

RATIONAL ACOUSTICS

Smart SPL

USER GUIDE

RELEASE 9.1





Rational Acoustics Smaart SPL User Guide Copyright Notice

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1 – INTRODUCTION

WHAT IS SMAART SPL?

Rational Acoustics Smaart® SPL is a software application that enables you to measure, analyze, log, and remotely monitor sound levels. Smaart SPL is designed to provide a powerful and comprehensive sound level measurement toolset while maintaining an approachable and intuitive interface.

Being a software application, Smaart SPL is hardware independent and can therefore be paired with a variety of microphones and interfaces, which allows its capabilities to scale from a single microphone logging application up to large, distributed systems that can monitor sound levels from a variety of locations using a networked audio protocol such as Audinate Dante®.

WHAT IS THIS GUIDE?

This guide is intended as a practical introduction to configuring and operating Smaart SPL. It will provide an explanation of the program's features and operation. As many of the topics surrounding sound level measurement are commonly misunderstood, this guide will also provide some background on the core concepts that are critical to making effective use of Smaart SPL's powerful toolset.

This guide is organized so that it can be read from start to finish. We have attempted to provide information about program features and options organically and in context, so if you are searching for information on a specific function, the table of contents and index will help you track things down. In addition, a full text search can be done if you are reading an electronic copy of this document.

You are also encouraged to consult Smaart's built-in helpfiles, which contain much of the same information but is organized to mirror the program's menus, dialogs and onscreen controls more directly.

Due to the nature of sound level measurement at concert levels, we will discuss a few considerations for choosing measurement hardware that go above and beyond equipment considerations for general measurement work. In addition, we recommend that you visit our website for the most up-to-date list of minimum computer hardware requirements and compatibility information.

A NOTE ABOUT SMAART'S HELP FILES

Smaart includes a comprehensive set of help files, which can be accessed from within the program by clicking *Help* in the program's menu bar (found at the top of the main program window on Windows operating systems, or across the top of the display on macOS computers). These helpfiles contain a concise reference for features and functions within Smaart and are intended as a companion to this user guide. The helpfiles and this user guide contain much of the same material but organized in different ways. While this guide attempts to group information conceptually, the helpfiles are organized to be in parallel with Smaart's user interface, to allow you to quickly "drill down" through menus, dialogs, and on-screen controls to find what you are looking for.

HOTKEY AND MOUSE CLICK NOTATION

Since Smaart SPL runs on both Windows and macOS, there are some minor differences in keyboard commands between the two versions. Specifically, the Control [Ctrl] key in Windows serves the same purpose as the Command [Cmd] key on macOS, and the [Alt] key in Windows is equivalent to the [Option] key on macOS.

In this document, we'll write the names of keys used for keyboard shortcuts ("hotkeys") in brackets. In cases where Mac and Windows have different key names, both names will appear inside the brackets separated by a slash. For example, [Ctrl/Cmd] means the [Ctrl] key on Windows or the [Cmd] key on macOS. As regards the mouse, left-click on Windows is a regular mouse click on macOS. A right-click for Windows users means [Ctrl] + click on macOS.

FULL SCALE (DBFS) VERSUS FULL SCALE

There exist two competing references for decibels in digital audio signals. Once convention references dBFS to the largest possible amplitude value in a digital system, such that 0 dBFS denotes the maximum possible digital amplitude value. We refer to this as "normalized Full Scale."

The second convention, preferred by the Audio Engineering Society (AES), references 0 dBFS to the *RMS* value of a full-scale peak-to-peak sinewave (i.e. 0.7071 normalized Full Scale, rather than 1.0). We will call this "AES Full Scale."

In Smaart, Full Scale decibel values are always referenced to normalized full scale, meaning that the RMS level of a full-scale digital sinewave is -3.01 dBFS.

SOFTWARE MANAGEMENT

To get started, you will need to install Smaart onto your computer and activate the installation.

MY.RATIONALACOUSTICS.COM

The online account portal at my.rationalacoustics.com is the centralized location by which Smaart software licensing is managed. If you don't already have an account, you should create one, and then you can register a new Smaart license by clicking the appropriately named "Register a new Smaart License" button on the Account Details page.

You can also upgrade licenses and add installs and download installers for macOS and Windows computers. Click the *Downloads* tab in the navigation bar to access downloadable installers for all the versions of Smaart that are registered to your account, both for Windows and macOS. Take care to download the appropriate version.

INSTALLING AND ACTIVATING

Once the installer is downloaded, run it and follow the prompts. Smaart SPL licenses are perpetual, and must be activated online. More detailed information about the activation process can be found in the Licensing Guides that appear in the Documentation section of the support portal at RationalAcoustics.com. If you require assistance with the licensing process, please contact Rational Acoustics support.

Following installation, the first time you run Smaart, you will be presented with an activation screen asking you to select your license type – perpetual or subscription. The next page will ask for your license code, followed by your my.rationalacoustics.com account information. The final screen will ask for your name and email address, as well as a Computer Name identifier of your choice. Then click *Activate* to activate your license.

BASIC COMPUTER REQUIREMENTS

You can find the most up-to-date minimum system requirements for Smaart v9 on our website at rationalacoustics.com. The minimum system requirements for Smaart v9 have changed from v8, and may be updated as new hardware, operating systems and program features become available. These minimum requirements are established to ensure reasonable performance when default settings are used. You should consider a more powerful computer if you plan to run many measurements at the same time.

2 – SPL CONCEPTS

In this chapter, we offer an overview of some core concepts that are important for SPL measurement and analysis. Due to the historical roots of sound level measurement and the inherent ties with hearing perception, a review of this information is recommended for all users.

WHAT IS SPL?

Sound Pressure Level (SPL) is a measure of the strength of air pressure variations caused by a sound wave. As sound waves travel, they force air into rapidly alternating states of high and low pressure (*compression* and *rarefaction*), and SPL is a measurement of the magnitude of those changes.

SPL measures pressure deviation, not ambient air pressure itself. To use a metaphor of a cork floating on the surface of the ocean, we're interested in how far the cork bobs up and down with passing waves, not measuring sea level.

Sound Pressure Level measurements are stated in decibels (dB SPL), which means they communicate a value by comparing it to an established reference. The reference for dB SPL that 0 dB SPL is set equal to a pressure fluctuation of 0.00002 Pascal (20 µPa). This is an extremely small pressure fluctuation that deflects the human eardrum by about the diameter of a hydrogen atom, and is roughly comparable to the quietest sound a human being can perceive under controlled conditions.

It also works out that 94 dB SPL is almost exactly equal to 1 Pascal (a unit of pressure equal to 1 Newton per square meter). Many sound level calibrators produce 94 dB SPL because it's a convenient "round number" value from a mathematics and engineering perspective. The threshold of physical pain lies somewhere around 120 to 130 dB SPL, which means that all the sounds you experience in your day-to-day existence (hopefully) fall between those two extremes.

SPL IS LOGARITHMIC

In mathematical terms, SPL is equal to twenty times the base ten logarithm of ratio of the rms (root mean square) sound pressure to the reference level of 20 µPa. This can be stated as

$$SPL = 20 \log_{10} \left(\frac{p}{p_{ref}} \right)$$

where p is the measured sound pressure and p_{ref} is the reference level of 20 microPascals.

