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JBL-SMAART: A PC-BASED SOUND SYSTEM OPTIMIZATION AND ACOUSTIC MEASUREMENT TOOL

S. Berkow

JBL Professional, Northridge California, U.S.A.
SIA SoftwareCompany, Inc., New York, New York, U.S.A.

1. INTRODUCTION

JBL-Smaart was developed to provide affordable real-time and disk-based processing of audio measurements and recorded audio data. JBL-Smaart runs on the Microsoft® Windows® operating system, using standard Windows-compatible audio hardware to provide audio input and output. The program is divided into two sections: a Real-Time module and an Analysis module.

The JBL-Smaart Real-Time module provides two channel spectrum analysis and real-time transfer function calculation as well as a Delay Locator™ feature. The Delay Locator also functions as an impulse response calculator. The JBL-Smaart Analysis module provides waveform displays, a disk based FFT generator and several frequency domain displays including individual FFT frames, a 3-dimensional spectrographic display (time vs. frequency with magnitude represented by color) and Time Slices™, magnitude vs. time plots for user selected frequency range(s). The Analysis Module also has the ability to calculate several quantitative acoustical quantities from data gathered in the Real-Time module's Delay Locator.

2. INDEPENDENCE FROM DEDICATED AUDIO HARDWARE AND STIMULUS GENERATION

The JBL-Smaart version 1.0 differs from most other acoustic measurement systems in that: 1) the program uses standard (Windows-compatible) audio hardware, not a dedicated Digital Signal Processing (DSP) platform, 2) no stimulus signal is generated by the program itself.

There are two reasons why JBL-Smaart does not generate stimulus signals internally: 1) the transfer function calculation is (within limitations) independent of stimulus, and 2) Windows provides only nominal hardware-independent support for synchronized full-duplex operation of audio hardware. To help understand this, a simplified discussion of the way in which Windows allows programs to control audio signals is provided below.

The Microsoft Windows multimedia services provide several Application Program Interfaces (APIs) that effectively isolate audio applications from concerns about specific audio hardware. Using these APIs, a Windows application, e.g., JBL-Smaart can utilize any Windows-compliant sound card to record audio data, play audio data, and change audio related parameters such as sampling rate and number of channels. The APIs exist in a layer of Windows multimedia system DLLs which translate application requests and pass them to vendor specific DLLs, which then pass vendor specific requests to the audio hardware drivers.

Windows provides two levels of APIs. "High Level" APIs provide simple ways to perform simple tasks such as playing wave files and recording data directly to wave files. Windows' "Low Level" APIs require more programming but provide a greater level of control. In particular, data may be played from and recorded to memory buffers, allowing rapid processing and real time monitoring. JBL-Smaart uses low level audio services to play, record, and monitor audio data.

If the installed audio hardware supports full-duplex operation, low level audio APIs allow the application to play and record audio data simultaneously. Unfortunately, the low level APIs do not include synchronization functions needed to precisely align playback and recording (in time). There are techniques that will let an application get close to signal alignment but Windows does not provide a standard interface to request precise alignment (to the sample level) from the hardware-specific drivers.

To achieve this level of control without restricting use of the program to specific or dedicated hardware, the JBL-Smaart development team found it would be necessary to convince and rely upon all of the hardware manufactures themselves to support highly synchronized full-duplex operations. We realized that this would be a potentially infinite task, given the number of parties involved, the frequency of updates, the lack of standards and documentation, and the small incentive for providing such a high level of level support for a small segment of their market. However we also realized that we could make useful measurements while satisfying the program design objectives of low cost and hardware independence simply by utilizing an external stimulus signal source.

3. REAL-TIME SPECTRUM ANALYSIS AND THE TRANSFER FUNCTION

The basic function of the real-time module is two-channel spectrum analysis. In this mode of operation, two independent channels of audio data may be transformed into the frequency domain and the magnitude of the signals frequency content displayed. The frequency domain data may be displayed in either narrow-band or fractional octave-band resolution,

The limiting factors when determining the ability of the spectrum analyzer to stay in "real time" are the speed of the computer, the sampling rate of the incoming data and the size of the FFT being used (larger FFTs require more processing power). In the Real-Time module, JBL-Smaart supports sampling rates up to 44.1kHz and FFT sizes up to 4096 points for real-time spectrum analysis. The frequency domain data generated from the FFT calculations may be displayed in narrow-band, or 1/1, 1/3 or 1/6 fractional octave band resolution (with the octave bands derived from the FFT data). As there is currently no standard for transforming FFT-generated data into fractional octave bands, JBL-Smaart attempts to mimic the ISO/ANSI standard filters as closely as possible and does not display fractional octave band data with less than 2 FFT data points per band.

