Choosing gear for your Smaart measurement system

Smaart

rational acoustics
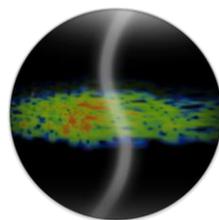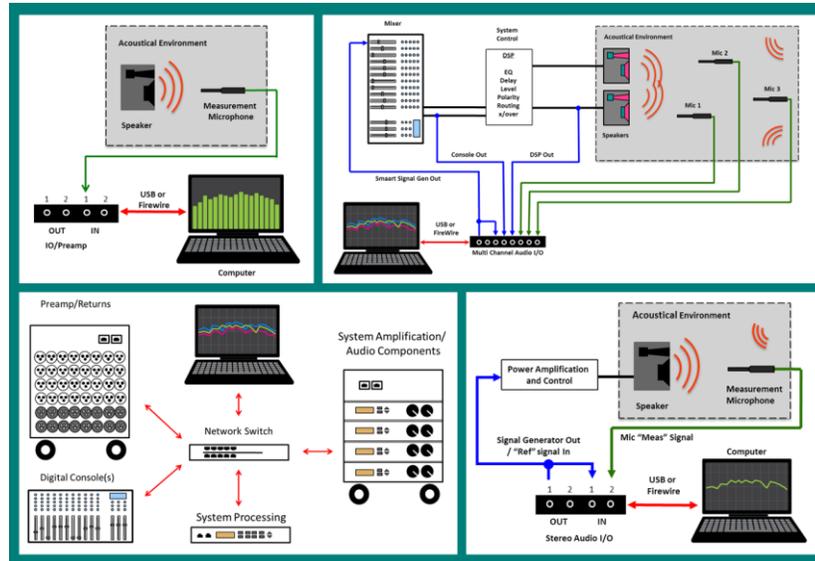
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Introduction

Smaart® is a software-based real-time spectrum and transfer function analyzer. Unlike hardware based, turn-key measurement systems, Smaart can be configured in any number of ways to suit a wide variety of measurement applications.There is really no such thing as a one-size-fits-all Smaart rig. But of course, that flexibility also implies that it is up to each user to configure a system that suits their own needs and budget.

This document is intended as a guide to help usersbetter understand some of the relative merits and trade-offs associated with the many choices for components of a Smaart-based measurement system. The better you understand those trade-offs, the better prepared you are to make cost-effective choices when selecting specific gear for any particular application.

Cabling aside, a complete Smaart measurement system basically consists of three main components:

1. Signal Processing - The Host Computer
2. Signal Acquisition - Audio Interface/Microphone Preamp(s)
3. Acoustical Probe - Measurement Microphone(s)

In each case, *what you need* is a function of *how you use*, or intend to use Smaart. Will you need a multi-channel interface to field many measurement microphones, or will a 2 channel I-O suffice? Does your rig need to be rugged and deployable, or will it be one that sits on a workbench? We cannot spend your money for you; so before you go out and purchase the most brand new, fancy-pants, acclaimed, or expensive equipment you must determine your application tolerances and operation priorities.

Fielding a Smaart rig goes beyond connecting the gear, pressing play and getting squiggly lines. For example, steps must be taken to plot calibrated levels or to transfer a license from one computer to another. Suffice it to say our goal here is twofold. Not only will we discuss measurement products and specifications, we'll also cover various peripheral equipment and example Smaart rig setups.

1. Choosing a Computer

Smaart software is available for both Mac and PC operating systems. The Smaart software is identical in every functional way regardless of operating system. Any differences are insignificant (hot-keys, menu bar locations, etc) and have no effect on the data generated by Smaart. Full versions of Smaart will not run on Windows RT or Apple iOS.

Before we dive into this discussion, let us note that if you are trying to determine whether or not Smaart can perform to your satisfaction on an existing computer, the simplest way to answer that question definitively is to download the 30-day demo and try it. Even if you are planning to purchase a multi-channel input device but haven't bought it yet (or the one you already own is on a truck somewhere in Kansas) you can simulate the load of multiple live measurements by simply setting up multiple measurements using the same sound card and running them all at once.

When considering a new computer for use with Smaart, there are two primary factors to consider: what amount of processing power will be needed, and which form factor will make the most sense for you. Oh, and also cost; three primary factors to consider…

Processing power

Smaart software will operate on lower-spec computers than what we recommend below, however more powerful computers will allow for more demanding measurement sessions. When purchasing new it is always a good idea to over-spec your CPU, GPU, and RAM in order to maximize the useful lifespan of the computer.

If you are planning on purchasing a "one size fits all" computer, it is important to consider the processing demands of other applications you wish to run. For example, a CPU intensive Digital Audio Workstation (DAW) or graphics intensive photo-editing client may require specifications that exceed the requirements for use with Smaart.

Recommended Requirements:

For smaller measurement setups, such as 2 RTA and a singlelive Transfer Function measurement running simultaneously (à la Smaart v7 Di, or older versions of Smaart) a computer that meets our minimum recommended system requirements will be more than sufficient with Smaart's default settings. In fact, the recommended requirements detailed below should be adequate for as many as five simultaneous transfer function measurements at default settings. After that you may notice your computer beginning to slow down. If you need to run more simultaneous live measurements (8+), you will need a more powerful computer.

