SIA-Smaart®

AcousticTools™

Acoustic Measurement and Analysis Software

User Guide

Version 4 for Windows®

Introduction

SIA-Smaart® Acoustic Tools™ provides a powerful, cost-effective set of tools for measuring and analyzing room acoustics and evaluating sound system intelligibility. The package consists of two program modules; the Intelligibility module and the Analysis module. The Intelligibility module allows you to measure the impulse response of a device or room/system and calculate both quanitative acoustical values for room analysis and objective intelligibility ratings for electro-acoustic systems. The Analysis module is the a file-based signal analysis program that provides both time and frequency based displays of wave file data. Both programs work with standard Windows audio waveform files (also called wave or *.wav files) and are designed primarily for use with impulse response data.

The SIA-Smaart Intelligibility Module

The Intelligibility module can calculate several objective speech intelligibility indices and quantitative acoustical values from impulse response data store in Windows wave files. The Percentage of Articulation Loss of Consonants (%ALCons), Clarity (early-to-late energy ratio) for some number of milliseconds (e.g., C50, C35, or C7), and both Speech Transmission Index and Rapid Speech Transmission Index (STI and RSTI), can all be calculated from the impulse response. The Intelligibility module can also be used to find delay times and reflections and calculate Early Decay Time (EDT), Direct to Reverberant level, and RT60 from the same impulse response data.

The program includes its own impulse response recorder that can calculate and record the impulse response of the system under test using either a Maximum Length Sequence (MLS) or stimulus-independent dual Fast Fourier Transform (FFT) transfer function technique. The MLS method requires the use of an internally generated MLS stimulus signal. The Dual-FFT method can use pink noise or virtually any other continuous signal (e.g., music) as a stimulus — this is the same technique used in SIA SmaartLive™ to find delay times. The Intelligibility module provides an internal pink noise generator for use in dual-FFT measurements however externally generated signals may be used as well. The Intelligibility module can also be used to analyze impulse data from any other source as long as the data is stored in, or can be converted to standard Windows wave files.

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The SIA-Smaart Analysis Module

The Analysis module reads audio data stored in any standard Windows wave (*.wav) file including impulse response files recorded in the Intelligibility module. The program can open one or two wave files at a time and can transform time-domain data in the wave file(s) into its frequency domain representation, using a series of Fast Fourier Transform (FFT) calculations. Both time and frequency characteristics of recorded signals can then be displayed, compared and analyzed a number of ways.

Like the Intelligibility module, the Analysis module is primarily designed for use in analyzing impulse response data and includes tools to help you find reflections, reverberation and decay information. Other possible uses for this program include speech analysis or any application where you need to analyze the time and frequency characteristics of recorded signals.

What's New In this Release

SIA-Smaart Acoustic Tools 4 features a completely redesigned user interface for both the Intelligibility and Analysis modules. The new interface design is intended to dramatically increase ease of use by reducing the complexity of on-screen control surfaces, providing easier access to the most frequently used program features, and making data readouts larger and easier to read. Several redundancies and inconsistencies in the two program modules have also been addressed.

In the Analysis module, the "Analyze" function has been removed since the All Bands Table in the Intelligibility module provides very similar functionality. The Intelligibility module also uses more conventional methods of finding and calculating quantitative acoustical values and produces more consistent and reliable results.

In the Intelligibility module, a stock set of Clarity Factor calculations for 10, 20, 50 and 80 milliseconds have been added to the All Bands Table to compensate for the removal of the Analyze function from the Analysis module. The ExSTI calculation has been removed since the program also calculates true STI and RSTI values and ExSTI is not a widely used or accepted measure of intelligibility. Additionally, %ALCons calculations are now limited to the octave band centered on 2 kHz because this is the only band in which this measurement is conventionally performed and for which the result is considered valid.

Contents

Chapter 1: Getting Started

System Hardware

Recommended Configuration

- Microsoft® Windows® 95, 98 or NT 4.0 (or higher)

- 166 MHz or faster Intel® Pentium® or compatible microprocessor

- 32 - 64 Mb system RAM

- 800 x 600 x 256-color Super VGA display

- Windows-compatible sound hardware with stereo line level input(s), 16-24 bit resolution, selectable sampling rates from 5512 to 48 k, capable of full-duplex operation

Minimum Configuration

- Microsoft® Windows® 95, 98 or NT 4.0 (or higher)

- 90 MHz or faster Intel® Pentium® or compatible microprocessor

- 16 - 24 Mb system RAM

- 800 x 600 x 256-color Super VGA display

- Windows-compatible sound hardware with stereo line level input(s), 16-bit resolution, 44.1 k and/or 48 k sampling rate capable of half- or full-duplex operation

When installed, the program and support files require approximately 12 Mb of hard disk space. Keep in mind that working with audio waveform (*.wav) files can use up a lot of "real" disk space. It's not unusual for a single waveform file to require 500 kb or more and due to the nature of the data, they don't compress very well (if you use disk compression software). The same is true of frequency-domain data files created by the SIA-Smaart® Acoustic Tools™ Analysis Module.

About Sound Hardware

SIA-Smaart Acoustic Tools is designed to work with any Windows-compatible sound device. Only the A/D and D/A portion of the computer's sound hardware are actually used so the software will work well with a wide variety of devices including "off-the-shelf" sound cards for desktop machines and the built-in sound sections of most notebook computers.

Two independent external line-level input channels (typically in the form of a single, stereo connector) are an absolute requirement for recording impulse response measurements using the Dual-FFT method and/or for time-locked MLS recording. We do ***not*** recommend using the microphone level inputs on most computer sound cards for any type of measurement. For microphone measurements, an external mixer or microphone preamp should be used with the computer's line-level inputs. Additionally, the sound hardware must be capable of full-duplex operation if you intend to use the Intelligibility module's internal signal generator as your stimulus signal source.

SIA-Smaart Acoustic Tools makes no use of the sound hardware's synthesis capability (if any). For our purposes the major differentiating factors between sound hardware are the maximum sampling rate, sampling resolution (bits per sample) and signal-to-noise (S/N) ratio. If your computer does not have sound hardware, lacks a line-level input, or you feel that its existing hardware may be problematic for any other reason, there are a number of add-on computer sound devices on the market that you can utilize for audio input and output.

When selecting an aftermarket sound hardware device for use with SIA-Smaart, we recommend you look for the following features and audio characteristics:

- Full-duplex (simultaneous play and record) capability

- 2 independent, external line-level input channels

- 16- to 24-bit sample resolution

- Digital inputs for use with external A/D converters (optional but highly recommended)

- User-selectable sampling rates: The Smaart Intelligibility module supports a number of different sampling rates from 5512 up to 48 kHz. We recommend choosing a sound card which supports the 44.1 and/or 48 kHz sampling rates and *at least* one additional sampling rate, somewhere in the range between 4 kHz and 12 kHz

External Hardware

In addition to your computer and SIA-Smaart Acoustic Tools, several pieces of outboard gear may be required to make measurements in the field. The following is a list of equipment you may wish to consider to complete your measurement kit:

- ***Measurement Microphone:*** We recommend the best omnidirectional microphone with the flattest frequency response you can reasonably afford. A high-quality microphone preamp and an external phantom power supply are also very useful additions to a measurement tool kit.

  Note: A high quality sound pressure level meter with an audio output can be very effective as a measurement microphone.

- ***Mixer or other level adjustment device:*** Although you can set relative signal levels at the computer in many cases, it is helpful to be able to adjust signal levels externally. Also, being able to quickly switch the signals reaching the sound card's inputs can greatly expedite the measurement process in many cases. A compact mixer with quiet microphone inputs and built-in phantom power is ideal, as it puts several pieces of "the puzzle" in one box.

- ***Cables and adapters:*** Y-cables are useful for tying the measurement system into sound systems. Also, as most sound cards use unbalanced (2-conductor) inputs, several sets of adapters which allow balanced to unbalanced connections may be necessary.

Installing SIA-Smaart Acoustic Tools

If you are familiar with the standard Windows software installation process, you probably already have SIA-Smaart Acoustic Tools up and running. If not, don't worry. It is a very straightforward procedure. All you need to do is start the automated installer program and follow the on-screen instructions. **We *strongly* recommend that you close all other Windows programs, *particularly* any automatic anti-virus and/or system monitor software you have running, before attempting to install this software.**

With Microsoft Windows running, place the SmaartLive/SIA-Smaart Acoustic Tools installation CD in your CD-ROM drive. The CD menu program should open automatically. If the CD menu did not start automatically (or you exited to shut down your virus checker, double-click the *My Computer* icon on the Windows desktop then right-click the icon for the CD and select *AutoPlay* from the pop-up menu. After the CD menu program opens, select *Install Smaart* from the top level menu then click *Smaart Acoustic Tools* in the submenu.

When the *Welcome* screen of the install wizard appears, click the *Next* button to continue. On the next screen, select *Install the Full Version (Serial Number Required)* and click *Next*. Enter your assigned *Install Name* and *Serial Number* on the next screen. **Please note that <u>both</u> your *Install Name* <u>and</u> *Serial Number* must be entered <u>exactly</u> as registered.** You should have received a registration card in the package with this manual confirming the *Install Name* and *Serial Number* assigned for your copy of Smaart Acoustic Tools. The *Install Name* and *Serial Number* must be typed exactly as they appear on the registration card.

Software Installation

After entering your registration information, the Install Wizard will guide you through the remainder of the installation process. You will be prompted to choose and installation directory (folder) and given the option to install a shortcut to SmaartLive on the Windows desktop. The installer will suggest "c:\Program Files\SIA-Smaart Acoustic Tools" as the target disk drive and folder for the Acoustic Tools program files. You can accept this default or select a different destination by clicking the *Browse* button.

If the installer detects that SmaartLive is also installed on your computer, you will be asked if you want to have jump modifications made to establish links between SmaartLive and Smaart Acoustic Tools. If you answer yes, it will automatically create or modify links files for all three programs to allow jumping and passing data between SmaartLive and the two Acoustic Tools program modules. See *Setting Up Links to Other Programs* on page 26 for more information.

If you have previously installed the 30-day demonstration version of SmaartLive, we recommend that you install the full (Standard or Upgrade) version of SmaartLive to the same folder used for the demonstration version. This will allow the full version the overwrite the demo during installation. If you want to install the full version to a *different* drive or directory, you should *uninstall* the demo version using the Windows *Add/Remove Software* control panel prior to installing the full version.

When you have made your selections, the installer will create the specified program directory, copy the program and support files into it, register the programs, and create shortcuts in the *Programs* submenu in the Windows *Start* menu and (optionally) the Windows desktop. When the installation is complete, click the *Finish* button to exit the installer.

That's it! If you were asked to restart your computer after installation was completed you should do so before attempting to use either of the two Acoustic Tools program modules. Otherwise you can use the software immediately. Just press the *Start* button on the Windows Taskbar, click on *Programs* in the menu, then click on the listing for *SIA-Smaart Acoustic Tools* and select either *SIA-Smaart Acoustic Tools Intelligibility* or *SIA-Smaart Acoustic Tools Analysis.* If you experience any problems during or after installation, refer to *Appendix A* of this manual, beginning on page 120, for troubleshooting and technical support information.

<u>*Note:*</u> If you have more than one sound device installed on your system the SIA-Smaart Intelligibility and Analysis modules will initially select the Windows default audio input and output (Wave-in and Wave-out) devices. To change the default device selections for either program, select *Devices* from the program's *Options* menu and choose the appropriate *Wave-in* and *Wave-out* devices and the sampling resolution (bits per sample) for each on the *Devices* tab of the *Options* dialog box.

Chapter 2: Intelligibility Module Functions and Commands

Navigating in the Intelligibility Module
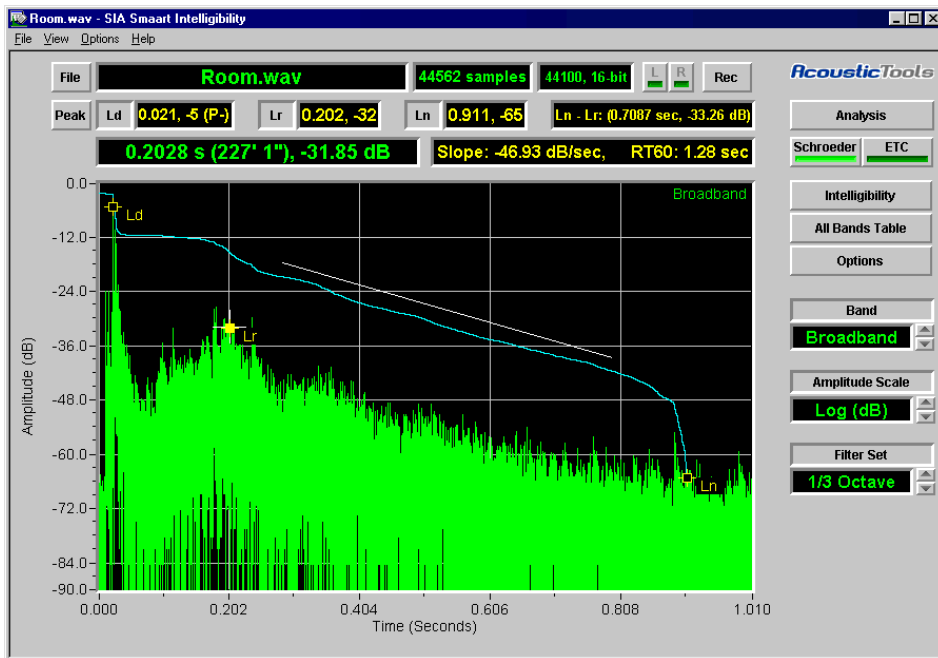
The Intelligibility module user interface is designed to put the functionality required for the most common impulse analysis tasks literally at your finger tips by giving you one-click access to the most commonly used functions. The use of pop-up menus and context-sensitive dialog boxes mean you may never have to open a menu to get at all but the most advanced features.

In many cases, simply the clicking the label or readout field for an on-screen control or readout will allow you to access nearly any associated parameter or option you are likely to need. Advanced users will find the high level of control and flexibility one would expect in a truly professional measurement and analysis application easily accessible through menu commands, hot keys, and the Options dialog box.

The following is a brief description of the basic elements that make up the Intelligibility module program window. Detailed information on Intelligibility module displays and functions can be found in *Intelligibility Module Functions,* beginning on page 10. The section on *Intelligibility Module Commands,* beginning on page 29, contains complete listing of all menu commands and options.

The Menu Bar

File View Options Help

Virtually all of the program functions that are available as on-screen controls and/or keyboard commands are also accessible through the pull-down menus on the *Menu Bar.* Menu commands also provide access to many less commonly used features.

To activate a pull-down menu from the Menu Bar, click on its name using your mouse or hold down the [Alt] key while pressing the key corresponding to the underscored letter in the menu name. When a menu is open, commands may be selected by clicking on their names with your mouse or by typing the underscored letter.

Menu commands followed by ellipses (three periods...) call dialog boxes. An arrow-head at the right of a menu item indicates that this command activates a nested "fly-out" submenu. For explanations of Intelligibility module menu commands, please refer to the Help file and manual sections titled *Intelligibility Module Commands* beginning on page 29.

File Information

File **Room.wav** 44562 samples 44100, 16-bit L R Rec

This section of the Intelligibility Module window provides information about the primary wave file currently loaded. The (Open) *File* button calls an *Open* file dialog box allowing you to select a wave file to load. The next field to the right displays the file name of the wave file after opening it.

Immediately to the right of the file name field are two fields that display the number of samples in the file and the sampling rate and resolution (bits/sample) at which the data was recorded. When a stereo wave file is loaded, the (L and R) *Channel* buttons select which channel is displayed. The Record *(Rec)* button opens the Intelligibility module's Impulse recorder.

Level Marker Coordinates

Peak | Ld | 0.021, -5 (P-) | Lr | 0.202, -32 | Ln | 0.911, -65 | Ln - Lr: (0.7087 sec, -33.26 dB)

The second row of display fields at the top of the Intelligibility module window give numeric coordinates for the time and amplitude positions of Ld, Lr, and Ln markers (see *Calculating Intelligibility from an Impulse Response* on page 10). The first number in each field is the time position for the corresponding marker in milliseconds followed by the amplitude value in decibels. The readout for the Ld marker has an additional notation indicating the polarity (P+ or P−) of the highest peak in the impulse response. The *Ld* marker is normally positioned directly on the this peak clicking the *Peak* button to the left of the *Ld* field will return it to the peak position it is moved.

The larger field on the right is a readout for either the pencil tool (see page 24) or the relationship between the *Lr* and *Ln* markers. When a line is drawn on the plot with the pencil tool this field displays the time and amplitude difference between the end points of the line as well as the distance the time difference represents. When the normal impulse response plot is displayed and no line is present, the readout area displays the time and amplitude difference between the *Lr* and *Ln* markers.

Cursor Readout

0.2028 s (227' 1"), -31.85 dB

The readout field immediately above the main plot area on the left gives you *x* and *y* axis coordinates for the Intelligibility module's precision tracking cursor when the mouse cursor is positioned over the main plot area. *X*-axis coordinates are given in seconds and distance (relative to the beginning of the file). Y-axis coordinates are given as decibels (dB) for the ETC and impulse plot when Logarithmic amplitude scaling is selected or as percentage of full scale (based on the bits per sample at which the wave file was recorded) for Linear scaling.

Slope Readout

Slope: -46.93 dB/sec,　　RT60: 1.28 sec

The *Slope* field immediately above the main plot to the right of the tracking cursor readout (see above) is a readout for either the pencil tool (see page 24) or relationship between the Lr and Ln markers (in dB scaling mode only). When a line is drawn on the plot with the pencil tool, this field displays the *Slope* of the line in dB/second, and the

equivalent RT60 value. In dB amplitude scaling mode, when the normal impulse response plot displayed and no line is present, the readout area displays slope and RT60 values based on the locations of the *Lr* and *Ln* markers.

Plot Area

The largest portion of the Intelligibility module's main window is devoted to the display area for impulse response and ETC data plots. The plot area will appear blank until a wave (*.wav) file is loaded or recorded. When a file is loaded, a time-domain (time vs. energy) representation of the wave file data appears in the plot area.

Jump To Other Programs

Clicking the button in any display mode will automatically close the Intelligibility module while opening the Intelligibility module and loading the file that was open in the Intelligibility module.

Clicking the Acoustic Tools logo above the Analysis button will pop up a menu that allows you to jump to, and pass data to other programs if there are jumps defined it imjumps.txt. *See Setting up Links to Other Programs* on page 26 for more information.

Display Controls

The controls to the right of the plot area in the main program window affect the appearance of the main impulse/ETC plot and open the Intelligibility module's secondary graph windows. For more information about these controls please refer to the following topics:

- *Intelligibility Graph,* page 36
- *All Bands Table,* page 38
- *All (Options),* page 47
- *Amplitude Scale Commands,* page 43
- *Octave and 1/3 Octave Band Filters,* page 46
- *About Intelligibility Module Filters,* page 25
- *Calculating Intelligibility from an Impulse Response,* page 10

Intelligibility Module Functions

Calculating Intelligibility from an Impulse Response

When a wave file is opened or a new file is recorded in the Intelligibility module, the file's raw time-domain audio data is displayed in the plot area with amplitude on the *y*-axis and time on the *x*-axis. The file name, number of samples in the file, and its sampling rate and resolution (8-24 bits/sample) are displayed in the file name and information fields at the top of the Intelligibility module window. If a stereo file is loaded, only one channel is displayed. You can switch between channels using the *L* and *R* (left/right) channel selector buttons to the right of the file name and information fields.

You can select whether to display either the unfiltered broadband data or a 1- or 1/3-octave passband using the *Band* spinner list to the right of the plot. The available (1- or 1/3-octave) bands in this list will depend on the currently selected .

Both Linear and Logarithmic (decibel) amplitude scaling are available however Log (dB) amplitude scaling is normally used when performing intelligibility calculations. Amplitude scaling is selected using the Amplitude Scale spinner located to the right of the plot.

Positioning the Level Markers

In dB scaling mode three markers appear on the plot labeled *Ld* (Level direct), *Lr* (Level reverberant), and *Ln* (Level noise). The time and amplitude coordinates for these three markers are used as the basis for %ALCons and Clarity calculations. The standard STI and RSTI calculations are not dependent on the level marker positions.

By default, the Schroeder Inverse Integration curve is calculated and displayed above the raw time-domain data of the impulse response. The Schroeder trace is intended mostly as a guide to help you identify the overall trends in the impulse response. The three level markers should normally be positioned on the impulse response trace itself.
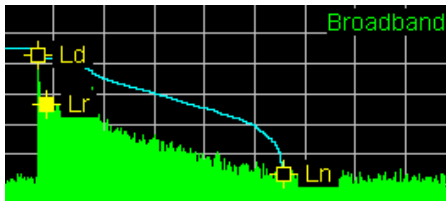
| Level marker positions for fairly a typical impulse response. | Level marker positions for an impulse response with a late buildup of reverberant energy. |
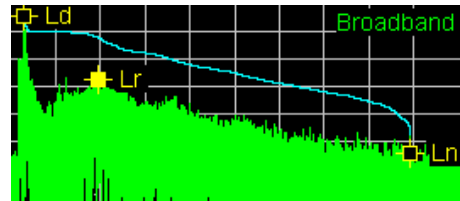
The *Ld* marker is automatically set at the position of the highest point in amplitude value found in the time-domain trace when a file is loaded. This will normally represent the arrival of direct sound. The *Lr* and *Ln* markers must be positioned manually. Any of the markers can be moved anywhere on the plot using your mouse. The time and amplitude position (milliseconds, dB) for each marker are displayed in the marker coordinate readout fields above the plot.

Peak

To calculate intelligibility values other than STI and RSTI from an impulse response, first confirm that the *Ld* marker is in the correct position. If it has been moved since the file was opened or the position is otherwise incorrect, click the (Set LD to) *Peak* to return it to the highest peak in the impulse response or manually move it to the top of the first arrival peak by dragging the marker with your mouse. The *Lr* marker should then be positioned at the top of the reverberant decay slope (sometimes called the "elbow" or "knee" of the impulse response), and the *Ln* and the *Ln* marker at the approximate point where the decay slope meets the noise floor, sometimes referred to as the "saddle."

You will need to repeat this procedure for each octave or 1/3-octave band you intend to analyze. The Intelligibility module "remembers" the marker positions for each passband during run-time and will return the makers to the correct position for each band as you switch between bands during a session. Please note that the marker positions are not stored when the file is closed.

The time and amplitude difference between the *Lr* and *Ln* markers, as well as the distance the time difference represents, the *Slope* (in dB/second) and the equivalent RT60 reverberation time are normally displayed in the slope readout and relative cursor readout fields above the plot.

Note that you can find this same information for any two points on the plot by holding down the [Shift] key as you click and "drag" (move the mouse while holding down the left mouse button) the mouse cursor across the plot to draw a line. While the line is present, the information in the slope readout and relative cursor readout fields is calculated relative to the end points of the line, rather than the *Lr* and *Ln* markers. Clicking once on the plot while holding down the [Shift] key erases the line. See *Finding Time and Amplitude Differences* on page 24 for more details.

The All Bands Table

All Bands Table

After positioning the level markers, for the passbands you want analyze, you can click the *All Bands Table* button to display a table of reverberation and decay characteristic, Clarity values and %ALCons (if applicable) for multiple octave- or 1/3-octave pass-bands. The *All Bands Table* will include all bands in the current filter set that have been displayed since the current file was opened.

All Bands Table

Band	RT60	EDT	DTR	%Alcons(S)	%Alcons(L)	C35	C10	C20	C50	C80
Broadband	1.34	0.37	29	×	×	7.80	7.14	7.68	7.83	8.02
63 Hz	1.93	0.78	4	×	×	-6.91	-8.22	-7.52	-4.95	-2.27
125 Hz	1.25	1.25	10	×	×	-2.02	-2.50	-2.08	-1.91	-1.05
250 Hz	1.07	0.57	21	×	×	2.07	1.46	1.93	2.16	2.89
500 Hz	1.32	0.73	14	×	×	0.83	0.15	0.73	0.87	1.15
1 kHz	1.12	0.61	17	×	×	-0.97	-1.95	-1.11	-0.92	-0.60
2 kHz	1.14	0.45	23	2.25	2.50	6.87	6.07	6.74	6.90	7.02
4 kHz	1.17	0.36	27	×	×	10.44	10.00	10.37	10.48	10.69
8 kHz	0.47	0.29	33	×	×	13.57	11.66	13.39	13.62	13.83

Save   Print   Close

The *Intelligibility* dialog box and the *All Bands Table* both have a *Save* button that allows you to export the values to an ASCII file for import into other applications or documents. Both can also be printed directly from the Intelligibility module.