Each 20 dB increase in SPL equates to a tenfold increase sound pressure. This allows us to compress the enormous range of sound pressures that human hearing is sensitive to (about a million to one) into a useful 120 dB range.

Another benefit of the logarithmic (dB) scale is that it is a closer approximation to our hearing perception, which responds to "percent change." In other words, we perceive a small increase to a low-level signal similarly to how we perceive a larger increase in a high-level signal, and the same "percent change" in both instances can be represented by the same dB value.

LOUDNESS VS LEVEL

It might be tempting to think of SPL as “how loud a sound is” but that’s not accurate. SPL is a measured level – that is, an objective measure of a physical quantity, just like height or weight, whereas *loudness* describes human perception, which is subjective and influenced by a number of factors besides SPL, including tonal balance, crest factor, intelligibility, the presence of distortion, and even listener preference for the program material. Understanding the distinction between *loudness* and *level* is critical to glean useful information from SPL measurement data.

For example, when listening to reinforced speech in low-intelligibility environments, people will often request for the operator to “turn it up,” because the lack of intelligibility can cause a perception that the signal isn’t loud enough. Likewise, mix engineers are familiar with the technique of using small amounts of harmonic distortion to increase the perceived loudness and presence of a signal without significantly increasing its level. Music playback at a certain SPL might be deemed far too quiet for a concert setting on Saturday night, and far too loud for a worship service the next morning – because even the context and physical setting play into our loudness perception. These are just a few examples of how, although *level* and *loudness* are certainly related, they are very different concepts.

An important result of this distinction is that our loudness perception is not able to reliably tell us whether sound levels are safe. It is possible to feel completely comfortable despite being exposed to unsafe sound levels, and it is also possible to feel that a sound is “too loud” when it’s objectively safe.

As with all measurement tools, the way forward lies in choosing the appropriate tool to answer the specific question that is being asked. If the questions are perceptual and artistic in nature (“Is the balance of this mix pleasing to the ear,” or “Is it artistically appropriate for this performance to be this loud?”), trained ears and experienced listening skills will provide the answers.

SPL measurement tools come to the forefront to provide objective answers to objective questions, such as “Is the show tonight the same level as the show last night,” “are we violating the noise ordinance,” or “are these sound levels harmful?”

UNDERSTANDING LOUDNESS

You may have encountered the statement that a 10 dB increase in SPL will be perceived as “twice as loud,” however due to the inherent divide between level and loudness, this is best viewed as a rough rule of thumb. For example, the dB increase needed to produce a doubling of perceived loudness is much smaller in the low frequency region. Experienced mix engineers exploit this divide to achieve mixes that feel loud and impactful at safe sound levels.

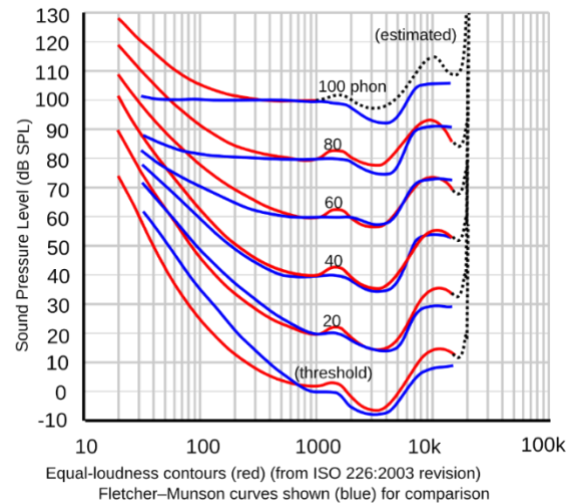


Figure 1 - Equal - loudness contours (red) and Fletcher-Munson curves (blue)

Formal attempts to quantify the relationship between SPL began when Harvey Fletcher and Wilden A. Munson published their research on the subject in 1933. Although their findings differ significantly from the currently accepted values, the work inspired further experimentation which resulted in the modern Equal Loudness Contours, standardized as ISO 226:2003. Figure 1 shows the original Fletcher-Munson curves in blue and the current Equal Loudness Contours in red.

The Equal Loudness Contours describe how our perception of tonal balance changes with level. At low levels, our hearing is most sensitive to frequencies around 3 kHz and rolls off towards the extremes of the audible spectrum, particularly the low frequency. As level increases, our response becomes less biased and tends toward a flatter response (although it does not flatten out completely, even at concert levels). Thus, a mix that sounds balanced at high levels might not also sound balanced at lower levels, providing a scientific basis for the liner notes of the Rolling Stones' 1969 album *Let It Bleed* ("THIS RECORD SHOULD BE PLAYED LOUD").

WHY IS SPL MEASURED?

If a sound level meter isn't used to tell us "How loud it is," then why use it? There are three primary motivations for taking a sound level measurement.

INFORMATIONAL

Often, sound level measurements are used for informational purposes, even though the measurement may not be required by any applicable policy or legislation. In the context of live events, sound level measurements are a powerful tool to help ensure consistency from show to show, a sort of "quality control" for the audience. Realtime sound level meters help provide mix engineers immediate visual feedback to help them control their levels and dynamic range throughout a performance.

In houses of worship, for example, which may offer several services on a weekend, mixed by different mix engineers or a team of volunteers, sound level monitoring is used to ensure that all the services maintain a similar level and dynamic footprint regardless of who is mixing.

Many live event spaces gather sound level measurements whenever a sound system is in use in the space, and the log data is kept on file in case any concerns about level policies need to be addressed after the fact.

NUISANCE NOISE

Nuisance noise regulations are common in many populated areas, where the sound from live events or loud music playback can carry across property lines and disturb people in nearby residences. The primary goal of many noise ordinance regulations is reducing noise pollution and nuisance to parties outside or adjacent to the event area. Nuisance regulations often require events to keep their levels under a certain limit as measured from the property line in question. A measurement taken at FOH mix position does not necessarily correlate to what the level will be in the far field, as the distance a sound system will carry depends on how it is designed as well as multiple environmental factors, however, once a rough correlation has been established between the two locations, setting a sound limit as measured from the FOH mix position is typically the most actionable way to address these concerns.

However, many of these nuisance regulations are poorly written, set arbitrary / unrealistic limits, use language that is too vague to be meaningful or enforceable, or specify an A-Weighted limit, which is ineffective because A weighting is insensitive to LF energy, often the most problematic from a nuisance noise point of view.

SOUND EXPOSURE

It is well established that both short-term and long-term exposure to high sound pressure levels can cause hearing damage. Sound exposure is a measure of total sound energy over a period of time. High sound levels increase a person's total sound exposure more quickly. Thus, there is not a single-number dB SPL value that can be considered "safe" or "unsafe," because it depends on the amount of time a person is exposed to that sound. Conceptually, sound exposure depends on both "how loud" and "how long."

In Smaart, sound exposure metrics function as dosimeters, with the meter reading the total sound exposure in percent dose. Smaart offers two different sound exposure dosimeter models, one based on OSHA's Permissible Exposure Limits (PELs) and one based on NIOSH's Recommend Exposure Limits (RELs).

