Choosing Gear for your Smaart Measurement System

rational acoustics

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Introduction

Smaart® is a software-based audio analyzer. Unlike hardware-based, turnkey measurement systems, Smaart can be used with a wide variety of hardware options to suit a range of measurement applications. There is no one-size-fits-all Smaart rig – each user must configure a system that suits their individual needs and budget.

This document is intended as a guide to help users better understand some of the relative merits and trade-offs associated with the many choices for components of a Smaart-based measurement system. This will help you make cost-effective choices when selecting specific equipment for a particular application.

A complete Smaart measurement system consists of three main components:

1. Host Computer *(Signal Processing)*
2. Audio Interface *(Signal Acquisition)*
3. Measurement Microphone(s) – *(Acoustical Probe)*

For each of these components, *what you need* is a function of how you use (or intend to use) Smaart. Does the rig need to be rugged and quickly deployable enough for touring, or will it sit on a workbench? Does the hardware need to be rack mountable, or fit in a backpack? Give some thought to your needs and priorities in order to make wise gear investments and end up with the right tools for the job. This document will explore some considerations in each of the above categories.

For more information on how to operate Smaart software, measurement theory, and application, see the Smaart v8 User Guide.

Choosing A Computer

Smaart software is available for both macOS and Windows operating systems. The software is functionally identical on both platforms. Full versions of Smaart will not currently run on mobile device operating systems.

If you already have a computer and are not sure whether it can run Smaart to your satisfaction, **the best way to know for sure is to download the demo version of the software and try it out**. Learn more about the demo and access downloads here.

Some additional considerations:

- When selecting a new computer for use with Smaart, keep in mind that Smaart will run comfortably on virtually all modern “office” and “workstation” computers, although more powerful computers will allow for more demanding measurement sessions with more simultaneous measurements.
- Smaart can be fairly graphics intensive, particularly when graphing multiple live plots, and this can form a bottleneck on “netbooks” or other light-duty computers with basic or integrated graphics processors. Important note:
- When purchasing a new computer, it is always a good idea to over-spec CPU (Central Processing Unit, the “brain” of your computer), GPU (Graphics Processing Unit, or “video card”) and RAM (memory) in order to maximize the useful lifespan of the computer.
- We’ve detected some problems with Smaart crashing while the program is being resized on laptops running Intel HD/UHD 600 on-board graphics (and likely any other Intel series chips that use the same driver). Our recommendation is to avoid Intel on-board graphics, as the crash does not occur with AMD and NVidia graphics cards.
- If you are planning a “one size fits all” computer, it is important to consider the processing demands of other applications, especially if you plan to run multiple applications simultaneously, as is often the case for computers used in a live production environment.
- Lastly, consider how you plan to get signals into and out of Smaart – what type of connection your audio interface requires – and make sure the computer has the proper ports (or adapters).

You can view the most up to date Recommended System Requirements for current software versions on our website. If you are considering updating your computer’s OS and have concerns about compatibility, or have any questions about hardware, visit our support portal for the most current information.

Choosing an Audio Interface

An audio interface is used to convert between the analog voltage signals that flow through our sound systems and digital values that computer software can understand and analyze, using Analog to Digital converters (A/D) and Digital to Analog convertors (D/A). In other words, the audio interface is how we transport signals into and out of the computer (which is why audio interfaces are also referred to as Audio I/O).

When using a measurement microphone to capture acoustic signals, a microphone preamplifier is used to amplify the relatively low signal levels created by the microphone to a higher level before they are sent to the A/D converter. Some of the signals you want to analyze may already be in digital form, and can be connected to an audio interface via AES/EBU or S/PDIF, or via a networked audio protocol such as Dante or AVB.

When choosing an audio interface, therefore, it's important to consider *how many signals* and *what type of signals* you want to be able to input simultaneously. At a bare minimum, for making typical two-channel (transfer function) acoustic measurements in Smaart, you will need an interface with at least one mic input (XLR connector with preamp) and one line-level input (typically TRS). If you plan to use multiple microphones to measure from multiple locations simultaneously, you will want a larger interface with more mic inputs – common options include 4-mic and 8-mic interfaces. Since measurement microphones are condenser microphones, the mic preamps must offer Phantom power (+48V) capability.