The Intelligibility Graph

Intelligibility

Clicking the *Intelligibility* button opens the Intelligibility dialog box. This window displays the results intelligibility calculations, and several quantitative acoustical values. With the exception of STI and RSTI, all of the values displayed in this window apply only to the passband currently displayed in the main program window.
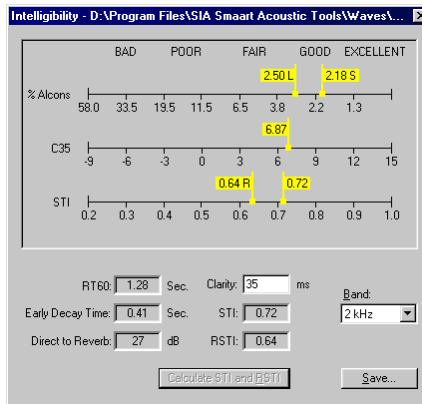
The Intelligibility dialog window is "mode-less," meaning you can leave it open while you continue to work in the main program window. When you move the marker positions on the main plot, values displayed in the Intelligibility dialog window update themselves automatically (where applicable) to reflect changes as they are made. All of the information shown in the Intelligibility graph window can be saved to an ASCII text file suitable for import into a spreadsheet or other applications by clicking the *Save* button at the bottom of the window.

Intelligibility Scales

The top half of the Intelligibility dialog box is devoted to the *Intelligibility Graph*. In this section, objective intelligibility values are plotted on three horizontal scales ranging from "Bad" to "Excellent" (left to right). Separate scales are provided for three types of intelligibility values:

- Percentage Articulation Loss of Consonants (%ALCons)
- Clarity (early-to-late energy ratio) at *n* milliseconds (C*n*)
- Speech Transmission Index (STI) and *Rapid* Speech Transmission Index (RSTI)

Values for the current measurement are indicated by yellow "flags" plotted on these three scales. Two of the scales have two "sub-types" so there are separate flags for each. The results for both the long and short forms of the %ALCons calculations are plotted on the *%ALCons* scale with the type indicated on the individual flags by the letters *"L"* for long form and *"S"* for short form. Note that %ALCons will appear only after setting the Ld, Lr, and Ln markers in the 1-octave passband centered on 2 kHz.

Two different values will appear on the *STI* scale after running the STI calculation routine: STI and Rapid STI (RSTI). The latter is indicated by a flag with the letter "R" after the number. The STI and RSTI calculations require fairly significant processing time so they are calculated only upon request when you click the *Calculate STI and RSTI* button. Once calculated, these values will not change for a given file. Their values are not dependent upon the level markers or the *Band* (broadband or passband) currently selected for display.

Other Quantitative Acoustical Values

The lower section of the Intelligibility dialog box is devoted to display fields for some additional acoustical values and controls for the Clarity scale and display passband.

RT60 — RT60 or "reverberation time" is an expression of the reverberant decay rate of the system under test. By convention, reverberation time is expressed as the amount of time it would take the system to decay 60 dB at the given rate of decay (RT60). The decay rate is determined from the relative positions of the *Lr* and *Ln* markers on the plot in the main program window. The value shown in this field applies only to the *Band* (broadband or passband) currently selected for display.

Early Decay Time — This is an expression of the rate at which the system under test completes earliest stage of its decay. This value is dependent on the relationship between the *Ld* and *Lr* markers on the main plot and applies only to the *Band* (passband) currently selected for display. As with RT60, the early decay time is expressed as the amount of time (in seconds) that would be required for the signal to decay 60 decibels at the given rate of decay.

Direct to Reverb — The direct-to-reverberant level. This is the relative difference between the level of direct sound to the level of reverberant energy at the top of the reverberant decay slope. This value (given in decibels) is simply the difference in level between the *Ld* and *Lr* markers however it is an important factor in the overall intelligibility of a room/system.

Clarity **—** The Clarity field sets the number of milliseconds used for the Clarity Factor scale in the upper portion of the window. For example, entering 35 in this field will set the clarity scale to C35.

STI and RSTI — The STI and RSTI fields show the calculated Speech Transmission Index values after running the *Calculate STI and RSTI* routine. Note that STI and RSTI values are calculated only when you click the *Calculate STI and RSTI* button. These calculations can take some time to process, particularly on long files, so they are done only on demand. This only needs to be done once for any file, however. The STI and RSTI values are ***not*** dependent on either the positions of the level (Ld, Lr, and Ln) markers or the passband currently selected for display.

Band — The *Band* selector serves the same function as the *Band* selector to the right of the plot in the main program window. It selects the passband for both the Intelligibility window and the main impulse response plot— either the unfiltered *Broadband* data or a 1- or 1/3-octave passband as determined by the filter set currently in use.

Recording an Impulse Response

MLS or Dual-FFT?

The first decision you will need to make before recording an impulse response measurement is which recording technique to use. The Intelligibility module's *Record Impulse* feature allows you to acquire the impulse response of a room/system under test using a Maximum Length Sequence (MLS) or dual Fast Fourier Transform (Dual-FFT) transfer function technique. MLS and Dual-FFT are basically just two different ways of comparing what goes into a system with what comes out. There are, however, fundamental differences between the two techniques.

MLS is a single-channel, stimulus-dependent measurement technique that requires the use of an internally generated MLS stimulus signal. Since the stimulus signal is *known,* only the output of the system under test needs to be recorded (hence the term "single-channel"). Dual-FFT is a *dual*-channel, stimulus-*independent* measurement technique that can use virtually any continuous signal, including pink noise (recommended for most acoustic/intelligibility measurement applications) or music, as a stimulus. This technique requires that you capture *both* the stimulus signal and the output of the system under test during the measurement procedure. Each of these two approaches has its advantages.

The MLS Method

Of the two techniques, MLS has the potential to offer somewhat superior noise rejection. Since the MLS stimulus signal is pseudo-random, not random, the stimulus is a precisely known quantity and may be easier to correlate with the measured system response. When using MLS to acquire an impulse response, the signal returned from the device or system under test is compared to the original, internally generated data sequence *exactly as sent* to the system under test. The MLS stimulus signal also has a lower crest factor than truly random noise so clipping the measurement system's Analog-to-Digital (A/D) converter is s little less of a concern.

Some possible *disadvantages* with MLS are the high concentration of energy in the top octaves and that this method is probably a little more demanding in terms of the computer's audio input/output (I/O) hardware. For MLS measurements, the computer sound hardware needs to be capable of full duplex operation (playing *and* recording simultaneously) and it is more critical that the analog I/O stages and D/A and A/D conversion process be as transparent as possible. Since the measured signal is compared directly to the original (digital) MLS data, it is unavoidable that some part of what you end up measuring will be the quality of the I/O hardware in the measurement system itself (i.e., the computer and sound hardware). Also, since the MLS stimulus signal is more "white" than pink noise, you need to exercise greater care to avoid damaging the high-frequency elements of loudspeaker systems under test.

Note that the MLS technique can be used with or without an external loopback on one audio I/O channel as a time reference (see diagram on page 18). Due to a limitation of Microsoft Windows, the loopback is required to accurately measure the delay time through the system under test when using MLS. This loopback signal is used *only* as a time reference and should not be confused with the *reference* signal in Dual-FFT measurements.

The Dual FFT Method

The Dual FFT impulse recording technique is stimulus-independent. The Intelligibility module provides an internal pink noise generator for use with the Dual-FFT impulse response recorder but virtually any continuous test signal generated inside or outside the computer may be used. In a Dual-FFT measurement, a test signal is split, preferably outside the computer, with one branch returning directly to one computer audio

input as a reference. The other branch is sent through the system under test and the output of the system is returned to a second computer audio input as the measurement signal (see diagrams on pages 20 and 21). These two signals are then compared in the frequency domain using a transfer function calculation and the results are converted back to the time domain for display and analysis.

Since both the reference and measurement signals travel through the same processes (analog I/O stages, A/D and D/A) leaving/entering the measurement system, the Dual-FFT technique can be more forgiving of less-than-perfect computer sound hardware. This technique can also be used with half-duplex sound hardware (capable only of playing *or* recording at any one time) provided an external test signal source is used.

The relative disadvantage of the Dual-FFT method is that it is typically necessary to average the measurement over a longer period of time to approach the kind of signal-to-noise ratios possible with MLS. This method also requires much more processing power than MLS (less of an issue now than in the past) and is theoretically less repeatable since the test signal is a more random quantity. In practice, you can obtain very solid, repeatable measurement results using either recording method and it is even possible to combine the two by using an MLS stimulus with the dual-FFT method.

Recording an Impulse Response Using MLS

The Maximum Length Sequence (MLS) technique for measuring the impulse response of a system is stimulus-dependent, requiring the use of an MLS stimulus signal. An MLS signal is "pseudo-random," generated using a system of bit-shift registers and a known group of "seed" numbers. Since the output of the system under test is compared to MLS data generated internally by the Intelligibility module, it is critical that the stimulus signal matches the internally generated data precisely.

It is not an absolute requirement that you use the Intelligibility module's internal MLS signal generator. Any two MLS stimulus signals of identical length, generated at the same sampling rate and resolution using identical seed numbers, should be identical. And in practice, most MLS generators do, in fact, use the same seed numbers however there is no accepted standard that says they must. Using the internal MLS generator is probably the easiest way to be absolutely sure the stimulus signal and program's internal MLS reference parameters are identical.
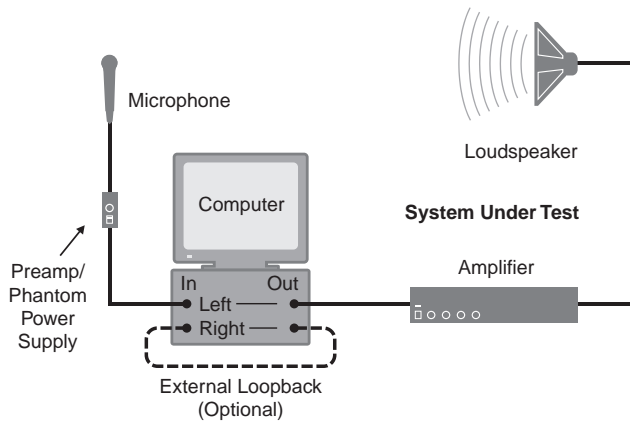
The diagram above shows a typical measurement system setup for MLS measurement. Note that the use of an external loopback, connecting one of the computer outputs to one of the inputs, is optional. The loopback is required (due to a limitation in Microsoft Windows) only as a time reference if you need to know the delay time through the system under test. You can make an MLS measurement that is valid in of every other respect without a loopback. When recording without a loopback, the initial peak in the impulse response, indicating the arrival of direct sound at the measurement microphone position, will be positioned arbitrarily at 10 or 100 milliseconds, depending on the MLS length.

Rec

To begin the recording process, open the Intelligibility module and click the Record *(Rec)* button. When the Record Impulse dialog box appears, click the MLS tab at the top of the dialog box to display the MLS recording controls.

MLS: Setting Measurement Parameters

If you will be using the internal MLS generator to make your measurement, set the *Level* control in the *MLS Signal Generator* section to a low number initially. Set the *Generate Channel(s)* control to match the computer's line output channel you are using to send the stimulus signal to the system under test or choose *Stereo* to send the signal on both output channels. If you are using the external loopback option, the *Generate Channel(s)* will be set automatically to *Stereo*.

In the *Impulse Output File* field, you can type the path and file name for the wave file to be used to store the measurement results or click the *Browse* button and use the *Open* file dialog box to navigate to the desired folder and specify a file name. If you will be using an external loopback as a time reference for the measurement, make sure the *Use Loopback for Time Alignment* check box is checked. Set the *Measurement Channel* selector to match the computer line input used for your measurement microphone/preamp. Notice that when the *Use Loopback for Time Alignment option* is checked, the *Reference Channel* is automatically set to the opposite channel.

When the *Averages* value is set to a number greater than 1, multiple MLS data sequences are recorded and the results are averaged together to produce the finished measurement. Each doubling of the number of averages will yield 3 dB more signal to noise in the finished measurement however the time required to make each measurement is increased. As a general rule, you will want to use higher numbers of averages in noisier environments and/or when you are restricted to using lower sound pressure levels to stimulate the room/system under test.

The sampling rate *(SR)* determines the frequency content of the completed measurement. To measure the full audio spectrum, a sampling rate of 44.1 k or greater is required. Using lower sampling rates will restrict the highest frequency obtainable to one half of the sampling rate (the *Nyquist* frequency).

The *MLS Length,* together with the sampling rate, determines the *Time Constant* of the sequence, also called the "time window." The *Time Constant* for the selected *MLS Length* and *SR* values is automatically displayed below the *MLS Length* field. To obtain a solid impulse response measurement, **the MLS time constant must be longer than the decay time of the system under test.**

As a general rule, a time window of .7 to 1.5 seconds is sufficient for most small- to mid-sized rooms. Larger rooms with longer decay times require a longer time window. When in doubt, it never hurts to go too long — in fact, it will help to improve the signal-to-noise ratio of the measurement. This does, however, increase both the time required to make the measurement and the resulting wave file size. Larger files will take more time to filter and analyze. If you find you have overestimated when the measurement is completed, you may want to use the *X Zoom* feature and *Save As* command to edit out some of the unneeded "noise tail."

MLS: Recording the Impulse Response

If you are using the internal *MLS Signal Generator,* click the *Generate MLS Signal* button to begin sending a signal and carefully bring the *Level* control up to desired signal level. Remember that the MLS signal is essentially white noise with most of its energy in the higher octaves. You need to exercise care not to overdrive high-frequency drivers in the loudspeaker system being used to stimulate the room. Note that you may also need to set the overall *Wave* output level in the Windows *Volume Control* to achieve the desired output level.

Adjust the measurement microphone preamp gain and/or the computer's *Line-In* level in the Windows *Recording Control* if necessary so that the measurement signal and loopback signal (if applicable) are running at about –12 dB on the *Input Meters* and click the *Record* button. The Intelligibility module will record audio data based on you measurement parameters, process the data and write the results to the specified output file. When the recording process is completed the *Record Impulse* dialog box closes automatically and the new file is displayed in the main *Plot Area.*

Recording an Impulse Response Using Dual FFT

The *Dual FFT* impulse response measurement technique is stimulus-independent. This technique can be used with internally or externally generated pink noise (recommended for most acoustical measurement applications) or virtually any other continuous signal (e.g., music) as a stimulus.
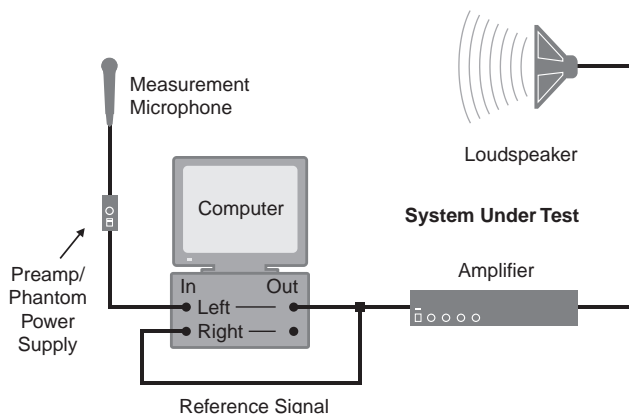
Dual FFT measurement setup using the internal signal generator
(requires full-duplex capable computer sound hardware).

When using this recording method, a test signal is split, normally outside the computer, with one branch returning directly to one computer audio input as a reference signal. The other branch is sent through the system under test and the output of the system is returned to the other input as the measurement signal.
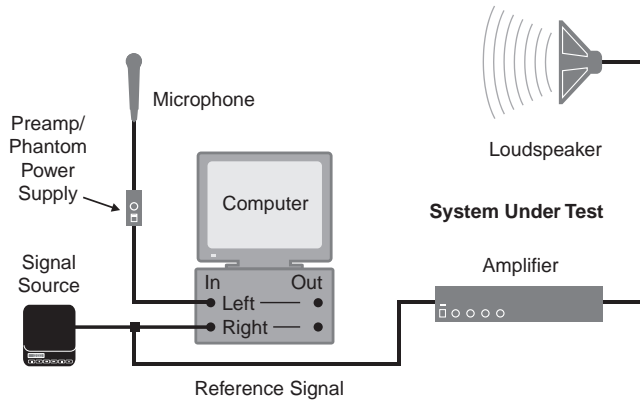
Dual FFT measurement setup using an external signal source.

The diagram above shows a typical measurement system setup for Dual FFT measurement using the internal signal generator as the reference signal source. The diagram below shows a Dual-FFT measurement setup using an *external* signal source.

Rec

To begin the recording process, open the Intelligibility module and click the Record *(Rec)* button. When the Record Impulse dialog box appears, click the Dual FFT tab at the top of the dialog box to display the Dual FFT recording controls.

Dual-FFT: Setting Measurement Parameters

If you intend to use the internal signal generator to make your measurement, set the *Level* control in the *Pink Noise Generator* section to a low number initially. In the *Impulse Output File* field, you can type the path and file name for the wave file to be used to store the measurement results or click the *Browse* button and use the *Open* file dialog box to navigate to the desired folder and specify a file name.

When the *Averages* value is set to a number greater than 1, the specified number of FFT frames are recorded and processed and the results averaged together to produce the finished measurement. Each doubling of the number of averages will yield 3 dB

better signal-to-noise in the finished measurement however the time required to make each measurement is increased. As a general rule, you will want to use higher numbers of averages in noisier environments and/or when you are restricted to using lower sound pressure levels to stimulate the room/system under test.

The sampling rate *(SR)* determines the frequency content of the completed measurement. To measure the full audio spectrum, a sampling rate of 44.1 k or greater is required. Using lower sampling rates will restrict the highest frequency obtainable to one half of the sampling rate (the *Nyquist* frequency).

The *FFT* size, together with the sampling rate, determines the *FFT Time Constant*, also called the "time window." The *FFT Time Constant* for the selected *FFT Size* and *SR* values is automatically displayed below the *FFT* field. To obtain a solid impulse response measurement, **the FFT time constant must be longer than the decay time of the system under test.**

As a general rule, an FFT time constant of .7 to 1.5 seconds is sufficient for most small- to mid-sized rooms. Larger rooms with longer decay times require a longer time window. When in doubt, it never hurts to go too long — in fact, it will help to improve the signal-to-noise ratio of the measurement. It does, however, increase both the time required to make the measurement and the resulting wave file size. Larger files will take more time to filter and analyze. If you find you have overestimated when the measurement is completed, you may want to use the *X Zoom* feature and *Save As* command to edit out some of the unneeded "noise tail."

Dual-FFT: Recording the Impulse Response

If you are using the internal *Pink Noise Generator,* click the *Generate Pink Noise* button to begin sending a signal and carefully bring the *Level* control up to desired level. Note that you may also need to set the overall *Wave* output level in the Windows *Volume Control* to achieve the desired output level. If you are using an externally generated signal, start sending signal to the system under test and the computer. In either case, adjust appropriate external gain controls and/or the computer's *Line-In* level in the Windows *Recording Control* as necessary so that the reference signal is running at about -12 dB on the *Input Meters.* Make sure the *reference* signal is being received on computer's *Left* input channel (indicated by the blue meter "LEDs" on the right) and the *measurement* signal from the microphone is on the *Left* input (indicated by the green meter "LEDs" on the left).

Adjust the measurement microphone preamp gain if necessary so that both the reference and measurement signals are running at about –12 dB on the *Input Meters* and click the *Record* button. The Intelligibility module will record audio data based on your measurement parameters, process the data and write the results to the specified output file. When the recording process is completed the *Record Impulse* dialog box closes automatically and the new file is displayed in the main *Plot Area.*

Finding Time and Amplitude Differences

Ln - Lr: (0.7087 sec, -33.26 dB)

In decibel (dB) amplitude scaling mode, the time and amplitude difference between the *Lr* and *Ln* markers on the impulse response plot, the *Slope* (in dB/second) and the equivalent RT60 reverberation time are normally displayed in the slope readout and relative cursor readout fields above the plot. The Intelligibility module has another feature that allows you to find this same information, as well as the *distance* the time difference represents (given the speed of sound), for *any* two points on the plot in any display mode.

Slope: -46.93 dB/sec,   RT60: 1.28 sec

If you press the [Shift] key on your keyboard while the mouse cursor is positioned over the main plot, the mouse cursor changes to a tiny pencil. If you hold down the [Shift] key while "dragging" the cursor (moving the mouse while holding down the left mouse button) across the plot, you will draw a line on the plot. While the line is present on the plot, the information in the *Relative Cursor Readout* is calculated relative to the end points of the line, rather than the *Lr* and *Ln* markers. Clicking once on the plot while holding down the [Shift] key will erase the line.

A Shortcut for Positioning the Ld, Lr, and Ln Markers

Notice that if you press the [Alt] key while the mouse cursor is positioned over the plot, the cursor mouse changes to a marker symbol with a plus sign in the center. Clicking the left mouse button while holding down the [Alt] key moves the selected level marker to the mouse cursor location.

You can select the *(Ld, Lr, or Ln)* markers using the [Tab] key or by simply clicking a marker with your mouse. You can also move the markers around on the plot using the arrow keys or by clicking on a marker and dragging it (moving the mouse while holding down the left mouse button) from one place to another. For more information on setting the *Ld, Lr, or Ln)* markers refer to *Calculating Intelligibility from an Impulse Response* on page 10.

Zooming With the Mouse

Your computer's mouse (or other pointing device) can be used to set both the time (*x*-axis) and amplitude (*y*-axis) ranges on the main Intelligibility module plot. Notice that if you press the [Ctrl] key on your keyboard while the mouse cursor is positioned over the main plot, the mouse cursor changes to a magnifying glass with a plus sign in the center. If you click on the plot while the [Ctrl] key is depressed, the plot will zoom in on both the *x* and *y* axes by a preset increment centered on the cursor location. Clicking and "dragging" (holding down the left mouse button while moving the mouse) while holding down the [Ctrl] key will draw a "rubber band" box on the plot. When the mouse button is then released, the area enclosed by the box will be expanded to the full *x* and *y* limits of the plot.

When you press the [Ctrl] and [Shift] keys simultaneously, the mouse cursor changes to a magnifying glass with a *minus* sign in the center. If you click on the plot while the [Ctrl] is depressed, the plot will zoom *out* on both the *x* and *y* axes by a preset increment, centered on the cursor location. To reset the time (*x*-axis) range of the plot to the full time range of the wave file and the amplitude (*y*-axis) range to the default range as specified on the Graph tab of the Options dialog box, click once with your mouse anywhere in the margins of the plot or press the [Home] key.

About Intelligibility Module Filters

All filtering in the SIA Intelligibility module is performed in the time domain. The program comes with a set of 1- and 1/3-octave digital (FIR) filters. Filter sets are selected using the *Filter Set* spinner to the right of the main plot. The passband is then selected from the *Band* spinner (shown below) or using the *Band* list box in the *Intelligibility* dialog window. The individual bandpass filters are stored in ASCII text format to allow advanced users to make change the stock filters or write their own.

The ASCII filter files are kept in the *Filters* sub-directory (folder) in the main program folder. The Intelligibility module creates its lists of available filters each time you start the program based on the information in another text file called "filterfiles.txt," also located in the *Filters* folder. This file contains a list of the individual filter file names and other information about each file such as the sampling rate, filter type, and list name.

In addition to the 1/1 and 1/3-octave filters "types," there is also a provision for a third, "User" filter type. User filters (if present) are made available from the *Band* spinner whether the 1-octave or 1/3-octave filters sets are selected. The Intelligibility module accepts either FIR or IIR time-domain filters. For more information on customizing the filter lists, please refer to the instructions included in filterfiles.txt.

Setting up Links to Other Programs

In addition to the built-in links between the Intelligibility and Analysis modules (see *Analysis* on page 30 and *Intelligibility* on page 70), both SIA-Smaart Acoustic Tools program modules can be configured to start and pass Wave (*.wav) or ASCII (*.txt) data files to many other programs automatically using their jump function. The Acoustic Tools jump function is configured by a pair of text files named *wrjumps.txt* and *imjumps.txt*.

The *wrjumps.txt* file defines jumps for the *Analysis* module and *imjumps.txt* defines jumps for the Intelligibility module. Both programs scan the SIA-Smaart Acoustic Tools *System* folder on start-up to look for these files. If the program's jump file exists and has one or more jumps configured, its jump function is enabled automatically. When the jump function is enabled, clicking on the *Acoustic Tools* logo that appears in the upper right corner of the main program window will pop up a jump menu.

The Acoustic Tools System folder is located in the Acoustic tools program folder. The default location of this folder is: c:\Program Files\SIA-Smaart Acoustic Tools\System.

