to plot an unlimited number of graphs simultaneously. This feature is somewhat like overlays except that there is no memory storage of the data. Unlike overlays, overplots can be used to overlay different types of curves having comparable units. For example, you can overplot a Schroeder plot and an energy-time-curve. The overplot feature is turned on and off by pressing Ctrl-F4. Autoscale is also automatically turned off during overplots to preserve constant vertical axis scaling. The overplot color is set through the Library Colors Main oVerplot command. The sign of the overplot color determines overplot line style. A positive color selects dotted lines while a negative number selects solid lines. You can also create macros that change the overplot color between traces to make multicolored graphs. Note that the overplot feature is automatically turned off if you do anything that temporarily erases the graphics screen such as pressing F1 for help or executing the Setup command.

Normally, *MLSSA* automatically changes the vertical scale to best fit the actual measurement data. This feature is called "autoscale". You can disable autoscale by pressing Shift-F4 in order to fix the vertical scale. Once in the "fixed" mode, the vertical scale factor can be changed manually using F6 (expand vertical), Shift-F6 (compress vertical) and the vertical offset changed through the Up arrow, Down arrow, PgUp and

PgDn keys. You can also numerically enter your own custom vertical scale factor and offset through the Ctrl-F6 graphics function. A fixed vertical scale stays in effect until you Quit *MLSSA* and return to DOS. In order to restore the same vertical scale later, you must first store the current fixed vertical scale to a setup file through the Setup Save command before quitting *MLSSA*. You can then restore the same vertical scale later simply by loading that setup file through the Setup Load command. Note that the time-domain and frequency-domain vertical scales can be set independently. In general, all other graphics setup information is stored in that setup file as well.

The status of the autoscale and overplot features for the main data (but not any overlay data) is displayed on the outside lower left-hand corner of the graph. These three status words have the following meaning:

"auto"	Autoscale is on
"fixed"	Autoscale is off
"overplot"	Overplot is on, autoscale is off.

Note also that *MLSSA* remembers the previous state of autoscale when overplot is activated. It returns to the original autoscale status (on or off) after overplot is deactivated.

The vertical scale and units for overlays need not match the vertical scale and units of the main trace. The type of vertical scale for plotting overlays is controlled by the Overlay Vertical-scale command, which is available in both the time and frequency domains. There are three options: Normal, Auto and Fixed. The Normal option causes overlays to be plotted with same vertical scale as the main trace. The Auto option causes overlays to be autoscaled independently of the autoscale function, if enabled, for the main data. In that case, the possibly different vertical scale labels for the overlay data are drawn just below the vertical scale labels for the main trace data. There thus can be two different vertical scales on the screen simultaneously when plotting overlays. See section 3.20 Overlays and Overplots for more information.

### 3.4 Statistics Commands

*MLSSA* provides a set of statistical functions that operate on data bounded by the marker and cursor positions. These are invoked by holding down the Alt key and pressing one of the alphabetic keys and the result is displayed in the message window below the graphics area. For example, Alt-m computes and displays the mean value and Alt-r computes and displays the RMS value. All statistics commands detect whether a given curve is plotted on a decibel (dB) or a linear scale and will make any necessary corrections, as required. For example, if the data is plotted on a dB scale and Alt-r is executed for the RMS value, *MLSSA* first converts all the data between the marker and cursor from dB to the underlying internal units, performs the RMS calculation on this internal data and finally converts the result back to dB for display in the message window. This conversion is automatic and does not disturb the displayed curve. Alt-h brings up the help screen, which describes all the Alt-key commands available. This help screen is reprinted below for reference.

The statistics commands normally operate on the main data curve, however, when an overlay is plotted over the main data through the Overlay Plot command, the statistics will be computed based on the overlay data instead.

**Alt-key functions summary:**

Alt-a All statistics: mean, rms, standard deviation, max and min.  
Alt-e Extrema: max, min and their respective locations.  
Alt-d Deviation: peak +/- deviation from mean.  
Alt-r RMS: root mean square.  
Alt-s Standard deviation from mean.  
Alt-v Variance from mean.  
Alt-m Mean value.  
Alt-p Peak-to-peak (max - min).  
Alt-c Crest factor.  
Alt-l Slope by linear regression.  
Alt-i Integrate (summation of points in frame).  
Alt-n Number of points framed by marker and cursor.  
Alt-f Frame: current cursor and marker positions  
Alt-x Delta-x between cursor and marker positions.  
Alt-y Delta-y between cursor and marker positions.

**NOTES:**

The Alt-key functions compute statistics on plotted data falling between the marker and cursor positions. Results are given in currently displayed units. If an overlay has been plotted, statistics are based on the overlay trace data. Otherwise, they are based on the main trace data. Press SHIFT-F3 to erase an overlay trace from the screen.

**Statistics commands help screen**

### **3.5 Utility Commands**

*MLSSA* utility commands are invoked by holding down the Alt key while pressing one of the numeric keys 1 thru 9. A list of all utility commands is displayed by pressing F1 from the main menu and paging down two pages. The utilities help screen is reprinted below for reference.

**Utilities HELP:**

Alt-0 Dumps any text or graphics screen "as is".  
 Alt-1 Prints any text or graphics screen "as is".  
 Alt-2 Displays available DOS and UMB memory in bytes.  
 Alt-3 Displays filter module serial numbers.  
 Alt-4 Captures graphics or text screens and saves them to files.  
 Alt-5 Display current MLSSA version.  
 Alt-6 Mark current time.  
 Alt-7 Display elapsed time from last mark (Alt-6).  
 Alt-8 Display a message. If a macro is executing wait afterwards.  
 Alt-9 Toggles Character mode on and off.  
 Alt-9 Toggles character mode on and off.  
 Alt-= Toggles between VGA and CGA graphics mode. Toggle to CGA mode then press Alt-Enter to run in a window.

**NOTES:**

Alt-1 prints graphics or text screens without modifying them in any way.  
 Alt-2 displays the largest available memory block in the "DOS" field.  
 Alt-2 displays the largest upper memory block in the "UMB" field.  
 Alt-3 displays your filter module's serial numbers.  
 Alt-4 captures graphics screens to PCX files or text screens to plain text ("TXT") files.  
 Alt-8 use this command in place of Macro Command wait or Macro Command Pause when in the expanded graphics mode (Shift-F1) to insure your message is visible to the operator.

**Utilities commands****3.6 Menu Commands**

*MLSSA* uses a hierarchical menu scheme. Menu commands are executed starting at the top menu level by typing the highlighted letter of a menu selection. This action immediately presents you with the next menu level selections. Pressing the highlighted letter of your selection at the new menu level continues you on to the next level and this process continues until the final level is reached. At the final menu level, you may be prompted (depending on the particular command) to enter a filename, a number or to select an option from a list. If you press F1 at any menu level, a help screen will appear which is relevant to the current menu context. At any menu level, short of the point at which the command is actually executed, you can abort the command and return to the top menu level by pressing the Esc (escape) key. When you reach the final menu level of a command, *MLSSA* will display the current value of the relevant parameter before you actually execute the command and change that parameter. Therefore, if you press Esc at this point you will have determined the current parameter setting without changing it to a new value. Alternatively, you can invoke the Setup command to display the most important current operational parameters.

After some practice you'll notice how quickly commands can be completed since most involve three or fewer keystrokes to get to the final menu level. For example, suppose the Acquisition Bandwidth command is invoked to change the measurement bandwidth to 10 kHz using a Chebyshev antialiasing filter. The keystrokes are:

```
ab10 <Enter> c
```

Typing "ab10<Enter>" Enter sets the instrument bandwidth and typing "c" selects the Chebyshev filter type.

### **3.7 Mouse Support**

*MLSSA* can make use of a mouse to move the graphics cursors and select or mark files from directory lists. If a mouse is detected on start-up, *MLSSA* displays the message "Mouse Detected". Note that when running under DOS a mouse driver TSR such as *MOUSE.COM* must be present in memory for the mouse to function. To use the mouse to move the graphics cursor first press the left mouse button. Pressing the left mouse button again disables mouse control which prevents inadvertent cursor movements if, for example, the mouse sits on a sloping desk and will move on its own when unattended. Mouse control of the graphics cursor is enabled/disabled separately for the time domain, frequency domain, waterfall, Nyquist and Bode screen graphics cursors. In all cases you enable/disable mouse control using the left mouse button. In the time and frequency domains, the right mouse button (when mouse control is enabled) sets the marker to the current cursor position, which is equivalent to pressing the F9 function key.

During commands that prompt you to enter a filename (for example Transfer Load), you can press the right mouse button instead of the Right arrow key to display a list of available files. You can then move the mouse to select the file you want and press the left mouse button to load that file. During commands that allow you to select multiple filenames, for example the Library Operations Frequency-files Average command, the left mouse button marks/unmarks files while the right mouse button ends the selection process (equivalent to pressing the Ctrl-Enter key combination).

Mouse control of the graphics cursors as well as for file selection is always disabled during macro record and macro playback. This means you cannot record mouse movements in a macro and any actual mouse movements during macro execution have no effect.

### 3.8 Setup Files

*MLSSA* allows you to control many parameters. To avoid programming *MLSSA* in minute detail, you normally use setup files to store all the details. Several standard setup files are provided (see Applications section) for the most common types of measurements. You can modify these standard setup files or, create your own very different setup files for special applications.

The Setup Load command loads a setup file and reprograms the entire operational state of *MLSSA* according to its contents. The Setup Save command reverses this by saving the entire current operational state of *MLSSA* to a setup file.

The Setup Comment Display command displays the comment, if any, stored in the last setup that was loaded. The Setup Comment Enter command allows you to enter a comment, which will automatically be written to the next setup file you save through the Setup Save command.

*MLSSA* setup files have the default filename extension ".SET".

During initialization, *MLSSA* looks in the *MLSSA* program directory for the special setup file NORMAL.SET and automatically loads this setup file if it exists. By saving your most commonly used setup conditions to NORMAL.SET, *MLSSA* will automatically load them every time you start the program without specifying any setup file.

When *MLSSA* is started, an alternate setup file can be specified on the command line. For example, the DOS command line,

```
MLSSA LOUDS
```

causes *MLSSA* to load the setup file LOUDS.SET during initialization. If no setup filename is specified at startup, *MLSSA* defaults to the setup file NORMAL.SET.

When a setup file is loaded either from the command line or through the Setup Load command, the entire operational state of *MLSSA* is determined by the contents of that setup file. Of course, you are free to change any individual *MLSSA* setup parameter manually. In that case, the operational state of *MLSSA* will no longer agree with the contents of the original setup file.

The Setup Reload command reloads the last setup file you had previously loaded without having to re-type its name. This command is handy when you have manually changed several setup parameters, such as the acquisition bandwidth or the FFT size for example, and wish to restore the default values.

The Setup display screen shows the most important, but not all, of the *MLSSA* setup parameters. These are the currently active setup parameters and may not agree with the contents of the last setup file you loaded if you had manually changed any parameters since loading it. The Setup Reload command will restore the original setup parameters however.

If, after manually changing one or more setup parameters, you wish to create a new setup file to store them for use at a later time, execute the Setup Save command and enter a new setup filename up to eight characters long. The then current *MLSSA* setup parameters and the setup comment, if any, will be written to this new setup file which will be regarded by *MLSSA* as the "last setup file" you loaded as shown in the lower left-hand corner of the Setup display screen. This is how setup files for different applications are created. You can create an unlimited number of different setup files in this way.

Note that setup files store the complete configuration of *MLSSA* except for screen colors, printer configuration, automatic filename data and microphone calibration data. These data are not stored in setup files because they are unlikely to change from one application to the next.

### 3.9 Time Domain

*MLSSA* operates in either the time domain or the frequency domain. In the time domain, you are presented with a graph whose horizontal axis is time. In this domain you can measure the impulse response, step response, energy-time curve, reverberant decay curve or simply acquire and display unprocessed data — just like an oscilloscope.

The Acquisition Mode command determines whether any pre-processing is performed prior to displaying the data when you execute the Go Once command to make a measurement. In the Scope mode, *MLSSA* operates as a digital scope acquiring and displaying whatever signal is connected to the input. In Cross-correlation mode, in contrast, *MLSSA* cross-correlates acquired data with the currently selected maximum-length sequence (MLS) stimulus before displaying the result. The cross-correlation operation yields the system's impulse response. See reference [1] and the Theory Notes section of this manual for more information regarding MLS methods. The Asynchronous-correlation mode, like the Cross-correlation mode, also computes the system impulse response but without the need for data acquisition that is synchronous with the MLS stimulus. With this mode, you can make use of digitally recorded stimulus or measurement data. See the Applications section titled "Remote Measurements Using DAT and CD Recorders" for details.

The Acquisition Equalization Pink-filter command enables a high-speed digital filter that further pre-processes the measured impulse response. This digital filter equalizes the measured impulse response to correct for the effects of pink-filtered MLS output on J1 of *MLSSA 2000* cards. You need to enable this filter when making MLS measurements using the pink-filtered MLS output on J1. When performing MLS measurements using the normal or white-MLS output on J3, you need to disable this digital filter by executing the Acquisition Equalization Disable command. Note that if you intend to always use a pink-filtered MLS for certain measurements you can execute the Setup Save command and save the Pink-filter equalization state to an existing or new setup file. Thereafter, every time you load that setup file the Pink-filter equalization will automatically be enabled. If you later decide to switch to the white-MLS stimulus output on J3 you need to execute the Acquisition Equalization command and select the Disable option.

Acquiring raw data or, measuring the impulse response is just the starting point for further time-domain analysis as well as for frequency-domain analysis. The View commands allow you to post-process the data to yield other time-domain curves. In addition, a built-in digital bandpass filter provides filtered versions for all derived curves. For example, from a wideband measurement of the room impulse response, you can derive Schroeder's reverberant decay curve (View Schroeder-plot Filtered command) for any octave band. The bandpass filter's center frequency and octave width are changed by the View Filter Center and View Filter Width commands respectively. The filter itself is a 6th-order IIR Butterworth type that meets IEC-225 specifications.

The View More Step-response Unfiltered command digitally integrates the measured impulse response to yield the step response.

The View Energy-time-curve commands compute various forms of Heyser's energy-time-curve from the impulse response.

Time domain data is saved and loaded from disk in binary form using the Transfer Save and Transfer Load commands respectively. *MLSSA* time-domain data files have the default filename extension ".TIM". The Info command displays important information associated with the currently displayed time-domain measurement or data file when one is loaded.

The Transfer Comment Enter command allows you to enter a plain text comment and the Transfer Comment Display command will display that comment in the message window. The current comment, if any, is also displayed in the Info screen. The Transfer Save command will prompt you to enter a comment to be stored in the TIM data file. You can either enter the file comment directly or press TAB to copy over the previously entered comment, if any, and store it in the saved data file. The Transfer Load command will overwrite the existing comment with the comment stored in the newly loaded data file.

The displayed time-axis units are fully programmable through the Units command. This feature allows you to display time in units of "feet" or "meters" of propagation distance based on the speed of sound.

The Go Repeat command is used to make repetitive measurements, as fast as *MLSSA* and your computer will allow. If you turn on overplot (Ctrl-F4) before executing this command, *MLSSA* will plot a family of curves to show drift or other long-term changes in the measurements.

The Go Average command also repeats measurements as fast as possible but instead of displaying each measurement it averages them together to reduce noise. The result is graphed only after the number of measurements you specified have been made. The main application of the Go Average command is to reduce noise in very noisy situations and only when pre-averaging is insufficient. It can also be combined with pre-averaging. See section 3.17 below titled "Improving SNR through Averaging and Pre-emphasis" for more information about averaging and pre-averaging.

### 3.10 FFT Modes

The FFT Execute command performs certain processing of the time-domain data to yield a frequency domain result. *MLSSA* enters the frequency domain whenever this command is executed. The particular processing performed depends upon the FFT mode. There are eleven FFT modes in all: Normal, Transfer-Function, Sensitivity, Voltage-Sensitivity, Impedance, Power-Spectrum, CMTF, PSD, Coherence, Incoherency and Response-Function. The FFT mode is selected by the FFT Mode command. A summary of the meaning of each FFT mode follows.

1) Normal mode. A single windowed (as programmed by the FFT Window command) FFT of the time-domain data falling between the marker and cursor is computed. The result is corrected for any window function processing losses and displayed in the same units as the time domain data from which it was derived. This mode is intended for general purposes when in the Scope acquisition mode. It is not intended for analyzing impulse response measurements.

2) Transfer-Function mode. Derives the system transfer function from the impulse response segment framed by the marker and cursor and uses the window function currently selected by the FFT Window command. The units of the transfer function are expressed as the ratio of the acquisition units to the stimulus units. Therefore,

normally, the transfer function will be displayed in units of dB volts/volts. The system transfer function can be interpreted simply as system gain vs. frequency.

3) Sensitivity mode. Similar to Transfer-Function except the result is scaled to yield the sound pressure level (dB-SPL) at 1 meter for a 1-watt sinusoid input at each frequency. The FFT Mode Sensitivity command requests that you enter the reference load impedance in ohms (usually 8 ohms for 2.83 volts RMS effective input) and the actual measuring distance in meters. If the actual measuring distance is not 1 meter, *MLSSA* will refer the sensitivity curve to 1 meter using the inverse square law. The Sensitivity mode phase curve is identical to the Transfer-Function mode phase curve.

4) Voltage-Sensitivity mode. Similar to Transfer-Function except the result is scaled to yield the sound pressure level (dB-SPL) at the current microphone position for a sinusoid input of the specified RMS reference voltage at each frequency. This mode is intended for adjusting surround-sound system levels. The FFT Mode Voltage-Sensitivity command requests that you enter a reference input voltage in millivolts (usually 200 millivolts for most surround-sound applications). The Voltage-Sensitivity mode phase curve is identical to the Transfer-Function mode phase curve.

5) Impedance mode. *MLSSA* measures impedance indirectly by first measuring the transfer function of a voltage divider formed by a known series resistance and the unknown load impedance. You enter the series resistance in ohms when entering this FFT mode and *MLSSA* will correctly derive the complex load impedance from the measured transfer function.

6) Power Spectrum mode. The power spectrum of all the current time-domain data — regardless of the marker and cursor positions — is computed using an optimal Blackman-Harris window, multiple FFTs and 50 % overlap processing. Units are acquisition units. The power spectrum is used to analyze random signals and noise and simply expresses the total signal power falling in each FFT bin. This mode should not be applied to an impulse response measurement. It is intended for analyzing data acquired in Scope acquisition mode only.

7) CMTF mode. Computes the Complex Modulation Transfer Function (CMTF) from all the measured impulse response data regardless of the current marker and cursor positions. To compute the CMTF for a specific band first filter the impulse response through the View Data Filtered command prior to FFT Execute.

8) PSD (Power Spectral Density) mode. Like Power Spectrum mode except the result is normalized to a 1 Hz effective bandwidth. Units are acquisition units per  $\sqrt{\text{Hz}}$ .

9) Coherence mode. Computes the coherence function for the transfer function that would have been derived if one were in Transfer-Function mode. This mode is only valid for MLS-derived impulse response measurements that are relatively free from reverberation.

10) Incoherency mode. Similar to Coherence mode. Incoherency equals  $\sqrt{(1 - \text{coherence})}$ . Used for MLS-derived distortion measurements.

11) Response-Function mode. Similar to Power Spectrum except that the result is normalized by the current stimulus amplitude. Units are the ratio of acquisition units to stimulus units.

### 3.11 FFT Windows

As mentioned in the previous section, you can program various time windows through the FFT Window commands in the time domain. *MLSSA* supports two basic types of time windows: Full windows and Half windows. A Full window tapers the framed time data (as selected by the current marker and cursor positions) at both ends with its peak falling exactly in the middle of the framed data. Thus, Full windows follow the normal or classical window shape from beginning to end. A Half window, in contrast, always peaks on the left-hand edge of the framed time data and only tapers data when moving toward the right-hand edge. In essence, only the right half of the full window shape is used. Half windows are normally used to window the impulse response. However, if one wants to window a room reflection surrounded by clutter, for example, a Full window is usually called for to prevent spectral leakage.

*MLSSA* offers the rectangular window plus four non-rectangular or data-tapering window shapes. These are the Cosine window also often called the Hann or Hanning window, the Hamming window, the Blackman-Harris window and the Kaiser-Bessel window. The Hamming and Hann windows are both classical windows while the Blackman-Harris and Kaiser-Bessel windows are both modern optimal windows that exhibit less spectral leakage than any classical window. The modern windows also offer a better tradeoff between spectral leakage suppression and frequency resolution than the classical windows, that is, they give the best possible frequency resolution for a given level of leakage suppression.

The rectangular window, which yields the best possible frequency resolution, is usually ideal for windowing out room reflections in loudspeaker measurements provided the impulse response has decayed to a negligible value before the first room reflection appears. However, if the impulse response is still noticeably above or below the zero line when the first room reflection arrives, it might be better to choose a Half-Hamming window to gradually taper the tail-end of the impulse response to prevent the abrupt truncation which will result in spectral leakage errors. The disadvantage of using any data-tapering (non-rectangular) window is that its frequency resolution is worse than the rectangular window. In any case, *MLSSA* always displays the true frequency resolution in the frequency domain as a color bar in the lower left corner of the graphics area. Both the window length (as determined by the marker and cursor locations) as well as the window shape are both accounted for in the color bar display. The true frequency resolution is also displayed numerically in the information screen under "FFT Parameters" whenever the Info command is executed from the frequency domain.

The modern Blackman-Harris and Kaiser-Bessel windows are best used on non-impulse response type data such as sine waves, music and noise since in these cases spectral leakage effects are much more likely to be severe with the rectangular window and not insignificant even with the classical data-tapering windows. Because of this, the Power Spectrum and PSD FFT modes both automatically use the Full Blackman-Harris window regardless of the current FFT Window selected. The disadvantage of both of these low-leakage windows is that their frequency resolution is slightly worse than the classical windows.

That is why the Half-Cosine or Half-Hamming windows are usually better for windowing an impulse response since spectral leakage is less likely to be a problem.

The FFT Window Display command graphically displays the currently selected window in the time domain. This command is used to visually verify the current window shape and position relative to the time data but has absolutely no effect on any post-processing operation.

See reference [18] for more information about FFT windows, their uses and limitations.

### 3.12 Adaptive Window™

The Adaptive Window is used to perform in-room frequency response measurements that more closely reflect the ear's subjective perception of spectral balance as compared to standard RTA-type measurements. In assessing the spectrum of a sound source, even in a reverberant room, the ear tends to largely ignore late room reflections in the treble region while giving them some weight in the mid-range and much more weight in the bass region. Because the Adaptive Window mimics this behavior, it is superior to standard RTA-type measurements in room equalization applications.

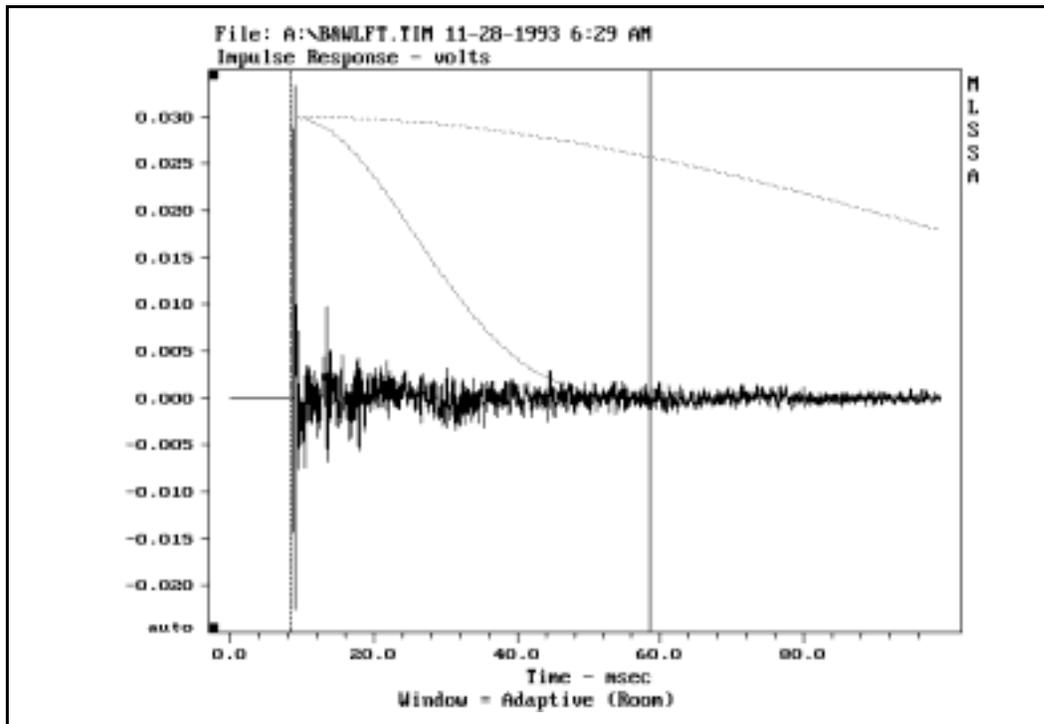
Another application of the Adaptive Window is to perform "anechoic" loudspeaker frequency response measurements that are truly anechoic in the mid-range and treble regions but which include the effects of the room in the bass region. This application uses a different Adaptive Window mode called the Speaker mode. The Speaker mode is also recommended for measuring loudspeakers in low quality anechoic chambers to minimize the effects of any residual chamber wall reflections.

The Adaptive Window essentially varies the length of a data-tapering window applied to the measured room impulse response *smoothly and continuously* with frequency between two predefined window lengths (see figure below). You normally use the marker and cursor to set the short window to include just the first 40-50 ms of the

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™ The term Adaptive Window is a trademark of DRA Laboratories

measured impulse response. This is because the ear tends to fuse together arrivals occurring within the first 50 ms treating them as a single arrival. Once set, the short window determines the high frequency resolution of the computed transfer function. The long window starts at the same position as the short window but its length always equals 1/2 the current FFT size and should normally be long enough to include nearly all the late arriving room reflections up to at least 250 ms to insure good accuracy in the bass region. You should not include reflections later than roughly 500 ms because the ear tends to perceive such very late reflections as reverberation or ambience.

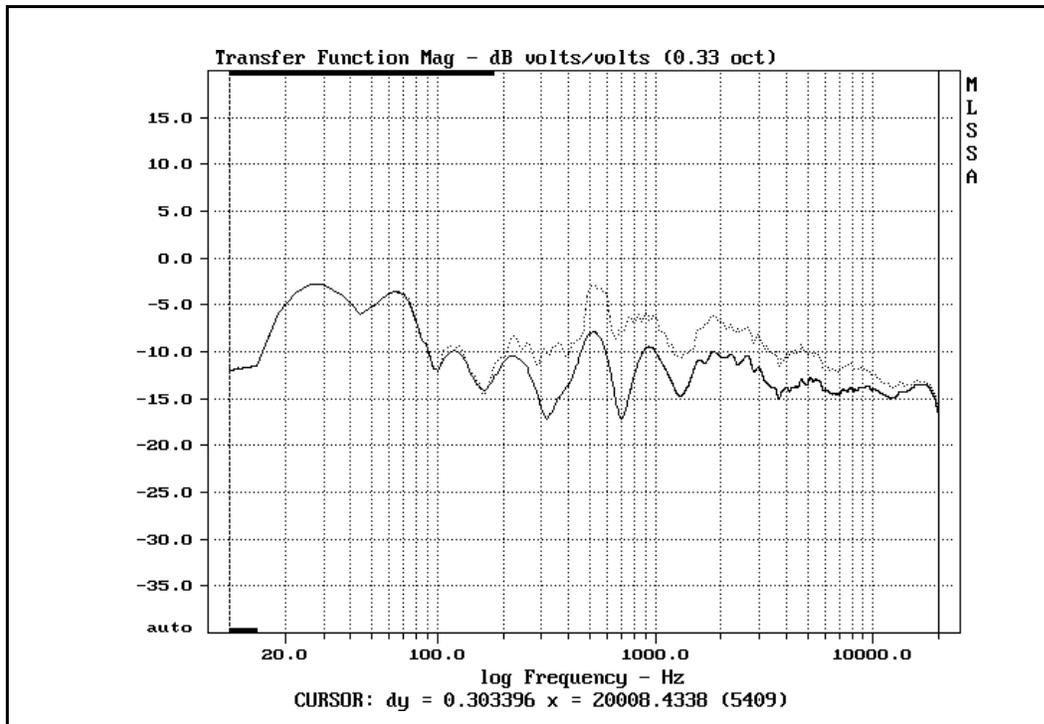


**The Adaptive Window display**

The long window determines the ultimate low frequency resolution of the measurement. In the frequency range between these two extremes, *MLSSA* automatically adjusts the window length to give a true frequency resolution corresponding to 22% of frequency or

about 1/3 octave. This algorithm was chosen because the ear's critical bands are roughly 1/3 octave wide. You display the two extreme windows in the time domain through the FFT Window Display command.

The computed adaptively-windowed transfer function (see figure below) therefore excludes all but the earliest reflections at high frequencies, gradually includes some later reflections in the mid-range region and, finally includes nearly all room reflections in the extreme bass region. It is important to note that this result is not only fundamentally different from any RTA measurement but also from a *MLSSA* measurement using a large fixed time window in conjunction with power smoothing in the frequency domain. Such a power-smoothed measurement (shown as the dotted curve) actually emulates an RTA and shares with it the same limitations of an actual RTA measurement. Power smoothing can still be applied to an adaptively windowed transfer function as shown in the figure.



**The adaptively windowed response (solid) vs. RTA-type response**

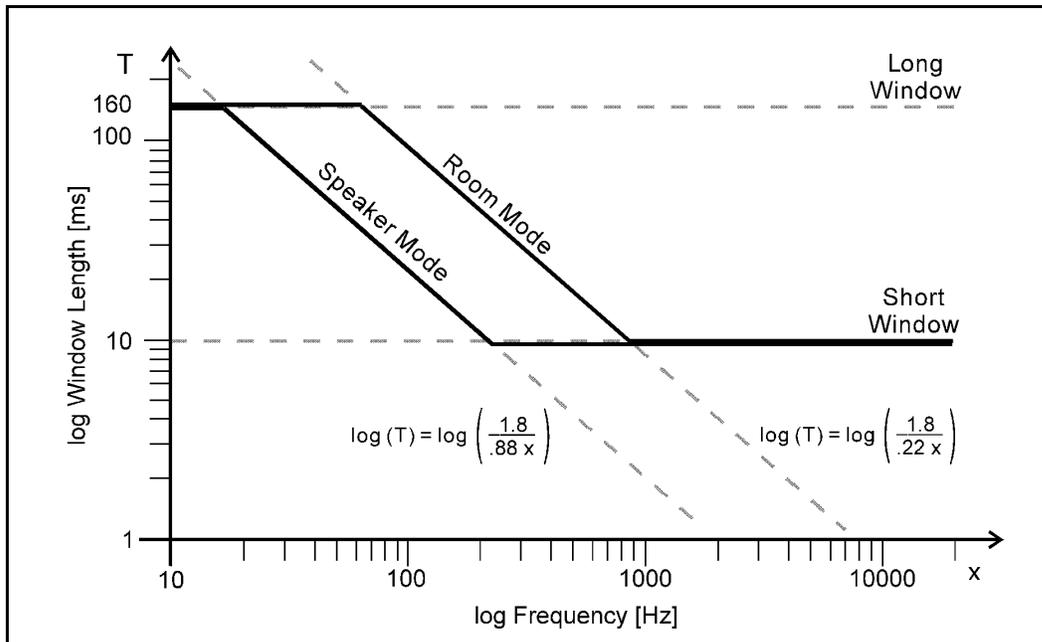
In addition, unlike an RTA-type measurement, an adaptively windowed measurement preserves phase information, which can be used by modern DSP-based equalizers to correct not only the magnitude response but also the phase response of the loudspeaker/room system. Nonetheless, the most common type of equalization in use today is minimum phase. You can export a minimum-phase version of an adaptively windowed transfer function to a standard text file through the Transfer Export Bode Minimum-phase command.

In the frequency domain, *MLSSA* displays two color bars, one at the bottom left and one at the top left of the graph. When an adaptively windowed transfer function is displayed, these two color bars will be of different lengths — the lower bar indicates the true frequency resolution in the extreme bass region while the upper bar indicates the

true frequency resolution in the extreme treble. A conventional fixed-window measurement will show color bars of equal length so that a quick glance at the graph is all that is required to see whether a particular measurement was made using a fixed window or the Adaptive Window.

There are two modes of operation of the Adaptive Window: Room mode and Speaker mode. You select the Adaptive Window and its mode of operation through the FFT Window Adaptive command. The Room mode is the one described above in which the transition region is such as to maintain a true resolution of 22% of frequency in the transition region. The Room mode is intended for typical room equalization applications.

The figure below shows how the Room and Speaker modes vary the window length with frequency. These curves assume a short window of 10 ms length set by the marker and cursor positions and, a large window of 160 ms as determined by the FFT size and the sample rate.



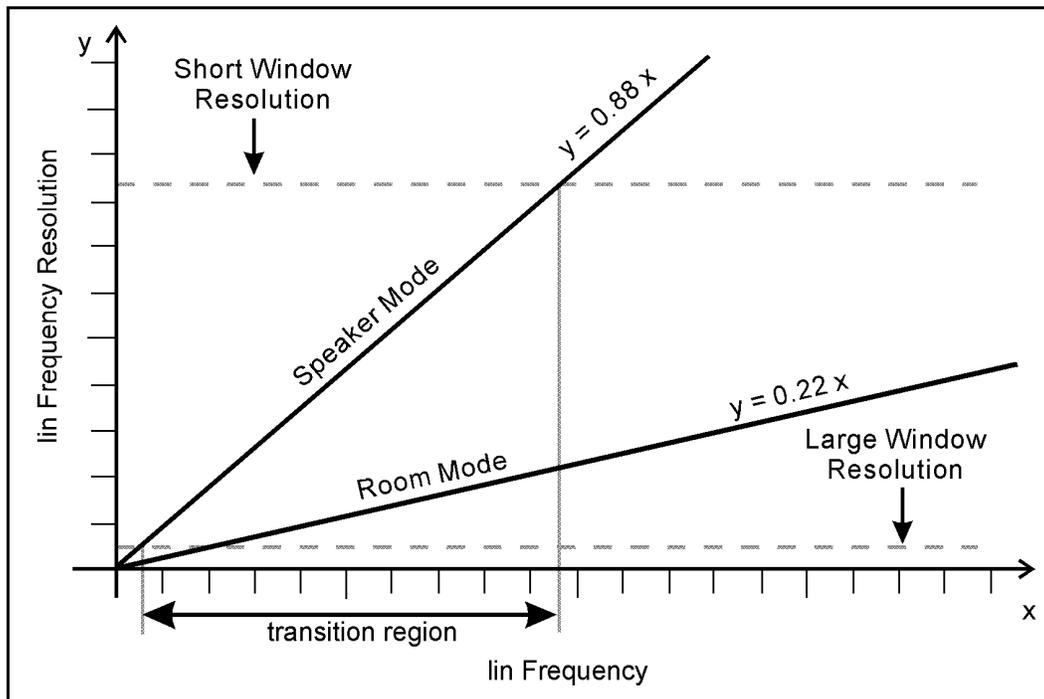
**Adaptive Window length vs. frequency**

The short window length is indicated by the lower dashed horizontal line while the long window is indicated by the upper dashed horizontal line. In this example, the Adaptive Window is fixed at 160 ms in the bass, smoothly decreases in length in the mid-range and finally becomes fixed again at 10 ms in the treble.

Note that the 10 ms short window and the 160 ms long window are used here for illustration purposes only. When making adaptively windowed in-room frequency response measurements a short window of between 40 and 50 ms is recommended. In addition, to achieve accurate bass measurements, a long window of 270 ms is recommended which implies a 32,768-point FFT when using a 20 kHz measurement bandwidth. Setup file ADAPTIVE.SET sets up these last two recommended parameters.

The Speaker mode, in contrast to Room mode, maintains a transition resolution equal to 88% of frequency which has the effect excluding room reflections until the very last

minute, so to speak, when the frequency is so low as to absolutely require a larger time window to preserve accuracy. When in the Speaker mode, the short time window is normally set to exclude all room reflections in order to ensure an anechoic transfer function in the mid-range and treble regions. In the bass region, however, where valid free-field measurements are not possible anyway without resorting to near-field techniques, *MLSSA* automatically expands the time window to include enough room reflections to maintain adequate frequency resolution. The Speaker mode can also be used for making loudspeaker measurements in an anechoic chamber. Because no anechoic chamber is perfectly anechoic, better accuracy is achieved using the Adaptive Window in Speaker mode as compared to using a large fixed window. Thus, the residual chamber reflections only influence the results in the bass but are excluded in the mid-range and treble regions.



**Adaptive Window resolution vs. frequency**

In the frequency domain, the Info command displays important data associated with particular adaptively windowed measurements. This information is listed under "FFT Parameters". First, whether the Room or Speaker mode was used followed by the transition region in Hertz. The transition region is defined as the frequency range over which the time window length is varying. Displayed below the transition region information is the true frequency resolution in Hertz for both the extreme bass region as well as the extreme treble region, that is, both below and above the transition region.

### 3.13 Frequency Domain

In the frequency domain, the true frequency resolution is graphically indicated either by not plotting the lower frequencies in which accuracy is impaired or, by moving color bars in the lower and upper left-hand corners of the graph. The color bars are normally of equal length and extend from zero frequency up to the lowest reliable frequency as determined by the time-frequency uncertainty principle. This display correctly accounts for both the time window duration as well as its shape. The mode of indicating the true frequency resolution (truncate or color bar) is selected by the Ctrl-F5 key combination. Note here that the two colors bars will be of different lengths only when the Adaptive Window was selected in the time domain. In that case, the lower color bar still indicates the true low-frequency resolution but the upper color bar now indicates the true high-frequency resolution.

Once in the frequency domain you can view the magnitude, phase or group-delay through the frequency-domain View commands. The View Bode-plot Actual-phase command displays both magnitude and phase response. The View Nyquist-plot command plots the data as real vs. imaginary. This form of data display is especially useful for impedance measurements. The View Bode-plot and View Nyquist-plot commands also plot overlays when one was displayed on the main frequency domain graph prior to executing either command.

Internally, frequency domain data consists of complex numbers. Frequency domain data is saved and loaded from disk in binary form using the Transfer Save and Transfer Load commands respectively. *MLSSA* frequency domain data files have the default filename extension ".FRQ". The Info command displays important information

associated with the currently displayed measurement or frequency-domain data file if one is loaded.

The Transfer Comment Enter command allows you to enter a plain text frequency-domain comment and the Transfer Comment Display command will display that comment in the message window. The current comment, if any, is also displayed in the Info screen. The Transfer Save command will prompt you to enter a comment to be stored in the FRQ data file. You can either enter the file comment directly or press TAB to copy over the previously entered comment, if any, and store it in the saved data file. The Transfer Load command will overwrite the existing comment with the comment stored in the newly loaded data file.

For decibel displays, the zero-dB reference level is set through the View Magnitude Reference Normal command. The View Magnitude Reference Microphone command selects an internal zero-dB reference based on microphone calibration data to display units of dB-SPL.

The View Magnitude sSmoothed commands allow you to display decibel or linear magnitude curves smoothed from 0.01 to 1.0 octave.

The View Magnitude sSmoothed command remembers the last entry and will reuse it in subsequent commands simply by pressing the Enter key when prompted for the octave smoothing desired. This eliminates the need to keep retyping numbers when executing these commands.

The View Magnitude Integrated commands display decibel magnitude integrated (summed) over standard IEC octave, 1/2-octave, 1/3-octave or 1/6-octave bands. These commands are normally used to display absolute dB-SPL of noise or sound. They should not be applied to transfer function measurements (e.g. loudspeaker or room response), as the result will show a +3 dB/octave bias error. Always use the View Magnitude sSmoothed commands on transfer function measurements.

The displayed frequency-axis units are programmable through the Units command. Units for phase angle can be selected as either degrees or radians using the View Phase uNits command.

A number of phase curves can be displayed including wrapped phase, unwrapped phase, minimum phase and excess phase. Excess phase is the difference between the actual measured phase and the minimum phase computed from the measured magnitude response. Excess group delay is the negative derivative of the excess phase and shows the excess delay through a system as compared to a truly minimum phase system. The excess phase, minimum phase and excess group delay curves are never equalized even if a reference was stored through the Reference Store command. All other curves are equalized if a reference is stored and enabled.

The accuracy level of the minimum phase, excess phase and excess group delay calculations are set by the View Phase Accuracy command. The High and Super accuracy levels require more memory and take longer to complete but are required for correct absolute phase and absolute acoustic center measurements of loudspeakers (see Applications section for details). Where only relative delays are important, for example, in checking the time coherence of loudspeaker systems, the Normal minimum-phase accuracy mode is adequate.

The View Phase Delay command lets you enter a phase delay correction to compensate for signal delays in the measured system without returning to the time domain. This command will also compute the phase delay correction when the Right arrow key is pressed and you are currently viewing an excess or unwrapped phase curve. The algorithm finds the value of phase delay that produces the flattest possible phase curve between the current marker and cursor positions. Normally, this command is used in conjunction with the excess phase curve to determine the proper phase delay correction when measuring loudspeaker absolute phase (see Applications section for details).

Group delay is the negative derivative of the unwrapped phase curve. You display group-delay through the View Group-delay Unsmoothed command. The View Group-delay Smoothed command displays group delay but smoothed from 0.01 to 1.0 octave.

The View Group-delay Excess command displays the excess group delay meaning the group delay in excess of a minimum phase system having the same frequency response as the actual system. Excess group delay is inherently smooth and so does not require further smoothing.

You return to the time domain by invoking the Exit command.

### 3.14 Frequency Domain Acquisition

You don't have to return to the time domain to make additional measurements from the frequency domain. *MLSSA* remembers the current marker and cursor positions (in the time domain), the current window type as well as the current FFT mode so that executing the Go Once command from the frequency domain automatically selects, windows and transforms the new data in the same way as when you entered frequency domain initially through the FFT Execute command.

The Go Repeat command is used to make repetitive measurements, as fast as *MLSSA* and your computer will allow. If you turn on overplot (Ctrl-F4) before executing this command, *MLSSA* will plot a family of curves to show drift or other long-term changes in the measurements.

The Go Level-repeat command is the same as the Go Repeat command but additionally the pink-weighted power average of the data falling between the marker and cursor is computed and displayed. This is the same result you would get using the Calculate Level command but is automatically repeated after each measurement. You also have the option, by pressing the TAB key, of using the Dolby standard frequency range of 500 Hz to 2 kHz instead of the marker/cursor frequency range. This command works only on curves displayed in units of dB and dB-SPL.

The Go Average command also repeats measurements as fast as possible but instead of displaying each measurement it averages them together to reduce noise. The result is graphed only after the number of measurements you specify have been taken. The main application of the Go Average command is to reduce measurement variance due to noise. The Go Average command uses complex averaging in all FFT modes except

the Power Spectrum, PSD and Response-Function modes where it uses power averaging.

The Go Spatial-average command computes the unweighted power average over the specified number of microphone locations and pauses after each measurement to allow time to reposition the microphone to a new location. This command also requests the frequency range that will be used to normalize the overall level at each location to the level at the reference location, which is assumed to be the microphone location of the first measurement. Use the mouse or cursor keys to select this frequency range or, press the TAB key to use the Dolby standard frequency range of 500 Hz to 2 kHz rather than the marker/cursor frequency range. Level normalization is necessary to prevent louder locations from dominating the results, in effect, weighting the spatially averaged frequency response in favor of the loudest location. The reference location is normally chosen to be the "sweet spot" or "hall center" location usually in the center of the intended listening area. The reference location is always the first location measured. To measure the next location, reposition the microphone and press any key or, for remote locations, connect a momentary contact switch between J4 pin-5 and J4 pin-4 (digital ground) and press this switch to commence the next measurement. This command also outputs a positive-going TTL pulse of 10 msec duration on J4 pin-2 after each measurement is completed to signal an external device such as a microphone multiplexer or turntable holding a single microphone to move to the next location.

### **3.15 Frequency Domain Calculations**

Several Calculate commands are provided in the frequency domain for special purposes.

The Calculate Distortion command determines even, odd and total harmonic distortion from the measured spectrum of sine wave stimulated measurements. Note that *MLSSA* does not provide a sine stimulus, which must come from some external source. In addition, *MLSSA* must be in Scope Acquisition Mode.

*MLSSA* locates the fundamental frequency automatically by searching for the first peak after the marker position. *MLSSA* then searches for the peak value of each individual

harmonic thus ensuring maximum accuracy. You have the option of entering the maximum harmonic number to be used in the harmonic distortion calculation. For example, entering 5 tells *MLSSA* to include all harmonics up to and including the fifth harmonic but no higher harmonics. If you enter 0, 1 or nothing followed by the Enter key *MLSSA* uses all harmonics up to current operating bandwidth and displays the maximum harmonic number in the harmonic distortion result. If you enter nothing and press the Tab key *MLSSA* uses the last maximum harmonic number you entered. For example, if you execute Calculate Distortion and enter 5 followed by the Enter key then later execute Calculate Distortion and press the Tab key *MLSSA* will calculate harmonic distortion up to the 5th harmonic. This feature is useful if you use the *DISTORT.MAC* macro to repeat the Calculate Distortion command for different fundamental frequencies since the initial maximum harmonic number you enter before starting the macro is used repeatedly. Note also that the maximum harmonic number is written out to the distortion export file if one is opened through the Calculate Export-distortion-file Open command (see below). There is a fifth column in this file that indicates the maximum harmonic used for each measurement.

As the fundamental frequency is increased, the maximum possible harmonic number decreases because it is limited by the measurement bandwidth. With the Calculate Distortion command, the maximum harmonic number actually used and displayed may be lower than the one specified when the fundamental frequency is high. For example, if you enter nothing and press Enter, all harmonics up to the measurement bandwidth are included. As the fundamental frequency is increased, the maximum harmonic number will decrease. Therefore, you may want to set the maximum harmonic number to a low value such as 5 to maintain the same maximum harmonic number over a wide frequency range. For example, if the measurement bandwidth is 20 kHz you can include the fifth harmonic for all fundamental frequencies up to 4 kHz.

If there is an erroneous peak near DC due to spectral leakage for instance, you can simply move the marker past it prior to executing this command since *MLSSA* only searches for the peak value after the marker position. The result is true harmonic distortion because any noise falling between the harmonics is ignored. If the vertical display is in decibels, *MLSSA* reports distortion in decibels also. If, however, the

vertical display is linear (selected through the View Magnitude Linear command), *MLSSA* reports distortion as a percentage.

The Calculate Export-distortion-file Open command opens a plain text file for exporting the fundamental frequency (Hz), the total, even and odd harmonic distortion in dB, as well as the maximum harmonic number, each time you execute the Calculate Distortion command. The Calculate Distortion results are always written to this file until it is closed through the Calculate Export-distortion-file Close command or when you Quit *MLSSA*. You can then simply print the contents of this file using the DOS PRINT command or use a word processor to print it. You could also import this file into a spreadsheet program to graph the data.

The Calculate Acoustic-center command determines the acoustic center of a driver based on the measured excess phase curve (selected through the View Phase Excess command). This calculation depends upon the selected frequency range as determined by the cursor and marker positions. You must first move the marker and cursor to select the frequency range of the driver to be measured. The reported distance is that measured from the microphone diaphragm to the driver's acoustic center. This distance is calculated from the measured time delay and the speed of sound in air, which can vary with air temperature and humidity. The speed of sound value used in this calculation is stored in the text file REVERB.INI located in the main *MLSSA* directory. This value for the speed of sound in this file can be changed depending upon current measurement conditions. Online calculators can be found on the Internet that compute the speed of sound based on air temperature and humidity. After obtaining the exact speed of sound for your conditions, simply edit REVERB.INI to change the speed of sound value stored there. Note that the Calculate Acoustic-center command also displays, in parentheses, the time delay in milliseconds. This time delay can be simply multiplied manually by the known speed of sound to get the distance value without editing REVERB.INI. See the Application section under "Loudspeaker Phase and Acoustic Center" for more details regarding acoustic center measurements.

The Calculate Inductance and Calculate Capacitance commands determine inductance and capacitance based upon an impedance measurement of the component in question. You use the marker and cursor to select the intended operational frequency

range of the component under test. *MLSSA* performs a least-squares fit to the imaginary part of the measured impedance over the selected frequency range. Using only the imaginary part of the impedance data to perform the calculation insures immunity to any series resistance in the test leads. Note, however, that these commands are not immune to stray capacitance or series inductance in the test leads.

The Calculate Level command computes the pink-weighted power-average of the data falling between the marker and cursor positions or, press the TAB key to use the Dolby standard frequency range of 500 Hz to 2 kHz instead of the marker/cursor frequency range. This command is normally used to compute the overall level of some measurement (Sensitivity, for example). It only works on curves scaled in units of dB or dB-SPL. Pink weighting means that the results are effectively unweighted if the same data were plotted uniformly on a logarithmic frequency scale and simply power averaged. If pink weighting were not used the result would give more emphasis to the higher frequencies in the selected band as compared to the lower frequencies since there are many more actual data points per octave at higher frequencies as compared to the low frequencies within the selected band.

The Calculate Weighted-level command calculates the A-, B- and C-weighted average energy level over the entire audio band to 20 kHz or to the selected measurement bandwidth, if less than 20 kHz. This command can be applied to any curve displayed in decibels. Its main purpose is to compute B-weighted sensitivity from a calibrated sensitivity vs. frequency curve. Like the Calculate Level command, this command compensates for the linear frequency spacing of the data points. Unlike the Calculate Level command, which is applied over a selected frequency range, the Calculate Weighted-level command is always applied over the entire audio band. The Calculate Weighted-level Flat command computes the unweighted average level over the entire audio band.

The Calculate SPL commands are covered below under the section "Absolute SPL".

### 3.16 Frequency Domain References

In the frequency domain, the Reference commands are used to create a baseline to equalize further measurements. This feature is required for accurate impedance, absolute phase and absolute sensitivity measurements and can optionally be used to correct for minor antialiasing filter errors in other measurements. To create an exact antialiasing filter reference, measure the transfer function of the antialiasing filter through a direct loopback connection between input and output and store this transfer function using the Reference Store command in the frequency domain. Alternatively, use the Reference Go command to measure and store the reference in one command from the frequency domain. This command allows you to make reference measurements without returning to the time domain. After the reference is measured and stored, subsequent measurements will automatically be corrected by *MLSSA* for any deviation from flat response or any phase errors of all the devices included in the reference measurement.

In cases where making a loopback connection is not convenient, you can execute the Reference Theoretical command instead. This command computes the theoretical *MLSSA* antialiasing filter response as well as the theoretical *MLSSA* output filter response and stores the combined response as the reference. This method is only approximate, however, as the real filters do not necessarily follow exactly their theoretical responses due, for example, to component tolerances. In addition, the theoretical reference cannot account for frequency response errors in other devices in the measuring chain such as power amplifiers.

You can save any transfer function measurement to a file using the Transfer Save command and load it later as a reference through the Reference Load command. The Reference iMport command allows you to import reference data from a text or binary (FRQ) file. See the section below titled "Importing and Exporting Data" for details.

The Reference Info command shows setup information associated with the present reference data. The Reference Reference command allows you to scale the reference by some constant (the default value is 1.0). The Reference Free command clears the reference data and frees up any memory used to store it.

The Reference Equalize-mode commands control how the reference is applied or whether it is applied at all. The Reference Equalize-mode Disable command turns off equalization but does not erase the reference data. The Reference Equalize-mode Equalize command turns on equalization by the reference if it exists in memory.

The Reference Equalize-mode Inverse-equalize command changes the way the reference data is applied to the main data. Normally, the reference is used to effectively remove or "equalize" the transfer function of some unwanted device or combination of devices in the measuring chain. This is accomplished by *complex division* of their combined transfer function, which is stored as a reference, into the main data. There are cases, however, when it is desirable to effectively insert some device or combination of devices into the measuring chain without any actual physical connection. This use of the reference feature is called inverse equalization since it does the exact opposite of equalization. Inverse equalization is performed by *complex multiplication* of the main data by the reference data.

MLSSA supports two distinct type of references intended for different purposes. The Main reference, discussed above, is typically used to correct for power amplifier and/or antialiasing filter errors in impedance or sensitivity measurements. A second reference called the Auxiliary reference is also available. You control the Auxiliary reference through the Reference Auxiliary commands. One use for the Auxiliary reference is to hold a previous measurement of a driver or loudspeaker for QC comparison purposes. Both references are normally applied by dividing them (on a complex basis) into the main data. However, since, on a decibel scale, division is equivalent of subtraction, MLSSA will display the difference in dB between the main data and the Auxiliary reference data yielding an error curve. Similarly, an equalized phase curve is actually the difference between the main data's phase curve and the reference data's phase curve. Note that minimum phase, excess phase and excess group delay curves are never equalized or inverse equalized by either reference because equalizing these particular phase curves would invalidate the minimum phase calculation (via the Hilbert transform of the log magnitude) used to generate them.

The Auxiliary reference behaves differently when it is used to store impedance measurements. Normally, the Auxiliary reference is divided into the main trace data on

a complex basis. This results in a decibel difference curve being displayed for frequency response measurements. Since decibels make little sense for impedance data, the Auxiliary reference is subtracted, on a complex basis, from the main data when both contain impedance data. This allows you to use the Auxiliary reference to store the residual impedance of your test leads, which will then automatically be subtracted, on a complex basis, from subsequent impedance measurements. This correction not only accounts for the residual DC resistance of your test leads but also their inductance permitting the use of longer test leads for impedance measurements without compromising accuracy. You simply short together the test leads, measure their impedance and execute Reference Auxiliary Store. Subsequent impedance measurements will then be corrected for the residual impedance of your test leads.

However, there are some QC applications in which dividing the Auxiliary reference into the main data makes sense for impedance measurements. You can force the Auxiliary reference to always be divided into the main data, on a complex basis, even for impedance data. To do this first Quit *MLSSA* and edit the text file *MLSSA.INI*, located in the main *MLSSA* directory. You'll need to change the sign of the entry for the "ISA bus recovery time" from a plus to minus. After saving this file with the change, you'll next need to re-start *MLSSA* for the change to take effect. With this change in effect you can no longer use the Auxiliary reference to correct for the residual impedance of your test leads.

If *MLSSA* has equalized data according to the Main reference, this is indicated by "(eq)" being appended to the title above the graphic display. If Inverse-equalize mode was selected then "(/eq)" is displayed. If *MLSSA* has equalized the data according to the Auxiliary reference, this is indicated by "(eq:aux)" being appended to the title. If Inverse-equalize mode was selected then "(/eq:aux)" is displayed. Note that measurements can be equalized or inverse-equalized according to both references simultaneously. For example, if the Main reference is in equalize mode and the Auxiliary reference is in inverse-equalize mode both "(eq)" and "(/eq:aux)" will be appended to the title.

Executing Transfer Save when either or both references are being applied as indicated by one of these symbols appearing, *MLSSA* will give you the option to either apply the

reference before saving or not. If you answer "yes" to equalize or inverse-equalize the data, the reference is permanently combined with the measurement data and saved to the specified FRQ file. To indicate that equalization or inverse equalization is now an integral part of the specified FRQ file as well as the main FFT data, "(equalized)" is appended to the FRQ filename, which now appears at the very top of the *MLSSA* screen. This indicates that the FRQ file data has already been equalized or inverse-equalized and *MLSSA*, recognizing this, will not equalize the data a second time when this could lead to erroneous results. Note that if you answer "no" in order not to equalize or inverse-equalize the data before saving then the data in the resulting FRQ file will not be the same as the displayed curve, which continues to be equalized by the reference data. Also, if you subsequently load an FRQ file that was saved as equalized or inverse-equalized "(equalized)" will be appended to the FRQ filename just as when the file was originally saved.

There are key functional differences between the Main and Auxiliary reference: The Main reference must always be a Transfer-Function and is never applied to FRQ files that were previously saved equalized or inverse-equalized except when Inverse-equalize mode is active which allows the possibility to undo a previous equalization of an FRQ data file. In addition, the Main reference must be used for the Impedance FFT mode for correction of any power amplifier or antialiasing filter errors.

The Auxiliary reference is a bit more flexible. It can hold any type of measurement data including impedance data. When in Equalize mode the Auxiliary reference is applied to previously equalized or inverse-equalized FRQ files containing data having the same FFT Mode as the reference data. This means you can use the Auxiliary reference to compare two equalized measurements, for example, for QC purposes provided they both were made using the same FFT Mode.

The Reference Auxiliary Equalize-mode Inverse-equalize command turns on inverse equalization by the Auxiliary reference. An important use of inverse equalization is the ability to simulate inserting an equalizer into a sound system in order to adjust it for the desired overall frequency response without the need to physically connect it up to the system being corrected. Suppose, for example, there is stored on disk an FRQ file holding an uncorrected loudspeaker/room Voltage-Sensitivity curve previously

measured. You simply load this FRQ file as the Auxiliary reference and select the Inverse-equalize mode. Then when you measure the equalizer's transfer function *MLSSA* will display the loudspeaker/room response with the equalizer's response included just as the loudspeaker/room response would measure were the equalizer actually inserted into the sound system. A further refinement of the Auxiliary reference when in Inverse-equalize mode is that the FFT Mode and units of the current Auxiliary reference are "passed through" and replace the units that would normally be in effect but only when *MLSSA* is operating in the Transfer Function FFT mode. A transfer function is gain vs. frequency and since gain is a unitless quantity, it makes sense to display the FFT Mode and units of the Auxiliary reference data, which in this example are Voltage-Sensitivity mode displayed in units of dB-SPL/200mV.