The good news is that almost all modern professional or "music industry (MI)" grade audio interfaces are of suitable quality for use with Smaart and will support base sample rates of 44.1 kHz and 48 kHz. Higher sample rates such as 96 kHz are not required for acoustic measurement but can offer some benefit when testing electronic equipment, as they allow the analyzer to "see" higher frequencies. The Rational Acoustics webstore offers a selection of recommended audio interfaces that we've found work well for Smaart measurement. If you have questions or concerns about whether a particular interface is right for your needs, please contact support via our support portal.

One special consideration exists when choosing audio interfaces and measurement microphones for SPL measurement at concert levels: the mic preamps on the interface must be able to accommodate the signal voltages produced by your measurement microphone at high acoustic levels. For more information on this, see "Special Considerations for SPL Measurement" below.

Audio Drivers

On macOS machines, Smaart will be able to see and utilize any audio device that the computer recognizes as a valid CoreAudio device. On Windows, Smaart can see and utilize any audio device that Windows recognizes as a valid WDM/Wave or ASIO device. Simple two-channel USB devices may be "plug and play." Multi-channel devices offering more than two inputs require ASIO drivers to be recognized by Windows (and therefore seen by Smaart).

The rule of thumb is: if your operating system can see the audio device, Smaart will be able to as well. Therefore, it's important to make sure that your device is connected, powered on, and recognized by your operating system before launching Smaart.

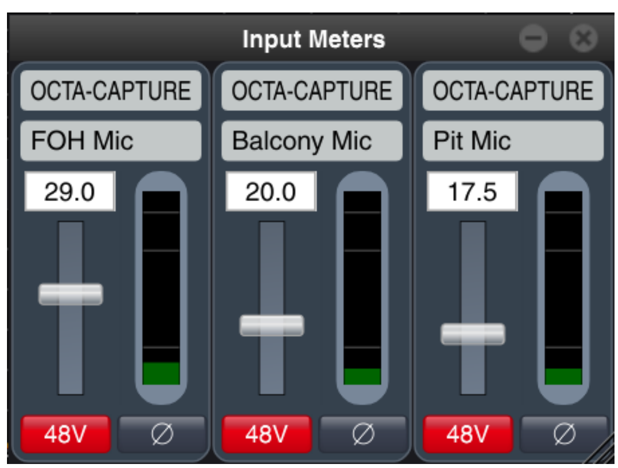
Rational Acoustics does not maintain a list of compatible audio interfaces for use with Smaart. In general, if the device you are considering is still supported by its manufacturer, it will still be supported on current operating systems and therefore compatible with Smaart.

Remote Control and Gain Reporting

Some interfaces have the ability to report their preamp gain levels to the computer, and have their gain settings controlled remotely via software.

In particular, the Roland OCTA-CAPTURE interface is an 8-preamp audio interface that has the ability for remote gain control. Smaart detects when a Roland OCTA-CAPTURE device is connected and will automatically enable user-adjustable gain and phantom power controls, allowing the preamps to be controlled directly from within Smaart.

This functionality is not required, as the gain can always be set manually using the front panel controls, and in general a measurement session that starts by setting the properly gain structure into the analyzer should proceed without issue. However, some users appreciate the convenience, particularly in a touring or install situation where the interface may be installed in a rack and not easily physically accessible to the user.



Choosing a Measurement Mic

A measurement microphone is our window into the acoustical environment. There are many models in common use with Smaart rigs, spanning a wide range of price points. To help you decide which mic is right for you, we will consider the required tolerances and desired use case.

Since measurement microphones are designed to capture an acoustic signal without coloring it, they are designed to be omnidirectional with a flat frequency response.