Note that *wrjumps.txt* and *imjumps.txt* are created automatically by the Acoustic Tools installer if SmaartLive was detected on the computer during installation and

you answered yes when the installer asked if you wanted to make jump modifications. Otherwise, you can easily create and/or edit the *wrjumps.txt* and *imjumps.txt* files yourself using any ASCII text editor such as the Windows Notepad.

The example of a Smaart Acoustic Tools jump definition shown below would create a command labeled "Excel" in the jump menu for the Intelligibility or Analysis module if added to *wrjumps.txt* and *imjumps.txt*. When selected from the jump menu, this command would open MicroSoft® Excel® and load the last ASCII text file saved by the program into a new Excel worksheet.

```
[Excel]
PATH=c:\Program Files\Microsoft Office\Office\Excel.exe
ARGS=
EXIT=N
SMAART=N
WAVE=N
TEXT=Y
```

Each jump must begin with a section title enclosed in brackets that will also become provides the text label for its jump menu command in Intelligibility or Analysis module pop-up jump menu. The *PATH* line must provide the complete path and file name of the target program. The *ARGS* line can be used to pass additional command line arguments to the target application if necessary.

The remaining lines are yes or no questions that should have either a "Y" or "N" after the equals sign (=) as follows:

- If EXIT=Y, the Smaart program you are jumping from shuts down as it starts the target program.

- The SMAART line is for use when jumping between Smaart Acoustic Tools and SmaartLive (4.1 or higher) and should be set to "N" for jumps to other programs.

- If WAVE=Y, the Smaart program you are jumping from will pass a wave file it has open (File 1 only in the Analysis module) to the target application. This line should be set to "N" if the TEXT line is set to "Y."

- If TEXT=Y, the last text file saved by the Smaart program you are jumping from will be passed to the target program. This line should always be set to "N" if WAVE=Y.

Making a Screen Capture

Microsoft Windows has a feature built in that allows you to capture the active window as a bitmap. These captured "screen shots" are an easy way to include Smaart data displays as illustrations in reports and other documents.

To make a screen shot, click on the window you want to capture with your mouse to make sure it is the "active window" and press [Alt] + [PrtScrn] (also be labeled as F13 on some keyboards). This copies an image of the active window to the Windows clipboard. The image on the clipboard can be pasted directly into some word processor and spreadsheet applications, others may require you to save the image as a bitmap file first and then import the file into the document.

To save the captured image as a bitmap using the Paint program included with Windows, open the program by clicking the *Start* button on the Windows Taskbar and selecting *Programs* > *Accessories* > *Paint*. In the Paint program select *Paste* from the *Edit* menu or press [Ctrl] + [V] to bring in the image from the clipboard. If you see a message saying that the image is too large and asking if you want to enlarge the bitmap, click *Yes.* Then, while the pasted image is still selected, open the *Edit* menu and select *Copy To.* In the *Copy To* dialog box navigate to the folder where you want to put the file, type a file name, and click the *Save* button.

Note: All SIA-Smaart applications let you set up your own display and printing *Color Schemes*. This feature can be very useful when working with screen shots. For example, you can set up a *Color Scheme* with white as the background color and dark traces to optimize the captured image for printing to a black and white printer. *Color Schemes* are accessible through the *Colors* tab of the *Options* dialog box in both the Intelligibility and Analysis.

Intelligibility Module Commands

File Menu Commands

Open

File Menu > Open

Keyboard Command = [Ctrl] + [O]

Clicking the *File* button in the upper left corner of the main Intelligibility module window or selecting *Open* from the *File* menu calls the *Open* file dialog box allowing you to select standard Windows audio waveform (*.wav) file, also called a "wave" file, to be opened for analysis. Note that the standard Windows dialog box that has been modified to show the sampling rate, resolution (8 - 24 bits/sample), type (stereo or monaural), and the length (in samples and seconds) of the file you select.

If a stereo file is loaded, only one channel is displayed. You can switch between channels using the *L* and *R* (left/right) channel selector buttons to the right of the file name and information fields or the *Channel* commands in the *View menu*.

Save As

File Menu > Save As

Keyboard Command = [Ctrl] + [S]

This command calls a modified Windows *Save As* file dialog box, allowing you to create a new wave (*.wav) file or a batch of files based on the wave file currently loaded in the Intelligibility module. The top portion of the dialog box has the normal areas for entering a file name and selecting the target directory for the new file. The lower portion of the *Save As* dialog box offers additional options.

On the lower right of the Intelligibility module's *Save As* dialog box is a pair of "radio buttons" to select whether you want the new file to include the entire time range of the original file or just the time range currently displayed on the main Intelligibility module plot. On the lower left is a list box labeled *Bands to Save* with a listing of all available filters. To the left of each entry in the *Bands to Save* list is a check box. You can select whether to save the unfiltered (Broadband) data, a filtered version of the original data, or some combination by checking and un-checking these boxes.

When you save the unaltered broadband wave file data, the name for the new file will be the file name entered on the *File Name* line in the upper portion of the dialog box. Filtered versions are named automatically using the specified *File Name* plus a single letter indicating the filter type ("o" for octave, "t" for 1/3-octave, or "u" for user) and the filter name (in parentheses). For example, if you entered "room.wav" in the *File Name* field and chose to save the broadband data plus filtered versions for the octaves centered on 250 Hz, 500 Hz, and 1 kHz (using the stock 1-octave bandpass filters), you would create the following files:

room.wav   (the unfiltered broadband data)

room_o(250 Hz).wav   (the original data filtered for the octave centered on 250 Hz)

room_o(500 Hz).wav   (the original data filtered for the octave centered on 500 Hz)

room_o(1kHz).wav   (the original data filtered for the octave centered on 1 kHz)

Analysis

File Menu > Analysis
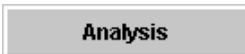
Analysis

This command opens the SIA-Smaart Acoustic Tools Analysis module, automatically loading the same wave file currently that is open in the Intelligibility module. Since neither program normally alters the wave file in any way, it is not a problem for the two programs to have the same file open at the same time. The Analysis module allows you to transform the wave file data into the frequency domain, using a series of Fast Fourier Transform () calculations, and display both time- and frequency characteristics several different ways. For more information about the Analysis module, please refer to the Analysis module section of your *SIA-Smaart Acoustic Tools User's Manual* or the Analysis module help file.

Record Impulse

File Menu > Record Impulse

    Rec

Keyboard Command = [Ctrl] + [R]

This command opens the *Record Impulse* dialog box where you will find controls for the Intelligibility module's impulse response recorder. The impulse response recorder allows you to measure the impulse response of the room/system under test using either a Maximum Length Sequence (MLS) or stimulus-independent dual Fast Fourier Transform (FFT) transfer function technique.

The *Record Impulse* dialog box has two pages or "tabs;" a Dual-FFT tab and an MLS tab. Each tab has it's own signal generator controls and input level meters. You will need to specify measurement parameters and an output file name for the impulse response data. The impulse response is written to a standard Window wave (*.wav) file. When you have selected your measurement parameters, click the record button. The intelligibility module will record and process data based on your selections, write the measurement results to the specified file name, and then close the dialog box and load the new file automatically. The measurement parameter controls on each tab are very similar but not identical.

The Dual FFT Tab

Recording an impulse response using the Dual FFT method is identical to the technique used to measure delays in the SIA SmaartLive so if you are familiar with SmaartLive, you probably already know how to use this feature. The interface is different but the measurement procedure is identical.

Pink Noise Generator — At the top of the Dual FFT tab are controls for the Intelligibility module's internal pink noise generator. This section contains a *Level* selector that sets the output gain in dB and a *Generate Pink Noise* button that starts and stops the signal generator. Use of the internal
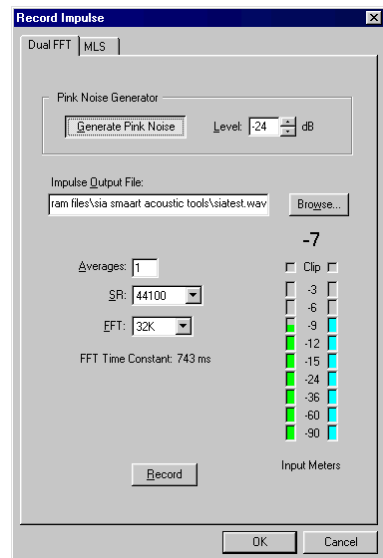
signal generator is optional as the Dual FFT recording method is stimulus-independent and works with virtually any type of continuous test signal including music.

Impulse Output File — This is where you specify the name of the wave file to be used to store the measurement results. You can type the path and file name for the output wave file in the text entry field or click the *Browse* button and use the *Open* file dialog box to navigate to the desired folder and specify a file name or select an existing file to overwrite.

Averages — This value determines the number of FFT frames the Intelligibility module will record. If a value greater than 1 is selected, the specified number of frames are recorded and then averaged together to calculate the impulse response. With every doubling of the number of the number of frames averaged, the signal-to-noise (S/N) ratio of the measurement will increase by 3 dB (the lower limit being the noise floor of the measurement system or the system under test, whichever is higher). The trade-off is that it takes longer to record and process the data for the measurement.

SR — Sets the sampling rate to be used to record the impulse response. This value determines the frequency content of the measurement. The highest frequency obtainable at a given sampling rate (the Nyquist frequency) will be equal to one half of the sampling rate. As a general rule, you will probably want to use a sampling rate of 44100 or greater to obtain a full-bandwidth impulse response. A lower rate might be preferable when measuring a subwoofer system, a band-limited system, or a room/system with a decay time significantly longer than five to six seconds.

FFT (size) — This value specifies the size (in samples) of the Fast Fourier Transform (FFT) used in the impulse response calculations. Together with the Sampling Rate, the FFT size determines the length of the FFT Time Constant or "time window" for the measurement as well as the size of the resulting wave file.

FFT Time Constant — The time window or length of time covered by an individual FFT Frame. This value is not directly editable as it is a function of the selected FFT size and Sampling Rate (FFT Size divided by Sampling Rate = FFT Time Constant). It is very critical that the *FFT Time Constant* be longer than the total decay time of the room/system under test.

The MLS Tab

The MLS recording method is a single-channel, stimulus-dependent measurement technique that compares the measured data to an MLS data sequence stored internally. The measurement parameters for MLS recording are substantially similar those for the Dual FFT Method but the MLS tab has a few additional options.

MLS Signal Generator — The use of the internal signal generator is again optional but in the case of MLS, it is the simplest way to ensure that the stimulus signal exactly matches the internal MLS data used in calculating the impulse response of the system under test. The *MLS Signal Generator* section includes a *Level* selector that sets the output gain in dB, a *Generate Channel(s)* selector used to set the output channel(s), and a *Generate MLS Signal* button that starts and stops the signal generator.

Impulse Output File — Specifies the name of the file to be used to store the measurement results.

Use Loopback For Time Alignment — Due to a limitation in Microsoft Windows, the Intelligibility module cannot, through software alone, determine the start time of the MLS stimulus signal accurately enough to provide the delay time through the system under test reliably. But since MLS is a single-channel technique, this limitation can be worked around by routing the signal from the unused computer audio output back to the unused input (see diagram on page 18) as a time reference. This check box should be checked when using an external loopback connection. When recording *without* a loopback signal, the initial peak in the impulse response will be positioned arbitrarily at 10 or 100 milliseconds, depending on the *MLS Length.*

Reference Channel — The input channel used to record the external loopback signal (if applicable). This parameter cannot be set directly. When the *Use Loopback For Time Alignment* box is checked, the *Reference Channel* is set automatically to the channel not used as the *Measurement Channel,* e.g., setting the *Measurement Channel* to *Left* automatically sets the *Reference Channel* to *Right.*
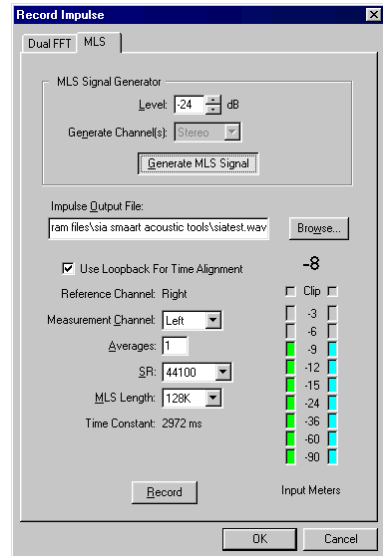
Measurement Channel — Sets the input channel to be used to record the output of the system under test.

Averages — Setting this value to a number greater than 1, causes the Intelligibility module to record and average the results of multiple MLS stimulus sequences to produce the finished measurement. Each doubling of the number of averages will yield 3 dB more signal to noise in the finished measurement (the lower limit being the noise floor of the measurement system or the system under test, whichever is higher) however the time required to make the measurement is increased.

SR — Sets the sampling rate to be used to record the impulse response measurement. Again, the highest frequency obtainable at a given sampling rate will be equal to one half of the sampling rate (the Nyquist frequency).

MLS Length — Like the *FFT* size parameter on the *Dual FFT* tab, this value specifies the size (in samples) of the MLS stimulus sequence. Together with the sampling rate, the *MLS Length* determines the size of the *Time Constant* or "time window" for the measurement as well as the size of the resulting wave file.

Time Constant — The time window for the impulse response measurement. This value is not directly editable but is a function of the selected FFT size and Sampling Rate (MLS Length divided by Sampling Rate = Time Constant). As with Dual FFT measurements, this value should always large compared to the total decay time of the room/system under test.

Create Impulse From File

File Menu > Create Impulse From File

Keyboard Command = [Ctrl] + [F]

The *Create Impulse From File* feature allows you to acquire the impulse response of a system from recorded data. You can use this feature to compare signals stored in two separate wave files or the Left and Right channels of a single stereo wave file. Possible uses for this feature might include:

• Applications where you need to acquire data without a computer

• Analyzing previously recorded data you already have on hand

• Situations where the amplifier room, or control room for a system is too far from the area you need to measure to return the measured signal to the computer

The *Create Impulse From File* function can use either the Dual-FFT method or MLS to calculate an impulse response. For the dual-FFT method the choice of test signals can be very flexible. For example, you could use a segment of a track from a music CD, a pink noise or other test signal track from a reference CD or a wave file sent through the system under test from a computer and then recorded remotely. The only real requirements are that the measurement file (or track) must be recorded using the same signal used to create the reference file and that the signal used has enough energy at the frequencies you need to measure. The reference file will be the "dry" signal as sent to the system under test. The measurement file will be the same signal, recorded after traveling through the system. When two separate files are used, they must also be recorded at the same sampling rate.

The MLS method uses only one wave file (the measurement file) and compares it to internally generated MLS data. This technique can yield very good results however there are a couple of important caveats that you should be aware of when using it. First, the MLS data used to record the measurement file must be identical in length to the internally generated sequence selected for comparison and generated using an identical group of seed numbers. Any two MLS stimulus signals of identical length, generated at the same sampling rate, using identical seed numbers should be identical. And in practice, many MLS generators do, in fact, use the same seed numbers however there is no accepted standard that says they must.

The second issue, when using the MLS technique, is that to obtain the best results, sampling rates should be consistent throughout the signal generation, output and recording processes. If you want to be absolutely safe on both counts, use the Intelligibility module's internal MLS signal generator (accessible through the MLS tab of the Record Impulse dialog box) to create your MLS stimulus signal. You can connect the computer directly to the system under test or record the internally generated signal to CD or DAT through a digital output. Be careful to note the sequence length and output the signal at a sampling rate supported by the CD or DAT recorder.

For best results, use a digital recording device to record the output of the system under test. If you plan to bring the measured signal back into the computer digitally, it should be recorded at the same sampling rate used to generate the MLS stimulus signal. If you will be bringing the signal back in through an analog input, it should be

recorded at the same sampling rate or higher. In either case, the measurement wave file should be recorded at the same sampling rate as the original stimulus signal.

If all this is starting to sound too complicated, don't worry. You can also obtain very good measurement results using MLS data with the dual-FFT technique. All you need is a wave file with the original MLS signal recorded at the same sampling rate and resolution as the measurement data file.

Intelligibility Graph

File Menu > Intelligibility Graph

Intelligibility

Keyboard Command = [Ctrl] + [I]

This command opens the Intelligibility dialog box where the results of all intelligibility calculations, the Intelligibility module supports are displayed. This dialog box is "mode-less," meaning you can leave it open while you continue to work in the main program window. Changes to the marker positions on the main plot are applied to the calculated values automatically (where applicable) as they are made.

The top half of the dialog box is the *Intelligibility Graph* where quantitative intelligibility values are plotted on three horizontal scales ranging from "Bad" to "Excellent" (left to right). Separate scales are provided for %ALCons, Clarity and STI.

Note that the results of two different forms of the %ALCons (long and short) calculations and two types of STI results may be plotted on these two scales. Also note that %ALCons will appear only after setting the Ld, Lr, and Ln markers in the 1-octave passband centered on 2 kHz ant that STI and RSTI values are calculated only when you click the *Calculate STI and RSTI* button. Once calculated, STI and RSTI values will not change for a given file.

The lower section of the Intelligibility dialog box has controls for some of the parameters that affect the intelligibility calculations and display fields for additional quantitative acoustical values. With the exception of *STI* and *RSTI* (if calculated) the acoustical values shown in this area apply only to the *Band* (passband) currently selected for display.

- **RT60** — Also called "reverberation time." RT60 is the amount of time it would take the system to decay 60 dB at the given rate of decay (RT60).

- ***Early Decay Time*** — The Early Decay Time (EDT) is rate at which the system under test completes earliest stage of its decay, again expressed as the amount of time (in seconds) that would be required for the signal to decay 60 decibels at the given rate of decay.

- ***Direct to Reverb*** — The direct-to-reverberant level (DTR). This is the relative difference between the level of direct sound to reverberant energy (at the top of the reverberant decay slope).

- ***Calculate STI and RSTI*** — Clicking this button will cause the Intelligibility module to calculate STI and RSTI from the impulse response data. These calculations can take some time to process, particularly on long files, but this only needs to be done once for any file. The STI and RSTI calculations are ***not*** dependent on either the positions of the level (Ld, Lr, and Ln) markers or the passband currently selected for display.

- ***Clarity —*** The Clarity field sets the number of milliseconds used for the Clarity Factor scale in the upper portion of the window. For example, entering 35 in this field will set the clarity scale to C35.

- ***STI*** and ***RSTI*** — When the STI and RSTI values are calculated for the current file, the results of the calculations are displayed in these two fields and on the *STI* scale in the upper portion of the dialog box. Again, the STI and RSTI values are ***not*** dependent on either the positions of the level (Ld, Lr, and Ln) markers or the passband currently selected for display.

- ***Band*** — The *Band* selector serves the same function as the *Band* spinner to the right of the plot in the main program window. It selects the passband — either the unfiltered *Broadband* data or a 1- or 1/3-octave passband — displayed on the impulse response plot. The currently in use determines the available passbands. The filter set is selected using the *Filter Set* spinner to the right of the plot in the main window.

- ***Save*** — This button opens a *Save As* file dialog box allowing you to save all of the intelligibility and acoustical values above to a standard ASCII text file suitable for import into a spreadsheet, word processor, or any other application that accepts ASCII files. If you check the *Save All Bands* table in the *Save As* dialog, the Intelligibility will automatically create a separate text file for each passband in the currently-selected filter set.

All Bands Table

File Menu > All Bands Table

> **All Bands Table**

Keyboard Command = [Ctrl] + [B]

This command opens the *All Bands Table* dialog box. This window lists quantitative acoustical and intelligibility values for the broadband data and multiple octave or 1/3-octave (depending on the filter set currently in use) passbands. The quantities listed in this table are all based on some combination of the level (Ld, Lr, and Ln) marker positions so only bands that have actually been displayed for the current file during the current Intelligibility module session will be included in the table. Bands are added to the *All Bands Table* automatically as you change the passband for the main display and set or confirm the marker positions.

The *All Bands Table* lists (from left to right) the *RT60*, Early Decay Time *(EDT)*, Direct-to-Reverberant level *(DTR)*, *%ALCons* short *(S)* and long *(L)* forms (for the 2 kHz octave band only), the user-selected Clarity factor (C*n*) and additional Clarity values for 10, 20, 50 and 80 milliseconds (C10, C20, C50 and C80). Note that the *Intelligibility Graph* can show you most of these same values (with the exception of the last four Clarity columns) for a single passband. The *All Bands Table* provides an easy way to compare these values for different passbands.

Below the text table are three buttons labeled *Print, Save and Close.* Clicking the *Print* button will, by default, take you to the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before printing. If you have un-checked *"Show this dialog before printing"* either in the *Custom Print Information* dialog box or the *Printing* tab of the *Options* dialog box, the *Print* button will take you directly to the *Print* dialog box.

The *Save* button opens a *Save As* file dialog box to allow you to dump the information in the table to a standard ASCII text file suitable for import into a spreadsheet, word processor or other application. Note that the standard Windows file dialog box has been modified to allow you to enter a text *Comment* to be included in the output file. The *Close* button closes the *All Bands Table* dialog box.

Set All Values to Default

File Menu > Set All Values to Default

The user-definable options for the Intelligibility module are very flexible and can sometimes be confusing, especially at first. The Intelligibility module stores the state of nearly all of these options when you exit the program so if a problem does arise, chances are good that it will still be there the next time you start the program.

Any time you wish to restore the Intelligibility module to its "factory" default settings, select *Set All Values to Default* from the *File* menu. This will reset all virtually user-definable program parameters to their original default states. The only exceptions are that the *Color Scheme* and device selections are left as set on the Colors and Devices tabs of the *Options* dialog box.

Print

File Menu > Print

Keyboard Command = [Ctrl] + [P]

The *Print* command first takes you to the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before printing. The *Custom Print Information* dialog box can be used to set all the same printing options as the Printing tab of the Options dialog box. After setting title and page options, click the *OK* button in the *Custom Print Information* dialog. A standard Windows *Print* dialog box will then open to allow you to select a printer and set up your printer options before sending the document to print.

Note: If you un-check either the *"Show custom print dialog before print and print preview"* check box on the *Printing* tab of the *Options* dialog box or the check box labeled *"Show this dialog before print and print preview"* in the *Custom Print Information* dialog box, both the *Print* and *Print Preview* commands (see next page) will bypass the *Custom Print Information* dialog box and take you directly to the *Print* dialog box.

Print Preview

File Menu > Print Preview

The *Print Preview* command first takes you to the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before proceeding. When you click the *OK* button in the *Custom Print Information* dialog box, the Intelligibility goes to Print Preview mode to show you "what-you-see-is-what-you-get" (WYSIWYG) preview of how the printed page should look. You can then use the *Print* button to send the document to the currently selected printer immediately and exit *Print Preview* mode or select *Close* to return to the normal Intelligibility module program window without printing the document.

Print Setup

File Menu > Print Setup

The *Print Setup* command opens a standard Windows *Print Setup* dialog box where you can select a printer and set up the page size, orientation, and paper source for hard copy output. Depending on the type of printer selected, there may be additional options you can access by clicking the *Properties* button in the *Print Setup* dialog.

View Menu Commands

Zoom

Zoom Out To Limits

View Menu > Zoom > Zoom Out To Limits

Keyboard Command = [Home]

This command returns the time (*x*-axis) range of the main plot to the full time range of the wave file and resets the amplitude (*y*-axis) range to the default range as specified on the Graph tab of the Options dialog box. A shortcut for this command is to simply click once with your mouse anywhere outside the plot in the margins of the plot area.

Undo / Redo Zoom

View Menu > Zoom > Undo / Redo Zoom

Keyboard Commands:
Undo Zoom = [Ctrl] + [Z]
Undo Zoom = [Ctrl] + [A]

As you zoom in and out on the main plot, the Intelligibility module remembers each zoom view and allows you to browse back and forth through the previous views using these two commands.

Zoom X In / Out

View Menu > Zoom > Zoom X In / Out

Keyboard Commands:
Zoom X In = [Ctrl] + [Down Arrow]
Zoom X Out = [Ctrl] + [Up Arrow]

These two commands increase or decrease the horizontal (*x*-axis) "magnification" by decreasing the displayed time range of the main plot. Note that you can also zoom in on both the *x* and *y* axes of the plot and zoom out using your mouse. For more information on this feature, please refer to the topic titled *Zooming With the Mouse* in the *Shortcuts* section.

Move Range Left / Right

View Menu > Zoom > Move Range Left / Right

Keyboard Commands:
Move Range Left = [Ctrl] + [Left Arrow]
Move Range Right = [Ctrl] + [Right Arrow]

When you are zoomed in on the horizontal (*x*-axis) range of the main plot, only a portion of the wave file's full time range is displayed. These two commands move the displayed range left or right to allow you to view the information before or after the range currently displayed.