OSHA PEL	
Sound level (dBA)	Duration (Hours:Minutes)
85	16:00
90	8:00
95	4:00
100	2:00
105	1:00
110	0:30
115	0:15
120	0:07

OSHA (Occupational Safety and Health Administration) issues PELs for sound exposure in an occupational setting. The OSHA exposure dose is based 90 dBA for an eight-hour time-weighted average (see Leq, above). The context of the standard is workplace safety, which is the basis for the 8-hour exposure time. The PEL for other durations is determined using a 5 dB **exchange rate**, which means that every 5 dB level increase cuts allowable exposure time in half. This means the PEL for a one-hour concert would be 105 dBA.

The chart at left shows the OSHA PEL for various exposure durations. Maximum level for a given exposure time H in hours can also be estimated with the formula **$SPL = 105 - 16.6 \log H$** . It is a common misconception that observing OSHA exposure limits will protect against hearing damage. The OSHA limits are not intended for hearing loss prevention; they indicate the levels at which employers are legally obligated to protect their employees. In other words, use the OSHA sound exposure measurements (*Exposure O* in Smaart) to answer questions about whether sound levels constitute a workplace safety hazard from a legal perspective.

NIOSH REL	
Sound level (dBA)	Duration (Hours:Minutes)
82	16:00
85	8:00
88	4:00
91	2:00
94	1:00
97	0:30
100	0:15
103	0:07

By contrast, NIOSH (National Institute for Occupational Safety and Health) issues RELs (Recommended Exposure Limits) which aim to protect hearing by reducing the risk of noise-induced hearing loss. The NIOSH limit is 85 dBA for an eight-hour time-weighted average. Unlike OSHA, the NIOSH standard uses an exchange rate of 3 dB, which is consistent with the current scientific consensus.¹ (Since 3 dB represents a doubling of power, it follows that the same exposure limit will be reached in half the time.) The chart at left shows the NIOSH REL for various exposure durations. Maximum level for a given exposure time H in hours can also be estimated with the formula $SPL = 94 - 10\log H$. Use NIOSH exposure dose measurements (*Exposure N* in Smaart) to answer questions about whether an

event presents a hearing health risk.

A-Weighting (see Frequency Weighting on page 13) is used for most health and safety regulations concerning sound exposure, although it may not be intuitively obvious why this is the case, as the A-Weighting curve bears little resemblance to the tonal response of human hearing at high levels. However, the current audiological consensus is that that A-Weighted sound level measurements correlate well with statistical data on noise-induced hearing loss, and reasonably approximate the diffuse-to-DRP transfer function of the human hearing mechanism. For this reason, the A Weighting curve is considered by audiological experts to be effective at characterizing sound exposure risk.^{2,3}

However, additional research has shown that LF energy can present a hearing health risk at high levels.⁴ For this reason, many sound level limits specify a secondary C-weighted limit.

Human hearing employs several mechanisms that act to temporarily desensitize the inner ear in the presence of high sound levels, to reduce the amount of energy hitting the cochlea. This loss of sensitivity is called a **threshold shift**. The excess energy causes stress and fatigue in the hair cells (stereocilia) of the inner ear, and in some cases the loss can become permanent.

It is important to note that the discomfort (or lack thereof) in the presence of loud sounds is not a reliable indicator of whether a noise level is safe from a hearing health standpoint. Some people think that they've "conditioned" their ears to be more resistant to loud noises. In reality, this is not the case, since repeated exposure to high sound levels can raise the discomfort threshold, so you can "feel fine" and still be subjected to potentially damaging sound levels.

¹ <https://www.cdc.gov/niosh/docs/98-126/pdfs/98-126.pdf?id=10.26616/NIOSH PUB98126>

² https://www.who.int/pbd/deafness/Hearing_loss_due_to_recreational_exposure_to_loud_sounds.pdf

³ <https://www.who.int/activities/making-listening-safe>

⁴ https://www.aes.org/technical/documents/AESTD1007_1_20_05.pdf

FREQUENCY WEIGHTING

Smaart SPL offers multiple options for frequency weightings, including A-, C-, Z- (unweighted), and octave-banded filters. When discussing SPL measurement values, it is always important to state which frequency weighting was used, since different weightings produce different values. Care should be taken to choose the appropriate frequency weightings for the measurement task at hand, as some are more useful than others depending on what we are hoping to learn.

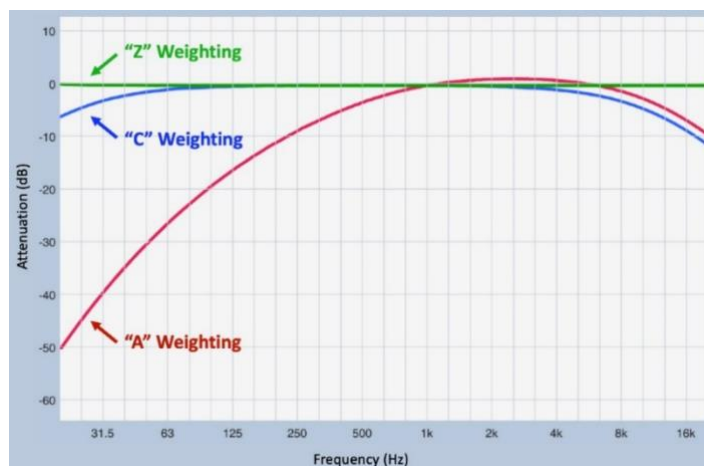


Figure 2: Z-, C-, and A-Weighting curves

THE GENESIS OF FREQUENCY WEIGHTING

Although accounts are sparse, it is generally accepted that the Fletcher-Munson research was used as the basis for a series of frequency weighting curves that were originally intended to alter the response of sound level measurements in a way that mimics the tonal response of human hearing at various levels, with the goal of creating a measurement that “looks like it sounds.” The A-Weighting curve, for example, is a rough match for the inverted original 40-phon contour. The intent was that the user would apply different weighting curves based on the measured level to allow the reading to match their expectations more closely.

Besides the technical flaws in this approach (the standardized curves departed from the already-inaccurate Fletcher-Munson results to make the filters easier to implement with the analog electronics of the day), there is a more fundamental conceptual pitfall, in the form of circular reasoning: We use weighting curves to change the measured level, while the measured level is used to choose the weighting curve. One must question the point of taking an objective measurement and then weighting it until we feel better about the result – it would be like changing the calibration on one’s bathroom scale until it reads a more flattering weight.

We can dodge this bullet entirely by remembering that SPL is not an indicator of perceived loudness, but rather a tool to answer objective questions about sound level. Despite their well-intentioned but misguided historical roots, the A- and C-Weighting curves remain useful in modern SPL measurement applications. The fact that the B-Weighting curve, despite being the most appropriate match for the tonality of human loudness perception at concert levels, has since been removed from most applicable Standards and sound level measurement devices, speaks to the fact that modern sound level measurement tools are not intended to answer questions about perceived loudness.⁵

A-WEIGHTING

The A Weighting curve consists of a significant low-frequency roll-off, exhibiting an attenuation of about -20 dB at 100 Hz, in combination with a slighter high-frequency roll-off. A-Weighting tends to correlate well with energy that lies in the human vocal range, and is commonly used for real-time level monitoring by concert mix

⁵ It should therefore come as little surprise that “Loudness” measurement algorithms such as ITU-RE BS.1770-3, intended specifically to characterize the perceived loudness of digital signals, do use a modified version of the B weighting curve.

engineers. As such, the majority of sound level limit policies at live events specify an A-Weighted limit as averaged over a period of time.