Even an inexpensive measurement microphone can be expected to produce a nearly-omnidirectional polar pattern and nominally flat frequency response between 50 Hz and 5 kHz, and be "reasonably" flat between 30 Hz and 18 kHz. For a large number of measurement applications, a relatively inexpensive measurement microphone that fits this general description might be all you need for your Smaart rig.

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

- Expand the flattest part of its response curve to higher and lower frequencies
- Extend its dynamic range, increasing Max SPL or decreasing self-noise, or both
- Tighten unit-to-unit tolerances for overall frequency response
- Provide individually measured frequency response data for each microphone
- Increase the useful operating temperature range
- Ruggedize, moisture-proof or otherwise extend environmental capabilities

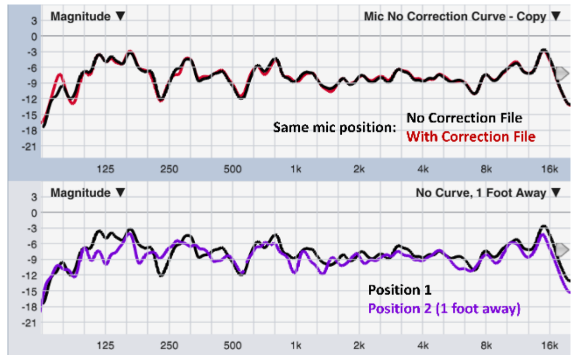
Although expensive measurement microphones are objectively "better" in some or all of the above ways, not every application may require or even benefit from the added expense.

In a typical field measurement scenario, the 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 an "excellent" measurement mic can easily be lost. Some touring engineers choose to travel with modestly priced measurement microphones to make it easier to cope with a lost or damaged microphone on the road.

Mic Correction Files

Some measurement mics include files with individually-measured frequency response data that can be loaded into Smaart to correct for deviations in the microphone's magnitude response. With mics of reputable quality, these deviations tend to be very minor throughout most of the audible range (tenths of a dB). Although this is important information for laboratory test conditions, these small deviations are dwarfed by the natural variance that occurs when measuring in acoustic spaces simply by moving the mic a short distance.

The top pane of the image at right shows a measurement taken from the same microphone position with (red) and without (black) a correction file in use, and the bottom pane shows the variance caused by moving the microphone a foot away.



For this reason, mic correction files (often referred to as "calibration files") are *not* necessary for typical field measurement work.

However, there is a benefit to buying a microphone that includes individual correction data: the confidence that the manufacturer has tested the microphone and it is performing within its design tolerances. In other words: if your mic is of high enough quality to come with a correction file, it is probably of high enough quality not to need it.

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

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 at higher frequencies, where the diameter of the diaphragm starts to become significant with respect to wavelength. Omnidirectional measurement mics can be classified as Free Field, Diffuse Field/Random Incidence, or Pressure Field based on these directional characteristics, which are generally restricted to the top octave of the response and most noticeable in the nearfield of a loudspeaker.

In practice, the difference is that free field microphones are designed to correct for this loading effect when pointed directly at the source, whereas diffuse field microphones are designed to be pointed straight up (90° off axis from the source). The most important thing is to be consistent with how you aim the microphone as you move it around the space.

Special Considerations for SPL Measurement

SPL measurement at concert levels places some unique demands on the measurement equipment, and we must consider these for optimal results.

At concert levels, the microphone is subjected to acoustic pressures several orders of magnitude higher than during a typical measurement session. As a rule of thumb, for typical live music material, the instantaneous peaks will be approximately 35 dB above the average A-weighted SPL. Therefore, the measurement microphone should be rated for a Max SPL of at least 135 dB – 140 dB to safely accommodate signal peaks without overload. (When it comes to SPL measurement, even occasional overloads are problematic as they flush averaging buffers and potentially invalidate large time ranges of log data.)

The next thing to consider after max SPL is the voltage that the mic creates at those levels. Mics with relatively high sensitivity will produce large voltages that can overload a typical mic preamp. The microphone and audio interface must be chosen to work together as a pair, by selecting an interface with a max mic input level able to accommodate the voltages created by the microphone during the event.