Windows:

Operating System: Microsoft XP or newer (32 & 64 – bit)
CPU: 2 GHz Dual-Core Intel Processor or faster (or compatible)
RAM: 2 GB or greater
Video: Graphics processor with 128 M dedicated video RAM, minimum 1024 x 600 pixel display
Sound Hardware: Audio Hardware with OS compatible ASIO, Wav/WDM drivers.

Macintosh:

Operating System: Mac OSX or newer (32 & 64 – bit)
CPU: 2 GHz Dual-Core Intel Processor or faster (or compatible)
RAM: 2 GB or greater
Video: Graphics processor with 128 M dedicated video RAM, minimum 1024 x 600 pixel display
Sound Hardware: Audio Hardware with OS compatible CoreAudio drivers.

CPU

The computer's Central Processing Unit (CPU) is responsible for number crunching, audio streaming, memory management and housekeeping functions – basically everything that isn't graphics processing. Smaart is optimized for use with Intel processors. Computers with AMD processors will also run Smaart just fine, however we recommended that you over-spec those machines. If you have the option, consider going with a quad-core (or greater). Smaart is a multi-threaded application that can make very effective use of multi-core processor architectures.

RAM

Random Access Memory (RAM) is where applications like Smaart live when they're running and where your computer stores data that it needs to access very quickly. Smaart's memory requirements really aren't as crazy as they used to be, but that's mainly because everything *else* is using more RAM these days. If you run other applications alongside Smaart you can still run out of physical RAM fairly easily. When that happens, the OS will start using disk space to supplement your available RAM, which can slow your system to a crawl. The good news is, if you're not already maxed out, adding RAM is one of the easiest and least expensive ways to upgrade a computer. If you're spec'ing a new machine, consider 8 gigabytes as a target just to give yourself a little headroom and some room to grow.

Video (GPU)

The Graphics Processing Unit (GPU) renders data generated by the CPU into images for display. Smaart is a CPU *and* graphics intensive program, especially in situations with many active live plots. It is not uncommon, particularly on laptop computers with chipset or on-chip graphics coprocessors or lighter-duty mobile GPUs, to find yourself in a situation where your CPU can generate more data than the graphics hardware can plot. Another potential bottleneck is video memory access. When the GPU and CPU share the same memory bus, which tends to a common case on laptop computers, memory access can become congested when both processors become very busy.

The good news is that on-chip graphics controllers on some of Intel's higher-end CPUs (e.g., Core i3, i5 and i7) appear to benchmark pretty favorably in comparison to the current crop of mobile class GPUs, both of which significantly outperform previous generations of motherboard chipset graphics and "mobile" class GPUs. But the current crop of GPUs found on desktop graphic cards and "gamer" class laptops can still run circles around all of the above.

Best case for graphics processing is a desktop-grade GPU with dedicated video RAM. Next best would be newer integrated controller or mobile class GPU using a shared system memory partition. This is not to say that if your current laptop has an older chipset graphics controller, it might not still do a respectable job of running Smaart up to a point. Just don't expect miracles.

Form Factor

Laptop or Desktop?

Once upon a time you almost had to make a choice: If you wanted power you bought a desktop, and if you needed portability you bought a laptop. These days the boundaries between portability and performance aren't as clear. Many new laptops meet or exceed similarly priced stock desktop machines. And given that it's typically easier to bring the computer to where the sound system is than vice versa, it isn't surprising that laptop computers are the most common type used with Smaart.

Each year, laptops get thinner, more efficient, better, faster, stronger, and so on. But while laptops are becoming sleeker, in some cases it has been at the expense of wired connectivity. So when shopping for a new laptop purchase, consider what types of connections you intend to make. For example, not all laptops come with FireWire or RJ45 Ethernet ports as standard equipment. Some don't even have USB ports. This is problematic if you want to run Smaart because we need to get audio data in and out of the machine; and for now, USB, Firewire (also called IEEE 1394) and wired Ethernet are really the only games in town.

Desktop and rack mount computers offer the most flexibility in terms of hardware extensibility and connectivity. They're also big and heavy and won't fit in your backpack (unless we're talking Mac mini's, in which case you still need a keyboard, mouse and monitor) and require someplace to plug in all their power cables. But if you really need a lot of computing power there's almost no limit to how far you can go in loading them up with goodies.

Tablet PCs and iPads

Smaart will run just fine on Windows tablet computers that meet our minimum hardware requirements for desktop and laptop computers. There is currently no version of Smaart or Smaart Di that runs on Windows RT, Apple iOS or Android. The Smaart Tools app for iOS published by Studio Six Digital can provide iPad and even iPhone users with a useful subset of Smaart's single-channel measurement capabilities, but by no stretch of the imagination does it represent a complete replacement for our Windows or MacOS applications.