In addition, many environmental and nuisance noise regulations on the books in the United States specify A-Weighting. Although this is largely a holdover from the now-defunct Noise Control Act of 1972, A-Weighting is a poor choice for nuisance noise applications, particularly in the context of live events. This is because the low frequencies, which are most likely to propagate far distances outdoors and transmit most efficiently through structures, and therefore most likely to generate noise complaints, are de-emphasized by the A-Weighting curve, leaving the measurement unable to accurately characterize the sounds it is designed to regulate.

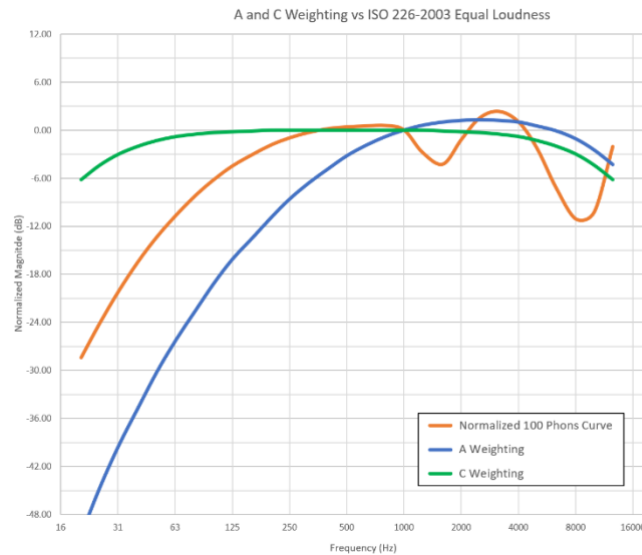


Figure 3 - A- and C-Weighting vs 100 phon ELC

A-Weighting is also used for most health and safety regulations concerning sound exposure, although it may not be intuitively obvious why this is the case, as the A-Weighting curve bears little resemblance to the tonal response of human hearing at high levels. However, the current audiological consensus is that that A Weighted sound level measurements correlate well with statistical data on noise-induced hearing loss, and reasonably approximate the diffuse-to-DRP transfer function of the human hearing mechanism. For this reason, the A Weighting curve is considered by audiological experts to be effective at characterizing sound exposure risk.^{6, 7}

However, additional research has shown that LF energy can present a hearing health risk at high levels.⁸ For this reason, many sound level limits specify a secondary C-weighted limit (see following).

C-WEIGHTING

The C-Weighting curve is effectively broadband, exhibiting a slight roll off at the frequency extremes. For this reason, C-Weighting is commonly used to characterize broadband content such as live music, including the subwoofer region. The slight rolloffs are sometimes considered useful for restricting the measurement to the range of energy commonly emitted by a sound system, and de-emphasizing low frequency structure-borne vibrations, rumble, and so forth. It is often stated that C-Weighting resembles the tonal response of human hearing at concert levels. However, in Figure 3 which compares the A- and C-Weighting curves against the 100-phon Equal Loudness Contour, we see that this is not the case.

⁶ https://www.who.int/pbd/deafness/Hearing_loss_due_to_recreational_exposure_to_loud_sounds.pdf

⁷ <https://www.who.int/activities/making-listening-safe>

⁸ https://www.aes.org/technical/documents/AESTD1007_1_20_05.pdf

Thus it is better to stick by the rule of thumb that we want to use C-Weighting for when we wish to include most of the low frequency energy in our measurement, and A-Weighting for when we don't.

C-A WEIGHTING

C-A Weighting is exactly what it sounds like – the difference between the C-Weighted and A-Weighted sound levels. C-A weighted measurements will show low values when most of the signal's energy is concentrated higher in the frequency spectrum, such as a speech signal, and higher values when more of the energy is concentrated at low frequencies, such as live music. C-A is often used as an indicator of overall spectral balance of a mix, and is useful for evaluating potential low frequency sound exposure risk.

The current recommendation from the AES Technical Committee on Acoustics and Sound Reinforcement is that when the measured C-A exceeds 20 dB, that is an indicator that the signal contains significant energy distributed in the low-frequency region and the traditional A-Weighted sound exposure metrics may not fully characterize the potential risk.

Z-WEIGHTING (UNWEIGHTED)

Unweighted (also called "Z-weighted") measurements are taken with a flat frequency response or, in other words, no frequency weighing applied to the measurement. It is most often used when the measurement needs to accurately characterize energy at the extremes of the audible range (below 30 Hz and above 10 kHz). It is of limited use in live sound applications but it can be useful for loudspeaker testing applications and acoustical evaluations.

OCTAVE BANDED MEASUREMENTS

Smaart SPL offers the ability to measure sound levels in octave-wide bands, on center frequencies from 16 Hz through 16 kHz. This is useful to study and document how sound level energy is distributed over frequency. For example, examining energy within the 63 Hz octave band can help indicate whether a level overage occurred because of the PA system, or a cheering crowd. Octave-banded sound level data can also help characterize nuisance noise along a property line.

TIME WEIGHTING AND TIME AVERAGING

Besides frequency weighting, another important consideration when measuring sound level is how we treat the signals in the time domain. Since real-world signals change in level from moment to moment, our measurements can produce different results depending on if they look at the signal over shorter or longer periods of time. Sound level measurements in Smaart fall into three different categories of time-domain weighting: *exponential average*, *equivalent continuous sound level* (Leq), and *Peak* (instantaneous level).

EXPONENTIAL AVERAGE / TIME WEIGHTING

Exponential averaging, also sometimes referred to as *time weighting*, is the meter ballistic that is the most common on basic sound level meters. It comes in two varieties, *SPL Fast* and *SPL Slow*. Conceptually speaking, exponential averaging is like applying a low-pass filter to the signal in the time domain, which "damps" the

reading against quick jumps but keeps the reading trending towards the current level. This *time constant* behavior is similar to the behavior of a charging capacitor, or the movement of the needle on a VU meter.

SPL Slow uses a time constant of 1 second, and *SPL Fast* uses a time constant of 125 milliseconds. Although these two time weighting options are the most commonly available, and are popular choices for mix engineers to keep an eye on during events, they don't offer enough context to allow us to make meaningful statements about level trends over time.

EQUIVALENT CONTINUOUS LEVEL

A very useful metric for sound level monitoring applications is Equivalent Continuous Sound Level, or *Leq*. *Leq* is a measurement of total sound energy over a period of time, presented as a single dB value. It can be thought of as “area under the curve,” or an answer to the question “what is the equivalent steady-state level over the same amount of time?” Since live events tend to have high dynamic range and high crest factor, *Leq* is typically the most useful metric for evaluating level trends (“what was the level of the last verse / song / set?”). For this reason, *Leq* measurements spanning ten or fifteen minutes are the most implemented sound level limits at live events.