The Reference Auxiliary Equalize-mode Disable command turns off equalization or inverse equalization by the Auxiliary reference.

### **3.17 Controlling Signal Acquisition**

The Acquisition Bandwidth command sets the instrument measurement bandwidth, the antialiasing filter characteristic and the sampling rate to standard values. You can also change the filter bandwidth, filter characteristic and sampling rate independently to non-standard values through the Acquisition Filter Bandwidth, Acquisition Filter Select and Acquisition Clock Internal commands respectively.

*MLSSA 2000* cards are able to sample at exactly 96 kHz. To use this feature execute the Acquisition Bandwidth command. Note the two selections named \*24 and \*32. The '\*' indicates that these two bandwidths use a 96 kHz sampling rate. The 24 kHz bandwidth defaults to a Butterworth filter but you can also choose the Thomson (Bessel) filter. Only the Chebyshev filter is used for the 32 kHz bandwidth sampling at 96 kHz. All other bandwidth selections use exactly the same sampling rates as older *MLSSA* cards.

The Acquisition Clock External command allows you to select an external clock source for special applications. The external clock must be a negative-going TTL pulse applied to J4 pin-3. Pulse width must lie between 250 nanoseconds and 1 microsecond for

proper operation. Connect the external clock signal after the external clock option is selected in software to prevent a conflict between the internal and external clock sources. See Appendix E for important information regarding the use of external digital signals with *MLSSA*.

The Acquisition Filter Bandwidth command sets the cut-off frequency of the programmable 8-pole antialiasing filter. The Acquisition Filter Gain command sets the filter's passband gain in 1 dB steps. The Acquisition Filter Select command allows you to choose between a Butterworth, Chebyshev or Thomson (Bessel) antialiasing filter roll-off characteristic. The current filter characteristic, bandwidth and gain are displayed in the Setup screen and the filter characteristic, bandwidth and gain used for a particular measurement is displayed on the Info screens.

Normally, *MLSSA* performs autoranging during signal acquisitions to maximize the dynamic range of each measurement, that is, to utilize as many A/D converter bits as possible. Autoranging means that the gain of the antialiasing filter is automatically adjusted during signal acquisition in 1 dB steps to obtain the best dynamic range. There are three autorange modes: Normal, Loose and Tight as selected through the Acquisition Autorange command. The Loose mode will adjust the filter gain up only if the dynamic range would otherwise be less than 25% meaning that less than 25% of the A/D converter's full scale range would otherwise have been used. In the Normal autorange mode, filter gain is adjusted to insure a dynamic range of at least 50% and in the Tight mode at least 75%.

The Tight autorange mode is obviously preferable to the others except that it can result in infinite cycling of the filter gain if the input signal level changes too much during signal acquisitions. If cycling occurs, try switching to the Normal or Loose mode. If cycling still occurs you may be forced to disable autoranging entirely through the Acquisition Autorange Disable command and then set the filter gain manually through the Acquisition Filter Gain command. The Acquisition Filter Gain command allows you to manually set filter gain in 1 dB steps anywhere from -14 dB to +54 dB.

Note that the actual dynamic range of any measurement can be viewed by executing the Info command. The dynamic range is displayed in the lower left-hand corner of the resulting information screen.

If you intend to measure unpredictable signals, such as music or certain kinds of noise, use the Loose mode or disable autoranging through the Acquisition Autorange Disable command and set the filter gain manually through the Acquisition Filter Gain command. This is advisable when trying to measure the power spectrum of music for example. When autoranging is disabled and the filter gain is set too high an "A/D saturation" error will occur. In such cases reduce the filter gain manually and try again until a valid acquisition sequence is completed.

Thus even though *MLSSA*'s A/D converter resolution is only 12 bits, the autorange feature insures that most of these bits are being used at all times. Note the fundamental difference between *MLSSA*, which is a measurement system that generates its own stimulus and standard 16-bit digital audio recording. Digital audio requires 16-bit resolution because music exhibits a very large dynamic range and is unpredictable prohibiting the use of autoranging. Thus, most of the time, far less than 16 bits of resolution are used during digital audio recording.

The Acquisition Mode command specifies what, if any, preprocessing will be applied to acquired data before displaying it. The Scope acquisition mode merely scales the acquired data to appropriate units prior to display. The Cross-correlation acquisition mode cross-correlates the acquired data with the currently selected MLS stimulus to obtain the impulse response. The Asynchronous-correlation acquisition mode is similar to Cross-correlation but does not require exact data acquisition synchronization with the MLS stimulus. This mode is most often used to make remote MLS measurements with a DAT recorder.

The Acquisition Equalization Pink-filter command enables a high-speed digital filter that further pre-processes the measured impulse response. This digital filter equalizes the measured impulse response to correct for the effects of pink-filtered MLS output on J1 of *MLSSA 2000* cards. You need to enable this filter when making MLS measurements using the pink-filtered MLS output on J1. When performing MLS measurements using

the white-MLS output on J3, you need to disable this digital filter by executing the Acquisition Equalization Disable command. Note that if you intend to always use a pink-filtered MLS for certain measurements you can execute the Setup Save command and save the Pink-filter equalization state to an existing or new setup file. Thereafter, every time you load that setup file the Pink-filter equalization will automatically be enabled or disabled. If you later decide to switch to the white-MLS stimulus output on J3 you need to execute the Acquisition Equalization command and select the Disable option.

The Acquisition Length command determines how many time-domain points are computed and displayed. When in the Scope acquisition mode *MLSSA* acquires and displays the number of points specified by this command. When in the Cross-correlation acquisition mode, however, *MLSSA* will always acquire the full MLS period but the cross-correlation algorithm will only compute and display the number of impulse response points specified in the Acquisition Length command. Lower values take less computation time.

The trigger source is selected by the Acquisition Trigger Select command. The Stimulus option uses the stimulus generator as the trigger source. The Auto option causes *MLSSA* to start acquiring data as soon as a Go command is executed. The Minus and Plus options select an external TTL compatible trigger source connected to J4 pin-6. The Minus option causes triggering on the negative going edge of the external trigger while the Plus option cause triggering to occur on the positive going edge. See Appendix E for important information regarding the use of external digital signals with *MLSSA*.

The trigger delay is set by the Acquisition Trigger Delay command. Negative trigger delays are permitted and place *MLSSA* into its pre-trigger acquisition mode. This mode is handy when you must observe data that occurs before the trigger event. For most applications, including impulse response measurements, the trigger source is set to Stimulus and the trigger delay to zero.

The Acquisition Units command programs the units used for acquired data. You may use any label up to five characters in length. This command requires that you specify

both the unit label (e.g. mV for millivolts) and the appropriate scale factor (e.g. 1000 mV/volt).

### **3.18 Improving SNR using the Pink-filtered MLS and by Averaging**

When performing in-room frequency response or acoustical parameter measurements, you should use *MLSSA*'s pink-filtered MLS output on J1 (top phono connector) instead of the white-MLS output on J3 (bottom phono connector). Note that this feature is available on *MLSSA 2000* cards only. If you have an older *MLSSA* card, you must leave J1 disconnected at all times.

When using the pink-filtered MLS output on J1, you also need to enable pink filter correction through the Acquisition Equalization Pink-filter command. This command directs the *MLSSA* software to automatically digitally filter the impulse response with the exact inverse of the pink filter response before displaying and performing further processing. The pink-filtered MLS provides low frequency pre-emphasis to the MLS, which can significantly improve the signal-to-noise ratio (SNR) in the bass region. A pink-filtered MLS is also easier to listen to in an enclosed space compared to a white-MLS and provides a more random-like amplitude distribution which provides better distortion rejection than using a white-MLS. For more detailed information see section 3.41 titled "Pink-filtered MLS Output — Characteristics and Uses".

Further SNR improvements are possible using pre-averaging and averaging. *MLSSA* will quickly average up to 16 MLS periods prior to performing a cross-correlation operation using the Acquisition Pre-average commands. Pre-averaging is used to further increase the signal-to-noise ratio (SNR) of an MLS measurement when operating in noisy environments. You can pre-average up to 16 MLS periods for a 12 dB SNR improvement. The Acquisition Pre-average Cycles command selects the number of MLS periods to be pre-averaged while the Acquisition Pre-average Mode command controls how pre-averaging will be performed. In Concurrent mode, raw data is averaged concurrently in real time with signal acquisition yielding the fastest speed. The Alternate mode, in contrast, skips every other MLS period to average the data and should be used if your computer is too slow for Concurrent mode.

Further averaging (beyond 16 MLS periods) can be performed by using the Go Average command from the time domain. This command allows you to perform an almost unlimited number of averages. The difference between the Go Average command and pre-averaging is that the former averages the impulse response, not the raw data, after the cross-correlation operation and is therefore inherently slower than pre-averaging which averages the raw data before cross-correlation in real time. By combining pre-averaging and the Go Average command it is possible to achieve very long averaging times with a minimum of computational overhead, that is, only slightly slower than real time operation. For example, by selecting 16 pre-average cycles and 10 Go Averages you can obtain the same SNR as doing 160 pre-averages with only a small additional time penalty on Pentium class computers. In addition, the Go Average command can be terminated prematurely at any time by pressing Esc. In that case, all the valid data acquired up until that time is used in the average calculation. That is, Esc does not abort the command but only shortens it — the final curve displayed is still correct based on the actual amount of valid data acquired. Note that you should not attempt to use the Go Average command in the time domain when in Asynchronous-correlation acquisition mode since effective averaging requires exact time alignment between successive measurements.

When measuring the Power Spectrum or power spectral density (PSD) of raw input data such as noise or music in the Scope acquisition mode, it is not valid to use time domain averaging or pre-averaging as described above. Instead, you can improve accuracy of such measurements by selecting a longer Acquisition Length and using the Go Average command from the frequency domain. The Go Average command in the frequency domain performs spectral averaging which is automatically performed on a power basis whenever the Power Spectrum, PSD or Response-Function FFT mode is active. In all other FFT modes, the frequency domain Go Average command will average the data on a complex basis thus preserving both magnitude and phase information. The use of power vs. complex averaging is determined automatically by software.

### 3.19 Stimulus Generator

*MLSSA* provides a maximum-length sequence (MLS), positive pulse and programmable step function as stimuli. Of the three possibilities, the currently active stimulus is chosen through the Acquisition Stimulus Select command. You should not use the Step stimulus to measure the step response of loudspeakers because better results can be obtained by post-processing the MLS-derived impulse response through the View More Step-response command in the time domain. Besides, both the Step and Pulse stimuli result in a large DC offset at the output when they are inactive which could damage your loudspeaker if your power amplifier is DC coupled.

The output amplitude of the active stimulus is changed through the Acquisition Stimulus Amplitude command or by pressing the + and - keys located on the cursor keypad of your keyboard or the normal = and - keys. To control stimulus amplitude using these keys you must first enable the stimulus by pressing the Del key. Pressing the + or = key increases the stimulus amplitude in 1 dB steps while pressing the - key decreases its amplitude in 1 dB steps. Pressing the Del key again turns off the stimulus.

The output current capability of *MLSSA 2000* cards is 100 ma for both the normal unfiltered stimulus output on J3 and the pink-filtered version output on J1. This permits you to make direct impedance measurements using the maximum *MLSSA* output voltage of 5.23 volts as set by the Acquisition Stimulus Amplitude command. Older *MLSSA* cards limit the output current to 10 ma max which limits you to 1 volt stimulus output maximum when making direct impedance measurements.

*MLSSA 2000* cards contain precision 1 watt, 75.5-ohm resistors in both the white-MLS and pink-MLS outputs. These resistors are accurate to 0.1% to insure the highest possible accuracy of *MLSSA* impedance measurements. When making direct impedance measurements using the *MLSSA 2000* card be sure to enter exactly 75 ohms as the series resistance in the FFT Mode Impedance command. Do not enter 75.5 ohms as this will result in reduced accuracy. The precision 75.5-ohm series resistors are intentionally 0.5 ohm high in order to compensate for the input impedance of the *MLSSA* antialiasing filter, which is effectively in parallel with the 75.5-ohm series resistor.

When the stimulus is activated and has a non-zero amplitude, *MLSSA* indicates this by displaying two 1/16 notes just outside the lower right corner of the graphics area. This indicator is useful when making remote measurements such as in an anechoic chamber when it is not possible to actually hear the stimulus.

The Acquisition Stimulus Mode command determines whether the stimulus is on continuously (Continuous mode) or just during an acquisition sequence (Burst mode). This command performs the same function as pressing the Del key.

The MLS stimulus is programmed through the Acquisition Stimulus Program MLS command. You can choose a 12, 14, 15 or 16th-order MLS corresponding to sequence periods of 4095, 16383, 32767 or 65535-points respectively. Long sequence periods (32767 points and higher) are recommended for audio and acoustic measurements to prevent time aliasing in reverberant environments. However, the shorter sequences can be used advantageously in dead or anechoic spaces for faster measurements but they provide less noise immunity compared to the longer sequences. Short (4095 points) sequences are normally recommended for impedance measurements because they are not affected by either room reverberation or room noise.

The Pulse stimulus repetition period is programmed through the Acquisition Stimulus Program Pulse command.

The Step stimulus high and low times are programmed through the Acquisition Stimulus Program Step command.

The Acquisition Stimulus Units command sets the stimulus output units. You may use any label up to five characters in length. This command requires that you specify both the unit label (e.g. mV for millivolts) and the appropriate scale factor (e.g. 1000 mV/volt).

### 3.20 Overlays and Overplots

*MLSSA* can store (in memory) a copy of the currently displayed data for future comparison with new data in either the time or the frequency domain. The Overlay Store command stores the currently displayed data as an overlay while the Overlay Plot command plots the overlay data over new data. The overlay data is normally plotted with dotted lines to distinguish the overlay data from the current data. You can change the overlay color through the Library Colors Main Overlay command. Entering a negative overlay color causes *MLSSA* to plot them as solid lines. The overlay data is kept in memory until it is overwritten by new data, cleared using the Overlay Free command or until you quit *MLSSA*.

Overlays are always post-processed in the same way as the main data. For example, executing View Phase Wrapped in the frequency domain will cause *MLSSA* to display the phase response of the main data. Likewise, executing Overlay Plot will cause *MLSSA* to plot the phase response of the overlay data over the phase response of the main data.

In the frequency domain, the cursor readout display (enabled/disabled by F5) shows the difference or delta between the main data and the overlay data at the current cursor position ( $dy = ??$ ) whenever an overlay is plotted on the screen. Pressing Shift-F3 erases the overlay from the screen (but not from memory) causing just the main data values to be read out in the cursor display ( $y = ??$ ). In the time domain, when an overlay is first plotted, the delta cursor display is also activated ( $dy = ??$ ) but moving the cursor causes the display to immediately revert to reading out just the main data ( $y = ??$ ) even though the overlay trace remains on the screen. To see the delta value for another point you must execute Overlay Plot after first moving the cursor to a new data point.

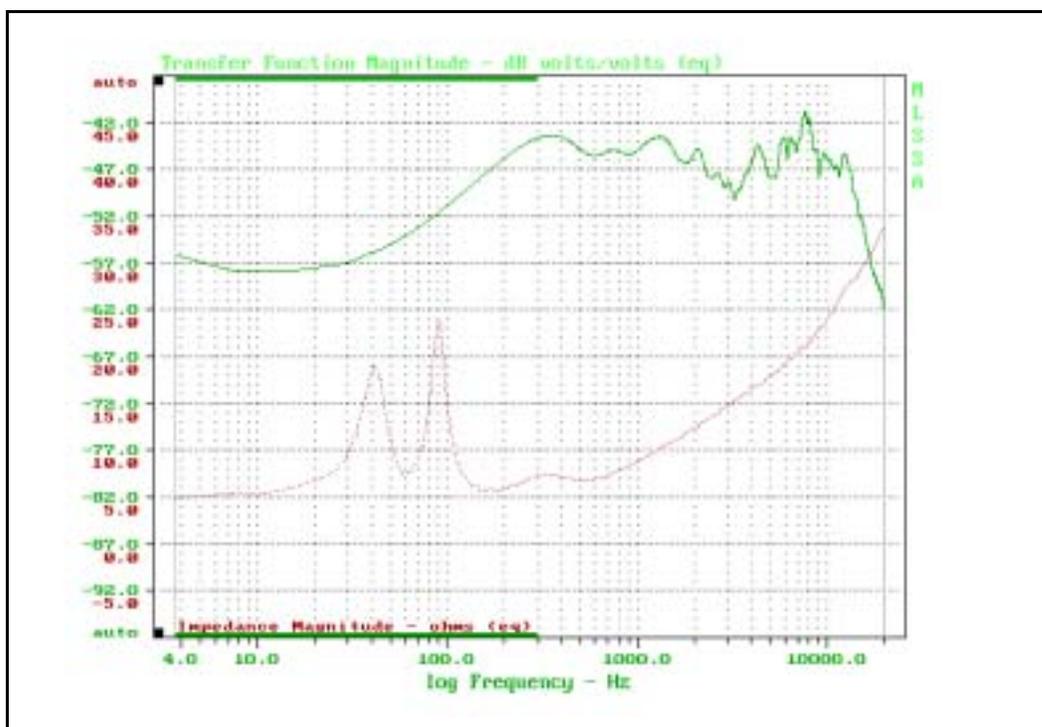
Whenever an overlay is plotted in either domain, all Alt key statistics functions are computed from the overlay data rather than the main data.

The Overlay Info command displays status information associated with the current overlay data.

You can also load and plot an overlay from a data file through the Overlay Load command. Loading a new overlay does not erase a previous overlay from the screen so you can use this command to plot multiple overlays. Note, however, that the overlay buffer in memory always contains data from the last overlay loaded.

The Overlay sAve command saves the current overlay data to a disk file. It works just like the Transfer Save command.

The vertical scale and units for overlays need not match the vertical scale and units of the main trace. The type of vertical scale for plotting overlays is controlled by the Overlay Vertical-scale command, which is available in both the time and frequency domains. There are three options: Normal, Auto and Fixed. The Normal option causes overlays to be plotted with same vertical scale as the main trace. The Auto option causes overlays to be autoscaled. In that case, the possibly different vertical scale labels for the overlay data are drawn just below the vertical scale labels for the main trace data. There are thus two vertical scales on the screen simultaneously, as shown in the figure below. A description of the overlay data is displayed in the lower-left inside corner of the graphics area. The units of the overlay data need not match the units for the main trace. The Fixed option allows you to enter your own arbitrary vertical scale for the overlay data.



### Simultaneous Display of Loudspeaker Frequency Response and Impedance

If an impedance curve is overlaid on top of a frequency response or other curve displayed in decibels, the impedance magnitude is displayed in units of ohms, not dB ohms. This makes it easy to display and print a loudspeaker's frequency response simultaneously with its impedance. An example is shown in the figure above. This feature is best used with the *MLSSA Remote Controlled Analog Interface*, which facilitates performing both loudspeaker impedance and frequency response measurements without swapping cables or changing the *MLSSA* setup.

The Overlay Erase command erases any displayed overlay from the screen but not from memory. This command is equivalent to pressing Shift-F3.

In the frequency domain, the Overlay Compare command compares the overlay data with the display data and computes mean error, +/- peak deviation from the mean and

standard deviation for points framed by the marker and cursor. You do not have to plot the overlay to use this command.

In the frequency domain, the Overlay iMport command allows you to import overlay data from a text file or from a *MLSSA* binary FRQ file having a different frequency spacing than the current main frequency domain data. See the section below titled "Importing and Exporting Data".

In addition, in the frequency domain, the Overlay Offset command allows you to plot overlays with the specified decibel offset. This feature allows you to easily compare curves that may have different average decibel levels. Note that this command only changes the overlay's vertical offset on the screen not its internal data. This command only works when data is plotted on a decibel scale.

Use the overplot feature (Ctrl-F4) when there is insufficient memory for overlay storage or, when you want to display different curves derived from the same internal data. For example, you can display the magnitude response of a transfer function, turn on overplot followed by View Magnitude sMoothed Decibels. The result will be a smoothed version of the same response plotted over the unsmoothed response. Such a combination cannot be graphed using overlays.

Overplot also permits successive measurements to be plotted on top of one another. For example, if you turn on overplot followed by Go Repeat, *MLSSA* will plot each measurement on the screen without erasing the previous measurement. This feature can be used to observe drift or other subtle changes over time.

When overplot is on and you execute Transfer Load, the new file data will be overplotted only if the file data has the same length and spacing as the original data. Otherwise, overplot is automatically turned off and only the loaded file data will be plotted.

Overplot is disabled automatically whenever you execute any command that must erase or re-scale the graph including pressing F1 for help, using the Transfer Load

directory feature or, loading a file having a different time or frequency spacing or length as compared to the current data.

### 3.21 Importing and Exporting Data

Standard *MLSSA* data files are binary files to reduce disk space requirements. You can also import text time-domain data files using the Transfer Import command and export time and frequency data to text files using the Transfer Export commands. *MLSSA* expects imported time-domain data to be arranged in a text file as follows:

Zero (0).  
Sampling interval in milliseconds.  
Total number of data points to follow.  
List of all data points.  
Title string.

**IMPORTANT:**

When creating a text file for import using a text editor or word processor be sure to add at least 1 leading space at the beginning of each line, that is, before entering any numerical data. If you are using a word processor also be sure you save the file as a plain text file, that is, without special word-processor formatting characters.

In the time domain, the Transfer Export View command exports the currently displayed data as selected by the View commands. Only those data falling between the marker and cursor positions are exported. To export a Schroeder plot, for instance, you would first execute the View Schroeder-plot Unfiltered command followed by the Transfer Export View command.

The Transfer Export Data command exports all the un-post-processed time-domain data regardless of the marker and cursor positions. The format of the exported data is the same as described above for the Transfer Import command.

There are three frequency-domain export commands called Transfer Export View, Transfer Export Bode and Transfer Export Nyquist.

The Transfer Export View command exports whatever frequency domain curve is currently displayed. Only those data falling between the marker and the cursor are exported. The text file format for the Transfer Export View command consists of two comma-delimited columns:

<frequency>, <value>

where <frequency> is the frequency in Hz and <value> is the value of the currently displayed frequency domain curve for that frequency. Note that <value> could be magnitude in dB, phase in degrees or even impedance in ohms as determined by the currently displayed frequency domain curve.

The Transfer Export Bode Actual-phase command exports both the magnitude and the actual measured phase of data falling between the marker and cursor positions. The exported magnitude and phase data are identical to those displayed by the View Bode-plot Actual-phase command. The Transfer Export Bode Minimum-phase command exports the measured magnitude and the computed minimum phase for frequencies falling between the marker and cursor positions. The exported magnitude and phase data are identical to those displayed by the View Bode-plot Minimum-phase command. The Transfer Export Bode Excess-phase command exports the measured magnitude and the computed excess phase for frequencies falling between the marker and cursor positions. The exported magnitude and phase data are identical to those displayed by the View Bode-plot Excess-phase command. The text file format for all the Transfer Export Bode commands consists of three comma-delimited columns as follows:

<frequency>, <magnitude>, <phase>

where <frequency> is the frequency in Hz, <magnitude> is the magnitude of the data in dB, except for impedance data where <magnitude> is in ohms; and where <phase> is the phase in degrees or radians depending upon the phase units programmed by the View Phase uNits command.

The Transfer Export Nyquist command exports the real and imaginary parts of the data falling between the marker and cursor positions. The exported data are identical to those displayed by the View Nyquist-plot command. The text file format for the Transfer Export Nyquist command consists of three comma-delimited columns as follows:

<frequency>, <real>, <imaginary>

where <frequency> is the frequency in Hz, <real> is the real part of the data and <imaginary> is the imaginary part. Note that microphone calibration, if any, is lost when exporting using the Transfer Export Nyquist command as *MLSSA* performs microphone calibration only when data is converted to decibels. If you need to export data that is correctly scaled and calibrated, use the Transfer Export Bode commands described above.

Each of these frequency-domain export commands allows you a choice of the exported frequency spacing. These are Linear, Octaves, Half-octaves, Third-octaves, Sixth-octaves and Custom. The Linear option exports all the data falling between the marker and cursor on a linear frequency scale exactly as it is stored by *MLSSA* internally. The other options cause the data to be decimated (i.e. some internal data points are skipped over) in such a way that the output frequency spacing is *approximately* logarithmic. For example, the Octaves option exports only those internal data points lying nearest to the IEC-standard octave frequencies. This results in a much smaller output text file but entails some loss in information. The Custom option allows you to specify any fractional octave export spacing from 0.001 to 1.0 octave.

The Custom option remembers the last value you enter and will reuse automatically simply by pressing the Enter key when prompted for the fractional octave export resolution. This eliminates the need to keep retyping numbers when executing these commands.

You can also import data into the frequency domain from either text files or binary (FRQ) files. *MLSSA*'s frequency-domain import capabilities use cubic spline interpolation to convert any frequency spacing in a given file to match that of the currently displayed frequency domain trace. The imported file's frequency spacing may

be linear, logarithmic or any other spacing as long as there are a sufficient number of data points to accurately define a smooth curve. Using too few points can result in wild swings in the spline interpolation. In text files, always be sure there are sufficient points near bends in the curve to define them unambiguously. Very few points are needed in the straight portions of the curve.

Various frequency domain commands allow you to import text files, which must be in the Bode type format described above for the Transfer Export Bode commands, this is, frequency in Hertz, magnitude in dB and, phase in degrees, all comma delimited. In addition, you can optionally include a comment enclosed inside double quotation marks as the very first line of the text file. After importing, the comment, if any, defined on the first line of the text file can be displayed using the Transfer Comment Display command.

In the frequency domain, the Transfer Import Text, Overlay iMport Text and Reference iMport Text commands import data in Bode format: ascending frequency in Hz, magnitude in dB, phase in degrees. These three columns should be delimited by commas but space delimited text files can also be imported. The first line in the text file can be any plain text enclosed in double quotes to be interpreted and imported as the comment. These commands assume that the type of data (FFT mode) in the file is the same as the currently displayed curve. Note that if the text file is to be interpreted as impedance data, *MLSSA* assumes that the magnitude is in ohms and not in dB ohms. In addition, if you omit the phase column in an imported text file, *MLSSA* assumes a zero phase angle at all frequencies.

Importing references is a little different from the other import commands. The Reference iMport Text command always assumes the file data is a transfer function type even if the main data is a different type (e.g. Impedance). The Reference Auxiliary iMport Text command offers you a choice. The selections are Default and Transfer-function. The Default option interprets the file data according to the currently displayed data's FFT mode while the Transfer-function option interprets the file data as a transfer function.

Except for the first line, any other plain text lines preceding the numerical data are ignored, whether enclosed in double quotes or not. This allows for directly importing LEAP GDT files. Note that some exported LEAP GDT files contain duplicate frequencies. *MLSSA* will refuse to import such files and will issue an error message showing the line number where the duplicate frequency lies within the file. In such cases, you must use a text editor to edit out the offending line before you can import the file successfully into *MLSSA*. Also, note that some other LEAP export files are in descending order of frequency. *MLSSA* does not import such files. *MLSSA* always presumes that the data in a text file is listed in order of ascending frequency. Any duplicate or out-of-order frequencies will result in an error and the import command will abort.

Upon importing a text file *MLSSA* will prompt you to "Enter Y to mark data as equalized". If you answer Y for "yes", then the imported data will be so marked and the string "(equalized)" will be appended to the filename title string at the top of the screen. Note that a file marked as equalized will not be equalized further by the Main reference. Press N or Esc and the imported file will not be marked as equalized.

**IMPORTANT:**

When creating a text file for import using a text editor or word processor be sure to add at least 1 leading space at the beginning of each line, that is, before entering any numerical data. If you are using a word processor also be sure you save the file as a plain text file, that is, without special word-processor formatting characters.

The Transfer Import Binary command imports data from standard *MLSSA* binary (FRQ) files. This command acts essentially as an interpolation utility allowing you to directly compare FRQ files having different internal frequency spacings. Unlike text file imports, binary imports retain the FFT mode of the data defined in the file. Therefore, importing a binary file only has the effect of interpolating it to match the frequency spacing of the currently displayed data. You can also import overlays through the Overlay iMport Binary command, references through the Reference iMport Binary and Reference Auxiliary iMport Binary commands.

Several *MLSSA* commands result in a text screen full of results including the Calculate STI commands, the Calculate Acoustics command and the Library Acoustics commands. The data contained in such text screens can be exported using the Alt-4 capture utility, which exports the text data to a text file having a default extension of ".TXT". First, press Alt-9 to switch *MLSSA* into Character mode, which displays all text screens in a consistent 25-line text video mode. By first switching to Character mode and then recalculating the STI or Acoustics screen, you can use Alt-4 to export it to a plain text file. Returning to the time domain and pressing Alt-9 again restores graphics mode. Without first switching to Character mode the Alt-4 command will capture these text screens as graphics data and save them to PCX files.

The Alt-4 utility gives you two text export options. First, answer 'y' to the option to delete all IBM line-drawing characters used to draw any table outlines and comma-delimit the output data. The default delimiting character is a comma but you can select another delimiting character by editing REVERB.INI and changing the comma to another character such as a semicolon. If the output text file already exists another option presented is: Overwrite or Append. The Overwrite option destroys any data previously stored in the text file and replaces it with the current screen text data. The Append option adds or appends the current text data to the end of the existing file without overwriting the old data but instead forming a larger output file containing the data from one or more text screens. You can also export the Setup and Info screens using the Alt-4 capture utility after first switching to Character mode through the Alt-9 command.

## 3.22 Printer Support

*MLSSA* will print the graphics screen by invoking the Printer Print Screen command or by pressing the F1 key for waterfall, Nyquist and Bode plots. In any case, you must first correctly configure both *MLSSA* and your printer. The Printer Setup Type command selects the printer language (printer codes) that will be sent to the printer. The HP-PCL option is suggested for most modern printers, most of which accept Hewlett Packard PCL (Printer Control Language) printer codes, including both laser printers and inkjet printers but you may need to first configure your printer to accept PCL codes. Refer to your printer documentation for details. Often, the configuration software supplied with the printer will allow you to select PCL as the default printer language.

The LaserJet printer type is suggested only for older HP LaserJet models and for some older non-HP printers that support PCL. The IBM, Epson and Okidata printer types are suggested only for older IBM, Epson and Okidata printer models and not for modern printers made by the same manufacturers. All modern printers should accept PCL and will usually work by selecting the HP-PCL printer type via the Printer Setup Type command and configuring your printer to accept PCL printer codes.

Normally, the printer output is sent to the parallel printer port called LPT1. You can redirect printer output to another port through the Printer Setup Port command. When printing under Windows to a shared printer on a local area network (LAN), you should select the PRN printer port, which is the generic printer device name under DOS. Note also that selecting the PRN port may slow down the printing speed slightly.

You can also print selected text portions of the screen through the Printer Print Text and Printer Print Message commands. Because only plain text is printed by these commands, they are often much faster than the Printer Print Screen command. The Printer Print Text command prints the graphics header field, title field as well as the contents of the message window (between the two horizontal lines). The Printer Print Message command prints only the contents of the message window.

Graphics screen printouts can be in either Portrait or Landscape orientation when supported by the printer as set by the Printer Setup Direction command. Printouts in

Portrait mode appear in the same orientation as the screen image while Landscape printouts are turned sideways to fill the whole page. Note, however, that some laser printers may not contain enough memory to print in Landscape orientation.

The position of the graphics screen on the printed page can be adjusted through the Printer Setup Margins command. This may be necessary on some printers to center the printout. You will have to experiment with the margin settings to get the look that suits you.

By default, *MLSSA* automatically sends a form feed to the printer after executing any print command. You can disable this automatic form feed through the Printer Setup Formfeed Disable command. This is useful for printing two graphic screens on one page. After printing the two graphics screens or one graphic screen followed by a text screen (for example, the Setup screen) you can execute a manual form feed from the printer itself or use the Printer Eject-page command.

All printer output (text and graphics) can optionally be re-directed to a printer file through the Printer File Open command. The Printer File Close command closes an open printer file and redirects printer output to the printer again. The Printer File Print command prints the contents of any previously created printer file.

Any file containing compatible printer codes can be linked to *MLSSA* graphics printouts through the Printer Link Header and Printer Link Footer commands. When linked, *MLSSA* will automatically print the header file above and/or the footer file below all graphics printouts. Header and footer files can contain either plain text or graphics created by other programs as long as the printer codes in them are compatible with your particular printer. This feature can be used to attach information (including product model numbers, company logos etc.) to all *MLSSA* graphics printouts. It can also be used to attach a status screen previously created as a printer file (e.g. Setup Status) or through the screen capture utility (Alt-4) to the graphics printouts. *MLSSA* remembers the current header and footer files even between sessions. Therefore, the only way to disable printing of the header or footer is through the Printer Unlink Header, Printer Unlink Footer or Printer Unlink Both commands.

For fastest printing speeds, such as in QC applications, use HP LaserJet or compatible laser printers. Execute the Printer Setup Type command and select the HP-PCL option. With this printer type selected, header and footer files, including plain text files, attached to the graphics screen are automatically rotated when changing from Portrait to Landscape orientation. Thus the header and footer are automatically printed in the same orientation as the graphics printout assuming there are no orientation printer control codes embedded in the header or footer files.

### 3.23 Color Printing

Because *MLSSA* is a DOS-based program, it is not possible to print directly to a Windows color printer. However, it is possible to print indirectly to the current Windows printer by first capturing the graphics screen to a PCX file, loading that PCX file into a Windows-based graphics program and printing from it. This is best accomplished by changing the way the Printer Driver Dump command works. Normally, the Dump commands are used to print through the *Pizazz 5 for DOS* (PZ5) printer driver (see next paragraph). However, all Dump commands can optionally be re-directed to invoke the screen Alt-4 capture command. The advantage over the Alt-4 screen capture command is that the Dump commands allow you to enter a comment, which will be captured as well. This feature is enabled through the Printer Driver Mode command. The Capture-to-PCX-file mode re-directs all Dump commands to the screen capture utility. The PZ5 mode directs all Dump commands to the *Pizazz 5 for DOS* printer driver (see below). Note that if *MLSSA* does not detect *Pizazz 5 for DOS* at start up, then the mode is automatically set to Capture-to-PCX-file. This mode also allows you to use the *Pizazz 5 for Windows* Edit program to print any captured *MLSSA* screen to the Windows printer. To do this, simply use the Printer Driver Dump command to capture a *MLSSA* screen with an optional comment to a PCX file. Next, from Windows, start *Pizazz 5 Edit* and click File Open to open that PCX file. Print this file using the File Print command to print to the current Windows printer. If this is a color printer be sure to answer "yes" to the option "Adjust colors for printing? (Y/N)" during the Dump sequence. To print another screen, return to *MLSSA* and Dump another *MLSSA* screen to the same PCX filename, which you can invoke simply by pressing the Up arrow key. Next, return to the *Pizazz 5 Edit* program and click on the same PCX file as

before, which will be listed in the File menu. This same PCX file, containing the new image data, will be re-loaded and can then be printed. Although more cumbersome than normal black & white printing, this method allows you to print to any printer supported by Windows, even USB printers not directly supported by *MLSSA* or by *Pizazz 5 for DOS*.

*Pizazz 5 for DOS* and *Pizazz 5 for Windows* are both bundled with *MLSSA* and the Dump commands directly support color printing and screen capture using *Pizazz 5 for DOS*. *Pizazz 5 for DOS* is a terminate and stay resident (TSR) printer driver that provides direct color printing capability and supports many more monochrome printers than *MLSSA*'s internal printer drivers. To use the Dump commands to print to a supported color printer through *Pizazz 5 for DOS*, execute the Printer Dirver Mode command selecting the PZ5 option.

Note that when you first installed *Pizazz 5* you were asked to select your printer model. If your current color printer is different, you must re-install *Pizazz 5* and select your current printer model. Place the *Pizazz 5* installation disk into drive A: and from Windows click Start then Run and type "a:setup" and then click OK. After the initial installation screen appears click OK and you will see the next screen which allows you to select your DOS printer model and the printer port it is connected to. Hopefully, your color printer model can be found and selected from this list. You must finish the re-installation process for the new color printer and printer port selections you made to take effect.

If you do not already have a shortcut to *MLSSA* on your Windows 95/98 desktop to enable printing using *Pizazz 5 for DOS* you first need to create one. To create a shortcut click Start then Find then Files-and-folders. Enter *MLSSA.EXE* in the Named-field and the full DOS path (drive and directory) into the Look-in field then click Find-now. Point to the [*MLssa*] program icon and while holding down the left mouse button drag it to the Windows desktop then release to create the shortcut. Left click the shortcut name and edit it to read "*MLSSA-PZ5*".

Right click your *MLSSA* shortcut icon then click Properties. Click the Program tab and then edit the "Cmd line:" field as you wish adding a setup file for example after the

program name. Otherwise, you should append a '?' after the program name in the "Cmd line" field so that Windows will always prompt you for command line parameters before starting *MLSSA*.

Under the "Batch file" field, enter the full DOS path to the batch file PZ5.BAT located in the *MLSSA* directory. Normally, this line would read:

**C:\MLS\PZ5.BAT**

If the directory where *MLSSA* is installed is not C:\MLS you will have to edit this line as required.

To start *MLSSA* with *Pizazz 5 for DOS* just double-click the shortcut icon. To print using *Pizazz 5 for DOS* from inside *MLSSA* you simply use the Printer Driver Dump command.

Since *Pizazz 5 for DOS* takes up around 24 KB of memory, it may be wise to create a second shortcut to *MLSSA* that does not load *Pizazz 5 for DOS*. Simply copy the shortcut and delete the "Batch-file" field entry. That way, in cases where maximum memory is more important than color printing, you can start *MLSSA* from this second shortcut.

Various *MLSSA* commands control printing using *Pizazz 5 for DOS*. The Printer Driver Colors command selects the color mapping. The Normal option hands *Pizazz 5 for DOS* the colors exactly as they appear on the screen. This is usually undesirable because *Pizazz 5 for DOS* will print a black background on white paper, which not only looks unappealing but also rapidly depletes your supply of black ink. The Adjust option corrects this problem by switching to a white background and adjusting the colors for contrast. For example, pure yellow appears with high contrast against a black background but looks washed out against a white background. Therefore, the Adjust option prints yellow as gold. The other colors are similarly mapped to achieve good contrast on white paper without totally changing your color scheme. If you are not satisfied with the color mapping you can change it by editing the printer palette in text file PALETTES.INI located in the *MLSSA* program directory. See the paragraph at the

end of this section for details. When dumping to a monochrome printer via *Pizazz 5 for DOS* it is often desirable to convert all colors to pure black-and-white to maximize contrast. You do this by selecting the Black-and-white option of the Printer Driver Colors command.

The Printer Driver Options command brings up a screen from which you can change various *Pizazz 5 for DOS* options. Change from portrait to landscape printing or, save the screen to a graphics file instead printing it. You can also choose dynamic cropping which allows you to print or capture only a portion of the screen. See the *Pizazz 5 User's Guide* for more details about the *Pizazz 5 for DOS* options.

The Printer Driver Dump command calls on *Pizazz 5 for DOS* to print or capture the screen. Use the Printer Driver Dump command to print the time domain or frequency domain screens in color using *Pizazz 5 for DOS*. Press Shift-F1 to color print other *MLSSA* screens, such as the waterfall screen, that do not include an explicit Dump command. In general, the word "Dump" as in the Setup Dump command, denotes printing using *Pizazz 5 for DOS* while the word "Print" as in the Setup Print command, denotes pure black-and-white printing using *MLSSA*'s internal printer drivers which, by the way, are very much faster than *Pizazz 5 for DOS* when printing to HP LaserJet printers.

Note that in order for these commands to work properly, the DOS environment variable *MLSSADUMP* must exist and point to the *Pizazz 5 for DOS* program *PZ5.EXE*.

If you installed *Pizazz 5 for DOS* when you installed *MLSSA*, the install program will have already set up the *MLSSADUMP* environment variable. If you installed *Pizazz 5 for DOS* separately from *MLSSA*, you must add the following line to your *AUTOEXEC.BAT* file,

```
MLSSADUMP=[drive:][dir]\PZ5.EXE
```

where *[drive:]**[dir]* is the full DOS path to the *Pizazz 5 for DOS* program directory. For example, if your *Pizazz 5 for DOS* program files reside in *E:\PZ5* then you would enter the line,

**MLSSADUMP=E:\PZ5\PZ5.EXE**

into your AUTOEXEC.BAT file and re-boot your computer.

The text file PALETTES.INI file located in the *MLSSA* program directory defines the contents of the color palette registers of the video graphics hardware. In EGA and VGA modes, there are 16 palette registers to define. Each register holds a palette color numbered from 0 (black) to 63 (bright white). The first palette register set in the file defines the 16 screen colors displayed in EGA and VGA modes. The second palette register set defines the 16 colors used when dumping EGA and VGA screens to a color printer via *Pizazz 5 for DOS* with the Adjust or Black-and-white option selected from the Printer Driver Colors command. Screen and printer palette register sets for CGA mode are defined at the end of the file. There are only two CGA palette registers, the first defines the background color and the second defines the foreground color.

### **3.24 Capturing Graphics and Text Screens**

In many cases, it is desirable to be able to capture *MLSSA* graphics screens in a form that can be imported into word processors or desktop publishing programs. *MLSSA* includes a built-in screen capture utility that can be called upon at any time by pressing Alt-4. Note that *Pizazz 5 for DOS* can also be used to capture graphics screens. See the previous section titled "Color Printing" for details.

If the current video mode is a Graphics mode, the Alt-4 command will prompt you for the name of a PCX output file. If you do not specify a file extension then ".PCX" is assumed since this is the standard extension for PCX files. (Note: To re-use the previous PCX filename that you entered for the prior capture, just press the Up arrow key.) After entering the PCX filename, there follows several options. The first option is "Adjust colors for printing? (Y/N)". Answer "yes" to adjust colors for best appearance when later printed on white paper. For word processors that can only handle monochrome images, it's best to convert the color screen image to pure black & white and to save the image in reverse video. Answer "no" to skip the color adjustment for printing and the next option "Convert to pure black & white? (Y/N)" appears. Answer "yes" and the next option "Save in reverse video? (Y/N)" appears. Answer "yes" to save

in reverse video which looks best when later printed on white paper. For programs that support full color images you can skip the color to black & white conversion option by answering "no" to the "Convert to pure black & white? (Y/N)" option. In that case, the capture utility offers another option, which will reverse black & white only. This option is recommended for capturing screens in full color to be later printed in full color from inside a desktop publishing program. Reversing black & white avoids printing an unsightly pure black background on white paper while retaining the true color information in the image. Note, however, that the exact colors are retained in this case (not adjusted for printing) meaning that yellow will appear washed out when printed on white paper.

One disadvantage of the Alt-4 screen capture command is that it captures the screen without allowing you to enter any comment. This shortcoming can be rectified by changing the way the Dump command works. Normally, the Dump commands are used to print through the *Pizazz 5 for DOS* (PZ5) printer driver. However, all Dump commands can optionally be re-directed to invoke the screen Alt-4 capture command. The advantage is that the Dump commands allow you to enter a comment, which will be captured as well. This feature is enabled through the Printer Driver Mode command. The Capture-to-PCX-file mode re-directs all Dump commands to the screen capture utility. The PZ5 mode directs all Dump commands to the Pizazz 5 printer driver. Note that if *MLSSA* does not detect *Pizazz 5 for DOS* at start up, then the mode is automatically set to Capture-to-PCX-file. This mode also allows you to use *Pizazz 5 for Windows Edit* to print any captured *MLSSA* screen to the Windows printer. To do this, simply use the Printer Driver Dump command to capture a *MLSSA* screen with an optional comment to a PCX file. Next, from Windows, start *Pizazz 5 Edit* and click File Open to open that PCX file. Print this file using the File Print command. To print another screen, return to *MLSSA* and Dump another *MLSSA* screen to the same PCX filename. Next, return to the *Pizazz 5 Edit* program and click on the same PCX file as before, which will be listed in the File menu. This same PCX file, containing the new image data, will be re-loaded and can then be printed. Although more cumbersome than normal printing, this method allows you to print to any printer supported by Windows, even USB printers not supported by *MLSSA* or by *Pizazz 5 for DOS*.

Every *MLSSA* screen, including those containing only text such as the Setup, STI, *SPO* screens, all help screens as well as the directory/file lists are all displayed in the same video graphics mode as normal *MLSSA* graphics. This is to prevent Windows from constantly switching *MLSSA* between full-screen and a window.

One disadvantage of graphics mode is that it is impossible to export a text screen, for instance, the STI screen, to a text file using the Alt-4 capture command. Pressing Alt-9 however switches *MLSSA* into Character mode, which displays all screens in a consistent 25-line text video mode. By first switching to Character mode and recalculating the STI screen, you can then press Alt-4 to capture it to a plain text file. Return to the time domain and press Alt-9 again to restore graphics mode.

In Character mode the image is not captured, only the information in plain text format. Therefore, when in Character mode, the Alt-4 command is really a text data export utility. There are two text mode options of the Alt-4 command. If you answer 'y' to the first option all IBM line-drawing characters, which are used to draw any table outlines, if present, will be deleted and the output data will be comma-delimited. The default delimiting character is a comma but you can select another delimiting character by editing REVERB.INI and changing the comma to another character such as a semicolon. If the capture file already exists another option presented is: Overwrite or Append. The Overwrite option destroys any data previously stored in the file and replaces it with the current screen text data. The Append option adds or appends the current text data to the end of the existing file without overwriting the old data but instead forming a larger output file containing the data from one or more text screens.

### **3.25 Waterfall Plots**

*MLSSA* provides four different types of waterfall plots. The word "waterfall" refers generically to any type of 3D display in which data is plotted in slices where each slice is displaced from its neighbor in the front-to-back or z direction. *MLSSA* uses its waterfall display for four different post-processing functions:

1. Cumulative Spectral Decay (CSD).
2. The Wigner distribution.

3. The Energy-Time-Frequency (ETF) or spectrogram.
4. Frequency-File-Display (FFD).

You select the post-processing function mode through the Waterfall Mode command.

The ETF mode is the simplest of the three to understand and is primarily intended for room acoustic evaluations. The CSD and Wigner modes, in contrast, are intended primarily for anechoic loudspeaker evaluations. The FFD mode offers a general-purpose means of plotting multiple frequency domain data files and finds its primary application in plotting loudspeaker polar response data.

*MLSSA* allows you to control how post-processed data is displayed through the Waterfall X-axis, Waterfall Y-axis and Waterfall Z-axis commands.

The x-axis is just the frequency axis, which is oriented horizontally. You set the displayed frequency range through the Waterfall X-axis Range command. This command is effectively a zoom function for waterfall plots. The Waterfall X-axis Scale command sets the type of frequency axis scaling. Your choices are liNear, Log or IEC-log, which are the same choices for the main display as selected by pressing F4 from the frequency domain.

The y-axis represents amplitude and is oriented in the vertical direction. The Waterfall Y-axis Decibels/div command sets the y-axis scale factor for all decibel plots. For most purposes, a scale factor of 6 dB/div is recommended. In addition, when plotting decibel data, the waterfall display forms a floor below which no data is plotted. This floor appears as the plane formed by the x and z-axis. The purpose of the waterfall floor is to aid in visual perspective and the in the interpretation of the data. The Waterfall Y-axis Floor command allows you the choice of fixing the floor level to a particular decibel value or having *MLSSA* determine the optimal floor level automatically (Auto option). Note that the Decibels/div and Floor settings have no effect on the Wigner distribution, which is always plotted on a linear vertical scale. The Wigner distribution floor level is always zero and negative values are plotted below the floor level. The Waterfall Y-axis Wigner-scale command allows you the choice of setting the Wigner vertical scale factor manually or having *MLSSA* determine the optimum scale factor automatically. Note

however, that the Auto mode sometimes fails to prevent the Wigner plot from exceeding the display limits. If this occurs you should set the Wigner scale factor manually.

The z-axis usually represents time delay and is oriented front-to-back. In the CSD, ETF and Wigner modes the z-axis always represents time and is scaled automatically by *MLSSA*. In the FFD mode, however, you must tell *MLSSA* what the z-axis represents. In this mode, *MLSSA* will plot any set of autonamed frequency domain files starting with file number specified by the Waterfall Z-axis First-file command as the first (rear-most) slice. For example, suppose a particular set of frequency files comprise loudspeaker polar response data taken at 10-degree increments starting from the on-axis position. In this case, the z-axis represents an angle measured in degrees. The Waterfall Z-axis Units command sets the particular z-axis units you wish to use ("degs" in this example) and the Waterfall Z-axis Increment command sets the change in these units between data files (10 degrees in this example). The Waterfall Z-axis Offset command sets the label value of the first (rear-most) slice (zero in this example).

The Waterfall Octave-smoothing command allows you to smooth individual waterfall slices in the ETF and FFD modes only.

The Waterfall Reference command allows you to select a particular waterfall slice to act as a reference for all the slices. This means that all waterfall slices will be plotted as differences, in dB, relative to the selected reference slice. When you activate this feature *MLSSA* will display the message "(Equalized to slice ?)" on the waterfall screen where ? is the particular slice index entered in this command. Normally, slice index 1 or the rear-most slice is chosen as the reference slice. If you want to use another slice as a reference, move the waterfall cursor to that slice and note its index number. The index of a slice is displayed in parentheses at the end of the cursor position display line. Note also that the waterfall reference has no effect in the Wigner distribution mode but does affect the CSD, ETF and FFD modes.

The Waterfall Hidden-lines Visible command shows all parts of all the waterfall slices. You can optionally show normally hidden lines in any waterfall display. With this feature enabled, hidden lines are displayed but in a different color than the visible lines. The display color is controlled via the Library Colors Waterfall Hidden-cursor command.

The Waterfall Perspective Reverse command displays the backside view of any waterfall plot. This feature is useful in the Wigner display, which can sometimes hide the information near zero delay.

The Waterfall Text-export-file command exports waterfall data to the specified text file whenever Waterfall Execute is invoked. The file is overwritten with new data every time you invoke Waterfall Execute. To stop exporting data, execute Waterfall Text-export-file again and press the Enter key to specify a null export file.

The format of the exported text file allows for easy importing into MS Excel. The exported text file includes an extra column preceding the first column of data, which labels each row of data. For example, if the exported data is loudspeaker polar response measurements, the first column will be a list of all the angles of the polar data. Similarly, an extra row is included above the first row of data, which labels each column of data. Each column always corresponds to a frequency in Hz. The waterfall data is exported in all waterfall modes including Frequency File Display (FFD) for polar measurements, Cumulative Spectral Decay (CSD), Energy Time Frequency (ETF) and Wigner distribution. Once imported into Excel, any and all of Excel's graphics and processing functions can be applied to the waterfall data.

The Waterfall Text-export-file command also prompts you for the desired frequency spacing of the exported waterfall data. Enter a fractional-octave value from 0.01 to 2.0 or enter a value of zero to export the waterfall data at full resolution and with linear frequency spacing. For more information see the section below titled "Exporting Waterfall Data to Excel — Circular Polar Plots".

### 3.25.1 Cumulative Spectral Decay Mode

The system transfer function can be displayed in three dimensions as cumulative spectral decay plot (CSD). To generate this plot, *MLSSA* computes the FFT of successively left-shifted and windowed versions of the impulse response segment selected by the marker and cursor. The first slice of the CSD plot (at zero delay) is the same as what you would get with the FFT Execute command when in the Transfer-Function FFT mode.

Successive slices are computed by shifting the windowed impulse response segment (as determined by the marker and cursor positions) left by a specified number of time-domain points, applying a special waterfall window and then computing the FFT. This process is repeated many times generating the successive slices, which form the CSD plot. You set the number of time points to shift between slices through the Waterfall Shift command and you set the total number of slices to process and display through the Waterfall Cycles command. You generate a CSD plot through the Waterfall Execute command.

The result is called a cumulative spectral decay plot because, for any specific frequency, this plot shows the decay of the system output for a pure tone input that is suddenly turned off. Thus you can easily observe loudspeaker cabinet or other resonances which persist after the excitation ceases.

Another way of interpreting a CSD plot is as a spectrogram of the tail of the selected impulse response segment. Therefore, an impulse response that decays quickly will have a CSD waterfall that also decays quickly and visa versa. The advantage of the CSD over just the impulse response or, an energy-time-curve (ETC) display of the decay is that the CSD plot shows you the frequency-decomposed decay while the ETC lumps together all frequencies.

The FFT window used is the same window selected by the FFT Window command. This is done to insure consistency between CSD plots and transfer functions. Normally, you select a rectangular FFT window.

A second special window function is formed internally by convolving a unit step function with the selected standard window function. This results in a step function with a smeared rising edge. The window function used is selected through the Waterfall Window Select command. The 10% to 90% rise time of this waterfall window is set by the Waterfall Window Rise-time command. The rise-time actually represents how fast the hypothetical excitation tone is turned off. Small rise time values improve time resolution at the expense of frequency resolution while larger values improve frequency resolution at the expense of time resolution. This is a fundamental tradeoff of CSD analysis. Typical values for the window rise time range from 0.01 to 0.35 msec.

The FFT size used for the CSD calculation is set by the Waterfall FFT-size command. The displayed frequency range is set by the Waterfall X-axis Range command and the frequency-axis scale (log or linear) is set by the Waterfall X-axis Scale command.

Due to the time-frequency uncertainty principle, low frequencies and long delay times in a CSD plot are unreliable. *MLSSA* automatically determines these unreliable regions of the CSD plot and masks them from view. The unreliable regions of the CSD plot are indicated by a floor pattern of dotted lines as distinguished from the reliable data that are plotted as solid lines.

See references [2] and [6] for more information regarding CSD plots.

### 3.25.2 Wigner Distribution Mode

The Wigner distribution is a different 3D display of the transfer function. You generate this plot by first invoking the Waterfall Mode command and choosing the Wigner-distribution option. Thereafter, the Waterfall Execute command will generate the Wigner distribution of the impulse response segment selected by the marker and cursor. Actually, *MLSSA* first forms the complex analytic signal from the impulse response before computing the Wigner distribution. This provides a more useful form that is easier to interpret. As with the CSD plot, *MLSSA* also applies the current FFT window to the selected segment as well as a special waterfall window. In the case of the Wigner distribution, the waterfall window is a complex version (since the analytic signal is complex) of the standard window function selected through the Waterfall Window Select command. Note, however, that this window is not modified by convolution with a unit step as for CSD plots.

Also, note that the Wigner distribution can go negative so it is always plotted on a linear amplitude scale with a floor level of zero energy. Normally, *MLSSA* determines the best vertical scale factor automatically but the top part of the Wigner distribution may sometimes be chopped off. If this occurs, set the Wigner distribution vertical scale factor manually through the Waterfall Y-axis Wigner-scale Set command. To return to autoscale mode execute the Waterfall Y-axis Wigner-scale Auto command. These commands affect only the Wigner distribution.

See reference [7] for more information regarding the Wigner distribution.

### 3.25.3 Energy-Time-Frequency Mode

The energy-time-frequency (ETF) or simple spectrogram is useful in room acoustic evaluations. In this mode, *MLSSA* analyzes all the impulse response data starting from the current marker position. Normally, when analyzing room impulse response measurements, you set the marker at the first arrival position. Each slice is generated by applying the selected waterfall window function and then applying the FFT. The ETF waterfall window is a full window as long as the Waterfall FFT-size that tapers the time data on either side and is centered initially at the marker position. Each successive slice is generated by shifting the window to the right by the Waterfall Shift parameter and performing a new FFT. The shift parameter should normally equal one half the waterfall FFT size. You can also smooth the individual ETF slices through the Waterfall Octave-smoothing command.