Screens and Displays

Smaart v7 and Di both utilize a single-window user interface design. You only need one monitor to view all of the information that Smaart can show you. With that said, Smaart can potentially present you with a *lot* of information. If you typically run other applications alongside Smaart, an additional monitor or two might still be a nice-to-have, so that you don't have to keep switching from window to window as you work and everything you need to keep an eye on is visible all the time.

2. Choosing an Audio I-O

Before we can start analyzing signals from microphones or other analog sources in Smaart, we have to turn them into digital signals. This primarily requires an A/D converter to turn voltages into numbers and a D/A to turn numbers into voltages for output (also called an ADAC if one device does both jobs). Since all of the measurement quality microphones that we know of are condenser mic's and most of the ones we commonly use are small diaphragm types, we probably also need to provide preamplification and phantom power to get the signal from a microphone into the A/D at a useful level.

Any of these functions could conceivably live almost anywhere. Most computers have a built-in stereo sound chip that could be used in conjunction with an external mic preamp or a battery powered mic. In networked digital sound systems, preamp and A/D functions may be located anywhere on the network and served up to clients over Ethernet. But most commonly in our world, all of these functions are packaged into a device with two or more mic/line level analog input and output channels that connects to the computer via USB or Firewire (IEEE 1394). We generically refer this type of everything-in-one-box solution as an Audio I-O (also called a recording interface or sometimes, anachronistically, a sound card).

If you are purchasing a new Audio I-O, the proper choice is largely defined by your intended usage; what type systems are going to be measured, under what conditions, and whether or how the rig is going to be transported. No one solution works for every user and application. Often it may be desirable to have a basic 2-channel I-O for simpler tasks and another with more input channels for larger systems and more complex jobs requiring more measurement positions.

General Audio I-O Requirements

In general, the minimum requirements for making two-channel (transfer function and impulse response) acoustical measurements in Smaart are at least one balanced, Mic level XLR input with phantom power and one line level input. A line level output for generating test signals is technically optional but normally required in practice. Fielding multiple microphones obviously requires additional mic inputs.

Microphone Inputs

General requirements for microphone inputs include:

- Sufficient gain to boost your least sensitive microphone to a useful level for the A/D converter – at least 40 to 60 dB of gain is pretty good rule of thumb.
- A maximum input level capable of accepting the output of your most sensitive microphone without overloading, when measuring the loudest sounds that you will need to measure. A maximum input capability of at least +6dBu should be fine for general use, but high sensitivity microphones that also have high SPL capabilities – like the Earthworks S30 or some large diaphragm condenser mic's– may require a +10dBu if used at high SPL.
- A phantom power supply – ideally switchable for each input – that is capable of feeding your most power hungry microphone(s).
- Low noise. EIN (equivalent input noise) at maximum gain ideally should be <120 dBu un-weighted, or as close that figure as you can get.

Line Inputs

- Line level inputs should ideally be balanced and have a maximum input capability (at minimum gain setting) of at least +22dBu. This will allow you to accept balanced analog line level reference signals from professional audio sources without clipping in show conditions. *Note that many home studio or MI market products assume a -10dBV line level standard and may require an attenuator when used with proaudio equipment.*
- Line inputs should provide adjustable gain up to about +30dB or better.

Analog-to-Digital (A/D) Converters

- Although 16-bit A/D converters will work OK with Smaart, there's no reason to consider purchasing any device (if you could even find one) that can't do 24 bits/sample.
- Smaart requires a sample rate of at least 44.1k samples/sec for multi-time-window (MTW) transfer function measurements, and the additional high frequency range extension that you get with a 48k sample rate is a definite nice-to-have. Higher sample rates such as 88.2k or 96k can sometimes be useful for testing electronic equipment and other lab/bench work but are generally not required for acoustic measurement.
- 20Hz - 20kHz overall dynamic range should be on the order of 110dB (A-weighted) or better.
- Overall (S:N) ratio of 95 - 105dB would be a reasonable expectation.

A Note on Audio Drivers

On Windows systems, Smaart can see any audio device that Windows recognizes as a valid WDM/WAV or ASIO device. On Mac OS, CoreAudio is the only game in town. Simple two-channel USB I-O devices that conform to the USB Audio Class specification may be truly "plug-n-play," utilizing device drivers built into both operating systems. Other devices, particularly multi-channel (>2) and those whose settings can be remotely controlled via software, may require device drivers and/or control software to be installed before they can be used with Smaart.

Gain Reporting

While not a requirement as such, an input device whose input sensitivity and current input gain settings are known to Smaart enables you to calibrate for sound level measurement (SPL or LEQ) given just the microphone sensitivity, and maintain calibration if the mic gain is changed. Support for this capability was introduced in version Smaart 7.3. As of the publishing of this document, the only device currently capable of reporting its input gain to Smaart is the Smaart I-O, however we expect more devices to join this list over time.

3. Measurement Microphones

Measurement microphones are our window into the acoustical environment. They come in various shapes and sizes and a wide range of price points. The key to deciding which mic is right for you, lies in knowing what you are using it for, or more specifically: how it is intended to be used, and what sort of tolerances you need.