The number in the suffix indicates the measurement period in minutes (*Leq 1*, *Leq 10*, etc). For brevity, the frequency weighting used is commonly indicated in the suffix as well (*LAeq 10*, *LCeq 15*, etc). In *Smaart SPL*, you can configure an unlimited number of *Leq* measurements with averaging times from 1 second to 24 hours.

PEAK / INSTANTANEOUS

The “Peak” measurement always shows the maximum instantaneous level reached within a measurement interval. It is useful for evaluating compliance with regulations seeking to protect employees against high level impulses such as those created by industrial machinery, explosives, and firearms. The C-Weighted variant (*Peak C*) is often used as a secondary limit for sound level regulation at live events to protect the audience from intense broadband events.

CALIBRATION

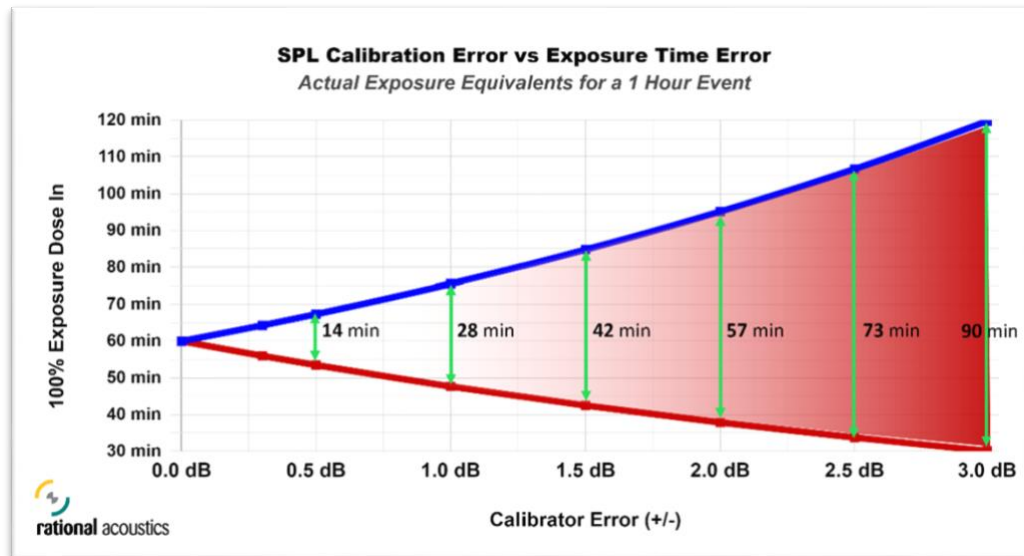
Since *Smaart* is a software application that can analyze audio signals from a wide variety of sources, it doesn't know what acoustic level at the microphone caused the digital signal values that it's seeing. The process of *calibration* involves producing a known acoustic level at the microphone, so *Smaart* can draw a correlation between a digital signal level and the acoustic level that caused it. This allows *Smaart* to establish the sensitivity of the entire measurement signal chain from microphone to convertor. Once *Smaart* has been calibrated, it will be able to accurately display sound level measurements. The calibration process is covered in detail on page 25.

ACCURACY AND COMPLIANCE

When it comes to equipment used for sound level measurement, achieving higher accuracy typically requires more costly hardware, so it's a good idea to evaluate your requirement for accuracy before making purchases. The accuracy required depends heavily on the goal of the measurement. A mix engineer who is simply looking

to ensure night-to-night consistency has no requirement for accuracy – even a poorly-calibrated measurement is useful, as long as it is stable over time – by allowing the mixer to “hit the same numbers” every night.

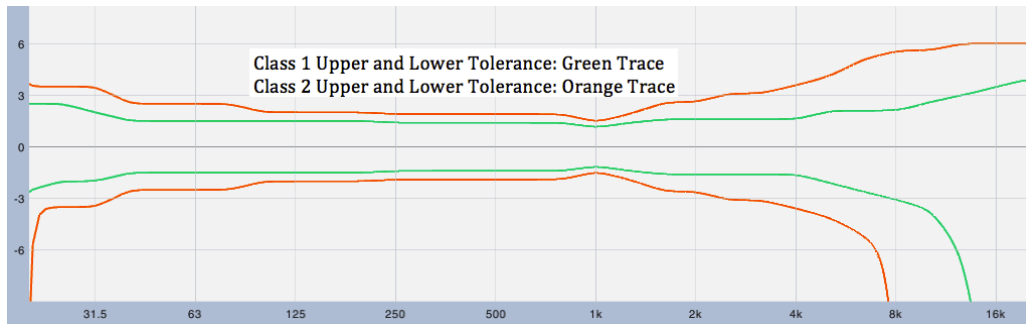
If the measured value needs to be meaningful – for example, when evaluating compliance with sound level regulations, then accuracy is important, as decisions will be made based on that level data. Sound exposure measurements have the highest requirement for accuracy. Consider that a calibration error of 3 dB equates to a 100% error in NIOSH sound exposure, which can result in individuals being exposed to twice the intended noise dose. THE FIGURE BELOW shows the relationship between calibration error and uncertainty in exposure time.



It is our recommendation that, for critical measurement applications such as compliance or sound exposure, accuracy should be within +/- 1 dB tolerances.

STANDARDS

IEC standard 61672-1 sets forth two categories for sound level meters, Class 1 and Class 2, with Class 1 being the more stringent specification. The standard dictates a variety of performance specifications covering every part of the metering and logging process, from mic tolerances and windscreens all the way through to data logging format and how the interface presents measurement data to the user. Probably the most relevant difference is the frequency response tolerances of the microphones used for logging. They are quite similar for most of the audible band, but a Class 1 microphone must fall within more stringent tolerances at the extremes of the spectrum.



COMPLIANCE

In the context of SPL metering, we use the term “compliance” to describe being in accordance with established standards. Given the rather serious consequences of error – hearing damage and potential legal trouble, to start – it is important that we can trust the data we are collecting. Note that compliance is not strictly a question of accuracy – a high-quality measurement mic, connected to Smaart SPL via a high-quality audio interface, and calibrated with a high-quality calibrator, will certainly provide an accurate measurement. It is not, however, a legally verifiable measurement, and so does not rise to the standard of “Class Compliant”.

By contrast, class-compliant systems such as the 10EaZy system are “closed” measurement chains – the hardware is designed to comply with IEC 61672-1 standards, manufacturer-calibrated before it is sold, and cannot be altered or tampered with in the field. The resulting logfiles therefore have sufficient integrity to be admissible as evidence in a court of law or legal proceeding.

In many jurisdictions, violations can carry hefty fines, so it is important to consider whether your SPL metering needs would be better met with class-compliant hardware, along with whether you want to leave the determination in the hands of a well-meaning local official with an inexpensive handheld meter or smartphone. In these situations, a good (class-compliant) offense can be the best (legal) defense.