### 3.25.4 Frequency File Display Mode — Polar Plots

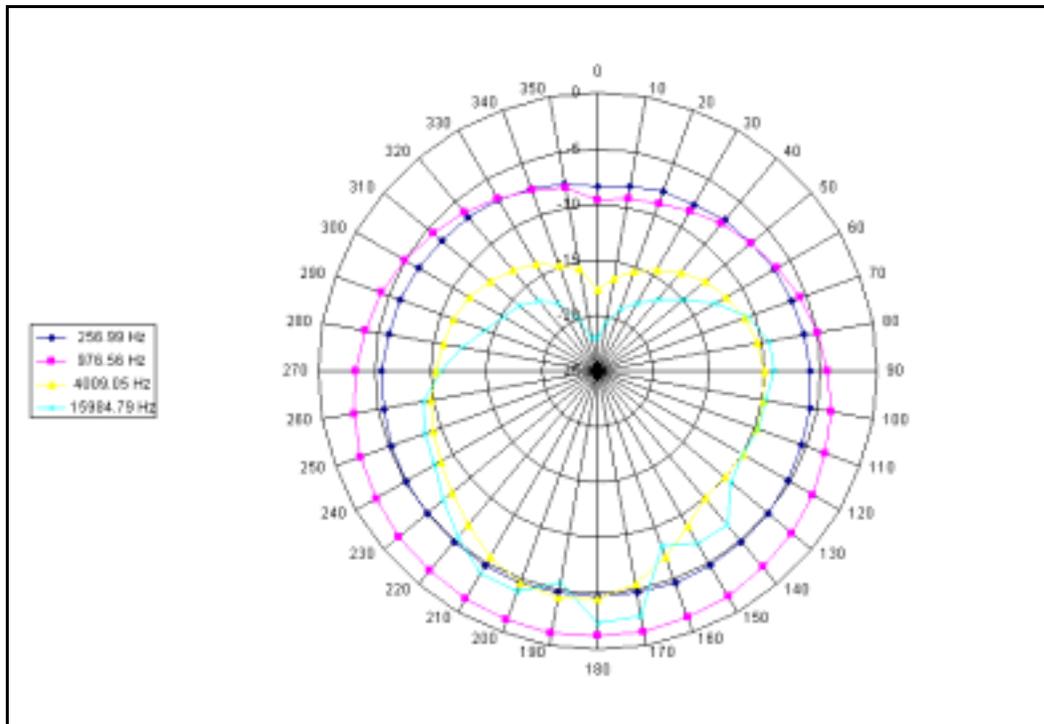
The FFD mode displays any set of autonamed (see section titled "Automatic Filename Generation" below) frequency domain data files. Files displayed are of the form <PATH>\<ROOT>\*.FRQ where <PATH> and <ROOT> are those currently defined by the Transfer Autaname Path and Transfer Autaname Root commands respectively. File <PATH>\<ROOT>-?.FRQ where ?? is the autonamed file number defined by the Waterfall Z-axis First-file command, is always plotted as the first (rear-most) slice followed by successively higher numbered autonamed files. The number of files plotted is set by the Waterfall Cycles command. Nonexistent files are represented by an empty slice (floor level) and files are always plotted on a decibel scale. The Waterfall Shift command sets the file-number increment to be used. Normally, the shift parameter is set to one (1) causing all files to be displayed contiguously. A shift value greater than one (1) will result in a noncontiguous display. For example, if the first file number is 1 and the shift parameter equals 2 then file numbers -01, -03, -05 etc. will be plotted with files -02, -04, -06 etc. being skipped.

In the FFD mode you must tell *MLSSA* what the z-axis represents since this cannot be determined from the data files. For example, suppose a particular set of data files comprise loudspeaker polar response data taken at 10-degree increments starting from the on-axis position. In this case, the z-axis represents an angle measured in degrees. Use the Waterfall Z-axis Units command to set the particular z-axis units you wish to use ("degs" in this example) and the Waterfall Z-axis Increment command to set the increment in these units between data files (10 degrees in this example). The Waterfall Z-axis Offset command sets the value of the first (rearmost) slice (zero in this example). You can also smooth the individual slices through the Waterfall Octave-smoothing command.

The main application of the FFD waterfall mode is to display loudspeaker polar response measurements. For detailed application information, see section 6.6 titled "Loudspeaker Polar Response".

### 3.25.5 Exporting Waterfall Data to Excel — Circular Polar Plots

The Waterfall Text-export-file command exports waterfall data to the specified text file whenever Waterfall Execute is invoked. The format of this text export allows for easy importing into MS Excel. The exported text file includes an extra column preceding the first column of data, which labels each row of data. For example, if the exported data is loudspeaker polar response measurements, the first column will be a list of all the angles of the polar data. Similarly, an extra row is included above the first row of data, which labels each column of data. Each column always corresponds to a frequency in Hz. Once the waterfall data is imported into Excel, it's a simple matter to use Excel's radar chart option to plot polar data in a circular fashion. The waterfall data is exported in all waterfall modes including Frequency File Display (FFD) for polar measurements, Cumulative Spectral Decay (CSD), Energy Time Frequency (ETF) and Wigner distribution. Once imported into Excel, any and all of Excel's graphics and processing functions can be applied to the waterfall data.



**Example Circular Polar Plot using Excel**

The Waterfall Text-export-file command also prompts you for the desired frequency spacing of the exported waterfall data. Enter a fractional-octave value from 0.01 to 2.0 or enter a value of zero to export the waterfall data at full resolution and with linear frequency spacing. Note however that when the Waterfall FFT-size is large in the case of CSD, ETF and Wigner waterfall plots, or when the FFT size of the displayed FRQ files is large when in the Frequency File Display waterfall mode, a huge amount of data is exported when linear frequency spacing is selected. When importing such files, Excel may not import many of the highest frequency data columns due to Excel's limit of 230 data columns maximum. By entering a small value such as 0.1 octave, *MLSSA* will export the waterfall data at a frequency spacing of 0.1-octave which greatly reduces the number of high frequency data columns thus overcoming Excel's upper limit on the number of columns. If you are only interested in graphing polar data at a handful of

frequencies, enter a large value. For example, by entering a 2.0-octave frequency spacing, *MLSSA* will export the data at frequencies spaced by two whole octaves; this is, at roughly 125 Hz, 500 Hz, 2 kHz, 8 kHz. This option is ideal for graphing polar data since all four columns can be selected at once to generate an Excel radar chart that includes all four frequencies.

From *MLSSA* invoke Waterfall Execute to generate and export the polar data to the specified text file. To import the text file, press Alt-space from *MLSSA* to switch to the Windows desktop. Start Excel and select Data then Get-External-Data followed by Import-Text-File. Enter or select the filename defined previously by the *MLSSA* Waterfall Text-export-file command and click Import. The Excel import wizard will appear. Under "Original data type", click the Delimited option and then click Next. Click the Comma button to select comma delimiting and be sure to de-select all other delimiters. Click Finish then OK. The exported text file is now in Excel including the header information. The first line of the header will be the time domain comment, which can be modified in *MLSSA* by the Transfer Comment Enter command from the time domain. If no comment was defined prior to Waterfall Execute, the phrase "MLSSA waterfall data" will be displayed on the first line. In the FFD waterfall mode, row 13 of the Excel spreadsheet displays contains the exported frequencies in Hz while column A lists the exported z-axis values in units of "degs" (degrees) of angle. The waterfall z-axis units are "msec" (milliseconds) for CSD, ETF and Wigner plots. Note that you can switch at will from *MLSSA* to Excel and back again, re-export and re-import the same file repeatedly with different waterfall settings, different waterfall data or a different time domain comment until you get the results you want.

To plot polar data in a radar chart you need to first select one or more columns of polar data. Note that if less than a full 360 degrees of polar data is available in each column, you need to select a sufficient number of blank cells below the data cells to represent a full 360 degrees. For example, if there are just 19 cells of data in each column representing 0 through 180 degrees at 10-degree intervals, you need to include 17 additional blank cells below the data cells, that is, select 36 cells per column for a total range of 360 degrees. Even though the extra cells contain no data, selecting them along with the cells that do contain the data is necessary to produce a radar style graph

with a full 360 degrees in a circle. To facilitate selecting the correct number of empty cells, return to *MLSSA*, invoke Waterfall Cycles and enter 36, assuming 10-degree resolution polar data. Next, invoke Waterfall Perspective and select the Reverse option. Waterfall Execute will now output 36 angles ranging from 0 to 350 degrees into the export file even though only 19 rows will contain polar data. Switch back to Excel and re-import the file. Note that column A now contains angles 0 thru 350 degrees. Select the column or columns of polar data you wish to include in the polar plot using the mouse while holding down the left mouse button being sure to select enough empty cells to make up a range of 360 degrees.

Once you have selected at least one column of polar data including the correct number of pad cells to make up a range of 360 degrees, click Insert then Chart then select the radar chart type and then click Next. Click the Series tab and then the "Category (X) axis labels" area so that the cursor appears there. Move the mouse back over the spreadsheet and select the column of angle values, which appear in column A and include 36 cells to make up a range 360 degrees. This last step labels the polar plot angles in degrees.

To add more columns of polar data, click the Add button then click the values area and press backspace several times to clear it. Select another column of polar data at a different frequency, being sure to select the same number of empty pad cells as before to make up 360 degrees. You can optionally rename each data series according to the frequency value displayed at the top of each data column. Press Next twice. Click the button indicated to display the chart as a new spreadsheet.

The above procedure is not intended to be exhaustive in explaining how to use Excel to graph exported *MLSSA* waterfall data. See the Excel documentation and online help for making fine adjustments to the graphic presentation.

### 3.26 Speech Transmission Index

MLSSA calculates the speech transmission index (STI) and the rapid STI or RASTI from all the available impulse response data regardless of the marker and cursor positions. The STI is an objective measure of speech intelligibility. RASTI is a simplified version of STI, which takes slightly less computation time but is less accurate than the full STI. You perform these post-processing operations through the Calculate STI Full, Calculate STI Rapid, Calculate STI Men and Calculate STI Women commands in the time domain.

STI and RASTI measurements properly account for the background noise provided the white-MLS stimulus is passed through a speech-weighting filter prior to applying it to the system to be measured. A simple speech-weighting filter is shown in the Applications section 6.21 titled "Speech Intelligibility — STI and RASTI".

MTF Matrix (Calibrated)							
Frequency-Hz	125	250	500	1000	2000	4000	8000
level dB-SPL	84.6	91.5	92.1	89.3	84.7	79.5	62.6
m-correction	1.000	1.000	1.000	0.999	0.999	0.999	0.986
0.63	0.960	0.989	0.986	0.961	0.880	0.617	0.173
0.80	0.956	0.988	0.985	0.961	0.879	0.613	0.168
1.00	0.952	0.987	0.984	0.960	0.877	0.615	0.168
1.25	0.949	0.986	0.984	0.960	0.877	0.626	0.175
1.60	0.944	0.984	0.983	0.960	0.877	0.629	0.175
2.00	0.940	0.982	0.981	0.959	0.878	0.621	0.172
2.50	0.936	0.979	0.979	0.957	0.876	0.622	0.181
3.15	0.932	0.976	0.978	0.955	0.877	0.618	0.175
4.00	0.919	0.972	0.975	0.955	0.873	0.612	0.163
5.00	0.907	0.968	0.973	0.952	0.877	0.622	0.168
6.30	0.887	0.962	0.968	0.951	0.870	0.619	0.168
8.00	0.867	0.952	0.963	0.948	0.873	0.615	0.167
10.00	0.838	0.943	0.957	0.944	0.871	0.619	0.160
12.50	0.803	0.930	0.950	0.940	0.865	0.612	0.146
octave MTI	0.859	0.977	0.989	0.942	0.782	0.570	0.269

STI value= 0.753 (0.808 modified)    ALcons= 2.9%    Rating= EXCELLENT

ESC to exit or F1 to print. MLSSA: STI

Typical Full STI screen

The figure above shows a typical full STI screen computed by the Calculate STI Full command. MLSSA analyzes the impulse response in 7 octave bands (across) and 14

modulation frequencies (down). The matrix data consists of 14 modulation indexes computed at modulation frequencies ranging from 0.63 to 12.5 Hertz in 1/3-octave intervals. The m-correction row displays the auditory masking correction [8]. For a perfect system, all the modulation indexes would be nearly 1.000. The 14 modulation index values in each band are combined to form the octave modulation transmission index (MTI) values, which are shown on the last row. These in turn are combined to determine the overall STI value. Finally, the %ALcons is computed from the overall STI value. The seven channel transmission indexes are combined on a weighted basis to determine the overall STI value. Two sets of weighting factors are supported, the original weighting factors given by Steeneken and Houtgast [8] and the modified weighting factors given by French and Steinberg. The STI value using the modified weighting factors is displayed in parentheses.

The original and modified weighting factors are listed in the table below.

	125	250	500	1000	2000	4000	8000 Hz
Originals	.13	.14	.11	.12	.19	.17	.14
Modified	0.0	.074	.149	.198	.337	.241	0.0

Note that *MLSSA* uses all seven octave bands for STI when the measurement bandwidth is set to 12 kHz or above. When the measurement bandwidth is lower, *MLSSA* will only analyze those octaves that fall within the selected measurement bandwidth. For example, if you select a 6 kHz bandwidth through the Acquisition Bandwidth command, *MLSSA* will compute STI based on bands 1 through 6 with band 7 (8 kHz) being skipped. You might want to do this to correlate *MLSSA* STI measurements with others in the literature that only go to 4 kHz. In addition, you should reduce the bandwidth if the reverberation time exceeds 7 seconds to avoid time-aliasing errors.

The Calculate STI Men and Calculate STI Women commands use a revised STI method that accounts for the differences between male and female talkers. The revised method is specified by the international standard IEC 60268-16 Third Edition [17].

Please refer to this standard for details on the revised STI method. Both the octave weighting factors as well as the reference speech spectrum differ between these two commands and both are different from the original STI weighting factors and reference speech spectrum. Although the revised STI method implies the need for two speech filters, that is, one for male and one for female talkers, *MLSSA* is able to make do with a single speech filter due to its ability to correct for minor errors in the reference speech spectrum in software. The revised STI speech spectra according to IEC 60268-16 Third Edition are listed in the third and fourth columns of text file *SPEECH.INI* located in the main *MLSSA* directory.

The Calculate STI Men and Calculate STI Women commands, in conformance with IEC 60268-16 Third Edition (2003-03), also include MTF corrections that are more sophisticated than the original STI algorithm. First, the absolute threshold of hearing is accounted for. Thus, revised STI values will decrease with decreasing SPL at the listening position, even in cases where there is negligible interfering noise. Second, the auditory masking correction factor depends upon the absolute SPL level in each octave band at the listening position. The original STI algorithm used by the Calculate STI Full command, in contrast, employs an auditory masking factor of 0.000316, which is independent of the absolute SPL. Because both corrections (masking and absolute hearing threshold) defined by IEC 60268-16 Third Edition require knowledge of absolute octave-band SPL levels at the listening position, that is, where the measurement microphone is positioned, you must use the Library Microphones command to enter your microphone sensitivity and microphone preamp gain. You must also select this microphone/preamp combination prior to performing measurements. If no microphone/preamp combination was defined and selected prior to performing measurements, the Calculate STI Women and Calculate STI Men commands will refuse to perform the analysis. In cases where you have archived old TIM files that do not contain (or contain spurious) microphone calibration data, use the Library Microphones Modify-file command to enter, as best as can be determined from memory or from notes, your best estimate of the microphone sensitivity and microphone preamp gain existing at the time when the original measurement was performed. Whenever the Calculate STI Women or Calculate STI Men commands are successfully executed, the second row in the MTF matrix, labeled "m/thresh-cor" (masking/threshold-

correction) contains the MTF correction factors for each octave band that incorporate both corrections described above: the more sophisticated level-dependent auditory masking correction as well as the absolute hearing threshold correction, both according to IEC 60268-16 Third Edition.

The figure below shows Rapid Speech Transmission Index (RASTI) computed from the same impulse response. RASTI is a simplified version of STI that only considers two octave bands (500 Hertz and 2 kHz) and nine modulation frequencies. You compute RASTI through the Calculate STI Rapid command.

MTF Matrix (Calibrated)							
Frequency-Hz	125	250	500	1000	2000	4000	8000
Level dB-SPL			92.2		84.9		
0.71					0.879		
1.00			0.989				
1.41					0.886		
2.00			0.988				
2.80					0.883		
4.00			0.987				
5.60					0.886		
8.00			0.986				
11.20					0.882		

RASTI value= 0.885    ALcons= 1.4%    Rating= EXCELLENT

ESC to exit or F1 to print. MLSSA: RASTI

**Typical RASTI screen**

Note that *MLSSA* uses brick-wall (infinite roll-off) octave analysis filters for full STI while the RASTI calculation uses 6-pole IIR filters having a roll-off rate of about -18/dB octave. For this reason, full STI is recommended for characterizing sharply bandlimited systems such as Codecs.

STI and RASTI calculations will warn you if there is a change in the setup from STI.SET that might result in questionable STI or RASTI readings by displaying the word

"Uncalibrated" at the top of the screen. In order to assure accurate measurements, you should measure a full 65535-point impulse response over a minimum 12 kHz bandwidth (3 kHz for RASTI) without averaging or pre-averaging and with a speech filter inserted. Thus you need to start *MLSSA* with the STI.SET setup file and the /6 option for "Calibrated" STI and RASTI measurements.

Minor errors in the speech filter can be corrected. You correct for errors in the response of your speech filter by editing text file SPEECH.INI consisting of four columns of octave levels. The first column is the actual speech filter response integrated over whole octaves. The second column is the reference speech spectrum of the original STI algorithm. The third column is the revised STI reference speech spectrum for male voices. The fourth column the revised STI reference speech spectrum for female voices. *MLSSA* reads all four data sets at start-up and corrects STI and RASTI for any deviation between the actual speech filter response and the specified speech spectrum. The Calculate STI Full and Calculate STI Rapid commands use the second column as the reference speech spectrum. The Calculate STI Men command uses the third column and the Calculate STI Women command the fourth column as the reference speech spectra respectively. Note that corrections for the speech spectrum or speech filter are only calculated and applied to "Calibrated" STI or RASTI measurements, which require a full 65535-point impulse response.

All STI and RASTI screens display the octave band levels (signal plus noise) in units of dB-SPL (or dB ref 1 volt if no microphone calibration data is available). The displayed levels are also corrected for any minor differences between the applicable reference speech spectrum and the speech filter response, both defined in SPEECH.INI. Therefore, the displayed levels cannot be regarded as octave SPL present at the microphone position. Rather, these are virtual levels that would exist if the response of the speech filter exactly agreed with the applicable reference speech spectrum.

See section 6.21 titled "Speech Intelligibility — STI and RASTI" for more information.

See references [8], [11], [12] and [17] for more information regarding STI and RASTI.

### 3.27 Automatic Filename Generation

Data files are saved through the Transfer Save command from both the time and frequency domains. Normally, you explicitly enter your own filename. *MLSSA* will optionally generate a new numbered filename when the Right arrow key is pressed. The filename generated depends upon the parameters set by the Transfer Autaname commands. When the file is saved, *MLSSA* will also log that filename and any comment to an optional log file.

Autonamed data files have the form <PATH>\<ROOT>-<NUMBER>.<EXT> where <EXT> is "TIM" for time-domain files and "FRQ" for frequency domain files. Frequency and time domain data files use independent numbers, which are automatically incremented whenever you use the Right-arrow key to save an autonamed file. The Transfer Autaname Path command sets an optional MS-DOS path for the location of autonamed data files and the optional log file. The Transfer Autaname Root command sets the root name and the Transfer Autaname Number command sets the starting number.

Changing the path via the Transfer Autaname Path command normally automatically resets the autoname number to zero. To change the path without resetting the autoname number simply append a backslash to the entered path name, for example, enter C:\DATA\ instead of C:\DATA.

The Transfer Autaname Log-file Create command creates an optional log file for autonamed files having a common root name. The name of this file has the form <PATH>\<ROOT>-00.LOG. *MLSSA* also creates or appends a master log file MASTER.LOG — located in the main *MLSSA* directory — with the name of the newly created log file plus the log file description. Once a log file is created, all autonamed files saved through the Transfer Save Right-arrow command are automatically logged.

The Transfer Autaname Log-file List command lists the contents of the specified log file or the master log file MASTER.LOG. If you only press the Enter key, *MLSSA* will automatically list the current log file contents.

When loading (or saving) autonamed files through the Transfer Load or Transfer Save commands, you need only to enter a hyphen as the first character followed by the file number. For example, if the autaname path is C:\MLS and the root is JOB01, then entering "-01" will cause the filename C:\MLS\JOB01-01.TIM to be loaded or saved from the time domain. In general, in any command requiring a filename or path, anything you do that is preceded by a hyphen as the first character is always expanded according to the current autaname parameters. For example, during Transfer Load, by entering a hyphen followed by the Right arrow key, *MLSSA* will display a list of all current autonamed files.

The highest number autonamed filename is limited only by the 8-character DOS limit on filenames. That is, if you want to go up to 3 digit file numbers you must limit the root name to 4 characters or less.

### 3.28 Creating and Running Macros

*MLSSA* incorporates a keystroke macro processor. Macros are formed by recording a series of *MLSSA* commands. Later, you can play back the macro to repeat those commands without retyping them.

To start recording a macro you press the Ctrl-r key combination. To stop recording the macro press Ctrl-r a second time. When macro recording is in effect, all the commands you enter are recorded in a macro. You can save the recorded macro to a macro file through the Macro Save command. Previously saved macros can be reloaded through the Macro Load command.

You execute a macro through the Macro Execute command. The Macro Debug Enable command causes a macro to single-step during execution for testing purposes.

If Ctrl-u is entered during macro record when *MLSSA* is prompting you for a filename or other input string, the resulting macro will automatically pause during execution to allow the operator to enter a specific (and possibly variable) filename or string.

Normally, you load a previously created macro through the Macro Load command and execute this macro through the Macro Execute command. With instant macros, however, you can load and execute a macro through a single key combination. Instant macros are accessed through function keys Alt-F1 through Alt-F10. To make a previously created macro into an instant macro you simply give it a name corresponding to the desired function key. For example, if you save a macro with the name ALT-F5.MAC then pressing the Alt-F5 key combination will automatically load and execute that macro. Note that there can be only 10 instant macros in a given directory corresponding to function keys Alt-F1 through Alt-F10.

When you save a macro, you can also enter a title that describes its purpose. This title can be displayed later through the Macro Info command, which also displays the macro filename and size.

The Macro Free command frees up the memory used for the macro buffer. You use this command when you need more memory for other operations but don't need the macro facility.

Five alphanumeric string variables are available at any time by pressing Ctrl-v through Ctrl-z. Pressing any one of these key combinations causes a pre-defined string to be entered or displayed, depending upon the context. Each string can be up to 58 characters long and can be used to store, for example, long DOS paths or speaker model names and numbers. They are not true macros however because they cannot store action keys such as the Enter key. They can only store alphanumeric characters. They can be used to store frequently needed strings to save re-typing them. *MLSSA* remembers the values of all string variables between sessions.

These five string variables are entered and/or modified through the Macro Variable command. If you later press Ctrl-v through Ctrl-z from the top menu level, the corresponding string is simply displayed in the message window for reference. During macro execution, this feature can be used to display a message or prompt a macro user for input.

Invoking the same key combination at a lower menu level when, for example, *MLSSA* is prompting you to enter a filename, causes the same string to be entered, as if typed by hand. These variables can be changed and invoked during macro record and macro execute as well as when no macro is executing.

Note that by executing the Macro Variable command during macro record and entering Ctrl-u rather than an actual string, the macro user can dynamically change the value of that string variable during macro execution. The newly entered string can then be used by the same or another macro as a message, filename, file prefix, comment, serial number or other string by pressing the appropriate Ctrl key combination. For example, if during macro record Ctrl-u is entered as the input to the Macro Variable ctrl-V command then during macro execution the user can enter a new value for the Ctrl-v string variable.

The macro processor does not support nested macros but you can loop and chain macros. To loop a macro, simply invoke the Macro Execute command as the last command in the macro during recording. When the macro is executed, it will repeat indefinitely or until the Esc key is pressed. Note that any executing macro can be terminated by pressing the Esc key.

You can also chain macros. To do this, invoke the Macro Load command as the last command during macro record. When the macro is executed, it will automatically load and execute the second macro specified by the Macro Load command. Any number of macros can be chained in this way. Chained macros can be used to break up a large macro into several smaller macros.

There is also a special macro filename called AUTOEXEC.MAC. When *MLSSA* is first started, it looks for this special macro filename in the current directory. If it finds AUTOEXEC.MAC, *MLSSA* loads and executes it automatically.

You can also have *MLSSA* load and run another macro file at start up through the /m command line switch. For example the command line:

```
MLSSA loads /mAMACRO
```

starts *MLSSA* with setup file *LOUDS.SET*, immediately loads it, and runs the macro file *AMACRO.MAC*. For a complete list of all command line switches and their functions start *MLSSA* with the */?* switch for help.

The graphics display is normally updated during macro execution, which might slow down some complex macros. The Ctrl-F3 graphics command allows you to blank the graphics display thus speeding up macro execution. To unblank the graphics display and show the final results press Ctrl-F3 a second time.

There is also a special set of macro commands that begin as Macro Command. These commands are used to direct the macro processor to do certain tasks during macro execution. For example, the Macro Command Pause command causes macro execution to pause until any key is pressed. The Macro Command Message command causes a message to appear in the message window during macro execution. For example, suppose you want an operator to connect a new loudspeaker before proceeding with a test. You could display "Connect next speaker" using the Macro Command Message command and wait for the operator through the Macro Command Pause command. When you execute the macro the message will appear and macro execution will pause until the operator presses a key.

The Macro Command Headline command also allows you to display a short message but in larger type positioned in the center of the screen. Note, however, that this macro command works only if you had previously blanked the graphics display (Ctrl-F3). If the graphics display is not blanked then no message will appear during macro execution.

The Macro Command Exec, Macro Command Run, and Macro Command Call commands all allow you to spawn another program from an executing macro. As soon as the child program finishes, control returns to the macro, which continues executing. These commands allow you to write custom programs for QC or other purposes that can be spawned from inside a macro. The Macro Command Exec command first swaps all of the *MLSSA* program and data from conventional memory to either EMS memory or, if EMS is not available, to your hard disk thus freeing up as much conventional memory as possible for the child program. The Macro Command Run command does not swap out *MLSSA* leaving less conventional memory for the child

program, which can read *MLSSA*'s data arrays from memory. This command is also faster than the Exec command since no swapping operations are performed. The Macro Command Call command is similar to Macro Command Run except that the *MLSSA* screen is not blanked. This command is intended to save time when the spawned program does not need to draw anything on the screen, for example, in order to control some external hardware.

DOS commands and DOS batch files can be executed from inside macros through the Macro Command Run and Macro Command Exec commands. To do this you must invoke the DOS command interpreter COMMAND.COM, which is usually located in the root directory on PCs running DOS or, in the WINDOWS directory on PCs running Windows 95/98. For example, suppose you wish to run batch file TEST.BAT and exit back to *MLSSA* immediately after completion to resume macro execution. To do this you would enter the following command string when prompted by the Macro Command Exec command or Macro Command Run command during macro record:

```
C:\WINDOWS\COMMAND /c test.bat
```

The /c switch indicates that the string following it is to be interpreted as a DOS command. To keep the DOS command interpreter running after completion of the DOS command or batch file, use the /k switch as in:

```
C:\WINDOWS\COMMAND /k dir/w
```

When this line is executed, the current DOS directory will be displayed in wide format and then the DOS command prompt will appear. The user is free to execute more DOS commands manually, if desired. When finished, the user types Exit on the DOS command line to exit to *MLSSA*, which resumes macro execution. If you enter no string after the /k switch then the DOS command prompt will appear immediately.

To step through a batch file for debugging purposes, add the /y switch before the /c or /k switch as in:

```
C:\WINDOWS\COMMAND /y /c test.bat
```

Note that the path to COMMAND.COM can be stored as a string variable through the Macro Variable command to save re-typing it each time you want to execute a batch file or DOS command. For example, entering "C:\WINDOWS\COMMAND" as the Ctrl-v string variable allows you to execute the same batch file from Macro Command Run or Macro Command Exec simply by entering:

```
<ctrl-v> /y /c test.bat
```

The Macro Command Load-setup command permits a macro to load a new setup file without displaying the Setup screen. This command lets you quickly change the *MLSSA* setup without disturbing the current display screen. Note that it must be executed from the time domain however.

The Macro Command Signal and Macro Command Wait commands are used to synchronize *MLSSA* macros with external hardware devices. The Macro Command Signal command causes a positive-going TTL pulse of 10 msec duration to be output on J4 pin-2 during macro execution. The Macro Command Wait command causes macro execution to pause until a positive going TTL pulse is detected on J4 pin-5. The input pulse duration must exceed one sample period. For example, if the sample rate is set for 33 kHz then the input pulse must be at least 30 microseconds in duration.

The Macro Command IF-PASSED-QC Execute command and Macro Command IF-FAILED-QC Execute command conditionally chain to another macro depending upon whether the last QC Compare operation resulted in a PASS or FAIL. You are responsible, however, for chaining back to the original calling macro by inserting a Macro Load command as the last command in any macro executed by these conditional commands. Similarly, Macro Command IF-PASSED-QC Signal and Macro Command IF-FAILED-QC Signal both output a 10 ms pulse on J4 pin-2 depending upon the last QC Compare result. Macro Command IF-FAILED-QC Alarm suspends macro execution and continuously beeps until Esc is pressed if there was a QC failure.

The Macro Command IOOP command is used to re-execute a section of a macro up to 32767 times. To form a loop within a macro use Macro Command marK first to mark the point in the macro where you want the loop to begin and then Macro Command

LOOP where you want the loop to end. Macro Command LOOP prompts you to enter the number of loops, up to 32767, to perform during macro execution. Note that you can have more than one loop within a macro but loops cannot be nested.

You can edit macros by converting a macro file to a plain text file and using a standard text editor, such as EDIT supplied with DOS, to make any changes. Then you convert the corrected text file back into a macro file to run. These conversions are performed using the MAKEMAC.EXE utility program. MAKEMAC converts macro files into text files or text files into macro files. Macro files contain binary data and have the default extension "MAC" while text files have the default extension "TXT". The normal procedure for creating a macro is to press Ctrl-r and record the macro including any mistakes then use MAKEMAC to convert the macro file to a text file. Then you edit the text file to correct any errors and use MAKEMAC a second time to convert it back to a macro file.

You can also run MAKEMAC and specify all options on the command line in order to run it from DOS batch files. The DOS command line format is:

```
MAKEMAC filename filename M|T [Y to overwrite files]
```

MAKEMAC assumes the first filename to have the file extension "MAC" unless you specify another file extension. MAKEMAC assumes the second filename to have the extension "TXT" unless you specify another file extension. The third parameter must be an "M" or "T" to indicate either a macro or a text file conversion. The fourth optional parameter "Y" is to indicate you want MAKEMAC to overwrite an existing output file without asking permission. You must enter at least three parameters or MAKEMAC defaults to prompting you for each input parameter.

Within the macro text file, non-printing characters are represented by special key words enclosed in angle brackets. For example, <Esc> refers to the Esc key, <Enter> refers to the Enter key, <PgDn> refers to the PgDn key and <F1> refers to the F1 key etc. You can also precede a key name by "ctrl-", "shift-" or "alt-". For example, <ctrl-right> refers to the Ctrl-Right-arrow key combination. MAKEMAC also compresses multiple keystrokes to improve readability and reduce the amount of typing required. For

example, <Ctrl-Right 10> means that the Ctrl-Right-arrow key combination is pressed 10 times in a row. MAKEMAC ignores the Enter and Tab characters in the text file so you can use these characters to create white space. To enter these keys as part of the macro use <enter> and <tab> respectively. Note that normal blank space characters are not ignored but are interpreted literally. Anything included in double quotation marks is ignored allowing you to create comments. If you want MAKEMAC to interpret angle brackets or double quotation marks literally, you must double them (e.g. "", <<, >>).

To specify a macro title, enclose it in angle brackets and quotation marks. For example, <"This is a macro title">.

### **3.29 Microphone Calibration**

Before *MLSSA* can give you absolute SPL readings, it must know the characteristics of the microphones you intend to use to make measurements. All of the information needed should be on the calibration data sheet supplied by the microphone manufacturer.

Note that if you are using the *MLSSA RCAI*, important additional information about microphone calibration is contained the section titled "Assigning Microphones to Mic-mux Inputs" found in the *MLSSA RCAI Reference Manual*.

*MLSSA* will allow you to enter data on up to 16 microphone/preamp combinations. Even though many users will have only one microphone, there is another use for having 16 microphone/preamp combinations. Many microphone preamps have a gain switch that controls preamp gain in 10 dB increments. If you have such a preamp, you may want to enter the same microphone data multiple times but with different gain values to cover the different preamp gain settings. This may be useful if you expect to make measurements over a very large SPL range because some preamps will clip on very loud sounds when set for maximum gain.

Execute the Library Microphones command from the main menu. Next, execute the Define command by pressing 'd'. *MLSSA* will request the microphone number you wish

to enter. Enter a '1' Enter to select microphone position 1. Next, enter the microphone's manufacturer, model, serial number and sensitivity in mV/Pascal. If your microphone's sensitivity is given in units of mV/microbar, multiply by 10 to get mV/Pascal. Finally, *MLSSA* will prompt you to enter the preamp gain and display a recommended gain to the nearest 10 dB. The recommended gain is calculated to optimize SNR and dynamic range but is not a requirement. Be sure to enter the actual gain of your preamp in dB whatever it is. Microphone #1 is now defined.

The data you entered for microphone #1 can be used for measurements only if it is selected. To select a microphone press 's' for Select and enter the microphone number you want (1 in this case). Note that you can also select microphone 0 which tells *MLSSA* that no microphone is connected to the input. The Select command is important because it's the only way *MLSSA* knows what is connected to its input. If you are making measurements that don't use a microphone (e.g. impedance), you should select microphone 0. Otherwise, select the microphone number that corresponds to the actual microphone/preamp combination that is currently connected to the input. Note that if the *MLSSA RCAI* is connected, the Select command is not available and is replaced by the Assign command. See the *MLSSA RCAI* Reference Manual for details.

The accuracy of SPL measurements is only as good as the microphone data you entered. If your microphone is old or if you are uncertain of your preamp's gain you should calibrate them both using a microphone calibrator. To calibrate a microphone/preamp combination you first select it using the Select command. Press 'c' for Calibrate and *MLSSA* will prompt you to enter the dB-SPL output level of the calibrator. This number is usually printed on the outside of the calibrator. Next *MLSSA* will prompt you to turn on the calibrator and press any key. Make sure the microphone is inserted into the calibrator and that the calibrator stays on until the calibration cycle is complete. If calibration is successful, *MLSSA* will update the microphone sensitivity entry. After calibration, overall accuracy will be as good as the accuracy of the calibrator.

The Library Microphones Modify-file command is useful in those cases where a measurement was performed with the wrong microphone number selected or, when incorrect microphone data was entered. This command prompts you for a TIM file from

the time domain or an FRQ file from the frequency domain. You can also override these default file extensions manually but only TIM and FRQ data files can be modified. Upon completion, this command will overwrite the microphone data stored in the header of the specified TIM or FRQ data file with the new microphone data corresponding to the microphone number you entered.

*MLSSA* can also correct for microphone frequency response variations through the Reference Auxiliary iMport Text Transfer-function command. To use this feature you need to create a text file in Bode format specifying the frequency response of the microphone you wish to correct. Microphone frequency response data is often plotted on the calibration data sheet supplied by the microphone manufacturer. See Applications section for more information. High quality instrument grade microphones are usually flat within +/- 1 dB and such small deviations from flatness do not usually warrant any correction.

### **3.30 Sound Level Measurements**

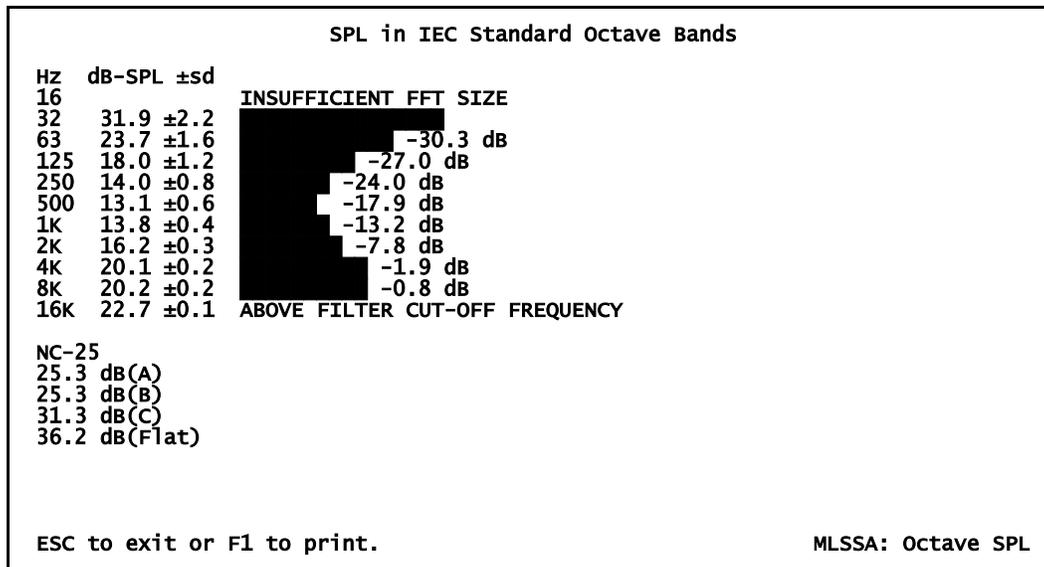
*MLSSA* makes absolute sound level measurements in the time domain as well as the frequency domain. Weighted sound levels (e.g. A, B or C weighting) can only be computed from the frequency domain. In addition, you must have previously entered accurate sensitivity data on your microphone through the Library Microphones command to make any accurate sound level measurements.

In the time domain, the Calculate spL command computes the unweighted sound level from the measured impulse response or microphone signal. If the signal is first filtered using the View Data Filtered command, the computed sound level refers to the portion of the signal falling within the filter bandpass. For example, if you measure the impulse response of a room over a 10 kHz bandwidth and execute the Calculate spL command the result will display unweighted sound level in units of dB-SPL of the MLS stimulus over a 10 kHz bandwidth. If you then filter the impulse response using a 1 kHz, 1 octave filter through the View Data Filtered command and repeat the Calculate spL command the displayed sound level will reflect the MLS level in that octave band only. You can therefore compute the sound level for any octave band simply by changing the

digital filter's center frequency through the View Filter Center command and recomputing through the Calculate SPL command.

To measure sound levels in the frequency domain you must be in the Scope acquisition mode and the Power Spectrum FFT mode. After acquiring raw noise or sound data you execute the FFT Execute command to enter the frequency domain then you use the Calculate SPL commands to compute and display sound levels. These commands compute and display A-weighted, B-weighted, C-weighted, unweighted or octave-band sound levels in units of dB-SPL. The frequency domain method has the advantage of being able to compute the weighted levels and is faster than the time domain method if you want multiple narrow-band levels. However, you cannot apply the frequency-domain method to impulse response measurements.

The Calculate SPL Inclusive command computes and displays all the frequency-domain sound level measures on one data screen as shown in the figure below.



**Room Noise Octave-band SPL Display**

The message "INSUFFICIENT FFT SIZE" means that the current frequency resolution is too coarse to compute the SPL accurately in those bands so labeled. You can increase the FFT size to increase frequency resolution but this will require more processing time and increase the amount of random error. The message "ABOVE FILTER CUT-OFF FREQUENCY" means that the antialiasing filter cut-off is too low to allow accurate SPL measurements in those octaves so labeled. You can often eliminate this message by increasing the bandwidth through the Acquisition Bandwidth command but then frequency resolution will be made coarser.

The expected standard deviation for each octave SPL value is displayed immediately after each dB-SPL value. The expected standard deviation is computed based upon amount of data used to make the measurement. Thus increasing the Acquisition Length and/or using the Go Average command from the frequency domain both increase the total amount of data used in the calculation which in turn decreases the expected random error in the measured level. Note, however, that time domain averaging is not valid for sound level measurements and *MLSSA* will refuse to compute the Power Spectrum of acquired data that was first averaged in the time domain. When executed from the frequency domain, however, the Go Average command spectrally averages data acquired over very long periods if desired. This command can be stopped prematurely at any time by pressing Esc but *MLSSA* will still display the results based on all of the data acquired up to that time.

The noise criterion (NC) rating is also calculated and displayed. The NC rating is defined as the lowest NC curve that is not exceeded by the actual octave-band noise spectrum. In the figure, the NC rating shown is NC-45. The difference in decibels between the rated NC curve and the actual octave-band noise level is displayed to the right of each octave's bar graph. Note that, by definition, these differences must always be negative.

For 1/2, 1/3 or 1/6-octave noise measurements, use the View Magnitude Integrated command. This command is similar to the Calculate SPL Octaves command except that the result is displayed as a normal *MLSSA* frequency graph and therefore allows bandwidths narrower than one octave to be displayed. Energy is *integrated* (summed) over each octave, 1/2-octave, 1/3-octave or 1/6-octave band and the result is displayed

as a stair-step style graph. This command contrasts with the View Magnitude sSmoothed command, which *averages* the energy over the specified band. Note also that before using the View Magnitude Integrated command you must first select the Microphone reference through the View Magnitude Reference Microphone command. Otherwise, the resulting curve will not be scaled properly in units of dB-SPL.

### 3.31 Energy-time Curves

*MLSSA* supports several energy-time curve options.

The window function associated with an energy-time curve is applied in the frequency domain not in the time domain as is typical. For example, executing the View Energy-time-curve Unfiltered half-Hann command means that a half-Hann window is applied in the frequency domain. Thus the highest frequencies are attenuated and do not contribute much to the ETC. This is the type of energy-time curve recommended for loudspeaker measurements.

The View Energy-time-curve Unfiltered Blackman-Harris command, in contrast, applies a full Blackman-Harris window meaning that both the lowest and the highest frequencies are attenuated and only mid-band frequencies contribute to the result. Note that the Blackman-Harris option yields the fastest computation speed. The View Energy-time-curve Unfiltered Unwindowed command gives all frequencies equal weight.

*MLSSA* also supports a filtered energy-time curve, which is computed through the View Energy-time-curve Filtered command. This command applies a full Hamming window over the frequency range specified by the View Filter commands. For example, if a 1-octave, 1-kHz filter is selected then only frequencies falling between 707 Hz and 1414 Hz contribute to the result and frequencies near 1 kHz are given the most weight. These windows should not be confused with the time windows, which are controlled by the FFT Window commands.

What is often called the "integration time" of the ETC is just the time resolution of the ETC, which equals roughly  $1/B$  where  $B$  is the bandwidth over which the ETC is

computed. This means that filtered ETCs will generally have less time resolution (larger integration times) than wideband or unfiltered ETCs. If you display a filtered ETC having a 707 Hz frequency range as in the example above, the time resolution or integration time is about  $1/707 = 1.4$  milliseconds. However, even if the bandwidth of the ETC is wide (small integration time) its ability to distinguish between closely spaced arrivals is still limited by the nature of the ETC itself as explained in [10].

You can make reverb/direct calculations on any energy-time curve. However, the results may differ from the reverb/direct calculations made on wideband or filtered impulse response data because of the influence of the window functions mentioned above. Only the unfiltered and unwindowed energy-time curve is guaranteed to yield the same reverb/direct ratio as the unfiltered impulse response.

See reference [10] for more information on ETCs.

### **3.32 Acoustical Parameters**

In the time domain, *MLSSA* measures many acoustical parameters including, reverb/direct ratio, early/late ratios, center time and strength. All of these parameters are derived from the room's impulse response, which may be either unfiltered (wideband) or filtered (narrow-band). You filter the impulse response through the View Data Filtered command. The filter center frequency and bandwidth are controlled through the View Filter Center and View Filter Width commands respectively.

The reverb/direct ratio shows the ratio in dB between the energy of the reverberant field and the energy of the direct sound. The command Calculate Reverb/direct performs this function but you must position the cursor to divide the impulse response between the direct sound and the reverberant field. The direct sound includes the initial spike in the impulse response up until the first room reflection. The reverb/direct ratio can also be computed for any energy-time curve but the results may differ from those obtained from the impulse response because of the frequency-domain window functions used to compute the various energy-time curves. Only the unwindowed and unfiltered energy-time curve is guaranteed to yield the same reverb/direct ratio as the unfiltered impulse response.

The early/late ratio, also known as clarity, is the ratio, expressed in dB, between the energy of the early arriving reflections (including the direct sound) and the energy of the late arriving reflections (mostly reverberation). The early/late ratio is considered a measure of acoustical clarity where higher values mean greater clarity. The dividing point between early and late arrivals is usually 80 milliseconds (C80) measured from the first arrival but *MLSSA* allows you to specify any early time period (C??) when you execute the Calculate Early/late command. There is no need to position the cursor since *MLSSA* is able to determine the first arrival time on its own. The Calculate Early/late command also computes and displays the definition (D??) defined as the ratio, expressed as a percentage, of the early arrival's energy to the total energy in the impulse response.

Note that the Calculate Early/late command, if terminated prematurely by pressing the Esc key, will display the time of the first sound arrival in milliseconds. If desired, you can then move the cursor to this position by pressing Shift-F2 and entering this time value.

Center time (TS) is another measure of acoustical clarity. Center time is defined as the centroid of the impulse response energy relative to the first arrival. You calculate center time from the impulse response through the Calculate Center-time command. A high center time indicates poor clarity and vice versa.

Acoustical strength is defined as the energy of the impulse response relative to the energy of the same source at a distance of 10 meters. You determine acoustical strength through the Calculate strenGth command. You must position the cursor to divide the impulse response between the direct and reverberant fields. *MLSSA* uses the direct sound portion of the impulse response and the inverse square law to predict the source level at 10 meters. The result is then divided into the total impulse response energy to yield acoustical strength, which is displayed in dB. See reference [9] for more information.

The View Schroeder-plot Filtered and Unfiltered commands cause *MLSSA* to compute and display the reverse integral of the squared impulse response. Schroeder [3] proved that this curve represents the true reverberant decay curve. However, Schroeder's

theorem holds exactly only if a) the measurement is perfectly noiseless and, b) the integration is performed over infinite time. Since neither condition is ever realized in practice, *MLSSA* incorporates algorithms designed to correct the decay curve both for the influence of any background noise (as well as any noise created by loudspeaker nonlinearities) and the fact that the measured impulse response is finite in length. Both corrections are enabled or disabled by the View Schroeder-plot Compensation Enable and Disable commands. If enabled, the symbol "(comp)" is displayed in the header of the graph. *MLSSA* may not be able to accurately perform these corrections even when enabled if certain conditions are not met. If one or more of these conditions are not met, the symbol "(comp)" is not displayed even if Compensation is enabled. See the "Tabular Acoustical Parameters" section below for a more detailed discussion of these conditions.

*MLSSA* computes the early decay time (EDT-10dB) and true reverberation time from Schroeder-plot decay curves using linear regression over the decay range determined by the marker and cursor positions. The EDT-10dB parameter is defined as the reverberation time measured over the first 10 dB of decay and is considered a better measure of the subjective reverberation time as compared to the true reverberation time, which is measured over some linear portion of the decay curve. You generate the appropriate decay curve through the View Schroeder-plot Unfiltered or View Schroeder-plot Filtered commands and then use the Calculate T60 command to compute either EDT-10dB or the true reverberation time. To measure EDT-10dB, execute the Calculate T60 command, press the Home key, press Ctrl-F2 and enter -0.01 to move the cursor to the start of the decay. Press F9 or the right mouse button to fix the marker there then press Ctrl-F2 and enter -10 to place the cursor at the -10 dB point on the decay curve. Finally, press Enter to calculate the early decay time.

True reverberation time RT-20dB is measured over a -5 to -25 dB decay range. To measure RT-20dB, execute the Calculate T60 command, press the Home key, press Ctrl-F2 and enter -5.0 to move the cursor to the -5 dB point on the decay. Press F9 or the right mouse button to fix the marker there then press Ctrl-F2 and enter -25 to move the cursor to the -25 dB point on the decay curve. Finally, press Enter to calculate RT-20dB.

### 3.33 Tabular Acoustical Parameters

MLSSA will automatically compute and tabulate the most important acoustical parameters in IEC-standard octave bands through the Calculate Acoustics command. The figure below shows a typical Calculate Acoustics screen.

IEC 1/1-Octave Band Acoustical Parameters									
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000	
S [dB-SPL]	60.2	64.3	68.5	73.7	78.8	78.9	79.4	SPL- weighted Averages	
N [dB-SPL]	33.4	34.1	38.3	43.1	43.6	44.8	49.6		
SNR [dB]	26.8	30.2	30.3	30.6	35.1	34.1	29.7		
C50 [dB]	-6.85	-4.15	-1.69	-0.41	4.76	5.30	8.81		3.442
C80 [dB]	-0.51	-0.21	1.01	2.23	7.04	7.80	11.94		5.846
D50 [%]	17.1	27.8	40.4	47.7	75.0	77.2	88.4		63.439
TS [ms]	114.7	108.1	104.2	88.2	43.9	36.8	15.3		62.548
EDT-10dB [s]	1.390	1.630	1.670	1.487	0.939	0.864	0.939		1.182
RT-20dB [s]	1.614	1.308	1.293	1.296	1.270	1.198	0.861		1.265
(-5:-25) r	-0.996	-0.995	-0.999	-0.999	-0.999	-0.998	-0.996		-0.999
RT-30dB [s]	1.560	1.332	1.342	1.284	1.247	1.166	0.841		1.254
(-5:-35) r	-0.997	-0.998	-0.999	-1.000	-1.000	-0.999	-0.999		-1.000
RT-USER [s]	1.705	1.258	1.322	1.297	1.237	1.178	0.895		1.253
(-10:-25) r	-0.992	-0.993	-0.999	-0.999	-0.999	-0.998	-0.995		-0.999

File: C:\VIEW\SAROP-2.TIM 4-18-95 1:12 PM  
SAROP center house

ESC to exit or F1 to print. MLSSA: Acoustics

#### Acoustical Parameters in Tabular Form

For each octave band MLSSA computes the signal level (S) in dB-SPL, the background noise level (N) in dB-SPL and the signal-to-noise ratio (SNR) in dB. The displayed levels show the MLS signal level present in each octave band. These S values should not be interpreted as the room's frequency response since they are determined by integrating over whole octaves and hence show a +3 dB/octave bias due to the white MLS spectrum.

Also shown are the clarity at 50 milliseconds (C50), the clarity at 80 milliseconds (C80), the definition at 50 milliseconds (D50), the center time (TS) and the early decay time over the first 10 dB of decay (EDT-10dB). Finally, MLSSA computes three RT<sub>60</sub> values

(RT-20dB, RT-30dB and RT-USER) corresponding to the three decay ranges shown in parentheses.

All parameters are corrected for the influence of background noise when a valid estimate of the noise can be performed. If the impulse response is less than 65535 points in length, *MLSSA* tries to analyze the initial time-of-flight delay gap in the measured impulse response in order to get a good estimate of the background noise. This method results in poor accuracy if the delay gap is less than about  $5/B$  seconds where  $B$  is the current octave bandwidth in Hz (e.g. for the 1 kHz octave band  $B = 1000 \cdot 0.707 = 707$  Hz). If this condition is violated, the Calculate Acoustics command will no longer attempt to estimate the background noise and apply noise correction to the parameters for those octave bands. This is indicated by a -- appearing in the N and SNR fields. If, however, you measure a full 65535-point impulse response, *MLSSA* uses the last 4096 points of the impulse response instead of the initial time-delay gap to estimate the noise level. This will usually satisfy  $5/B$  requirement except for the 32 Hz octave band. Such tail-end background noise estimates are unaffected by the length of the initial time-of-flight delay gap but, you must start *MLSSA* with the /64K option and select a full 65535-point Acquisition Length with a 16th-order MLS for this method to be enabled. Otherwise, *MLSSA* defaults to the using the initial time delay gap.

When the SNR exceeds 10 dB, *MLSSA* uses the displayed noise level to correct the decay curve used to compute EDT-10dB and the three RT values. If the SNR falls below 10 dB, its entry in the table will blink and noise correction will no longer be applied to those parameters but will still be applied to the other parameters in that octave band. Nonetheless, all parameters should be regarded as suspect whenever the SNR falls below 10 dB.

The decay ranges used to compute the three  $RT_{60}$  values can be changed by editing the text file REVERB.INI although only RT-USER should be changed as both RT-20dB and RT-30dB are set up to conform to ISO 3382 [13]. If the required decay range is not available because the measured impulse response is too short, *MLSSA* displays a "T" to indicate this. To insure an adequately long measured impulse response, always start *MLSSA* with the /64K option or, you can edit MLSSA.INI to make this option the default.

You should also make sure the Acquisition Length is 65535 points and that the 16th-order MLS stimulus is selected.

Below each displayed RT value is a negative number known as the correlation coefficient. The correlation coefficient shows how closely the corresponding decay curve fits a perfectly straight line over the decay range indicated. A correlation coefficient of -1.000 indicates a perfect fit while smaller negative values indicate that the decay curve may include a bend or other nonlinearity. A correlation coefficient smaller than -0.95 indicates a nonlinear reverberant decay process, which makes the corresponding RT value suspect. When the correlation coefficient's magnitude is smaller than 0.95, its entry will blink to indicate that the RT value above it is suspect.

The Calculate Acoustics command prompts you for the starting octave number (1=32 Hz thru 5=500 Hz). If you type only Enter however, the starting octave is the last one selected or the default given in REVERB.INI.

You can also edit REVERB.INI to change from full octave analysis to 1/2 or 1/3 octave analysis. In such cases, the Calculate Acoustics command will cover a smaller total frequency range since only seven bands are ever analyzed and displayed at once. To cover a larger frequency range, execute the Calculate Acoustics command several times entering a higher starting band equal to 7 plus the previous starting band each time.

The eighth data column in the Calculate Acoustics screen presents the SPL-weighted average of each parameter over some specified range of octaves. The default range is the 500 Hz thru the 4 kHz octave band inclusive but you can edit REVERB.INI to change to another range. This column is normally used to display the mid-range average value for each parameter except S, N and SNR. All averages are weighted by the total SPL in each octave after accounting for the +3dB/octave bias of the MLS spectrum. Thus, for example, if the frequency response of the speaker/room system is perfectly flat, the octaves 500, 1k, 2k and 4k will all be given equal weight in computing the average parameter values across those octaves.

*MLSSA* detects the first arrival time relative to an estimate of the background noise. Knowing the exact first arrival time is critical to computing many of the parameters. You can program a different detection threshold by editing REVERB.INI. An entry of zero causes *MLSSA* to use the cursor position as the first arrival time. When this option is selected, all commands requiring the first arrival time will pause and request that you move the cursor manually to the first arrival position.

The detection algorithm can be fooled when there is insufficient gain in the microphone preamp. This can lead to the appearance of a small spike at time zero in the impulse response due to electrical crosstalk, which can be mistaken for the acoustical first arrival. Most condenser microphones require about 20 to 30 dB of gain for optimum performance with *MLSSA*. The optimum preamp gain is displayed when you first program the microphone information in the Library Microphones screen.

Normally, *MLSSA*'s Calculate Acoustics command will not analyze non-MLS data such as balloon pops or pistol shot data acquired in the Scope Acquisition mode. You can analyze such data by editing REVERB.INI and changing the corresponding mode from 0 to 1. In this mode, Calculate Acoustics will not attempt to noise correct the results if noise compensation is disabled through the View Schroeder-plot Compensation command in the time domain. This was done because noise compensation can easily become erroneous on non-MLS measurements that are contaminated by non-stationary noise. You can also use this mode to analyze MLS measurements both with and without noise compensation to see its effect on the parameters. In the normal mode (0), Calculate Acoustics always attempts to perform noise compensation regardless of whether it was selected in the time domain but it will refuse to analyze non-MLS measurements.

The Calculate Acoustics command as well as the Calculate Early/late and the Calculate Center-time commands, when these operate on filtered data, account for the average effective delay introduced by the 6th-order Butterworth IIR filter used to filter the wideband impulse response. This delay compensation affects only C50, D50, C80 and TS (Center time) in the Calculate Acoustics screen.

### 3.34 Lateral Energy Fraction

Lateral energy fraction is that proportion of the total sound reaching the listener that was reflected from the sidewalls of the hall. This parameter is also sometimes referred to as spatial impression since high values of lateral energy are associated with the listener feeling enveloped by the sound.

The Calculate Acoustics command computes two measures of lateral energy fraction, LF and LFC, based on two impulse response measurements: one taken using a figure-of-eight microphone which was stored as an overlay and, a second measurement taken using an omni directional microphone. Some microphones implement both polar characteristics (omni and figure-of-eight) as controlled by a switch. If you use two separate microphones, you need to remember to select a new microphone number in the Library Microphones screen when switching microphones to account for likely differences in either their sensitivity or their preamp's gain. Otherwise, the computed values of LF and LFC will be invalidated unless both microphones happen to have identical sensitivities and preamp gains. You also need to make sure the null in the figure-of-eight microphone is pointed toward the sound source for maximum rejection of the direct sound.

If the figure-of-eight overlay is present and plotted on the screen, Calculate Acoustics will compute and display LF and LFC as shown in the sample screen below. The other parameters are also computed and displayed but they are based on the omni directional measurement only. The figure-of-eight measurement must be stored as the overlay and the omni measurement as the main data. If they are reversed, LF and LFC will be computed incorrectly. You must remember to execute Overlay Plot before Calculate Acoustics to compute LF and LFC, otherwise the overlay data is ignored and LF and LFC are not computed.

LC and LFC are defined in ISO 3382 [13]. LF is the standard measure while LFC is a cosine-weighted according to Kleiner [14]. Below is a sample Calculate Acoustics screen when an overlay is plotted containing the figure-of-eight impulse response.