It is a relatively simple and inexpensive proposition nowadays to produce an omnidirectional condenser microphone with respectably flat frequency response between 50Hz and 5kHz, reasonably flat response from 20 Hz -18 kHz, and dynamic range that is generally usable for measurements between 30-130 dB SPL. And for a large number of applications, a relatively inexpensive microphone meeting that general description may be all you really need for your Smaart rig.

The cost of a measurement microphone starts to increase when you:

- Expand the flattest part of its response curve into higher frequencies
- Extend its dynamic range, byincreasing max SPL or decreasing self-noise
- Tighten tolerances for overall frequency response
- Provide individually-measured frequency response data for each microphone
- Require precisely matched response between two or more microphones
- Increase the useful operating temperature range
- Ruggedize, moisture-proof or otherwiseextend environmental capabilities

All of this to say that you can spend HUGE amounts of money on measurement microphones if you like, and for some applications one might actually need to. But not every application requires a terribly expensive measurement microphone.

For field measurement applications in general, environmental factors such as background noise and wind can limit the repeatability of measurements to the point that the difference between a pretty good measurement mic and a really excellent one can easily be lost. Consider also that the environments in which sound engineers often work in can be chaotic and fraught with danger for microphones, and losing modestly priced workhorse mic to theft or accident may be a lot easier to cope with than the loss of a prized, laboratory quality instrument.

Omnidirectional Microphone Types: Free field, Diffuse Field, Pressure Field

Omnidirectional microphones are chosen for acoustical measurements because we are interested in both direct and diffuse sound. Intuitively, one might expect "omnidirectional" to mean a microphone that is equally sensitive in all directions across the full audio spectrum, but in practice even the very best omnidirectional measurement microphones exhibit some directional characteristics. This is especially true at higher frequencies, where the diameter of the diaphragm startsto become significant with respect to wavelength.

The chart on the following page compares the directional response of two mid-priced measurement microphones to an ACO Pacific reference mic. The Beyerdynamic MM1 is a *Diffuse field* omni (sometimes called a *Random Incidence* mic) optimized for smooth off-axis response. This type of microphonewill exhibit some build-up at high frequencies when pointed directly at a discrete sound source such as a loudspeaker.

The iSEMcon EMX-7150 is a *Free field* omni, designed to be pointed directly at whatever you're measuring. It rolls off in the high end at 90° off axis.

Measurement Microphone Comparison
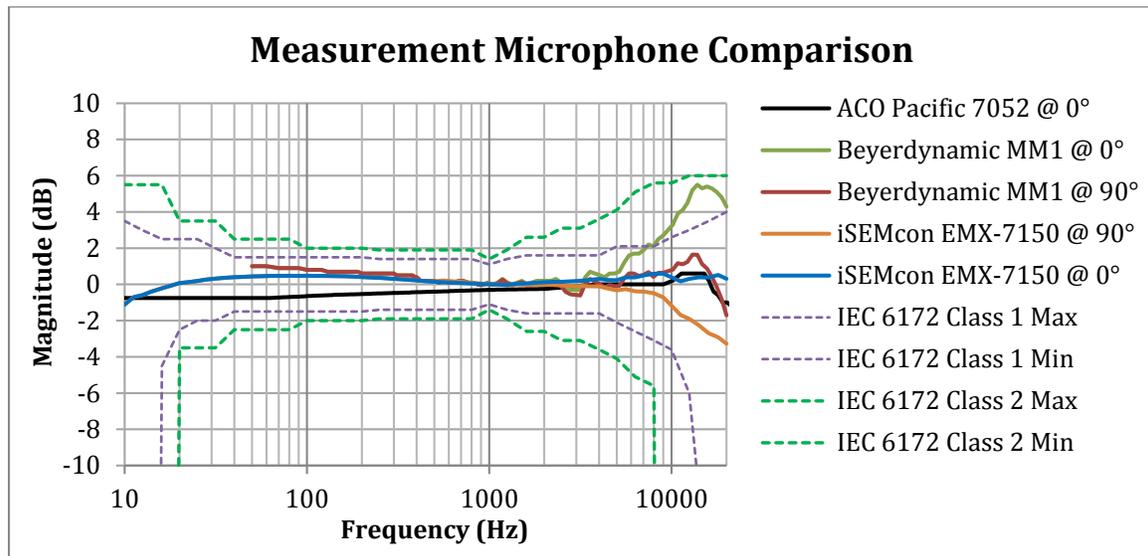
In technical terms, *Free field* refers to an environment where sound from a given source is free to radiate outward forever, without encountering any reflective surfaces. *Diffuse field* ideally means an environment where sounds can arrive from any direction with equal probability, such that it's impossible to identify any single source. A third omnidirectional microphone type called *Pressure field,* is optimized for use in very small enclosed spaces, in acoustical couplers or at boundary surfaces.

In *practical* terms, we most often find ourselves measuring sound from loudspeakers in spaces where our ears can at least tell which direction the sound is coming from. And generally speaking, unless you are measuring from within a few inches or centimeters of a reflecting surface (and if so, why?), interference from reflections tends to be concentrated most in the frequency ranges where smaller diaphragm omni mic's become almost ideally omnidirectional, regardless of type. The main difference between the three basic types in that case, is simply their on-axis vs. off-axis high frequency response characteristics.