Of course, Class-Compliant level hardware systems can be significantly more costly than a non-class compliant system capable of producing the same measurement data within the same margins of accuracy. Therefore, the question of whether Class-Compliant hardware is worth the extra investment for your application could be framed as “Do we just need to know what the levels are, or do we need to be able to prove it?”

3 – SELECTING HARDWARE FOR SPL MEASUREMENT

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

MICROPHONE MAXIMUM SPL

At concert levels, the measurement microphone in the audience area 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 seen by the microphone will be approximately 35 dB above the average A-weighted SPL. Analysis of sound level data gathered over the course of many live events suggests that the measurement microphone should be rated for a Max SPL of at least 135 dB – 140 dB to safely accommodate these instantaneous peaks without overload. Although infrequent overload events may not cause much concern in the context of live sound mixing, 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. As many measurement microphones are driven into severe distortion at concert levels, it is important to confirm the Max SPL specifications of the chosen microphone to be certain that it can perform cleanly at these levels.

MICROPHONE SENSITIVITY

After Max SPL, the next thing to consider is the voltage that the measurement mic creates at its maximum input level. Microphones with high sensitivity will produce large voltages that can overload a typical mic preamp. Therefore, the microphone and audio interface must be chosen to work together as a pair, by selecting an interface whose inputs can accommodate the voltages created by the microphone during the event.

Measurement microphone sensitivity is usually indicated as mV / Pa, which is the output voltage created when the capsule is exposed to 94 dB SPL (1 Pascal). Since our target of 140 dB SPL is, in linear terms, 200 times greater than 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 may overload the microphone preamps on many interfaces. A lower sensitivity measurement microphone, with a sensitivity of about 6 mV/Pa, for example, would produce about 1.2Vrms (+3.8 dBu) at 140 dB SPL, which can be comfortably accommodated by the preamps on most audio interfaces.

SOUND LEVEL CALIBRATORS

A sound level calibrator is used to produce 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 REL), 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.

4 – WORKING WITH SMART SPL

We'll start by taking a tour through the major elements of the Smart SPL graphical user interface (GUI).

INTERFACE TOUR

The Smart SPL user interface consists of a central SPL History Timeline display area, which is flanked by the Data Bar on the left side (highlighted in red in the figure below), the Control Bar on the right (highlighted gold), a docked meter bridge above (purple) and time range statistics below (blue).

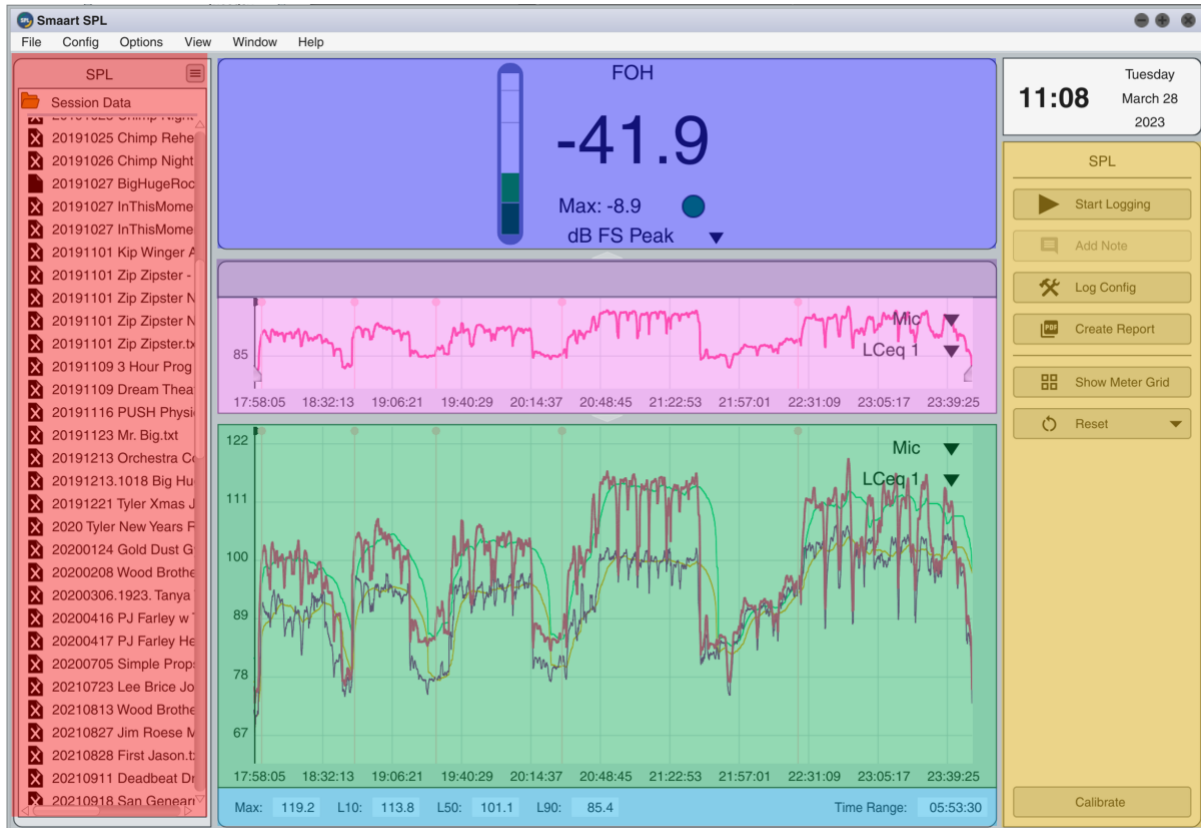


Figure 4 - the Smart SPL user interface. Data Bar in red, Control Bar in gold, docked meter bridge in purple, time range statistics in blue, and the SPL history timeline in the center, divided into the navigation pane (pink) and the detail pane (green).

SPL HISTORY TIMELINE

The SPL History Timeline is divided into two panes. The upper pane is the Navigation pane, which always shows the entire duration of the currently displayed logfile, and is shaded pink in the figure above. The lower pane is the Detail pane, which allows you to zoom in on a designated time range and view multiple metrics overlaid at once, and is shaded green in the figure above.

CURSOR READOUT AREA

One of the most important analysis tools in Smart is the mouse cursor. In addition to operating the program's controls, the cursor is used to find precise values for any point of interest in the SPL History Timeline. When you position the cursor over data in the plot, the Cursor Readout Area at the top of the history timeline

displays information about the data under the cursor. The text in the cursor readout area is color-coded to match the data it describes.



Figure 5 - The cursor readout area displays information about the data under the cursor. Note that the cursor readout is color coded to match the data in question.

CONTROL BAR

The Control Bar is found along the right side of the main program window. The Control Bar, true to its name, houses controls for the common functions in Smaart SPL. At the top of the Control Bar is a docked clock that displays the current system time.

DATA BAR

The Data Bar, located along the left side of the program window, is home to stored logfile data. Once logging is stopped, the logfile will appear in the data bar. The Data Library is a live view of the stored data files in your computer's Documents folder, inside the **Smaart SPL\Data\SPL Logs** subdirectory.