IEC 1/1-Octave Band Acoustical Parameters									
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000	
S [dB-SPL]	60.2	64.3	68.5	73.7	78.8	78.9	79.4	SPL- weighted Averages	
N [dB-SPL]	33.4	34.1	38.3	43.1	43.6	44.8	49.6		
SNR [dB]	26.8	30.2	30.3	30.6	35.1	34.1	29.7		
C50 [dB]	-6.85	-4.15	-1.69	-0.41	4.76	5.30	8.81		3.442
C80 [dB]	-0.51	-0.21	1.01	2.23	7.04	7.80	11.94		5.846
D50 [%]	17.1	27.8	40.4	47.7	75.0	77.2	88.4		63.439
TS [ms]	114.7	108.1	104.2	88.2	43.9	36.8	15.3		62.548
EDT-10dB [s]	1.390	1.630	1.670	1.487	0.939	0.864	0.939		1.182
RT-20dB [s]	1.614	1.308	1.293	1.296	1.270	1.198	0.861		1.265
(-5:-25) r	-0.996	-0.995	-0.999	-0.999	-0.999	-0.998	-0.996		-0.999
RT-30dB [s]	1.560	1.332	1.342	1.284	1.247	1.166	0.841		1.254
(-5:-35) r	-0.997	-0.998	-0.999	-1.000	-1.000	-0.999	-0.999		-1.000
RT-USER [s]	1.705	1.258	1.322	1.297	1.237	1.178	0.895		1.253
(-10:-25) r	-0.992	-0.993	-0.999	-0.999	-0.999	-0.998	-0.995		-0.999
LF [%]	72.6	50.4	61.0	60.9	41.4	33.3	27.2	44.052	
LFC [%]	72.6	50.4	61.0	60.9	41.4	33.3	27.2	44.052	

File: C:\VIEW\SAROP-2.TIM 4-18-95 1:12 PM  
SAROP center house  
ESC to exit or F1 to print. MLSSA: Acoustics

**Acoustical Parameters with LF and LFC**

### 3.35 Spatially-Averaged Acoustical Parameters

The Calculate Acoustics command is able to analyze only a single impulse response measurement at a time. The Library Acoustics command lets you analyze a set of up to 14 TIM files taken at different microphone and/or source positions to determine the spatially averaged acoustical parameters in accordance with ISO 3382 [13]. You can also display the acoustical parameters for each position (TIM file) individually as well as display how each parameter varies with position. Thus, the Library Acoustics command is really a superset of the Calculate Acoustics command except that only Calculate Acoustics is able to compute LF and LFC.

To select a set of files to analyze, execute Library Acoustics Select-files and enter the directory where the TIM files reside or just press the enter key if they reside in the current default directory. Pressing the Right arrow key or right mouse button displays the current set of autonamed files if they exist. Next, you mark all the files you want to analyze and press Enter to return to the Library Acoustics menu.

The Calculate command (all commands from here on assume you are starting from the Library Acoustics menu) analyzes each selected file, in turn, until all files have been analyzed. After completion, the most important spatially averaged parameters are displayed in what is called the Averages data screen shown in the figure below.

File: C:\ROOMS\TEST.ACO - Average of 4 TIM files								
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000
EDT-10min [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	
EDT-10mean [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	1.143
EDT-10max [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	
RT-20 min [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	
RT-20 mean [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	1.948
RT-20 max [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	
(-5:-25) warn	r N	T						
RT-30 min [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	
RT-30 mean [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	1.939
RT-30 max [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	
(-5:-35) warn	r N	T						
RT-U min [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	
RT-U mean [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	1.910
RT-U max [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	
(-10:-25)warn	r N	T						

Warnings: T= IR too short; r= r < 0.95; N= SNR < 10 dB or no noise correction.

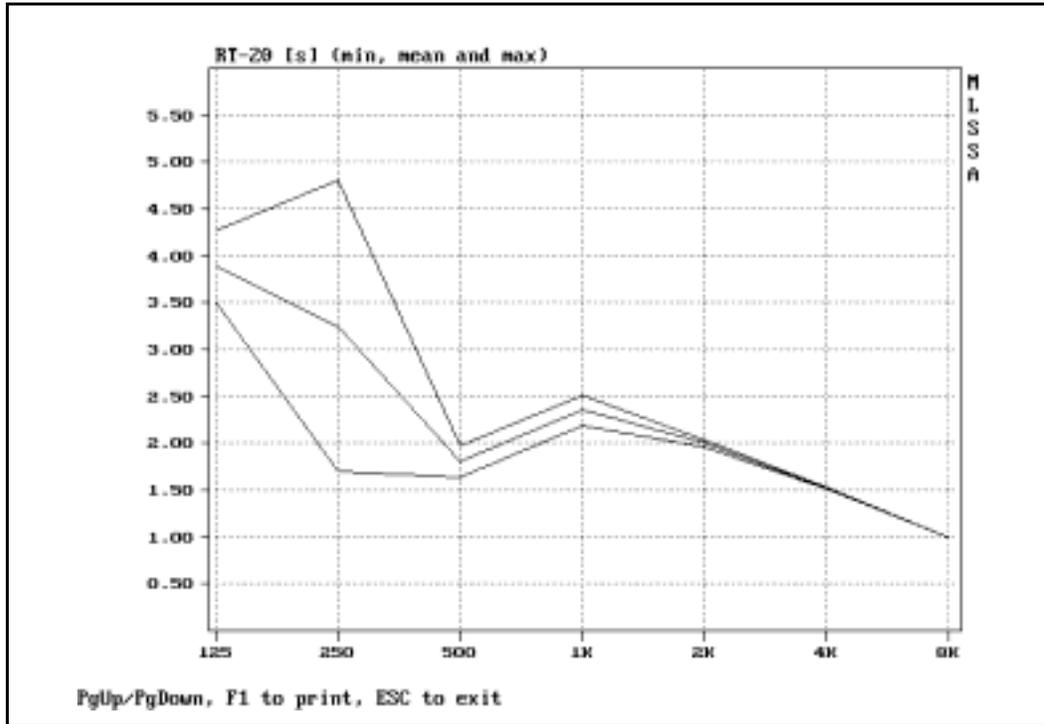
LIB ACOUSTICS: Calculate Select-files View Transfer Print Graph Options Free  
F1 for Help or ESC to exit MLSSA: Lib Acoustics

### Spatially-averaged Acoustical Parameters

The mean, minimum and maximum are displayed for parameters EDT-10dB, RT-20dB, RT-30dB and RT-USER. There are three possible warnings: 1) a "T" indicates that the impulse response of at least one of the TIM files was too short to perform a proper regression analysis over the specified decay range, 2) an "r" indicates that the absolute value of the correlation coefficient of at least one of the TIM files was smaller than 0.95 and, 3) an "N" indicates that on at least one of the TIM files it was not possible to perform the noise correction.

The spatially averaged RT parameters can also be graphed as shown in the next figure. You use the Graph command and then PgUp and PgDn to scroll through each spatially averaged RT parameter. You type Esc to return to the main Library Acoustics menu.

The graph's vertical scale can also be changed through the Options Graph-scaling command.



**Graph of spatially averaged RT-20dB vs. frequency**

Starting from the main spatially averaged data screen, you use PgDn to scroll through the acoustical parameter tables for each individual file. Each is similar to the table generated by the Calculate Acoustics command. After all files have been displayed, further PgDn key strokes scrolls through another set of data screens, one for each individual parameter, showing its mean value and standard deviation computed across all the analyzed TIM filenames which are listed in the first column of each parameter screen. These parameter screens are useful for observing how each parameter varies with microphone and/or source location. You can also jump to the start of any set of

data screens through the View Averages, View Files and View Parameters commands or just use the PgUp and PgDn keys if you wish.

The Options Starting-band command lets you change the lowest octave or starting band to be analyzed without editing REVERB.INI. After changing the starting band, you must execute the Calculate command again to recompute the parameters with the new starting band. The Options Bandwidth command lets you change from full octave analysis to 1/2 or 1/3 octave analysis. In such cases, the Calculate command will cover a smaller total frequency range since only seven bands are ever analyzed and displayed at once. The Options Decay-range command lets you change the decay range for the RT-USER parameter. Again, you must recompute the parameters for any change to take effect. The Options Graph-scaling has already been explained above.

The Transfer Save command saves all the post-processed parameter data to a binary data file having the default extension ".ACO". You can later load ACO files through the Transfer Load command without the need to recompute the data from the original TIM files.

The Library Acoustics command needs to allocate some memory to process all the files and store the results. If you need this memory for other purposes, you can free it through the Free command before returning to the time domain.

### 3.36 File Operations

*MLSSA* performs a variety of useful mathematical operations on both time (TIM) and frequency (FRQ) domain data files. For time-domain data files, use the Library Operations Time-files commands and for frequency-domain data files, use the Library Operations Frequency-files commands.

For time and frequency file operations, the output files will generally have the same header information (using the Info command) as the first input file specified. The precursor field in the Info command will display the particular operation that generated the output file. The size of the output file will equal the size of the largest input file.

Input and output filenames can also overlap. For example, you can square a file by multiplying by itself and overwriting the same file with the results.

The available frequency-domain file operations are:

- Average
- Sum
- Difference
- Product
- Ratio
- Equalize
- Conjugate
- sMooth
- reciprOcal
- spLice
- IFT
- QC-statistics

The Library Operations Frequency-files Difference command, for instance, can be used to compute an error curve that can subsequently be loaded and displayed. Note, however, that to create an error curve expressed in decibels you must use the Library Operations Frequency-files Ratio command.

Except for the spLice operation, the input files must have the same frequency spacing. They must also be of the same type (e.g. Impedance, Transfer-Function etc.) except for the Equalize operation, which requires that the second file be of the Transfer-Function type. The QC-statistics operation is a specialized function covered in detail in the "QC PASS/FAIL Functions" section below.

Most operations can be performed on either a complex or a power basis. The Complex option performs the indicated operation using the full complex frequency-domain data including both magnitude and phase. The Power option performs the indicated operation on the squared magnitude of the complex data meaning all phase information

is lost in the output file. The Power option should be used with the Average operation, for instance, when spatially averaging frequency response data.

The Frequency-files Average Weighted command allows you to perform a weighted average on an unlimited set of frequency files while specifying a separate weighting factor for each one. It is used by the SNDPWR.MAC sound power macro, for example. This command can also be used to form the weighted average of several low frequency near-field loudspeaker measurements to determine the composite low frequency response. In that application, the weighting factors would simply be the diameters of the respective drivers, ports or passive radiators involved.

When using the Average commands or any command requesting multiple filenames, you are prompted to enter a DOS path that points to where the source files reside instead of entering a particular filename as is the case for most other commands such as Transfer Load. Pressing the Enter key alone specifies the current default directory path. If you do enter a directory name, remember to precede it with a backslash according to DOS convention. You can also delimit the number of filenames displayed by entering the first few characters of the filenames you want to average. For example, if the files you want to average reside in C:\MLS\DATA and they all start with JOB2A then you would enter C:\MLS\DATA\JOB2A followed by the Enter key. In that case, the command will display only those files residing in C:\MLS\DATA that start with JOB2A. Another alternative is to list only the current autonamed data files by pressing the Right arrow key. In that case, the path and files prefix are those specified by the Transfer Autaname commands. In any event, once you are presented with a list of filenames you must mark those you want to average. To mark all files press Ctrl-Space or, use the Space and the arrow keys or mouse to mark only selected files. After marking all the files you want to average, press the Ctrl-Enter key combination or the right mouse button to continue.

You can substitute any real or a complex constant for the second filename in two-file operations. Constants are distinguished from filenames by enclosing them in square brackets. For instance, the real number 0.5 would be entered as [0.5] while the complex number (1 - 0.25j) would be entered as [1, -0.25]. For example, you can easily scale an FRQ file down by one half by executing Library Operations Frequency-files

Ratio Complex and entering the real constant [2] in place of the second filename. This constant is interpreted as the complex number  $(2 + 0j)$ .

The IFT operation takes the inverse FFT of the indicated frequency-domain file and creates a *MLSSA* time-domain output file. If the frequency-domain input file is a Power-Spectrum, PSD or Response-Function type, the output file will be the autocorrelation function of the original data. The autocorrelation function is an important statistical measure of random data. Otherwise, the output file will be a windowed version of the original time-domain data. The IFT operation will only work when the original FFT was an even power of two (e.g. 1024 not 1000 points).

The *spLice* operation is used to smoothly join together two different frequency domain measurements at some specified frequency. This command is intended to splice together a high-frequency anechoic loudspeaker measurement to a low-frequency near-field measurement of the same loudspeaker. Typically, the two input FRQ files will differ in frequency spacing, phase delay as well as in absolute levels. The *spLice* operation interpolates the data as required to get uniform frequency spacing and then automatically adjusts the level and phase delay of the low frequency data to match that of the high frequency data. The resulting output file can then be loaded and displayed in the frequency domain through the Transfer Load or Transfer Import commands.

If you press the Enter key in response to the splice frequency, *MLSSA* automatically splices the data at the lowest reliable frequency of the HF file as determined by the time-frequency uncertainty principle. The phase is also spliced with appropriate phase delay adjustments made to the low frequency file data in order to join the two phase curves together smoothly.

Note that in order to minimize the output file size, the *spLice* operation normally selects the smallest possible output FFT size that will still give adequate frequency resolution down to 20 Hz. If you want to maximize the output file frequency resolution and hence its FFT size, enter the splice frequency manually as a negative number.

The available time-domain file operations are:

Average  
Sum  
Difference  
Product  
Convolution  
X-correlation

The time-domain input files must all have been measured using the same sampling rate.

The X-correlation operation performs cross-correlation.

You can substitute any constant for the second filename in two-file TIM file operations. Constants are distinguished from filenames by enclosing them in square brackets. For instance, the real number 0.5 would be entered as [0.5]. For example, you can easily scale a TIM file down by one half by executing Library Operations Time-files Product and entering the constant [0.5] in place of the second filename.

### 3.37 File Browser and Filebox Features

The *MLSSA* Directory feature displays a complete list of available files and allows you to browse and select a particular file, sub-directory or drive simply by moving a cursor using the cursor keys or the mouse. This feature is available to all *MLSSA* commands which prompt you for an input filename including Transfer Load, Setup Load and the Library Operations commands by pressing the right arrow key or the right mouse button. You can also direct *MLSSA* to display any directory or sub-directory on your disk and delimit the number of filenames displayed. For example, if you execute Transfer Load from the time domain and enter C:\MLS\DATA\JOB1 followed by the right arrow key or the right mouse button, *MLSSA* will display only those files in the C:\MLS\DATA sub-directory that begin with "JOB1" and have the extension "TIM". Thus, there is no need to enter wildcards (\* or ?) in most cases but wildcards can be entered if desired.

Drives and directories are enclosed in square brackets to distinguish them from normal files. To enter a directory, move to it and press Enter or the left mouse button. Once in a new directory you will see a new list of files and a list of sub-directories, if they exist. Click the [up arrow] icon to move to the beginning of the list of available files in the current directory. Click the [down arrow] icon to move to the beginning of the list of available sub-directories and drives. Clicking the [..] icon moves you to the parent directory. Clicking a drive letter moves you to the root directory for that drive. From there, you can move to any sub-directory on that drive. In this way, you can reach and load any file on your system.

Note that the [.] icon refers to the current directory. If you click on it when *MLSSA* is prompting you to enter a filename, this action is interpreted as an Esc press and returns you to the original filename prompt. However, if *MLSSA* is prompting you for a DOS path such as is required by the Transfer Autaname Path command or the DOS change-Directory command, clicking the [.] icon or any file selects the current drive and directory as the path you want to enter.

Pressing F2 or the 'i' key while a *MLSSA* data file or setup file is highlighted, causes a screen to appear displaying key information about the data in that file. This screen is the same as what the Info command would display if the file were already loaded. In fact, *MLSSA* is just peeking into the highlighted file to let you decide if you want to use it or not. The comment, if any, contained in the file is also displayed. This command is useful for finding a specific setup file (SET), time domain data file (TIM) or frequency domain data file (FRQ) when you cannot remember the filename. Highlighted text files with extensions TXT, LOG, PQC, BAT and HLP can be displayed and browsed by pressing F2.

Some *MLSSA* commands (e.g. Library Operations Frequency-files Average commands) expect multiple input filenames. To accommodate these commands, the *MLSSA* Filebox feature allows you to mark as many or as few files as desired. You mark or unmark a file simply by moving the cursor over it and pressing Spacebar or, by pressing the left mouse button. You can also mark or unmark all files by pressing Ctrl-Spacebar. Clicking the [.] icon also marks or unmarks all the listed files in the current directory.

You press the Ctrl-Enter key combination or the right mouse button to finish a file selection session.

The DOS change-Directory command is used to change the default directory from within *MLSSA*. This command is used to redirect all Transfer commands to the new directory without the need to specify it explicitly. *MLSSA* always returns you upon exit to the default directory you were in when you originally started *MLSSA*.

Note the default directory for setup files, macro files and QC-limits files is always the *MLSSA* program directory regardless of the default DOS directory as set by the DOS change-Directory command. This is to allow you to run the same macros to process different data files stored in different directories simply by changing directories. When a macro loads a setup file, for instance, it will assume it resides in the *MLSSA* program directory but when it goes to fetch a data file (such as a TIM or FRQ file) it will assume it resides in the default DOS directory as set by the DOS change-Directory command. Thus, there is no need to copy your macro, setup or QC-limits files to where the data files are located or to explicitly enter a path to them. For the same reason the default directory for the Macro Command Run, Macro Command Call, Macro Command Exec and Macro Command Load-setup commands is also the *MLSSA* program directory regardless of the default DOS directory. Of course, any default or assumed directory can always be overridden by entering a full DOS path to the target file.

You can also override these defaults on a case-by-case basis by entering the characters “.\ “ before a setup, macro or QC-limits filename. In that case, *MLSSA* will look for the file in the default DOS directory. For example, when executing Setup Load and entering .\LOUDS, *MLSSA* will look for setup file LOUDS.SET in the default DOS directory.

In all *MLSSA* versions prior to 10W, all files including setup and macro files were presumed to reside in the default DOS directory. *MLSSA* users who wrote macros under the older *MLSSA* versions and would like to run them unmodified under later versions can force *MLSSA* to use the older convention. To do so, edit text file *MLSSA*.INI located in the *MLSSA* program directory and add a minus sign before the entry for the “Default macro buffer size in keystrokes”. Upon re-starting *MLSSA*, all

files, data, setup and macro will be loaded from the default DOS directory unless an explicit path is given.

### 3.38 Executing DOS Commands

The DOS Shell command frees up nearly all of conventional memory then starts a copy of the DOS command processor COMMAND.COM so you can execute normal DOS commands and run other programs without quitting *MLSSA*. To return to *MLSSA*, just type EXIT at the DOS prompt. You can also use DOS Shell to start a second copy of *MLSSA* without destroying the state of the current copy. After quitting the second copy and typing EXIT, your original copy of *MLSSA* will be restored to exactly the same state with exactly the same data as when you originally executed DOS Shell. The old copy of *MLSSA* and its data is stored in extended memory if available. Otherwise, it is written out to a temporary file on your hard disk. Therefore, this command may be a bit slow if there is no extended memory available on your computer.

The DOS MLSSADATA-Shell also shells out to DOS but the *MLSSA* program and its data are retained in conventional memory leaving much less memory to run other applications as compared to the DOS Shell command. The advantage of this command over DOS Shell is that other programs can access *MLSSA*'s measurement data directly from memory. See Appendix D for details.

You can also execute DOS commands or run DOS batch files from a *MLSSA* macro through the Macro Command Run and Macro Command Exec commands. See the previous section titled "Creating and Running Macros" for details.

The DOS Unlink command provides a means of deleting files without shelling out to DOS. By pressing the right arrow key or the right mouse button from the frequency domain you can display and mark any number of FRQ files to delete. Executing this command from the time domain displays TIM files instead. You can enter any file extension you wish however in order to delete other types of files. For safety's sake, *MLSSA* prompts you to verify each file before deleting it.

### 3.39 QC PASS/FAIL Functions

In the frequency domain, *MLSSA* provides for comprehensive automated QC PASS/FAIL testing. Upper and lower QC limits curves are defined in a special text file having the following format.

```
"Example QC Limits File (Title)"
  "Hz"      "dB"      "dB"      "dB" "Abs offset" "Polarity" "Lines"
892:1121, -1.0,  +1.0,      -60    1,      -2,      0
10,      -2,      +2,      -60
1000,    -2,      +2,      -60
10000,   -2,      +2,      -60
15000,   -2,      +2,      -60
20000,   -5,      +5,      -60
```

This QC limits example file is named QCTEST.TXT and resides in your *MLSSA* directory. The first line contains the title string, which must be enclosed in double quotes. Then you can enter as many comments as you wish as long as they are enclosed in double quotes and provided there are no intervening empty lines.

**IMPORTANT:**

When creating a QC limits text file using a text editor or word processor be sure to add at least 1 leading space at the beginning of each line, that is, before entering any numerical data. If you are using a word processor also be sure you save the file as a plain text file, that is, without special word-processor formatting characters.

The first column of the first data line in the QC limits file defines either a specific frequency in Hz or, a range of frequencies to average. To specify a frequency range, you enter two numbers separated by a colon as shown. In the example file, this range defines a band 1/3-octave wide surrounding 1 kHz, which corresponds to some loudspeaker sensitivity specifications. When a range is specified, *MLSSA* computes the pink-weighted RMS value from the actual data falling in that range. (Note that the power average computed is here is the same given by the Calculate Level command when the marker and cursor are set to select the same frequency range.) By entering a single number, without a colon, you can check just a single data point.

The next two entries in the first line specify the lower and upper QC offset limits. These limits are interpreted according to the fourth entry labeled "Absolute offset check". If set to 1, *MLSSA* will first compute the RMS value of the data falling in the specified range (892 to 1121 in this case) and compare that result directly to the offset limits given in the 2nd and 3rd entries. If these limits are exceeded, a QC FAILURE message is immediately displayed and no other QC checks are performed. If the RMS value is within these offset limits, what happens next depends upon whether you have selected floating or fixed QC limits curves through the QC-limits Offset command. If you had selected the Floating option, the computed RMS value is next added to both the upper and lower QC limits curves specified by the remaining lines in the file in the second and third columns. These two offset QC limit curves are next plotted on the screen and numerically compared to the measured curve falling between the marker and cursor. Thus the QC limits curves will float up and down within the range defined by the offset limits given in the 2nd and 3rd columns of the first data line, to accommodate different absolute signal levels. However, if you had selected the Set option in the QC-limits Offset command and entered a zero (0), the QC limits curves will not be able to float up and down, that is, they will be plotted and used literally on an absolute basis.

Entering a zero (0) in the "absolute offset check" column disables the absolute limits checking option. In that case, *MLSSA* no longer explicitly checks the computed RMS value against the offset limits specified in the second and third columns of the first data line. Instead, assuming the Floating option was selected in the QC-limits Offset command, it first clips the computed RMS value according to the offset limits and then adds the clipped value to both the lower and upper QC limits curves. Thus, with absolute offset checking disabled, the offset limits given in the second and third columns of the first data line are checked only indirectly when the measured curve is compared to the floating limits curves.

In the case where absolute offset checking is disabled and the Floating offset mode is disabled, *MLSSA* totally ignores the RMS value of the data as well as the upper and lower offset values, that is, it ignores the first, 2nd and 3rd entries of the first data line and simply compares the measured curve to the QC limit curves. In this case, you can

manually offset both limit curves by some fixed amount by entering a nonzero value in the QC-limits Offset Set command.

You can also instruct *MLSSA* to check the polarity of the time domain response by entering a nonzero value in the fifth column of the first data line labeled "Polarity check". Both the impulse response (IR) or the step response, or any combination, can be checked as shown in the following table.

Polarity checking codes:

0	no polarity check
+1	check IR only for positive polarity
+2	check step response only for positive polarity
+3	check both IR & step response for positive polarity
+4	check for positive IR but negative step response polarity
-1	check IR only for negative polarity
-2	check step response only for negative polarity
-3	check both IR & step response for negative polarity
-4	check for negative IR but positive step response polarity

Since the IR polarity depends primarily on the tweeter output in multi-driver loudspeaker systems while the step response polarity depends primarily upon the woofer output, you can choose the appropriate combination for your particular loudspeaker design.

The last parameter in the sixth column of the first data line is the "line interpolation" parameter. Entering a '1' for this parameter causes *MLSSA* to use straight-line interpolation between the data points for the limit curves rather than the normal cubic spline interpolation. These lines appear straight when displayed on the current frequency scale whether log or linear. Note, however, that loading a QC file when displaying a linear scale and switching (F4) to a log scale later will cause the same lines to appear curved and vice versa. This is because *MLSSA* interpolates the data differently depending on the frequency scale in effect at the time the QC Load-limits command is executed. This feature is useful for creating simple textbook style QC limits curves with a minimum number of data points. Note, however, that it is very difficult to create smooth curves using straight-line interpolation in which case you should enter a "0" this parameter to go back to cubic spline interpolation.

The remaining data lines in the QC-limits file define the lower limit curve (2nd column), upper limit curve (3rd column) and an optional incoherency upper limit curve (4th column). The incoherency column instructs *MLSSA* to check incoherency distortion levels after testing the upper and lower response levels. There is no lower limit on incoherency. If you omit the last column, incoherency will no longer be checked during QC operations. Note also that you cannot use incoherency testing in typical reverberant environments. Incoherency measurements are valid only for true anechoic or near-field measurements.

Once you have created a QC limits file, you load it from the frequency domain through the QC-limits Load command. The QC-limits Compare command initiates a QC PASS/FAIL operation in addition to actually plotting both the upper and lower QC limits curves. The comparison is restricted to the frequency range defined by the current cursor and marker positions. If there is a failure, *MLSSA* beeps and a message appears indicating what caused the failure. Codes associated with each possible failure mode are listed below.

Error codes for QC failures:

```
"QC FAILURE: 1, Exceeded upper offset limit"  
"QC FAILURE: 2, Exceeded lower offset limit"  
"QC FAILURE: 3, Crossed upper limit at ?? Hz"  
"QC FAILURE: 4, Crossed lower limit at ?? Hz"  
"QC FAILURE: 5, Incoherency exceeded limit at ?? Hz"  
"QC FAILURE: 6, Incoherency exceeded limit at ?? Hz"  
"QC FAILURE: 7, +Polarity (IR)"  
"QC FAILURE: 8, +Polarity (Step Response)"  
"QC FAILURE: 9, +Polarity (IR & Step Response)"  
"QC FAILURE: 10, +Polarity (IR) & -Polarity (Step Response)"  
"QC FAILURE: 11, -Polarity (IR)"  
"QC FAILURE: 12, -Polarity (Step Response)"  
"QC FAILURE: 13, -Polarity (IR & Step Response)"  
"QC FAILURE: 14, -Polarity (IR) & +Polarity (Step Response)"
```

If there is no failure and you have defined the upper incoherency limit column, *MLSSA* goes on to automatically compute incoherency and compare it to the upper incoherency limit. If all checks pass the message "QC PASSED!" appears.

In creating the QC lower and upper limits curves be sure you specify enough data points to accurately describe the curves you want. *MLSSA* normally uses cubic spline interpolation to draw a very smooth curve between these points. Using too few points, however, can result in wild swings in the interpolation. A general rule is to specify more points near sharp changes in the slope of the desired limit curves. In those regions where the desired curves are of a constant slope, very few points are needed to describe them. The last parameter in the sixth column of the first data line is the "line interpolation" parameter. Entering a '1' for this parameter causes *MLSSA* to use straight-line interpolation between the data points for the limit curves rather than the normal cubic spline interpolation, thus avoiding any overshoot or ringing problems

Sometimes, it may be necessary to re-edit the QC limits file several times either to add or remove data points or, to make minor adjustments to various entries in order to achieve the QC functionality you want as well as the desired upper and lower QC limit curves. To speed this process, from Windows, minimize *MLSSA* by pressing Alt-Space-bar then start Notepad (Programs Accessories Notepad) to load and edit the QC limits file in question. After editing, click File then Save to save it to disk but do not exit Notepad yet. Next, click on *MLSSA* to maximize, then execute QC-limits Load and enter QC filename in question or, if you had previously loaded it, simply press the Up arrow and Enter keys to re-load the modified QC limits file without having to retype its filename. Next, execute the QC-limits Compare command to see if your changes solved any ringing, overshoot or other problem. If not, simply minimize *MLSSA* once again, go back to the Notepad editor, make further changes and save again. Repeat as many times as needed until the functionality and displayed limit curves are what you want, then exit Notepad. When running under DOS, the procedure is a little more involved: Execute the DOS Shell command and load the QC limits file in question into a DOS text editor such as EDIT to perform any necessary adjustments. Next, exit the text editor with changes saved, then type Exit and press Enter from DOS to resume *MLSSA*. Execute the QC-limits Load command followed by the Up arrow and Enter keys to re-load the modified QC limits file. Execute QC-limits Compare to see if the limit curves and the QC functionality are what you expected. If not, repeat the whole procedure.

The QC-limits Go commands initiate measurements followed automatically by the QC-limits Compare command.

When graphics display blanking is enabled (Ctrl-F3), the final QC PASS/FAIL result is displayed in larger type for easier visibility from a distance. In addition, for even greater visibility, the "QC PASSED!" message is framed by a broad green bar while a "QC FAILED" message is framed by a broad red bar. To restore the normal graphics display press Ctrl-F3 a second time.

Even though *MLSSA* will optionally compute and check incoherency, it and its upper limit curve are not normally displayed. To actually plot incoherency and see its upper limit curve you must return to the time domain and select the Incoherency FFT mode. Next, enter FFT Execute to return to the frequency domain. Now if you execute QC-limits Compare, *MLSSA* will compare and plot the incoherency limit only. This option intended only for debugging purposes.

Incoherency failure codes 5 and 6 differ only in when they can occur. Code 6 occurs when incoherency is checked invisibly (normally) while code 5 occurs when incoherency is checked while being actually displayed on the screen (debug mode).

When checking incoherency during QC Compare, *MLSSA* will smooth it prior to checking against the limit curve if the magnitude was smoothed. For example, if you display the magnitude response smoothed to 0.1 octave, *MLSSA* automatically smoothes the incoherency to 0.1 octave as well prior to checking it against the incoherency QC limit curve.

You change the color used to plot the QC limits curves through the Library Colors Main QC-plot command. Entering a negative color value causes the QC limits to be plotted as solid lines instead of dotted lines.

Macros can be written to perform sophisticated QC operations. For example, a macro can be used to save each measurement to an FRQ file and record the QC PASS/FAIL result and error code as the file comment. You do this by pressing the Right arrow key when prompted by the Transfer Save command to enter a comment. The Right arrow

key causes the current message displayed in the message window to be used as the file comment. In addition, if you use *MLSSA*'s autonamed file system to save the data and had previously opened a log file, all the QC PASS/FAIL information including the error codes will also be written to the log file. The log file, in turn, can be scanned later by another program to determine the reason for any failure.

If all the QC measurement data is stored in a set of FRQ files and, all files are located in the same directory (such as a set of autonamed files), it is possible to calculate some basic statistics on these data files through the Library Operations Frequency-files QC-statistics command. This command prompts you to mark the set of files to be analyzed and prompts you for the root name of the output filenames. The command then appends a special 2-character identifier to the specified output root name to generate four distinct output filenames. These are the mean or average file computed on a power basis created by appending ^M to the root, the minimum file created by appending ^L to the root, the maximum file created by appending ^H and, the standard deviation file created by appending ^S to the root name. If an output filename were to exceed six characters in length, it is first truncated to six characters before appending these special identifiers. For example, if the output root name you entered was QCSTATS, the actual output filenames will be QCSTAT^M.FRQ, QCSTAT^L.FRQ, QCSTAT^H.FRQ and QCSTAT^S.FRQ. These statistics files, in turn, can be loaded into the frequency domain to plot them.

The mean response file can be loaded as the Aux reference through the Reference Aux Load command in which case *MLSSA* will display the difference, in decibels, between the mean response and the actual response of the driver under test. Furthermore, all QC Compare operations will now operate on this difference curve rather than the raw driver response curve. When using the mean file as the reference in this way you can easily generate upper and lower limit curves based on some multiple of the standard deviation file. For example, assume you want to pass every driver within +/- 2 standard deviations from the mean. To generate the appropriate upper limit curve, multiply the ^S (standard deviation) file by a constant of [2.0] using the Library Operations Frequency-files Product Power command and save the result to another file called say STD\_2.FRQ. Next, load this file into the frequency domain through the Transfer Load

command and then export it to a text file through the Transfer Export View Sixth-octaves command. Once in text format, use a text editor such as Windows program Notepad or, the DOS program EDIT, to copy and paste these data in order to create the upper limit in your QC file. The negative of these data can then be used to form the lower limit curve. With this setup, QC Compare will pass every driver that falls within +/- 2 standard deviations of the mean response.

There are also macro commands specifically related to QC operations. The Macro Command IF-PASSED-QC Execute command will chain to another macro only when the last QC Compare operation resulted in a PASS. The Macro Command IF-FAILED-QC Execute command will chain to another macro only when the last QC Compare operation resulted in a FAILURE. The Macro Command IF-PASSED-QC Signal command and Macro Command IF-FAILED-QC Signal command cause a positive-going TTL pulse of 10 msec duration to be output on J4 pin-2 during macro execution according to the last PASS/FAIL result. The Macro Command IF-FAILED-QC Alarm command causes the executing macro to terminate and a continuous beep to commence until an operator presses Esc.

### **3.40 Running Multiple Sessions under Windows 98SE**

You can start a second or third copy of the *MLSSA* software under Windows 95/98. Each copy of can then be set up to make different kinds of measurements. For example, one copy can be configured to make impedance measurements while the second is configured to make frequency response measurements. You switch between the different copies using the Windows Alt-Tab task switching shortcut: hold down the Alt key then press Tab successively to switch from one task to the next. Release both keys to resume the selected task.

Each copy of the *MLSSA* software must share the same card. Therefore, at startup each copy notes the current time within 2 seconds resolution and stores a copy of that time on the *MLSSA* card. Thereafter, when you execute any command that requires the use of the card, such as the Go Once command or any Acquisition command, *MLSSA* first checks with the card to see if the correct startup time is still stored on it. If not, the current copy assumes that another copy with a different startup time has

reprogrammed the card. The current copy then automatically and completely reprograms the card according to its setup parameters before proceeding to execute the command. Reprogramming the card takes only a small fraction of a second and is signaled by a short high-pitched beep.

Do not attempt to start a second copy of *MLSSA* while another copy is executing a macro that is performing measurements. In such cases, the two copies of *MLSSA* will fight over the card with unpredictable and usually unpleasant results. For the same reason, do not start a second copy until after the first has completely initialized itself. You can safely have one copy of *MLSSA* run a macro and then start a second copy in Analysis mode. Analysis mode lets you load data files and perform analysis such as waterfall plots but you cannot make any measurements because *MLSSA* is prevented from reprogramming the card. Analysis mode is enabled by the /a switch when you start *MLSSA*.

Several different *MLSSA* shortcuts can be created on the Windows desktop each having a different name and specifying different command line options and switches for performing different kinds of measurements. For example, a new shortcut named "MLSSA Analysis" can be created that starts *MLSSA* with the /a switch. To do this point to the first shortcut on your Windows desktop and while holding down the right mouse button drag the copy to another location and release, then click Copy-here. Click on the name of the new shortcut and edit it to read "MLSSA Analysis" and then click on the desktop to finish. You will also need to change the window name and start-up command line. To do this right-click on the newly created shortcut, click Properties and then click the Program tab. Enter "MLSSA Analysis" as the window name, which is the first field in the Programs dialog box. Next edit the Cmd-line field by appending "/a" followed by "/nAnalysis" to the command line, for example,

```
C:\MLS\MLSSA.EXE room /a /nAnalysis
```

then click Apply and then OK to finish. The /a switch starts *MLSSA* in Analysis mode while the /n or name switch causes *MLSSA* to display the word "Analysis" in the upper left-hand corner of the *MLSSA* screen. The /n switch is handy when you have several

copies of *MLSSA* open and are constantly switching between them. By giving each copy a unique name, you'll never be confused as to which is which.

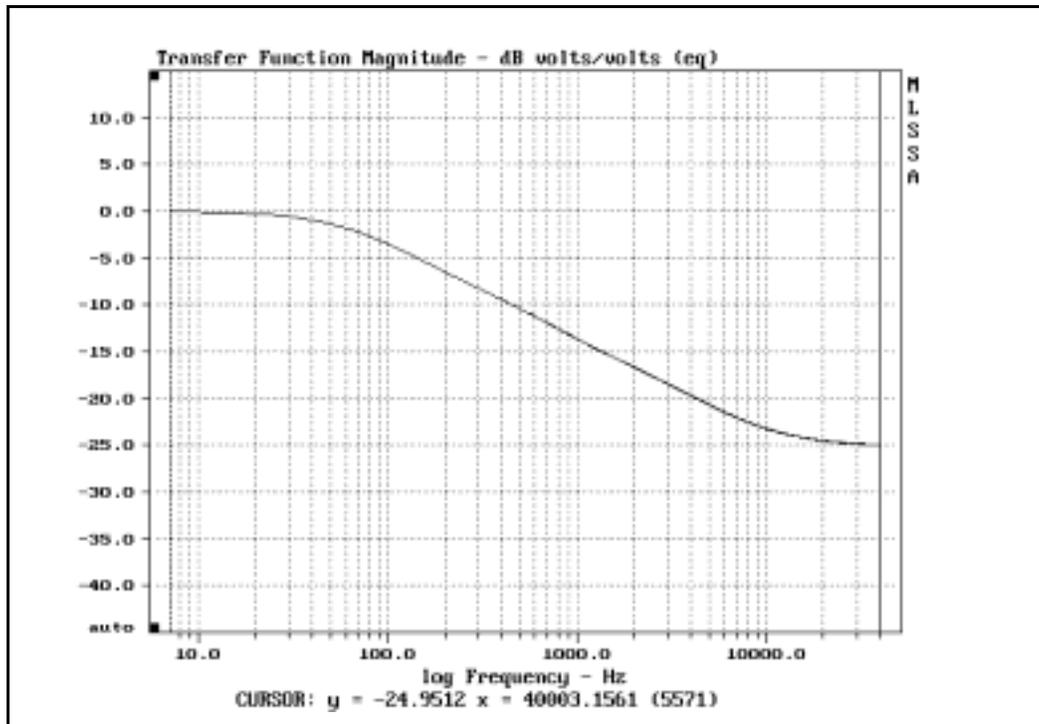
Three other command line switches facilitate using Windows shortcuts to *MLSSA*. The /c switch starts *MLSSA* in Character mode. The /m switch specifies a macro file to commence executing as soon as *MLSSA* starts. The /v switch starts *MLSSA* in the specified video graphics mode: /v0 = CGA, /v1 = EGA and /v2 = VGA. For example the Cmd-line

```
C:\MLS\MLSSA.EXE Louds /c /mAMACRO "/nQC Macro"
```

starts *MLSSA* in Character mode with setup file LOUDS.SET, runs the macro AMACRO.MAC and displays the name "QC Macro" in the upper-left hand corner of the screen. Note that the /n switch is enclosed in double quotes in this case to cause Windows to interpret "/nQC Macro" as a single command line argument. The double quotes are not required when you specify a name that does not include any spaces. A name specified using the /n switch cannot exceed 10 characters in length including any spaces but not including the switch characters "/n". In case you forget how to use any of these switches start *MLSSA* with the /? switch for help.

### 3.41 Pink-filtered MLS Output – Characteristics and Uses

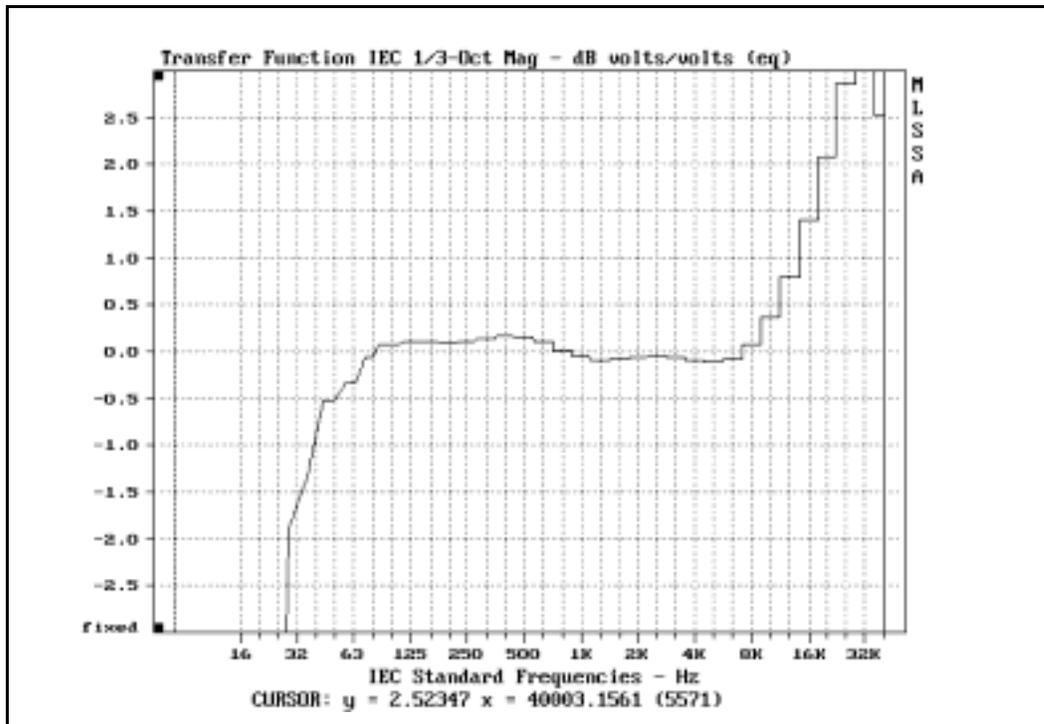
*MLSSA 2000* cards allow you to choose between a normal or white-MLS, having a flat spectrum, output on J3 or, a quasi pink-filtered MLS output on J1. The figure below shows the frequency response of the precision on-board quasi-pink filter. Note that the response is truly pink, at  $-3$  dB/octave, only in the center of the audio band between 100 Hz and 10 kHz. Below 100 Hz, the response flattens out. This is to prevent overdriving woofers unable to handle large bass signals. Above 10 kHz, the curve also flattens out. This is to insure that there is sufficient signal to prevent degrading the signal-to-noise ratio in the ultrasonic region. Overall, the filter provides 25 dB of bass pre-emphasis.



**MLSSA 2000 card quasi-pink filter frequency response**

Typical rooms exhibit much more noise in the bass region as compared to the treble region. This quasi pink filter frequency response was chosen to optimize the signal-to-noise ratio of MLS measurements over the entire frequency range to 40 kHz while providing a true pink-noise MLS signal in the critical center portion of the audio band as required when making room acoustics measurements according to ISO 3382.

The figure below shows the filter's frequency response integrated in 1/3-octave bands. A true pink filter would show a flat result over the entire frequency range. The figure shows that *MLSSA's* quasi-pink filter is truly pink within +/- 0.2 dB from 80 Hz to 8 kHz and within +/- 0.5 dB from the 50 Hz to 10 kHz.

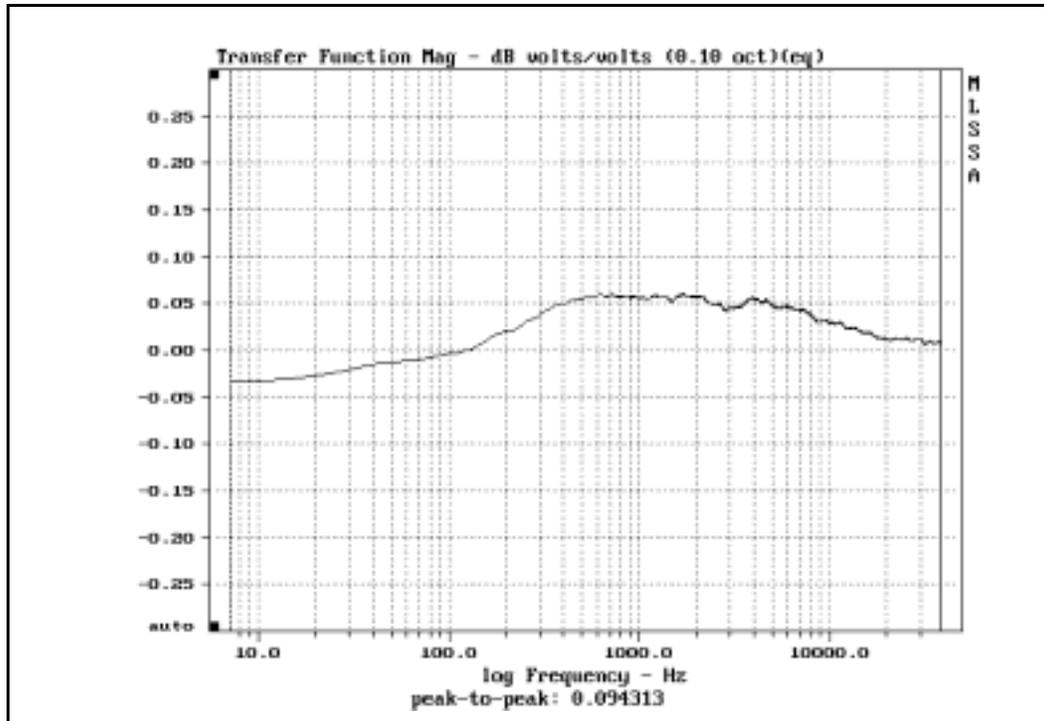


**MLSSA 2000 card quasi-pink filter, deviation from true pink response**

The *MLSSA* software includes a precision high-speed digital filter with a transfer function (including both frequency and phase response) that is the exact inverse of the transfer function of the quasi-pink filter. Thus, the digital equalization filter corrects both the phase and amplitude response of the quasi-pink filter. You enable this digital inverse filter through the Acquisition Equalization Pink-filter command. You disable it through the Acquisition Equalization Disable command. The state of this selection is stored in setup files so you can create different setup files that either enable or disable this equalization feature to suit different applications.

The equalization error is less than  $\pm 0.1$  dB, that is, the frequency response of the inverse filter is matched to the quasi-pink filter on the *MLSSA 2000* card within  $\pm 0.1$  dB from DC to 40 kHz. The equalization phase error is less than  $\pm 1$  degree from DC

to 30 kHz. The figure below shows the typical equalization error, which is usually much less than  $\pm 0.1$  dB. The correction is applied directly, in the time domain, to the measured impulse response before display and post-processing. Therefore, the option to use the quasi pink-filtered MLS is virtually transparent to the normal operation of *MLSSA*.



***MLSSA 2000* card quasi-pink filter, typical error after digital equalization**

The quasi pink-filtered MLS is output on J1 (top phono connector) while the unfiltered or white-MLS is output on J3 (bottom phono connector). Both outputs are always active. The quasi pink-filtered MLS is recommended for making in-room frequency response and room acoustical measurements to improve the signal-to-noise ratio. The quasi pink-filtered MLS is also advantageous for making loudspeaker frequency response measurements in noisy rooms. Use the quasi pink-filtered MLS when measuring

loudspeakers at high power levels to avoid tweeter damage. However, impedance measurements are usually best made using the white-MLS.

The quasi pink-filtered MLS has a lower slew rate and lower high frequency content than the normal MLS and is therefore less likely to produce slew rate limiting in power amplifiers. The quasi pink-filtered MLS also has roughly a Gaussian PDF or amplitude distribution. An MLS test signal with a Gaussian PDF is more tolerant of nonlinear distortions in the system under test than an unfiltered or white MLS. Therefore, when measuring systems with significant distortion levels, especially at high power levels, the quasi pink-filtered MLS is preferable.

If you do not own a *MLSSA 2000* card you should leave J1 unconnected at all times. A *MLSSA 2000 Mothercard Upgrade Kit* is available to upgrade older *MLSSA* cards to the latest version. *MLSSA 2000* cards are easily identified by a copyright date of 2000 etched on the back of the card. Older *MLSSA* cards have earlier copyright dates.

### **3.42 Editing Filenames and Comments**

*MLSSA* allows you to edit text data using the Left and Right arrow keys as well as the Del, Ins, Home and End keys. You can edit newly entered text to correct typing errors or edit a previously entered filename or comment to create a new one. You can edit filenames, comments and most other strings with the exception of whole numbers and real numbers. In the later cases you can only edit using the Backspace key and/or Ctrl-Backspace which deletes the whole line.

You begin editing a line by pressing the Left arrow key one or more times to move the cursor to the point in the line where you wish to make changes. The Left and Right arrow keys move the text cursor, which is indicated by the highlighted character. Typing normally inserts new text before the highlighted character. You can turn off insert mode by pressing the Ins key. When insert mode is off, new text overwrites the highlighted character and characters to the right. Pressing the Ins key toggles insert mode on and off. Insert mode is the default because it is the most commonly used. The Del key deletes the currently highlighted character as well as the space that it filled.

The Home key moves the text cursor to the first character on the line. The End key moves the cursor one space beyond the last character on the line, ready for you to append more characters. Note that pressing the Enter key at any time enters the whole line seen on the screen even if the text cursor is somewhere in the middle of the line. The Backspace key deletes the character immediately to the left of the text cursor. The Down arrow key normally deletes the entire line, regardless of the text cursor position. It also has an alternate function when entering comments. See next section for details. In most cases, the Up arrow key recalls the last entry you made for a particular filename, which can then be re-used again or edited.

Note that some *MLSSA* commands, such as Transfer Load, interpret the Right arrow key as a command to invoke the file browser. This feature works only when the cursor is beyond the end of the text line. Otherwise, the Right arrow key moves the cursor to the right, one character at time. To invoke the file browser after editing a line, press the End key followed by the Right arrow key. Pressing the Right mouse button invokes the file browser straight away, even when the text cursor is still in the middle of the line.

### 3.43 Recalling Previously Entered Comments

*MLSSA* constantly logs all comments you enter for any purpose, both file comments and printout comments, into a 1024 character history buffer. The number of comments stored depends upon their length. For example, if the average comment length is 50 characters then the last  $1024/50 = 20$  comments you entered can be re-used or edited. More comments can be recalled if they are shorter, on average. The contents of the comments history buffer are automatically saved to disk when you quit *MLSSA* and automatically re-loaded from disk when you re-start. *MLSSA* will therefore remember a certain number of your comments indefinitely. If more comments are entered than the buffer can hold, the oldest one is deleted to make room for the newest. To delete your comment history completely, delete the file COMMENTS.INI located in the *MLSSA* program folder or directory.

To access the comments stored in the buffer, simply press the Up arrow and/or Down arrow keys when *MLSSA* is prompting you to enter a comment. These keys scroll through the comments history buffer. You can re-use any one of them and edit it, if

desired. The PgUp key moves to the oldest available comment while the PgDn key moves to the end of the list.

When prompted to enter a comment you can also elect to press the Right arrow key. Doing so recalls the current comment, if any. Pressing Right arrow key a second time copies the current contents of message window, which you can edit and use as a comment. Pressing Right arrow again toggles to the current comment again. Note that the current comment is the one that is currently attached to the data in question. For example, if you execute Transfer Load and load file DATA.FRQ then the current comment becomes the comment that is stored in DATA.FRQ. You can still use the Up and Down arrow keys if you decide not to use the current comment. There are no restrictions on the order or number of key presses.

*MLSSA* commands Transfer Load and Transfer Save offer a time saving feature using the Up arrow key. When these commands are prompting you to enter a filename, pressing the Up arrow key causes the last filename you had loaded or saved the last time you had used that command, to be re-entered for you. This feature is also available to the Transfer Import and Transfer Export commands in both the time and frequency domains, and also Waterfall Text-export-file, QC-Limits Load and several other commands that specify a filename. Note unlike the comments, the last filenames are not saved when you Quit *MLSSA*.

### 3.44 Startup Switches and Options

Command line switches allow you to specify or override certain *MLSSA* modes of operation. A switch is simply the forward slash character '/' followed by one or more special characters. More than one switch can be specified on the same command line but all switches must be placed after the setup file name, if any.

MLSSA Command Line Switches:

/3	32K maximum acquisition length
/6	64k maximum acquisition length
/a	Analysis mode
/c	Character mode
/v0	CGA graphics mode
/v1	EGA graphics mode
/v2	VGA graphics mode
/m<filename>	Run the macro <filename>
/n<string>	Display the name <string>, 10 characters max
/?	Display all command line switches available

The switch */?* is a special help switch that displays the list of all the other available switches, as in the table above, but does not start *MLSSA*.

The */3* and */6* switches are used to configure the maximum acquisition length you can select through the Acquisition Length command. The */3* switch limits the maximum impulse response length to 32k points. This does not mean you cannot use the 64k MLS stimulus, you still can. Rather, the purpose of the */3* switch is to free up half of memory that would otherwise be needed to store the full 64k points of the measured impulse response. Since the impulse response of virtually all loudspeakers is confined to well under 32k points, this mode is suggested when performing loudspeaker measurements. The */6* switch, in contrast, permits storing impulse responses of up to 64k points in length, which are often necessary when measuring room acoustical parameters and speech intelligibility (STI). The default value of the maximum acquisition length is controlled by line 10 of the *MLSSA.INI* file, located in the main *MLSSA* directory. If you do not specify either the */3* or the */6* switch on the command line, the maximum acquisition length you can select using the Acquisition Length command will be whatever is specified by line 10 of the *MLSSA.INI* file. The */3* and */6* switches override the default specified in *MLSSA.INI*. The 64k maximum acquisition

length option, whether specified by the /6 switch or by line 10 of the MLSSA.INI file, should not be used unless the vast majority of your work involves room acoustics or speech intelligibility. The /6 switch severely limits the amount of free memory available for frequency domain post-processing functions required for typical loudspeaker measurements. Room acoustical parameter measurements as well as STI and RASTI speech intelligibility measurements generally do not require much frequency domain processing but do require measuring very long impulse responses. The following command line, for example, starts *MLSSA* to measure STI using setup file STI.SET and adds the /6 switch to override the default maximum acquisition length of 32k points in MLSSA.INI file:

```
MLSSA sti /6
```

The next switch to discuss is the /a switch that starts *MLSSA* in Analysis mode. Analysis mode prevents all software access to the *MLSSA* card. Analysis mode is suggested in cases when another copy of *MLSSA* running, for example, a macro to acquire loudspeaker polar response data. Suppose you want to start a second copy of *MLSSA* on the same PC in order to do some unrelated analysis work. If you start a second copy of *MLSSA* while another copy is busy performing measurements, the result will likely be a crash. Starting the second copy of *MLSSA* with the /a switch is the solution.

The /c switch starts *MLSSA* in character mode. This is equivalent to starting *MLSSA* without the /c switch and then immediately pressing Alt-9 to switch to character mode.

The /v0, /v1 and /v2 switches start *MLSSA* in the CGA, EGA or VGA graphics modes respectively. These switches override the default graphics mode specified by line 4 of the MLSSA.INI file.

You can have *MLSSA* load and run a previously saved macro file at start up through the /m command line switch. For example the command line:

```
MLSSA loads /mtestproc
```

starts *MLSSA* with setup file LOUDS.SET and immediately, from the time domain, loads and runs the macro file TESTPROC.MAC.

The /n switch allows you to enter an alphanumeric string up to 10 characters in length that *MLSSA* displays in the upper left hand corner of the screen. The /n switch is used if you intend to run more than one copy of *MLSSA* under Windows simultaneously and need to keep track of which copy is which.



## 4. *MLSSA* Commands Tutorial

This section takes you on a guided tour of *MLSSA* and includes exercises designed to acquaint you with its various commands. To start you'll need to run the program by typing *MLSSA* followed by the Enter key from the DOS command line or, double clicking on the *MLSSA* program icon from Windows 95/98. Next, load the time data file *ROOM.TIM* by typing a "t" for Transfer, then an "l" for Load. Enter the filename as "ROOM" without the extension and press Enter to load the file. This file holds the impulse response of a loudspeaker/room combination as previously measured by *MLSSA*.

### 4.1 Graphics Commands

The keyboard cursor keys and function keys allow interactive control of the *MLSSA* graphics display. A list of all graphics commands can be displayed by pressing F1 from the main menu and paging down one page. The graphics cursor is a solid vertical line, which can be moved left and right by pressing the Left and Right arrow keys. Normally, each time you press a Right or Left arrow key the cursor moves one data point. Cursor movement can be increased by holding down the Ctrl key simultaneously with an arrow key. Pressing the Home key causes the cursor to move to the left display edge while pressing the End key causes it to move to the right display edge.

You can also use a mouse, if installed on your computer, to move the graphics cursor. Press the left mouse button to enable mouse control of cursor movement. Pressing the left mouse button again disables mouse control which prevents inadvertent cursor movements if, for example, the mouse sits on a sloping desk and will move on its own when unattended. The right mouse button (when mouse control is enabled) sets the marker to the current cursor position, which is equivalent to pressing the F9 function key.

You can also move the cursor directly to its final destination by using the F2, Shift-F2 or Ctrl-F2 cursor commands. F2 moves the cursor to the specified data point index starting with index 0. Shift-F2 moves the cursor to the specified X-axis value, which

may be in units of time or frequency depending upon which domain you are in at the time the command is invoked. Ctrl-F2 moves the cursor right to the specified Y-axis value in currently displayed units.

