White Paper

Measuring and Optimizing Sound Systems: An introduction to JBL Smaart™

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JBL Smaart is a general purpose acoustic measurement and sound system optimization software tool, designed for use by audio professionals. Running on a Windows® computer and utilizing almost any standard Windows compatible (MCI compliant) sound card, **JBL Smaart** offers an accurate, easy-to-use and affordable solution to many of the measurement problems encountered by sound system contractors, acoustical consultants and other audio professionals.

JBL Smaart was designed by a team consisting of acoustical consultants, sound system designers, mixers, and installers. The goal of the project was to create a tool which would provide easy access to information which will help systems sound better, by identifying potential problems and quantifying system performance. To meet this goal both a real-time module and disk-based analysis module were developed. Perhaps most importantly, the features within **JBL Smaart** suggest a methodology for optimizing sound systems. This methodology is developed and explained in this paper. The method described is not intended as an inflexible procedure, rather as the starting point for you to modify as you like and assist you in understanding and optimizing sound system performance.

1) What am I trying to measure and why?

Before making measurements of a sound system it is critical to ask yourself, "What am I trying to measure and why?"

The performance of a sound system, whether it is a permanently installed system, touring sound system, or some hybrid of the two, is determined a number of ways, both qualitatively and quantitatively. The following is a list of some of the most important questions to ask when determining the level of performance of a given system.

- **Frequency Response:** Does the system have the ability to deliver sound over the intended frequency range, within the expected deviations?

• **Power Handling:** Can the system handle the desired amount of power without excessive distortion or failure?

• **Coverage:** Does the system provide sufficient coverage of required areas at all frequencies?

• **Subjective Quality:** This is always the most important criteria. Does the system meet the audience/owners/performers/operators expectations for perceived sound quality?

• **Stability:** Does the system feed back with microphone(s) open and the gain set to a useful level? (This is obviously not important if the system is playback only.)

• **Noise:** Is the system noisy? Are hums, buzzes and other unwanted noise present in the system?

• **Configuration:** Do you understand the system configuration? Some sound systems have groups of speakers driven from a single source. Others are divided into several sections, each controlled by a different set of control circuitry (such as equalizers, delays, crossovers etc.).

• **Operation:** Are all components of the system working?

No piece of hardware or software (even **JBL Smaart**) can accurately answer all of these questions by itself. Tuning a sound system requires an understanding of the hardware, a discerning ear, accurate measurements, and a disciplined and systematic approach. We doubt that any two system tuners approach the problem exactly the same way. Also, the process necessarily differs, depending on the complexity of the system in question and whether it is an existing system, a touring system, or a new system to be optimized.

However, there are several steps we feel are necessary to any successful exercise in sound system measurement and optimization. The order in which they would be followed might differ according to the tuner's personal preference and the task at hand. The procedure outlined here is based on our own experience and assumes a system that is already in place.

The Transfer Function & System Configuration

At the heart of **JBL Smaart's** Real-Time Module is the transfer function calculation. The transfer function (stated simply) is a comparison between the input and output of a system. The transfer function of a system can be displayed in either the time or frequency domain. To make the results of this calculation easy to understand, **JBL Smaart** displays the result in the frequency domain. To accomplish this calculation **JBL Smaart** compares two signals, each of which are input to your computer's sound card.

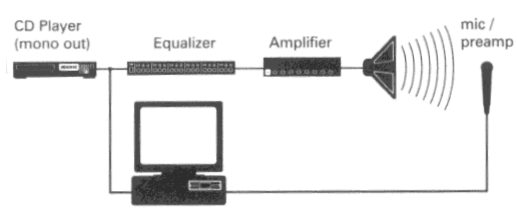

Fig. 1

Figure #1 displays a typical setup for using **JBL Smaart** to make transfer function calculations. The use of two signals, measured at the input and output of a system, allows **JBL Smaart** to display a comparison of the input and output as a function of frequency. This configuration allows the test signal to be noise, music or almost any broadband signal. The Coherence function feature of **JBL Smaart** is used to check the validity of the data, thus helping users to understand whether that data is reliable. For the transfer function calculation to be valid the two input signals must be aligned in time. This can be accomplished using the Delay Locator and Internal Signal Delay features included within **JBL Smaart**.