MENUS

Smaart's main menus span the top of the program window on Windows, and the top of the display on macOS. The top-level menus collect functions into general categories (File, Config, Options, View, Window, and Help). In this guide, carets will indicate the navigation path through the menus and any submenus. For example, *Options > Preferences* means to open the *Options* menu and select *Preferences*.

THE SMAART V9 FILE STRUCTURE

Smaart SPL stores all its captured measurement data and configuration data in your computer's Documents directory, in a folder called Smaart SPL. Inside the Smaart SPL folder are several subfolders, including the Config folder, which houses Smaart's config data and any saved configurations, and the Data folder, where saved measurement data is kept, in a subdirectory named SPL Logs.

MANAGING CONFIGURATIONS

The *Manage Configurations* dialog, accessible from the *Config* menu, will allow you to save the current configuration at any time by clicking *Save As*, entering the name for the configuration, and clicking *OK*. The *Restore Defaults* button will completely restore Smaart to its default state, including all configurations and settings. This can be helpful for troubleshooting purposes as well.

Choose a stored configuration from the list and click Recall, and confirm with *Yes* to restore it. Use the *Delete* button to delete a stored configuration, *Copy* to duplicate it, and *Overwrite* to update the stored configuration with the current program configuration.

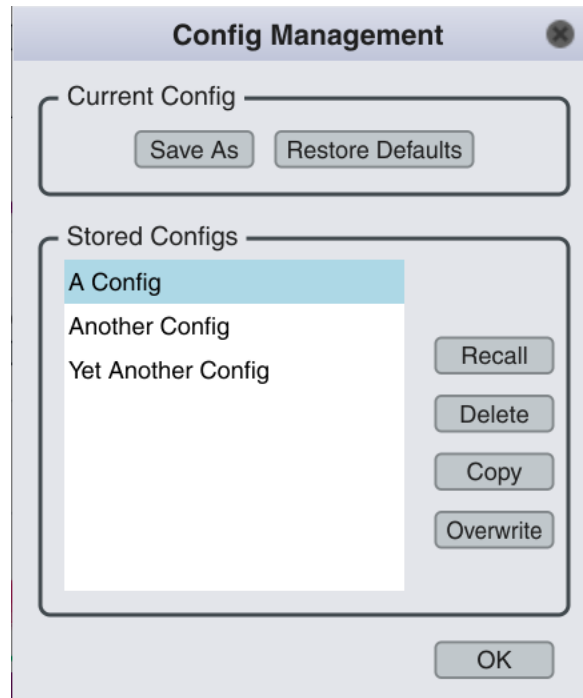


Figure 6 - the Config Management dialog

CALIBRATING SMAART FOR SPL MEASUREMENT

All of the measurement data produced by Smaart is produced by analyzing the incoming digital signals. Sound level measurements are unique in that they directly describe the strength of the fluctuations in air pressure occurring at the microphone location. In order to produce SPL measurement data, Smaart needs to know what acoustic level at the microphone caused the incoming signal levels. This is accomplished by running the sound level calibration routine.

Configuring Smaart SPL to measure sound level data is a two-step process. First, selecting the input devices and input channels you wish to use, and then calibrating those input channels for sound level measurement.

CHOOSING INPUT SIGNALS

In order to start our work, we need to get input signals into Smaart – we do this by selecting which input device(s) we wish to use, and which channels on those devices. This is done in the Input Config dialog, which Smaart will automatically display on first launch. You can also access the Input Config dialog at any time by clicking *Config > Input Config* or via the hotkey [Alt/Option] + [A].

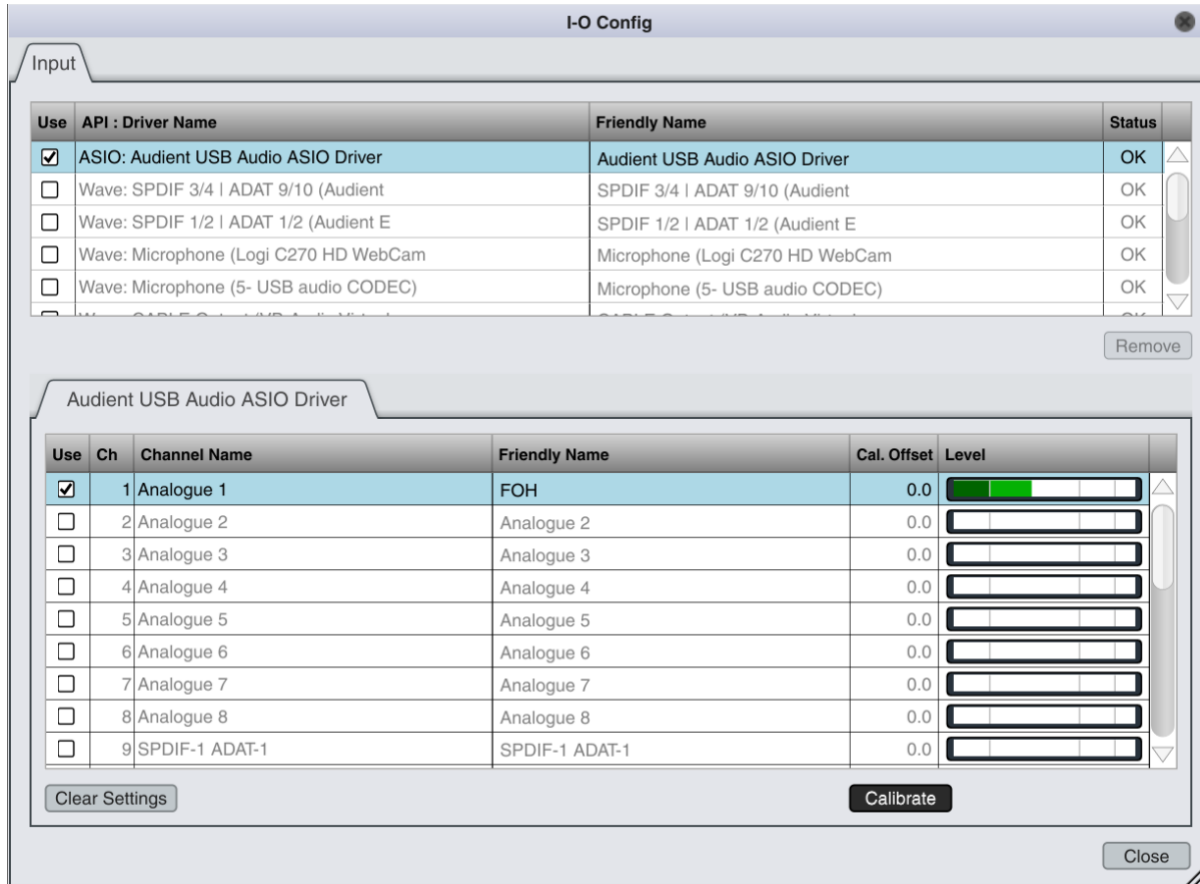


Figure 7 - the Input Config dialog

The IO Config dialog in other editions of Smaart v9 (Suite, RT and LE) has two tabs, *Input* and *Output*. Since Smaart SPL only monitors incoming signals but doesn't produce any output signals of its own, its IO Config dialog only has a single tab, labeled *Input*.