For best HF accuracy:

- Use a free field microphone on-axis for predominantly direct sound measurements of loudspeakers and other discrete sound sources.

- Use a diffuse field response microphone, typicallypointed straight up, for diffuse reverb measurements or background noise surveys, where no single sound source dominates the noise field. Consult the manufacturer's documentation for specific recommendations regarding the optimal angle of incidence for direct sound (free field) measurements.

- A pressure field microphone may also be used successfully for free field measurements if positioned at 90° to the source axis, so that a pressure wave from the sound source passes in front of the diaphragm, rather than striking it head-on.

Calibration Files

Note that if the manufacturer provides individually-measured frequency response data for your microphone, another way to improve the overall accuracy of spectral and transfer function magnitude measurements in Smaart is to use a calibration file. This enables Smaart to use the microphone's measured response to compensate for any lumps and bumps in its magnitude response curve. Please refer to the Smaart help file for more information on creating and applying calibration files.

Frequency Response and IEC 61672 "Types" (1 or 2)

A term you often hear in connection with measurement microphones is the word "Class," followed by the number 1 or 2. These refer to tolerances for sound level meters (SLMs) and measurement microphones laid out in IEC 61672 (and other derivative or harmonized national standards). Type 1 denotes a precision instrument suitable for laboratory use and critical field measurement applications. Type 2 is a general purpose field instrument.

We mainly concern ourselves with frequency response however the type designation also takes other factors into account, such as temperature and barometric pressure ranges and long-term stability. To say outright that a microphone or SLM is Class 1 or Class 2 means that it satisfies *all* applicable requirements for that designation. This adds significantly to both the design requirements and production costs and it is generally not something you get with lower and mid-priced microphones.

So, is it worth it to spend the extra money on a fully Class-compliant microphone? One reason to consider going this route might be if you need a mic guaranteed to hold within fairly small variances its nominal frequency response across a relatively extreme temperature range, e.g., from -10° to 50° C (14° - 122° F) for Class 1. However if you normally work in venues where an audience would find it comfortable to take in a show, then perhaps operating temperature range isn't a huge concern. Another reason to consider a fully class compliant microphone might be if you planned on using it with a standards-compliant sound level meter (SLM), which Smaart is not.

The DSP processes for sound level measurements in Smaart were actually designed to the old Class 0 (laboratory instrument) specifications. And given a high-quality mic and preamp, when properly calibrated, Smaart can provide sound level measurement data that rivals or exceeds the accuracy of many SLMs. Unfortunately there is no provision in current standards for certifying DSP processes independently of data acquisition hardware. If you are required to produce official reports, wherein standards traceability for your measurement equipment is required, for example certifying compliance with noise regulations, then a dedicated SLM or other standardized metering system is likely still your best option.

Max SPL, Self-Noise and Sensitivity

In addition to frequency response, the microphone specifications that we tend to pay the most attention to are max sound pressure level (SPL), self-noise and sensitivity. As you might guess, the more you are willing to pay for a microphone, the better you can do with all of the above. But in practice, lower spec'd microphones can be perfectly suitable for field measurements and daily alignment work. In general, the most affordable measurement-grade microphones that we have seen are small

diaphragm condenser mics with metalized PET film diaphragms. We will focus on the relative strengths and weaknesses of those.

Max Sound Pressure Level (SPL)

Small diaphragm condenser microphones naturally tend to have max SPL figures ranging from pretty good to very good, by virtue of being small. The smaller the surface area of the diaphragm, the stiffer it tends to be. The stiffer the diaphragm, the more energy it takes to move it. At the lower end of the range you mayseeMax SPL figures like 110-115 dBA, that a really loud amplified musical performance mightchallenge, but 130-ish or better for <3% THD is pretty typical. Keep in mind that the threshold of pain in human hearing is around 130 dBA. So unless you often find yourself measuring gunshots, jackhammers or jet engines from very close distances, it's debatable whether you'll really need the 150-160 Max SPL that you can get with higher-priced metal diaphragm measurement microphones.

Self-Noise

Another factor that could occasionally come into play is that small diaphragm condenser mics tend to have higher self-noise than their larger diaphragm cousins. Broadband self-noise figures of 30 dBA or higher are typical for 0.25" (~6mm) microphones. This is rarely an issue for most Smaart users, but it if you are called upon to measure something very quiet, such as background noise levels in a recording studio or a *very* quiet auditorium; it's conceivable that if could be. If your normal operating environment tends more toward spaces that are at least as noisy as a typical cinema room, chances are good that a small diaphragm mic will work just fine for you.

Sensitivity

Sensitivity is the amount of electricity your microphone generates for a given amount of sound. It is typically quoted in millivolts per Pascal (e.g., 6 mv/Pa) or sometimes in dBV, referenced to 1 volt per Pascal. The latter is just a decibel representation of the former. For example 15 mV/Pa is equivalent to -36.5 dBV ($20 \cdot \mathrm{Log}10(0.015) \approx 36.5$). In SPL terms, one Pascal is equal to 94 dB, so you can actually measure a microphone's sensitivity yourself with a standard sound level calibrator and a volt meter.