**Exercise 1:**

Press F1 and PgDn to display the graphics functions Help screen. Press Esc to return to the main display. Move the cursor left by holding down the Left arrow key for a few seconds. Now press the Ctrl-Left arrow key combination and note the increase in the horizontal increment. Move the cursor from one end of the display to the other by alternately pressing the Home and End keys. Move the cursor to the 100th data point by pressing F2 and entering 100 followed by the Enter key. Move the cursor to the 3-millisecond position by pressing Shift-F2 and entering 3 followed by the Enter key.

The cursor shape can be changed from a simple vertical line to a cross hair by pressing Shift-F5. Pressing Shift-F5 a second time returns the cursor shape to its original format. The exact numerical data represented by the cursor position can be displayed continually by pressing F5. Pressing F5 a second time disables the cursor display.

**Exercise 2:**

Enable the cursor display by pressing F5. Using the left and right arrow keys or the mouse, move the cursor and watch the displayed value change. The number in parenthesis is the index of the current data point. Press Shift-F5 to change the cursor to a cross hair. Press Shift-F5 to return to normal.

In addition to the cursor, *MLSSA* also displays a dashed vertical line called the marker. The marker is positioned by first moving the cursor to the desired location and then pressing F9 (or the right mouse button if mouse control is enabled). Similarly, the cursor can be moved to the marker position by pressing Shift-F9. Ctrl-F9 moves the cursor relative to the marker by the entered value, which can be positive or negative. For example in the time domain, Ctrl-F9 followed by 10 <Enter> moves the cursor 10 ms ahead of the current marker position while entering -10 will place it 10 ms behind the marker. Together, the marker and cursor are used to frame areas of the display for

various other functions including zoom, FFT and statistics calculations. The marker position is unaffected by any of the cursor keys and can only be changed by pressing F9, the right mouse button or re-sizing the display by pressing Ctrl-PgDn. The area of the display framed by the marker and cursor positions can be zoomed by pressing F10. Zooming expands or compresses the display both horizontally and vertically. The vertical scale is automatically adjusted for best visibility of the signal within the zoomed region. Pressing Shift-F10 restores the display to its original condition, that is, the marker and cursor positions are automatically restored regardless of any movements occurring during the zoomed condition.

**Exercise 3:**

Press F9 or the right mouse button to move the marker to the current cursor position. Move the cursor to the extreme left by pressing the Home key. Now press F10 to zoom this area to see the fine detail. Press Shift-F10 to return to the original display.

In some applications, it is convenient to move the marker and cursor together in lock step. You can do this by pressing Ctrl-F10.

**Exercise 4:**

Press Ctrl-F10 to enable cursor/marker locking. Move the cursor by pressing the Right-arrow key and note that the marker follows. Press Ctrl-F10 again to disable cursor/marker locking.

Function keys F7 and F8 respectively scroll the display left and right. Each keystroke scrolls the display by one display width. Of course, if the entire signal is already being displayed, horizontal scrolling is not possible. If more signal is available, horizontal scrolling will also automatically occur if you attempt to move the cursor beyond a display edge. You can move to the start or end of a signal by pressing the Ctrl-Home or Ctrl-End key combinations respectively. These commands will initiate scrolling when necessary to bring the signal's end-point into view. In contrast, the Home and End keys move the cursor only to the current display edges and never initiate scrolling.

**Exercise 5:**

Press Ctrl-End to move the cursor to the signal's end-point. Press Ctrl-Home to move the cursor to the signal's starting point. Press F8 to see more signal to the right. Press F7 to scroll back. Press the End key to move the cursor to the right display edge. Hold down the Right arrow key or move the mouse to the right and note that scrolling occurs automatically.

Vertical scrolling is activated by the Up, Down, PgUp and PgDn keys. The vertical arrow keys scroll by one vertical division while the page keys scroll by three vertical divisions.

**Exercise 6:**

Press the Up, Down, PgUp and PgDn keys and observe the effect on the display.

A signal can be graphed as individual data points or the points may be connected by lines to form a continuous curve. Function key F3 selects the active display format. Function key F4 changes the horizontal axis from a linear to a logarithmic scale and back again.

**Exercise 7:**

Press F3 to plot points. Press F3 again to plot points connected with lines. Press F4 to change to a logarithmic scale. Press F4 to return to a linear scale.

Function key F6 expands the vertical axis and magnifies the signal. Conversely, function key Shift-F6 compresses the vertical scale and shrinks the signal. Ctrl-F6 allows you enter a custom vertical scale. These commands allow you to override the vertical scale factor selected automatically by *MLSSA* during a zoom operation.

**Exercise 8:**

Press F6 and observe the effect on the display. Press Shift-F6 to restore the original vertical scale factor.

In addition to expanding a portion of the signal, the zoom command can also be used to compress a signal.

**Exercise 9:**

Move the cursor to the signal start by pressing Ctrl-Home. Press F9 to set the marker there. Press Ctrl-End to move the cursor to the signal end-point. Press F10 to display all the data at once.

The Ctrl-PgDn key combination re-sizes the display. The marker is moved to index 0 and the cursor is moved to the current default display width. This command is useful for undoing a complex set of graphics operations. The Ctrl-PgUp key combination also re-sizes the display but the marker position is not reset to index zero as with the Ctrl-PgDn command. Use Ctrl-PgUp when you want to remember a special time or frequency domain data point using the marker.

**Exercise 10:**

Press Ctrl-PgDn and observe that the display returns to its original state.

Pressing Shift-F7 allows you to change the default display width. Thereafter, pressing the Ctrl-PgDn or Ctrl-PgUp key combination will re-size the display based on the new default display width. Note also that the default display width is stored in setup files. This means that loading a new setup file may change the default display width.

**Exercise 11:**

Press Shift-F7 and enter 4096 followed by the Enter key. Note that the display is re-size to reflect this new default display width.

*MLSSA* does not normally draw a grid on the graphics screen to maximize drawing speed. However, printouts often require a grid so that measurements can be read

directly off the paper. *MLSSA* will draw a grid over the graphics display when you press Shift-F8.

Exercise 12:

Press Shift-F8 and observe that a grid appears over the graphics display.  
Press Shift-F3 to erase the grid.

You can also have *MLSSA* display a grid continually by pressing Ctrl-F8. To restore normal (no grid) operation press Ctrl-F8 again.

Normally, every time *MLSSA* performs a new measurement it re-scales the vertical axis automatically for a best fit to the available data. This autoscale mode is active whenever the word "auto" is displayed on the lower left outside corner of the graphics area. Sometimes the autoscale feature is undesirable. You disable it by pressing Shift-F4 and the word "fixed" is displayed indicating that the vertical scale is now fixed. Actually, the vertical scale can still be adjusted manually by pressing F6, Shift-F6, Ctrl-F6, Up, Down, PgUp and PgDn. Once you have set up the desired fixed vertical scale you can optionally save it to any setup file through the Setup Save command and later restore it through the Setup Load command. Note also that different fixed vertical scales can be defined for the time-domain and frequency-domain displays.

Exercise 13:

Press Shift-F4 and observe that the word "fixed" appears to indicate a fixed vertical scale. Press Shift-F4 again to restore autoscale mode.

*MLSSA* normally erases the graphics screen whenever a new measurement is to be plotted. In some cases, however, it may be desirable to observe two or more plots simultaneously. This is made possible by the overplot feature, which is activated by pressing Ctrl-F4. When overplot is on this is indicated by the word "overplot" being displayed. Autoscale is also turned off during overplot to insure that all plots will have the same vertical scale. The overplot feature is deactivated by pressing Ctrl-F4 again or by any action (e.g. pressing F1 for help), which temporarily erases the graphics screen.

**Exercise 14:**

Press View Schroeder-plot Unfiltered ("vsu") to display the wideband reverberant decay curve for ROOM.TIM. Press Ctrl-F4 to enable the overplot feature. Press View Energy-time-curve Unfiltered Blackman-Harris ("veub") to overplot the energy-time-curve of the same room. Press Ctrl-F4 to disable the overplot feature.

Note that the overplot feature should only be used when the various curves have comparable units (dB in the example above). You change the overplot color through the Library Colors Main oVerplot command. The overplot color can be also changed while the overplot feature is active allowing multicolored plots. Note also that overplots differ from overlays in that the data of the former is not retained in memory but only on the graphics screen.

Pressing Shift-F1 expands the graphics display vertically to fill the whole screen for better visibility. In this mode, menus share the same screen area with the horizontal axis labels. You toggle between them by either pressing the Esc key or the Enter key. To go back to the old graphics display press Shift-F1 a second time. You should probably stick to the normal (unexpanded) display until you become more familiar with *MLSSA*.

## 4.2 Statistics Commands

*MLSSA* statistics commands compute signal statistics on data framed by the marker and cursor. Statistics commands are invoked by holding down the Alt key while pressing one of the alphabetic keys. A list of all statistics commands is displayed by pressing Alt-h.

If an overlay has been plotted, the statistics commands operate on the overlay data rather than on the main trace data. To compute statistics on the main trace simply erase the overlay by pressing Shift-F3. The overlay can always be re-plotted with the Overlay Plot command.

**Exercise 15:**

Hold down the Alt key and press the "h" key. A list of all statistics commands should now be displayed. Press Esc to resume. Press Alt-a to display several signal statistics.

### 4.3 Menu Commands

*MLSSA* menu commands simple to learn. You need only remember the following rules.

- 1) You make a menu selection at any level by typing the highlighted letter of the desired command.
- 2) Pressing the Esc key at any time returns you to the main menu and aborts any command that was in progress.
- 3) Pressing F1 at any menu level displays context sensitive Help.

In the time domain (where you should be now) *MLSSA* displays the following main menu:

**TIME DOMAIN MENU: Go View FFT waterfall Acquisition Setup Transfer Macro  
Overlay Calculate Printer DOS Units Library Info Quit**

Most words in the main menu represent a broad category of functions. For example, the word "Acquisition" represents all parameters related to programming of the hardware including the stimulus generator and the antialiasing filter. If you now press the letter "a" the Acquisition menu will be entered.

**Exercise 16:**

Press "a" to enter the Acquisition menu.

**ACQUISITION: Mode Autorange Bandwidth Length Pre-average Equalization Interface  
Clock Stimulus Trigger Filter Units Delay**

Pressing function key F1 at this point will display a general Help screen for the Acquisition commands.

To display Help for specific Acquisition commands you must be continue to a lower (more specific) menu level.

Note that the word ACQUISITION replaces the words "TIME DOMAIN MENU" preceding the main menu. This reminds you that you are now at a lower menu level. To get to the next lower level, make another selection.

Exercise 17:  
Press "s" to enter the Acquisition Stimulus menu.

**ACQUISITION STIMULUS: select Amplitude Mode Units Program**

Now the menu is prefaced by the two words "ACQUISITION STIMULUS" which indicate the path taken to this menu level.

Exercise 18:  
Press F1 to display the Help screen.

At the Acquisition Stimulus menu level, the Help screen refers only to commands at this level.

Exercise 19:  
Press Esc to resume. Press "a" to enter the Acquisition Stimulus Amplitude menu.

**ACQUISITION STIMULUS AMPLITUDE volts:**

Now you can see that in the message window (two horizontal lines above menu) *MLSSA* is displaying the current stimulus amplitude. In addition, in the lower left hand corner of the screen is a prompt indicating the acceptable range of new amplitude values that can be entered. At this point, you have two choices; you can abort the command by pressing Esc or you can enter a new value for the stimulus amplitude.

Exercise 20:  
Enter "1.5" and press Enter to change the stimulus amplitude to 1.5 volts.

Note that *MLSSA* now displays the new stimulus amplitude in the message window and returns to the main menu.

Some menu commands don't require that you enter a value but rather present you with a list of options from which to select.

Exercise 21:  
Press "afs" to display the Acquisition Filter Select menu.

**ACQUISITION FILTER SELECT: Thomson Butterworth Chebyshev**

Note that in this case the prompt in the lower left hand corner says "Select option from list". In this case, you select the type of antialiasing filter by typing the highlighted letter of the desired filter type.

Exercise 22:  
Press "b" to select the Butterworth filter option.

So far, you have seen how to change some individual acquisition parameters using menu commands. Because there are a large number of *MLSSA* parameters, setting them up individually every time you ran the program would be tedious and time consuming. Instead, you can use the Setup Save command to save the entire operational state of *MLSSA* to a setup file.

Exercise 23:  
Press "s" to enter the Setup menu and display the setup status.

The Setup command displays a summary of the most important *MLSSA* parameters. Note that the stimulus amplitude and antialiasing filter changes you made are reflected

here. To save these changes to a new setup file you can use the Setup Save command.

**Exercise 24:**

Press "s" to enter the Setup Save command. Enter the filename "NEW" followed by Enter to save the current state to the file NEW.SET.

Note that *MLSSA* automatically adds the filename extension "SET". To prove this you could use the directory option in the Setup Load command.

**Exercise 25:**

Press "l" to enter the Setup Load command. Press Right arrow or the right mouse button to see the directory of current setup files. Note that NEW.SET is listed. Use the arrow keys or the mouse to move the cursor over the filename NORMAL.SET then press Enter or the left mouse button to load this file. The previous changes you made to the stimulus level and antialiasing filter type have now been undone. Press Esc to return to the main menu.

As demonstrated above, during commands which prompt you to enter a filename you can press the right mouse button instead of the Right arrow key to display a list of available files. You can then move the mouse to select the file you want and press the left mouse button to load that file. During commands that allow you to select multiple filenames, for example the Library Operations Frequency-files Average command, the left mouse button marks/unmarks files while the right mouse button ends the selection process (equivalent to pressing the Enter key).

#### 4.4 Utility Commands

*MLSSA* utility commands are invoked by holding down the Alt key while pressing one of the numeric keys 1 thru 9. A list of all utility commands is displayed by pressing F1 from the main menu and paging down two pages. The utilities functions are listed in the table below.

Alt-0	Dumps any text or graphics screen "as is".
Alt-1	Prints any text or graphics screen "as is".
Alt-2	Displays available DOS and UMB memory in bytes.
Alt-3	Displays filter module serial numbers.
Alt-4	Captures graphics or text screens and saves them to files.
Alt-5	Display current MLSSA version.
Alt-6	Mark current time.
Alt-7	Display elapsed time from last mark (Alt-6).
Alt-8	Display a message. If a macro is executing wait afterwards.
Alt-9	Toggles Character mode on and off.
Alt-=	Toggles between CGA and VGA graphics modes.

### Utility commands

Pressing Alt-4 captures the current graphics screen and writes it to a PCX file. A PCX file stores graphics data in a standard format originally used by a drawing program known as PC Paintbrush. Nowadays, most desktop publishing programs have the ability to import graphics screens saved in the PCX format. You can therefore use the Alt-4 command to save any graphics screen (including waterfall screens) to a PCX file and then import the image into a publishing program to produce reports. Another use for this utility is to save graphics screens and review them later through the Library Load-PCX command. The Library Load-PCX command will load and display any valid PCX file even one not created by *MLSSA*'s screen capture utility. After loading a PCX file, you can optionally print it through the Alt-1 print utility.

#### Exercise 26:

Press Alt-4 to enter the screen capture utility. Enter "TEST" followed by Enter to save the current screen to the file TEST.PCX. Answer 'y' to the prompt "Adjust colors for printing? (Y/N)". *MLSSA* will sound a short high-pitched beep when the capture is successfully completed. Now load the newly created PCX file by pressing Library Load-PCX ("ll") and entering "TEST" followed by Enter. Press Esc to restore the original display.

Pressing Alt-= causes *MLSSA* to enter CGA graphics mode thus allowing you to run *MLSSA* in a window under Windows 95/98. In CGA mode under Windows 95/98 *MLSSA* will also run as a background task.

Exercise 27:

Switch from running full screen to running in a window. Hold down the Alt key then press the = key and release. While still holding down the Alt key press the Enter key. To switch back to running full screen in VGA graphics mode press Alt-= again.

This Alt-4 capture utility can also be used to export screens (e.g. RASTI, STI, Setup or Info screens) to plain text files but first you need to switch to Character mode by pressing Alt-9. Text screens are saved to plain text files having the extension "TXT" to distinguish them from PCX files, which have the extension "PCX". Thus, you can use the Alt-4 capture utility effectively as an export utility for the data contained in text screens.



## 5. Theory Notes

### 5.1 Maximum-length Sequences

*MLSSA* measures a linear system's impulse response using a pseudorandom stimulus called a maximum-length sequence (MLS). ML sequences have both theoretical and practical advantages over other test signals commonly used to measure linear systems such as narrow pulses, white noise and swept sine signals. These advantages are:

- 1) The MLS stimulus has a low crest factor for high signal energy content. Thus, MLS measurements provide very high signal-to-noise ratios.
- 2) Unlike white noise, the MLS stimulus is non-random and exactly repeatable. Therefore, there is no need to simultaneously measure the stimulus input with the system output as required by dual-channel FFT techniques.
- 3) The MLS stimulus is periodic. Consequently, signal truncation effects do not appear as long as exactly one MLS period is used to compute the system impulse response by cross-correlation. There is no need for data windows and their attendant compromises as required by other methods.
- 4) Mathematical properties of the MLS stimuli permit efficient computation of very long impulse responses. When a large FFT is applied to an extended wideband impulse response the result is a wideband transfer function with very fine frequency resolution. Hence, there is no need for zoom processing with the MLS approach.

When using ML sequences for linear system measurements there are a few theoretical points to keep in mind:

- 1) The MLS power spectrum follows a  $\sin^2(x)/x^2$  envelope which falls by 1.6 dB at one third of the sample rate. You should always choose a sample rate at least 3 times the bandwidth of interest to maintain a flat MLS spectrum. *MLSSA* does this automatically through the Acquisition Bandwidth command.

- 2) The MLS repetition period should be longer than the expected duration of the system impulse response. If not, the starting and ending sections of the measured impulse response will overlap in a phenomenon called time aliasing. A certain amount of time aliasing is tolerable. A rule-of-thumb in room acoustics measurements is to choose an MLS period, which is at least 1/3 as long as the room reverberation time. Thus, you can safely measure the impulse response of a room having a reverberation time of up to 3 seconds using an MLS period of 1 second.
- 3) The MLS signal must be applied to the system long enough for its output to reach a steady state condition. *MLSSA* insures that this condition is met as long as you satisfy condition #2 above.
- 4) The MLS method has no response at DC. In most applications, this is an advantage because DC offsets are automatically rejected.
- 5) The MLS method is accurate only for *time-invariant* systems meaning their transfer function does not change significantly over the measurement interval. Therefore, you should not attempt to make room measurements while the room acoustics are changing. This might occur if enough people are milling around a room during a measurement session. In addition, some of the new sound enhancement systems are time-variant in nature and should not be measured with an MLS. If this is suspected turn off the enhancement system and repeat your measurements.
- 6) Non-linearity or distortion results in fixed-pattern noise in the measured impulse response. This fixed-pattern noise is normally spread out over the entire length of the impulse response just like ordinary noise so that windowing the impulse response has the beneficial side effect of reducing the influence of distortion on transfer function or frequency response measurements.

*MLSSA* will generate four different MLS periods shown in the table below.

<i>MLS order</i>	<i>MLS period</i>
12	4095 points
14	16383

15	32767
16	65535

The MLS signal is periodic with period  $2^N-1$  where N is the order of the sequence. The MLS generator is clocked by the sample rate clock. Therefore, the period of a MLS, in seconds, is equal to the sampling interval times the sequence period in points. For example, for a 100 kHz sampling rate (10 microseconds sampling interval) a 4095-point MLS will have a period of 40.95 milliseconds. *MLSSA* does this calculation for you when you select the Setup command. You select the MLS period through the Acquisition Stimulus Program MLS command. Shorter sequences provide faster measurements but with less noise immunity and possible time-aliasing in reverberant environments. Time aliasing can usually be recognized by the appearance of noticeable noise in the time-of-flight gap of the measured impulse response, which cannot be reduced by averaging or pre-averaging.

Note however that the appearance of such noise can also be due to excessive system nonlinearity. Nonlinearity or distortion shows up in an MLS measurement as fixed-pattern noise meaning noise that does not change from one measurement to the next and so cannot be reduced by averaging or pre-averaging. Normally, the fixed-pattern noise due to typical nonlinearities is small compared to the background noise and can be ignored. If the nonlinearity is excessive, however, such as can occur with clipping distortion in power amplifiers or overdriven loudspeakers the fixed-pattern noise can dominate. If you encounter a situation in which the noise appearing in the initial time-of-flight gap seems excessive and cannot be reduced by averaging try reducing the MLS drive level. If the noise is reduced or goes away entirely it was due to excessive nonlinearity and not to time-aliasing or background noise.

## 5.2 True Frequency Resolution

The system transfer function is computed by applying an FFT to the impulse response or a portion of it as in the case of anechoic loudspeaker measurements. The displayed frequency resolution is given as the sample rate divided by the FFT size. Displayed frequency resolution is just the FFT bin separation. However, the true frequency resolution of a measurement is at best equal to  $1/T$  where T is the length of the impulse

response segment used in the FFT calculation (i.e. the width of a rectangular time window). Thus, it is possible to have a very fine displayed resolution but a rather coarse true resolution. For example, you could apply a 32768-point FFT to a 1024-point segment of impulse response. In that case, the large FFT essentially acts to interpolate the true frequency resolution to form a much finer displayed resolution. Data-tapering window shapes yield even worse resolution than a rectangular window. In the frequency domain, *MLSSA* indicates the true frequency resolution by a color bar in lower left-hand corner of the graphics area. The data falling in the frequency range indicated by the color bar should be regarded as suspect.

For example, if the time window  $T = 5$  milliseconds the lowest accurate frequency is 200 Hz. The true frequency resolution at all higher frequencies will, in this case, also be 200 Hz meaning that any two peaks in the actual response that are within 200 Hz of each other will appear as a single peak on the graph.

To obtain better true resolution you must measure an extended impulse response and use an equally long time window and FFT size to compute the transfer function. At low frequencies, even finer resolution can be achieved by lowering the measurement bandwidth. *MLSSA* will allow you to go down to a 1 kHz bandwidth sampled at 4 kHz. Under these conditions, a 65535-point impulse response will be 16 seconds in duration leading to a true resolution of 0.065 Hz when using an equally long rectangular time window, as set by the marker and cursor, and a 65536-point FFT to compute the transfer function.

### 5.3 Antialiasing Filter

Aliasing occurs in sampled data systems when frequencies are present in the analog input which are greater than  $1/2$  the sample rate (Nyquist frequency). These frequencies then appear in the sampled data as phantom low frequencies (aliases), which can result in large measurement errors. Typically, an antialiasing filter consists of a fixed cut-off frequency low-pass filter, which serves to remove the offending high frequencies before sampling and digitizing takes place.

The ideal anti-aliasing filter would pass all frequencies below and completely attenuate all frequencies above the cut-off frequency. Furthermore, this ideal 'brick wall' filter would introduce no phase shift or time delay within its passband. Although such a filter is unrealizable, several approximations to the brick wall filter have been derived, each enhancing some characteristic at the expense of another. For example, the Bessel approximation achieves excellent phase linearity at the expense of a drooping amplitude response and a gradual cut-off characteristic. However, its phase linearity permits the Bessel filter to attenuate the high frequency components of a signal with minimum distortion to its waveform. At the other extreme, the Chebyshev approximation achieves a sharp cut-off characteristic at the expense of poor phase linearity as well as introducing amplitude ripples into the passband. Another popular approximation is the Butterworth filter which offers a very flat amplitude response.

To insure at least 76 dB of alias suppression, you should select a sample rate (using the Acquisition Clock command) at least 3 times the filter bandwidth for the Chebyshev filter and at least 4 times the filter bandwidth for the Thomson and Butterworth filters. If you violate these recommendations, *MLSSA* will warn you of possible aliasing errors. The Acquisition Bandwidth command automatically chooses a safe sample rate for the desired measurement bandwidth and filter characteristic.

Different filter characteristics have advantages in different applications. The Chebyshev filter allows you to select a lower sample rate but creates more phase shift than other filter types and has +/- 0.5 dB of passband ripple. However, you can correct filter errors in the frequency domain by the Reference Store command. Simply measure the impulse response of the filter alone by looping back the *MLSSA* output to input. Next, compute the filter's transfer function by the FFT Execute command. Finally, store this reference transfer function using the Reference Store command. Future measurements will now be equalized to remove the filter's amplitude and phase errors.

*MLSSA* will also compute the theoretical antialiasing filter response and store this as a reference through the Reference Theoretical command. The advantage of this command is that it avoids the need to set up and measure a loopback condition but at the expense of accuracy, since the real filter's response will not exactly match its theoretical response due to component tolerances.

## 5.4 Minimum and Excess Phase

A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response. Mathematically, the phase response of a minimum phase system equals the Hilbert transform of its log magnitude response (i.e. magnitude expressed in decibels). This relationship is also reciprocal: the log magnitude response of a minimum phase system equals the Hilbert transform of its phase response. Because the Hilbert transform requires an infinite integral over all frequencies, however, any numerical computation of minimum phase is necessarily an approximation.

To cope with this problem, *MLSSA* provides three levels of minimum phase accuracy as selected through the View Phase Accuracy command. The three accuracy levels are denoted Normal, High and Super. The Normal accuracy level is the least accurate but requires the least amount of memory and is fast. The main error associated with the Normal level is that the computed minimum phase will show a small phase delay error. A small phase-delay error is not a problem if one is making relative measurements such as determining the time coherence between drivers in a multi-driver loudspeaker system. However, if you wish to make accurate absolute phase or acoustic center measurements, the phase delay must be measured as accurately as possible. Furthermore, in addition to a phase delay error there will also be a small residual phase nonlinearity error in the computed minimum phase at the Normal accuracy level. Going to higher accuracy levels reduces both the phase-delay error and the residual phase nonlinearity of the computed minimum phase. Higher accuracy levels take longer to compute and require more memory. The High accuracy level is recommended as a good compromise between computation time and accuracy for most applications. The Super accuracy level can be used to make the highest accuracy phase measurements. When using a large FFT size (greater than 4096 points), however, there may not be enough free memory available to support the High or Super accuracy levels. In such cases you can either reduce the FFT size or choose a lower accuracy level.

The main purpose for computing minimum phase is to use it to compute the excess phase. A system's excess phase is simply the difference between its actual phase and its theoretical minimum phase. In other words, it is the phase shift in excess of the

minimum possible phase shift. Therefore, if a system is truly minimum phase it should be possible to obtain a nearly flat excess phase curve when the proper value of phase-delay correction is applied through the View Phase Delay command. The excess phase is usually used to determine how close a measured system is the true minimum phase condition. The derivative of the excess phase curve is the excess group delay. The View Group-delay Excess command displays this curve. The excess group delay is often easier to interpret than the excess phase curve. A flat excess group delay indicates a true minimum phase system but does not require any phase-delay correction for proper interpretation as does the excess phase curve. Phase delay errors show up in the excess group delay only as an overall level change and not as a slope change as with the excess phase curve.



## 6. Applications

### 6.1 Loudspeaker Anechoic Response

The "anechoic" response of a loudspeaker is determined by first measuring the combined room/loudspeaker impulse response and then using the marker and cursor to select out only the initial "anechoic" segment of the impulse response. The interval defined by the marker and cursor is often referred to as the time window. With the Transfer-Function FFT mode in effect, an FFT Execute command on this selected segment yields the "anechoic" transfer function of the loudspeaker.

When making loudspeaker measurements a 32767 or 65535-point MLS is recommended to guard against time-aliasing effects. However, under favorable (low noise and low reverberation) conditions shorter MLS periods can be used to increase measurement speed. You change the MLS period through the Acquisition Stimulus Program MLS command.

Start *MLSSA* with setup file LOUDS.SET by typing

```
MLSSA LOUDS
```

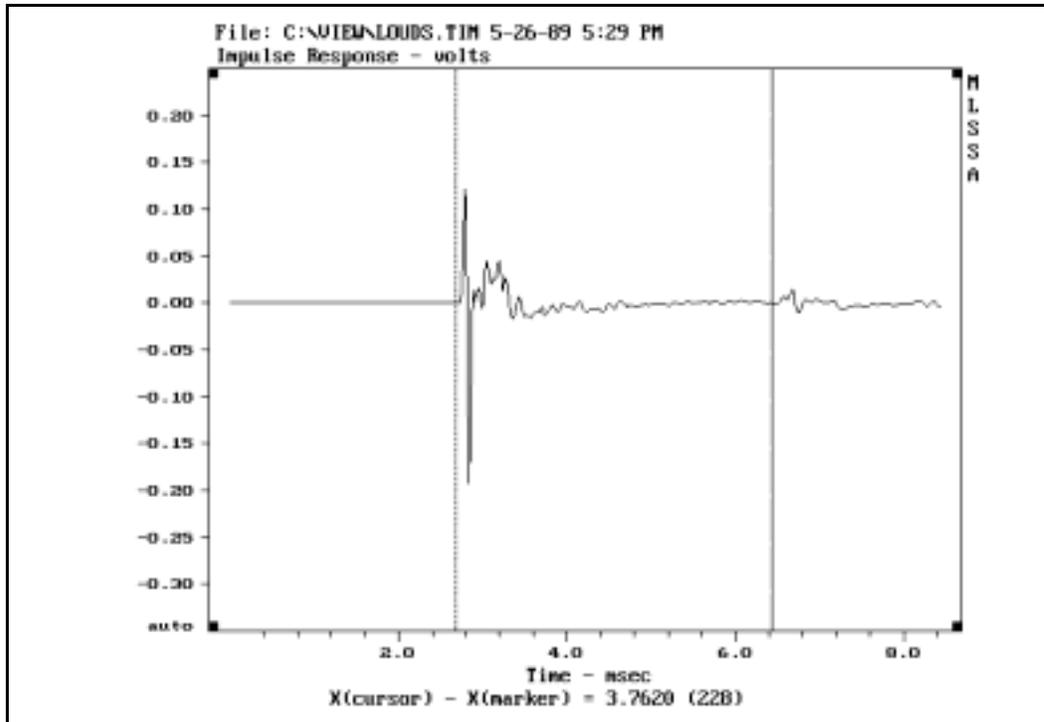
or load this file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the input connector J2 of the board (center phono connector). Position the measurement microphone about 2 meters directly in front of the loudspeaker on the tweeter axis.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Make the first measurement by executing a Go Once command. *MLSSA* may take a little extra time on the first reading to optimize the filter gain through its autorange feature. Further readings will not take as long.

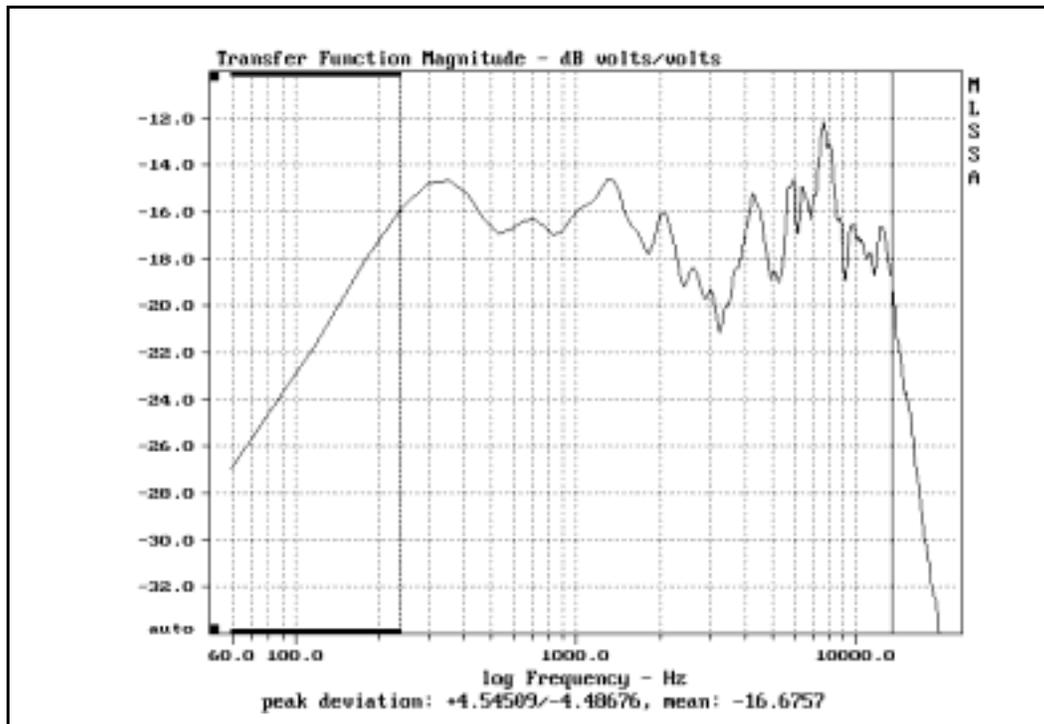


**Selecting the "anechoic" segment**

*MLSSA* will calculate and display the impulse response of the loudspeaker/room combination. Move the cursor so it is positioned just before the initial transient and

press function key F9 or the right mouse button to move the marker to this position. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Move the cursor just before the first reflection (usually due to the floor). You can also press function key F10 to zoom this "anechoic" portion of the response but this is not required. Press Alt-x to display the time window duration. You can also use the FFT Window Display command to draw the exact shape of the time window over the impulse response.

To measure the "anechoic" transfer function, execute the FFT Execute command. MLSSA will enter the frequency domain and display the magnitude of the speaker's frequency response as shown in the figure below.



Measured "anechoic" frequency response

The true frequency resolution of the "anechoic" transfer function measurement using a rectangular time window equals  $1/T$  where  $T$  is the duration of the impulse response segment selected by the marker and cursor. Therefore, if  $T = 5$  milliseconds the lowest accurate frequency in the transfer function is 200 Hz. The true resolution at all higher frequencies will also be 200 Hz meaning that any two peaks in the actual frequency response that are within 200 Hz of each other will appear as a single peak on the graph. The limited frequency resolution is due to the physical measurement setup and is the same for *all* measurement methods, not just for *MLSSA*. Although *MLSSA* will display frequencies below  $1/T$  Hz, these frequencies cannot be considered valid and should be ignored. In the frequency domain, *MLSSA* displays color bars in the lower and upper left-hand corners of the graphic area. These indicate the true frequency resolution of the measurement warning you that the data in this low frequency region is suspect. You can also totally blank the suspect data by pressing Ctrl-F5. Pressing Ctrl-F5 again unblanks the low frequency data.

To measure the peak deviation from flatness over a particular frequency range, first position the marker and cursor to select the range you want followed by the Alt-d (for deviation) command to compute the mean value and the peak deviation from the mean.

The default *MLSSA* time domain window shape used for loudspeaker measurements is the rectangular window because it offers the best frequency resolution. If the speaker impulse response has not decayed to nearly zero just before the first room reflection arrives, however, you should probably use a tapering window such as the Half-Hamming window. The reason is that a rectangular window will abruptly truncate such an impulse response causing a sharply falling edge, which will result in spectral leakage effects. These often appear as ripples in the mid-range frequency response. To select this data-tapering window from the time domain, execute the FFT Window Hamming Half command. You can also display the exact window shape and position through the FFT Window Display command. The disadvantage of a data-tapering window shape is that it yields poorer frequency resolution as compared to a rectangular window. Thus, the color bar indicators in the frequency domain will extend to a higher frequency when using a data-tapering time window. To return to the rectangular window, execute FFT Window Rectangular command.

There are several approaches to increasing the true frequency resolution in order to improve low frequency accuracy. One method is to select the Adaptive Window through the FFT Window Adaptive Speaker command and increase the FFT size to 32768 points through the FFT Size command. This setup will automatically include some of the room reflections as required at the lowest frequencies where accuracy would otherwise be impaired. This is not necessarily a bad thing considering that loudspeakers are always played in a room. On the other hand, different rooms will influence the low-end response of a given loudspeaker differently. To perform measurements using the Adaptive Window in Speaker mode substitute the setup file LOUDS-AW.SET for LOUDS.SET.

If available, you should perform your measurements in an anechoic chamber but no anechoic chamber is totally reflection free, especially at low frequencies. MLSSA's Adaptive Window feature in Speaker mode is recommended when performing measurements in an anechoic chamber to reduce the influence of the residual chamber reflections except at low frequencies. Starting from the LOUDS.SET setup, you execute the FFT Window Adaptive Speaker command to select this feature and increase the FFT size to 32768 points through the FFT Size command. Alternately, you can load the setup file LOUDS-AW.SET to make these changes. As before, the time window at high frequencies is still determined by the anechoic window defined by the marker and cursor positions. At low frequencies, however, the time window gradually lengthens to include the more delayed reflections. The maximum Adaptive Window length is always 1/2 the current FFT size or 16384 points with an FFT size of 32768 points. This setup provides adequate accuracy down to 20 Hz. If you cannot see any chamber reflections in the impulse response a time window of 10 ms is suggested. After setting the marker just before the first arrival press Ctrl-F9 followed by 10 to set the short time window to 10 ms. To display the minimum and maximum windows, execute the FFT Window Display command. By using the Adaptive Window as described here, the residual chamber reflections will still have an effect below about 200 Hz but above this frequency the measurement will now be truly anechoic even if the chamber is of low quality.

Another approach is to reposition the loudspeaker and microphone to increase the arrival time of the first room reflection relative to the arrival time of the direct sound. Placing the loudspeaker on a tall stand in a large room with a high ceiling can achieve this effect. Outdoors, you can further delay reflections but wind and noise can take their toll when performing measurements out-of-doors.

When an anechoic chamber is not available, the best method for achieving true anechoic results over the full audio frequency range is to use the method described in the section below titled "Loudspeaker Near-field Measurements".

Execute the Waterfall Execute command. *MLSSA* will calculate and display a cumulative spectral decay (CSD) plot of the loudspeaker based on the same "anechoic" segment of the impulse response. The CSD display can reveal delayed resonances that show up as ridges parallel to the time axis. You can move the cursor to the top of such a ridge to determine the frequency of the resonance.

If the displayed time range of the CSD plot is too short to see all of the decay, you can increase it either by increasing the shift parameter through the Waterfall Shift command or, by increasing the number of waterfall slices through the Waterfall Cycles command.

Note that the lower frequency limit of  $1/T$  discussed above for transfer function measurements also applies to the CSD. In fact, the zero-delay slice of the CSD is actually the magnitude of the transfer function.

Due to the time-frequency uncertainty principle, low frequencies and long delay times in a CSD plot are unreliable. *MLSSA* automatically determines these unreliable regions of the CSD plot and masks them from view. The unreliable regions of the CSD plot are indicated by a floor pattern of dotted lines as distinguished from the reliable data that are plotted as solid lines.

For more information on CSD plots, see under "Waterfall Plots" in the Reference chapter of this manual.

## 6.2 Loudspeaker Time Coherence

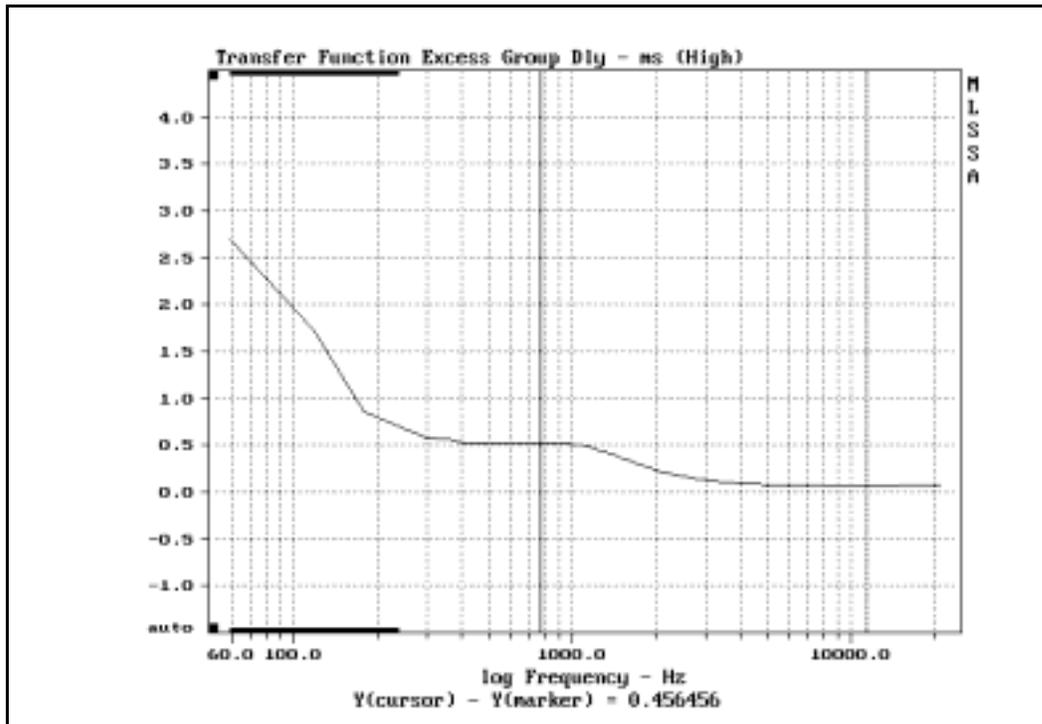
Loudspeaker time coherence refers to the possibility of differing arrival times from different drivers in multi-driver loudspeaker systems. *MLSSA* includes several post-processing algorithms designed specifically to measure loudspeaker time coherence. Perhaps the easiest to use and interpret is the excess group delay curve. Excess group delay refers to the signal delay above that expected based solely on the anechoic frequency response of the loudspeaker. In precise terms, excess group delay is the excess over the absolute group delay you would measure if your loudspeaker happened to be a true minimum phase system having exactly the same frequency response as the real loudspeaker.

For a minimum phase loudspeaker system, excess group delay would be flat over the entire audio range. Some loudspeaker designers believe that a flat excess group delay curve ranging from the lower mid-range through the treble region is desirable. Extending a flat excess group delay into the deep bass region is probably of little value however.

To measure excess group delay first follow the procedure above for performing an "anechoic" transfer function measurement of your loudspeaker. From the frequency domain, execute View Group-delay Excess to display the excess group delay. See the example excess group delay curve below.

This curve is reasonably flat in the treble region and jumps up to another plateau in the mid-range region. These two flat excess group delay regions indicate that the crossovers in this system are minimum phase but that the drivers' acoustic centers do not lie in the same plane. The time difference between these two plateaus shows the relative time misalignment between the tweeter and the mid-range driver. You can get a good quantitative estimate of the time delay by moving the cursor to the treble region, pressing F9 or the right mouse button to set the marker there, followed by moving the cursor down into the mid-range plateau. Pressing Alt-y will display the difference in milliseconds between these two frequency points. This particular loudspeaker system could be made minimum phase at least over the mid-range thru treble regions simply by

repositioning the tweeter to the rear of the mid-range driver by a distance equivalent to a delay of 0.46 ms based on the speed of sound in air.



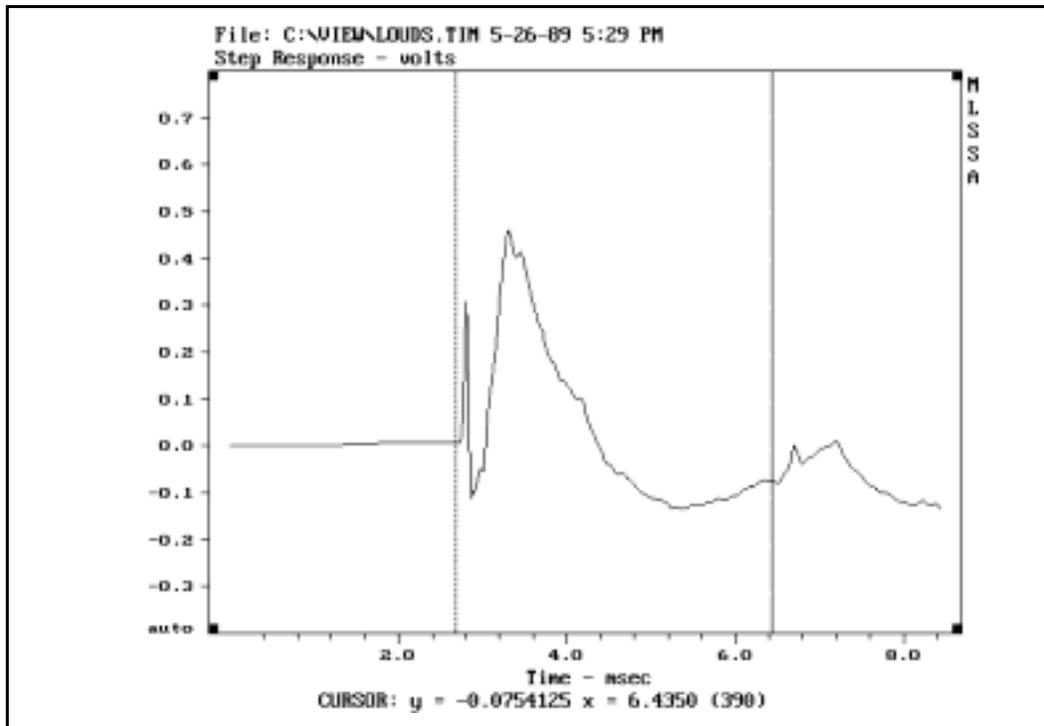
**Excess group delay curve**

Note, however, that many loudspeaker systems do not use minimum phase crossovers and will therefore not exhibit flat plateau regions in their excess group delay curves, as does the example system above. Such systems cannot be made minimum phase by any simple repositioning of their drivers although some improvement in their time coherence might be achieved through such efforts.

A more precise but more complicated method of measuring time coherence is to use the excess phase curve. This alternate method is described in detail in the section below titled "Loudspeaker Phase and Acoustic Center".

Another simple method of checking time coherence is to display the step response in the time domain. Although the step response shows something about time coherence, it is rather qualitative in nature as compared to the excess group delay which is much more quantitative showing precise delays and the frequencies at which they change. The step response is therefore not very useful in determining what design changes should be taken to improve time coherence. The step response, by its very nature as an integral of the impulse response, is heavily weighted toward the bass region meaning that a loudspeaker having fairly good time coherence in the mid-range and treble regions, as indicated by its excess group delay, may show a step response that looks much poorer.

You display the step response by executing View More Step-response Unfiltered from the time domain after measuring the loudspeaker impulse response. Ideally, the step response should look like a perfect saw tooth first rising abruptly from the baseline and then decaying linearly with time. Typically, however, poor time coherence will result in a step response that looks more like the example below. The tweeter signal arrives first causing the initial spike. The mid-range and woofer signals arrive sometime later resulting in a second upward bump in the curve.



**Loudspeaker step response**

The techniques described in this section all measure time coherence relative to a true minimum phase system. As such, they have the advantage of being insensitive to other devices in the measuring chain such as power amplifiers, microphones and *MLSSA*'s antialiasing filter. This because all these devices are usually minimum phase. To properly measure the absolute group delay or the absolute phase response of a loudspeaker, these other devices must be properly corrected or at least their effects minimized in order to obtain accurate results. Refer to the section titled "Loudspeaker Phase and Acoustic Center" below for the procedures required to perform absolute group delay and absolute phase measurements.

### 6.3 Loudspeaker Sensitivity

Loudspeaker sensitivity measurements are facilitated using the *MLSSA Remote Controlled Analog Interface (RCAI)*, which is recommended for this purpose. However, the following procedure assumes you do not own or have access to an *RCAI*.

The procedure to measure loudspeaker sensitivity is very similar to the procedure for normal "anechoic" loudspeaker transfer function measurements described in section 6.1 above. However, because sensitivity is an absolute measurement you must either program *MLSSA* to account for the gain of your power amplifier or, for highest accuracy, perform a loopback measurement of your power amplifier and store it as a reference. In addition, you must have previously entered valid microphone data and selected the microphone currently connected to the input using the Library Microphones commands.

**WARNING:**

Common ground power amplifiers having relatively low output power capability (100 watts into 4 ohms or less) are recommended. The use of high power audiophile amplifiers especially those having floating or bridged output circuits should be avoided. The high power capability of these amplifiers can result in inadvertent damage to *MLSSA*'s antialiasing filter or to the amplifier. In addition, floating amplifier output circuits can result in high ground currents since *MLSSA*'s input and output circuits are single ended.

Start *MLSSA* with setup file LOUDS.SET or SENS.SET if you had previously created it. Although *MLSSA* displays a sensitivity curve referred to 1 meter most sensitivity measurements today are taken at 2 meters actual distance. Wherever you position your microphone, you should accurately measure the actual microphone-to-baffle distance in meters and record this number for use below.

**NOTE:**

*MLSSA* can be programmed with your power amplifier's actual gain through the Acquisition Stimulus Units command as described below. In that case, whenever you change the stimulus amplitude through the Acquisition Stimulus Amplitude command or the =, + or - keys, the actual peak amplifier output voltage applied to the loudspeaker will be displayed instead of *MLSSA*'s output voltage. However, this step is not necessary provided a loopback reference loopback measurement your power amplifier is made. If you do not need to know the drive level at the output of your power amplifier you can skip the following paragraph to save time and proceed to the next paragraph describing how to make a reference loopback measurement.

If you have not previously created setup file SENS.SET, or suspect that the gain of your power amplifier has changed, follow the procedure of this paragraph, otherwise skip to the following paragraph. Temporarily disconnect your measurement microphone from the *MLSSA* input and reconnect the *MLSSA* input to the power amplifier output, preferably taking it from directly across the loudspeaker terminals to correct for any loading effect upon your power amplifier and/or any cable effects. Execute FFT Mode Transfer-Function followed by Go Once followed by FFT Execute to measure the transfer function of your power amplifier. Execute View Magnitude Linear to display the gain vs. frequency curve of your power amplifier. Press Shift-F2 followed by 1000 and the Enter key to position the cursor to 1000 Hz and note the cursor readout, which shows the gain of your power amplifier at 1 kHz. Make a note of this value and execute Acquisition Stimulus Units and enter the unit label "volts" followed by the gain of your power amplifier previously noted followed by the Enter key. Execute Reference Go to re-measure and store the power amplifier transfer function as the Main reference. Execute View Magnitude Decibels to restore the normal dB display then execute Exit to return to the time domain.

Execute the FFT Mode command and select the Sensitivity option. Enter the nominal impedance of the loudspeaker you wish to measure (e.g. 4 or 8 ohms). Note that today 8 ohms (2.83 volts) is often specified as the standard or reference impedance even when measuring the sensitivity of 4-ohm loudspeakers. Enter also the actual measuring distance in meters previously measured. Execute Setup Save to save all these parameters to setup file SENS.SET.

Temporarily disconnect your measurement microphone from the *MLSSA* input and reconnect the *MLSSA* input to the power amplifier output, preferably taking it from directly across the loudspeaker terminals to correct for any loading effect upon your power amplifier and/or any cable effects. Execute FFT Bypass followed by Reference Go to measure the transfer function of your power amplifier and store it as the Main reference. Execute Exit to return to the time domain. If you wish also now to update the reference load impedance or actual measuring distance, execute FFT Mode Sensitivity followed by the new load impedance and/or new measuring distance in meters. Execute Setup Save to save any changes to SENS.SET.

Reconnect the *MLSSA* input to your microphone preamp output and make a measurement by executing a Go Once command. *MLSSA* will calculate and display the impulse response of the loudspeaker/room combination. Move the cursor so it is positioned just before the initial transient and press function key F9 or the right mouse button to move the marker to this position. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Move the cursor just before the first reflection (usually due to the floor) to select the "anechoic" portion of the impulse response.

Execute the FFT Execute command. *MLSSA* will enter the frequency domain and display the sensitivity curve in units of dB-SPL/watt. Please note the meaning of this curve. The displayed sensitivity curve shows you, for each frequency, the dB-SPL that your loudspeaker *would* produce at 1-meter distance when driven by a pure sine wave of 1-watt amplitude referred to the specified load impedance. Thus, the sensitivity curve measured by *MLSSA* does not depend on the actual MLS power applied to your loudspeaker during the measurement and you should not worry about the exact MLS drive level unless it is too low to overcome the room noise or so high as to overdrive and possibly damage your loudspeaker. If you do want to change the drive level only change it through *MLSSA* using the Acquisition Stimulus Amplitude command, =, - or + keys. If you change the gain of your power amplifier to change the drive level the sensitivity curve will be off unless you update SENS.SET or perform a new reference measurement of your power amplifier.

Sensitivity is sometimes specified over a 1/1-octave band centered on 1 kHz. To perform this calculation, first execute Calculate Level and then position the cursor to 707 Hz by pressing Shift-F2 and entering 707. Press F9 or the right mouse button to set the marker there. Press Shift-F2 again and enter 1414 Hz. You have now selected a band 1 octave wide centered at 1 kHz. Press the Enter key to compute and display the average loudspeaker sensitivity over this frequency range. Another alternative is to smooth the sensitivity curve to 1/1 or other fractional octave through the View Magnitude sMooth Decibels command. Thereafter, the average sensitivity over the specified smoothing bandwidth can be obtained for any frequency simply by moving the cursor there and observing the cursor readout display.

#### **6.4 Loudspeaker Phase and Acoustic Center**

The phase response of a loudspeaker is determined from the "anechoic" transfer function as detailed in 6.1. Because measured phase response is very sensitive to the antialiasing filter you should first perform a loopback measurement through your power amplifier and store it as a reference through the Reference Store command prior to making your measurements. Alternatively, you can use the theoretical antialiasing filter response through the Reference Theoretical command. This option avoids having to perform a reference measurement. It is also less accurate although probably acceptable in most cases. Note also that this option cannot account for any phase irregularities in your power amplifier.

Another factor in absolute phase measurements is your microphone. For highest accuracy, you should use an instrument quality microphone having a bandwidth of 40 kHz or more with less than 1 dB of ripple over the audio band. The high bandwidth and low ripple assure a low phase shift over the audio band. If you only intend to perform excess phase or acoustic center measurements, however, microphone effects can usually be ignored since most microphones exhibit nearly minimum phase behavior even if their frequency response is not flat.

Phase curves can be either wrapped or unwrapped. Wrapped phase curves are restricted to a -180 to +180 degree range ( $-\pi$  to  $+\pi$  radians) and will therefore show sharp discontinuities. Unwrapped phase curves, in contrast, have an unlimited range

and do not normally exhibit discontinuities. *MLSSA* forms the unwrapped phase curve from the wrapped phase curve by detecting those discontinuities and adding either +360 or -360 degrees as appropriate. This phase unwrapping process can be accomplished only when the actual phase changes relatively slowly and smoothly over the desired frequency range. Therefore, you should not try to display the unwrapped phase of room measurements or loudspeaker measurements that include early reflections at high frequencies because, in such cases, the phase response will be too erratic to be properly unwrapped.

The wrapped phase is displayed through the View Phase Wrapped command while unwrapped phase is displayed through the View Phase Unwrapped command. You can also display decibel magnitude and wrapped phase at the same time through the View Bode-plot Actual-phase command. Phase units (radians or degrees) are selected through the View Phase uNits command.

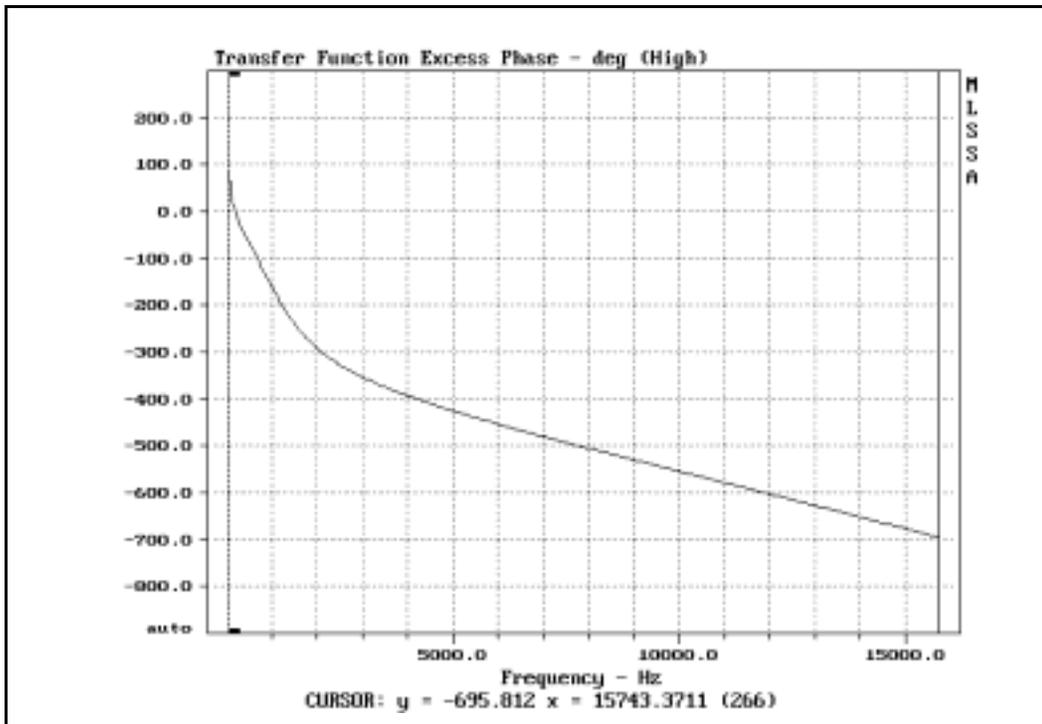
A concept central to all phase measurements is the notion of phase delay. A pure delay inserted into a system adds a negative linear phase component to its actual phase curve. That is, adding a pure delay causes the phase curve to tilt downward from left to right when displayed on a *linear* frequency scale. Conversely, removing a delay causes the corresponding phase curve to tilt upwards. Thus, a phase delay can only rotate the phase curve around its value at zero frequency without changing its basic shape when displayed on a linear frequency scale. For this reason you should change to a linear frequency scale when performing the following measurements by pressing F4 twice.