Zoom Y In / Out

View Menu > Zoom > Zoom Y In / Out

Keyboard Commands:
Zoom Y In = [–]
Zoom Y Out = [+]

These two commands increase or decrease the vertical (*y*-axis) "magnification" by decreasing the displayed amplitude range of the main plot. Note that you can also zoom in on both the *x* and *y* axes of the plot and zoom out using your mouse. For more information on this feature, please refer to the topic titled *Zooming In and Out on the Plot With the Mouse* in the *Shortcuts* section.

Move Range Up / Down

View Menu > Zoom > Move Range Up / Down

Keyboard Commands:
Move Range Up = [Page Up]
Move Range Down = [Page Down]

When you are zoomed in on the vertical (*y*-axis) range of the main plot, only a portion of the wave file's full amplitude range is displayed. These two commands move the displayed range up or down to allow you to view the information above or below the range currently displayed.

Amplitude

View Menu > Amplitude

> **Amplitude Scale**
> Log (dB)

The *Amplitude Scale* commands (below) are used to set the type of scaling used on the amplitude (y)-axis of the main Intelligibility module plot. Amplitude scaling for the impulse response plot can also be selected using the *Amplitude Scale* spinner to the right of the plot in the main program window.

Linear Scale

Keyboard Command = [Ctrl] + [N]

Displays the (A/D) amplitude values on a linear scale.

Decibel Scale

Keyboard Command = [Ctrl] + [D]

Converts (linear) A/D amplitude values to decibel values.

Show

Schroeder Trace

View Menu > Show > Schroeder Trace

> **Schroeder**

Keyboard Command = [S]

In addition to plotting the raw time-domain data of the impulse response, the Intelligibility module can display a second trace on the plot that is the result of a Schroeder Reverse Time Integration calculation performed using the impulse response data. The Schroeder trace can be useful for identifying trends in the impulse response, particularly the decay slope, and can be helpful in determining reverberation time (RT60).

Schroeder Trace is a "toggle" command. Clicking the *Schroeder* button (shown above) or selecting *Schroeder Trace* from the *Show* sub-menu of the *View* menu displays the Schroeder trace. When the Schroeder is present, clicking the same button or selecting the same menu command again will remove the trace from the plot.

Schroeder On Top

View Menu > Show > Schroeder On Top

Keyboard Command = [Ctrl] + [H]

When the Schroeder trace is displayed on the Intelligibility module plot, selecting *Schroeder On Top* from the *Show* sub-menu of the *View* menu moves the Schroeder trace to the "top" of the plot's *z*-axis stacking order. The *z* order of the plot is of concern in this case mainly because the precision tracking cursor always tracks the top trace when *Track Nearest Data Point* is turned on.

Schroeder On Top is a "toggle" command. Selecting the same menu command again will return the impulse response trace to the top of the *z* order.

ETC

View Menu > Show > ETC

Keyboard Command = [Ctrl] + [E]

This command calculates and displays the Energy Time Curve (ETC) based on the (broadband or band-limited) time-domain impulse response data. When *ETC* is selected, the *ETC* plot replaces the normal impulse response plot in the Plot Area of the main Intelligibility module program window. The *ETC* display mode can be useful in identifying reflections and looking at the decay characteristics of a system under test. Although the *ETC* plot does resemble the impulse response plot displayed on a decibel scale, it can make discreet reflections, in particular, somewhat more prominent and easier to see.

Note that a number of program features are disabled in *ETC* display mode. For example, the ETC cannot be used as the basis for the intelligibility calculations and linear amplitude scaling is not applicable to this type of display. *ETC* is a "toggle" command. Clicking the same button or selecting the same menu command again will restore the standard impulse response plot.

Marker (Ld, Lr, Ln)

View Menu > Marker (Ld, Lr, Ln)

The *Marker (Ld, Lr, Ln)* sub-menu commands allow you to select and position the level markers using keyboard or menu commands. Note that you can also select markers by simply clicking them with your mouse and move the markers around on the plot by clicking and dragging them (moving the mouse while holding down the left mouse button) from one place to another. Or to "snap" the selected marker to the mouse cursor location, hold down the [Alt] and click the left mouse button once.

Select Next Marker

Keyboard Command = [Tab]

Cycles marker selection through the Ld, Lr, and Ln markers.

Move Marker Left

Keyboard Command = [Left Arrow]

Moves the selected *(Ld, Lr, or Ln)* marker to the left.

Move Marker Right

Keyboard Command = [Right Arrow]

Moves the selected *(Ld, Lr, or Ln)* marker to the right.

Move Marker Up

Keyboard Command = [Left Arrow]

Moves the selected *(Ld, Lr, or Ln)* marker upward (in amplitude).

Move Marker Down

Keyboard Command = [Down Arrow]

Moves the selected *(Ld, Lr, or Ln)* marker downward (in amplitude).

Set Ld to Peak

Peak

Keyboard Command = [Shift] + [P]

Resets the Level direct (Ld) marker to the highest peak in the impulse response.

Track Nearest Data Point

View Menu > Track Nearest Data Point

The Intelligibility module features both a standard (arrow) mouse cursor and a precision tracking cursor. The relative time and amplitude position of the precision cursor is continuously displayed on a readout above the plot. The arrow cursor can always be moved freely around the plot and can be used to position the Level markers. When Track Nearest Data Point is turned on, the precision cursor "snaps" to the nearest data point on the "top" trace in the *z*-axis stacking order while following the horizontal movement of your mouse. When this feature is turned off, the tracking cursor freely follows the mouse cursor.

The *z*-axis stacking order is determined by the selected display options for the Schroeder reverse time integration curve. If the Schroeder trace is not displayed, the precision cursor tracks the impulse response trace by default. When the Schroeder trace is present, the precision cursor will track the Schroeder trace if *Schroeder On Top* is selected (see page 44).

Octave Band Filters and 1/3 Octave Band Filters

View Menu > Octave Band Filters and 1/3 Octave Band Filters

These two commands are used to select the filter set used for band-limited impulse response and ETC displays. The selected filter set determines the choices available on the *Bands* spinner and the *Bands* list box in the *Intelligibility* dialog box.

Channel

View Menu > Channel

When a stereo wave file is loaded, only one of the two channels is displayed. You will be asked to select which channel you want to display when the file is initially opened but if you want to change your selection later the *Left* and *Right* commands in the *Channel* submenu may be used to select either channel for display at any time.

Options Menu Commands

All

Options Menu > All

Options

Keyboard Command = [Alt] + [O]

This command opens the *Options* dialog box with the last tab used on top. Any tab may be selected any time the *Options* dialog box is open.

Color

Options Menu > Color

The SIA-Smaart Intelligibility module allows you to customize the colors of virtually everything on the screen and even use your own bitmap files as a background. Color and background options are loaded as sets called *Color Schemes.* Several ready-to-use color schemes (including two with "metallic" bitmapped backgrounds) are included with the program and you can easily define your own. The *Colors* tab of the Options dialog box allows you to create, edit and manage *Color Schemes* for the Intelligibility module program window and displays.

The Colors tab of the *Options* dialog box is used to create and manage color schemes for the Intelligibility Module. The Color Schemes section of the Colors tab has a list of all available Color Schemes along with buttons for creating, editing, and deleting user-defined schemes. A pair of list boxes below the Color Schemes section allow you to select separate color schemes for display and printing purposes.

Color Schemes

New...

Clicking the *New* button calls the *Edit Color Scheme* dialog box, allowing you to create a *new* color scheme. In this dialog box, you can change colors for various display elements and optionally, a Windows bitmap (*.bmp) file to use as a background for the program Window. Color selections initially shown in the dialog box for each element are based on the *Color Scheme* currently in selected in the list. Clicking any of the solid-colored squares opens a standard Windows Color dialog box allowing you to choose a standard color or specify a custom color to be used for the selected element.

Edit..

The *Edit* button calls the *Edit Color Scheme* dialog box to allow you to edit an existing (user-defined) color scheme selected in the list. Note that the stock color schemes supplied with the Intelligibility module may be used as the basis for a new user-defined color scheme but cannot be edited or deleted themselves.

Delete

Clicking the *Delete* button deletes a (user-defined) *Color Scheme* selected in the list.

Below the *Color Scheme* section are two list boxes

- Display Color Scheme — Selects an existing Color Scheme to be used for display purposes (the colors shown on your computer screen).

- Printing Color Scheme — Selects an existing Color Scheme for use when printing from the Intelligibility module.

Devices

Options Menu > Devices

If your computer has more than one sound card installed, the *Devices* tab of the *Options* dialog box allows you to select which of these the Intelligibility module should use. This tab is also where you set the sampling resolution (bits per sample) for each device. The *Devices* options tab has for drop-down list fields:

- ***Wave In*** — Selects the sound device to be used for audio input (recording).

- ***Wave Out*** — Selects the computer sound device to be used for output from the internal pink noise and MLS signal generators.

- ***Wave In Bits Per Sample*** — Sets the sampling resolution for the selected audio input (recording) device.

- ***Wave Out Bits Per Sample*** — Sets the sampling resolution for the selected audio output device. This setting also determines the output resolution of the internal signal generators.

At the bottom of the *Devices* tab is a check box labeled *"Warn before recording if the signal generator is not running."* Un-checking this check box disables the warning message normally given if you click the *Record* button on either tab of the Record Impulse dialog box when the internal signal generator is not running.

Graph

Keyboard Command = [Alt] + [G]

The *Graph* tab of the *Options* dialog box has the following options for controlling the appearance of the main plot and how the Intelligibility module displays data:

Title — At the top of the *Graph* options tab is a text input field used to set the title line displayed above plot in the main program window and above the Intelligibility Graph in the Intelligibility dialog box.

Amplitude Range — The *Amplitude Range* section has four text fields for setting the upper (Max) and lower (Min) limits of both the *Current* and *Default* amplitude (*y*-axis) ranges of the main plot. Note that the *Current* amplitude range can also be set using the Zoom Y In/Out and Move Range Up/Down commands or by using your mouse to zoom in and out. The *Default* range is the range displayed when you open a new wave file or when you use the Zoom Out To Limits command. The options available in the *Amplitude Range* section will depend on whether a *Linear* or *Decibel Amplitude Scale* is currently selected as the range may be set independently for each scaling mode.

In *Decibel* scaling mode, the *Min* and *Max* amplitude range values are set in decibels (dB). When calculating decibel amplitude values, the Intelligibility module regards "full-scale" (the highest possible amplitude value obtainable from the wave file) as zero dB and all other values are given as "dB down" from 0. The method used to calculate dB values from the wave file's linear amplitude units results in a maximum possible decibel range of 0 to approximately −210 so the maximum *displayable* range for the plot is +10 dB to −220 dB.

In *Linear* amplitude scaling mode, the amplitude range of the plot is set as *percentages* of the maximum possible positive and negative amplitude values obtainable from the wave file. The default range is normally set to +/− 100%.

Speed of Sound — The Intelligibility module uses the *Speed of Sound* setting to calculate distance equivalents for time values. The speed of sound value can be specified in either feet or meters, depending on which of the "radio buttons," is selected. The unit of measure selected here also determines whether distance equivalents are displayed in feet or meters elsewhere in the program. The default speed of sound value is 1120 feet per second or 341.359 meters per second.

Time Range — The controls in *Time Range* section set the current *x*-axis range for the main plot in milliseconds. The default time range is always the full length of the wave file and the time scale is always linear. Note that the current displayed time range for the plot can also be set using the Zoom X In/Out and Move Range Left/Right commands or by using your mouse to zoom in and out.

Save Filter Data in Memory — When this box is checked, filtered versions of the wave file data are held in memory after they are calculated so that you can switch between previously calculated passband views almost instantly. This is particularly useful when you are looking at the lowest (octave or 1/3-octave) passbands where the calculation time for filtering can be significant even on a very fast machine. You will probably want to leave this box checked unless your computer has very limited system resources available and you need to minimize the amount of memory the Intelligibility module requires.

Clarity — This value, specified in milliseconds, determines the Clarity Factor displayed in the Intelligibility Graph and in the All Bands Table. For example, if this value is set to 50 milliseconds (the default setting), the Intelligibility module will use C50 as the Clarity Factor in intelligibility displays. Changing this number to 7 or 35 will change the Clarity Factor to C7 or C35 respectively.

Meter

Options Menu > Meter

The settings on the *Meters* tab of the *Options* dialog box control the appearance and behavior of the *Input Level Meters* in the Record Impulse dialog box. The meters are calibrated to regard the maximum digital output of the A/D system of the sound card being used (normally 16-24 bits) as 0 dB with all other values given as "dB down" from zero.

The following meter options are user-definable:

- ***LED Levels*** — The Intelligibility module allows you to specify amplitude values for each individual (Left/Right) pair of "LEDs."

- *Peak Hold* — When this box is checked, the meters will continue to indicate the highest amplitude level encountered in the input signal for a specified period of time after the level drops off.

- *Hold Peak For* — Specifies the length of time (in seconds) the meters will hold peak levels when *Peak Hold* (see above) is enabled.

- *Averages* — Setting this value to a number higher than one will cause the meters to display a running average of the specified number of samples rather than simply reacting as fast as it can to changes in the input signals. This will tend to stabilize the meter reading, which may be desirable given the random, or pseudo-random stimulus signals typically used in recording impulse response measurements. Note that this increased stability comes at the cost of degrading the meter ballistics.

Meter LED Colors

The Input Level Meters are designed to display three colors, the normal "LED" color (Light Green for the Left input and Light Blue for Right) plus Yellow and Red. The meter "LEDs" will display input levels in the normal LED color up to the amplitude threshold specified for Yellow, then Yellow up to the threshold for Red, and Red above that level.

- *Yellow if above* — Sets the threshold amplitude at which the LEDs change to Yellow.

- *Red if above* — Sets the threshold amplitude above which the LED color changes to Red.

Numeric Readout — The numeric value that appears above the Input Meters in the Record Impulse response can be set to track either the current signal level *(Show Current Value)* or the peak value *(Show Peak)* by selecting the appropriate "radio button" in this section. Note that the *Show Peak* option for the numeric readout may be used with or without the *Hold Peak* option (see above) enabled for the meters themselves.

Printing

Options Menu > Printing

The Printing tab of the Options dialog box controls the appearance and content of pages printed directly from the Intelligibility module. The following options are user-definable:

- **Title (1, 2, and 3)** — You can specify up to three lines of title text to appear at the top Intelligibility module printouts. If any of the three *Title* fields is left blank, that line is omitted on the printed page.

- **Print Type** — The two radio buttons in this section determine what type of information will be printed. When the *Impulse and Intelligibility Graphs* option is selected, the printed page will include both the impulse response plot and the Intelligibility Graph. If the *All Bands Table* option is selected, only the All Bands Table is included on the printout.

- **Print File Name** — When this box is checked, the name of the wave file currently loaded in the Intelligibility module (on which the printed information is based) will be included in the title area at the top of the printed page.

- **Print Today's Date** — If this box is checked, the current date (as set on your computer) will be inserted in the title area at the top of the printout. If you want to use a date other the current system date of your computer on the printout, un-check this box and type the desired date as one of the *Title* lines.

- **Show custom print dialog before print and print preview** — When this box is checked, selecting either the Print or Print Preview commands will open the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before proceeding with the selected operation. The *Custom Print Information* dialog box includes all of the same options as the *Printing* tab of the *Options* dialog box. If you un-check this check box, the Intelligibility will bypass the *Custom Print Information* dialog box in *Print* and *Print Preview* operations and execute the selected command immediately.

Intelligibility Module Keyboard Shortcuts

The following is a list of the Intelligibility keyboard commands (hot keys). Note that all of these commands are also available as menu commands and in some cases, as on-screen buttons or controls.

Key names on the list below are enclosed in [brackets] to differentiate them from descriptive text. Some commands use more than one keystroke, for example the accelerator key sequence to show/hide the Schroeder Trace is "[Ctrl] + [S]." This means you must hold down the *Control* key while you press the *S* key.

Display

Decibel Scale = [Ctrl] + [D]

Linear Scale = [Ctrl] + [N]

Schroeder Trace = [S]

Schroeder Trace On Top = [Ctrl] + [H]

ETC = [Ctrl] + [E]

Intelligibility Graph = [Ctrl] + [I]

All Bands Table = [Ctrl] + [B]

File

Open File = [Ctrl] + [O]

Save As = [Ctrl] + [S]

Record Impulse = [Ctrl] + [R]

Create Impulse From File = [Ctrl] + [F]

Print = [Ctrl] + [P]

Marker (Ld, Lr, Ln)

Move Selected Level (Ld, Lr, Ln) Marker = [Arrow] Keys (Up, Down, Left, Right)

Select Next Level Marker = [Tab]

Set Ld Marker To Peak = [Shift] + [P]

Options

All Options = [Alt] + [O]

Graph Options = [Alt] + [G]

Zoom and Range

Undo Zoom = [Ctrl] + [Z]

Redo Zoom = [Ctrl] + [A]

Zoom Out To Limits = [Home]

Zoom X In = [Ctrl] + [Down Arrow]

Zoom X Out = [Ctrl] + [Up Arrow]

Move Range Left = [Ctrl] + [Left Arrow]

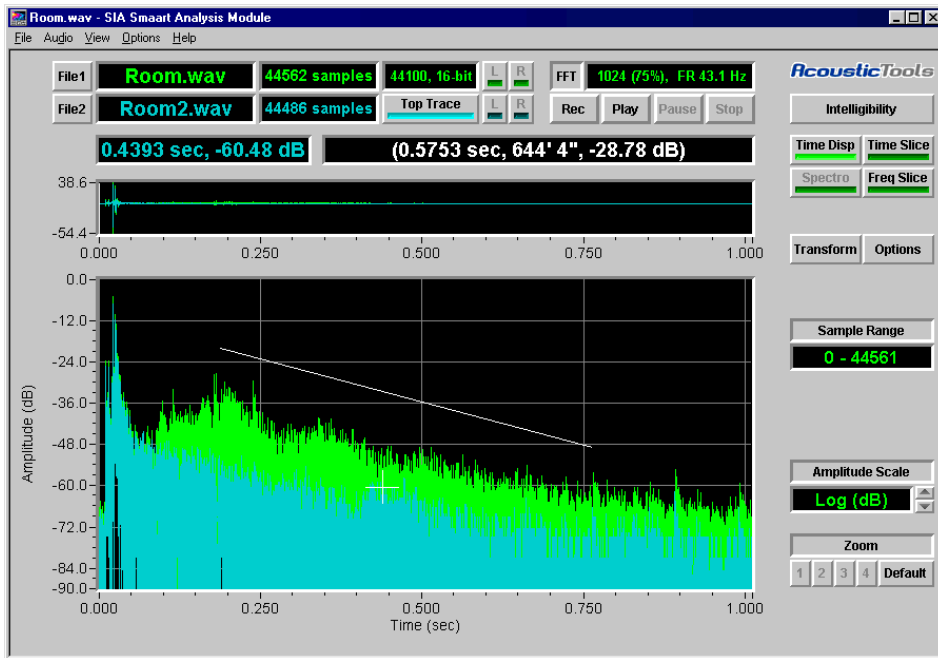Move Range Right = [Ctrl] + [Right Arrow]

Zoom Y In = [+]

Zoom Y Out = [–]

Move Range Up = [Page Up]

Move Range Right = [Page Up]

Chapter 3: Analysis Module Functions and Commands

Navigating in the Analysis Module

Like the Intelligibility module, the Acoustic Tools Analysis has a completely redesigned user interface that speeds access to the most frequently used program functions. Again, simply the clicking the label or readout field for an on-screen control or readout will allow you to access nearly any associated parameter or option you are likely to need for all but the most advanced features. Readouts have been enlarged and where appropriate, the text is color coded to match the associated trace color. This helps make it easier to understand at a glace exactly what you are seeing.

The following is a brief description of the basic elements that make up the Analysis module program window. Detailed information on Analysis module displays and functions can be found in *Analysis Module Functions,* beginning on page 59.

The Menu Bar

Virtually all of the program functions that are available as on-screen controls and/or keyboard commands are also accessible through the pull-down menus on the *Menu Bar.* Menu commands also provide access to many less commonly used features.

To activate a pull-down menu from the Menu Bar, click on its name using your mouse or hold down the [Alt] key while pressing the key corresponding to the underscored letter in the menu name. When a menu is open, commands may be selected by clicking on their names with your mouse or by typing the underscored letter.

Menu commands followed by ellipses (three periods...) call dialog boxes. An arrow-head at the right of a menu item indicates that this command activates a nested "fly-out" submenu. For explanations of Analysis module menu commands, please refer to *Analysis Module Commands* beginning on page 68.

File 1 Information

This section of the Analysis module window provides information about the primary wave file loaded in the Analysis module. The (Open) *File 1* button calls an *Open* file dialog box allowing you to select a wave file to load. The next field to the right displays the file name of the wave file after opening it.

Immediately to the right of the file name field are two fields that display the number of samples in the file and the sampling rate and resolution (bits/sample) at which the data was recorded. Clicking on either of these two fields with your mouse will open the *File Information* dialog box (see page 69) with additional details about the file(s) currently loaded. When a stereo wave file is loaded, the *(L* and *R)* channel buttons select which channel is displayed.

FFT Information

The FFT information field immediately to the right of the *File 1* information fields provides details about the frequency domain data calculated for the current wave file(s) by the Analysis module's *Transform* function (see page 71). The values

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displayed here include the FFT size (in samples) the overlap percentage, and the frequency resolution of the FFT. Clicking anywhere in this field calls the *Transform* dialog box to allow you to transform or re-transform the wave file data for the Analysis module's frequency-based displays.

File 2 Information

This section provides information about a *second* wave file loaded in the Analysis module. The (Open) *File 2* button calls an *Open* file dialog box allowing you to select a wave file to load. The next field to the right displays the file name of the wave file after opening it. Clicking this field with your mouse pops up a menu that will allow you to close the second wave file or select another file to open as File 2.

Immediately to the right of the File 2 file name field is a field that displays the number of samples in the file (the sampling rate and bits/sample for File 2 must be identical to File 1). Clicking on this field with your mouse will open the *File Information* dialog box (see page 69) with additional details about both files currently loaded. The *Top Trace* button cycles the *z*-axis stacking order in multiple trace displays. If File 2 is a stereo wave file, the *(L* and *R)* Channel buttons select which channel is displayed.

Audio Controls

The *Audio* controls provide access to the Analysis module's *Record* function and the means to start, pause and stop audio playback of an open wave file. When two files are loaded the file currently displayed it front will be played. For more information, see *Audio Menu Commands,* beginning on page 75.

Tracking Cursor Readout

This field gives you x and y axis (along with z in Spectrograph mode) coordinates for the Analysis module's precision tracking cursor when the mouse cursor is positioned over the main plot area or the Waveform display. Coordinates are given in seconds, dB, amplitude percentage or Hertz, as appropriate for the display type.

Relative Cursor Readout

(0.5753 sec, 644' 4", -28.78 dB)

In *Time Display* and *Time Slice* modes, you can use your mouse cursor find the time and amplitude/magnitude difference between two points in the main display by holding down the left mouse button and drawing a line on the plot. The line you draw remains visible on the plot when you release the mouse button, and the difference between the line's end points are displayed in this field. In *Time Slice* mode (only), the decay slope in decibels per second and equivalent RT60 value are also calculated and displayed here. See *Delta/Decay Rate Calculation* on page 102 for more details.

The Waveform Display
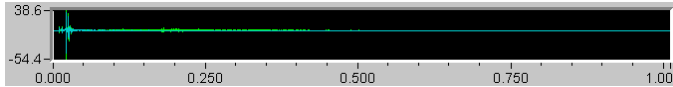
The *Waveform Display,* located above the *Plot Area,* is a simple time-domain representation of the waveform (*.wav) file(s) open in the Analysis module. This display serves as a visual locator for the data being displayed main Plot Area. In *Time Display, Time Slice,* and *Spectrograph* modes, locator bars in the Waveform Display indicate the boundaries of the currently-displayed time range. In *Frequency Slice* mode, the Waveform Display indicates the location of the currently-displayed FFT frame.

The amplitude (y-axis) scale of the Waveform display is specified as a percentage of the full (digital) amplitude scale of the wave file. For example, when a 16-bit file is loaded, 100% represents a range of $+/- 32767$ A/D units. The amplitude scale of the waveform display is set from the *Waveform* tab of the *Options* dialog box (see page 99).

The Plot Area

The *Plot Area* is the Analysis module's main display area for data plots in all display modes. Both the Waveform display (see above) and the Plot Area will appear blank until a wave (*.wav) file is loaded. When a new file is loaded, a time-domain (time vs. energy) representation of the wave file data appears in the Waveform Display and a more detailed version (the *Time Display*) is shown in the Plot Area.