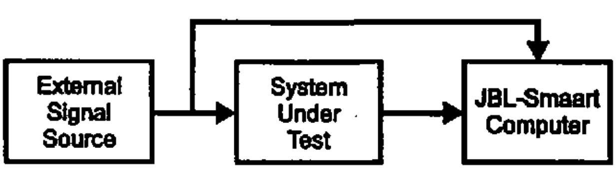

Figure #1: Simple block diagram of measurement setup.

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Using 486 or Pentium processors, it is possible to compare two signals in the frequency domain, in real-time, by complex division of the FFT data of each input channel. As displayed in Figure #1, a typical test setup provides a test signal (external to the measurement software), which is split, with one 'branch' of the signal sent directly to the computer and the other branch sent through the system to be tested (see Figure #2).

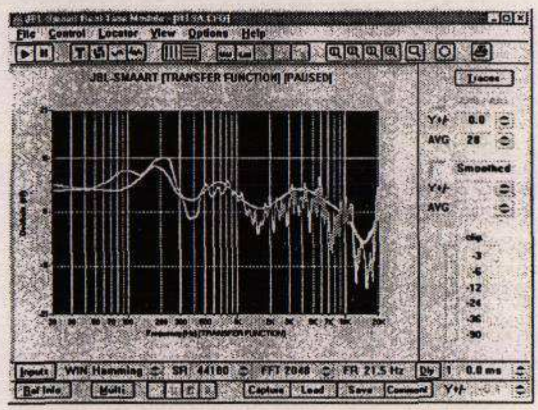


Figure #2: Transfer Function display. Two measurements are displayed, a loudspeaker measurement was made and stored and an equalizer measurement was inverted to allow direct comparison with the stored measurements.

This Transfer Function (frequency domain comparison of two signals) is valid under the following conditions:

- The two input signals are "aligned" in time
- The test signal used has sufficient energy duration and is well-behaved in the frequency bands of interest.
- There is no dynamic processing on either signal path

To address the alignment of the two signals, JBL-Smaart contains a digital signal delay internal to the program (implemented in software). This single channel signal delay is assignable to either channel, and can be set in 0.10 millisecond steps, from 0.0 sec. to 250 mSeconds. For Transfer Function measurements, the delay inherent in the A/D converter of the sound card can be disregarded as it is common to both input channels.

The sufficiency test(s) for suitable test signals can be determined by careful measurement of the coherence function between the two channels once they have been aligned in time. Practically, for applications in professional audio, filtered noise, typically pink, is often used as a suitable test signal. JBL-Smaart provides a coherence function calculation, as a post process of the transfer function data, to help determine the validity of transfer function data as a function of frequency.

4. THE LOCATOR AND THE IMPULSE RESPONSE

The signal alignment criteria for the real-time transfer function requires that the user be able to measure the relative delay between two channels. Two techniques are commonly used to measure the delay "between" the two signals; a correlation routine which focuses in on the actual delay time using an iterative process, or the measurement of the impulse response between the two signals, (which in practice will be the impulse response of the entire system under test). JBL-Smaart provides the user with the latter technique. A transfer function between the two input signals is calculated and the inverse FFT is used to calculate the time domain representation of the system's impulse response. In practice it is often very easy to find the propagation delay of a system by simple visual examination of the impulse response.