A basic problem in loudspeaker phase measurements is that there is a large acoustical delay associated with the space between the loudspeaker and microphone. Placing the marker just before the first arrival removes most of this delay in the time domain. Any residual delay can be removed through the View Phase Delay command in the frequency domain. This command allows you to manually enter any desired phase delay correction or, if the Right-arrow key is pressed, it will calculate the phase delay correction necessary to produce the flattest possible phase curve between the marker and cursor positions. Once entered, this delay correction will thereafter be automatically applied to all phase related curves including wrapped phase, unwrapped

phase, excess phase, group-delay and excess group delay. Minimum phase curves are never delay corrected however.

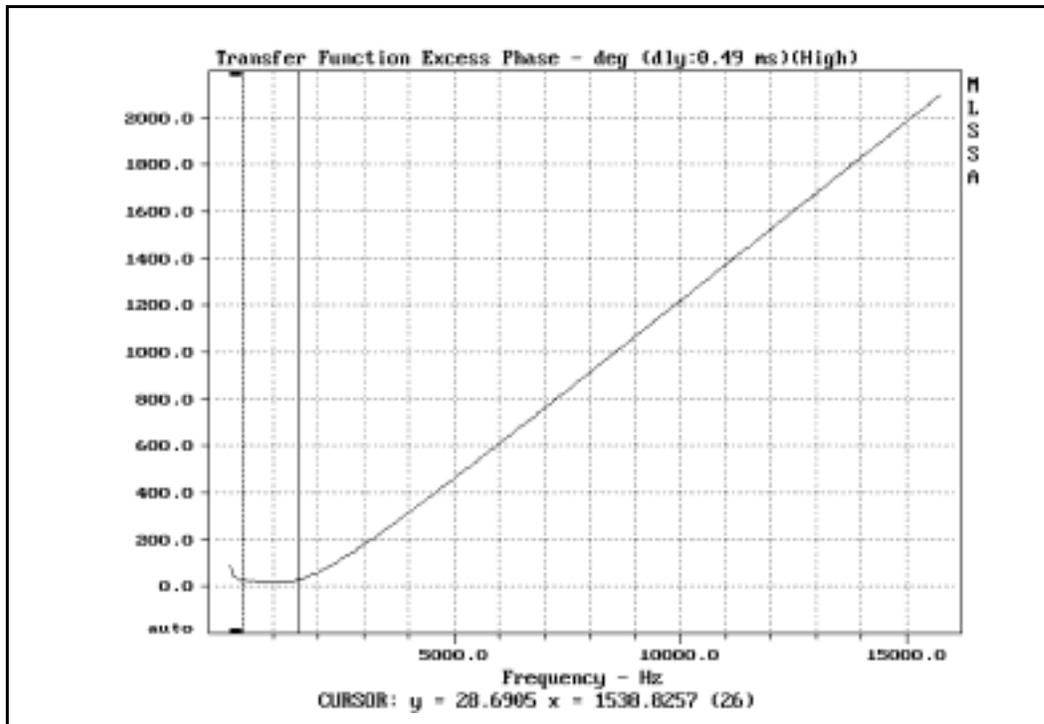
An understanding of minimum phase is helpful in determining the appropriate phase delay correction for a given loudspeaker phase measurement. See the "Theory Notes" section of this manual for a tutorial on minimum and excess phase.

While most loudspeaker drivers usually exhibit nearly minimum phase behavior, this is not necessarily the case for complete loudspeaker systems due to possible non-minimum phase crossovers or, if the acoustic centers of the drivers do not lie in the same plane. In fact, a truly minimum phase loudspeaker *system* is a rarity. The figure below shows the excess phase curve for a 3-way loudspeaker system measured on-axis. Note that in this figure the frequency scale is linear and extends to about 15 kHz. You execute View Phase Excess to display this curve.



**Multi-driver loudspeaker excess phase curve**

It should be clear from this excess phase curve that, taken as a whole, this loudspeaker system is non-minimum phase. The individual drivers and the crossover network, however, are minimum phase. This is evident from the fact that this curve consists of two straight lines that are smoothly joined at the 2500 Hz crossover point. These two lines have different slopes because the acoustic centers of these drivers do not lie in the same plane. Thus, this loudspeaker system could be made minimum phase over most of the audio range by proper repositioning of the tweeter. Note that there is also another crossover point at around 250 Hz due to the woofer but the low frequencies should not be considered due to frequency resolution limitations of this measurement as indicated by the color bars. You could include the woofer in this analysis if you have access to an anechoic chamber or follow the procedure described in the section below titled "Loudspeaker Near-field Measurements".

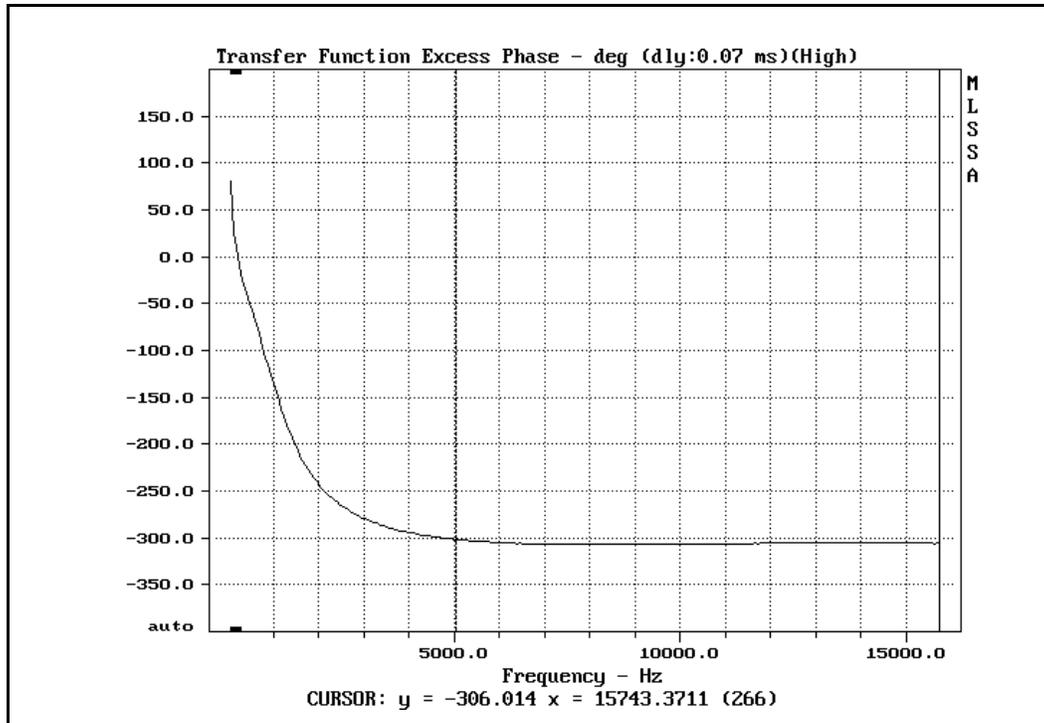


**Excess phase corrected for phase delay of mid-range driver**

The figure above is the same excess phase curve corrected for phase delay measured over the mid-range driver's frequency range. This was done by positioning the marker and cursor to select the 300 Hz to 1500 Hz range then executing the View Phase Delay command followed by the Right arrow and Enter keys. Note that a phase delay correction of 0.49 milliseconds is displayed in the title string above the graphics area.

The figure below is also the same excess phase curve corrected for phase delay measured over the tweeter's frequency range. This was done by positioning the marker and cursor to select the 5000 Hz to 16 kHz range and again executing the View Phase Delay command followed by the Right arrow and Enter keys. A phase delay of 0.07 milliseconds is now displayed. Therefore, the difference in arrival times between these two minimum phase drivers is  $0.49 - 0.07 = 0.42$  milliseconds. Note that this

measurement of time coherence using the excess phase curve is more precise than the one based on the excess group delay curve as described above in the section titled "Loudspeaker Time Coherence". However a quick look at that example showed a time difference of 0.46 milliseconds based on the excess group delay which is not too far from the 0.42 millisecond value shown here.



**Excess phase corrected for tweeter phase delay**

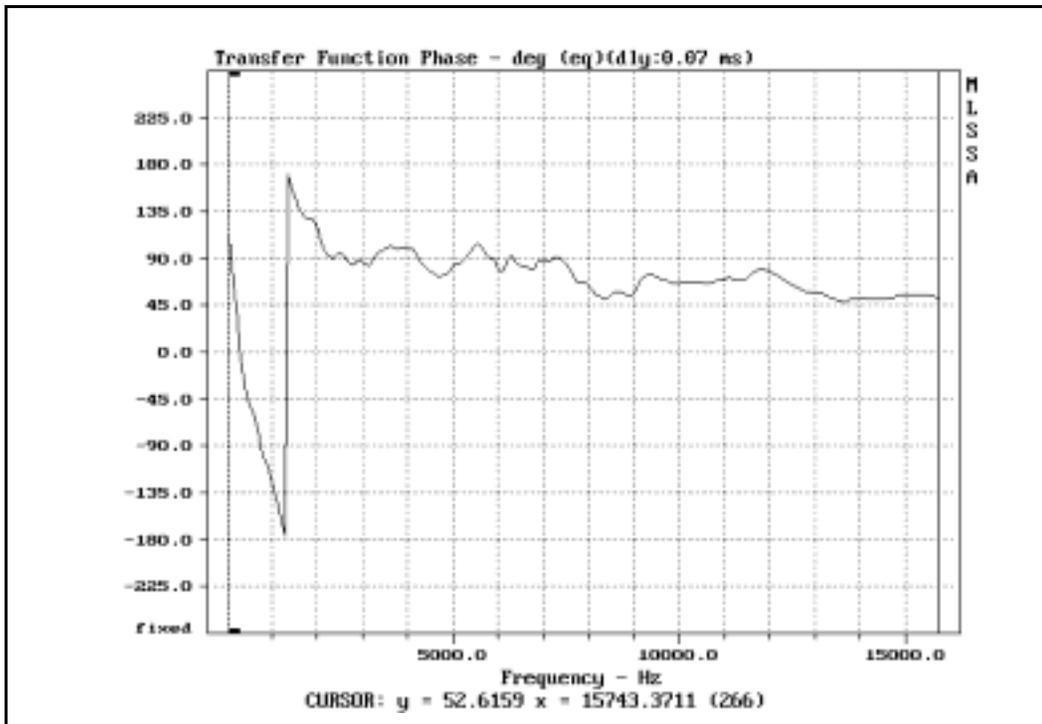
To determine the tweeter's acoustical center, execute the Calculate Acoustic-center command and record the result. The result is the acoustic center of the tweeter measured from the microphone's acoustic center. The acoustic center of small diameter condenser microphones is typically the microphone's diaphragm position. The Calculate Acoustic-center command works only on excess phase curves but it is not necessary to determine the proper phase delay correction as in the example above. A

faster method is to simply position the marker and cursor to select the driver's frequency range followed by the Calculate Acoustic-center command.

The Calculate Acoustic-center command computes the distance measured from the microphone diaphragm to the driver's acoustic center. This distance is calculated from the measured time delay and the speed of sound in air, which can vary with air temperature and humidity. The default speed of sound value used in this calculation is stored in the text file REVERB.INI located in the main *MLSSA* directory. This value applies to an air temperature of 22° C. If the ambient temperature is different, especially when the microphone-to-driver distance is large, the value for the speed of sound stored in this file should be modified for better accuracy. Online calculators can be found on the Internet that compute the speed of sound based on air temperature and also some account for humidity, although humidity has a relatively small effect upon the speed of sound compared to temperature, which has a fairly significant effect. After obtaining the exact speed of sound for your conditions, Quit *MLSSA* and edit REVERB.INI to change the speed of sound value stored there. When you must restart *MLSSA* the new speed of sound value will take effect. Note that the Calculate Acoustic-center command also displays, in parentheses, the time delay in milliseconds. This time delay can be simply multiplied manually by the known speed of sound to get the distance value without editing REVERB.INI.

You can use an accurate tape measure to determine the distance from the baffle to the microphone diaphragm position. The difference between this distance and the acoustic center distance is the tweeter's acoustical center relative to the baffle. Note that for highest accuracy of acoustical center measurements, select the High or Super minimum-phase accuracy level through the View Phase Accuracy command.

After the phase-delay correction has been determined for the tweeter, it can be used to correct the entire measured phase response. Simply execute the View Phase Wrapped or View Phase Unwrapped command and the phase-delay correction will be applied to them. Furthermore, because they are equalized, these ordinary phase curves will not include the effects of the antialiasing filter and will therefore be an accurate measure of the loudspeaker's true phase response. See figure below.



**Equalized and phase-delay corrected phase response**

Note the obvious ripples in the loudspeaker's actual wrapped phase curve as compared to its excess phase curve. The lack of ripples in the excess phase curve is because the drivers and crossovers comprising this loudspeaker system are indeed minimum phase devices. That is, the ripples in the actual phase curve are exactly compensated by equal ripples in the minimum phase curve.

You can display the magnitude response and the wrapped phase response together through the View Bode-plot Actual-phase command. You can also display the delay-corrected absolute group-delay through the View Group-delay Unsmoothed command.

To save the result with all corrections applied, enter the Transfer Save command and answer "y" to save equalized by the Main reference and "y" again to save delay corrected.

## 6.5 Loudspeaker Near-field Measurements

This section explains how to properly measure and combine individual woofer and port (or passive radiator) near-field measurements to obtain the correct overall low frequency anechoic loudspeaker frequency response. The method was originally explained by Don Keele in a 1974 award winning AES paper [16]. Any number of port and radiator near-field transfer function measurements are combined by forming their weighted complex sum. Each measurement is weighted according to the diameter of the port or radiator in question.

There is no need to equalize the near-field measurements to remove the effects of the antialiasing filter since it is common all the port or driver measurements. However, you do need to maintain a common filter bandwidth, filter type, sampling rate and FFT size between the woofer, port or passive radiator near-field measurements. Normally, the easiest procedure is to load setup file LOUDS-NF.SET which programs an antialiasing filter bandwidth of 1 kHz, a sampling rate of 4 kHz and an MLS length of just 4095 points for fast measurements.

Start *MLSSA* with setup file LOUDS-NF.SET by typing

```
MLSSA LOUDS-NF
```

or load this file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the input connector of the board J2 (center phono connector). Center the microphone and position it as close to the diaphragm or dust cap as possible without touching. In the case of open ports, position the microphone's front face flush with the enclosure surface.

Press the Del key to enable the 4095-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level. The level can be lowered by pressing the normal - key or, the - key on the cursor keypad. You should use a much lower drive level than for normal far-field measurements to avoid overloading your microphone preamp. Such overloads cannot be detected automatically by *MLSSA* and can result in severe errors. If you suspect preamp clipping you can temporarily place *MLSSA* into Scope mode through the Acquisition Mode command and measure the raw preamp output. Clipping will show up as a flat-topped waveform. Press the Del key again to turn off the MLS stimulus.

Make the first measurement by executing a Go Once command. Use FFT Execute to compute the near-field transfer function of the port or driver in question. Save the transfer function to an FRQ data file by executing Transfer Save from the frequency domain. Use Go Once from the frequency domain to repeat this procedure for all ports, drivers or passive radiators. Each measurement must be saved to a separate FRQ file. A good idea is to use filenames that start with the same prefix and use a suffix that clearly identifies the port or driver in question later on when it is time to combine the files.

Measure the diameter of each port, driver or passive radiator and record these data on a piece of paper for future reference. The diameters of the drivers should be measured to include about 1/3 of the surround on each side.

If any port or radiator is not circular, you can alternatively use the square root of its area to compute its effective diameter. For example, assume a loudspeaker having an elliptical 6" by 9" passive radiator. These are nominal dimensions but in practice, you should actually measure the diameters involved including about 1/3 of the surround on each side. The area of an ellipse is  $(\pi ab)/4$  where a and b are the minor and major diameter respectively. This area of this passive radiator is then 42.41 sq inches and thus its effective diameter is  $\sqrt{42.41} = 6.51$  inches.

When radiators or ports are mounted in vastly different planes further corrections are required as explained below.

To properly combine all the near-field measurements execute Library Operations Frequency-files Average Weighted Complex. Enter the directory where the near-field measurement files reside followed by the file prefix you selected or just the file prefix if they reside in the current directory. *MLSSA* will list all the FRQ files beginning with the specified prefix. Use Spacebar or the left mouse button to mark all the near-field FRQ files you just measured. Press the Ctrl-Enter key combination or the right mouse button to exit the filebox. *MLSSA* will next display each filename in turn and prompt you to enter the weighting factor for each. Enter these as the diameters you had previously recorded making sure each corresponds to the correct filename. Finally, enter an output filename to save the composite near-field low frequency response.

The resulting anechoic low frequency response can either be spliced to a far-field Transfer-Function or Sensitivity measurement using the Library Operations Frequency-files spLice command or, displayed as an offset overlay on top of the far-field anechoic response.

The Library Operations Frequency-files spLice command will prompt you for the splice frequency in Hz. If you enter nothing and press the Enter key, *MLSSA* will select a splice frequency determined by the true resolution of the high frequency far-field measurement file. This command adjusts the decibel level and phase delay of the low frequency near-field file to smoothly connect it to the high frequency far-field file. The output file contains the spliced measurements, which can be loaded into the frequency domain and displayed through Transfer Load.

Note that in order to minimize the output file size, the spLice operation normally selects the smallest possible output FFT size that will still give adequate frequency resolution down to 20 Hz. If you want to maximize the output file frequency resolution and hence its FFT size, enter the splice frequency manually as a negative number.

You can optionally apply *MLSSA*'s inverse FFT operation (Library Operations Frequency-files IFT) to a spliced transfer function to obtain the full-range impulse response saved in a TIM file. You can then compute a CSD plot or step response from the computed impulse response.

In case a port or woofer is mounted in a vastly different plane than the others, you need to account for the acoustical delay between it and them. This is most easily accomplished in the frequency domain through the View Phase Delay command. Suppose a passive radiator is mounted at the rear of the enclosure with the woofer in front and, suppose further that the enclosure is 2 feet deep. This requires that a delay of about 2 ms be added to the passive radiator output before combining it with the other radiators. To do this, load the rear port's FRQ file and execute View Phase Delay and enter -2 ms. Next save the result to the original file and answer 'yes' when asked to apply the phase delay correction to the saved data. You can now combine the FRQ files according to the procedure given above.

Here are some points to remember when making near-field measurements. Near-field loudspeaker measurements correctly predict the true anechoic far-field response only in the driver's piston range of operation below roughly  $f = 4290/d$  Hz where  $d$  is largest dimension of the largest diaphragm or port in inches.

Another limitation of the near-field method is that it does not account for sound diffracted from the enclosure edges. Diffraction will affect the desired far-field response but will not affect the measured near-field response. While this is a theoretical limitation, in practice, diffraction effects from typical enclosures are small below about 250 Hz because the wavelengths involved at such low frequencies are typically much larger than the enclosure dimensions.

In the case of unusually large enclosures, cognizance should be taken of possible errors due to diffraction effects. Since diffraction tends to increase loudspeaker output in the forward direction with increasing frequency, diffraction errors will typically result in a near-field response that is lower than actual far-field response near the splice frequency. The effects of diffraction are compensated near the splice frequency, however, by the spLice command, which automatically adjusts the level of the near-field measurement to match the level of the far-field measurement at the splice frequency. However, since diffraction has less effect at frequencies below the splice frequency its net effect, if present, will be to cause the bass roll-off rate of the loudspeaker to look a little better than its actual far-field anechoic roll-off rate. In other words, a loudspeaker's bass response will appear more extended than reality when measured with the near-

field method if the enclosure dimensions are large enough to cause significant diffraction in the region at and below the selected splice frequency. Diffraction effects can be minimized by measuring the far-field response of a large loudspeaker in a larger room elevated on a stand in order to achieve a lower valid far-field frequency limit and hence allow a lower splice frequency to the near-field measurement.

## 6.6 Loudspeaker Polar Response

The polar response of a loudspeaker is determined by making successive "anechoic" transfer function or sensitivity measurements at equally spaced angles and then using *MLSSA's* Frequency File Display waterfall mode to display the results. The best way to perform polar measurements is to mount the loudspeaker on a turntable with the tweeter's acoustic center approximately centered on the turntable's axis of rotation.

Start *MLSSA* with setup file POLAR.SET by typing

```
MLSSA POLAR
```

or load this file through the Setup Load command.

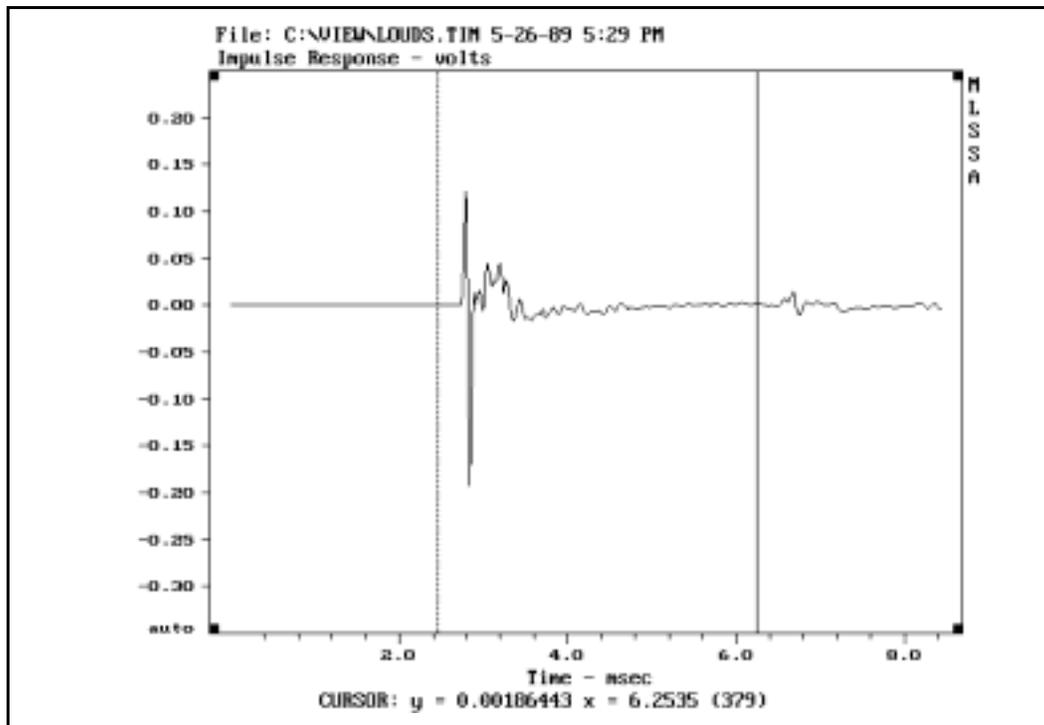
Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the input connector of the board J2 (center phono connector). Position the measurement microphone 2 meters directly in front of the loudspeaker.

Press the Del key to enable the 65535-point MLS stimulus and press = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or

the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Make the first measurement by executing a Go Once command.



#### Selecting the "anechoic" segment for polar response measurements

*MLSSA* will calculate and display the impulse response of the loudspeaker/room combination. Move the cursor so it is positioned about 0.25 milliseconds before the initial transient and press function key F9 or the right mouse button to set the marker to this position (see figure above). Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Move the cursor to a point about 0.25 milliseconds before the first reflection (usually due to the floor). Note that this setup is slightly different from the normal "anechoic" procedure of 6.1 above. These 0.25 millisecond guard bands allow for up to 3 inches of eccentricity in the

loudspeaker/turntable combination and have no effect on magnitude response measurement accuracy.

To check, rotate the loudspeaker 180 degrees and execute Go Once again. The "anechoic" segment should still fall between the marker and cursor positions. If not, re-center the loudspeaker on the turntable and retest at both the zero and 180-degree positions.

An alternative to the use of large guard bands and the need to carefully center the loudspeaker on the turntable is to place *MLSSA* into its Asynchronous-correlation mode through the Acquisition Mode command. In this mode, *MLSSA* automatically forces the first arrivals to the same time position even when the microphone-to-loudspeaker distance changes greatly between measurements. The main disadvantage of using this mode is longer measurement times due to the extra computations required to properly align the arrival times on the graph. You should perform one manual measurement using Go Once in this mode before setting the marker and cursor positions. Even when in Asynchronous-correlation mode, however, it's still a good idea to provide some guard bands of perhaps 0.1 milliseconds.

Rotate the loudspeaker back to the 0 degree position. Execute FFT Execute to enter the frequency domain. Execute Transfer Autaname Path and set the drive and directory you wish to store the polar response data. Execute Transfer Autaname Root and set the root name for the files that will hold the polar data. You are now ready to collect the polar response data.

Execute Macro Load and load the macro file POLAR.MAC. Invoke the Macro Execute command to start the data collection macro. This macro first initializes the Autaname number to -01. It then repetitively performs a Go Once followed by Transfer Save followed by Right arrow and Enter to save the polar response data to autonamed frequency files. It also outputs a positive TTL pulse on J4 pin-2 (through the Macro Command Signal command) to signal an automated turntable to rotate to the next position. If you are using a manual turntable, rotate it by 10 degrees and press Space to continue to the next measurement. Be sure to move away from the loudspeaker

while a measurement is in progress to prevent body reflections from influencing the results. Repeat the procedure until the macro is completed.

As an alternative to pressing Space after every turn, you can connect a push-button switch between J4 pin-5 and J4 pin-4 (digital ground). Then you just momentarily press this switch after rotating the turntable to have the macro continue to the next measurement.

**WARNING:**

Applying more than 5 volts to J4 pin-5 could result in damage to the board. For static discharge protection, you should connect a 5.1-volt zener diode across the switch. Connect its anode to digital ground and the cathode to the TTL input side. This arrangement serves to clamp any stray voltage between -0.6 and +5.1 volts to prevent static damage.

If you have an automated turntable which supplies a positive TTL pulse *after* it has rotated to the next position and come to a complete stop, simply connect this signal to J4 pin-5 for totally automated polar measurements.

To plot the polar data just acquired, exit to the time domain through the Exit command and invoke the Waterfall Execute command. You can also plot the data with smoothing through the Waterfall Octave-smoothing command.

Using the Waterfall Reference command you can select the on-axis slice (slice index # 1 with the current waterfall setup parameters) as the reference slice. Then when you invoke Waterfall Execute all the off-axis slices will be equalized to the on-axis reference slice thus showing only the changes in the speaker's frequency response relative to its on-axis frequency response. To again plot the actual frequency response at each angle, execute Waterfall Reference again and press the Enter key to disable equalization.

You can modify POLAR.MAC for different angular increments and different starting angles using the MAKEMAC utility. Then use the Waterfall Z-axis commands to reflect any changes you made to the starting angle or increment.

If you wish your waterfall plotted in units of dB-SPL/watt, you must make your measurements in the Sensitivity FFT mode and have selected a calibrated microphone.

The Waterfall Text-export-file command can be used to export the polar data to a text file which can then be imported into MS Excel to produce circular polar plots. For details see section 3.25.5 above titled "Exporting Waterfall Data to Excel — Circular Polar Plots".

## **6.7 Loudspeaker Sound-power and Directivity**

Macro file SNDPWR.MAC (the text file version is SNDPWR.DOC) allows convenient determination of loudspeaker sound power response and directivity index from free-field polar response measurements made according to section 6.6 above. The sound power response of a loudspeaker is the power average of its frequency response measured in all directions. This macro, however, determines power response from polar response measurements taken in just two speaker orientations: one with the loudspeaker in its normal orientation (standing up) and the second with the loudspeaker lying on its side. There are 48 measurements in all taken in 15-degree increments. These data are weighted prior to averaging in order to correctly calculate the total sound power output. Then the directivity index is computed as the ratio of the on-axis frequency response to the sound power response.

The setup file and procedure for making sound power response and directivity measurements is the same as that described in the section above titled "Loudspeaker Polar Response". See SNDPWR.DOC for additional information.

## 6.8 Loudspeaker Impedance

Impedance measurements using a power amplifier are greatly facilitated using the *MLSSA Remote Controlled Analog Interface (RCAI)* which is highly recommended for this purpose. However, the following procedure assumes you do not own or have access to an *RCAI*.

*MLSSA* measures the complex impedance indirectly by measuring the transfer function of a voltage divider formed by connecting an accurately known resistance in series with the driver or loudspeaker to be measured. There is no requirement for constant voltage or constant current conditions. Any value series resistor can be accommodated but 75 ohms is recommended for most measurements. Values down to 10 ohms can also be used with a power amplifier to increase the effective drive current for a given drive voltage.

*MLSSA* can measure driver impedance without a power amplifier by using a Y-adaptor having two RCA phono plugs connected to a common RCA phono jack. You connect one plug of the Y-adaptor into the *MLSSA* white-MLS output (J3) and the other plug into the *MLSSA* input (J2), thus forming a loopback connection. The common phono jack at the other end of the Y-adaptor connects to the driver to be measured. Test clips are available with a female BNC termination. By attaching a phono-jack-to-male-BNC adaptor to the BNC-terminated test clips, you can easily connect them to the common end of the Y-adaptor using a standard audio patch cable. The cable length should be no longer than 1.5 meters to avoid excessive cable resistance and inductance.

Direct impedance measurements are recommended and preferable for several reasons. Because no other electronics are involved, the possibility of ground loops is eliminated thus ensuring low noise measurements. In addition, *MLSSA 2000* cards contain precision 1-watt, 75.5-ohm resistors in both the white and pink MLS outputs. These resistors are accurate to 0.1% to insure the highest possible accuracy of *MLSSA* impedance measurements when using the direct method.

**WARNING:**

All standard setup files intended for measuring impedance assume that the series resistor is exactly 75.0 ohms. DO NOT enter 75.5 ohms into the FFT Mode Impedance command as this will result in reduced accuracy. The precision 75.5-ohm series resistors found on all on *MLSSA 2000* cards are intentionally 0.5 ohm high in order to compensate for the input impedance of the *MLSSA* antialiasing filter, which is effectively in parallel with the 75.5-ohm series resistor.

The output current capability of *MLSSA 2000* cards is 100 ma for both the pink and white outputs. This permits you to make direct impedance measurements using the maximum *MLSSA* output voltage of 5.23 volts as set by the Acquisition Stimulus Amplitude command. Note that impedance measurements are best made using the normal or white-MLS output on J3 (bottom phono connector) but you can use the pink-filtered MLS output on J1 (top phono connector) if desired.

**WARNING:**

If you do not own a *MLSSA 2000* card, do not attempt to use a stimulus amplitude higher than 1 volt when following this procedure. Doing so, may cause current limiting in the output stage and result in erroneous impedance measurements. In addition, the 75-ohm output series resistor on older *MLSSA* cards is rated at only 1/4 watt has only a 1% tolerance.

There are three setup files provided for direct impedance measurements: IMP1K.SET, IMP20K.SET and IMP20KHR. IMP1K.SET measures impedance from 1 Hz to 1000 Hz and should be used to measure low frequency driver impedance of woofers or mid-range units. IMP20K.SET measures impedance from 4 Hz to 20,000 Hz at 4 Hz frequency resolution and should be used to measure tweeters or to measure the full range impedance of loudspeaker systems or drivers. IMP20KHR provides full-range impedance measurements at 1 Hz frequency resolution. The procedure below is given for IMP1K.SET but you can substitute IMP20K.SET or IMP20KHR if desired.

Start *MLSSA* with one of the impedance setup files, for example, IMP1K.SET by typing

```
MLSSA IMP1K
```

or load the setup file through the Setup Load command.

Execute the FFT Bypass command to enter the frequency domain. Execute the Reference Go command to perform and store a reference loopback measurement. Check that the test leads are open when making reference loopback measurements.

For higher accuracy impedance measurements, you can optionally correct for the residual impedance of your test leads as follows: Short together the ends of the test leads. With the test leads shorted together, execute Go Once followed by Reference Auxiliary Store. The residual complex impedance of your test leads is now stored as the Auxiliary reference. The stored complex cable impedance will automatically be subtracted, on a complex basis, from all subsequent impedance measurements.

Connect the test leads to the driver to be measured. Execute the Go Once command to measure the complex impedance of the driver or loudspeaker system. You can display a Nyquist plot of the impedance through the View Nyquist-plot command. You can also display its Bode plot (magnitude and phase) through the View Bode-plot Actual-phase command.

If you decide to use a power amplifier, the following guidelines should be followed. Use only common ground power amplifiers having relatively low output power capability. The use of high power audiophile amplifiers especially those having floating or bridged output circuits should be avoided. The high power capability of these amplifiers can result in inadvertent damage to *MLSSA's* antialiasing filter or to the amplifier. In addition, floating amplifier output circuits can result in high ground-loop currents since *MLSSA's* input and output circuits are single ended. Also, be sure the reference loopback measurement includes the power amplifier and that the power amplifier has a low output impedance relative to the series resistor used. A series resistance of 75 ohms is recommended. However, lower values can be used down to about 10 ohms, to increase the effective drive level for a given drive voltage.

Connect the selected series resistor, of suitable power rating, directly to the hot terminal of the power amplifier output. Connect the free end of the power resistor to a piece of heavy gauge wire with an alligator clip at the opposite end for connecting to the driver.

Connect a second piece of heavy gauge wire, of equal length, to the ground terminal of the amplifier output with a second alligator clip for connecting to the driver. To prevent ground loops, it's advisable to connect another piece of heavy gauge wire from ground terminal of the power amplifier to the L-bracket of the *MLSSA* card or, directly to the frame of the PC.

Connect the hot lead of the *MLSSA* input (J2) to the series power resistor on the side opposite the power amplifier's hot terminal, that is, where the series resistor and hot test lead join together.

Using an audio patch cable, connect the *MLSSA* output (J3) to the amplifier's input and power up the amplifier.

Connect the loudspeaker or driver to the test leads.

Execute the FFT Mode command and select the Impedance option. Enter the series resistance of the out-board series resistor. For highest accuracy, it is advisable to measure this resistor with a high quality DVM.

Press the Del key to enable the MLS stimulus and then use the = key or the + key on the numeric keypad to slowly increase the stimulus level until the MLS noise is clearly audible from the driver or loudspeaker.

**NOTE:**

After setting a stimulus level appropriate for your power amplifier you can save it to IMP1K.SET or another setup file through the Setup Save command. Thereafter, there will be no need to re-adjust the stimulus level each time upon starting *MLSSA*.

Disconnect the unit under test and execute the FFT Bypass command to enter the frequency domain. Execute the Reference Go command to perform and store a reference loopback measurement. Always check that the test leads are open when making reference loopback measurements.

Reconnect the unit under test and Execute Go Once to measure its complex impedance.

**NOTE:**

To make impedance measurements over other bandwidths use the Acquisition Bandwidth command to change and then redo your reference measurement. *Any change in bandwidth, filter type, sampling rate or FFT size requires a new reference measurement.*

## 6.9 Loudspeaker Incoherency Distortion

Incoherency measures the fraction of the output signal that is not linearly related to the input signal. Mathematically incoherency equals  $\sqrt{1 - \text{coherence}}$ . More importantly, incoherency measures everything that is not coherent with the input signal including hum, noise, distortion, driver noises (buzz and rub) as well as reverberation. Because reverberation will increase incoherency, you generally need to eliminate or reduce reverberation when making incoherency measurements. Reverberation can be attenuated either by making measurements in an anechoic chamber or by making near-field measurements. A near field measurement is performed by placing your microphone within 0.5 inches of the driver. Because the sound source and microphone are so close together the SPL of the direct sound is high enough to mostly overwhelm the reverberant field. Thus near-field incoherency measures primarily driver intermodulation distortion assuming the driver is free from buzz and rub noises.

A disadvantage of the near field method is that the results may be sensitive to microphone position due to cone breakup. That is, different parts of the driver's radiating surface may show different distortion characteristics. This effect can be avoided by performing the measurement some distance from the loudspeaker in an anechoic chamber or, by performing it in a small test box having a very short reverberation time.

Start *MLSSA* with setup file INCOHER.SET or load this setup file from the Setup menu. This setup file places *MLSSA* in the Incoherency FFT mode, selects a 16383-point MLS

and provides 16 cycles of concurrent pre-averaging to reduce noise and emphasize distortion.

Unless you have access to an anechoic chamber, position the measurement microphone in a quiet environment within 0.5 inches of the driver you wish to measure. The driver to be measured can also be part of a multi-driver loudspeaker system.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Note that incoherency is a relative measurement so there is no need to make a loopback measurement or for absolute microphone calibration data. Press the Del key again to turn off the stimulus.

Make the first measurement by executing a Go Once command. *MLSSA* will calculate and display the impulse response of the driver. Execute the View Energy-time-curve Unfiltered Unwindowed command. The purpose of this step is to insure that no residual reverberation will contaminate the incoherency calculation. Observe that the energy level decays precipitously but then reaches a point where it remains relatively constant. Move the cursor so that it is positioned just after the initial reverberant decay. The data to the right of the cursor should be noisy but more-or-less uniform. The exact position is not critical as long as it is beyond the reverberant decay region. Execute the View Data Unfiltered command to restore the original impulse response data.

Execute the FFT Execute command. *MLSSA* will enter the frequency domain and display the incoherency in dB. You can also display incoherency as a percentage through the View Magnitude Percent command. Note that incoherency will typically rise outside the driver's intended frequency range. This is normal and results from the fact that intermodulation components are generated that fall outside the driver bandwidth due to mechanical non-linearity while the out-of-band linear components will typically be

attenuated by crossovers or by the driver's linear frequency response shape. Therefore, when interpreting incoherency measurements you should ignore data falling outside the intended frequency range of the driver.

Note that you can store the incoherency as an overlay by executing Overlay Store then return to the time domain and execute FFT Mode Transfer-function followed by FFT Execute and Overlay Plot. *MLSSA* will now show both the system transfer function and its incoherency together. To re-configure for further incoherency measurements, return to the time domain and execute FFT Mode More Incoherency. FFT Execute will again display incoherency only.

### 6.10 Loudspeaker Harmonic Distortion

*MLSSA* will measure total harmonic distortion with an external low-distortion sine wave source. For this measurement, *MLSSA* emulates a standard FFT analyzer. The spectral lines created by harmonic distortion are measured and added together to calculate the odd (OHD), even (EHD) and total harmonic distortion (THD). Individual harmonics can also be isolated and measured by direct cursor readout.

Harmonic distortion of loudspeakers should not be measured in a normal room in the far field because the room's frequency response, which normally contains wild swings, will boost or attenuate the harmonics differently than the fundamental. Therefore, measure harmonic distortion either in an anechoic chamber or in the near field. A near field measurement is performed by placing your microphone within 0.5 inches of the driver. Because the sound source and microphone are so close together the SPL of the direct sound is high enough to mostly overwhelm the reverberant field.

A disadvantage of the near field method is that the results may be sensitive to microphone position due to cone breakup. That is, different parts of the driver's radiating surface may show different distortion characteristics. This effect can be avoided by performing the measurement some distance from the loudspeaker in an anechoic chamber or by performing it in a small test box having a very short reverberation time.

Start *MLSSA* with setup file *DISTORT.SET* by typing

```
MLSSA DISTORT
```

or load the setup file through the Setup Load command.

The setup file *DISTORT.SET* is adequate for measuring harmonic distortion using excitation frequencies of 20 Hz and higher including all harmonics up to 25 kHz. This setup file puts *MLSSA* in Scope acquisition mode and sets the Normal FFT mode. The acquisition length is 32768 samples and the FFT size is 32768 points. The window type is a full Blackman-Harris window to minimize spectral leakage.

Connect a low-distortion sine wave generator to a power amplifier to drive the loudspeaker or driver to be measured. Connect your microphone preamp output to the *MLSSA* input J2 (center phono connector).

Execute the Go Once command. After autoranging, *MLSSA* will acquire 32768 points of the sine wave. Note that *DISTORT.SET* sets the marker and cursor to select the entire 32768 points of the sine wave. If this were not the case, the distortion results would likely be wrong especially at low fundamental frequencies. Execute the FFT Execute command and *MLSSA* will compute and display a windowed 32768-point FFT of the sine wave plus any harmonics present. When there is no harmonic distortion, only a single spectral line will be visible at the fundamental frequency. When harmonic distortion is present, other spectral lines will be visible at integer multiples of the fundamental frequency, these lines are the harmonics of the fundamental.

You determine odd, even and total harmonic distortion through the Calculate Distortion command and press TAB. This command computes distortion for all harmonics up to the antialiasing filter cut-off frequency. The result is true harmonic distortion because noise falling between the harmonics is not included in the calculation. Note however that sometimes leakage effects can cause a large spurious peak to appear at DC or zero frequency. If this spurious peak is higher than the sine wave's fundamental frequency peak, the Calculate Distortion command will choose this spurious peak as the fundamental and give the wrong answer. If this happens move the cursor to a point

just past this spurious peak and press F9 to set the marker there. From now on, the Calculate Distortion command will skip over the spurious peak when searching for the fundamental frequency.

The Calculate Distortion command gives you the option of entering the maximum harmonic number to be used in the harmonic distortion calculation. For example, entering 5 followed by the Enter key tells *MLSSA* to include all harmonics up to and including the fifth harmonic but no higher harmonics. If you enter 0, 1 or nothing followed by the Enter key *MLSSA* uses all harmonics up to current operating bandwidth and displays the maximum harmonic number in the harmonic distortion result. If you enter nothing and press the Tab key *MLSSA* uses the last maximum harmonic number you entered. For example, if you execute Calculate Distortion and enter 5 followed by the Enter key then later execute Calculate Distortion and press the Tab key *MLSSA* will calculate harmonic distortion up to the 5th harmonic. This feature is useful if you use the *DISTORT.MAC* macro to repeat the Calculate Distortion command for different fundamental frequencies since the initial maximum harmonic number you enter before starting the macro is used repeatedly. Note also that the maximum harmonic number is written out to the distortion export file if one is opened through the Calculate Export-distortion-file Open command (see below). The fifth column in this file indicates the maximum harmonic used for each measurement.

As the fundamental frequency increases the maximum possible harmonic number decreases because it is limited by the measurement bandwidth. With the Calculate Distortion command, the maximum harmonic number actually used and displayed may be lower than the one specified when the fundamental frequency is high. For example, if you enter nothing and press Enter all harmonics up to the measurement bandwidth are included. As the fundamental frequency is increased, the maximum harmonic number will decrease. Therefore, you may want to set the maximum harmonic number to a low value such as 5 to maintain the same maximum harmonic number over a wide frequency range. For example, if the measurement bandwidth is 20 kHz you can include the fifth harmonic for all fundamental frequencies up to 4 kHz.

You can read out the level of individual harmonics by first positioning the marker at the fundamental frequency. Since the cursor is already at the fundamental frequency just

press F9 or the right mouse button to move the marker there. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Next, move the cursor to the third harmonic and press Alt-y to display the decibel difference between the fundamental and the third harmonic. This result is the amount of third harmonic distortion. Before executing the Calculate Distortion command again, be sure to move the marker back to its original position. If that was the left-hand display edge, simply press Ctrl-PgDn.

To have *MLSSA* express harmonic distortion as a percentage execute View Magnitude Linear and repeat the Calculate Distortion command.

Calculate Distortion results can optionally be exported to a text file through the Calculate Export-distortion-file Open command. Specify a filename to export the distortion results. After an export file is opened, the Calculate Distortion command will automatically write out the fundamental frequency as well as the total, odd and even harmonic distortion as well as the maximum harmonic number to this file. You can then change your oscillator frequency to a new value execute Go Once followed by Calculate Distortion TAB to measure and save the harmonic distortion for the new fundamental frequency etc. A macro file named DISTORT.MAC will repeat these commands for you indefinitely until you press Esc. Use Macro Load to load this macro and Macro Execute to start it. When as many frequencies are measured as required, press Esc to stop the macro and close the export file through the Calculate Export-distortion-file Close command. Note that the export file is also closed automatically upon quitting *MLSSA*. You can load this text file into a word processor to print the results or, into a spreadsheet program to graph them.

## **6.11 Microphone Response Correction**

*MLSSA* will allow you to import microphone frequency response taken from the microphone's calibration sheet as a reference for correcting out deviations from flatness. This is generally not required or recommended for the highest quality instrumentation grade microphones, which are normally flat within 1 dB. In such cases, errors in the calibration curve or errors in reading it may be large enough to defeat any possible benefit. In addition, if you use the Auxiliary reference for microphone response

correction it will not be available for other important purposes. A rule of thumb is to correct microphone response only if it deviates by 1 dB or more from flatness in the frequency range of interest.

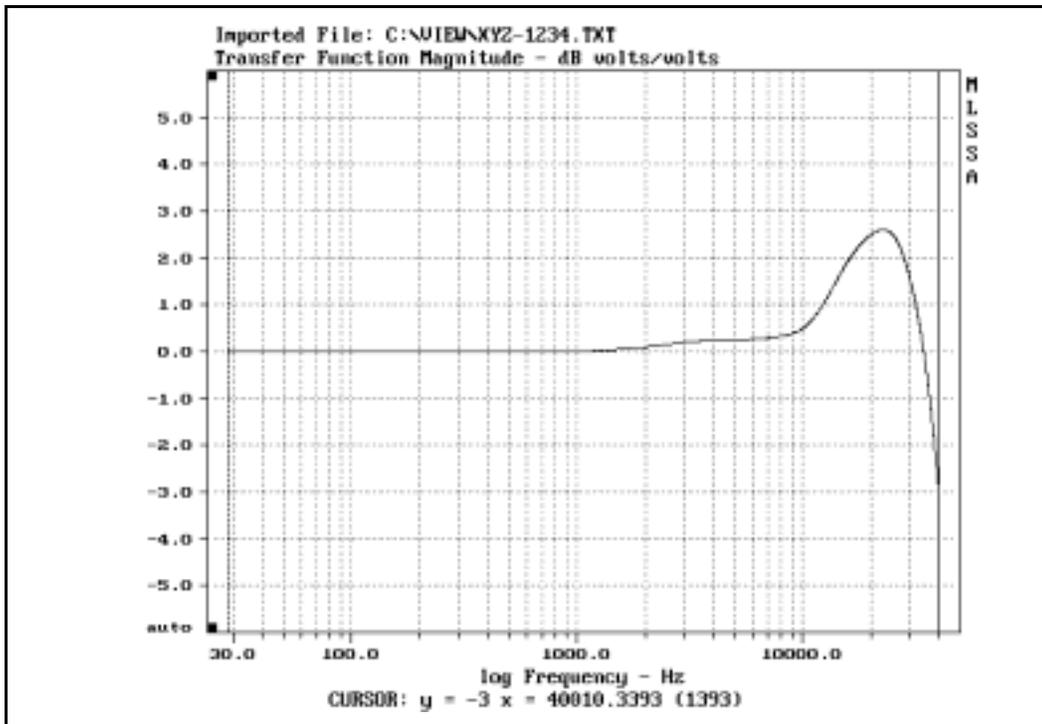
The example below shows a text file created to store the microphone frequency response taken from its calibration curve. The first line, enclosed in double quotes, is normally used to identify the microphone. It's also a good idea to select a filename that readily identifies the microphone. Most microphones are very flat at low frequencies so there normally is no need to include low frequency data points. In any event, the low frequency microphone response should average out to 0 dB. If not, you should subtract any decibel offset from the calibration curve before creating the correction file. The reason for this is to maintain correct microphone sensitivity calibration, which is usually referenced to some low frequency (e.g. 400 Hz or 1000 Hz). Also, at higher frequencies, be sure to pick enough points off the calibration curve to accurately describe any wiggles. Normally, as the example below shows, very few points are needed to get an accurate response since *MLSSA* will spline interpolate the data when it is imported.

**IMPORTANT:**

When creating a text file for import using a text editor or word processor be sure to add at least 1 leading space at the beginning of each line, that is, before entering any numerical data. If you are using a word processor also be sure you save the file as a plain text file, that is, without special word-processor formatting characters.

```
"Microphone: XYZ Company Model 1234, serial # 10777"  
"Hz" "dB"  
1000.00, 0.00  
2000.00, 0.1  
3000.00, 0.2  
5000.00, 0.25  
8000.00, 0.33  
10000.0, 0.5  
15000.0, 1.75  
20000.0, 2.5  
25000.0, 2.50  
30000.0, 1.5  
40000.0, -3.0
```

To test for accuracy of the interpolation, first import the text file through the Transfer Import Text command from the frequency domain and press 'n' to decline marking the imported data as "equalized". It's also a good idea to first set *MLSSA*'s bandwidth to 40 kHz through the Acquisition Bandwidth command to display the microphone's entire frequency response curve. The figure below shows the result of importing the text file above. If the imported curve closely matches the microphone's calibration curve, you can load it as an Auxiliary reference through the Reference Auxiliary iMport Text Transfer-function command. Thereafter, *MLSSA* will correct out microphone response errors automatically. Note that if you subsequently change the acquisition bandwidth or FFT size you will need to re-import the same text file in order to re-interpolate it to insure the same frequency spacing as the main data display.



**Imported Microphone Frequency Response**

If too few points are entered in the text file, interpolation may result in wild swings around some data points. If this occurs you need to pick more points off the calibration curve in those areas where the interpolation is oscillating and try again.

Note that if phase response data on your microphone is available, you can enter it also into the text file (in degrees) following the dB response magnitude. Thus, it is possible to correct for microphone phase as well as magnitude response errors. However, phase response is not usually recorded on microphone calibration data sheets.

## 6.12 QC PASS/FAIL Testing

In the frequency domain, *MLSSA* provides for comprehensive automated QC PASS/FAIL testing. Upper and lower QC limits curves are defined in a special text file. See the section titled "QC PASS/FAIL Functions" in the Reference chapter of this manual for details on how to create this file.

Once you have created a QC limits file, you load it from the frequency domain through the QC-limits Load command. The QC-limits Compare command initiates QC PASS/FAIL operation in addition to actually plotting both the upper and lower QC limits curves. The comparison is restricted to the frequency range defined by the current cursor and marker positions. If there is a failure, *MLSSA* beeps and a message appears indicating what caused the failure. If there is no failure and you have defined the incoherency distortion limit column, *MLSSA* goes on to automatically compute incoherency and compares it to the upper incoherency limit. If all checks pass the message "QC PASSED!" appears. Note that the QC-limits Go commands initiate measurements followed automatically by the QC-limits Compare command.

When graphics display blanking is enabled (Ctrl-F3), the final QC PASS/FAIL result is displayed in larger type for easier visibility from a distance. In addition, the result is framed by a broad red bar when it's a failure and a broad green bar when it's a pass. This command also disables all graphics commands thus preventing the operator from accidentally moving the cursor or marker positions that define the frequency range of the QC Compare command. To restore the normal graphics display and graphics functions press Ctrl-F3 a second time. With the screen blanked, you can also use a macro and the Macro Command Headline command to display other messages in large type in place of the graphics field.

Even though *MLSSA* will optionally compute and check incoherency, it and its upper limit curve are not normally displayed. To actually plot incoherency and see its upper limit curve you must return to the time domain and select the Incoherency FFT mode. Next, enter FFT Execute to return to the frequency domain. Now if you execute QC-limits Compare, *MLSSA* will compare and plot the incoherency limit only. This option intended only for debugging purposes.

You can additionally use the Reference Auxiliary Load command to load a reference measurement of the desired loudspeaker or driver response using a measurement of a reference unit. Thereafter, *MLSSA* will display the decibel difference or error between the reference unit and the unit under test. Use of the Auxiliary reference to store the desired target response is recommended because it results in simpler QC limits curve shapes which are easier to create and can even be used for different speakers or drivers. For example, if your error specification is simply +/-5 dB over the whole test frequency range then the upper limit curve is simply a flat line at +5 dB and the lower limit curve is another flat line at -5 dB regardless of the actual driver or speaker response. The incoherency test, if selected, is unaffected by the presence of any reference that may be present.

To completely automate the QC test procedure you can create a macro that guides the operator through the procedure. You can also use the Transfer Autaname commands to set up a directory and root name to save the data to FRQ files. See the section of the Reference chapter titled "Automatic Filename Generation" and "Creating and Running Macros" for more information. To save the PASS/FAIL result as a comment in the data file you can press the Right arrow key at the "SAVE comment: " prompt of the Transfer Save command.

### 6.13 Room Response — Adaptive Window

For best subjective equalization, you should measure room response using *MLSSA*'s Adaptive Window feature rather than using an RTA and pink noise source. *MLSSA* measures loudspeaker/room response by measuring a long-duration impulse response and then applying an adaptively windowed FFT to obtain the transfer function. An adaptively windowed FFT essentially varies the length of the time window with frequency such that a short time window is used at high frequencies while a gradually longer time window is used at the low frequencies. See the Reference section for more information on the Adaptive Window feature.

After computing the adaptively windowed transfer function, the magnitude is smoothed to reveal the 1/3 octave adaptively windowed room response. You can also determine

the spatially averaged adaptively windowed room response by using the Go Spatial-average command from the frequency domain.

Start *MLSSA* with setup file ADAPTIVE.SET by typing

```
MLSSA ADAPTIVE
```

or load the setup file through the Setup Load command from the time domain.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you should measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone within the intended listening area within the room.

**NOTE:**

If the room you are attempting to measure is unusually noisy, you can execute Acquisition Pre-Average Cycles and enter up to 16 cycles of pre-averaging to improve the signal-to-noise ratio (SNR) of your measurements by up to 12 dB.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging the loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Execute the Go Once command. *MLSSA* will compute and display the impulse response of the loudspeaker/room combination.

Use the mouse or arrow keys to position the marker just before the first arrival. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Press F9 or the right mouse button to position the marker there. Press Ctrl-F9 and enter 50 to move the cursor just 50 ms ahead of the marker position. You have now selected the first 50 milliseconds of the loudspeaker/room response as the high-frequency window length. At low frequencies, however, the Adaptive Window gradually expands to include more and more room reflections but can never become larger than 1/2 the FFT size. ADAPTIVE.SET selects an FFT size of 32768 points, which sets an effective low frequency time window of about 270 milliseconds. To display both time windows execute FFT Window Display.

Next, enter the frequency domain through the FFT Execute command. The true low-frequency resolution is indicated by the length of the color bar located in the lower left-hand corner of the graphics area. The true high-frequency resolution is indicated by the length of the color bar located in the upper left-hand corner of the graphics area.

The room response at the current microphone position is displayed as a Voltage Sensitivity curve in units of dB-SPL/200mV. This curve shows you at each frequency the SPL you would measure with a 200 mV RMS sinusoid input to the system. The 200-millivolt reference was chosen because it is common calibration input level used in Dolby surround sound systems. You can choose another reference input voltage by returning to the time domain, executing the FFT Mode Voltage-Sensitivity command and entering a different voltage reference in millivolts. After FFT Execute, the Voltage Sensitivity curve will reflect any change you made.

This magnitude response is smoothed to 1/3 octave. You can change to a different smoothing fraction by executing View Magnitude sSmoothed Decibels and entering another smoothing fraction.

Use the Go Spatial-average command to determine the spatially averaged room response taken at several (usually 4) representative positions in the intended listening area. The Go Spatial-average command computes the unweighted power average of the specified number of measurements and pauses between measurements to permit re-positioning of the microphone. Press any key to continue or, for remote locations,

connect a momentary contact switch between J4 pin-5 and J4 pin-4 (digital ground) and press this switch to continue.

**WARNING:**

Applying more than 5 volts to J4 pin-5, could result in damage to the board. For static discharge protection, you should connect a 5.1-volt zener diode across the switch. Connect its anode to digital ground and the cathode to the TTL input side. This arrangement serves to clamp any stray voltage between -0.6 and +5.1 volts to prevent static damage.

The Go Spatial-average command is greatly facilitated using the *MLSSA Remote Controlled Analog Interface (RCAI)* which includes a four-input microphone multiplexer. When connected to *MLSSA*, the *RCAI* automatically selects each connected microphone, in turn, when the Go Spatial-average command is invoked.

The Go Spatial-average command normalizes the level measured at each location to the reference location, which is assumed to be the first location measured. Normalization is necessary to prevent louder locations from weighting the spatially averaged frequency response in their favor. Press TAB to use the Dolby standard frequency range for normalization or, select your own custom range using the marker and cursor. *MLSSA* normalizes each measurement to the pink-weighted power-average calculated over the selected frequency range. The reference location should be chosen as the "sweet spot" or "hall center" location. Position your microphone first at the reference location since this is always the first location measured. When prompted, move your microphone to the next location and press any key to continue. Repeat this procedure until all locations are measured.

Note that since ADAPTIVE.SET selects the Asynchronous-correlation acquisition mode, it is not necessary to return to the time domain and reposition the time window each time to account for changes in the loudspeaker/microphone distance when the microphone is moved.