Mic sensitivity is usually indicated as mV / Pa, which is the output voltage at 1 Pascal (94 dB SPL). Since our target of 140 dB SPL is 200 times 94 dB SPL, you can quickly determine the mic's output voltage at 140 dB SPL by multiplying its sensitivity by 200.

For example, if your measurement mic has a sensitivity of 36 mV/Pa, then at 140 dB SPL, it will produce an output voltage of 7.2Vrms (+19 dBu), which will overload most mic preamps.

A lower sensitivity measurement microphone that we have found works very well for SPL measurement is the iSemCon EMX-7150, with a sensitivity of about 6 mV/Pa. At 140 dB SPL, that mic produces about 1.2Vrms (+3.8 dBu), which can be comfortably accommodated by most preamps.

Sound Level Calibrators

In order to measure SPL, Smaart needs to know what acoustic pressure at the mic caused the signal levels it's seeing. This can be accomplished by running the calibration routine using a sound level calibrator.

Calibration is *not* required for standard transfer function and spectrum measurements. It is *only* required for measuring SPL.

A sound level calibrator works by generating a known SPL at the diaphragm of the microphone, which Smaart can correlate with the incoming signal level, thus establishing the sensitivity of the entire signal chain, from mic to interface.

There are two basic types (electronic and piston-phone) and three standardized classes of accuracy (Classes 1, 2, and LS). Class LS calibrators are intended for lab use and are seldom seen by most humans. Class 1 devices have output tolerances of +/- 0.4 dB, and Class 2 devices have output tolerances of +/- 0.75 dB. Most calibrators generate a 1 kHz, although some produce lower frequencies such as 250 Hz.

Sound level calibrators, like microphones, are not all created equal. **A fully class compliant calibrator will always come with a certificate of calibration**, ensuring that it meets all standardized requirements for its classification. If a calibrator does not come with a certificate of calibration, all bets are off. We have seen inexpensive calibrators vary by as much as 2 dB from a Class 1 reference device, and as much as 3 dB from unit to unit. For sound exposure measurements (NIOSH dose), 3 dB represents 100% error, which is far too inaccurate to be meaningful.

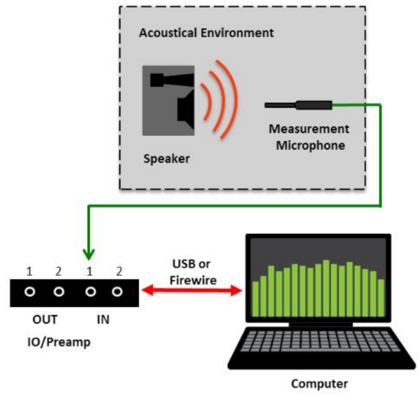
A realistic goal for SPL calibration accuracy is to be within +/- 1 dB, and that can be easily accomplished with a well-made and certified Class 2 calibrator or better. As with microphones, full compliance testing adds significantly to production costs. If you don't see any mention of calibration and 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.

Example Measurement Rig Setups

2x2 Audio Interface

A basic two-in, two-out audio interface is a common solution for quick deployment and portability, and is capable of performing the bulk of measurement field work when multi-channel and multi-microphone measurements are not required.

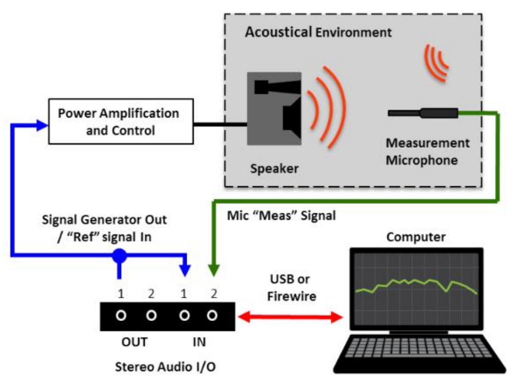
Although dual-channel measurements are preferred for system alignment work, single-channel analysis, such as the setup shown at right, remains very useful for a variety of tasks including monitoring the tonal balance of a mix with RTA or spectrograph, feedback elimination, and measuring SPL and Leq. Single-channel configurations can also be used to gather "direct" impulse response measurements using an external impulsive noise source (shot/pop/clap).