The Analysis module requires the waveform file's time-domain data to be transformed into the frequency domain using the *Transform* feature (see page 71) before any of the

other display modes are enabled. Once the wave file has been transformed and the companion frequency-domain (*.sia) file created, you can select any of the four primary display modes.For more information about Analysis module display modes, please refer to *Analysis Module Functions* beginning on page 59.

Jump to Other Programs

Clicking the *Intelligibility* button in any display mode will automatically close the Analysis module while opening the Intelligibility module and loading the file that was open as File 1 (in the Analysis module).

Clicking the Acoustic Tools logo above the Intelligibility button will pop up a menu that allows you to jump to, and pass data to other programs if there are jumps defined it wrjumps.txt. See *Setting Up Links to Other Programs* on page 26 or more information about this feature.

Display Modes

The *Display Mode* buttons switch the Analysis module between its four primary display modes. For information about Analysis module display modes, see *Analysis Module Functions* beginning on page 59.

Mode-Specific Controls

The controls in the area will below the *Display Mode* buttons (see above) will vary based on the primary display mode currently selected. The *Transform* and *Options* buttons are always visible along with the *Rainbow* and *Power Spectrum* buttons in *Time Slice* and *Frequency Slice* modes respectively. Below these buttons you will see range and scale and/or banding controls appropriate for the primary display currently selected. See *Analysis Module Functions,* beginning on page 59, for more information about Analysis module display modes.

Analysis Module Functions

The Time Display

The *Time Display* is a detailed view of the wave file's time vs. amplitude characteristics. The Time Display is the only display mode available prior to transforming a wave file for the first time (using the *Transform* command).

The default time (x-axis) scale shows the entire waveform. You can zoom to a specific portion of the waveform using the arrow keys, the *Time Zoom* commands in the View menu, or by clicking and dragging (horizontally) the with the mouse in the Waveform Display. The Locator Bars in the Waveform Display indicate the limits of the time range currently being displayed on the main plot.

The vertical (y-axis) *range* of the plot can be changed using the *Amplitude > Zoom* and *Move* commands in the View menu or specified on the Time Display tab of the Options dialog box. The amplitude *scaling* options for the Time Display plot are:

Log (dB) — Logarithmic amplitude scaling takes the linear A/D amplitude data and calculates a decibel value (relative to 32767) for each amplitude unit (except 0).

Lin (%) — Linear amplitude scaling represents amplitude linearly as a percentage of the full scale of the wave file (based on bits per sample used to record the data).

Absolute (%) — Absolute Value Scaling uses a linear amplitude scale, however the absolute value of each data point is plotted (all negative values are converted to positive). As with Linear scaling, amplitude values are given as a percentage of the full scale of the wave file.
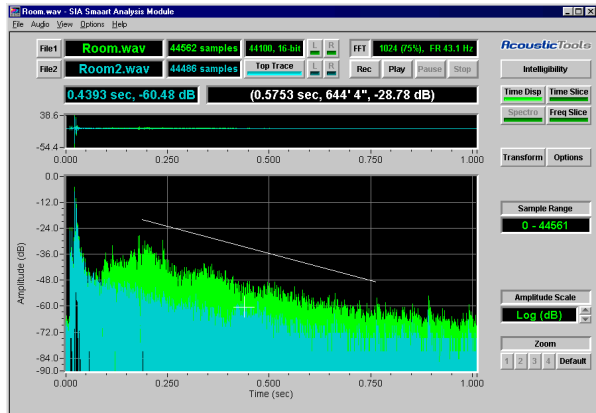
(0.5753 sec, 644' 4", -28.78 dB)

In Time Display mode you can use your mouse to find time and amplitude difference between any two points on the plot. Just drag the mouse (hold down the left mouse button while moving) across the plot to draw a line between the two points. When you release the mouse button, the difference between the end points of the line is displayed on the slope readout (shown above) located directly above and to the right of the Waveform Display. To remove the line, click once on the plot. The units used in the Time Display cursor and slope readouts are user-selectable. You can choose to display any combination of time (in seconds), samples, feet, or meters by selecting the appropriate options on the *Time Display* tab of the options dialog box.

All distance values in the Analysis module are calculated from the *Speed Of Sound* value, also set from *Time Display* options. The default Speed Of Sound value is 1120 feet (341.35 meters) per second.

The Time Slice

The *Time Slice* display is similar to the Time Display in that it plots *time* on the x-axis vs. *magnitude* on the y-axis however there are a couple of important differences. First, the time slice shows you data for only a narrow frequency range. Bandwidth options available for this display mode are *Narrowband, Octave* or *1/3-Octave*. The other big difference may be less obvious but in Time Slice mode, you are actually looking at FFT data instead of the raw, time-domain audio data shown in the Time Display. This is why the Time Slice trace appears much smoother than the band-limited time domain displays in the Intelligibility module. Time domain displays plot one data point for each *sample* in the wave file whereas on the Time Slice plot, there is one data point for each *FFT frame.*

The time resolution of the Time Slice plot as well as the bandwidth of the *Narrowband* Time Slice display is determined by the frequency resolution of the FFT size selected when the wave file is *transformed*.

Band

500.0 Hz

The current center frequency and bandwidth are shown on the *Band* and *Banding* spinners that appear to the Right of the plot in Time Slice mode. The center frequency the Time Slice plot is selected using the *Band* spinner or the *Next Slice* and *Previous Slice* commands under *Slice* in the *View* menu.

Rainbow

The *Rainbow* display simultaneously plots the normal Time Slice and two sidebands. By default, the sidebands are 1 or 1/3 octave up and down from the center frequency depending on the selected bandwidth (the Narrow Band option is not available in the Rainbow display). Specific sideband frequencies can be specified on the Rainbow tab of the *Options* dialog box. The three traces can be Normalized, or aligned in amplitude at a specific time location holding down the [Shift] key while clicking on the plot with the left mouse button. Pressing the [N] key toggles normalization on and off.

The Time Slice can be very useful for determining decay rates in a room impulse response. Because it tends to be much smoother than simple time domain displays, overall trends in the impulse response are easy to see. You can draw a line on the Time Slice plot by holding down your left mouse button while moving the mouse cursor between the two points on the plot. When you release the mouse button, the decay rate in dB/sec. and equivalent RT60 values are displayed in the relative cursor readout immediately above the plot to the right of the standard cursor readout.

The Frequency Slice

Freq Slice

The Frequency Slice shows magnitude (y-axis) versus frequency (x-axis) for a given frame — in effect, a still-frame spectrum analyzer. The FFT data may be displayed in Linear, Logarithmic, Octave, or 1/3-Octave frequency scaling. The highest magnitude value found in all the

FFT frames calculated for the wave file is regarded as 0 dB. All other magnitude values on the Frequency Slice plot are given relative to this 0 dB point.

In Frequency Slice mode you can "step through" each of the FFT frames calculated for the file using the *Next Slice* and *Previous Slice* commands under *Slice* in the *View* menu ([SpaceBar] or [Shift] + [SpaceBar] on your keyboard) or using the *Frame* spinner that appears to the right of the plot in Frequency Slice mode. Also, because each FFT frame represents a different time location, the (Time Zoom) pan controls (the Left and Right arrow keys) can be used in this mode to step through "slices."
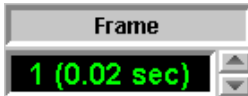
Frame locator bars in the provide a visual reference for the time position of the data from which the currently displayed FFT was calculated. Selecting *Go To Slice,* under *Slice* in the *View* menu or clicking the *Frame* spinner displays the *Go To* dialog box, allowing you to specify the frame you want to display by time location or FFT frame number. You can also select a time location in Frequency Slice mode by clicking on the trace in the *Waveform Display* with your left mouse button.

The frequency (x-axis) scaling of the Frequency Slice plot is set by the Frequency Scale spinner that appears ti the right of the plot in Frequency Slice mode. The range and scale of the plot's vertical *(y)* axis may be changed using the *Amplitude > Zoom* and *Move* commands in the *View* menu. The amplitude range for this display can also be set from the *Frequency Slice* section of the *Slice* tab in the *Options* dialog box. The label above the plot shows the current resolution (narrowband, 1-octave, or 1/3-octave) the frame number, and the start time of the frame currently displayed.

The *Power Spectrum* is a special type of Frequency Slice display. The Power Spectrum is a plot of the average of all, or a selected range of FFT frames calculated for a wave file. The power spectrum display is activated by pressing the *Power Spectrum* button that appears to the right of the plot in Frequency Slice mode.
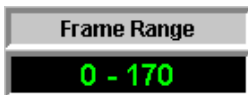
When *Power Spectrum* is on, you can also set the time range over which power spectrum is calculated by clicking the *Frame Range* readout to the right of the plot and entering a range of FFT frames in the pop-up dialog. The time range over which the Power Spectrum is calculated can also be specified using time coordinates or FFT frame numbers on the *Power Spectrum* tab of the Options dialog box.

The Spectrograph

Spectro

The *Spectrograph* is a "topo-graphic" representation of 3-dimensional data, with *time* along the x-axis, *frequency* along the y-axis, and *amplitude* represented by color. This display is similar in concept to the so-called "waterfall" display, however we feel the Spectrograph display is much easier to read and interpret than a waterfall plot once you become accustomed to it.

By default, the all *frequency-based* displays in the Analysis module regard the highest magnitude value actually *encountered* in the wave file as 0 dB. The magnitude value for every other frequency data point is calculated relative to this 0 dB point. Selecting *Full-Scale* on the *Calibration* tab of the *Options* dialog box re-scales the Spectrograph to regard the maximum magnitude *obtainable* (given the wave file's sampling resolution) as 0 dB. The *Full-Scale* and *Custom* calibration options are particularly useful when comparing multiple wave files.

A legend area below the plot shows the number of colors used and the *Dynamic Range* of the Spectrograph. The value of dB/color is displayed to the right of the color scale bar. The magnitude range represented by each color used is determined by the number of colors used and the specified dynamic range. These parameters are set from the *Spectrograph* tab of the *Options* dialog box. The Spectrograph may be displayed in up to 236 colors.

Banding

Log (NB)

The frequency range of the Spectrograph plot's y-axis is set using the four user-configurable Frequency Zooms. The frequency (y-axis) *scale* can be linear or
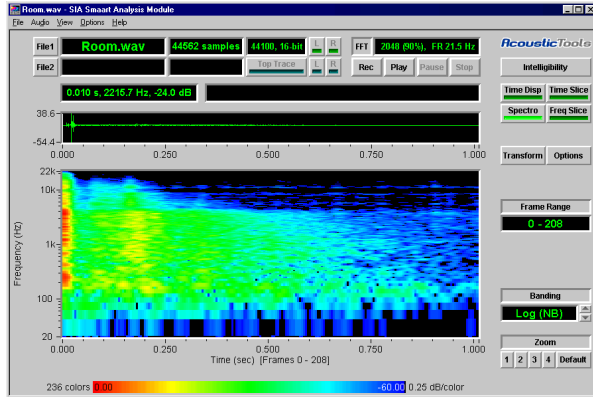
logarithmic. Frequency scaling may be selected using the *Banding* Spinner that appears to the right of the plot in Spectrograph mode or on the *Spectrograph* tab of the *Options* dialog box.

Frame Range
0 - 170
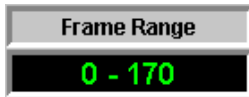
The Spectrograph plot is actually made up of vertical bars, with each bar representing one FFT frame. The vertical bars are divided into sections representing individual FFT data points with the magnitude for each point indicated by color. By default, all FFTs calculated for a wave file are displayed to show the entire time range of the file along the *x* axis. You can select a specific time range using the Time Zoom commands, by selecting a time range in the *Waveform* display, or by clicking the Frame Range readout and entering a range of FFT frames in the pop-up dialog box.

Two options are available on the *Spectrograph* tab of the Options dialog box that alter the way the individual sections of the vertical bars are plotted. The option used in the illustration above is the default *Uniform* setting. This option displays the *Spectrograph* as a continuous color map. Selecting *Relative* varies the (horizontal) width for each section relative to its magnitude value (higher magnitudes are wider).

Comparing Two Wave Files

The Analysis module allows you to open two files at the same time and compare their time- and frequency-domain data in *Time Display, Time Slice,* and *Frequency Slice* display modes. This feature is useful for assessing the effectiveness of acoustical treatment in a room or any other application that requires comparing two signals. The Analysis module can also create a new wave file that represents the difference between the two files.

File1 **Room.wav** **44562 samples** **44100, 16-bit** **L** **R**

To load a second file into the Analysis module, first open File 1 as you normally would the click the File 2 button above and to the left of the plot. This button calls a Windows Open file dialog box prompting you to select a wave file. The second wave file must be recorded at the same sampling rate and resolution (8-24 bits/sample) as File 1 and must be of the same type (stereo or mono) but does not have to be exactly the same size.

After you have made your selection, the Analysis module will automatically switch to Time Display mode and plot a signal trace for each of the two files in different colors (light green for File 1 and light blue for File 2 if the default Color Scheme is used). When a second file is loaded, it is displayed "on top" of File 1 in the *z*-axis stacking order by default. In other words, the *File 2* trace will cover the *File 1* trace, both in the *Waveform* display and on the *Time Display* plot in the main plot area, anywhere the two traces overlap. To bring the *File 1* trace to the top, click the *Top Trace* button. Clicking this button again brings *File 2* back to the top and so on.

When stereo wave files are loaded, the Analysis module displays only one channel for each file. The Channel (L/R) selector buttons to the right of the *File 1* and *File 2* information fields at the top left of the Analysis module window are used to select which Channel is displayed. There are separate channel selector buttons for each file so you can select either channel of either file for display independently. Note that by loading the same stereo wave file as both File 1 and File 2, you can use this feature to view both channels of the file simultaneously.

Important Note: The file displayed on top is always the object of all commands that act only on one file at a time such as the *Save Wave Zoom As,* and *ASCII Save* commands. Also, the tracking cursor always tracks along the top trace when *Track Data* is enabled.

Adjusting the File 2 Time Offset

When two files are loaded, the maximum displayable time range will be equal to the length of File 1. If File 2 is longer, its time range is truncated. To adjust the time position of File 2, for example to align peaks in the two traces, use the keyboard commands below. These commands are available only in Time Display mode.

Fine Adjustment

- Hold down the [Ctrl] key while pressing the [Left Arrow] to move the File 2 trace one pixel to the left.

- [Ctrl] + [Right Arrow] moves the File 2 trace one pixel to the right.

Coarse Adjustment

- [Ctrl] + [Shift] + [Left Arrow] moves File 2 left in larger increments.

- [Ctrl] + [Shift] + [Right Arrow] moves File 2 right in larger increments.

Clear File 2 Time Offset

- [Ctrl] + [R] removes all time offset.

Note: You can also specify the File 2 offset in samples in the *File Information* dialog box. To open the *File Information* dialog, select *File Info* from the *File* menu or click either of the two fields immediately to the right of the file name readout for *File 1* where the number of samples, sampling rate and resolution for the file are displayed.

Transforming File 2

FFT 1024 (75%), FR 43.1 Hz

Before the Analysis module can display frequency information for the two wave files, both must be transformed using identical transform parameters. When two files are loaded, transforming *File 1* will automatically transform *File 2* using the same parameters and the frequency-domain information for File 2 is stored in a temporary file. If File 1 has already been transformed, selecting *Time Slice* or *Frequency Slice* mode automatically transforms *File 2* (only) using the same parameters as were previously used file for File 1. Note that the *Spectrograph* display mode, and the *Rainbow* display in *Time Slice* mode are not available when two files are loaded.

Analysis Module Commands

File Menu Commands

Open File 1

File Menu > Open File 1

Keyboard Command = [Ctrl] + [O]

This command calls the *Open* file dialog box allowing you to select a standard Windows audio waveform (*.wav) file — also called a "wave" file — to be opened for analysis. This dialog box is modified to show the sampling rate, resolution (8- to 24-bit), type (stereo or monaural), and the length, in samples and seconds, of the file you select. When a wave file is opened, the *Waveform Display* and *Time Display* plots appear. If a stereo file is loaded, only one channel is displayed. You can switch between channels using the *L* and *R* (left/right) channel selector buttons.

Close File 1

File Menu > Close File 1

Closes the wave file. If two files are loaded, this command closes both.

Open File 2

File Menu > Open File 2

Keyboard Command = [Ctrl] + [Shift] + [O]

This command calls an Open file dialog box allowing you to select a second wave file to be loaded for comparison with the first. File 2 must be of the same type (monaural or stereo) sampling rate and resolution (bits per sample) as File 1. The data from the second file is plotted in a different trace color.

By default, the second file is displayed "on top" of File 1 in the *z*-axis stacking order. In other words, the File 2 trace will cover the File 1 trace, both in the Waveform Display and on the main Time Display plot, anywhere the two traces overlap. To bring the File 1 trace back to the top, click the *Top Trace* button. Clicking this button again brings *File 2*

back to the top and so on. The file displayed on top is the object of all commands that act only on one file at a time such as the Channel selection (for stereo files), *Save Wave Zoom As,* and *ASCII Save* commands. Also, the precision cursor always tracks along the top trace when *Track Data* is enabled.

Note: By loading the same stereo wave file as both File 1 and File 2, you can use this feature to view both channels of the file simultaneously.

Close File 2

File Menu > Close File 2

Keyboard Command = [Ctrl] + [Shift] + [L]

Closes File 2. File 1 remains open.

File Info

File Menu > File Info

Opens the *File Information* dialog box. This dialog box gives you the following information about the file(s) currently open in the Analysis Module:

- Sampling rate
- Sampling Resolution (8- to 24-bit)
- File type (stereo or mono)
- The name of each file
- The length of each file in samples and seconds
- The text comment in the *.sia file (created when the wave file is transformed), if applicable

When two files are loaded the *Shift* field displays the time offset, in samples, for File 2. You can adjust the offset by typing in a new value in this field. The time offset for File 2 can also be changed using menu, keyboard, or mouse commands. Please refer to *File 2* (View menu commands) on page 84 for more information.

Intelligibility

File Menu > Intelligibility

This command opens the SIA-Smaart Intelligibility module, automatically loading the same wave file currently that is open in the Analysis Module. Since neither program normally alters the wave file in any way, it is not a problem for the two programs to have the same file open at the same time. For more information about the Intelligibility module, please refer to Chapter 2 of this manual, beginning on page 6.

Save Wave Zoom As

File Menu > Save Wave Zoom As

This command allows you to save just the displayed portion of the time range of a wave file loaded in the Analysis module as a new wave file. After selecting a portion of an open file you wish to save using the *Time Zoom* commands (see page 78), select this command to call the *Save As* file dialog box. In the dialog box, select (or create) the destination folder then enter a file name and click the *Save* button. The new wave file is immediately loaded into the Analysis module after saving. The original wave file is not altered in any way.

Note: If two files are loaded, this command applies only to the one displayed in front.

Save Difference As

File Menu > Save Difference As

With two files loaded, this feature allows you to create a new wave file that represents the *difference* between the two. *Save Difference As* subtracts the amplitude value of the file displayed in front, sample by sample, from the one in back. The *Change "Z" Order* command can be used to control which file gets subtracted from which. After completing its calculations, the Analysis module opens a you will be prompted to select a destination folder and enter a file name for the new wave file. The *new* file is immediately loaded into the Analysis module after saving. The *original* files are not altered.

ASCII Save

File Menu > Save Difference As

In Time Display and Frequency Slice modes, the *ASCII Save* command creates a standard ASCII text file from the data displayed on the plot. ASCII output files have a header with the creation date and information about the wave file followed by time- or frequency-domain data (depending on the display mode in use) in numeric form:

- ASCII files created in Time Display mode have a single column of amplitude values for each displayed time-domain sample.
- ASCII files created in Frequency Slice display mode are formatted as a five-column table with information about each data point in the FFT frame currently displayed.

After converting the data to ASCII format, the Analysis Module opens a *Save As* file dialog box. In the dialog box, select a destination folder and enter a name for the text file. You can also enter comment text to be included in the file header if you like.

Notes:

When two files are loaded, only the one currently displayed in front is exported.

In Time Display mode, the *Time Zoom* feature can be used to select just a portion of the wave file for export. If you are "zoomed in" on the time axis, only the samples within the displayed time range are exported.

Transform

File Menu > Transform

Keyboard Command = [Ctrl] + [T]

The Analysis module requires the *time-domain* data contained in waveform files to be *transformed* into the *frequency domain* before many of it's features can be accessed. The Analysis module calculates frequency-domain data for the waveform file using a series of Fast Fourier Transform (FFT) calculations. The *Transform* command calls the *Transform Waveform File* dialog box, prompting you to select *FFT Size, Overlap,* and *Data Window* parameters and enter a text *Comment* if you wish.
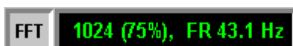
Selecting Transform Parameters

Selecting an appropriate FFT size, Overlap and Data Window can sometimes require some trial and error. The FFT size is the number of time-domain samples to include in each FFT. The combination of the FFT size selected and the wave file's sampling rate determine the frequency resolution of the FFTs (the number of Hertz between each FFT data point).

The frequency resolution for the currently-selected FFT size is shown to the right of the *Size* selector in the FFT Parameters section of the dialog box (FR = *n* Hz). Larger FFT sizes yield better frequency resolution, but can sometimes hide important changes in the data, due to the longer amount of time encompassed by the FFT.

Overlapping acts to smooth the data, often providing the *Spectrograph* with clearer information. Note that overlapping data increases the number of FFT calculations and therefore, the time required to create and display the frequency domain data. *Overlap* is specified as the percentage of data each FFT shares with the previous one.

Data Windows are used to reduce "edge effects" associated with the finite length of FFTs. The Analysis Module provides several different Data Window options. If you don't have another preference, the (default) Hanning or Hamming window types generally work well. The option labeled "None" (also called a "rectangular" data window) is typically not used.

After transformation, the FFT size and Overlap, and FFT frequency resolution (in Hertz) are displayed in the FFT information field at the top of the Analysis module window to right of the fields that display the number of samples, resolution and sampling rate for File 1. The frequency-domain data derived from the wave file data is stored in a separate file having the same name as the source wave file except that the file name extension ".sia" is used. This process is nondestructive to the wave file data and may be repeated as many times as you like. The original wave file is not altered in any way.

The **.sia* file is created in the same folder (directory) as the source wave file. As long as these two files are kept together in the same folder, the Analysis module will automatically find and load the existing *.sia file when the *.wav file is loaded again.

Configuration

File Menu > Configuration

The *Configuration* commands are used to import and export configuration files. Configuration data for the Analysis module is stored in the Windows registry. The program's registry stores the state of nearly all user-definable parameters.

Export

This command is used to extract a copy of the entire Analysis module registry from the Windows registry (registration database) to a (*.reg) file on disk. Selecting *Export* opens a standard Windows file *Open* dialog box allowing you to specify a file name and destination folder for the new file. This feature is useful for backup purposes or moving your preferences to a new computer.

Import

The *Import* command opens a standard Windows *Open* file dialog box, allowing you to import a previously-saved registry (*.reg) file into the Windows registry. This replaces all registry entries for the Analysis module with the information in the registry file.

Print

File Menu > Print

Keyboard Command $=$ [Ctrl] $+$[P]

The *Print* command first takes you to the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before printing. The *Custom Print Information* dialog box can be used to set all the same printing options as the *Printing* tab of the *Options* dialog box. After setting title and page options, click the *OK* button in the *Custom Print Information* dialog. A standard Windows *Print* dialog box will then open to allow you to select a printer and set up your printer options before sending the document to print.

Note: If you un-check the *"Show custom print dialog before print and print preview"* check box on the *Printing* tab of the *Options* dialog box or the check box labeled *"Show this dialog before print and print preview"* in the *Custom Print Information* dialog box, both the *Print* and *Print Preview* commands (see next page) will bypass the *Custom Print Information* dialog box and take you directly to the *Print* dialog box.

Print Preview

File Menu > Print Preview

The *Print Preview* command first takes you to the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before proceeding. When you click the *OK* button in the *Custom Print Information* dialog box, the Analysis goes to Print Preview mode to show you "what-you-see-is-what-you-get" (WYSIWYG) preview of how the printed page should look. You can then use the *Print* button to send the document to the currently selected printer immediately and exit *Print Preview* mode or select *Close* to return to the normal Intelligibility module program window without printing the document.

Print Setup

File Menu > Print Setup

The *Print Setup* command opens a standard Windows *Print Setup* dialog box where you can select a printer and set up the page size, orientation, and paper source for hard copy output. Depending on the type of printer selected, there may be additional options you can access by clicking the *Properties* button in the *Print Setup* dialog.

Exit

File Menu > Exit

Keyboard Command = [Alt] + [F4]

Ends your SIA-Smaart Analysis module session and returns you to the Windows desktop.