After the spatially averaged adaptively windowed room response is determined it is wise to save it to an FRQ file through the Transfer Save command for further operations such as for EQ adjustments which are discussed below in section 6.14.

To measure the bandlimited pink-noise sensitivity of the system, execute the Calculate Level command. This command computes and displays the pink-weighted power averaged of the displayed Voltage Sensitivity curve over the frequency range defined by the marker and cursor or, press the TAB key to use the Dolby standard frequency range of 500 Hz to 2 kHz instead of the marker/cursor frequency range. This command is especially useful for channel balance adjustments of surround sound systems which are much more accurate due to the Adaptive Window as compared to using a pink noise generator and sound level meter. The later method can give subjectively very poor results depending upon the particular loudspeaker-room interactions. See section 6.15 below for the procedure for adjusting surround sound channel levels.

### 6.14 Fast Room Equalization

As described in the previous section, *MLSSA* makes accurate room response measurements using the Adaptive Window feature including the spatially averaged room response if required. If spatial-averaging is required, room response measurements can be quite time consuming due to the time it takes to reposition the measuring microphone. Even when spatial averaging is not required, room measurements still require more time compared to electronic measurements since a long period MLS is required in room measurements both to avoid time aliasing and also to insure a high SNR. Rather than continually re-measure the room response in order to adjust your equalizer and gain levels it is faster, easier and more accurate to measure the uncorrected room response once without any equalizer or other processing included and save this result to an FRQ file. Then you connect *MLSSA* to measure the response of your equalizer/processor electronics only and let *MLSSA* automatically compute and display the combined equalizer/processor/room response. Since equalizers and processors are electronic devices, they can be measured repetitively much faster and more accurately than the room response even when no spatial averaging is involved.

This section documents the best way to use *MLSSA* to equalize a room by means of a parametric or fractional-octave equalizer as a post-processing operation. Before proceeding, however, you should first measure the room's uncorrected response for each channel according to the section 6.13 above without an equalizer or other processing in the signal chain. The best way to make a room response measurement is to connect *MLSSA*'s output to the power amplifier input for each channel you wish to measure since this insures all equalization, level controls and other processing are not included in your room measurements. The following procedure assumes that you have already measured the uncorrected room response for a particular channel according to section 6.13 above and saved it to an FRQ file.

Start *MLSSA* with setup file EQUALIZE.SET by typing

```
MLSSA EQUALIZE
```

or load this setup file through the Setup Load command from the time domain.

Connect the stimulus output on J3 (bottom phono connector) to the equalizer input. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Execute the Go Once command. *MLSSA* will compute and display the impulse response of the equalizer. Next, enter the frequency domain through the FFT Execute command. *MLSSA* will display the transfer function of the equalizer.

Load the previously measured uncorrected room response as the Auxiliary reference through the Reference Auxiliary Load command. Note that you must use the Auxiliary reference and not the Main reference for this procedure to work. If the FFT size or sample rate of the room FRQ file differs from the current values as determined by EQUALIZE.SET, you can use the Reference Auxiliary iMport Binary command to interpolate the reference data's frequency spacing to match EQUALIZE.SET. Note that EQUALIZE.SET sets the Auxiliary reference mode to Inverse-equalize so that the displayed response, after loading the room response file as the Auxiliary reference, is

the combined room/equalizer response. Additionally, the FFT Mode and units displayed above the graphics area will now be that of the room response Auxiliary reference data.

Execute Go Level-repeat and position the marker and cursor to select the mid-band region or, press the TAB key to use the Dolby standard frequency range of 500 Hz to 2 kHz instead of the marker/cursor frequency range. Go Level-repeat continually re-measures the equalizer transfer function, which is automatically combined with the uncorrected room response stored as the Auxiliary reference. Thus what you see is the corrected room response and overall level that you will actually achieve when the equalizer is inserted back into the sound system. On most 486 computers, Go Level-repeat completes a new measurement about once every 2 seconds and on Pentium computers, it executes about twice as fast. These measurement times are much faster than re-measuring the spatially averaged room response. In essence, the displayed curve is the room response you *would* measure according to the procedure of section 6.13 above were the equalizer actually inserted into the sound system signal chain but the rate of re-measurement is much faster since you are really only measuring the equalizer.

Adjust the equalizer controls for the desired target response and adjust the overall gain control, if present, for the desired broadband sound pressure level, which is displayed in the message window. Press Esc to terminate Go Level-repeat.

Sometimes it is useful to view the equalizer's response in isolation especially in order to initially adjust some parametric equalizers. To see the equalizer's response in isolation press Esc to stop Go Level-repeat and then execute Reference Auxiliary Equalize-mode Disable. Execute Go Level-repeat again to continually re-measure and display only the equalizer's frequency response. To go back to viewing the combined equalizer/room response press Esc to stop Go Level-repeat and then execute Reference Auxiliary Equalize-mode Inverse-equalize. Execute Go Level-repeat again to continually re-measure the combined equalizer/room response.

For documentation, execute Transfer Save and save the combined loudspeaker/room/equalizer response as a new FRQ file. Overwriting the uncorrected

room response file you loaded as the Auxiliary reference is not recommended since you may want to tweak the equalizer settings again which will require the uncorrected room response FRQ file. Answer 'yes' when Transfer Save you prompts you whether or not to save the data inverse equalized. The resulting FRQ file now holds the corrected room response, which will take effect when you insert the equalizer/processor with its current settings back into the system.

To make further room measurements you can go back to the time domain and use Setup Load to reload ADAPTIVE.SET and again follow the procedures of section 6.13 above.

## 6.15 Dolby Surround Sound Level Adjustments

An important application of *MLSSA's* Adaptive Window feature is to properly adjust the channel levels of Dolby surround sound systems for proper balance and specified acoustic SPL output.

The usual method is to use the internal test noise source generated within all Dolby Pro Logic decoders. This test signal is usually pink noise bandlimited to a range of 500 Hz to 2 kHz and is supplied to each channel in turn.

Using *MLSSA* to set up surround sound levels requires that the MLS signal replace the internal test signal provided within the decoder. This requires some tricks to fool the decoder so that it directs the MLS test signal to the desired channel. These tricks are summarized below. The decoder must be in the Dolby Pro Logic surround-sound mode for these tricks to work. Do not use the Stereo or Bypass modes if the decoder has them.

Connect the stimulus output on J3 (bottom phono connector) to the appropriate decoder input. If you own a *MLSSA 2000* card, you should measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

*Left:* To direct the MLS to the left channel speaker, connect the *MLSSA* output to the decoder's left channel input and leave the decoder's right channel input unconnected.

*Right:* To direct the MLS to the right channel speaker, connect the *MLSSA* output to the decoder's right channel input and leave the decoder's left channel input unconnected.

*Center:* To direct the MLS to the center channel speaker, connect the *MLSSA* output to *both* the left and right inputs of the decoder. Use a commonly available 'Y' adapter to make this connection. Note that this connection will result in a 3 dB increase in SPL so you need to adjust the center channel level to be 3 dB higher than specified. For example, if the required calibration level is 75 dB-SPL you need to adjust the center channel gain for a reading of 78 dB-SPL to match it to the left and right channels.

*Surrounds:* To direct the MLS to the surround speakers, connect the *MLSSA* output to the decoder's left channel input and connect an inverted version of the *MLSSA* output to the decoder's right channel input. The inverted version of the MLS must be exactly the same level as the normal *MLSSA* output only with a 180-degree phase shift. Note that because the MLS is driving both the left and right inputs there will be a 3 dB increase in SPL in the surrounds so you need to adjust the level of the surrounds to be 3 dB higher than specified.

The recommended procedure is as follows:

Start *MLSSA* with setup file ADAPTIVE.SET by typing

```
MLSSA ADAPTIVE
```

or load the setup file through the Setup Load command from the time domain. Enable pink filter correction through the Acquisition Equalization Pink-filter command.

*Left Channel:* Connect the stimulus output J1 (top phono connector) to the left channel decoder input and leave the right channel decoder input unconnected. Also insure that the decoder is in Dolby Pro Logic surround-sound mode and that any subwoofers, if

they exist, are operational. Set the decoder's master volume control to the reference position if applicable for your decoder.

Connect the measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone in the most central location within the intended listening area. The microphone used should have a flat response, be omni directional, be pointed towards the ceiling and mounted on a mike stand adjusted to typical seated ear height. To get accurate SPL readings you should have already entered the microphone sensitivity and preamp gain into *MLSSA* through the Library Microphones Define command and selected it as the currently active mike using the Library Microphones Select command.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the stimulus to a level loud enough to mask any background noise. The exact MLS level is not critical because *MLSSA* will automatically account for whatever level you choose when making the actual measurements. An MLS level that is too low can result in noisy measurements while an MLS level that is too high can overdrive the speakers resulting in too much distortion for good measurements or even damage the speakers. Press the Del key again to turn off the MLS.

Execute the Go Once command. *MLSSA* will measure and display the impulse response of the left channel loudspeaker/room combination.

Use the mouse or arrow keys to position the marker just before the first arrival, which is the first large initial spike visible in the impulse response. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Press F9 or the right mouse button to position the marker there. Press Ctrl-F9 and enter 50 to move the cursor 50 ms ahead of the marker position. You have now selected the first 50 milliseconds of the left channel loudspeaker/room response as the high-frequency window length. At low frequencies the Adaptive Window gradually expands to include more and more room reflections but can never become larger than 1/2 the FFT size. ADAPTIVE.SET selects an FFT size of 32768 points, which sets an

effective low frequency time window of about 270 milliseconds. To display both time windows execute FFT Window Display.

Next, enter the frequency domain through the FFT Execute command.

The left channel room response is displayed as a Voltage Sensitivity curve in units of dB-SPL/200mV. This curve shows you at each frequency the SPL you would measure with a 200 mV RMS sinusoid input to the decoder. The 200-millivolt reference was chosen because it is a common calibration input level used in some, but not all, Dolby surround sound systems. If a different calibration input level is called for by the system you are adjusting, enter the correct reference input by returning to the time domain and executing the FFT Mode Voltage-Sensitivity command and enter the correct input voltage reference in millivolts. After FFT Execute, the Voltage Sensitivity curve will reflect any change you made. For example, if you entered a 500 millivolt input voltage reference, the Voltage Sensitivity curve units will now be dB-SPL/500mV.

Execute the Go Level-repeat command and press the TAB key to select the Dolby standard frequency range of 500 Hz to 2 kHz. This command continually re-measures the adaptively windowed room response and re-calculates the SPL level for the current channel after each measurement. The SPL level is displayed in *MLSSA's* message window along with the frequency range in parentheses. If the room were anechoic, this level would be exactly the same level you would read using a sound level meter with 200 mV RMS of bandlimited pink noise applied to same decoder inputs to which *MLSSA* is currently connected. The only difference is that the *MLSSA* measurements are subjectively more accurate in a normal (non-anechoic) room. Adjust the decoder's left channel level control until you get the specified SPL for the system you are adjusting displayed in the message window. Press Esc to terminate Go Level-repeat.

If your system is equipped with stereo or mono subwoofers, execute Go Level-repeat again but this time move the cursor to the subwoofer crossover frequency and press F9 or, the right mouse button, to set the marker there. Next, move the cursor down to the low frequency cut-off limit of the subwoofer and press Enter to start the level measurements, which will now be based on the subwoofer output only. The displayed level may need to be boosted or cut using the appropriate (left or mono) subwoofer's

level control to match the specified level previously set for the main left channel loudspeaker. If your system is also equipped with a left channel EQ and assuming you have already adjusted it according to the procedure of section 6.16 below, you should also now adjust the left channel's EQ band nearest the subwoofer crossover frequency for a smooth transition from the left channel speaker to the subwoofer. Do not attempt to adjust the subwoofer's EQ or the other bands of the left channel EQ in this procedure. Instead, follow the procedure of section 6.16 below to do all other EQ adjustments. Press Esc to terminate Go Level-repeat.

*Right Channel:* Connect the stimulus output to the right channel decoder input and leave the left channel decoder input disconnected.

To measure the right channel it is not necessary to return to the time domain. With the microphone in the same position, simply execute Go Level-repeat pressing TAB again to select the Dolby frequency range and adjust the right channel's level to get the same reference SPL as the left channel. Press Esc to terminate Go Level-repeat.

If your system is equipped with stereo or mono subwoofers, execute Go Level-repeat again but this time move the cursor to the subwoofer crossover frequency and press F9 or, the right mouse button, to set the marker there. Next, move the cursor down to the low frequency cut-off limit of the subwoofer and press Enter to start the level measurements, which will now be based on the subwoofer output only. The displayed level may need to be boosted or cut using the appropriate (right or mono) subwoofer's level control to match the specified level previously set for the main right channel loudspeaker. If your system is also equipped with a right channel EQ and assuming you have already adjusted it according to the procedure of section 6.16 below, you should also now adjust the right channel's EQ band nearest the subwoofer crossover frequency for a smooth transition from the right channel speaker to the subwoofer. Do not attempt to adjust the subwoofer's EQ or the other bands of the right channel EQ in this procedure. Instead, follow the procedure of section 6.16 below to do all other EQ adjustments. Press Esc to terminate Go Level-repeat.

*Center Channel:* Connect the stimulus output to the right and left channel decoder inputs using a 'Y' adapter.

With the microphone in the same position, execute Go Level-repeat pressing TAB to select the Dolby frequency range. Note that you must adjust the center channel's SPL reading to be 3 dB higher than what you set for the left and right channels. This 3 dB difference is only an artifact of connecting *MLSSA's* output to *both* the left and right inputs of the decoder and not because it is an MLS signal rather than pink noise. A pink noise source connected in the same way would give the same 3 dB increase in level. If you do not adjust the center channel to read 3 dB higher than the left and right channels then the output of the center speaker will be 3 dB too low in normal operation of the decoder.

If your system is equipped with mono or stereo subwoofers and also with a center channel EQ which you have already adjusted according to the procedure of section 6.16 below, you should also now adjust the center channel EQ band nearest the subwoofer crossover frequency for a smooth transition from the center channel speaker to the subwoofer. Do not attempt to make any other EQ adjustments in this procedure. Instead, follow the procedure of section 6.16 below to do all other EQ adjustments. Press Esc to terminate Go Level-repeat.

*Surround Channel:* Connect the stimulus output to the left channel decoder input and an inverted version to the right channel decoder input.

With the microphone in the same position, execute Go Level-repeat pressing TAB to select the Dolby frequency range. Like the center channel, you must adjust the surround level to get an SPL reading that is 3 dB higher than the left and right channels. If you do not adjust the surround channel to read 3 dB higher than the left and right channels then the output of the surround speakers will be 3 dB too low in normal operation of the decoder. Press Esc to terminate Go Level-repeat.

### **6.16 Dolby Surround Sound EQ Adjustments**

An important application of *MLSSA's* Adaptive Window feature is to make proper EQ adjustments of Dolby surround-sound systems equipped with left, right and center channel equalizers as well as those additionally equipped with subwoofer equalizers.

Choose four (4) equally spaced microphone locations within the intended listening area. The first microphone position, called the reference position, should be the same one used for level adjustments, that is, located in the most central position within the intended listening area. The microphone used should have a flat response, be omnidirectional, be pointed towards the ceiling and mounted on a mike stand adjusted to typical seated ear height. Mark each of the four measurement positions (#1, #2, #3 and #4) with a piece of paper or masking tape so that the microphone can be quickly repositioned to the same locations for each channel.

This procedure assumes that the system EQ is a separate unit from the decoder. This permits the MLS test signal to be applied to the EQ inputs directly thus bypassing the decoder completely. If, however, the EQ you are adjusting is integrated with the decoder, you must refer to section 6.15 above on the tricks required to fool the decoder into directing the MLS test signal to the appropriate channel's EQ and speaker. Assuming your EQ is a separate unit; disconnect all EQ inputs from the decoder outputs for the remainder of this procedure.

**NOTE:**

If the room you are attempting to measure is unusually noisy, you can execute Acquisition Pre-Average Cycles and enter up to 16 cycles of pre-averaging to improve the signal-to-noise ratio (SNR) of your measurements by up to 12 dB.

Start *MLSSA* with setup file ADAPTIVE.SET by typing

```
MLSSA ADAPTIVE
```

or load the setup file through the Setup Load command from the time domain.

*Left Channel:* Connect the stimulus output to the left channel EQ input. Be sure all other EQ inputs are unconnected. Engage the bypass switch on the EQ.

Connect the measurement microphone and preamp to the *MLSSA* input (center phono connector). Position the measurement microphone at the reference position (#1) and adjust the stand to place the mike at normal ear height pointed towards the ceiling.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the stimulus level to a level loud enough to mask any background noise. An MLS level that is too low can result in noisy measurements while an MLS level that is too high can overdrive the speakers resulting in too much distortion for good measurements or even damage the speakers. Press the Del key again to turn off the MLS.

Execute a Go Once command. *MLSSA* will compute and display the impulse response of the left channel loudspeaker/room combination.

Use the mouse or arrow keys to position the marker just before the first arrival, which is the first large initial spike visible in the impulse response. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Press F9 or the right mouse button to position the marker there. Press Ctrl-F9 and enter 50 to move the cursor 50 ms ahead of the marker position. You have now selected the first 50 milliseconds of the left channel loudspeaker/room response as the high-frequency window length. At low frequencies the Adaptive Window gradually expands to include more and more room reflections but can never become larger than 1/2 the FFT size. ADAPTIVE.SET selects an FFT size of 32768 points, which sets an effective low frequency time window of about 270 milliseconds. To display both time windows execute FFT Window Display.

Next, enter the frequency domain through the FFT Execute command. *MLSSA* will display the frequency response for the reference microphone position. Execute Go Spatial-average and press TAB to select the Dolby standard frequency range of 500 Hz to 2 kHz. This frequency range will be used by *MLSSA* to normalize overall sound level at each microphone position to the reference position. Normalization is necessary to insure that louder mike locations do not skew the spatial averaging in their favor.

Note that the Go Spatial-average command pauses between measurements to permit re-positioning of the microphone. Press the Enter key to continue or, for remote locations, connect a momentary contact switch between J4 pin-5 and J4 pin-4 (digital ground) and press this switch to continue.

**WARNING:**

Applying more than 5 volts to J4 pin-5, could result in damage to the board. For static discharge protection, you should connect a 5.1-volt zener diode across the switch. Connect its anode to digital ground and the cathode to the TTL input side. This arrangement serves to clamp any stray voltage between -0.6 and +5.1 volts to prevent static damage.

The Go Spatial-average command also outputs a positive-going TTL pulse of 10 msec duration on J4 pin-2 after each measurement to signal an external microphone multiplexer to select the next mike. This procedure assumes, however, that you will be repositioning the microphone manually.

*MLSSA* will beep to indicate it is ready to perform the first measurement at the reference location. Press the Enter key to make this first measurement. When prompted by *MLSSA* to make the next measurement, move your microphone to the location marked #2 press Enter and wait for *MLSSA*, then to #3 and finally to location #4. *MLSSA* will display the uncorrected spatially averaged room response for the left channel speaker.

Execute Auxiliary Reference Store to store this room measurement as the Auxiliary reference. Return to the time domain, execute Setup Load, and load setup file EQUALIZE.SET. This setup file chooses a 16K point MLS and also puts the Auxiliary Reference into inverse-equalize mode which will cause *MLSSA* to add (rather than subtract) the measured EQ response to the stored room response. Disconnect the microphone from the *MLSSA* input and disconnect the EQ's left channel output from the left channel power amplifier input. Reconnect the *MLSSA* input to the EQ's left channel output. You have now configured *MLSSA* to measure the equalizer's left channel in isolation. Execute Go Once followed by FFT Execute. Execute Go Repeat to have

*MLSSA* continually re-measure your EQ. Before making your EQ adjustments, be sure to enable it by disengaging the bypass switch.

If the system contains no subwoofers, adjust the EQ for a flat response over the full audio range supported by the system. If the system does contain subwoofers, adjust the EQ for flat response only from the subwoofer crossover frequency to 20 kHz. In either case, do not attempt to adjust for a rolled-off target response in the treble. This is required for RTA measurements in an attempt to compensate for their inherent time blindness but you should aim for a flat measured response with *MLSSA* when using the Adaptive Window. The left channel is now correctly equalized except for final subwoofer EQ adjustments, which are covered below. You should now reconnect the equalizer's left channel output back to the left channel power amplifier input.

Sometimes it is useful to view the EQ response in isolation especially in order to initially adjust some parametric equalizers. To see the equalizer's response in isolation press Esc to stop Go Repeat and then execute Reference Auxiliary Equalize-mode Disable. Execute Go Repeat again to continually re-measure and display only the equalizer's frequency response. To go back to viewing the combined equalizer/room response press Esc to stop Go Repeat and then execute Reference Auxiliary Equalize-mode Inverse-equalize. Execute Go Repeat again to continually re-measure and display the combined equalizer/room response.

*Right and Center Channels:* Connect the stimulus output to the right (or center) channel equalizer input. Be sure all other equalizer inputs are unconnected. Engage the bypass switch on the equalizer.

Reconnect the measurement microphone's preamp to the *MLSSA* input. Position the measurement microphone again to the reference location (#1).

Return to the time domain, execute Setup Load, and load ADAPTIVE.SET. Execute a Go Once command. *MLSSA* will compute and display the impulse response of the uncorrected right (or center) channel loudspeaker/room combination.

Use the mouse or arrow keys to position the marker just before the first arrival, which is the first large initial spike visible in the impulse response. Press F9 or the right mouse button to position the marker there. Press Ctrl-F9 and enter 50 to move the cursor 50 ms ahead of the marker position.

Enter the frequency domain through the FFT Execute command. Execute Go Spatial-average and press TAB to select the Dolby standard frequency range of 500 Hz to 2 kHz.

*MLSSA* will beep to indicate it is ready to perform the first measurement at the reference location. Press the Enter key to make this first measurement. When prompted by *MLSSA* to make the next measurement, move your microphone to the location marked #2 press Enter and wait, then to #3 and finally to location #4. *MLSSA* will display the uncorrected spatially averaged room response for the right (or center) channel.

Execute Auxiliary Reference Store to store this room measurement as the Auxiliary reference. Return to the time domain and execute Setup Load to load setup file EQUALIZE.SET. Disconnect the microphone from the *MLSSA* input and disconnect the equalizer's right (or center) channel output from the right (or center) channel power amplifier input. Reconnect the *MLSSA* input to the equalizer's right (or center) channel output. You have now configured *MLSSA* to measure the equalizer's right (or center) channel in isolation. Execute Go Once followed by FFT Execute. Execute Go Repeat to have *MLSSA* continually re-measure your equalizer. Before making your EQ adjustments, be sure to enable it by disengaging the bypass switch. If the system does not contain subwoofers, adjust the equalizer for flat response over the full audio range supported by the system. If the system does contain subs then adjust for a flat response only from 100 Hz and up.

The right (or center) channel is now corrected and you should reconnect the equalizer's right (or center) channel output back to the right (or center) channel power amplifier input.

*Left and Right Subwoofers:* Connect the stimulus output to the left (or right) channel subwoofer equalizer input. Be sure all other equalizer inputs are unconnected. Also, disconnect the main left (or right) channel EQ outputs but leave the EQ's left (or right) subwoofer output connected. By doing this only the subwoofer will be driven by the MLS test signal allowing you to increase the MLS level for better noise masking without the risk of overdriving the main left (or right) speaker. Engage the bypass switch on the equalizer.

Reconnect the measurement microphone's preamp to the *MLSSA* input. Position the measurement microphone again to the reference location (#1).

Return to the time domain, execute Setup Load, and load ADAPTIVE.SET.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the stimulus level to a level loud enough to mask any background noise. An MLS level that is too low can result in noisy measurements while an MLS level that is too high can overdrive the speakers resulting in too much distortion for good measurements or even damage the speakers. You should be able to increase the MLS level far more for the subwoofers than for the main speakers because the subs only reproduce the bass frequencies of the MLS. This increase in level improves accuracy for the subs because room noise is usually greatest in the bass region. Press the Del key again to turn off the MLS.

Execute a Go Once command. *MLSSA* will compute and display the impulse response of the uncorrected left (or right) subwoofer/room combination.

Use the mouse or arrow keys to position the marker just before the first arrival, which is the first large initial spike visible in the impulse response. Note that the subwoofer's initial spike will not be nearly as sharply defined as the main speaker's will. This is normal and is because the subwoofer can only reproduce the bass frequencies. Press F9 or the right mouse button to position the marker there. Press Ctrl-F9 and enter 50 to move the cursor 50 ms ahead of the marker position.

Enter the frequency domain through the FFT Execute command. Execute Go Spatial-average and press TAB to select the Dolby standard frequency range of 500 Hz to 2 kHz.

*MLSSA* will beep to indicate it is ready to perform the first measurement at the reference location. Press the Enter key to make this first measurement. When prompted by *MLSSA* to make the next measurement, move your microphone to the location marked #2 press Enter and wait, then to #3 and finally to location #4. *MLSSA* will display the uncorrected spatially averaged room response for the left (or right) subwoofer.

Execute Auxiliary Reference Store to store this measurement as the Auxiliary reference. Return to the time domain and execute Setup Load to load setup file EQUALIZE.SET. Disconnect the microphone from the *MLSSA* input and disconnect the equalizer's left (or right) subwoofer output from the left (or right) subwoofer power amplifier input. Reconnect the *MLSSA* input to the equalizer's left (or right) subwoofer output. You have now configured *MLSSA* to measure the EQ's left (or right) subwoofer channel in isolation. Execute Go Once followed by FFT Execute. Execute Go Repeat to have *MLSSA* continually re-measure your EQ's subwoofer channel. Before making your EQ adjustments, be sure to enable it by disengaging the bypass switch. Adjust the subwoofer equalizer for flat response only below the subwoofer's crossover frequency (usually 100 Hz). Do not attempt to perform subwoofer level or subwoofer splice adjustments now. These adjustments are handled in section 6.15 above.

The left (or right) subwoofer is now corrected and you should reconnect the equalizer's left (or right) subwoofer output back to the left (or right) subwoofer power amplifier input. Also, reconnect the main left and right EQ outputs to the main speaker's left and right power amplifier inputs. Reconnect all EQ inputs back up to the decoder outputs.

Follow the procedure of section 6.15 above to set the correct SPL for each channel and also to adjust subwoofer levels and subwoofer splices.

## 6.17 Room Response — RTA-type

This section is included for those who want to perform an old-fashioned RTA type of room response measurement using *MLSSA*. The Adaptive Window procedure of section 6.13 above however is superior and recommended.

*MLSSA* can measure the overall 1/3 octave smoothed response of a loudspeaker/room combination in a way that emulates standard real-time analyzer (RTA) measurements using a pink noise stimulus. *MLSSA* makes this measurement by measuring a long-duration impulse response and then applying a large FFT to obtain the room's transfer function. The magnitude of the transfer function is then smoothed to reveal the 1/3 octave room response. You can also determine the spatially averaged room response by averaging several response measurements using the Go Spatial-average command.

Start *MLSSA* with setup file RTA.SET by typing

```
MLSSA RTA
```

or load the setup file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone at a typical listening position within the room.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or

the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Execute the Go Once command. *MLSSA* will compute and display the impulse response of the loudspeaker/room combination.

Use the arrow keys to position the marker just before the first arrival. Press F9 or the right mouse button to position the marker there. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Press Ctrl-F9 and enter 200 to move the cursor 200 ms ahead of the marker position. You have now selected the first 200 milliseconds of the loudspeaker/room response for further analysis.

Next, enter the frequency domain through the FFT Execute command.

The magnitude response of a typical room will generally be very ragged because of the many reflections but RTA.SET causes *MLSSA* to display the 1/3 octave smoothed magnitude, which is closer to what the ear perceives. You can change to a different smoothing fraction by executing View Magnitude sMoothed Decibels and entering, for example, 0.1 for 0.1 octave smoothing.

To determine the spatially averaged room response you take several (4 to 7) measurements at more-or-less uniformly spaced positions around the desired listening area and then average them together on a power basis. The Go Spatial-average command computes the power average of the specified number of measurements and pauses between measurements to permit re-positioning of the microphone. Press any key to continue or, for remote locations, connect a momentary contact switch between J4 pin-5 and J4 pin-4 (digital ground) and press this switch to continue.

**WARNING:**

Applying more than 5 volts to J4 pin-5, could result in damage to the board. For static discharge protection, you should connect a 5.1-volt zener diode across the switch. Connect its anode to digital ground and the cathode to the TTL input side. This arrangement serves to clamp any stray voltage between -0.6 and +5.1 volts to prevent static damage.

Note that since RTA.SET selects the Asynchronous-correlation acquisition mode, it is not necessary to return to the time domain and reposition the time window each time to account for changes in the loudspeaker/microphone distance.

After the spatially averaged room response is displayed, it is wise to save it to an FRQ file through the Transfer Save command for further operations such as equalizer adjustments which are discussed above in section 6.14.

## 6.18 Room Acoustical Parameters — Manual Methods

*MLSSA* measures reverberation by first measuring the loudspeaker/room impulse response and then reverse integrating the square of the impulse response to form a Schroeder reverberant decay curve. The best-fit slope of the decay curve is then determined by linear regression techniques to measure the  $RT_{60}$  reverberation time.

Start *MLSSA* with setup file ROOM.SET by typing

```
MLSSA ROOM /6
```

or load the setup file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone at a typical listening position within the room.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Execute the Go Once command. *MLSSA* will measure and display the impulse response of the room. Execute View Schroeder-plot Unfiltered to obtain the wideband reverberant decay curve.

A typical decay curve is initially quite nonlinear and then straightens out. The initial 10 dB of decay, called the early-decay region, does not represent the true reverberant decay. The true reverberant decay shows up as the relatively straight portion of the curve following the early-decay region.

Press the Home key to move the cursor to the start of the display. Press Ctrl-F2 and enter -0.01. The cursor will move to the start of the early-decay region. Press F9 or the right mouse button to set the marker. Note that if you intend to use the mouse, be sure to enable mouse control by first pressing the left mouse button. Press Ctrl-F2 again and enter -10.0. The cursor will move to the -10 dB location on the decay curve. Execute the Calculate T60 command to compute the early decay time (EDT). See the following section titled "Tabular Room Acoustical Parameters" for tabular reverberation time display.

To measure true reverberation time you must position the marker and cursor to select a relatively straight segment of the decay curve. The segment located between -5 and -25 dB positions is usually a good choice. Execute the Calculate T60 command to calculate the true reverberation time.

To compute  $RT_{60}$  for an octave band centered at 1 kHz, execute the View Schroeder-plot Filtered command and repeat the Calculate T60 command. You can now use the View Filter Center command to change the filter center frequency and generate a new decay curve for another octave band again followed by the Calculate T60 command to compute reverberation time.

Note that reverberation measurements are sensitive to measurement noise, which can be reduced by averaging through the Go Average command in the time domain. You can further increase the S/N of the measurement through the Acquisition Pre-average Cycles command which will cause *MLSSA* to average up to 16 MLS periods before computing the impulse response. This increases the S/N by 12 dB.

System nonlinearity, such as speaker distortion, will result in fixed-pattern noise appearing in the impulse response. Noise due to nonlinearity cannot be reduced by averaging or pre-averaging. Sometimes, however, noise due to distortion can be reduced by *lowering* the MLS signal level to prevent overdriving the loudspeaker. Nevertheless, *MLSSA* will correct for measurement noise whether due to background noise or to distortion through the View Schroeder-plot Compensation command (now enabled). *MLSSA* estimates measurement noise in the impulse response and then uses this information to correct the Schroeder plot. Noise correction has limits, however, especially at very low frequencies where room noise is typically quite high. As a rule of thumb, noise correction will be effective if the S/N ratio in a particular band is 10 dB or greater.

The reverberant decay of the room can also be displayed as a 3D energy-time-frequency plot. To do so first return to a wideband impulse response display by executing the View Data Unfiltered command. Next, place the marker at the first arrival position. Finally, execute Waterfall Execute. This 3D plot shows how the spectrum of the reverberant energy changes over time. Typically, the higher frequencies decay more quickly than the lower frequencies. You can also smooth the waterfall slices through the Waterfall Octave-smoothing command.

*MLSSA* can measure the reverb/direct ratio, early/late ratio, center time and strength from the impulse response.

Locate the notch separating the direct and reverberant sound. This usually occurs within the first 50 milliseconds of the impulse response. First, zoom in on this initial segment by appropriate positioning of the marker and cursor followed by function key F10 to zoom. Next, position the cursor in the notch, which should now be visible. Execute Calculate Reverb/direct and *MLSSA* will display the reverb/direct ratio in decibels. Execute Calculate strenGth and *MLSSA* will display the acoustical strength in decibels. Acoustical strength is defined as the sound level relative to the level of the same source at a distance of 10 meters. Execute Early/late followed by "80" followed by Enter to compute the 80-millisecond early/late ratio (C80) and the definition at 80 milliseconds (D80). Note that all of these are wideband (10 kHz) measurements.

To make octave band specific measurements you must first filter the impulse response. Execute View Data Filtered and, if necessary, reposition the cursor in the notch separating the direct and reverberant sound. Execute these various Calculate commands again to get results for the octave band centered at 1 kHz. Other octave band measurements can be made by changing the filter center frequency through the View Filter Center command.

You can also calculate the reverb/direct ratio on any displayed energy-time curve. However, results may differ from those derived from the impulse response due to the frequency-domain window functions used in the computation of energy-time curves.

## 6.19 Tabular Room Acoustical Parameters

MLSSA will compute the most important acoustical parameters in tabular form through the Calculate Acoustics command. The table below is a typical Calculate Acoustics screen.

IEC 1/1-Octave Band Acoustical Parameters									
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000	
S [dB-SPL]	60.2	64.3	68.5	73.7	78.8	78.9	79.4	SPL- weighted Averages	
N [dB-SPL]	33.4	34.1	38.3	43.1	43.6	44.8	49.6		
SNR [dB]	26.8	30.2	30.3	30.6	35.1	34.1	29.7		
C50 [dB]	-6.85	-4.15	-1.69	-0.41	4.76	5.30	8.81		3.442
C80 [dB]	-0.51	-0.21	1.01	2.23	7.04	7.80	11.94		5.846
D50 [%]	17.1	27.8	40.4	47.7	75.0	77.2	88.4		63.439
TS [ms]	114.7	108.1	104.2	88.2	43.9	36.8	15.3		62.548
EDT-10dB [s]	1.390	1.630	1.670	1.487	0.939	0.864	0.939		1.182
RT-20dB [s]	1.614	1.308	1.293	1.296	1.270	1.198	0.861		1.265
(-5:-25) r	-0.996	-0.995	-0.999	-0.999	-0.999	-0.998	-0.996		-0.999
RT-30dB [s]	1.560	1.332	1.342	1.284	1.247	1.166	0.841		1.254
(-5:-35) r	-0.997	-0.998	-0.999	-1.000	-1.000	-0.999	-0.999		-1.000
RT-USER [s]	1.705	1.258	1.322	1.297	1.237	1.178	0.895		1.253
(-10:-25) r	-0.992	-0.993	-0.999	-0.999	-0.999	-0.998	-0.995		-0.999

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SAROP center house

ESC to exit or F1 to print. MLSSA: Acoustics

### Acoustical Parameters in Tabular Form

Start MLSSA with setup file ROOM.SET by typing

```
MLSSA ROOM /6
```

or load the setup file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a MLSSA 2000 card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone at a typical listening position within the room.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Execute the Go Once command. *MLSSA* will compute and display the impulse response of the loudspeaker/room combination.

Execute the Calculate Acoustics command. You can select the range of octaves to be analyzed by entering the starting octave number (1= 32 Hz thru 5= 1000 Hz). If you press only the Enter key, however, the starting octave is the last one selected or the default given in REVERB.INI.

For each octave band *MLSSA* computes the signal level (S) in dB-SPL, the background noise level (N) in dB-SPL and the signal-to-noise ratio (SNR) in dB. The displayed levels show the MLS signal level in each particular octave and are given only for reference. These S values should not be interpreted as the room's frequency response since they are determined by integrating over whole octaves and hence show a +3 dB/octave bias due to the white-MLS spectrum.

Also shown are the clarity at 50 milliseconds (C50), the clarity at 80 milliseconds (C80), the definition at 50 milliseconds (D50), the center time (Ts) and the early decay time over the first 10 dB of decay (EDT-10dB). Finally, *MLSSA* computes three  $RT_{60}$  values (RT-20dB, RT-30dB and RT-USER) corresponding to the three decay ranges also shown in parentheses.

When the SNR is over 10 dB, *MLSSA* uses the displayed noise level to correct the three RT values. If the SNR falls below 10 dB, its entry in the table will blink and noise correction will not be applied to EDT-10dB and the RT values for that octave band only.

Below each displayed RT value is a negative number known as the correlation coefficient. The correlation coefficient shows how closely the corresponding Schroeder plot is to a perfectly straight line over the indicated decay range. A correlation coefficient of -1.0 indicates a perfect fit while smaller values indicate that the Schroeder plot may have a bend or other irregularity. A correlation coefficient smaller than -0.95 may indicate nonlinear reverberant decay, which can be verified through the View Schroeder-plot command for the indicated octave band. See section 6.18 for details.

You can also edit REVERB.INI to change from full octave analysis to 1/2 or 1/3 octave analysis. In such cases, the Calculate Acoustics command will cover a smaller total frequency range since only seven bands are ever analyzed and displayed at once. To cover a larger frequency range, you can perform multiple Calculate Acoustics commands on the same wideband impulse response each time specifying a starting band that equals 7 plus the previous starting band.

The Calculate Acoustics command can also be used to calculate two measures of lateral energy fraction, LF and LFC, based on two impulse response measurements: one taken using a figure-of-eight microphone which is stored as an overlay and, a second measurement taken using an omni directional microphone. Some microphones implement both polar characteristics (omni and figure-of-eight) as controlled by a switch. If you use two separate microphones, you need to remember to select a new microphone number in the Library Microphones screen when switching microphones to account for likely differences in either their sensitivity or their preamp's gain. Otherwise, the computed values of LF and LFC will be invalidated unless both microphones happen to have identical sensitivities and preamp gains. You also need to make sure the null in the figure-of-eight microphone is pointed toward to sound source for maximum rejection of the direct sound.

If the figure-of-eight overlay is present and plotted on the screen, Calculate Acoustics will compute and display LF and LFC in addition to the other acoustical parameters as

shown in the figure below. The other parameters are based on the omni directional measurement only. The figure-of-eight measurement must be stored as the overlay and the omni measurement as the main data. If they are reversed, LF and LFC will be computed incorrectly. You must also execute Overlay Plot before Calculate Acoustics to compute LF and LFC, otherwise the overlay data is ignored and LF and LFC are not computed.

IEC 1/1-Octave Band Acoustical Parameters								
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000
S [dB-SPL]	60.2	64.3	68.5	73.7	78.8	78.9	79.4	SPL- weighted Averages
N [dB-SPL]	33.4	34.1	38.3	43.1	43.6	44.8	49.6	
SNR [dB]	26.8	30.2	30.3	30.6	35.1	34.1	29.7	
C50 [dB]	-6.85	-4.15	-1.69	-0.41	4.76	5.30	8.81	3.442
C80 [dB]	-0.51	-0.21	1.01	2.23	7.04	7.80	11.94	5.846
D50 [%]	17.1	27.8	40.4	47.7	75.0	77.2	88.4	63.439
TS [ms]	114.7	108.1	104.2	88.2	43.9	36.8	15.3	62.548
EDT-10dB [s]	1.390	1.630	1.670	1.487	0.939	0.864	0.939	1.182
RT-20dB [s]	1.614	1.308	1.293	1.296	1.270	1.198	0.861	1.265
(-5:-25) r	-0.996	-0.995	-0.999	-0.999	-0.999	-0.998	-0.996	-0.999
RT-30dB [s]	1.560	1.332	1.342	1.284	1.247	1.166	0.841	1.254
(-5:-35) r	-0.997	-0.998	-0.999	-1.000	-1.000	-0.999	-0.999	-1.000
RT-USER [s]	1.705	1.258	1.322	1.297	1.237	1.178	0.895	1.253
(-10:-25) r	-0.992	-0.993	-0.999	-0.999	-0.999	-0.998	-0.995	-0.999
LF [%]	72.6	50.4	61.0	60.9	41.4	33.3	27.2	44.052
LFC [%]	72.6	50.4	61.0	60.9	41.4	33.3	27.2	44.052

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 SAROP center house  
 ESC to exit or F1 to print. MLSSA: Acoustics

### Acoustical Parameters with LF and LFC

LC and LFC are defined in ISO 3382 [13]. LF is the standard measure while LFC is a cosine-weighted according to Kleiner [14].

## 6.20 Spatially-Averaged Room Acoustical Parameters

To determine the spatially averaged acoustical parameters you take several (5 to 10) impulse response measurements at more-or-less uniformly spaced positions around the room and then execute the Library Acoustics to analyze them.

Start *MLSSA* with setup file ROOM.SET by typing

MLSSA ROOM /6

or load the setup file through the Setup Load command.

Connect the stimulus output on J3 (bottom phono connector) to the input of a power amplifier, which in turn, is connected to drive the loudspeaker. If you own a *MLSSA 2000* card, you can alternatively choose to measure using the pink-filtered MLS output on J1 (top phono connector). In that case, you need to enable pink-filter equalization by executing the Acquisition Equalization Pink-filter command.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Position the measurement microphone at a typical listening position within the room.

Press the Del key to enable the 65535-point MLS stimulus and press the = key or, the + key on the cursor keypad, to increase the sound level enough to mask any background noise. Do not increase the level too much to avoid overdriving and possibly damaging your loudspeaker. The stimulus level can be lowered by pressing the normal - key, or the - key on the cursor keypad. Note that each key press increases or decreases the stimulus level by 1 dB. Press the Del key again to turn off the MLS stimulus.

Execute the Go Once command. *MLSSA* will compute and display the impulse response of the loudspeaker/room combination.

Execute Transfer Autaname Path and enter the drive and directory where you wish to store your impulse response measurements to be spatially averaged later. Execute Transfer Autaname Root and enter the file root name or prefix. Save the first impulse response measurement by executing Transfer Save Right-arrow Enter. Move to the next location and repeat until all locations are measured and saved.

After you have measured and saved as many locations as desired, execute Library Acoustics Select-files followed by the Right-arrow key or the right mouse button. *MLSSA* will display a list of the autonamed data files you just created. Press Ctrl-Space or the left mouse button to mark all files followed by the Ctrl-Enter key combination or

the right mouse button. The Library Acoustics Calculate command analyzes a set of up to 14 TIM files to determine the spatially averaged acoustical parameters according to ISO 3382 [13].

The Calculate command (all commands from here on assume you are starting from the Library Acoustics menu) analyzes each selected file, in turn, until all selected files have been analyzed. After completion, the most important spatially averaged parameters are displayed in the Averages data screen shown in the figure below.

File: C:\ROOMS\TEST.ACO - Average of 4 TIM files								
Band Parameter	3 125	4 250	5 500	6 1000	7 2000	8 4000	9 8000	500- 4000
EDT-10min [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	
EDT-10mean [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	1.143
EDT-10max [s]	1.559	2.240	1.625	0.995	1.563	0.251	0.113	
RT-20 min [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	
RT-20 mean [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	1.948
RT-20 max [s]	1.417	--	1.912	2.224	2.054	1.576	1.053	
(-5:-25) warn	r N	T						
RT-30 min [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	
RT-30 mean [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	1.939
RT-30 max [s]	1.021	--	2.001	2.108	2.022	1.610	1.098	
(-5:-35) warn	r N	T						
RT-U min [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	
RT-U mean [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	1.910
RT-U max [s]	0.878	--	1.786	2.215	2.019	1.622	1.184	
(-10:-25)warn	r N	T						

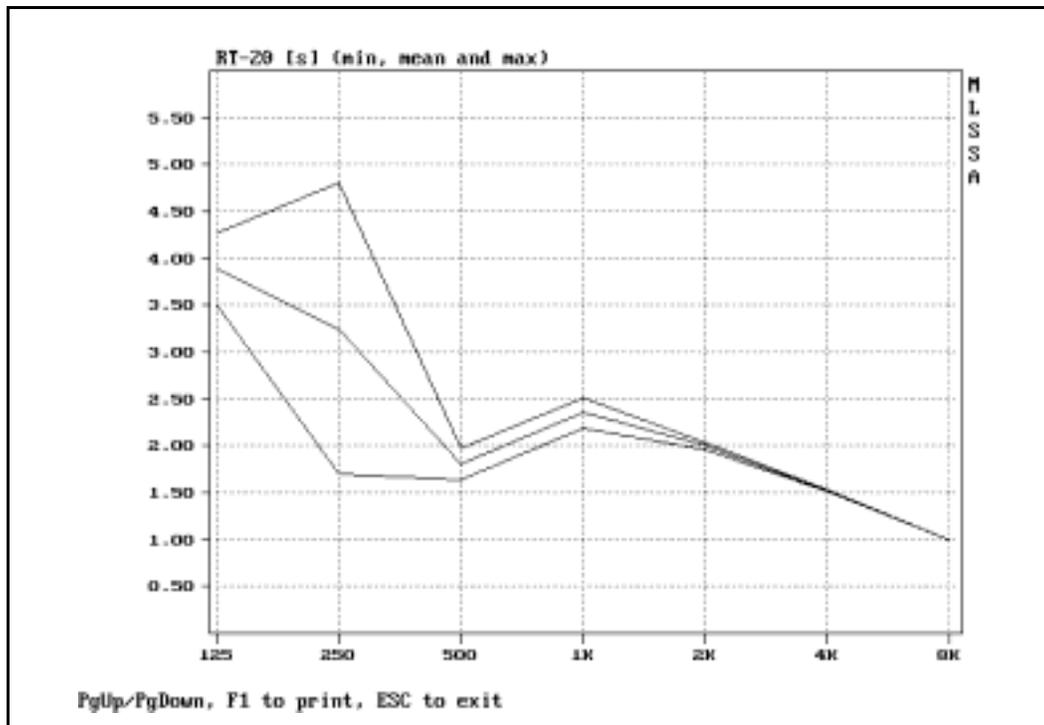
Warnings: T= IR too short; r= r < 0.95; N= SNR < 10 dB or no noise correction.

LIB ACOUSTICS: Calculate Select-files View Transfer Print Graph Options Free  
F1 for Help or ESC to exit MLSSA: Lib Acoustics

### Spatial-averages of selected acoustical parameters

The mean, minimum and maximum are displayed for parameters EDT-10dB, RT-20dB, RT-30dB and RT-USER. There are three possible warnings: 1) a "T" indicates that the impulse response of at least one of the TIM files was too short to perform a proper regression analysis over the specified decay range; 2) an "r" indicates that the absolute value of the correlation coefficient of at least one of the TIM files was smaller than 0.95 and; 3) an "N" indicates that on at least one of the TIM files it was not possible to perform the noise correction.

These spatially averaged RT parameters can also be graphed as shown in the next figure. You use the Graph command and then PgUp and PgDn to scroll through each RT parameter. You type Esc to return to the main Library Acoustics menu. The graph's vertical scale can also be changed through the Options Graph-scaling command.



**Graph of spatially averaged RT-20dB vs. frequency**

Starting from the main spatially averaged data screen, you use PgDn to scroll through the acoustical parameter tables for each individual file. Each is similar to the table generated by the Calculate Acoustics command. After all files have been displayed, further PgDn key strokes scrolls through another set of data screens, one for each individual parameter, showing its mean value and standard deviation computed across all the analyzed TIM filenames which are listed in the first column of each parameter screen. These parameter screens are useful for observing how each parameter varies

with microphone and/or source location. You can also jump to the start of any set of data screens through the View Averages, View Files and View Parameters commands or just use the PgUp and PgDn keys if you wish.

The Options Starting-band command lets you change the lowest octave or starting band to be analyzed. After changing the starting band, you must execute the Calculate command again to recompute the parameters with the new starting band. The Options Bandwidth command lets you change from full octave analysis to 1/2 or 1/3 octave analysis. In such cases, the Calculate command will cover a smaller total frequency range since only seven bands are ever analyzed and displayed at once. The Options Decay-range command lets you change the decay range for the RT-USER parameter. Again, you must recompute the parameters for any change to take effect. The Options Graph-scaling has already been explained above.

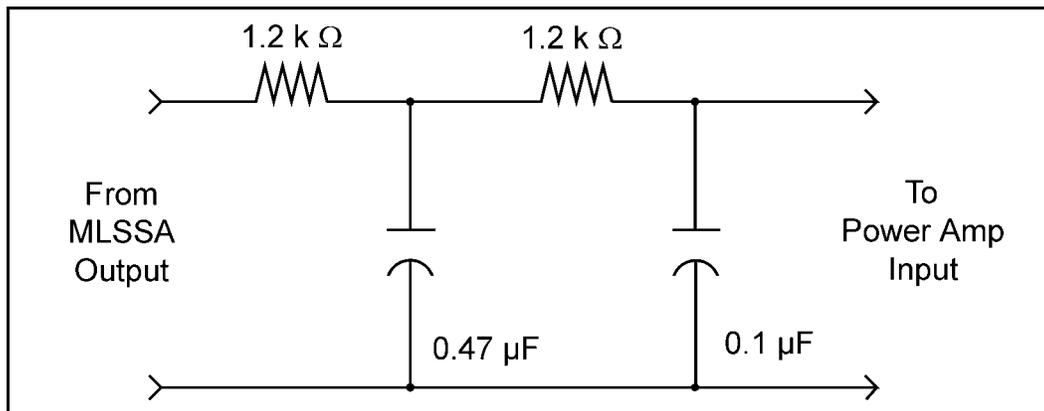
The Transfer Save command saves all the post-processed parameter data to a binary data file having the default extension ".ACO". You can later load ACO files through the Transfer Load command without the need to recompute the data from the original TIM files.

Any parameter screen or combination of screens can be exported to a text file through the Alt-4 screen capture utility and choose the option that deletes all IBM line-drawing characters and comma-delimits the output data. But first you must press Esc return to the time domain and press Alt-9 to switch *MLSSA* into Character mode followed by Library Acoustics to display these screens in 25 line text video mode. You can now use the Alt-4 command to export any screen to a plain text file. Returning to the time domain and pressing Alt-9 followed by Library Acoustics again restores graphics mode. Without first switching to Character mode, the Alt-4 command will capture these screens as graphics data and save them to PCX files.

## 6.21 Speech Intelligibility — STI and RASTI

*MLSSA* measures the speech transmission index (STI) and the rapid speech transmission index (RASTI) from a long-duration impulse response measured over a bandwidth of 3 kHz or greater. A minimum impulse response duration of 1 second is also required. Using setup file *STI.SET* mode ensures that these conditions are met. Note also that *STI.SET* starts *MLSSA* in the Asynchronous-correlation acquisition mode. This mode allows you to perform remote STI and RASTI measurements using DAT and CD recorders. See section 6.23 below titled “Remote Measurements Using DAT and CD Recorders” for details.

A simple passive speech filter is recommended for all STI and RASTI measurements. The recommended speech filter (see schematic below) assumes that the load impedance exceeds 10 kohm, which should be the case for most available power amps. It must be driven by the *MLSSA* white-MLS output (J3). Driving the speech filter from the pink-MLS output will result in potentially large measurement errors.



**Recommended STI/RASTI Speech Filter**

If you have the *RCAI*, connect the speech filter’s input to the RCA jack labeled WHITE located in the *MLSSA/AUX* area on the rear panel of the *RCAI*. Next, connect the speech filter’s output to the RCA jack labeled AUX IN. Finally, execute the Acquisition Interface Source Aux command to select the speech-filtered MLS as the stimulus.

Minor errors in the speech filter can be corrected. You correct for errors in the response of your speech filter by editing text file SPEECH.INI that consists of four columns of octave levels. The first column is the actual speech filter response integrated over whole octaves. The second column is the reference octave speech spectrum of the original STI and RASTI algorithms. The third column is the revised STI reference speech spectrum for male voices [17]. The fourth column is the revised STI reference speech spectrum for female voices [17]. *MLSSA* reads all four data sets at start-up and corrects STI and RASTI for any deviation between the actual speech filter response and the applicable reference speech spectrum. The Calculate STI Full and Calculate STI Rapid commands use the second column as the reference speech spectrum. The Calculate STI Men command uses the third column and the Calculate STI Women command the fourth column as the reference speech spectra respectively. Note that corrections for the speech filter's deviation from the applicable reference speech spectrum are only calculated and applied to "Calibrated" STI or RASTI measurements using a 65535-point impulse response. This means that if you use a shorter length impulse response and do not present the specified speech spectrum to the system under test, the STI or RASTI value will be in error to the extent that background noise is the dominant factor. To insure a 65535-point impulse response always use setup file STI.SET and be sure to start *MLSSA* with the /6 switch.

To determine the actual speech filter output spectrum, measure it by a direct loopback connection to *MLSSA* started with setup file SPEECH.SET. Measure the transfer function in the normal manner, that is, Go Once followed by FFT Execute. The filter response must be displayed on an integrated octave scale for proper results. To do this load macro SPEECH.MAC followed by Macro Execute from the frequency domain. This macro will display in the message window the integrated-octave filter response relative to the 250-Hz octave starting from the 125-Hz octave. Press Space to display each octave level value in turn. You should write down the results and manually enter them into the "Speech Filter" column in SPEECH.INI to make a permanent record of the response of your particular speech filter. Note that SPEECH.MAC will not display the speech filter octave level values if you are currently running *MLSSA* with its display in expanded mode. If this is the case, press Shift-F1 to unexpand the display prior to running SPEECH.MAC.

Note that you should not try to edit SPEECH.INI to correct gross errors in the speech spectrum such as using no speech filter at all. Nor can you use it to correct overall level errors as *MLSSA* always internally normalizes each column of data in SPEECH.INI to the same A-weighted sound level. Generally, octave level differences between the speech filter and reference speech spectrum should not exceed +/-6 dB. For larger corrections, an equalizer can be inserted into the measuring chain cascaded with the speech filter, or used by itself, and adjusted to get a reasonable speech-like spectrum, which can then be refined further by entering the actual measured spectrum into the "Speech Filter" column of SPEECH.INI.

Speech intelligibility is also affected by the directivity of the sound source. Therefore, a mouth simulator or small test loudspeaker having the same directivity characteristics of the human mouth/head system should be used for highest accuracy when assessing the intelligibility of unamplified talkers. A mouth simulator is not required, however, when speech will be delivered to the audience through a sound system. In such cases, measurements should be made through the sound system.

When using a mouth simulator, an acoustic measurement of its output spectrum needs to be taken at a close distance (0.5 to 1.0 meters) directly in front of a mouth simulator or small test loudspeaker. In addition, all room reflections need to be windowed out of the measurement. See section 6.1 titled Loudspeaker Anechoic Response for more details. After measurement of the anechoic acoustic frequency response taken directly in front of the mouth simulator, one that also includes the speech filter and/or equalizer driven by the white-MLS stimulus, execute macro SPEECH.MAC to compute the octave-band levels of the measured speech spectrum. Enter the results into the "Speech Filter" column of SPEECH.INI. Note that if the results differ from the "Reference" column by more than +/- 6 dB, you should adjust the equalizer as required and repeat the measurement.

In addition to providing a speech-like spectrum, the speech filter also serves to smooth out the high slew rate binary white-MLS to form a gentler noise-like signal having a nearly Gaussian (bell shaped) amplitude distribution. Such a Gaussian amplitude distribution is necessary to properly account for any non-linear distortion present in the measurement chain.

A speech filter is especially important for the Calculate STI Women and Calculate STI Men commands because these apply additional corrections that account for the absolute threshold hearing, as well as applying a variable auditory masking factor, both of which depend upon absolute octave-band SPL levels that mimic those found in actual speech. Without a speech filter, the 4K and 8K octave levels will typically be much higher than encountered in normal speech resulting in reduced accuracy.

Furthermore, because they require knowledge of absolute SPL levels in each octave band at the listening (microphone) position, the Calculate STI Women and Calculate STI Men commands require accurate microphone sensitivity and preamp gain data, which you will need to enter into the Library Microphones screen. Don't forget to select the microphone you are using prior to performing measurements. If no microphone was selected prior to performing measurements, the Calculate STI Men and Calculate STI Women commands will refuse to do the calculations. In cases where you have old TIM files without microphone calibration data (or containing inaccurate calibration data), use the Library Microphones Modify-file command to enter, as best as can be determined from memory or from notes, your best guess as to the microphone sensitivity and microphone preamp gain existing at the time when the original measurement was performed.

After setting up the mouth simulator where live talkers will be expected to talk, as described above, you are ready to perform accurate STI measurements.

Start *MLSSA* with setup file STI.SET by typing

```
MLSSA STI /6
```

or load the setup file through the Setup Load command. Note the command line switch /6 allows you to measure a full 65535-point impulse response, which is required for "Calibrated" STI or RASTI measurements. The word "Uncalibrated" is displayed if you use the wrong setup or analyze an impulse response less than 65535 points or 1.8 seconds in length. Furthermore, speech filter response corrections are disabled if you fail to issue the /6 switch, unless *MLSSA* is instructed to automatically start with the 64K max acquisition length as set by line 10 of in *MLSSA.INI*.