After making a measurement in the configuration displayed in Figure #1, the results may be stored and the system reconfigured as shown in Figure #2:

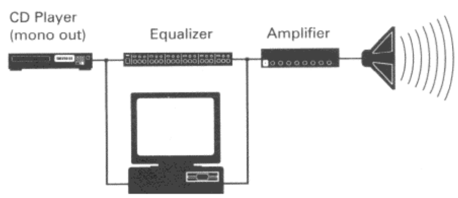

Fig. 2

Measuring the transfer function of the equalizer (Figure #2) is particularly useful when the resulting transfer function is inverted and overlaid on top of the previously stored loudspeaker measurement. The need to switch between measurement points within a system suggests that the use of an external mixer, as configured in Figure #3, may prove extremely useful in the field:

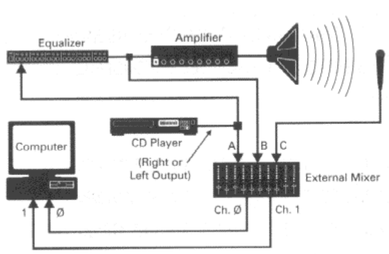

Fig. 3

A Step-by-Step Approach

STEP 1: Evaluation Listening
Before you begin measuring a sound system, we strongly recommend listening to it! You should attempt to qualitatively answer the questions listed on the previous page. This will require you to move around and listen to each section or subsection of the system. Explore the edges of the coverage pattern to see where the various elements are covering and where they are not. It may also be helpful to turn off parts of the system in order to make a more detailed evaluation of various subsystems and components.

Practical Note: Unless you designed the system, take some time to try and understand what the system designer had in mind and how the various elements relate to each other.

STEP 2: Identify Potential Problems
Examining the list of questions above, are there any obvious problems that need to be addressed? For example, unwanted noise, such as hums and buzzes associated with ground loops and “dirty” power, can degrade system performance and should be addressed before **JBL Smaart** testing is begun. Loose and intermittent connections should be fixed. A gain structure that leaves the system hissing should be explored and corrected.

Practical Note: The system must be stable before trying to make measurements. Systems that seem to be changing gain or have noises that come and go are not good candidates for critical measurement. Spend some time sorting things out first.

STEP 3: Select Measurement Points & Positions

This is one of the most important steps in the process. You need to select measurement points that will show you what you need to see. There are two kinds of measurement points: electrical and acoustic.

Electrical measurement points are used to sample the input or output of a particular piece of equipment. If you want to measure a piece of equipment, or the result of a string of series-connected pieces of equipment, make the connections at the input and the output.

Acoustic measurements are made with a microphone. When making transfer function measurements, a reference signal is also required. The connection for the reference signal should be made at the input of the speaker system, the input of the processor if it is a processed system, or at the input of the system's equalizer(s).

Microphone selection and placement are very critical. The microphone must be a known quantity. A measurement microphone should be of high-quality construction, having the flattest frequency response characteristics that you can reasonably afford. When selecting the microphone position, ask yourself two questions: "Is this a useful place to make a measurement?", and "What other things will the microphone pick up in this location that might affect the measurement?" Reflections into the side or back of a measurement microphone can seriously reduce the accuracy of a measurement. Think "mirror", and look around for surfaces that might catch you unaware!

Practical Note: Acoustical reflections of sound energy from large (and some not so large) surfaces may generate "comb filters" in the measured signal. The result is a system of dips in the frequency response that are evenly spaced in frequency. They are easiest to see when the frequency axis is set to linear, as they appear to be a set of valleys evenly spaced across the plot.

STEP 4: Compare Positions

In making acoustic measurements of any system, it is important to make a number of measurements to make sure that you are not being fooled by something affecting the measurement (such as reflections). Move the microphone around and look at what happens to the measured frequency response. Also, you may wish to put sound absorbing materials on the floor between the source and microphone to reduce the effect of the reflected sound energy from the floor (this is another good place to think "mirror")...