Audio Menu Commands

| Rec | Play | Pause | Stop |

Record

Audio Menu > Record

The *Record* command opens the *Record Wave File* dialog box. This dialog box is a simple wave file recording control panel. To record a new wave file on your computer's hard disk:

- Enter a name for the new file in the *Output File* field. The Browse button calls a standard Windows Open file dialog box, allowing you to select the destination folder (directory) for the file.

- Enable the Stereo "check box" if you want to record a stereo wave file.

- Check the level of the signal(s) coming into the sound card inputs on the input level meters — the right channel meter is only active if the Stereo check box is enabled. If the clip indicators light, this indicates that the input signal is overloading the sound card inputs and should be adjusted before recording.

- Select the sampling rate. This determines frequency content of the wave file. 44.1 k or 48 k is required for full-bandwidth audio. 22050 is sufficient for speech.

- Select the "Bits per sample" (the digital resolution). 16-bit resolution is recommended for most applications.

- Click the *Record* button. The counter begins running and the *Record* button changes to a *Done* button.

The length of the recording is limited only by the amount of disk space available. When you are finished recording, click the *Done* button to close the dialog box and immediately load the new wave file into the Analysis module. To exit the dialog box without loading the recorded file, click *Cancel.*

Play

Audio Menu > Play

Keyboard Command = [Shift] + [P]

The *Play* command begins audio playback of a wave file open in the Analysis module. If two files are loaded, the one displayed on top is played. If you are "zoomed in" on the time axis, only the displayed portion of the file is played. During play, a progress indicator shows the approximate time position of the data being played.

Pause

Audio Menu > Pause

Select this command to pause the audio playback. Repeat the command (toggle off) to resume the playback at the point at which it was paused.

Stop

Audio Menu > Stop

Stops audio playback.

View Menu Commands

Mode

The *Mode* commands switch the Analysis module between its four primary display modes:

Time Display

View Menu > Time Display

Time Disp

Keyboard Command = [D]

The *Time Display* plots a detailed view of the time-domain (time vs. amplitude) data in a wave file. The Time Display is the only display mode that can be accessed before transforming a wave file. For more information, please refer to *The Time Display* on page 59.

Time Slice

View Menu > Time Slice

Time Slice

Keyboard Command = [T]

Time Slice mode plots magnitude vs. time *for a given frequency range* (Narrow Band, Octave, or 1/3-Octave). You could think of this plot as a *horizontal* slice of the Spectrograph. The Time Slice display is particularly useful for looking at impulse response data. For more information, please refer to *The Time Slice* on page 60.

Frequency Slice

View Menu > Frequency Slice

Freq Slice

Keyboard Command = [F]

The *Frequency Slice* displays a magnitude vs. frequency plot of a selected FFT frame ("slice") — in effect, a still-frame spectrum analyzer. You could also think of this display as a (horizontal) plot of a *vertical* slice of the Spectrograph. For more information, refer to *The Frequency Slice* on page 62.

Spectrograph

View Menu > Spectrograph
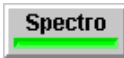
Spectro

Keyboard Command = [S]

The *Spectrograph* is a topographical representation of the traditional "waterfall" display. This "three-dimensional" plot puts time on the x-axis, frequency on the y-axis, and represents amplitude using color. For more information, please refer to *The Spectrograph* on page 64.

Channel

View Menu > Channel

L   R

When a stereo wave file is open in the Analysis module, the *Channel* commands select which channel (Left or Right) is displayed. If two stereo files are loaded, the displayed channel for each file is selected independently — the Channel commands apply to the wave file displayed *on top*. To view both channels of a stereo wave file simultaneously, open the same file as both File 1 and File 2 then select the Right channel on one copy and the Left channel on the other.

Time Zoom

Move Right / Left

View Menu > Time Zoom > Move Right / Left

Keyboard Commands:
Move Right = [Right Arrow]
Move Left = [Left Arrow]

When you are zoomed in on the horizontal (*x*-axis) range of the main plot, only a portion of the wave file's full time range is displayed. These two commands move the displayed range left or right to allow you to view the information before or after the range currently displayed.

Zoom In / Out

View Menu > Time Zoom > Zoom In / Out

Keyboard Commands:
Zoom In = [Ctrl] + [Down Arrow]
Zoom Out = [Ctrl] + [Up Arrow]

These two commands increase or decrease the horizontal (*x*-axis) "magnification" of the Time Display, Time Slice and Spectrograph plots by decreasing the displayed time range of the main plot. Note that when the time axis scale is changed, the boundaries of the currently-displayed time range are indicated by range locator bars in the *Waveform* display.
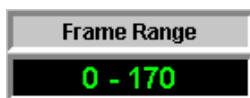
In *Time Display* mode the precise sample range displayed in the *Sample Range* readout to the right of the plot. In *Time Slice* and *Spectrograph* modes the *Sample Range* readout is replaced by a *Frame Range* readout that shows the *FFT frame* range currently displayed. Clicking either of these readouts with your mouse will pop up a dialog box allowing you to specify a time range in samples or FFT frame numbers (depending on the display mode).

Mouse Shortcut: The time range for the *Time Display, Time Slice* and *Spectrograph* display modes can also be set by clicking and dragging (horizontally) with the mouse on the *Waveform* display. See page 100 for more information.

Amplitude

Zoom In / Out

View Menu > Amplitude > Zoom In / Out

Keyboard Commands:
Zoom Out = [+]
Zoom In = [–]

These two commands increase or decrease the vertical (*y*-axis) "magnification" of the *Time Display, Time Slice,* and *Frequency Slice* plots by decreasing the displayed amplitude/magnitude range.

Move Up / Down

View Menu > Amplitude > Move Up / Down

Keyboard Commands:
Move Up = [Page Up]
Move Down = [Page Down]

When you are zoomed in on the vertical (*y*-axis) range of the main plot, only a portion of the wave file's full amplitude range is displayed. These two commands move the displayed range up or down to allow you to view the information above or below the range currently displayed.

<u>Note</u>: To reset the range of any Analysis module plot to the program defaults, use the *Default Zoom* command (see page 87).

Linear Scale

View Menu > Amplitude > Linear Scale

Keyboard Command = [Ctrl] + [L]

Sets the vertical *(y)* axis of the *Time Display* plot (only) to *Linear* amplitude scaling. When *Linear* scaling selected amplitude values are given as a percentage of full scale based on sampling resolution (bits per sample) at which the wave file was recorded.

Absolute Value Scale

View Menu > Amplitude > Linear Scale

Keyboard Command = [Ctrl] + [V]

Displays the A/D amplitude values on a linear scale with all negative values converted to positive. Amplitude values are given as a percentage of full scale based on sampling resolution (bits per sample) at which the wave file was recorded.

Decibel Scale

View Menu > Amplitude > Linear Scale

Keyboard Command = [Ctrl] + [D]

Sets the vertical *(y)* axis of the *Time Display* plot (only) to logarithmic amplitude scaling. A/D amplitude units are converted to decibel values.

Slice
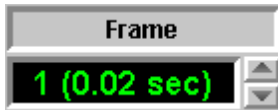
The *Slice* commands are used to navigate between FFT frames and frequency ranges in *Frequency Slice* and *Time Slice* modes (respectively). Note that you can also navigate between bands/frames using the *Band* spinner that appears to the right of the main plot in *Time Slice* mode or the *Frame* spinner that replaces the Band spinner *Frequency Slice* mode.

Next Slice

View Menu > Amplitude > Slice

Keyboard Command = [Space Bar]

"Moves" the display to the next FFT frame or *Time Slice* (depending on the current display mode).

Previous Slice

View Menu > Amplitude > Slice

Keyboard Command = [Shift]+ [Space Bar]

"Moves" the display to the previous FFT frame or *Time Slice* (depending on the currently selected display type).

Go to Slice

View Menu > Amplitude > Slice

Keyboard Command = [Ctrl]+ [G]

In Frequency Slice mode, this command opens a dialog box prompting you to select an FFT frame for display by time or frame number.

Mouse Shortcut: You can select an FFT frame in *Frequency Slice* mode by placing the mouse cursor on a point in the Waveform Display and clicking the left mouse button.

Loop Through Slices

View Menu > Amplitude > Slice

The *Loop through Slices* command displays all of the FFT frames calculated for the wave file sequentially, returning to the beginning of the file when it reaches the end. Repeating the *Loop through Slices* command toggles the feature off.

Frequency

Zoom (1, 2, 3 and 4)

View Menu > Frequency > Zoom (1, 2, 3 and 4)

Keyboard Commands = [1], [2], [3], or [4]

The numbered *Frequency Zoom* commands are used to select from the four preset frequency *ranges* for the *Frequency Slice* and *Spectrograph* display modes. The range of each *Zoom* is user definable and is set from the *Zooms* tab of the Options dialog box. The frequency *scale* commands below set the type of frequency scaling used on *Frequency Slice* and *Spectrograph* plots and as the *resolution* (bandwidth) of the *Time Slice* display.

Narrowband

View Menu > Frequency > Narrowband

Keyboard Command = [5]

Sets the *Time Slice* display to *Narrowband* (single FFT bin) resolution.

Linear Narrowband

View Menu > Frequency > Linear Narrowband

Keyboard Command = [6]

Plots the frequency axis of the *Frequency Slice* and *Spectrograph* displays in Linear frequency scaling.

Logarithmic Narrowband

View Menu > Frequency > Logarithmic Narrowband

Keyboard Command = [7]

Plots the frequency axis of the *Frequency Slice* and *Spectrograph* displays using Logarithmic frequency scaling.

Octave

View Menu > Frequency > Octave

Keyboard Command = [8]

Changes the *Frequency Slice* and *Time Slice* displays to octave band resolution.

1/3-Octave

View Menu > Frequency > 1/3-Octave

Keyboard Command = [9]

Changes the *Frequency Slice* and *Time Slice* displays to 1/3-octave band resolution.

<u>Note:</u> If octave bands in the Frequency Slice display appear "ghosted" at low frequencies, it means the frequency resolution of the FFT is insufficient for the data in those bands to be completely reliable. To increase frequency resolution, increase the FFT size when transforming the wave file.

Mask

The *Mask* feature is useful when analyzing signals known to be contaminated by persistent extraneous noise sources such as fluorescent lights, electric motors, etc. Masking is applied in Frequency Slice mode but the effects are also visible on the Spectrograph plot when Masking is active. See page 103 for more information on setting up frequency masks in the Analysis module.

Use Mask

View Menu > Mask > Use Mask

Keyboard Command = [Ctrl] + [M]

To apply a mask, switch to Frequency Slice mode, select Narrowband resolution, and select *Use Mask* to activate Masking. With the mouse, click and drag from left to right to draw a horizontal line across the frequency range you wish to mask at the highest magnitude you want to allow for those frequencies. When you release the mouse button you will see the effect of the mask. Repeat this process for any other frequencies you wish to mask.

After a mask has been created, you can add more frequencies at any time using the same procedure. Repeating the *Use Mask* command toggles masking on and off. The mask is kept in memory until you clear it (see below) or exit the Analysis module.

Clear Mask

View Menu > Mask > Clear Mask

Keyboard Command = [Ctrl] + [Shift] + [M]

Clears all masking information from memory, allowing you to create an entirely new mask (rather than adding to an existing mask).

File 2

The *File 2* commands control the relative time offset for File 2 when two wave files are open in the Analysis module. The *Move* (time offset) commands above are enabled in *Time Display mode only.* Note that File 2 time offset may also be specified (in samples) in the File Information dialog box.

Move Left

View Menu > File 2 > Move Left

Keyboard Command = [Ctrl] + [Left Arrow]

Fine (negative) time offset adjustment (moves the File 2 trace one pixel to the *left*).

Move Right

View Menu > File 2 > Move Right

Keyboard Command = [Ctrl] + [Right Arrow]

Fine (positive) time offset adjustment (moves the File 2 trace one pixel to the *right*).

Move Left Fast

View Menu > File 2 > Move Left Fast

Keyboard Command = [Ctrl] + [Shift] + [Left Arrow]

Coarse (negative) time offset adjustment for the File 2 trace.

Move Right Fast

View Menu > File 2 > Move Right Fast

Keyboard Command = [Ctrl] + [Shift] + [Right Arrow]

Coarse (positive) time offset adjustment for the File 2 trace.

Reset to No Shift

View Menu > File 2 > Reset to No Shift

Keyboard Command = [Ctrl] + [R]

Clears all time offset adjustment.

Cursor

Track Data

View Menu > Cursor >Track Data

With this option selected, the tracking cursor "snaps" to the top trace in the "Z" order (see below), moving from data point to data point along the trace as you move the mouse from left to right.

Move Left / Right

View Menu > Cursor > Move Left / Right

Keyboard Commands:
Move Left = [Alt] + [Left Arrow]
Move Right = [Alt] + [Right Arrow]

When the Track Data (see above) is enabled, the (Cursor) *Move Left* and *Move Right* c*o*mmands can be used to move the precision tracking cursor to the right or left by one pixel or one sample or data point (depending on the display mode and magnification).

Set FFT Center to Peak

View Menu > Cursor > Set FFT Center to Peak

Keyboard Command = [Ctrl] + [Alt] + [P]

The *Set FFT Center to Peak* command sets the center point for the first FFT frame to be calculated by the Transform routine to the highest peak found in the wave file. A marker will appear on the plot and in the Waveform Display to indicate the location of the specified center point. The marker is initially a red and white striped line that changes to a solid red line after the file has been re-transformed.

Normally, the Analysis module's *Transform* routine calculates the FFT series used for frequency-domain displays beginning with the first sample in the wave file. One potential problem with this is that the peak in the impulse response may fall at a point in the FFT frame where it is attenuated by the *Data Window* function used in the *Transform* routine.

All the data windows provided in the Analysis module attenuate the time-domain data closest to the beginning and end of each FFT frame to the largest extent but leave the samples closest to the center of the frame untouched for the most part. Setting the

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center point for the first FFT in the transform series on the initial peak in the impulse, which normally represents the arrival of direct sound, ensures that the peak data is altered as little as possible by the data window in the first FFT frame.

Move FFT Center Left

View Menu > Cursor > Move FFT Center Left

Keyboard Command = [Ctrl] + [Alt] + [Left Arrow]

Moves the selected FFT center point to the left by one pixel or one data point depending on the display "magnification." If you are zoomed in on the time (x) axis to the extent that individual data points are more than one pixel apart, the center point will move one data point to the left.

Move FFT Center Right

View Menu > Cursor > Move FFT Center Right

Keyboard Command = [Ctrl] + [Alt] + [Left Arrow]

Moves the selected FFT center point to the left by one pixel or one data point depending on the display "magnification."

Clear FFT Center

View Menu > Cursor > Move FFT Center Right

Keyboard Command = [Ctrl] + [X]

Clears the FFT center point selection from the Time Display and resets the default starting point for the Transform routine the first sample in the wave file.

Setting the FFT Center Point Using Your Mouse

You can also set/move the center point for the first FFT frame to any point on the Time Display plot by holding down the [Ctrl] key while clicking on the plot with your mouse.

Notes:

There must be enough samples between the beginning of the file and the selected FFT center point for the first half of the FFT. If the selected center point would cause the first FFT to "overrun" the beginning of the wave file, the program will default back to beginning the first FFT on the first sample in the wave file when the file is transformed. The FFT center point can only be set/changed in Time Display mode.

Change "Z" Order

View Menu > Change "Z" Order

Keyboard Command $=$ [Ctrl] $+$ [F]

Cycles through the (z-axis) "stacking order" of all displayed traces. In other words, when multiple traces are displayed, repeating this command will bring each trace in turn to the "top of the stack." This is useful for making the trace that most interests you easier to see, but it is also how you assign which trace the precision cursor tracks. When tracking is enabled, the precision always tracks the trace that is displayed on top.

Default Zoom

View Menu > Default Zoom

Keyboard Command $=$ [Shift] $+$ [Up Arrow]

Returns the both the X and Y axes of the plot to the default settings for the current display mode (as defined on the *Time Display, Slice,* or *Spectrograph* tabs of the *Options* dialog box).

Rainbow

View Menu > Rainbow

Keyboard Command $=$ [R]

In Time Slice mode, pressing the *Rainbow* button displays three *Time Slice* traces simultaneously; the "slice" for the selected center frequency plus two sidebands. By default the sidebands are one octave or 1/3-octave up and down from the center frequency depending on the resolution selected (Narrowband resolution is not available for the Rainbow display). Specific (octave) center frequencies for all three traces may be selected on the Rainbow tab of the Options dialog box.

Normalized

View Menu > Normalized

Keyboard Command = [N]

The *Rainbow* display plot can be "normalized" so that all three *Time Slice* traces have the same amplitude at a point you select. To simultaneously enable *Normalized* and select the normalization point, hold down the [Shift] key while clicking on the plot (typically near a peak on one of the traces) with the left mouse button. When *Normalize* is on, two vertical white lines on the plot indicate the normalization point. To move the normalization point, hold down the [Shift] key while clicking another on the plot. Pressing the [N] key, or selecting the *Normalized* command from the View menu, toggles normalization on and off.

Power Spectrum

View Menu > Power Spectrum

Power Sp.

Keyboard Command = [P]

In Frequency Slice mode, the Power Spectrum command plots a single trace that represents the average of all, or a specified range, of the frames calculated for a wave file. The locator bars in the Waveform display above the main plot, and the FFT range over which the power spectrum was calculated is shown in the *Frame Range* readout to the right of the plot.

Frame Range
0 - 170

The time range over which the Power Spectrum is calculated may be specified, using time coordinates or FFT frame numbers, on the *Power Spectrum* tab of the *Options* dialog box or by clicking the *Frame Range* readout to the right of the plot and entering a range of FFT frames in the pop-up dialog. *Power Spectrum* is a toggle command. Repeat the command or click the Power Spectrum button again to return to the normal, single-FFT Frequency Slice display.

<u>Note:</u> The mathematical technique used to calculate the *Power Spectrum* is an RMS running average of the complex frequency data.

Options Menu Commands

Most of the Analysis module's user-configurable options and properties are set in the *Options* dialog box. The Options dialog box is organized into 12 separate "pages" for different types of settings. We also refer to these pages as "tabs" because each has an index tab at the top that is always visible (in the top portion of the dialog box window). To bring a page to the front when the dialog box is open, click on its tab.

The Options *menu* commands open the Options dialog box with the corresponding tabbed page already on the top of the stack. There are also keyboard shortcuts to some of the more frequently accessed tabs.

All

Options Menu > All

Options

Keyboard Command = [Alt] + [O]

Opens the Options dialog box with the last tab used on top (any tab may be selected any time the Options dialog box is open).

Calibrate

Options Menu > Calibrate

The *Calibrate* tab of the Options dialog box is used to calibrate multiple files to a single amplitude/magnitude reference. This feature helps to simplify comparison of two files and can be used to calibrate measurements stored in files to some external reference for analysis or comparison purposes. There are three basic calibration options:

- ***Calibrate to Current File*** — Sets the 0 dB point for all frequency-based displays (Time Slice, Frequency Slice, and Spectrograph) to the highest value actually found in the current wave file (or the file displayed on top if two files are loaded).

- ***Calibrate to Full Scale*** — Sets the 0 dB point for frequency-based displays to the highest magnitude value theoretically obtainable given the wave file's sampling resolution (8 to 24 bits per sample).

- ***Calibrate to Custom Level*** — Sets the 0 dB point for all frequency-based displays to the level specified in the *0 dB* field of the *Custom Level* section. Custom magnitude values are specified as "dB down" from the theoretical maximum for the file's

sampling resolution (8 to 24 bits per sample).

- **Set Custom Level To Current File** — Clicking this button plugs in the highest magnitude value actually found in the current wave file (or the file currently displayed on top if two files are loaded) as the 0 dB point for the custom calibration option.

Color

Options Menu > Color

The SIA-Smaart Analysis module allows you to customize the colors of virtually everything on the screen and even use your own bitmap files as a background. Color and background options are loaded as sets called *Color Schemes.* Several ready-to-use color schemes (including two with "metallic" bitmapped backgrounds) are included with the program and you can easily define your own. The *Colors* tab of the Options dialog box allows you to create, edit and manage *Color Schemes* for the Analysis module program window and displays.

The Color Schemes section of the Colors tab has a list of all available Color Schemes along with buttons for creating, editing, and deleting user-defined schemes. A pair of list boxes below the Color Schemes section allow you to select separate color schemes for display and printing purposes.

Color Schemes

New...

Clicking the *New* button calls the *Edit Color Scheme* dialog box, allowing you to create a *new* color scheme. In this dialog box, you can change colors for various display elements and optionally, a Windows bitmap (*.bmp) file to use as a background for the program Window. Color selections initially shown in the dialog box for each element are based on the *Color Scheme* currently in selected in the list.

Edit..

The *Edit* button calls the Edit Color Scheme dialog box to allow you to edit an *existing* (user-defined) Color Scheme selected in the *Color Schemes* list.

Delete

Clicking this button *deletes* the user-defined Color Scheme selected in the list on the left.

- **Display Color Scheme** — Selects an existing Color Scheme to be used for display purposes (the colors shown on your computer screen).

- **Printing Color Scheme** — Selects an existing Color Scheme for use when printing from the Analysis module.

Devices

View Menu > Devices

If your computer has more than one sound card (or driver set) installed, the *Devices* tab of the Options dialog box allows you to select which devices the Analysis module should use.

- **Wave In** — Selects the device to be used for audio recording.

- **Wave Out** — Selects the device to be used for audio output (wave file playback).

Frequency Zooms

View Menu > Frequency Zooms

Keyboard Command = [Z]

This section of the *Options* dialog box controls frequency ranges for the four *Frequency Zooms.*

- **MIN** — Sets the lowest frequency displayed when the zoom preset is selected.

- **MAX** — Sets the highest frequency to display. Entering the word "Nyquist" in this field automatically sets the MAX frequency to the highest frequency obtainable at the wave file's sampling rate (Nyquist Frequency (Hz.) = Sampling Rate ÷ 2).

Meters

The settings in the Meters tab of the Options dialog box control the appearance and behavior of the Input Level Meters in the Record Wave File dialog box. The meters are calibrated to regard the maximum digital output of the A/D system of the sound card being used normally 16-24 bits) as 0 dB.

Peak Hold

- **Peak Hold** — When this box is checked the input level meters will continue to indicate the highest input level encountered in the input signal for a specified period of time after the level drops.

- **Hold Peak For** — Specifies the length of time (in seconds) the meters hold peak levels when Peak Hold (see above) is enabled.

Meter Colors

The input level meters in the Record dialog are designed to display three colors, the normal "LED" color (Light Green for input 0 and Light Blue for input 1 when the default Color Scheme is used) plus Yellow and Red. The meter "LEDs" will display input levels in the normal LED color up to the amplitude threshold specified for Yellow, then Yellow up to the threshold for Red, and Red above that level.

- **Yellow if above** — Sets the amplitude threshold at which the meter LED color changes to Yellow.

- **Red if above** — Sets the amplitude threshold above which the meter LED color changes to Red.

Appearance

Meter (Options for the appearance of the area immediately surrounding the meters).

- **Raised** — The meter area will appear to protrude from the background.

- **Inset** — The meter area will appear to be sunken into the background.

- **Flat** — The meter area will be indistinguishable from the background.

LEDs (Options for the appearance of the meter "LEDs.")

- *Raised* — The meter area will appear to protrude from the background.
- *Inset* — The meter area will appear to be sunken into the background.

LED Levels

The Analysis module allows you to specify amplitude values for each individual pair (L/R) of "LEDs." Meter scaling is independent of any display calibration setting. All values must be given as a negative number of decibels (dB) where 0 dB is equal to the maximum (digital) output amplitude of the selected wave-in device's A/D section.

Power Spectrum

Options Menu > Power Spectrum

The default P*ower Spectrum,* an average of all FFT frames calculated for a wave file, is automatically computed when the wave file is transformed. You can change the time range of the Power Spectrum at any time by specifying a custom range on the *Power Spectrum* tab of the Options dialog box.

The beginning and end values for a custom range of FFT frames may be specified by FFT frame number or by time coordinates. The Power Spectrum is recalculated for the specified range when you click *OK* to exit the Options dialog box. If you entered the range using time values, the Analysis module will adjust the time range specified to coincide with the beginning and end of the closest FFT frames before recalculating.

Printing

Options Menu > Printing

The *Printing* tab of the Options dialog box controls the appearance and content of pages printed directly from the Analysis Module. The following options are user-definable:

Header

- *Title (1, 2, and 3)* — You can specify up to three lines of title text to appear in the header section at the top of Analysis Module printouts. If any of the three *Title* fields is left blank, that line is omitted on the printed page.