Connect the white-MLS stimulus output J3 (bottom phono connector) to the input of the passive speech filter. Connect the output of the speech filter to the power amplifier input, which in turn, should be connected to drive the mouth simulator or test loudspeaker. If the *RCAI* is connected, execute Acquisition Interface Source Aux and check that the speech filter's input is connected to the connector labeled WHITE and its output is connected to the connector labeled AUX IN, both connectors located in the MLSSA/AUX area on the rear panel of the *RCAI*. To drive a sound system's line level input, connect it to the SRC OUT connector located on the rear panel of the *RCAI*.

Press the Del key to enable the 65535-point white-MLS stimulus and repeatedly press the = key or, the + key on the cursor keypad, to increase the stimulus to the desired level. When measuring the STI or RASTI for unamplified talkers using a mouth simulator or small test loudspeaker, you measure the A-weighted sound level of the speech-filtered MLS signal at a location 1 meter directly in front of the mouth simulator or test loudspeaker and adjust to achieve an A-weighted sound level of 60 dB-SPL (according to IEC 60268-16 Third Edition [17]). The use of a sound level meter is recommended for this adjustment so as not to require re-configuring of *MLSSA* to measure A-weighted levels or require temporarily moving the microphone connected to *MLSSA* to this reference location. The measurement microphone is normally positioned far from the mouth simulator, representing a typical listening position.

If you are measuring the STI or RASTI of amplified talkers, apply the speech-filtered MLS through the same PA sound system and adjust the A-weighted level to be the same as for a typical amplified talker measured at a typical close-in position. This level need not be 60 dB, but rather the A-weighted level of a typical amplified talker when speaking through the same PA system. Note that all such level adjustments are critical if the background noise is expected to be high enough to significantly influence intelligibility. After adjusting the A-weighted level using a calibrated sound level meter; press the Del key again to turn off the white-MLS stimulus.

Connect a measurement microphone and preamp to the *MLSSA* input J2 (center phono connector). Using the Library Microphones Select command, select this microphone/preamp combination from the list of your microphones. Position the measurement microphone at a typical listening position within the room. If the *RCAI* is

connected, execute Acquisition Interface Mic-mux and select the mic-mux input (A,B, C or D) currently connected to your microphone preamp output. You may also need to execute the Library Microphones Assign command to assign the correct microphone and preamp calibration data to the currently selected mic-mux input.

Execute the Go Once command. *MLSSA* will measure and display the impulse response of the room or system. Execute the Calculate STI Full command to compute the original STI value or, the Calculate STI Men command to compute the revised STI value for male voices or, the Calculate STI Women command to compute the revised STI value for female voices. Press Esc to return to the time domain. Execute the Calculate STI Rapid command to compute the RASTI value. Note that all STI measurements are more accurate than any RASTI measurement yet require only slightly more time to calculate. Note also that you can abort any STI or RASTI calculation in progress by hitting the Esc key.

If the room's reverberation time is greater than about 7 seconds, you should also reduce the measurement bandwidth to avoid time-aliasing errors. If the reverberation time is greater than 7 seconds in any octave band, execute the Acquisition Bandwidth and select a 6 kHz bandwidth and a Chebyshev filter. You will now be able to tolerate reverberation times up to 14 seconds without error. In that case, however, the STI commands will not analyze the 8-kHz octave band, resulting in some loss in accuracy.

STI and RASTI are special derivative calculations based on the complex modulation transfer function (CMTF). You can also compute the system's CMTF directly for any given octave or fractional octave band using *MLSSA*'s built-in time-domain bandpass filter. The FFT mode is already set to CMTF as programmed by STI.SET. For example, after measuring the 65535-point impulse response, to look at the CMTF for the 125-Hz octave simply (from the time domain) execute View Data Filtered to display the 125-Hz octave-filtered impulse response. Next, execute FFT Execute to compute the CMTF. To look at a different octave CMTF, return to the time domain and execute View Filter Center to change the octave center frequency to say 500 Hz again followed by FFT Execute. Note that there may be some differences in the CMTF curves as compared to the MTF values displayed by the STI screens. Differences can occur because the STI function uses true brick wall (infinite roll-off) analysis filters while

*MLSSA*'s time-domain bandpass filter has a -18 dB/octave roll-off rate. In addition, the "m-correction" or "m/thresh-cor" correction factors, respectively, are applied to the MTF values displayed on the STI screens but these correction factors are not computed or applied to CMTF curves.

## 6.22 Sound Level Measurements and NC curves

Often in sound system work, it is necessary to measure absolute background noise or live sound levels. Although a good sound level meter can make these measurements, *MLSSA* can be configured to make them also. To measure sound levels in the frequency domain you must be in the Power Spectrum FFT mode and the Scope acquisition mode. Sound levels are calculated from the frequency domain through the Calculate SPL commands. These commands allow you to determine sound levels in units of dB-SPL in IEC standard octave bands as well as determining A-, B-, and C-weighted levels. *MLSSA* also computes and displays the noise criterion (NC) rating of background noise.

Start *MLSSA* with setup file NOISE.SET by typing

```
MLSSA NOISE /6
```

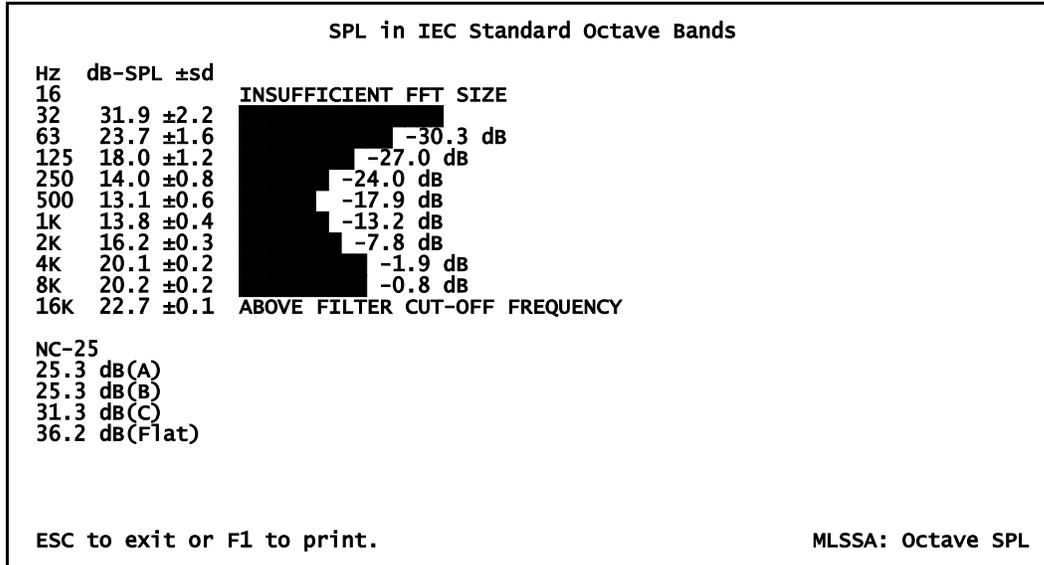
or load the setup file through the Setup Load command.

This setup file puts *MLSSA* in Scope acquisition mode and sets the Power Spectrum FFT mode. The acquisition length is 65536 samples.

Connect a calibrated measurement microphone to the *MLSSA* input connector J2 (center phono connector). Make sure you select the microphone you are going to use and that its correct sensitivity and preamp gain are entered into the Library Microphones screen. The accuracy of your SPL measurements will only be as good as your microphone. For highest accuracy, use a microphone calibrator in conjunction with the Library Microphones Calibrate command to calibrate your microphone and preamp.

Execute the Go Once command. After autoranging, *MLSSA* will acquire 65536 points of sound data. Execute the FFT Execute command and *MLSSA* will compute and display the power spectrum of the noise.

Execute the Calculate SPL Inclusive command to compute and display all the SPL measurements.



**Calculate SPL Inclusive command data screen**

You will notice in the sample screen above, that for the 16-Hz octave band *MLSSA* displays the message "INSUFFICIENT FFT SIZE". This means that the current frequency resolution is too coarse to compute the SPL accurately in that band. You can increase the FFT size to measure this band but this will require more processing time and increase the amount of random error. Notice also that the 16 kHz band displays the message "ABOVE FILTER CUT-OFF FREQUENCY". This message means that the antialiasing filter cut-off is within or below the 16 kHz octave band. You can eliminate this message by increasing the instrument bandwidth to 25 kHz but then frequency resolution will be made coarser. The default values for bandwidth and FFT

size in NOISE.SET are considered optimum for most applications but you can always change them to suit your needs.

The expected standard deviation for each octave SPL value is also displayed immediately after the dB-SPL value. The expected standard deviation is computed based upon amount of data used to make the measurement. Thus, increasing the Acquisition Length and/or using the Go Average command from the frequency domain both decrease the expected error in the measured SPL by acquiring and processing more data. Note, however, that time domain averaging is not valid for SPL measurements and *MLSSA* will refuse to compute the Power Spectrum of acquired data that was first averaged in the time domain. When executed from the frequency domain, the Go Average command can acquire and process data taken over very long time periods, if desired, and will spectrally average that data on a power basis. This command can be stopped prematurely at any time by pressing Esc but *MLSSA* will still display the results based on all of the data acquired up to that time.

The noise criterion (NC) rating is also calculated and displayed. The NC rating is defined as the lowest NC curve that is not exceeded by the actual octave-band noise spectrum. In the example above, the NC rating is NC-25. The difference in decibels between the rated NC curve and the actual octave-band noise level is displayed to the right of each octave's bar graph. Note that, by definition, these differences must always be negative.

Press Esc to return to the frequency domain menu. The power spectrum curve is also displayed in units of dB-SPL. This means dB-SPL per FFT-bin bandwidth. Since the FFT-bin bandwidth will change when either the sample rate or FFT size are changed, you should use care in interpreting this curve. For example, if you double the FFT size the FFT bin bandwidth will be halved and the dB-SPL in a particular bin will decrease by 3 dB. The power spectrum curve should only be used to determine the *relative* distribution of sound or noise energy over frequency.

Instead, for 1/2, 1/3 or 1/6-octave sound level measurements use the View Magnitude Integrated command. This command is similar to the Calculate SPL Octaves command except that the result is displayed as a normal *MLSSA* frequency graph and can

therefore compute and display results for bandwidths narrower than one octave. This command *integrates* (sums) energy over each octave, 1/2-octave, 1/3-octave or 1/6-octave band and the result is displayed as a stair-step style graph. Because this command integrates the power spectrum over bands of fixed width, the result, unlike the power spectrum itself, is insensitive to changes in FFT size and/or sampling rate. Note that before using the View Magnitude Integrated command you must first select the Microphone reference through the View Magnitude Reference Microphone command. Otherwise, the resulting curve will not be scaled properly in units of dB-SPL.

### **6.23 Remote Measurements Using DAT and CD Recorders**

In large acoustical spaces, it is sometimes inconvenient to make measurements with *MLSSA* due to the long cable lengths required. It is possible, however, to perform measurements using a DAT (digital audio tape) or CD recorder acting as a signal source and/or to capture the system response for later analysis by *MLSSA*. The Acquisition Mode command offers the Asynchronous-correlation mode just for this purpose. In this mode, it is no longer necessary for the MLS stimulus signal to trigger the data acquisition of the response, as is case for normal *MLSSA* measurements.

Small portable battery-operated DAT recorders are widely available and are recommended for this application. CD recorders can also be used. Note that Minidisk digital recorders use a lossy compression algorithm and are therefore incompatible with MLS measurements. Analog tape recorders cannot be used due to insufficient time base stability caused by tape stretch as well as by wow and flutter.

You can use a DAT recorder in two ways. One way is to prerecord the MLS stimulus signal for later playback as a remote stimulus source. The other application is to use a DAT recorder to capture the system response for later analysis by *MLSSA*. With two DAT recorders, it is even possible to make complete measurements without moving *MLSSA* to the site.

DAT and CD recorders use an internal crystal-controlled clock as a time base. The typical tolerance on these internal clocks is 100 ppm or 0.01%. Therefore, you must always ensure that the same recorder is used for both the record and play functions

and that it is fully warmed up prior to recording or playback. Otherwise, small time-base errors (> 1 ppm) between the recording and playback recorders and/or drift in the time base clock due to temperature changes will result in excess noise in the measurements. For example, you cannot prerecord a CD with MLS data and play it back on a standard CD player. The internal clock of the CD recorder and that of the CD player are not likely to match within 1 ppm given that each clock has a tolerance of 100 ppm.

The same requirement applies to the *MLSSA* card, which also employs a crystal-controlled clock. The same *MLSSA* card must be used to acquire the measurement data as was used to prerecord the MLS stimulus. Additionally, the PC containing the *MLSSA* card as well as the DAT recorders need to be fully warmed up prior to making any recordings or measurements to allow the internal clocks to stabilize. This is especially important if the DAT recorder is exposed to extreme cold or heat in transit to the site. In such cases, the DAT recorder needs time to adjust to normal room temperature before measurements can be performed.

Start *MLSSA* with setup file DATROOM.SET by typing

```
MLSSA DATROOM /6
```

or load the setup file through the Setup Load command.

This setup file puts *MLSSA* into the Asynchronous-correlation acquisition mode and sets the trigger delay to -1500 samples. The negative trigger delay moves the first arrival location to the right to simulate a typical initial time-of-flight gap. Otherwise, this setup file is the same as ROOM.SET.

If you intend to make remote STI or RASTI measurements, you should substitute STI.SET for DATROOM.SET since STI.SET also puts *MLSSA* into the Asynchronous-correlation acquisition mode and additionally properly configures *MLSSA* to make "Calibrated" STI and RASTI measurements.

To prerecord the MLS stimulus press the Del key to turn on the stimulus and press the = key or, the + key on the cursor keypad, to bring up the output to a reasonable level usually about 0.5 volts. Connect the line level input of your DAT recorder to *MLSSA*'s white-MLS or pink-MLS output and record at least 30 minutes of gap-free MLS data. Note that if you intend to perform only STI or RASTI measurements, you should insert a speech-weighting filter between the white-MLS output and the DAT recorder's line level input. On site, after allowing sufficient warm up time, you simply start playing back the prerecorded MLS into the room or sound system prior to initiating a measurement through the Go Once command. Allow at least one full MLS period to elapse before initiating a measurement to ensure that the room response has stabilized. *MLSSA* will compute the room's impulse response as usual except that the initial time gap is simulated (reflecting the negative trigger delay) and is not related to the actual time-of-flight delay between the sound source and microphone.

Instead of carrying *MLSSA* to the remote site, you can carry a second DAT or CD recorder and use it to record the system response. Record at least 10 or 15 seconds of data and note the elapsed time indicator on the recorder for each measurement so that the tape can be rewound to the correct starting point for playback. Later, you simply play back the system response into *MLSSA* and execute Go Once to make the actual measurement. Again, be sure to use the same particular DAT or CD recorder to play back the system response that was used to record it on site.

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15. Leo L. Beranek, "Concert Hall Acoustics" 1992", *Journal of the Acoustical Society of America*, vol. 92, No. 1, pp 1-39, July 1992.
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## Appendix A

### MLSSA Hardware Configuration Information

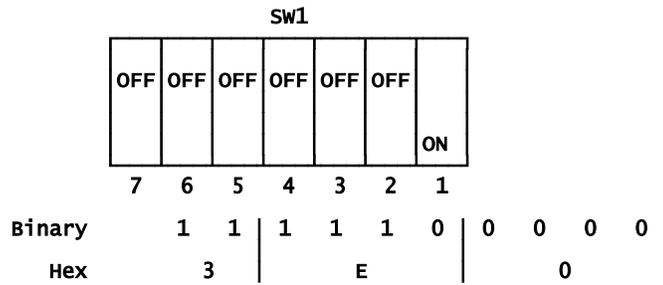
The *MLSSA* 2000 standard hardware consists of an A2D-160 2000 mothercard and two AFM-50 antialiasing filter modules. The two filter modules are cascaded to form a single 8-pole filter.

#### A2D-160 2000 Jumpers and Switches

Interrupt level 3 and DMA channel 1 are normally selected as shown in the illustration below. This group of jumper pads is located near the A2D-160 2000 expansion bus edge connector. To change to interrupt 5 remove the jumper at IRQ3 and reinstall at the IRQ5 position. Note that IRQ2 should never be selected. To change DMA channels you must change two jumpers. For example, to change to DMA channel 2\* you must move the jumper at DRQ1 to DRQ2 and the jumper at DACK1 to DACK2.

JP10	JP11	JP12
IRQ5 +                      + 	DRQ1 +                      + 	DACK3 +                      + 
IRQ3 +                      + 	DRQ2 +                      + 	DACK2 +                      + 
IRQ2 +                      + 	DRQ3 +                      + 	DACK1 +                      + 

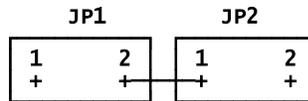
Switch positions 1 through 6 of SW1 are normally set to select an I/O base address of 3E0 hexadecimal. Switch position 7 is turned off to select two's complement A/D coding. Do not change switch position 7. To change the I/O base address change switch positions 1 through 6 as required.



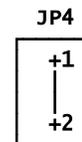
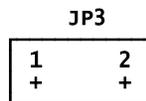
## Jumper & Switches

The remaining jumpers explained below are given for reference purposes only. You should not attempt to change any of the jumpers shown below.

The AFM-50 antialiasing filters are cascaded by jumpering pin 2 of JP1 to pin 1 of JP2. These jumpers are located near empty socket U1 on the A2D-160 2000 card.



JP3 must not have any jumper installed. Jumper JP4 must have a jumper installed.



JP6 must have a jumper installed. JP7 must not have any jumper installed. These jumper pads are located below jumper pads JP1 and JP2.



## AFM-50 Antialiasing Filter Modules

The A2D-160 2000 must have two AFM-50 antialiasing filter modules installed at locations designated AFM1 and AFM2. When installing the modules make sure that the four corner pin numbers (1, 15, 16 and 30) correspond between the AFM-50 and A2D-160 2000 and that the AFM-50 components are facing upwards.

Each AFM-50 should have jumper pads JP3 and JP4 jumpered as shown below.



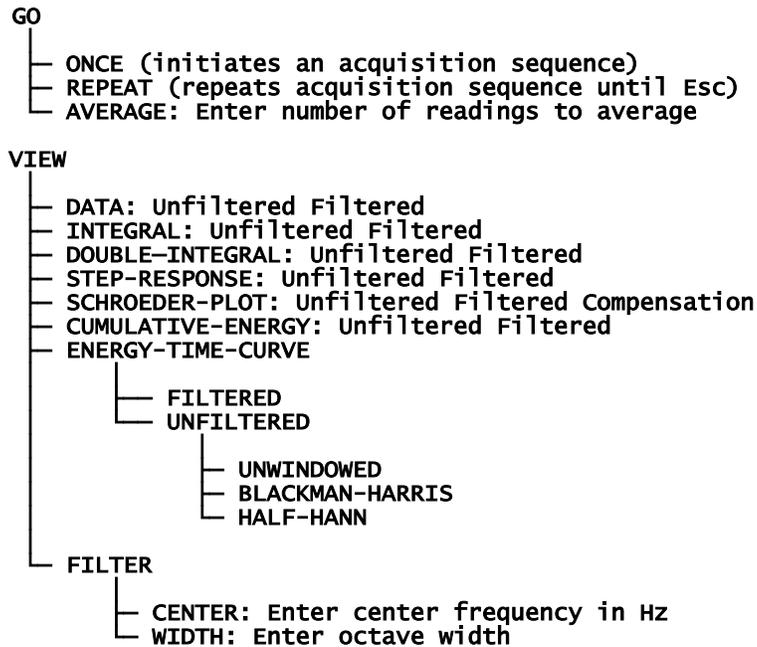
The AFM-50 module installed at AFM1 (located at the extreme end of the A2D-160) should have its JP1 jumpered and its JP2 unjumpered. The other AFM-50 should have its JP2 jumpered and its JP1 unjumpered.



## Appendix B

## MLSSA Menu Trees

## Time-Domain Menus



```
FFT
├── EXECUTE (executes FFT and enters frequency domain)
├── BYPASS (enters frequency domain w/o executing FFT)
├── SIZE: Enter FFT size in samples
├── WINDOW
│   ├── DISPLAY
│   ├── ADAPTIVE: Room Speaker
│   ├── RECTANGULAR
│   ├── HAMMING: Full Half
│   ├── COSINE: Full Half
│   ├── KAISER-BESSEL: Full Half
│   └── BLACKMAN-HARRIS: Full Half
└── MODE
    ├── NORMAL
    ├── TRANSFER-FUNCTION
    ├── POWER-SPECTRUM
    ├── IMPEDANCE: Enter series resistance in ohms
    ├── SENSITIVITY: Enter load impedance & distance
    ├── VOLTAGE-SENSITIVITY: Enter ref input voltage
    ├── CMTF
    ├── PSD
    ├── COHERENCE
    ├── INCOHERENCY
    └── RESPONSE-FUNCTION
```

```
WATERFALL
├── EXECUTE (waterfall plots)
├── MODE: CSD wigner ETF Frequency-file-display
├── FFT-SIZE: Enter FFT size in samples
├── CYCLES: Enter number of slices to plot
├── SHIFT: Enter number of points or files to shift
├── WINDOW
│   ├── SELECT: Rect. Cosine Hamming Blackman-Harris
│   └── RISETIME: Enter window risetime in msec
├── OCTAVE-SMOOTHING: Enter 0.1 to 1.0 octave
├── REFERENCE (enter reference slice)
├── HIDDEN-LINES: Visible Invisible
├── PERSPECTIVE: Normal Reverse
├── TEXT-EXPORT-FILE: (enter filename)
├── X-AXIS
│   ├── RANGE: Enter low and high frequency limits
│   └── SCALE: Log liNear IEC-log
├── Y-AXIS
│   ├── DB/DIV: Enter dB/div for non-wigner only
│   ├── FLOOR: Auto Set
│   └── WIGNER-SCALE: Auto Set
└── Z-AXIS
    ├── UNITS: Enter z-axis units for FFD mode
    ├── INCREMENT: Enter z-axis increment for FFD
    ├── OFFSET: Enter z-axis offset for FFD
    └── FIRST-FILE: Enter first file # for FFD
```

## ACQUISITION

- MODE: Scope Cross-correlation Asynchronous-corr.
- AUTORANGE: Normal Tight Loose Disable
- BANDWIDTH: 1 2 3 5 6 10 12 15 20 \*24 25 30 \*32 35 40 kHz
- LENGTH: Enter length in samples
- UNITS: Enter acquisition units label and factor
- DELAY: Enter delay in seconds
- RETRY: Enable Disable
- EQUALIZATION: Pink-filter Disable
- PRE-AVERAGE
  - CYCLES: Enter 1-16 blocks to pre-average
  - MODE: Concurrent Alternate
- CLOCK
  - INTERNAL: Enter sample rate in kHz
  - EXTERNAL: Enter sample rate in kHz
- STIMULUS
  - SELECT: Pulse Step MLS
  - AMPLITUDE: Enter amplitude in volts
  - MODE: Burst Continuous
  - UNITS: Enter units label and factor
  - PROGRAM
    - PULSE: Enter period in samples
    - STEP: Enter high/low times
    - MLS: Enter MLS order: 12 14 15 16
- TRIGGER
  - SELECT: Auto Stimulus Plus Minus
  - DELAY: Enter +/- trigger delay in samples
- FILTER
  - SELECT: Butterworth Thomson Chebyshev
  - BANDWIDTH: Enter bandwidth in kHz
  - GAIN: Enter filter gain in 1 dB steps
- INTERFACE (For controlling the RCAI)
  - MIC-MUX: A B C D
  - SOURCE: Pink White Aux Mute
  - DRIVE: Power-amp Direct
  - RESET
  - OPTIONS: Mic-mux-mode Spatial-averaging Z/loop-source

## SETUP (displays current setup info)

- |— LOAD: Enter setup filename to load
- |— RELOAD (reloads current setup file)
- |— SAVE: Enter setup filename to save
- |— COMMENT: Enter Display
- |— PRINT: Prints setup screen
- |— DUMP: Dumps setup screen using TSR printer driver

## TRANSFER

- |— SAVE: Enter binary filename to save.
- |— LOAD: Enter binary filename to load.
- |— IMPORT: Enter text filename to load.
- |— COMMENT: Enter Display
- |— EXPORT
  - |— DATA: Enter text filename
  - |— VIEW: Enter text filename
- |— AUTONAME
  - |— PATH: Enter path to files
  - |— ROOT: Enter root name of files
  - |— NUMBER: Enter filename starting number
  - |— LOGFILE: Create List

## MACRO

- EXECUTE (executes current macro)
- DEBUG: Enable Disable
- LOAD: Enter filename to load
- SAVE: Enter filename to save
- INFO: (displays info about macro)
- FREE (frees macro buffer storage)
- VARIABLE: ctrl-V ctrl-W ctrl-X ctrl-Y ctrl-Z

## COMMAND

- PAUSE (pauses during macro execution)
- DELAY (delays during macro execution)
- BEEP (beeps during macro execution)
- MESSAGE: Enter message to display
- HEADLINE: Enter headline to display
- RUN: Enter program name to run during macro
- CALL: Run program w/o screen changing
- SIGNAL (output a pulse on J4 pin-2)
- WAIT (wait for input signal on J4 pin-5)
- LOAD-SETUP (load setup file w/o screen)
- IF-PASSED-QC: Signal Execute
- IF-FAILED-QC: Signal Execute Alarm
- EXEC: Run big program during macro
- MARK (mark macro location for LOOP return)
- LOOP (enter loop repetitions)

## OVERLAY

- LOAD: Enter time-domain filename
- STORE (stores current data in overlay buffer)
- PLOT (plots overlay from overlay buffer)
- FREE (frees overlay storage)
- SAVE: Enter time-domain filename
- INFO (displays overlay info)
- VERTICAL-SCALE: Normal Auto Fixed
- ERASE (erases overlay from screen)

## CALCULATE

- STI: Full Rapid Men Women
- T60 (reverberation time)
- REVERB/DIRECT (reverberant to direct sound ratio)
- EARLY/LATE (early/late ratio)
- CENTER-TIME (centroid of impulse response)
- STRENGTH
- SPL
- ACOUSTICS: (Enter starting octave number)

## PRINTER

- PRINT
  - SCREEN (prints current graphics display)
  - TEXT (prints all text on screen)
  - MESSAGE (prints message window contents)
  - HEADER (prints header file if exists)
  - FOOTER (prints footer file if exists)
  - LINEFEED (advances the printer one line)
  - FORMFEED (ejects paper from printer)
- FILE: Open Close Print
- LINK: Header Footer
- UNLINK: Header Footer Both
- COPIES: Enter number of copies to print
- EJECT-PAGE (Ejects current page)
- DRIVER
  - DUMP: (prints screen using TSR driver)
  - COLORS: Normal Adjust Black-and-white
  - OPTIONS:
  - MODE: P25 Capture-to-PCX-file
- SETUP
  - TYPE: IBM LaserJet Epson Okidata  
HP-PCL
  - MARGINS: Enter print margins in inches
  - DIRECTION: Landscape Portrait
  - FORMFEED: Enable Disable
  - PORT: LPT1 LPT2 LPT3 COMM1 COMM2 PRN AUX

DOS

- CHANGE-DIRECTORY: (Changes default directory)
- SHELL (executes COMMAND.COM )
- UNLINK (deletes files)
- MLSSADATA-SHELL (executes COMMAND.COM w/ data)

UNITS: Enter time units and factor

LIBRARY

- BEEPER: Enable Mute
- COLORS: Main Waterfall Aux Text
- LOAD-PCX (load and display a PCX graphics file)
- GRAPHICS-MODE: CGA EGA VGA
- MICROPHONES: Select Define Remove Calibrate  
                  Modify-file Print dUmp
- OPERATIONS
  - TIME-FILES: Average Sum Difference Product  
                  Convolution X-correlation
  - FREQUENCY-FILES: Average Sum Difference  
                  Product Ratio IFT Equalize  
                  Conjugate sMooth reciprOcal  
                  spLice QC-statistics
- ACOUSTICS: Calculate Select-files View Transfer  
                  Print Graph Options Free
- PARAMETERS (access Speaker Parameter Option)

INFO (display information of current time-domain data)

QUIT (exits MLSSA to DOS)

## Frequency-Domain Menus

- GO
- ONCE (initiates an acquisition/FFT sequence)
  - REPEAT (repeats acquisition/FFT until Esc)
  - AVERAGE: Enter number of FFTs to average
  - SPATIAL-AVERAGE: Enter number of FFTs to average
  - LEVEL-REPEAT: (Go Repeat & Calculate Level)
  - IMPEDANCE: (Measures impedance using RCAI)
- VIEW
- REAL (real part of FFT data)
  - IMAGINARY (imaginary part of FFT data)
  - GROUP-DELAY: Smoothed Unsmoothed Excess
  - NYQUIST-PLOT (real vs imaginary plot)
  - BODE-PLOT: Actual-phase Minimum-phase Excess-phase
  - MAGNITUDE
    - LINEAR (magnitude on a linear scale)
    - SQUARED (squared magnitude)
    - DECIBELS (magnitude on decibel scale)
    - SMOOTHED
      - DECIBELS: Enter 0.1 to 1.0 octave
      - LINEAR: Enter 0.1 to 1.0 octave
    - PERCENT
    - INTEGRATED: Octaves Half-octaves  
Third-octaves Sixth-octaves
    - REFERENCE: Normal Microphone
  - PHASE
    - WRAPPED (normal phase curve)
    - UNWRAPPED (unwrapped phase curve)
    - MINIMUM (minimum phase curve)
    - EXCESS (actual - minimum phase curve)
    - DELAY: Enter phase delay correction in msec
    - ACCURACY: Normal High Super
    - UNITS: Radians Degrees

## ACQUISITION

- MODE: Scope Cross-correlation Asynchronous-corr.
- AUTORANGE: Normal Tight Loose Disable
- BANDWIDTH: 1 2 3 5 6 10 12 15 20 \*24 25 30 \*32 35 40 kHz
- LENGTH: Enter length in samples
- UNITS: Enter acquisition units label and factor
- DELAY: Enter delay in seconds
- RETRY: Enable Disable
- EQUALIZATION: Pink-filter Disable
- PRE-AVERAGE
  - CYCLES: Enter 1-16 blocks to pre-average
  - MODE: Concurrent Alternate
- CLOCK
  - INTERNAL: Enter sample rate in kHz
  - EXTERNAL: Enter sample rate in kHz
- STIMULUS
  - SELECT: Pulse Step MLS
  - AMPLITUDE: Enter amplitude in volts
  - MODE: Burst Continuous
  - UNITS: Enter units label and factor
  - PROGRAM
    - PULSE: Enter period in samples
    - STEP: Enter high/low times
    - MLS: Enter MLS order: 12 14 15 16
- TRIGGER
  - SELECT: Auto Stimulus Plus Minus
  - DELAY: Enter +/- trigger delay in samples
- FILTER
  - SELECT: Butterworth Thomson Chebyshev
  - BANDWIDTH: Enter bandwidth in kHz
  - GAIN: Enter filter gain in 1 dB steps
- INTERFACE (For controlling the RCAI)
  - MIC-MUX: A B C D
  - SOURCE: Pink White Aux Mute
  - DRIVE: Power-amp Direct
  - RESET:
  - OPTIONS: Mic-mux-mode spatial-averaging Z/loop-source

REFERENCE

- GO
- LOAD: Enter frequency-domain filename
- STORE (stores current FFT in reference buffer)
- IMPORT: Text Binary
- FREE (frees reference storage)
- EQUALIZE-MODE: Equalize Inverse-equalize Disable
- REFERENCE: Enter 0 dB reference for equalize
- INFO (displays reference info)
- THEORETICAL (stores theoretical AA filter)

AUXILIARY-REFERENCE

- LOAD
- STORE
- IMPORT: Text Binary
- FREE
- EQUALIZE-MODE: Equalize Inverse-eq Disable
- INFO:
- THEORETICAL

SETUP (displays current setup info)

- SAVE: Enter setup filename to save
- COMMENT: Enter Display
- PRINT: Prints setup screen
- DUMP: Dumps setup screen using TSR printer driver

TRANSFER

- LOAD: Enter binary filename to load
- SAVE: Enter binary filename to save
- IMPORT: Text Binary
- COMMENT: Enter Display
- EXPORT
  - VIEW: Enter filename for view data
  - NYQUIST: Enter filename complex data
  - BODE: Actual Minimum Excess-phase

AUTONAME

- PATH: Enter path to files
- ROOT: Enter root name of files
- NUMBER: Enter filename starting number
- LOGFILE: Create List

MACRO

- EXECUTE (executes current macro)
- DEBUG: Enable Disable
- LOAD: Enter filename to load
- SAVE: Enter filename to save
- INFO: (displays info about macro)
- FREE (frees macro buffer storage)
- VARIABLE: ctrl-V ctrl-W ctrl-X ctrl-Y ctrl-Z

COMMAND

- PAUSE (pauses during macro execution)
- DELAY (delays during macro execution)
- BEEP (beeps during macro execution)
- MESSAGE: Enter message to display
- HEADLINE: Enter headline to display
- RUN: Enter program name to run during macro
- CALL: Run program w/o screen changing
- SIGNAL (output a pulse on J4 pin-2)
- WAIT (wait for input signal on J4 pin-5)
- LOAD-SETUP (load setup file w/o screen)
- IF-PASSED-QC: Signal Execute
- IF-FAILED-QC: Signal Execute Alarm
- EXEC: Run big program during macro
- MARK (mark macro location for LOOP return)
- LOOP (enter loop repetitions)

#### QC-LIMITS

- GO: Once Average Repeat  
Spatial-average Level-repeat
- COMPARE: (Plot and compare QC limits)
- OFFSET: Floating Set
- LOAD: (load QC-limits from special text file)
- FREE (frees QC-limits storage)
- INFO (displays QC-limits info)

#### OVERLAY

- LOAD: Enter frequency-domain filename
- STORE (stores current data in overlay buffer)
- IMPORT: Text Binary
- PLOT (plots overlay from overlay buffer)
- COMPARE (compares current data to overlay data)
- FREE (frees overlay storage)
- OFFSET: (set dB offset for overlay plot)
- SAVE: Enter frequency-domain filename
- INFO (displays overlay info)
- VERTICAL-SCALE: Normal Auto Fixed
- ERASE (erases overlay from screen)

#### CALCULATE

- DISTORTION (calculates odd, even and THD)
- EXPORT-DISTORTION-FILE: Open Close (text file)
- SPL: A B C Flat Octaves Inclusive
- ACOUSTIC-CENTER (from excess phase curve)
- CAPACITANCE (from impedance)
- INDUCTANCE (from impedance)
- LEVEL: (dB curves only)
- WEIGHTED-LEVEL: (dB curves only)

```

PRINTER
├── PRINT
│   ├── SCREEN (prints current graphics display)
│   ├── TEXT (prints all text on screen)
│   ├── MESSAGE (prints message window contents)
│   ├── HEADER (prints header file if exists)
│   ├── FOOTER (prints footer file if exists)
│   ├── LINEFEED (advances the printer one line)
│   └── FORMFEED (ejects paper from printer)
├── FILE: Open Close Print
├── LINK: Header Footer
├── UNLINK: Header Footer Both
├── COPIES: Enter number of copies to print
├── EJECT-PAGE (Ejects current page)
├── DRIVER
│   ├── DUMP: (prints screen using TSR driver)
│   ├── COLORS: Normal Adjust Black-and-white
│   ├── OPTIONS:
│   └── MODE: PZ5 Capture-to-PCX-file
└── SETUP
    ├── TYPE: IBM LaserJet Epson Okidata
    │           HP-PCL
    ├── MARGINS: Enter print margins in inches
    ├── DIRECTION: Landscape Portrait
    ├── FORMFEED: Enable Disable
    └── PORT: LPT1 LPT2 LPT3 COMM1 COMM2 PRN AUX
    
```

DOS

- CHANGE-DIRECTORY: Changes default directory
- SHELL (executes COMMAND.COM )
- UNLINK (deletes files)
- MLSSADATA-SHELL (executes COMMAND.COM w/ data)

UNITS: Enter frequency units label and factor

LIBRARY

- BEEPER: Enable Mute
- COLORS: Main Waterfall Aux Text
- LOAD-PCX (load and display a PCX graphics file)
- GRAPHICS-MODE: CGA EGA VGA
- MICROPHONES: Select Define Remove Calibrate  
                  Modify-file Print dUmp
- OPERATIONS
  - TIME-FILES: Average Sum Difference Product  
                  Convolution X-correlation
  - FREQUENCY-FILES: Average Sum Difference  
                  Product Ratio IFT Equalize  
                  Conjugate sMooth reciprOcal  
                  spLice QC-statistics
- ACOUSTICS: Calculate Select-files View Transfer  
                  Print Graph Options Free
- PARAMETERS (access Speaker Parameter Option)

INFO (display information on frequency-domain data)

EXIT (returns to time-domain)

## Appendix C

### MLSSA File Structures

This appendix is included for those who wish to write post-processing programs and want to work with *MLSSA* binary data files. Binary files are more compact and load faster than text files. All file structure definitions are given in C language form and are defined in the header file *mlssa.h*. Note that the structure for setup files is given for completeness but is subject to change in future versions of *MLSSA*.

#### Time Domain Data Files

```

long identification_tag = 0xffffabcd;
int acquisition_algorithm; /* 0=Scope,1=Correlation,2=Async-corr. */
float delta_time; /* msec between data points */
long data_length; /* number of data points */
float data[data_length]; /* time data */
char title[80]; /* title string */
char comment[60]; /* user comment if any */
struct setup; /* file header (see below) */

```

#### Frequency Domain Data Files

```

long identification_tag = 0xffffbcde;
float delta_frequency; /* kHz between data points */
long data_length; /* number of complex points */
float fftdata[2*data_length]; /* complex FFT data */
char comment[60]; /* user comment if any */
struct setup; /* file header (see below) */

```

## Setup Files

```
long identification_tag = 0xffffdcba;
struct setup
{
    long fftsize; /* FFT size used for frequency domain */
    int window_type;
        /* 0=rect, 1=full-Hamming, 2=half-Hamming, 3=full-Hann,
           4=half-Hann, 5=full-Blackman-Harris, 6=half-Blackman-Harris */
    float sample_rate; /* sampling rate in kHz */
    float filter_band; /* Antialiasing filter cut-off in kHz */
    float filter_gain; /* Antialiasing filter gain */
    int db_filter_gain; /* Antialiasing filter gain in dB */
    long trigger_delay; /* Trigger delay in samples */
    int trigger_type; /* 0=Stimulus, 1=Auto,
                       2=Extern-minus, 3=Extern-plus */
    int stimulus_type; /* 0=Pulse, 1=Step, 2=MLS */
    long stimulus_period; /* period of pulse stimulus in samples */
    int stimulus_order; /* MLS order 12, 14, 15 or 16 */
    float stimulus_amp; /* Amplitude of stimulus in volts */
    int stimulus_on; /* 0=Burst stimulus, 1=Continuous stimulus */
    long acquire_size; /* Number of samples acquired/displayed */
    int acquire_algorithm; /* 0=Scope mode, 1=Cross-correlation */
    int filter_type; /* 0=Butterworth, 1=Bessel, 2=Chebyshev */
    int printer_type;
    int beeper_on;
    int dc_couple;
    int autorange_on; /* 0=autorange off, 1=autorange on */
    float units_factor; /* Acquisition units scale factor, units/volt */
    char units_label[11]; /* Acquisition units */
    float db_reference; /* Zero dB reference in frequency domain */
    float stim_units_factor; /* Stimulus units scale factor */
    char stim_units_label[11]; /* Stimulus units */
    int ratio_mode; /* 0=Normal, 1=Transfer-Function, 2=Impedance,
                    3=PSD, 4=Response-function, 5=Power Spectrum,
                    6=Coherence, 7=Incoherency */
    int phase_units; /* 0=degrees, 1=radians */
    int equalize_on; /* 0=No equalization, 1=equalize transform */
    long stimulus_low; /* Low period for step stimulus in samples */
    long stimulus_high; /* High period for step stimulus in samples */
};
```

```

int print_dir;
int print_port;
float print_hmargin;
float print_vmargin;
int print_ff;
int external_clock; /* 0=internal clock, 1=external clock */
int acquire_cycles; /* Number of pre-average cycles */
char tunits_label[11]; /* Time axis units */
float tunits_factor; /* Time axis units scale factor in units/ms */
char funits_label[11]; /* Frequency axis units */
float funits_factor; /* Frequency axis scale factor in units/kHz */
char impedance_units_label[11]; /* Impedance units */
float series_resistance; /* Series resistance for Z measurements */
float equalize_level; /* 0 dB reference for equalization */
int sincx_correction;
float fcenter; /* Center frequency of digital bandpass filter */
struct colortype cga_colors;
struct colortype ega_colors;
struct graph t;
struct graph f;
int print_size;
int wf_cycles;
int wf_step;
int wf_window;
int wf_logon;
long wf_fftsize;
float wf_lowf;
float wf_highf;
float wf_risetime;
float wf_falltime;
float wf_dbperdiv;
float foctave; /* octave bandwidth of digital bandpass filter */
char date_stamp[20]; /* Date stamp of acquisition */
float dyna_range; /* Dynamic range of acquisition */
float acquire_delay; /* Delay after GO command */
char source_name[35]; /* precursor data filename */
char reference_name[35]; /* reference filename */
char filename[35]; /* filename of this data if saved */
int wf_mode;
int wf_noautofloor;
float wf_floor;
int schroeder_correction;
float sensitivity_impedance; /* load impedance, sensitivity mode */
struct mictype mic; /* microphone calibration data */
int spl_mode; /* 1= display dB-SPL when mic data exists */
float wf_octave; /* energy-time-frequency octave smoothing */
int preaverage_mode; /* 1= concurrent, 0= alternate */
long t_delay /* zero time offset in samples */
float wf_energy_per_div /* wigner vertical scale factor */
float wf_z_axis_offset; /* z-axis offset */
float wf_z_axis_delta; /* z-axis scale factor */
char wf_z_axis_label[11]; /* z-axis units label */
int min_phase_accuracy; /* minimum phase accuracy level*/

```

```

float phase_delay; /* phase delay for delay corrected files*/
long go_average_readings; /* # of averages */
float sensitivity_distance;
float smooth_octave;
int findex;
float sensitivity_voltage;
int equalize_invert[2];
int equalize_enable[2];
int no_acquire_retry;
int acquire_eq_filter;
char setup_file_comment[60];
float interface_attenuation;
int interface_unit_number;
int interface_drive;
int time_overlay_style;
float time_overlay_lowamp;
float time_overlay_highamp;
int freq_overlay_style;
float freq_overlay_lowamp;
float freq_overlay_highamp;
};

```

For reference, some other structures included in the setup structure are reproduced below.

```

struct colortype
{
int graph_end;
int graph;
int cursor;
int highlight;
int plot;
int overlay;
int axis;
int qc_plot;
int logo;
int title;
int ticks;
int label;
int menu;
int level;
int overplot;
int domain;
int message;
};

```

```

struct graph
{
long maxpoints; /* default display length */
long slength; /* length of data */
long cindex; /* cursor index */
long mindex; /* marker index */
long highi; /* right display edge index */
long lowi; /* left display edge index */
float highamp; /* top of display level */
float lowamp; /* bottom of display level */
long sensnum; /* sensitivity level (don't use) */
int crosson; /* 1=crosshair on, 0= crosshair off */
int position; /* 1=cursor readout on, 0=cursor readout off */
float deltax; /* point to point X axis spacing */
int style; /* style bit definitions
            bit 0  plot points = 0, lines = 1
            bit 1  move cursor with marker = 1
            bit 2  grid always = 1
            bit 3  overplot on = 1
            bit 4  old autoscale when overplot on
            bit 5  screen blanked = 1
            bit 6  blank LF region below color bar
            bit 7  autoscale on overlay */
int logmode; /* 0=linear X axis, 1=log X axis */
char *hunits; /* horizontal axis units label */
char *title; /* displayed title string */
int autoscale; /* 0=autoscale off, 1=autoscale on */
float hunits_factor; /* horizontal units scale factor */
};

struct mictype
{
int number;
char mfr[13];
char model[9];
char serial[9];
char caldate[10];
float sensitivity;
float preamp_gain;
float db_reference;
int degree;
float coefs[8];
float c1;
float c2;
};

```

## ACO Files

```
long identification_tag = 0xffff8eda;
int nfiles; /* number of FRQ files used */
int starting_band; /* 0=16Hz, 1=32Hz, 2=63Hz etc. */
float low_reverb[3]; /* low end of decay range in dB for RT1, RT2,
RT3 */
float high_reverb[3]; /* high end of decay range in dB for RT1, RT2,
RT3 */
char filepath[80]; /* DOS path to FRQ files */
char filenames[nfiles][13]; /* list of FRQ filenames */
char filecomments[nfiles][60]; /* array of FRQ file comments /
acoustic parameters[nfiles][7]; /* array of structure acoustic, see
below */
char comment[60]; /* comment for ACO file */
int iso_increment; /* analysis bandwidth = iso_increment/6,
example 2/6= 1/3oct.*/
int nbands; /* number of bands processed /
int reverb_low_band; /* starting band for average */
int reverb_high_band; /* ending band for average */

typedef acoustic
{
float S;
float N;
float SNR;
float C50;
float C80;
float D50;
float Ts; /* center time */
float EDT;
float RT1; /* reverberation time, first range */
float r1; /* correlation coeff. For RT1 */
float RT2;
float r2;
float RT3;
float r3;
float LF;
float LFC;
} acoustic;
```

## Appendix D

### Programming Notes

This appendix is included for those who wish to write post-processing programs for use with the Macro Command Run or Macro Command Call commands. These commands create an MS-DOS environment variable called MLSSADATA which points to a structure which locates all the important *MLSSA* data either in memory or stored in temporary files. This structure is defined below.

```

struct mlssatype
{
float huge *adbuffer; /* main time data */
float huge *view_buffer; /* processed/displayed time data */
float huge *overdata; /* main overlay time data */
float huge *view_overdata; /* processed/displayed overlay data */
struct setup *st; /* setup structure for main time data */
struct setup *overst; /* setup structure for overlay data */
struct graph *t; /* graph structure for time domain */
long acquire_size; /* number of time-domain points */
float sample_interval; /* sample interval in milliseconds */
float huge *fftdata; /* main FFT data (complex) */
float huge *fdata; /* processed/displayed frequency data (real) */
float huge *offtdata; /* overlay FFT data (complex) */
float huge *ofdata; /* processed/displayed overlay data (real) */
float huge *refdata; /* reference FFT data (complex) */
struct setup *sf; /* setup structure for fftdata[] */
struct setup *oversf; /* setup structure for offtdata[] */
struct setup *refsf; /* setup structure for refdata[] */
}

```

```

struct graph *f; /* graph structure for frequency domain */
long nval; /* number of complex points in fftdata[] */
long ref_nval; /* number of complex points in refdata[] */
float deltaf; /* frequency spacing in kHz */
int tindex; /* time-domain display type index */
int findex; /* frequency-domain display type index */
float *parameters; /* pointer to array of 16 speaker parameters */
int *temp_handles; /* pointer to array of temporary file handles
  [0] = frequency domain overlay data (complex)
  [1] = saved time data when in frequency domain
  [2] = time domain overlay data (float)
  [3] = adbuffer time data (float)
  [4] = Main reference data (complex)
  [5] = Auxiliary reference data (complex)
  [6] = saved FFT data for FFT > 4096
      or Power Spectrum FFT mode (complex)
  [7] = time post-processed overlay when plotted
  [8] = general purpose , data is undefined */
char *message_save; /* last message displayed */
};

```

The indexes "tindex" and "findex" refer to the contents of the arrays "view\_buffer" and "fdata" respectively. The time domain display index "tindex" has the following meaning.

- 0 Data (IR or raw)
- 1 Filtered Data
- 2 Energy-Time Curve - dB (Blackman-Harris window)
- 3 Integral of Data
- 4 Integral of Filtered Data
- 5 Cumulative Energy - dB
- 6 Filtered Cumulative Energy - dB
- 7 Schroeder Plot - dB
- 8 Filtered Schroeder Plot - dB
- 9 Energy-Time Curve - dB (Unwindowed)
- 10 Energy-Time Curve - dB (half-Hann window)
- 11 Filtered Energy-Time Curve - dB
- 12 Step-response (unfiltered)
- 13 Filtered Step-response
- 14 Double-integral
- 15 Filtered double-integral

The frequency domain display index "findex" has the following meaning.

- 0 Magnitude - dB
- 1 Wrapped Phase
- 2 Group Delay - msec
- 3 Real part
- 4 Imaginary part
- 5 Unwrapped Phase
- 6 Linear Magnitude
- 7 Squared Magnitude
- 8 Smoothed Magnitude - dB
- 9 Minimum Phase
- 10 Excess Phase
- 11 Nyquist Plot (complex)
- 12 Bode Plot Actual-phase (complex)
- 13 Magnitude - Percent
- 14 Smoothed Magnitude (linear scale)
- 15 Integrated octaves
- 16 Integrated 1/2-octaves
- 17 Integrated 1/3-octaves
- 18 Integrated 1/6-octaves
- 19 Smoothed Group-delay
- 20 Bode-plot Minimum-phase (complex)
- 21 Bode-plot Excess-phase (complex)
- 22 Excess Group delay

For best compatibility, your program should be written in Microsoft C. The following C program, TESTDATA.C, shows how to gain access to the structure and hence to the data. Note that the file MLSSA.H is included to define this and other *MLSSA* data structures.

```
#include "mlssa.h"
#include <stdlib.h>
#include <stdio.h>
main(argc, argv)
int argc;
char *argv[];
{
int i;
char *string;
struct mlssatype *pd;
if ((string = getenv("MLSSADATA")) == NULL) exit(0);
if ((pd = (struct mlssatype *)atoi(string)) == NULL) exit(0);
for (i = 0; i < argc; i++)
printf("%s\n", argv[i]);
printf("\ncindex = %li, mindex = %li", pd->t->cindex, pd->t->mindex);
printf("\n\nPress any key to resume");
getch();
}
```

The call to "getenv" returns a string corresponding to the address of the structure. If the return string is NULL the program exits because it cannot gain access to the structure. The call to "atol" converts the environment string into a long integer, which is actually a far pointer to the structure. If the pointer is NULL, the program exits.

The program prints out any command line arguments that may have been passed and then prints out the cursor and marker locations for the time-domain graphics display. Note the two-level structure references needed for this task. To avoid long structure expressions you can optionally copy the members of this structure into separate variables defined in your program.

There are some important points to keep in mind when using this structure. Always check pointers to be sure they are not NULL before using them. Overlay buffers, for instance, are only allocated when an overlay is present. If there is no overlay, the overlay pointer variable will be a NULL pointer. In addition, when in the time domain, the frequency domain buffers "fftdata" and "fdata" will be NULL pointers.

The "fdata" pointer always points to the currently post-processed frequency domain data. For example, if the magnitude in dB is displayed then "fdata" points to this data at all times when in the frequency domain. Similarly, the "view\_buffer" pointer always points to the currently displayed time data.

Another way to access the measurement data is to read it from the temporary files. The DOS handles of these open files are contained in the array named temp\_handles[]. These temporary files stay open all of the time *MLSSA* is running. Therefore, you should not try to reopen them in your program. However, you should rewind a file so that its pointer points to the beginning of that file before trying to read its contents.

The Macro Command Exec command like Macro Command Run also spawns another program but all of *MLSSA* including its data is swapped out of conventional memory to permit larger child programs. This command does not create the MLSSADATA structure but you can still read the temporary files because both Macro Command Exec and DOS Shell create an environment variable named MLSSAHANDLES, which contains a list of the temporary file handles.

There are a few other important rules to adhere to when reading the temporary files. File handle index [6] contains the raw FFT data when in the frequency domain only when the FFT size exceeds 4096 and the FFT mode is Normal, Transfer Function, Impedance or Sensitivity. In those cases, that is, when FFT size is equal to or less than 4096 or the FFT mode is Power Spectrum, PSD, Coherence or Incoherency then the raw FFT data is in data array "fftdata" and is not present in any temporary file.

Do not attempt to write to any of the locations specified by the MLSSADATA structure. The purpose of this structure is to allow you to *read* the data only. Attempting to write into any *MLSSA* data structure could cause improper operation of *MLSSA*.



## Appendix E

### *MLSSA 2000* Card External Interface Connections

This section describes the analog and digital interface connectors located on the back panel 'L' bracket. Three RCA phono connectors and one 9-pin 'D' connector protrude through the back panel for interfacing to external devices and are labeled J1, J2, J3 and J4.

#### **Analog Connections**

RCA phono connectors J1, J2 and J3 are used for all analog connections.

J3, the bottom RCA jack, is the main analog output supplying a white-MLS from both *MLSSA 2000* cards as well as all earlier versions of the *MLSSA* card. Output impedance is 75 ohms. Maximum output voltage is 5.25 volts and maximum output current is 100 ma from *MLSSA 2000* cards and 10 ma from all older *MLSSA* cards.

J2, the center RCA jack, is the analog input having an input impedance of around 13 k ohms. Maximum input is 25 volts peak and damage to the AFM-50 antialiasing filters can occur with inputs exceeding 35 volts peak.

J1, the top RCA jack, is an analog output supplying a quasi pink-filtered version of the analog output on J3, usually an MLS, from *MLSSA 2000* cards. Output impedance is 75 ohms. Maximum output voltage is 5.25 volts and maximum output current is 100 ma.

Note that on pre-*MLSSA 2000* cards J1 does not supply any useful analog output and must be left unconnected at all times. Otherwise, incorrect operation or large measurement errors will result.

For reference, the following table summarizes the analog connections.

<i>Connector</i>	<i>Function</i>
J1 (top)	Pink-filtered version of Main Analog Output on J3 (75 ohms)
J2 (middle)	Analog input (13 k ohms)
J3 (bottom)	Main Analog output (75 ohms)

Maximum safe input rating for J2 is +/-35 peak volts. Both outputs are protected against short circuits and against thermal overload but care should be taken to prevent the outputs from being accidentally connected to a power amplifier, which can result in damage to the output circuits.

## Digital Connections

J4 is a 9-pin 'D' connector, which is used for all digital connections. The following table summarizes these signals. Note that all are TTL compatible and all inputs are pulled up to +5 volts using 4.7 k ohm resistors. You should never attempt to apply any analog signal to any of these pins.

For static discharge protection, all used inputs and outputs should be shunted by a 5.1-volt zener diode to digital ground (J4 pin 4). Connect the zener's anode to digital ground and its cathode to the TTL input or output line as required, preferably as close to J4 as possible such as right at the connector that plugs into J4. This arrangement serves to clamp any stray voltage between -0.6 and +5.1 volts to prevent damage caused by static discharge.

<i>J4 pin number</i>	<i>Digital Signal</i>
pin 1	+12 volts DC
pin 2	Macro Command Signal 10 ms pulse (output)
pin 3	Sample rate clock (input or output)
pin 4	Digital ground
pin 5	Macro Command Wait sense line (input)
pin 6	External acquisition trigger (input)
pin 7	MLS signal generator (output)

pin 8	External time base clock (input)
pin 9	Internal trigger not delayed (output)

The following is a detailed description of each digital signal available through J4.

1. This pin is +12 volts DC from the PC power supply.
2. Macro Command Signal pulse. A positive-going TTL pulse of 10 ms duration is output on this line when executing the Macro Command Signal during macro execution.
3. Sample rate clock. This TTL level signal acts as either an input or an output as controlled by software. Normally, pin 3 supplies the internally generated sample rate clock to an external device. The internal clock generator can be disabled through the Acquisition Clock External command, which places this pin into a high impedance state allowing an external sample rate clock signal to be applied. The external sample rate clock source must supply a negative-going TTL pulse having a pulse width between 250 nanoseconds and 1 microsecond. Longer or shorter pulse widths will result in a malfunction of the data acquisition hardware. The leading or negative-going edge of this signal is the active edge.
4. Digital ground. Digital ground reference for all external digital connections. This pin should be connected to ground on the device being interfaced to J4. Note that this pin should not be connected to analog ground of any external analog device. Analog ground is the shield connections of the RCA phono jacks J1, J2 and J3.
5. Macro Command Wait sense line. This Schmitt-triggered TTL input allows an external positive going pulse to be sensed by the Macro Command Wait command during macro execution. You can connect a push-button switch between this pin and J4 pin-4 (digital ground). Then, you momentarily press this switch to enable the macro to continue. For static discharge protection, you should also connect a 5.1-volt zener diode across this switch. Connect the zener's anode to digital ground and its cathode to the TTL input side.

6. External trigger. This Schmitt-triggered TTL input allows an external event to trigger a data acquisition sequence. This feature is selected by the Acquisition Trigger Select Plus (for an external positive-going edge trigger) or Acquisition Trigger Select Minus (for an external negative-going edge trigger) commands.
7. MLS generator digital output. This TTL level output is the digital output of either the on-board MLS generator or the Pulse or Step stimuli if selected by the Acquisition Stimulus Select command.
8. External time base clock input. Not supported by the *MLSSA* software,
9. Internal trigger output. This TTL output is the internally generated acquisition trigger. When the MLS generator is enabled, it is a positive-going pulse having a pulse width equal to the period of the sample rate clock.