To provide valid results, the Delay Locator / impulse response measurement requires that: 1) the input signal to each channel have sufficient broadband energy of sufficient duration, 2) The system under test behave linearly, and 3) that the FFT time constant selected for the calculation be large compared to the decay of the system under test. Once again sufficiency can be demonstrated by comparison with known results, or by calculation of the coherence function. The linearity requirement requires that all dynamic processing in the system, such as compressors, limiters etc. be bypassed during measurement,

The Delay Locator's FFT time constant is calculated by dividing the user-selected FFT size by a user-selected sampling rate. JBL-Smaart polls the sound card installed and lists all sampling rates supported by both the card and JBL-Smaart as options. Increasing the FFT size or decreasing the sampling rate will act to increase the Delay Locators FFT Time Constant. As the decay of many acoustic systems, such as large rooms or sounds systems with long propagation paths, often require FFT time constants with length of one or more seconds, large FFTs are required. JBL-Smaart's Delay locator provides FFT sizes from 128 to 65k points and sampling rates from 5k to 44.1kHz, providing a wide range of possible FFT time constants, from small fractions of a second to several seconds. As larger FFTs require more computation time, and the impulse response of the system is assumed not to change (the linearity requirement), the delay locator calculation is not done in real-time. Rather, a user defined amount of data is collected (i.e. recorded) and the impulse response calculation is done immediately, as a post process, often taking several seconds to complete. The user may choose to have a number of measurements made and the results averaged.

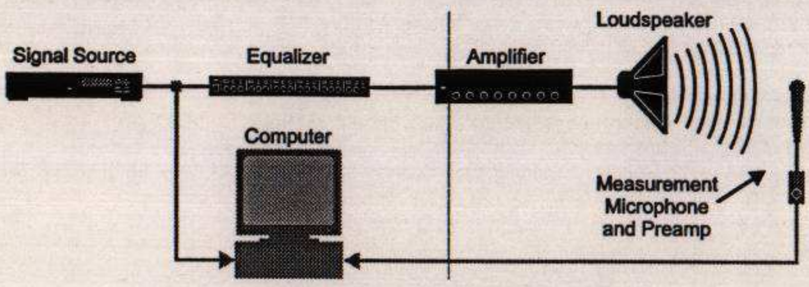
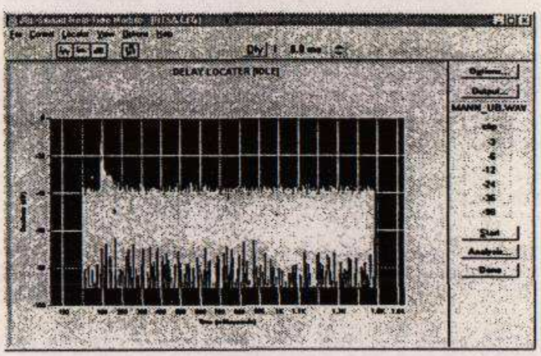


Figure #3: Simple measurement system setup.

It is important to note that the Delay Locator's transfer function-based impulse response measurement does not directly provide semi-anechoic results. Replacing, the "system under test" block in Figure #3 with an electronic system such as an equalizer, an impulse response will provide the time, magnitude and phase information describing state of the equalizer. Replacing the "system under test" with an amplifier / loudspeaker / microphone / acoustic-environment, see Figure #2, will yield the impulse response of the entire signal path, not the impulse of any one device in the signal chain. It is possible to post process the entire impulse response with a time window to achieve similar results to traditional semi-anechoic techniques.

Figure #4: Delay Locator
The output of the delay locator.
The amplitude scale is in decibels.



The output of the Delay Locator is stored as a Windows standard wave file and can be displayed on either a logarithmic, liner or absolute value amplitude scale (see Figure #4). Using the windows standard wave file format allows data calculated by the Real-Time module to be read by the JBL-Smaart Analysis module or any other program that works with windows wave files.

5. THE JBL-SMAART ANALYSIS MODULE

As the Real Time Module is used for spectrum analysis and delay/impulse response calculations, the JBL-Smaart Analysis Module provides numerous ways to manipulate disk based audio data, in both the time and frequency domain. Time domain data may be viewed on a user selectable time scale and either logarithmic, linear or absolute value amplitude scales.

The Analysis module can read standard 16-bit wave files containing either stereo or mono data. Currently JBL-Smaart supports wave files containing data at almost any sampling rate and any amplitude resolution up to 16 bits but does not support compressed audio file formats such as ADPCM. Audio data may be transformed from the time domain into the frequency domain, via FFT calculations ranging from 128 points to 32k points. Seven data windows and a user-selectable FFT overlap parameter are also provided.