Footer

- **Note (1 and 2)** — You can specify up to two lines of footer text to appear at the bottom Analysis Module printouts. If either of the *Note* fields is left blank, that line is omitted on the printed page.

- **Print Today's Date** — If this box is checked, the current date (as set on your computer) will be inserted in the title area at the top of the printout. If you want to use a date other the current system date of your computer on the printout, un-check this box and type the desired date as one of the *Title* lines.

Note: In addition to the note and date lines as specified in the *Footer* section above, Analysis Module printouts will also include a final footer line with the notation "Created by *(User Name)* using SIA Smaart." The *User Name* will be the name of the user to whom this copy of Smaart Acoustic Tools is registered. This line cannot be edited or omitted.

- **Show custom print dialog before print and print preview** — When this box is checked, selecting either the or commands will open the *Custom Print Information* dialog box to allow you to set title text and other options for the printout before proceeding with the selected operation. The *Custom Print Information* dialog box includes all of the same options as the *Printing* tab of the *Options* dialog box. If you un-check this check box, the Analysis module will bypass the *Custom Print Information* dialog box in *Print* and *Print Preview* operations and execute the selected command immediately.

Rainbow

Options Menu > Rainbow

The *Rainbow* tab of the Options dialog box has the following options for the *Time Slice* Rainbow display feature.

- **Center Frequency +/– 1 side band** — When this option is selected, the Rainbow display plots the "Time Slice" for the selected center frequency plus two sidebands one octave or 1/3-octave up and down from the center frequency depending on the resolution selected (Narrowband resolution is not available for the Rainbow display).

- **Specific frequencies** — Selecting this options allows you to select three specific (octave) "Time Slices" to be displayed when the Rainbow feature is enabled.

Slice

The Slice tab of the Options dialog box provides the following options controlling the appearance and behavior of the Time Slice and Frequency Slice displays.

Time Slice

- **Center Frequency** — Sets the center frequency for the Time Slice plot (the center frequency can also be changed using the *Next Slice* and *Previous Slice* commands).

Frequency Scale:

- **Narrowband** — Sets the Time Slice plot to narrowband (single FFT point) resolution.
- **1/3-Octave** — Sets the Time Slice plot to octave-band resolution.
- **Octave** — Sets the Time Slice plot to 1/3-octave-band resolution.

Amplitude Range:

- **Max** — Sets the *upper* end of the displayed magnitude (y-axis) range of the plot.
- **Min** — Sets the *lower* end of the displayed magnitude (y-axis) range of the plot.

Frequency Slice

Frequency Scale:

- **Linear** — Displays the Frequency Slice plot using linear frequency scaling.
- **Log** — Displays the Frequency Slice plot using logarithmic frequency scaling.
- **1/3-Octave** — Displays the Frequency Slice plot using 1/3-octave frequency scaling.
- **Octave** — Displays the Frequency Slice plot using octave-band frequency scaling.

Amplitude Range:

- **Max** — Sets the *upper* end of the displayed magnitude (y-axis) range of the Frequency Slice plot.
- **Min** — Sets the *lower* end of the displayed magnitude (y-axis) range of the Frequency Slice plot.

Spectrograph

The *Spectrograph* tab of the Options dialog box contains a number of options that control the appearance and behavior of the Spectrograph display.

Bar Width

The Spectrograph is made up of vertical bars representing each FFT frame that falls within the displayed time range. The vertical bars are divided into sections representing individual FFT data points. The following options control the way these individual sections of the vertical bars are plotted.

* **Uniform** — This option displays the *Spectrograph* as a continuous color map with each vertical bar having a uniform width from top to bottom.

* **Relative** — Varies the (horizontal) widths of individual sections in the vertical bars plotted *relative* to their magnitude value (higher magnitude values are wider).

Colors

* **Dynamic Range** — This value sets the magnitude range included in the *file-based* Spectrograph display *only.* All magnitude values in the Spectrograph display are given as "dB down" from zero so specifying a dynamic range of 50 plots a magnitude range from 0 dB to –50 dB.

* **Number** — Selects the number of colors to use in the Spectrograph plot (from 8 to 236 depending on your display hardware and drivers). This setting, along with the Dynamic Range, also determines the number of decibels represented by each color.

* **Min** — Selects the color used to represent the *minimum* magnitude value(s) shown on the Spectrograph plot. Clicking the colored square opens a standard Windows *Color* dialog box enabling you to choose a different color for the lowest value(s).

* **Max** — Selects the color used to represent the *maximum* magnitude values shown on the Spectrograph plot. Clicking on the colored square opens a standard Window *Color* dialog box enabling you to choose a different color for the highest value(s).

* **Default** — The Default button resets the *Min* and *Max* color selections to the program defaults.

- *Gray* — Plots the Spectrograph using shades of gray rather than colors. This option is useful for monochrome displays or for printing purposes.

<u>Note</u>: The Analysis module automatically generates the colors in the spectrum between the specified *Min* and *Max* colors using Hue, Lightness and Saturation (HLS) calculations.

Frames

The options listed in this section enable you to specify a time range for the Spectrograph display using FFT frame numbers (rather than the Time Zoom commands).

- *All* — Displays all the FFT frames calculated for a wave file to show the entire time range of the file.

- *Custom* — Enables the two edit fields to the right, allowing you to specify starting and ending FFT frame numbers for a custom time range you wish to display.

Frequency (Y) Scale

- *Linear* — Displays the frequency (*y*) axis of the Spectrograph plot on a *linear* scale.

- *Log* — Displays the frequency axis of the Spectrograph plot on a *logarithmic* scale.

<u>Note</u>: Calibration for all frequency-based display modes in the Analysis module (i.e., all display modes except Time Display) is now set from the tab of the Options dialog box.

Time Display

Options Menu > Time Display

The *Time Display* tab of the Options dialog box provides the following controls for the appearance and behavior of the Time Display plot.

Amplitude Scale

- **Linear** — Displays the amplitude (*y*) axis of the Time Display plot on a linear scale.

- **Absolute Value** — Displays the amplitude (*y*) axis of the Time Display plot on a linear scale with all negative values converted to positive.

- **Decibel** — Displays the amplitude (*y*) axis of the Time Display plot on a logarithmic (decibel) scale.

Amplitude Range

The *Amplitude Range* controls allow you to set both the Current and Default amplitude range to be displayed on the Time Display plot.

- **Max** — Sets the *upper* end of the displayed amplitude (y-axis) range of the plot.

- **Min** — Sets the *lower* end of the displayed amplitude (y-axis) range of the plot.

Show Delta Cursor In

Time values in the Analysis module can be displayed in any combination of time, samples, or distance equivalents depending on which of the following four options is selected.

- **Time** — Displays the horizontal difference in seconds.

- **Feet** — Displays the horizontal difference in feet.

- **Meters** — Displays the horizontal difference in meters.

- **Samples** — Displays the horizontal difference in samples.

Speed of Sound — The Analysis module uses the *Speed of Sound* setting to calculate distance equivalents for time values. The speed of sound value can be specified in either feet or meters, depending on which unit of measure is selected (see above). The default speed of sound value is 1120 feet per second or 341.359 meters per second.

Waveform

The Waveform tab of the Options dialog box contains the following options for controlling the amplitude *range* of the Waveform Display.

Amplitude Zoom

- **None** — Displays an amplitude range representing +/– 100% of the maximum possible amplitude given the wave file's digital resolution (+/– 127 for an 8-bit file, +/– 32 k for a 16-bit file).

- **Peak to Peak** — This option scales the y-axis scale of the *Waveform Display* equal to the largest (+/–) amplitude values actually encountered in File 1.

- **Custom** — Allows you to set a custom amplitude range for the Waveform Display using the two edit fields. The minimum (Min.) and maximum (Max.) values for the amplitude range are specified as a percentage of the maximum amplitude values possible given the wave file's digital resolution.

Default On Open

- **None** — Selects "None" (full scale) as the start-up default amplitude scaling for the Waveform Display.

- **Peak to Peak** — Selects Peak-to-Peak (see above) as the start-up default amplitude scaling for the Waveform Display.

Set All Options to Default

Returns all user-configurable settings, with the exception of the *Color Scheme* and *Device* selections, to the Analysis module's "factory" defaults.

Mouse Procedures

Most Analysis module functions can be accessed several different ways (e.g., menu commands, on-screen controls or hot keys). Some of these also have mouse short-cuts and there are a few functions that can *only* be accessed using your mouse (or other pointing device). This section details the Analysis module's most important mouse procedures and shortcuts:

- *Time Range Selection* (*Time Display, Time Slice* and *Spectrograph* modes)

  Select/Change (First) FFT Center Point (*Time Display* mode only)

- *Select Normalization Point for Rainbow Display* (*Time Slice* mode only)

- *Manual Delta/Decay Rate Calculation* (*Time Slice* and Time Display modes)

- *FFT Frame Selection* (*Frequency Slice* mode only)

- *Masking* (Frequency Slice mode)

Time Range Selection

In *Time Display, Time Slice* and *Spectrograph* modes, the main plot can display the entire wave file time range or any portion thereof. Vertical *Range Locator* bars in the *Waveform* display show the boundaries of the time range currently displayed.

To select a specific portion of the time range for display using your mouse, place the mouse cursor over the point on the plot in the *Waveform* display where you want the displayed range to begin and hold down the left mouse button.
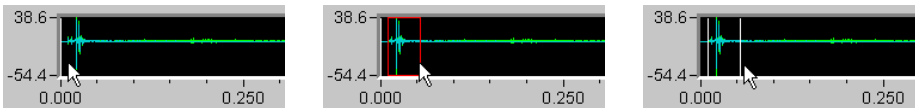
Drag the cursor to the right or left until you reach the point where you want the displayed range to end (a "rubber band" box will appear in the *Waveform* display to show your progress) and release the mouse button.

After you release the mouse button, the *Range Locator Bars* will snap to the boundaries of your selection and the main plot will change (zoom in or out) to display only the time range you selected. Clicking in the area outside the plot in the *Waveform Display* with the left mouse button or pressing [Shift] + [Up Arrow] on the keyboard will return the time range to the full extent of the wave file.

Select/Change (First) FFT Center to Peak

Setting the center point for the first FFT in the transform series on the initial peak in the impulse response ensures that the peak data is altered as little as possible by the data window function in the first FFT frame. All of the data windows provided in the Analysis module attenuate the samples closest to the beginning and end of each FFT frame to the largest extent, leaving the samples closest to the center of the frame untouched for the most part. To set or move the center point for the first FFT frame to any point on the *Time Display* plot, hold down the [Ctrl] key while clicking on the plot with your mouse. To reset the first FFT starting point to the beginning of the wave file, hold down the [Ctrl] key while clicking *off* the plot anywhere in the margins of the plot area. To clear the FFT center point selection press [Ctrl] + [X].

Note that specifying the first FFT center point only works if are enough samples between the beginning of the file and the selected FFT center point for the first half of the FFT. If the selected center point would cause the first FFT to "overrun" the beginning of the wave file, the program will default back to beginning the first FFT on the first sample in the wave file when the file is transformed.

Select/Change Normalization Point for Rainbow Display

The (*Time Slice* mode) *Rainbow* display plot can be "normalized" so that the three *Time Slice* traces have the same amplitude at a point you select. When *Normalize* is on, two vertical white lines appear on the plot to indicate the normalization point. To move the normalization point, hold down the [Shift] key while clicking on the plot (typically near a peak on one of the traces) with the left mouse button. Pressing the [N] key toggles normalization on and off.
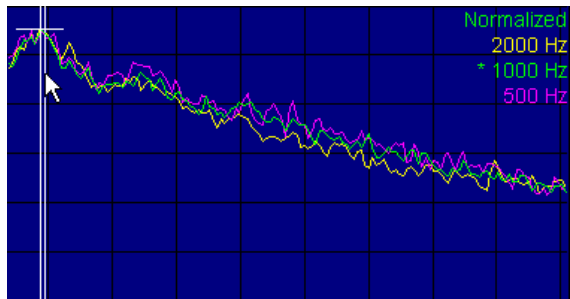
Delta/Decay Rate Calculation

In *Time Slice* mode you can use your mouse to calculate the decay rate for an impulse response recorded in a wave file. The same procedure can be used to find the time and amplitude difference between two points in the *Time Display* mode.

To perform this procedure, place your mouse cursor at the point on the plot where you wish to begin the decay rate calculation. Hold down the left mouse button and drag the cursor to draw ("rubber band") a line to the desired end point.
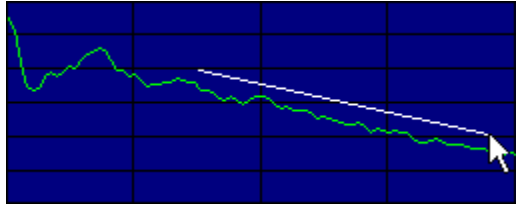
(0.5800 sec, -24.00 dB), Slope=-41.38 dB/sec, T60=1.4 sec

The line you draw remains visible on the plot when you release the mouse button. The relative time and amplitude difference between the end points of the line, the decay slope in dB per second, and equivalent RT60 value will be displayed in the relative cursor readout above the *Waveform* display to the right of the tracking cursor readout. To clear the line, click once on the plot with your left mouse button.

FFT Frame Selection in Frequency Slice Mode
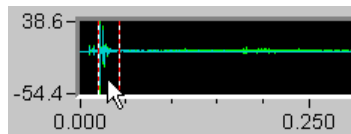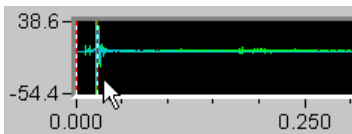
In *Frequency Slice* mode, two vertical striped *Frame Locator* bars in the *Waveform* display indicate the time position of the FFT Frame currently displayed in the plot area. To select a different FFT frame for display using your mouse, place the mouse cursor over any point in the *Waveform Display* and click the left button once. The *Frame Locator Bars* will move to the new location and the data from the selected FFT frame will be displayed on the main Frequency Slice plot.

Frequency Masking

Frequency *Masking* is useful when analyzing signals known to be contaminated by persistent extraneous noise sources such as fluorescent lights or electric motors. Masking is applied in Frequency Slice mode but the effects of the mask are also visible on the Spectrograph plot when Masking is on.

To create a mask, in Frequency Slice mode, first select Narrowband resolution and press [Ctrl] + [M] (or select *Use Mask* (under *Masking*

in the View menu) to activate Masking. With the mouse, click and drag from left to right to draw a horizontal line across the frequency range you wish to mask at the highest magnitude you want to allow for those frequencies.

When you release the mouse button you should immediately see the effect of the mask. Repeat this process for any other frequencies you wish to mask. You can add more frequencies to the mask at any time using the same procedure. The mask is kept in memory until you clear it (using the *Clear Mask* command) or exit the Analysis module.
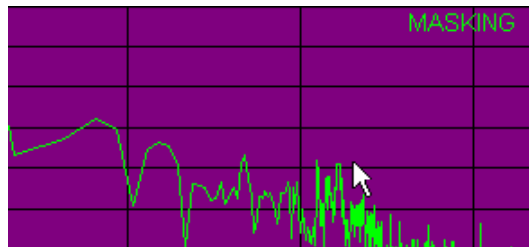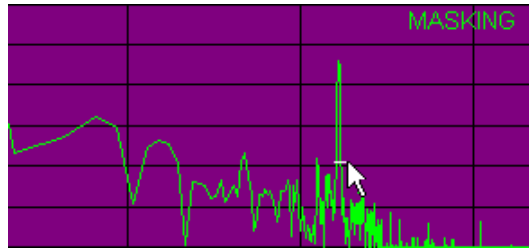
Analysis Module Keyboard Shortcuts

The following is a list of the Analysis module's keyboard accelerators, or "hot-keys." While there are alternate ways of accessing virtually all of these functions, learning the keyboard accelerators can significantly speed your work if you use the Analysis module extensively.

Key names on the list below are enclosed in [brackets] to differentiate them from descriptive text. Some commands use more than one keystroke, for example the accelerator for the Transform command is "[Ctrl] + [T]." This means you must hold down the *Control* key while you press the *T* key. For three-key accelerators, you would hold down the first key while pressing the third

Amplitude

[+/=] = Amplitude (Y) Zoom Out

[–] = Amplitude (Y) Zoom In

[PgUp] = Move Amplitude Range Up

[PgDn] = Move Amplitude Range Down

Time Display

[Ctrl] + [D] = Decibel Amplitude Scaling

[Ctrl] + [L] = Linear Amplitude Scaling

[Ctrl] + [V] = Absolute Value Amplitude Scaling

[Ctrl] + [Alt] + [P] = Set FFT Center to Peak

[Ctrl] + [Alt] + [Left Arrow] = Move FFT Center Point Left

[Ctrl] + [Alt] + [Right Arrow] = Move FFT Center Point Right

[Ctrl] + [X] = Clear FFT Center

[Ctrl] + Mouse click outside plot = Clear FFT Center Point Selection

Mouse: [Ctrl] + Clicking on Time Display plot to set/move first FFT center point.

Audio

[Shift] + [P] = Play wave file

Cursor

[Alt] + [Right Arrow] = Move Right

[Alt] + [Left Arrow] = Move Left

Display

[D] = Time Display

[F] = Frequency Slice

[S] = Spectrograph mode

[T] = Time Slice Mode

[P] = Power Spectrum (Frequency Slice mode only)

[R] = Rainbow Display (Time Slice only)

[N] *or* [Shift] + click on (Time Slice) Rainbow Display plot = Normalize

File

[Ctrl] + [O] = Open File 1

[Ctrl] + [P] = Print

[Ctrl] + [T] = Transform

File 2

[Shift] + [Ctrl] + [L] = Close File 2

[Shift] + [Ctrl] + [O] = Open File 2

[Ctrl] + [F] = Change "Z" Order

[Ctrl] + [R] = Reset to No Shift
(clear time offset)

[Ctrl] + [Left Arrow] = Move (File 2) Left
(negative time offset, fine adjust)

[Ctrl] + [Right Arrow] = Move (File 2) Right
(positive time offset, fine adjust)

[Shift] + [Ctrl] + [Left Arrow] = Move Left
Fast (negative time offset, coarse adjust)

[Shift] + [Ctrl] + [Right Arrow] = Move
Right Fast (positive time offset, coarse
adjust) Frequency

[1] = Select Frequency Zoom Range 1

[2] = Select Frequency Zoom Range 2

[3] = Select Frequency Zoom Range 3

[4] = Select Frequency Zoom Range 4

[5] = Narrowband Frequency Banding

[6] = Narrowband Linear Frequency Scaling

[7] = Narrowband Log Frequency Scaling

[8] = Octave Frequency Scaling/Banding

[9] = 1/3-Octave Scaling/Banding

Mask

[Ctrl] + [M] = Enable/Disable Masking

[Shift] + [Ctrl] + [M] = Clear Mask

Options

[Alt] + [O] = Options

[Alt] + [S] = Spectrograph Options

[Alt] + [Z] = Zoom Options

Slice

[SpaceBar] = Next Slice

[Shift] + [SpaceBar] = Previous Slice

[Ctrl] + [G] = Go To Slice (Frequency
Slice only)

Mouse Shortcut: Click in Waveform Display

Time Zoom

[Up Arrow] = Time zoom out

[Down Arrow] = Time zoom in

[Left Arrow] = Move Time Range Left

[Right Arrow] = Move Time Range Right

[Shift] + [Up Arrow] = Default Zoom (zoom
out to full time scale)

Mouse Shortcuts:
Click/drag in Waveform Display to
select range

Click off plot in Waveform Display
to clear zoom

Notes:

Not all keyboard commands are active in all display modes.

Keyboard accelerators are a subset of the menu commands. Please refer to *Analysis Module Commands,* beginning on page 68, for complete command descriptions.

Chapter 4: Reference

This section is intended to present "real-world" definitions of some of the basic concepts and terminology used by SIA-Smaart software products. The definitions given here are accurate with regard to SIA-Smaart, but should not necessarily be considered comprehensive or mathematically complete. We have provided a list of reference titles at the end of the section for those who wish to find a more in-depth treatment of the topics discussed here.

Percentage Articulation Loss of Consonants (%ALCons)

This index was traditionally measured using long word lists and a set of listener scoring tables. The %ALCons value can be calculated from the impulse response or energy time curve two ways, using either a long or short method. The long form calculation includes the ambient noise condition. The short form calculation does not. The %ALCons calculation, particularly the short form calculation, which ignores noise conditions, tends to give results that are very generous compared to more rigorous standards such as STI. %ALCons values between 0 and 4.0 are considered good, 12 to 4 percent are fair, below 12 is considered poor.

Clarity Factor (C*n*)

The acoustical quantity of Clarity is the ratio of the early sound to late sound. The division between early and late sound is given in milliseconds and expressed as C*n*, where *n* is a number of milliseconds (e.g., C10, C20, C50, etc.). For example, C50 is a decibel energy ratio of the sound arriving within 50 milliseconds of the direct sound compared to the overall reverberant decay. This tends to be one fairly accurate predictor of subjective intelligibility. Clarity values above 5 are considered good, -2 to 5 is fair, values below -2 are considered poor.

The default Clarity factor for the Intelligibility module is C50. Other commonly used Clarity factors include C35 and C7. You can change the Clarity factor to anything you like by a by changing the number of milliseconds in the *Clarity* field on the *Graph* tab of the *Options* dialog box.

Data Window Functions

Data Window functions help to reduce truncation errors arising from the segmenting of longer data series into FFT frames. Remember that the Fast Fourier Transform (FFT) requires a data series of finite length. The value of the samples immediately before and after the data series used in the FFT are unknown and this can lead to certain anomalies in the Fourier transform results.

All data windows types operate under the same basic principle. They "de-weight" the samples nearest to the beginning and end of the FFT data series to help reduce these truncation errors, or "edge effects." The only real difference between different the Data Window types is the "shape" of the "window." The Analysis module provides several different *Data Window* options including Hanning, Hamming, Blackman, Blackman-Harris, Parzen and Welch. The option labeled "None," for no data window (also called a "rectangular" window), should be avoided for most applications.

Decibels

The decibel (dB) is a unit used to express the logarithmic *ratio* of two amounts of power, voltage or any two values. Typically decibels are used when the two values may differ over a very large range. The need for logarithmic scales in acoustics and audio is a result of the wide range of sensitivity to sound intensities, frequencies and pressures that makes up the range of human hearing. For example, a decay of 60 dB represents a sound that has decayed by one million times in terms of energy! Most measurements in audio are based on either voltage or sound pressure and are expressed in decibels. The key point to understanding the decibel is that it is used to describe a *comparison* of two values.

Two useful points regarding decibels to remember:

• Decibels are always a *ratio* of one value to another. You must know the reference value in order to calculate the decibel values.

• 0 dB can mean several different things (hence much of the confusion with decibels).

For our purposes, in the context of SIA-Smaart Acoustic Tools, 0 dB means:

1. The input and output, or two inputs are equal in level for a given frequency, as in a *Transfer Function* display, **or**

2. The highest level that can be reached. This usage applies to the *Input Level Meters*, where a signal greater than 0 dB represents overload, and the Analysis module's *Spectrograph* display mode where all magnitude values are represented as "dB down" from zero.

In the case of the *Input Level Meters,* 0 dB represents a value equal to the maximum input voltage allowed by the sound cards' A/D converter. The range of the meters is set as a negative number of decibels *relative* to this value. Similarly, the dynamic range of the *Spectrograph* display is specified as a negative number of dB relative to the maximum digital amplitude resolution of the waveform file.

Note: For many sound cards, the maximum A/D converter input voltage is user-selectable. Check your sound card documentation and utilities for more information.

Direct to Reverberant Level

This is simply the amplitude difference between the level of the direct sound and the level of the top of the reverberant decay stage of the impulse response. This value is expressed in decibels and is of interest in both broadband and band-limited decay curves as it can play an important part in overall system intelligibility. There is no real benchmark for this value other than that higher numbers are better.

Energy Time Curve (ETC)

The ETC is the envelope of the decay of a system. The ETC is mostly used to look at reflections and decay rates of a system's response. The ETC is similar in some ways to the SIA-Smaart Analysis module's *Time Slice* display, however the ETC is calculated by using the Hilbert transform on the time-domain impulse response data whereas the Time Slice is calculated using a frequency-domain representation of a system's impulse response.

Fast Fourier Transform (FFT)

The *Fast Fourier Transform* (FFT) is a special case of the Fourier Transform, a mathematical technique used to *transform* time-domain data into frequency-domain data. The output of the transform is a set of complex numbers representing both frequency and phase information about the original time series. The term <u>*Fast*</u> Fourier Transform comes from the fact that if you specify the time-domain data to contain a "power of 2" samples, the transformation can be calculated very quickly by digital computers. Powers of 2 are the values of 2^n, where *n* is an integer (1, 2, 3, etc.).

	if *n* is:	the *power of 2* (2^n) is:
Examples:	8	256
	9	512
	10	1024
	11	2048
	12	4096

All frequency transformations in SIA-Smaart are *FFTs* and require the time record to be a power of 2 (samples) in length. For long files and real-time operations, SIA-Smaart will do the frequency domain transformation in pieces (frames), each a power of 2 (samples) in length.