So why do we care? Well for one thing, the lower the microphone sensitivity, the more you may need to boost its signal to reach a useful level on you're A/D converter. If your microphone's sensitivity is very low and your preamp doesn't provide sufficient gain to get the signal level to where you need it to be, that could be an issue. Something else to watch for is unusually high sensitivity. Typical sensitivities for small diaphragm mics tend to run in the 5 – 15 mV/Pa range. If you see a figure significantly higher than that for a small diaphragm mic, the trade-off *might* be higher power consumption, which could be an issue when the mic is used with wireless transmitters or anything else that runs on batteries.

IEC 61094-4 Measurement Microphone Types

When looking at higher-end measurement microphones, another cryptic term you may encounter is an IEC 61094-4 *Type* designation, consisting of the letters "WS", followed by a number a letter. These are pretty simple to decipher. "WS" stands for

Working Standard. The number and letter that follow refer to the size of the capsule and the pickup pattern. The number 1 designates a 1" (25 mm) diameter capsule, 2 stands for 0.5" (12 mm), and 3 means 0.25" (6 mm). The letters F, D, or P stand for free field, diffuse field or pressure field respectively.

Compliance with IEC 61094 assures (among other things) interoperability between microphone capsules and sound level calibrators from different manufacturers. Lower priced measurement microphones often have odd-sized tips that require an adapter (hopefully supplied by the manufacturer) to fit a standard sized acoustical calibrator opening.

4. Sound Level Calibrators

Although Smaart may not be certifiable as a standards-compliant sound level meter (unless perhaps you sent your entire Smaart rig to a lab to be tested as a "complete instrument"), Smaart can still provide you with often useful and really quite accurate SPL and LEQ measurement data when properly calibrated. Calibration can be accomplished by using a preamp and microphone with known sensitivity or by using a *sound level calibrator* to measure the sensitivity of the entire input chain, from the microphone capsule to the A/D converter.

A sound level calibrator works by acoustically coupling the capsule of your microphone to a small pressure chamber –an airtight fit with the microphone capsule is required – and then generating a calibrated test tone at a known sound pressure level inside the chamber. There are two basic types (electronic or piston-phone) and three standardized classes of accuracy (Classes 1, 2 and LS). Electronic acoustical calibrators use an oscillator circuit and transducer to generate a test tone. Piston-phone calibrators use a motor and piston(s) to create an acoustical sine wave mechanically.

The IEC/ANSI Classes mainly refer to frequency and level tolerances, distortion characteristics, operating temperature and humidity ranges, etc. Class LS (laboratory standard) calibrators are intended for lab use and are seldom seen by most humans. Class 1 and Class 2 devices have output level tolerances of +/− 0.4 dB and +/− 0.75 dB respectively. Ideally, the class rating for the calibrator should match the microphone or sound level meter (SLM). If you calibrate a Class 1 SLM using a Class 2 calibrator or vice versa, the lower rated instrument determines the presumed accuracy of the measurement.

The most common output level for all types of calibrators is 94 dB, which approximately equals one Pascal. Often, on electronic calibrators you may also see a +10 or +20 dB setting that can be handy for calibrating in very noisy environments, as long as it doesn't crowd the max SPL rating for your microphone. Most electronic calibrators put out a 1 kHz tone, which happens to be the frequency where the A weighting curve crosses unity gain and falls well into the 0 dB portion of the C curve. Piston-phone calibrators typically operate at a lower frequency such as 250 Hz, meaning you need to ensure that weighting filters are bypassed during calibration.

Some, but certainly not all electronic acoustical calibrators have a built-in pressure sensing microphone and feedback circuit that automatically adjusts their output level to compensate for changes in air pressure. The rest, along with piston-phone

calibrators, require the user to adjust their nominal output level based on an altitude chart or a barometer reading. If a barometer is strictly required, it *should* be included with the calibrator.

Sound level calibrators, like microphones, come in many shapes and sizes and a wide range of prices – one can spend anywhere from a couple of hundred to a few thousand US dollars for one. There tends to be a pretty large price differential between Class 1 and Class 2 calibrators. Also, the term "Class 2" sometimes gets tossed around pretty lightly as you get into the lowest price points, so it's important to know what you're buying. A fully class compliant calibrator will always come with a certificate of calibration, ensuring that it meets all standardized requirements for its classification (and it properly needs to be retested periodically by a test lab to fully maintain that rating).

As with microphones, full compliance testing adds significantly to production costs and it's something you may not get with lower priced products. Occasionally you will see two different prices for the same model; one with, and one without a lab calibration certificate. If you don't see any mention of calibration and it's a lower-cost device, then you probably want to assume that only a less stringent production line test of some kind was done. That doesn't necessarily mean that it might not still be "close enough" for your own purposes, but that's something you would want to confirm. We have seen inexpensive calibrators differ by as much as +/− 1.5 – 2 dB from a Class 1 reference device and as much as 2-3 dB from each other, when measuring several units of the same make and model.