Single Channel Measurement setup

System response measurements (**Transfer Function** measurements) are comparative measurements, taken by comparing what went into the system against what came out of the system. This measurement type is essential for sound system alignment tasks, as unlike signal channel (RTA) measurements, transfer function measurements are not "time blind." Dual channel measurements can characterize the system's response both in time and frequency domains, generating Magnitude, Phase, Coherence, and Impulse Response data.

In the example at right, Smaart's internal signal generator is used to excite the system. The output is split, to drive the sound system and also physically "looped back" into an input channel, which will act as the Reference signal for the transfer funtion meausrment.



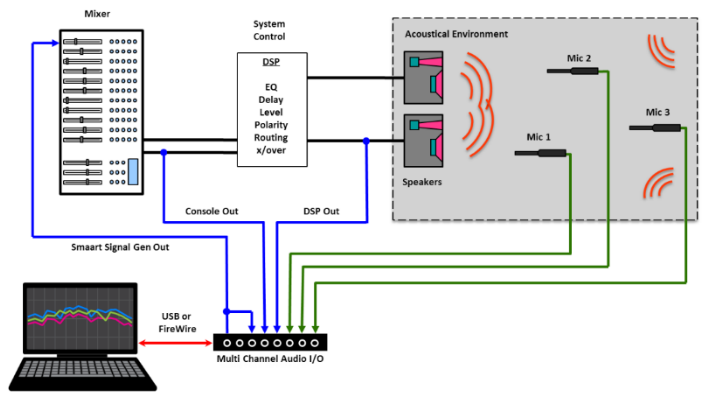
Dual Channel Measurement setup

Note: Although Smaart has the ability to internally reference the signal generator directly, ("software loopback"), this is not a recommended practice. This can result in erroneous delay times and clock drift, and in general it is recommended to use a hard-wired loopback, keeping both the measurement and reference signals on the same clock.

Multi-channel IO

Smaart can run and display as many simultaneous realtime Spectrum and Transfer Function measurements, and live averages of individual measurements, as your computer and hardware can accommodate (there is no software-imposed limitation). Multi-channel IO capability allows you to place multiple microphones around the space to compare measurements taken at different positions, which is a real time saver when working with larger, more complex systems.

In this example, we have three measurement microphones placed throughout the system's coverage area, as well as electronic measurement points pre-processing, post-processing, and a hardwired loopback.

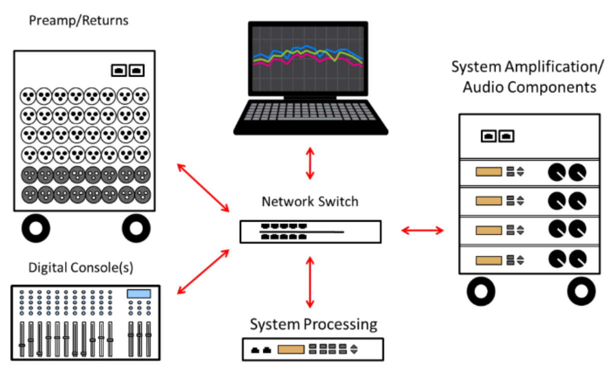


Multi-Channel Measurement Setup

This allows multiple transfer function measurements to isolate the response of various system elements. For example, a transfer function measurement between Console Out and DSP Out will allow direct measurement of the response of the system processing and EQ filters.

Network IO

The professional audio world is becoming increasingly networked – in some cases, the signal path is entirely digital from the preamp all the way to the power amplifiers. Digital signals can be routed into Smaart via a variety of methods depending on the specific digital networking protocols in use.



Networked Measurement Setup

In this example, the audio network is using the Dante network protocol, and Dante Virtual Soundcard has been installed on the computer, allowing Smaart to recognize the virtual soundcard as an audio interface. The ability to analyze any signal on the network provides extremely powerful measurement configuration possibilities.