The generic Windows wave file format allows for arbitrary extensions that, in theory, should not affect older programs that do not understand the extensions. This would allow JBL-Smaart to include extended data in its wave files while still allowing old programs to read the basic wave information. Unfortunately, many popular and well-established programs do not read wave files in a generic manner. To allow JBL-Smaart to add extra information to wave files would break compatibility with those well known programs.

For this reason, and to maintain the integrity of the original audio data files, the Analysis module stores frequency domain data from the FFT calculations (in complex pair format) in separate "companion" files. This process is virtually transparent to the user. As long as the companion (*.sia) file resides in the same directory as the wave file, the Analysis module will find it any time the wave file is loaded making both the time- and frequency-domain data immediately available for display.

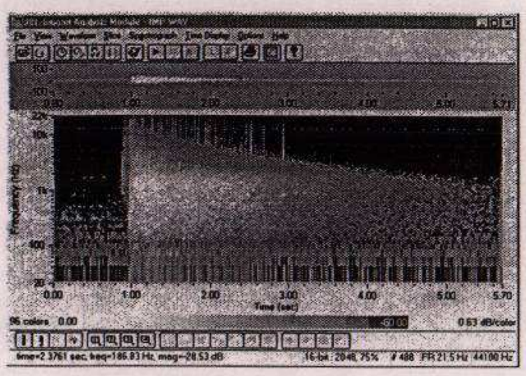
JBL-Smaart provides three main display types for frequency domain data:

- The Frequency Slice (Magnitude vs. Frequency)
- The 3D spectrograph (Frequency vs. Time with Magnitude by Color)
- The Time Slice (Magnitude vs. Time, selected freq.)

The Frequency Slice is a standard magnitude vs. frequency plot which can be viewed on a number of frequency scales. Narrow band data, can be displayed on either a linear or logarithmic frequency scale. Fractional octave band displays of 1/1, 1/3 or 1/6th octave displays are also available. A power spectrum average of all, or a user-selected range of frequency (FFT) frames many also be calculated and displayed on any of the frequency scales.

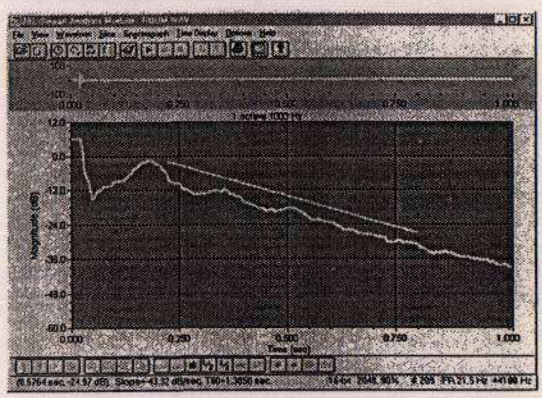
When designing JBL-Smaart we opted to provide a 3D spectrograph, rather than the traditional waterfall display (see Figure #5). A 3D spectrograph provides a topographical view of the signal, with time along the x-axis, frequency along the y-axis (on either linear or logarithmic scale) and magnitude represented by color. The user may specify the depth (in decibels) of the display to determine how much of the signal, based on magnitude is displayed. The number of colors used in the spectrograph plot, from 8 to 236 (that is not a typo, Windows reserves 20 entries in a 256-color palette for "system" colors) is also user-selectable. We have provided additional options which allow the spectrograph to be drawn with each FFT frame separated from adjacent frames by a spacer line and with the width of each FFT frame dependent on magnitude.

Figure #5: Spectrograph. Time on the x-axis, Frequency on the Y-axis (with a logarithmic frequency scale) and amplitude by color.



The JBL-Smaart Time Slice is a magnitude vs. time plot for a selected frequency range (see Figure #6). The frequency range may be specified as 1-octave, 1/3-octave or narrow band. Three Time slice plots representing three different frequency ranges may be viewed on a rainbow plot, with each frequency range displayed in a different color. Within the JBL-Smaart Time Slice display the user may draw a line on the plot to define a slope. Values for the slope are displayed both in units of dB/second and equivalent T60 values. Alternately, JBL-Smaart will calculate reverberation time, early decay rate, and various energy ratios for all 1/1 or 1/3 octave bands based on a peak location and other calculation parameters defined by the user.