5. Peripheral Equipment (Cables, Testers, and Other Audio Doohickies)

Take a look at any professional system engineers' workbox and you'll see all sorts of specialized gear, adapters, and assorted items put together through years of experience and necessity. A quality kit will enable you to:

1. Swiftly interface computer sound devices and interconnect professional audio equipment.
2. Field test and repair your cabling and, if applicable, perform maintenance on certain components.
3. Utility items such as tape of varying type and color (most common being electrical, gaffers, and spike tape), ground lifts and isolators, and a digital multi-meter.

Audio systems are becoming increasingly more networked. Lightweight cables transmitting digitalized audio streams are replacing the heavy copper snakes formerly used for those same tasks. BNC and RJ45 (and USB) connectors are now a staple in pro-audio. This makes it essential to have a cable tester with traditional, as well as digital I-O. For most situations, pass/fail diagnostic capacity is built into network control hardware and software monitoring - simply put, if a cable is failing, your network will not initialize. However, having a modern cable tester at the ready for checking your handiwork is a convenience that will prove to be worth its weight in gold. Luckily, a decent tester with digital I-O can be had for about USD $35-50.

The consummate professional may also consider convenience items such as a custom rack-mounted Keurig®or self-regulating cigar humidor; these items however are optional. Specialty mic clamps, pouches, accessories and cable bundles can be found on the Rational Acoustics web store.

# 6. Typical Measurement Rig Set-Ups

Below are some example Smaart measurement system setup diagrams. Basic connections are pretty much the same for any I-O you may encounter. We trust that you understand the audio cabling necessary connect your own equipment together.

Stereo 2x2 Audio I-O

A common solution for quick-deployment and ultra-portability employs a basic 2 in, 2 out audio I-O, such as the Smaart I-O. A 2x2 I-O is capable of performing the bulk of measurement fieldwork wherein Smaart's multi-channel capabilities are not required.

Single Channel

Although dual-channel measurements are preferred for system alignment work, single channel analysis (RTA, Spectrograph, or SPL) remains extremely useful for a for a variety of tasks including feedback elimination, lobe studies, SPL/LEQ measurements, input calibration or any other application where you are trying to determine what a signal is, and not what happened to it as it traversed a system under test. For any of the above, as well as impulse response measurements utilizing an external impulsive signal source (e.g., a balloon pop), this is as complicated as your setup need be.
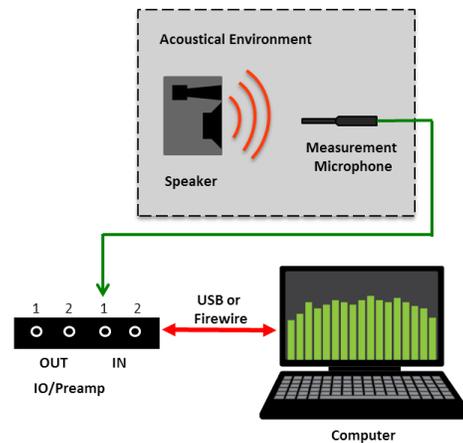
Dual Channel

Transfer Function, (TF) or system response measurements are the result of comparing a reference signal (pre system) and a measurement signal (post system). This measurement type is essential for aligning loudspeaker systems, as unlike RTA measurements, TF measurements are not "time-blind."This makes it possible to display Phase (time), Coherence (stability), and Magnitude (response).
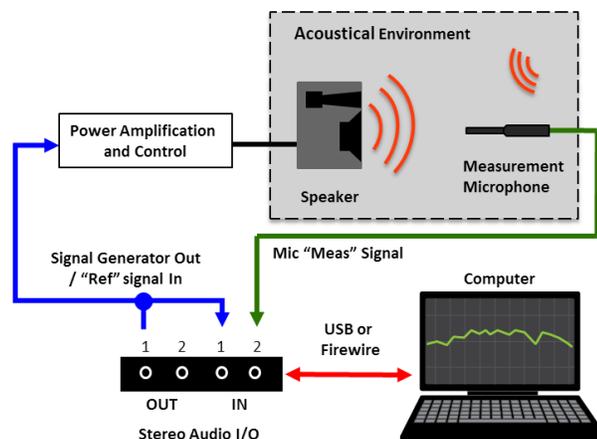
In this example, Smaart's internal Signal Generator is used to excite the system. The output of the generator

is routed toinput 1 of theI-O. Using a Y-split, the signal is physically loopedback (known as a hard-wired loop-back) into an input channel on the I-O. This input is then assigned to the Reference signal of a transfer functionmeasurement.

NOTE: While Smaart has the ability to reference transfer function measurements internally, by selecting the signal generator as your reference signal source, it is generally not a recommended practice. This often results in erroneous delay times as the latency from your I-O and the latency from the generator will not match. Some computers handle this better than others, but on the whole it is a better practice to use a hard wired loop back so that all of your signals are on the same clock.

Multi-channel I-O

In version 7, Smaart introduced the capability to run and view as many simultaneous real-time Spectrum and Transfer Function measurement – as well as live averages of individual measurements – as your computer can handle. Having multi-channel I-O capability enables you toset up multiple microphones to compare different measurement positions without a lot of running around. This can be a huge advantage when working with larger, more complex systems.