STEP 5: Set Equalizers and Delay Settings

Setting equalizers and delays can be very time consuming. There seem to be two distinct stages to the process: coarse adjustment and fine tuning. In the first stage, large adjustments to EQ and delay settings are used to make a system roughly correct. Sometimes, the sheer size of these adjustments may seem a little daunting, but if things are sounding right, you are probably OK. The next stage takes place when the system is getting close to right. At this point, changes of a few dB can make the difference between a good-sounding system and a great-sounding system. Learn to recognize this transition.

After making a number of changes to equalizer settings, it is important to go out and listen to what is happening to the system. Make sure it is moving in the right direction. Just because it looks good on an analyzer screen doesn't mean that it is right. Remember, you are working for the ears, not the instruments!

Important Notes:

- Always make delay adjustments before trying to make fine adjustments in equalizer settings. A combination of small delay and equalization changes can completely change the character of a delay system.

- Setting delays and equalizers can help make some poorly designed sound systems sound better. However, only in extreme and very rare cases is it possible to correct poor loudspeaker coverage with these types of devices.

STEP 6: Critical Listening

This is what it is all about. Here is where you take off the measurement hat and put on the listening hat. Put on a CD (or other program source) and "walk the system". Listen in the front rows and back in the cheap seats. Try it at low levels. Try it at high levels. Run it though its paces. Turn the source off and listen to everything in silence. Make sure that the noise floor is low enough not to affect the dynamic range of the system.

Use material that is familiar to you. Don't be afraid to listen to things others may not like. For this purpose, the best choice might be something you have heard so many times you don't even like it anymore. Only when you are very familiar with (several) program selections will you be able to use them as a basis to quickly and accurately evaluate a system by listening.

STEP 7: Stability Testing

It is very important to explore the stability of any critical sound system before it goes into service. Otherwise, you may find yourself in the uncomfortable position of trying to find and equalize feedback frequencies during a performance or other event. Obviously this is not a concern for playback only systems.

Unstable sound systems are those that have, at one or more frequencies, an overall gain, including the acoustic path, of more than one; in other words, feedback. It follows that a stable system has a comfortable margin of gain before feedback (GBF) at its intended operating level while delivering the intelligibility and frequency response characteristics required for its purpose.

Important Note:

- Feedback can damage audio components. Exercise caution when testing system stability. Feedback is particularly dangerous when it builds up very quickly and overdrives the system, causing overloads or clipping. It might be a prudent safety precaution to use a limiter or compressor during stability testing to help protect system components. Remember that non-linear devices such as limiters or compressors CANNOT be used during transfer function measurements.

Typical Causes of Instability

Instability, or feedback, is often the result of interaction between the off-axis response of the speaker system and the off-axis response of microphones. The biggest problems usually arise when narrow peaks in the off-axis responses of both loudspeaker and microphone coincide. These types of interactions can be very troublesome, and they are not as easy to control as the on-axis responses.

Other possible causes or contributors to stability problems include acoustical characteristics of the room and/or signal processing equipment; particularly reverberation units used in music reinforcement systems.

Detecting Instability

The simplest way to expose a stability problem in a sound system is to turn up the gain, slowly and carefully, until the system feeds back. Not a particularly elegant approach, but it almost always works. If feedback does not occur in the system until the gain is increased well beyond the intended operating level, and the system is free of any noticeable "ringing" at normal levels, it's pretty stable. If not, you will need to find ways to improve it. Depending on the situation, the best solution could be electronic, mechanical, acoustic, even educational, or some combination of the four.

Some Approaches to Stabilizing a Sound System

Stabilizing an unstable system, or giving a system more "margin" (GBF) primarily involves reducing the gain of the system at the problem frequencies. Given the nature of the problem, the most obvious solution is to apply equalization. Although equalization is not a panacea or a substitute for good system design, it is one of the most powerful tools you can bring to the task of stabilizing an existing sound system.