Figure #6: The Time Slice display.



6. MEASUREMENT TECHNIQUES AND FOCUS.

Since its introduction in May of 1996, JBL-Smaart has been used in a great many applications. One of the more popular uses has been optimization of large sound systems. The measurement system, consisting of a computer with sound card running JBL-Smaart and a small external mixer is connected into a sound system at various points. The user can then switch between measurement of signals from different insert points using the external mixer see Figure #7. In this configuration, the user would begin by measuring the delay propagation through each signal path of the system. When using a transfer function calculation, it is important to be aware of all sources of propagation delay. Many of the digital devices used in modern sound systems, such as digital equalizers and crossovers have unexpectedly long or program-dependent propagation times.

Once the impulse response of the system under test has been calculated using the Delay Locator, the propagation delay is determined by visual examination of the measured impulse response. To help make this measurement easier to read, it is advantageous to use the eternal mixer to adjust the levels reaching the computer to a point were they are approximately equal in amplitude. At this point the user may choose to examine the measured impulse response in more detail using the Analysis module features.

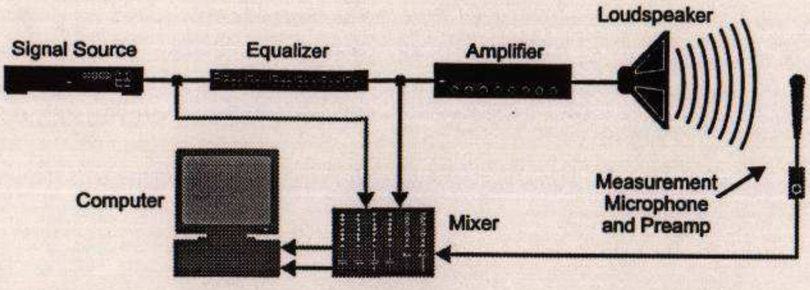


Figure #7: Typical field measurement setup.

Having determined the propagation delay, an internal signal delay is assigned to the channel providing the reference signal to achieve alignment (in time) of the two input signals. It is a good practice to then re-measure the impulse response of the system using the delay locator to ensure that the delay is set correctly and the two signals are, in fact, aligned. The Real-Time transfer function may then be calculated, and stored for comparison with other measured results.

It is a common practice for measurements of loudspeaker systems to be stored and compared with 'live' measurements of equalizers and crossover networks. The Real-Time module's ability to invert the display of a measured response and overlay stored measurements of other devices makes visual comparison very convenient. An advantage of this technique for setting equalizers, is that the bandwidth required for each filter, and the interaction of adjacent filters is easily seen and compared. The equalizer can also be adjusted and the changes can be both seen and heard in real time. This allows the user to obtain a "best-fit" equalizer response both objectively (visually) and subjectively (audibly) for to response of the system being optimized.

7. A TOOL IS JUST A TOOL

It might be appropriate to point out that JBL-Smaart was designed to be a useful field tool, not a scientific test instrument. The goal in designing JBL-Smaart was to address an area of sound system and acoustic measurement that in this age of hi-technology analog and digital audio systems remains amorphously defined and expensive to address: system evaluation and optimization. While great efforts are being made to provide useful and cost effective tools for system design, prediction of performance, acoustic simulations, writing specifications, 2D and 3D system drawings the optimization of sound systems and evaluation of room acoustics remains an area of great confusion. There is little agreement regarding the techniques, tool and even the ultimate goals for system optimization.

In providing JBL-Smaart to the acoustics and audio communities, we do not expect to solve all of the problems of sound system optimization. For example, setting delay times and equalizers will not solve coverage, power handling or many other system design and/or configuration problems (except in extremely rare cases). Further, the goals in setting equalizers and delays in sound systems are not always the same. In the case of a systems designed primarily for playback only, spectral balance might be the most important concern. In a live sound reinforcement system, stability, i.e., gain-before-feedback may be the primary goal of the system tuner. We hope that in JBL-Smaart, we have created a tool that will let acoustical consultants and sound contractors focus their energies on understanding their designs and installations and not their measurement system.