Traditional USB/FireWire Stand-Alone Interfaces

Eight-channel, single rack space packages are common for multi-channelaudio interfaces. The single rack space format makes portabilityin a rack case, or in your backpack, equally viable. In addition to fielding more microphones, having more inputs enables you to tapinto electronic measurement points as well.
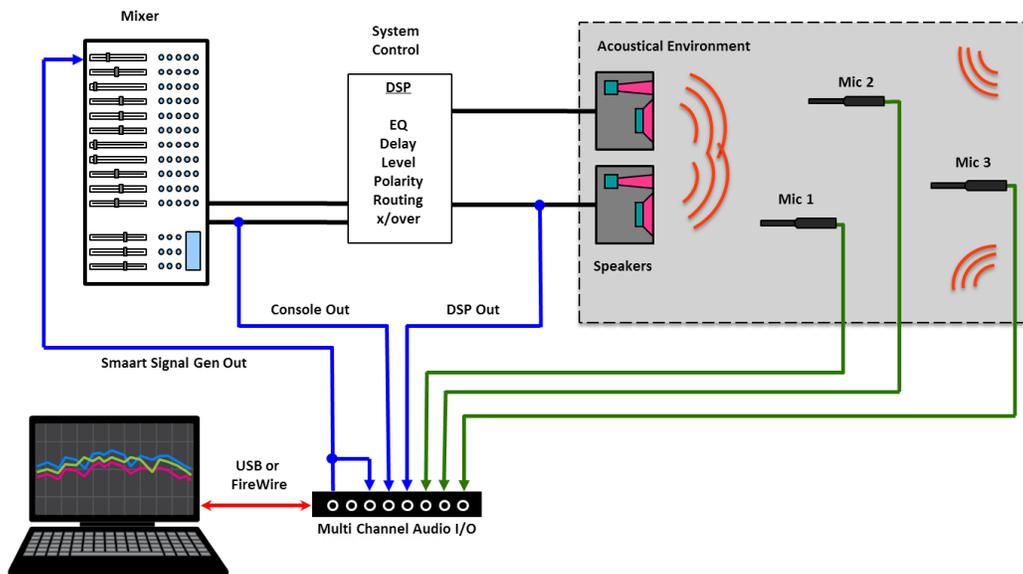
Above, we have three measurement microphones as well as electronic measurement points pre-system processing, pre-speaker system, and a hard-wired loopback pre-console. In this example, the Console Out and DSP Out can be used as reference signals that let you isolate the response of the system post-console or post-DSP, or as measurement signals, e.g., for graphing an EQ response in Smaart.

Network I-O

The ability to fully digitize your audio stream from the stage pre-amp, all the way to the speaker system has become a reality as of late. In some professional systems nowadays, the only analog signals present are those from the microphones to the stage conversion box; where they are digitized and distributed via the digital snake.
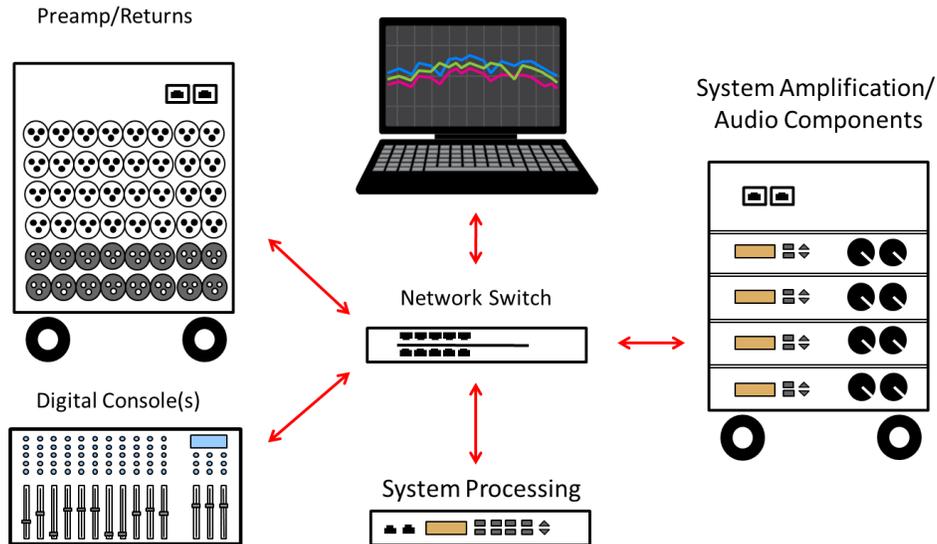
Dante Controller

Using Dante Network Protocol as an example, in a setup such as this the computer operating Smaart sees Dante Virtual Soundcard as an audio I-O device. All routing for the soundcard is done within Dante's Controller application. The computer running Dante Controller (in this example is also running or hosting the Smaart application) has the ability to see any signal connected in the network. With the ability to analyze any signal on the network, this can provide for some extremelyflexiblemeasurement configuration possibilities.