JBL Smaart can help you to quickly identify problem frequencies and apply equalization with great precision. But before you start turning knobs, consider that equalization affects the overall frequency response of the system. There are other strategies that might be equally effective, or more effective, and could afford you greater freedom to make the system sound good.

Mechanical Solutions and Acoustical Solutions

The physical position of microphones and loudspeakers in relation to each other can affect the feedback frequency (or frequencies). Reducing the gain at a problem frequency can sometimes be as simple as using a different microphone, or reorienting one already in use. This strategy is best employed when the microphone in question is intended to remain stationary. Moving or reorienting loudspeakers may also be a possibility. Mechanical solutions are most attractive when they can be applied without giving up any of the sound system's design goals.

Stability problems often arise when loudspeakers are placed close to (or behind) microphones. In such cases it may be possible to add some sound absorbing material or a baffle that reduces the speaker’s field at the microphone’s position or simply reduce the operating level(s) of the loudspeaker(s) in question.

Important Note:

• Moving microphones are moving targets. When speakers or performers move around with microphones, feedback frequencies may shift. Always try to perform stability testing in a manner that resembles how the system will actually be used.

Educational Solutions

An otherwise stable system may lose stability when a number of microphones are open at one time. In this case, the best solution might be to train the operator to keep microphones open only when they are actually in use.

Educating users in microphone technique can also be beneficial. Many people tend to grab microphones or lean very close when they speak. Both of these actions can cause problems. Grabbing a cardioid microphone can increase its physical gain at certain frequencies when the user's hand closes off the rear ports to the microphone element, making a stable system suddenly unstable. When people lean too close to a microphone, they can reflect some energy at problem frequencies back into the microphone themselves, possibly causing feedback.

Electronic Solutions

Some reverberation units can cause an otherwise stable system to become unstable. If this seems to be the case, try experimenting with other settings and/or reducing the overall level of electronic reverberation. Keep in mind that reverberation generators do what they do (very simply put) by feeding back some of the output of the system, through some system of delays, to the input.

With a simple enough system, polarity or phase changes could solve a feedback problem immediately. When the polarity is inverted, instead of positive feedback (something we don't like) we should get negative feedback (something that may be beneficial). However, in large, complex systems with multiple return paths many wavelengths long, phase or polarity changes might just tend to shift the feedback frequency, without increasing stability.

The most common solution to the problem of feedback is to use equalization to take out offending peaks. By peaks, we mean places in the spectrum where there is significantly more gain (or energy build-up) than others. The procedure involves:

- **Carefully,** running the system into feedback
- Identifying problem frequencies, and
- Setting up filters (i.e., EQ stages) to compensate

Important Note:

• We strongly recommend the use of parametric equalizers for this type of application.

How Much Equalization is Enough?

As you equalize a system to increase stability, keep in mind you are reducing gain, even though you are reducing it only at specific frequencies. In many cases, the frequencies in question have proportionately too much gain anyway. You may actually improve the system's frequency response at operating levels while increasing stability.

If there are two or more feedback frequencies, equalization tends to work best when the frequencies are fairly close together. You may find, after you have applied a number of filters at widely spaced frequencies, that all you have really accomplished has been to reduce the overall gain of the system – without really increasing its stability or GBF. It may be necessary to explore other solutions. (In some extreme cases it may be necessary to alter the system design to correct instability.)

STEP 8: More Critical Listening
If you, and everybody else concerned, are satisfied with the system, you're done. More likely, you will need to repeat some combination of Steps 2 through 7 to obtain the best possible performance. Optimizing a sound system is usually a gradual, cut-and-try, give-and-take process (which often takes more time than one would like or expect). We hope that **JBL Smaart** will help to make this process much easier for you.

CONCLUSIONS:

The ability to measure the transfer function between two signals (which have been aligned in time) allows sound system performance to be measured using a variety of test signals, including noise and music. This measurement technique provides an easily understood measurement which can be used to optimize sound system performance. The ability to overlay stored measurements allows both equalizers and delay units to be accurately set. **JBL Smaart** provides the capabilities to make these measurements, and benefits from the graphical user interface provided under the Windows operating system.