<u>*Note:*</u> It is certainly *possible* to calculate the Fourier Transform for a time record with *any* number of samples. But when the number of samples is not a power of 2, the number of calculations required can become very large. (On a PC, this could result in a very *slow* Fourier Transform.)

Frequency Resolution

The *frequency resolution* of an FFT is calculated from the sampling rate and FFT size. For a given FFT size, the frequency resolution of the FFT will be equal to the *sampling rate,* divided by 2 (the *Nyquist* frequency), divided by half the FFT size. This means that the data points along a frequency plot are distributed evenly along the frequency axis every *"Q"* Hertz, where *Q* is the frequency resolution. For example, sampling at 44,100 Hz gives you a Nyquist frequency of 22,050 Hz. An FFT size of 1024 yields 512 frequency data points. Divide 22,050 by 512 to get a frequency resolution of 43.07 Hz. This means that there is one data point every 43.07 Hz.

The table below shows frequency resolutions for some typical sampling rates and FFT sizes:

Sampling Rate	FFT size *(in samples)*	Frequency Resolution
44100 Hz	2048	21.5 Hz
44100 Hz	1024	43.0 Hz
22050 Hz	2048	10.7 Hz
22050 Hz	1024	21.5 Hz
11025 Hz	4096	2.6 Hz

Increasing the FFT size, or decreasing the sampling rate will increase the frequency resolution. Notice that both of these actions *increase the time it takes to collect a group of samples*. This is the real key to increasing frequency resolution.

Note: A linear distribution of FFT points, where there is one point every *n* Hertz, yields much better resolution at higher frequencies than at lower frequencies. In octave or fractional octave band display modes, the Analysis module will display low bands for which the frequency resolution is insufficient to resolve the band's frequency data as "ghosted."

Impulse Response

You could think of the impulse response as the "signature" of a system (e.g., a room and/or electroacoustic system). For our purposes, we can define the impulse response as the signal that describes the changes a test signal undergoes as it passes through a device or system under test. The broadband impulse response contains a wealth of information about the system including the delay through the system, frequency response, reflections, reverberation and decay. In fact, it is actually possible to use the impulse response as a filter to convolve a "dry" signal, such as speech or music, and hear what it would sound like if actually played through the same system.

Linear and Logarithmic Scales

In SIA-Smaart, both time and frequency plots may be viewed using either linear or logarithmic (amplitude or frequency) scaling. A linear scale plots the frequency or amplitude range *linearly* across the plot. On a linear frequency scale display, each *unit of measure* (e.g., Hertz) covers the same distance on the axis. On a logarithmic frequency display, each *decade* (power of ten), decibel, or octave (depending on the type of information displayed) covers the same amount of the axis.

For example, on a linear *frequency* plot covering the range or 20 Hz to 20 kHz, the entire right half of the plot would be devoted to the range of 10k to 20k, *a single octave.* On a logarithmic scale each decade: 10-100, 100 -1k, 1k - 10k, etc. would occupy an equal amount of the axis. 100 to 1000 Hz, a range of 900 Hertz covers the same distance on the plot as 1 k to 10k, ten times the number of Hertz.

Both linear and logarithmic scales (for frequency and amplitude) have their advantages. The choice of log or linear plots is based on experience and personal preference. The human ear hears logarithmically in frequency, however many people feel that harmonic structures and comb filters can be seen more easily on a linear display. SIA-Smaart Acoustic Tools allows you to switch between logarithmic and linear displays easily to find the type of plot that helps you best understand your data.

Maximum Length Sequence (MLS)

A Maximum Length Sequence (MLS) is a type of pseudorandom noise generated using a system of bit-shift registers and a known group of "seed" numbers. An MLS sequence is almost always a power of 2 (64, 128, 256, 512…) samples in length however this is not an absolute requirement. One attractive feature of MLS is that it is *repeatable* noise. Starting with the same seed numbers the same exact sequence can be generated any number of times. Also, when an MLS sequence (or string of MLS sequences) is played through a system under test and the results recorded, the measured results and the MLS seed numbers can be entered into a calculation called the Hadamard transform to calculate the impulse response of the very efficiently.

Octave-Band Resolution

Octave-band resolution provides a combination of logarithmic and linear frequency scales. On this type of display, each *octave* is represented as an equal width along the frequency axis. This correlates with human hearing, where each octave *appears* to sound like an equal interval, even though the number of Hertz *doubles* between each increasing octave! In SIA-Smaart, octave band values are derived from FFT data. The lowest bands appear as "ghosted" images (or are not shown) when the frequency resolution is insufficient for the data to be reliable *(see Frequency Resolution).*

Octave and fractional-octave resolution plots are often simpler to read than narrow band frequency data and can be useful for getting a quick overview of a signal.

Pink and White Noise

Pink and white noise are sounds or audio signals containing random (or pseudorandom) broadband energy. Both are commonly used as signal sources in audio testing. The difference between pink and white noise is obvious if you look at both using an RTA. Pink noise appears flat on an octave- or 1/3-octave band display, because it has equal energy per *octave*.
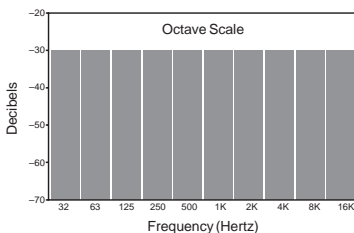
Pink noise on an octave band display.
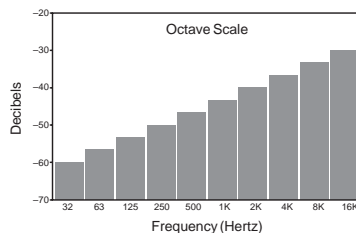
White noise on an octave band display.

White noise will appear to be sloping upward to the right on the same display (increasing in magnitude at higher frequencies). White noise has equal energy per *frequency* so each higher octave has 3 dB more energy than the octave below it. In fact, white noise has so much high frequency energy it can damage loudspeakers if used improperly. We recommend exercising great care when using it as a test signal for most electroacoustic measurement applications. When making an MLS measurement, keep in mind that the MLS stimulus signal is essentially white noise.

Schroeder Reverse Time Integration

Sometimes called Schroeder Inverse Integration, this is a technique for finding the decay rate of a room, first published by Manfred Schroeder. Before personal computers were powerful enough to calculate the actual impulse response of a system (using techniques such as MLS or a dual-FFT transfer function), Schroeder, et. al., suggested that a way to make measurements of reverberant decay in large halls was to use an impulsive source, record the data using a strip chart recorder, and integrate the decay by summing the energy, starting at infinity and working backward in time.

Speech Transmission Index (STI and RSTI)

This type of measurement is based on the assumption that human speech is essentially a fundamental waveform modulated by low frequency signals, STI uses the Modulation Transfer Function (MTF) for several octave bands to try and predict the ability of a system to convey spoken information. The MTF can be measured using a special speech-like amplitude-modulated test signal or calculated from the broadband system impulse response (the SIA Intelligibility module uses the latter method). STI is expressed as a value between 0 and 1 where 0 means completely unintelligible and 1 means perfect intelligibility, STI values above 0.7 are considered good to excellent. Values from 0.45 to 0.7 are fair, and values below 0.45 are poor.

The *Rapid* Speech Transmission Index (RSTI) is a shorter version of the STI calculation that uses data from just the 500 Hz and 2000 Hz octave bands. RSTI tends to yield slightly higher numbers then the more rigorous STI calculation which takes into account almost the entire audible spectrum.

Sampling Rate

The *sampling rate* in digital audio is the number of times per second an analog audio signal is sampled and digitized. The most important practical consideration is that the sampling rate limits the frequency content of the signal being digitized. A general rule is that sampling rate should be at least double the number of Hertz in the highest frequency you want to include in the digitized signal. The frequency that is equal to one half of the sampling rate is called the *Nyquist* frequency.

Some other issues affect the highest frequency you can accurately measure in sampled signals. In theory, you can measure to half the sampling frequency but

in practice things like aliasing, Sigma/Delta conversion and anti-aliasing filters make frequencies close to the Nyquist tricky to measure.

Compact discs run at a standard sampling rate of 44.1 kHz. Professional digital audio recording machines often sample at 48 kHz. Computer sound cards generally run at one of several user-selectable sampling rates such as 44.1 kHz, 22.05 kHz or 11.025 kHz. The SIA-Smaart Intelligibility module determines the available sampling rates by polling your computer's sound card each time it loads. Currently, the fastest sampling rate supported by any SIA-Smaart program module is 48 kHz.

General Rule*:* The higher the data sampling rate, the higher the frequency content of the digitized signal. However higher sampling rates also mean more data to be calculated, and possibly stored, *in a given period of time*.

Transfer Function

The transfer function is a complex *comparison* of two signals, typically a *reference* signal and a *measurement* signal. Most commonly, this comparison is made between the input and output of a device or system, such as an equalizer, sound system, or room. When recording an impulse response measurement using the *Dual FFT* method, the SIA-Smaart Intelligibility module records two channels of audio data, transforms the data using FFTs, and calculates the transfer function of the two input signals in the frequency domain. The results of the transfer function calculation are then transformed (using an *inverse* FFT) and displayed in the time domain to show the *impulse response* of the device or system under test.

Glossary of Audio and Acoustics Terms

Analog to Digital (A/D) Conversion: The process of digitizing an analog signal. This process almost always involves limiting the frequency content of the digitized signal.

Absolute Value: For any real number, the Absolute Value is the unsigned magnitude of the number. Example: the absolute value of -3, denoted $|-3| = 3$.

Amplitude: The size of a real number (e.g., a number of volts), in either the positive or negative direction. The term amplitude typically refers to numbers that are not complex or plotted on a logarithmic scale, such as the numbers stored in the A/D process.

Attenuation: A decrease in the level of a signal. Attenuation can refer to reduction in level for a specified frequency range or a decrease in the overall level.

Clarity: The acoustical quantity of Clarity is defined as the ratio of the early sound to late sound. The division between early and late sound is given in milliseconds and expressed as Cn where n is a value in milliseconds, e.g., C10, C20, C50, etc.

Coherence: A mathematical function which represents the linearity between two signals. The coherence is dependent on both signals and the number of averages taken. The coherence value is calculated in the frequency domain and yields a value between 0 and 1 for each frequency.

Compressors: Electronic devices that cause changes in gain (typically attenuation) as a function of the input level. These devices should NOT be used when making transfer function or impulse response measurements.

Crosstalk: Undesired energy in one signal (or channel) appearing from an adjacent signal or channel.

Data Window: A mathematical function used to smooth the negative effects of truncation that occurs when a finite number of FFT points are used to transform time domain data into the frequency domain. The Data Window(s) reduce the amplitude of the time domain data at the ends of the sample to be transformed.

Decay Rate: The rate at which a signal decays (diminishes in magnitude), usually a function of frequency and expressed in either decibels per second, or relative to the amount of time that would be required for the signal to decay 60 decibels at the given rate of decay. *(see Reverberation Time)*

Decibel: The decibel, often abbreviated as dB, is a logarithmic *ratio* between two values. In acoustics, decibels most commonly refer to the ratio of an input level to the output level of a system, or a given level compared to a fixed reference.

Dynamic Range: The difference in level between the highest and lowest signal a system can accept or reproduce.

Early Decay Time (EDT): An acoustical quantity which is the time required for a system, typically a room, to complete the earliest stage of its decay, typically the first 10 decibels. The Early Decay *Rate* can be expressed in units of decibels per second, or relative to the amount of time that would be required for the signal to decay 60 decibels at the given rate of decay.

FFT: The Fast Fourier Transform is a mathematical technique used to transform time domain data into the frequency domain. The term *"Fast"* refers to the fact that when the number of time domain samples is a power of 2 (16, 32, 64, 128, 256, et al.) the calculations can be performed very quickly by a digital computer.

Latency: The delay through a given unit or system. Latency is often referred to as the *throughput* delay of a device. It is typical for digital delays to have a small latency even when they are set to zero time delay.

Linear Scale: The term linear, refers to a set of values or scale of a graph on which values are evenly spaced. On a linear scale, each value (or unit) has equal dimension.

Logarithmic Scale: A scale where each *power* of a given number (e.g., ten) is given equal dimension.

Magnitude: A number assigned to a quantity so that it may be compared with other quantities. For complex quantities, the magnitude is the square root of the sum of the squares of the real and imaginary parts.

Nyquist Frequency: In digital audio, the Nyquist frequency is one half of the sampling rate. The Nyquist frequency represents the highest frequency contained within a signal sampled at a given sampling rate.

Octave-Band Resolution: Octave band resolution combines all data points in a given octave and displays a total energy value for each octave band (as opposed to a linear or logarithmic *narrowband* displays that plot the value of individual FFT data points). Standard octaves are centered on 16, 31.5, 63, 125, 250, 500 (et al.) Hertz.

Overlap: For the purposes of SIA-Smaart, overlap refers to the amount of data each successive FFT Frame shares in common with the one before. Overlapping FFT frames are analogous to shingles on a roof. When overlap is set to zero, each new FFT frame begins where the last one stopped, like beads on a string.

Parametric Equalizer: Equalizers are devices with one or more filters that affect the frequency content of a signal. On a *parametric* equalizers, the parameters of the filter(s) including gain or attenuation, frequency and bandwidth are user-adjustable.

Phase Shift: A timing difference in a signal (relative to some reference) at one or more frequencies, typically expressed in degrees.

Pink Noise: A random (or pseudorandom) signal in which, over a given averaging period, each *Octave-band* has an equal amount of energy

Propagation Delay: The time it takes for sound to travel from one place (typically a loudspeaker) to another place (typically a microphone).

Reverberation Time: The amount of time required for audio energy introduced into a system (typically a room) to diminish, or decay a specified number of decibels. Often expressed as an RT60 value.

RT60: Reverberation time. The amount of time required for a system, typically a room, to decay 60 decibels. *(see Decay Rate)*

Sampling Rate: The number of points per second used in the analog to digital conversion process. Typically expressed in Hertz.

Spectrograph: A three-dimensional plot, displayed in two dimensions with color representing the third dimension (or z-axis). The spectrograph is a topographical representation of the common waterfall display.

Spectrum: The frequency content of a given signal.

Speed of Sound: The speed of sound is dependent on the material of propagation, the temperature and several other factors. Typical values for the speed of sound in air are 1120 ft/sec, or 341.376 m/sec. This is the default value SIA-Smaart program modules use to calculate distance equivalents for time differences.

White Noise: A random (or pseudorandom) signal in which over a given averaging period, each *frequency* has equal energy.

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Appendix A: Troubleshooting

Installation Issues

SIA-Smaart Acoustic Tools is written for use with Microsoft Windows. It is a native 32-bit Windows application. The installation program will not run from the DOS prompt. (See *Section 1: Getting Started* for installation instructions.) Versions of Microsoft Windows previous to Windows 95/NT 4.0 are not supported. Windows must be running properly on your system before you can install or run this software.

A valid *Serial Number* and *Install Name* are required is required to install Smaart Acoustic Tools. The serial number and install name are keyed to the product code on your installation and will only work when installing from a CD with the identical product code. Your *Serial Number* and *Install Name* and installation *CD Product Code* are imprinted on the registration card included in each SIA-Smaart Acoustic Tools software package. Please keep your registration card in a safe place. This information may be required for you to receive technical support and product upgrades. Your install codes will also be required if the program files on your hard disk are deleted or corrupted and you need to reinstall the software for any other reason.

It is strongly recommended that you close all other Windows programs, including any automatic anti-virus and/or system monitor software you have running, before attempting to install this software. Antivirus programs in particular are known to cause conflicts with the installer program and may prevent the software from properly registering itself with the operating system. Installation problems related to other software usually manifest themselves after a seemingly successful installation when one or more of SIA-Smaart program module will not open. Shutting down all other running programs and repeating the installation will usually correct this.

If you experience problems *during* the installation, for example, if the installer program "hangs" or issues error messages before the installation process is completed, you may have a defective or otherwise unreadable installation disk. If you experience problems of this nature, contact SIA technical support for troubleshooting assistance. Technical support contact information is listed at the end of this chapter.

Computer Sound Hardware Issues

Windows-compatible sound hardware must be present and properly configured for your system to use recording features in the SIA-Smaart Intelligibility and Analysis modules. SIA-Smaart Acoustic Tools uses standard Windows Application Programming Interface (API) calls to access the sound card and should work properly with any 100% Windows-compatible audio device.

If the Intelligibility and/or Analysis modules will not recognize your sound hardware, check to see if you can record and play back wave files using the Sound Recorder and Media Player. These are standard Windows utilities, usually located in the Accessories section of the Programs menu (accessed by clicking the *Start* button on the Windows *Taskbar*).

If you cannot play and record using the Sound Recorder and/or the Media Player will not recognize your sound hardware, check to make sure both the hardware device and the software that came with it are properly installed. Sound hardware setup typically requires loading software drivers and utilities in addition to any hardware installation.

Depending on your system, the sound hardware driver software could be on a diskette supplied with the sound card or computer, the Windows setup disk(s), or both. Some sound cards also require the user to set jumpers or DIP switches on the card itself before installation. If you have upgraded your system to a new version of Windows from a previous version, you will probably need to obtain updated drivers for your sound hardware from the sound card or computer manufacturer as well.

It is not uncommon for manufacturers to discover device driver software problems after a card or computer ships. If you are sure your hardware and software drivers are properly configured and you continue to experience problems, contact the sound hardware or computer manufacturer. In many cases you can obtain updated driver software that will correct the problem(s).

If your computer has more than one sound hardware and/or MIDI I/O device or driver set installed, make sure the proper devices are selected for both audio and MIDI I/O on the Devices tab of the Options dialog box (accessible from the Options menu) in both the Intelligibility and Analysis modules.

Configuring Windows Audio Controls

`◀× 7:28 PM`
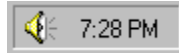
Beginning in Windows 95, the Windows Volume Control application provides a standardized interface for controlling the audio inputs and outputs of many or most Windows compatible audio devices. If you have trouble getting a signal into either SIA-Smaart Acoustic Tools program module from the computer's (line-in) inputs or suspect the computer's internal microphone may be enabled and contaminating your measurements, use the following procedure to access the *Record* (input) controls.

- Open Volume Control — *double*-click the speaker icon (shown above) on the Windows Taskbar or click the *Start* button and select *Programs* > *Accessories* > *Multimedia* > *Volume Control*

- In the Volume Control application, select *Properties* from the *Options* menu.

- Click the *Recording* "radio button," make sure the boxes for *Microphone* and *Line-In* are checked in the list below, and click *OK* to exit the Properties dialog box.

Notice that the title of the Volume Control window changes to *Recording Control.* Make sure the *Select* box for *Line-In* is checked, confirm that the balance control is centered and the fader is set to a useful level. If your computer is equipped with an internal microphone, it's a good idea to also *un-check* the *Select* box under the *Microphone* fader before exiting the Recording Control application.

Input Levels

It is important to maintain proper input signal levels when performing measurements of all types. For best results you need a good strong signal that does not "light" the Clip indicators above the meters. If the signal level is too low, you will get a poor signal-to-noise ratio. If it is too high, the input(s) will overload causing "clipping." This will compromise the accuracy of your measurements and could damage your computer. Overloading one input channel can also cause the other to behave erratically.

When using a test signal with a high "crest factor," such as pink noise, transient peaks in the signal may be too fast for the Input Level Meters to detect. We recommend keeping the overall input signal levels at about –12 dB for random noise and MLS stimulus signals, and music.

The *Input Level Meters* in the Intelligibility module *Record Impulse* dialog box and in the Analysis module *Record Wave File* dialog indicate the input signal level at the sound hardware's A/D converter. A/D converter input voltage ranges vary. Some sound cards allow you to set A/D input voltage levels using jumper settings or software. Some do not. It is often desirable to use a mixer, microphone preamp or other external device to control input levels.

If your computer has both microphone and line level inputs, be sure to avoid sending a line level signal to a microphone input. As a rule, we recommend you avoid using the microphone inputs. The quality of the preamp circuitry is typically very poor.

Problems With the Intelligibility Module Impulse Response Recorder

Poor Signal-to-Noise Ratio

If you do not see a clear peak in a new impulse response recorded in the Intelligibility module (using either the MLS or Dual FFT method) and/or the "noise floor" of the measurement is too high to see acoustical information about the room/system under test, it may be helpful to increase the gain of the measurement microphone and/or loudspeaker(s) used to stimulate the room. In reverberant spaces, the amplitude of the peak is a function of the signal-to-noise ratio of the measurement signal.

Another way to increase the signal-to-noise ratio of both MLS or Dual FFT measurements is to increase the number of averages by increasing the *Averages* value in the *Record Impulse* dialog box. If this value is greater than 1, the Intelligibility module records the specified number of FFT or MLS frames and averages the data from all recorded frames in the impulse response calculation. Each doubling of the number of frames averaged will yield 3 dB more signal to noise (down to the absolute noise floor of the system under test or the measurement system, whichever is higher).

Insufficient Time Window

If the impulse measurement plot appears erratic, without a single strong peak and/or with several peaks of nearly equal amplitude, the reason could be that the *FFT* or *MLS Time Constant* is too short. Check to insure that the time constant is set to a value larger than the decay time of the system under test and increase the FFT size/MLS Length (or decrease the sampling rate) if necessary.

"Negative" Delays (Dual FFT Only)

An oddity of the *Dual FFT* impulse recorder mathematics causes "negative" delays to be "wrapped around" and displayed at the end of the time scale. When this happens, the strongest peak in the impulse response will show up closer to the end of the measurement than the beginning, appearing to indicate an unbelievably long delay time through the system under test.

There are two likely causes for this problem.

1. The computer input channels are wired incorrectly and need to be swapped.

2. There is a delay in the *reference* channel.

In the first case, swapping the inputs to the computer will correct the problem. In the second case, you must either compensate for, or remove the delay in the reference channel. To accomplish this, you could delay the *measurement* signal by an equal amount (using an external delay device) or simply remove the device(s) introducing delay from the reference signal path.

Digital signal processing devices are likely suspects for reference signal delay. For example, many digital delay units introduce some delay, even when set to "bypass" and/or indicating 0 ms delay time. This is called *latency* or throughput delay. Other types of digital audio devices may also introduce unwanted delay.

Printing Problems

Problems with printing from SIA-Smaart applications often turn out to be the fault of incorrect, outdated or corrupt printer drivers. If you have printing problems, make sure you have the correct Windows software driver for your printer properly installed and configured. If necessary, reinstall the drivers from the Windows setup disk(s) or disk supplied with your printer using the Windows *Printers* control panel. If problems still persist, check the documentation that came with your printer for troubleshooting help. You may need to contact your printers manufacturer to obtain an updated driver.

Restoring the Default Configurations

User-definable options for the SIA-Smaart Acoustic tools program modules are very flexible and can sometimes be confusing, especially at first. Both the Intelligibility and Analysis modules store the state of nearly all of these options when you exit so if a problem related to configuration does arise, chances are good that it will still be there the next time you start the program.

Any time you wish to restore the Intelligibility module to its "factory" default settings, select *Set All Values to Default* from the *File* menu. To reset the Analysis module to its defaults, select *Set All Options to Default* from the *Options* menu. Resetting to defaults changes virtually user-definable parameters for the program back to their original states. The only exceptions are that the *Color Scheme* and audio input/output device selections are left as set on the *Colors* and *Devices* tabs of the *Options* dialog box.

On-line Help and Readme Files

If you have a problem or question about SIA-Smaart Acoustic Tools and cannot find the answer in this manual, check the on-line Help files and Readme file (if applicable). The Help files, accessible from the *Help* menus of both the Intelligibility and Analysis modules, may contain information not included in the manual. Your copy of Smaart Acoustic Tools may also include a file called Readme.doc, located in the main program folder, with additional information about the software. The default location for the Acoustic Tools program folder is C:\Program Files\SIA-Smaart Acoustic Tools.

Technical Support

Support for SIA software products is available through our web site, by e-mail or by telephone. On the SIA web site *(www.siasoft.com),* you can find Application Notes, Case Studies and answers to frequently asked questions as well as product news, updates, and an on-line web support forum. Maintenance updates, and information of interest to SIA-Smaart users is posted on the web site as it becomes available. For technical support via e-mail, the address is *support@siasoft.com.* The telephone number for SIA technical support (in the USA) is (+) 508 234-9877.

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