At the top of the Input tab is the devices table, which lists all audio devices that your operating system recognizes. While previous versions of Smaart required devices to be connected and recognized by the operating system before launching the software, Smaart v9 can dynamically detect input devices connected at any time. Once you connect your device, it should populate in the top pane of the IO Config dialog after a short delay, provided the proper drivers are installed.

To select a device for use in Smaart, click its check box in the *Use* column. The *API: Driver Name* column lists the name that the audio device (or its driver) reports to the operating system. On macOS, all devices will use CoreAudio. Windows machines may have ASIO and Wave API devices, and some devices may appear as both types.

Note: If you have both Wave and ASIO drivers installed for an IO device with more than two channels, it will typically show up as a single ASIO device, and multiple Wave devices, since the Wave API only supports two input channels per device.

In the *Friendly Name* column, you can assign an alternative name to a device if you choose – remember to press the [Enter] key after typing the name to apply your change.

The *Status* column indicates either *OK* or *N/C* for each device. *OK* means that Smaart was able to successfully connect to the device and it is ready for use, while *N/C* means “not connected”. *N/C* can mean *really* not connected – Smaart remembers the device was connected previously, but it is no longer present – or it can indicate some hardware or software problem prevented Smaart from communicating with the device. This could be a device driver issue, a loose cable, or the device needs to be restarted. You can remove an *N/C* device by selecting it in the list and clicking the *Remove* button to the bottom right of the devices table.

Enabling a device for use in the devices table will create a tab for it below the device table. The Input Channels table has 6 columns, the last of which is a live signal input meter for each input.

The check boxes in the *Use* column allow you to select the inputs you want to use in Smaart and ignore the rest. Note: clicking directly on the word *Use* in the column header will quickly select / deselect all inputs at once. The *Ch* column lists the channels by number, and *Channel Name* lists the driver-reported channel name for each.

You can assign a *Friendly Name* for each channel to help you identify it when you see it elsewhere within Smaart – make sure to press the [Enter] key to set the name after you’re done typing and move down to the next channel on the list.

The *Cal Offset* column indicates the calibration offset for sound level measurements. A calibration offset of zero means the input is not calibrated for sound level measurements. Once the input has been calibrated, the *Cal Offset* value displayed corresponds to the acoustic level at the microphone that will bring that input’s convertor to 0 dBFS.

Below the Input Channels table is a *Clear Settings* button which will reset the device’s configuration by clearing out all calibration offsets, microphone correction curves, and friendly names. To the right, the *Calibrate* button opens the *Amplitude Calibration* dialog and defaults its selection to the currently selected input.

CALIBRATION PROCEDURE

To complete the sound level calibration routine, you will need a sound level calibrator, which is a device that produces 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 input signal chain, from microphone to AD convertor.

A sound level calibrator is a precision device that fits snugly over the measurement microphone capsule and generates a known acoustic level. Most calibrators produce levels of 94, 110 or 114 dB SPL, but it is not critical which level your calibrator produces as long as it is known. If your calibrator offers a choice, there is a slight advantage to using the higher level because it increases the signal to noise ratio of the calibration process and makes external and ambient noise less likely to disrupt the process.

When you insert your measurement microphone into the sound level calibrator, do so slowly, to avoid shocking and potentially damaging the microphone diaphragm. Most calibrators use a rubber seal or gasket of some type to ensure a proper seal. The fit should be snug, without wiggling or play. If you can clearly hear the calibrator’s tone once you turn it on, you may not have a solid seal.

Once the calibrator is properly seated and is switched on, open Smaart’s *Amplitude Calibration* dialog by one of the following:

- Clicking the *Calibrate* button at the bottom of the Control Bar
- Clicking the *Calibrate* button at the bottom of the *IO Config* dialog
- via *Config > Amplitude Calibration*

The *Amplitude Calibration* dialog has dropdown menus at the top for you to select the device and input you wish to calibrate – ensure the selection is correct before you continue with the process. The *Offset* field displays the calibration offset for that input – the acoustic level that will bring the convertor to 0 dBFS. If the input hasn't yet been calibrated, this number will be 0.0. The level meter on the right shows the signal level for the selected input, with a bit of time averaging applied to smooth out the reading, and should reflect the level of the input signal as caused by the calibrator. Once you have adjusted the input preamp to achieve the desired digital level, click the *Calibrate* button to run the calibration routine.

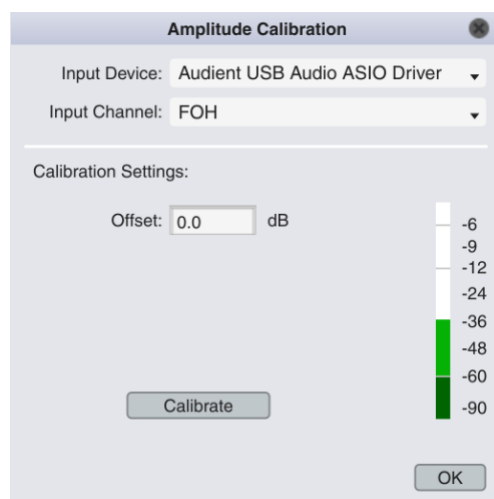


Figure 8 - Amplitude Calibration dialog

While the calibration process runs, avoid bumping the microphone and calibrator or making unnecessary noise. Once the process completes, it displays the measured digital level in dBFS RMS. Smaart will then ask you what acoustic sound level corresponds to this measured digital level. Enter the level produced by your calibrator into the *Set this value to:* field and press [Enter]. The *Offset Required* value displays the resulting calibration offset, which is the highest sound level that can be accommodated by your measurement setup in its current configuration without overloading your convertor. Click *Okay* to set the offset and close the dialog.

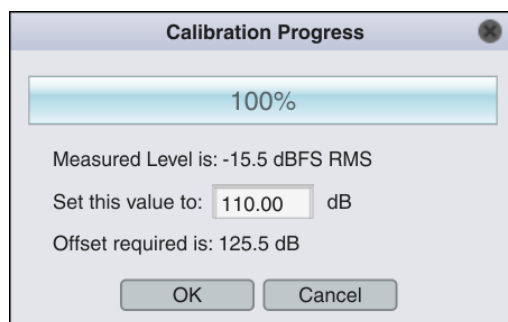


Figure 9 - the Calibration Progress dialog

Note that calibration describes the sensitivity of the entire signal chain from microphone to convertor, so if you plug in a different measurement microphone or adjust your interface's preamp gain, you will need to run the calibration procedure again to determine the new offset, unless you are using an interface with Gain Tracking (see below).

GAIN TRACKING

If you are using the Audient evo8, evo16, or the Roland OCTA-CAPTURE as your Smart SPL interface, Smart can communicate with the device to read and adjust preamp gain settings. If one of these devices is detected, a *Gain Tracking* checkbox will be displayed below the Channels list in the IO Config dialog. After the calibration process is completed, select the *Gain Tracking* check box to allow Smart to automatically adjust calibration as the preamp level is adjusted. Note that when using evo devices, the evo application should be launched before launching Smart in order for the Gain Tracking functionality to be enabled.

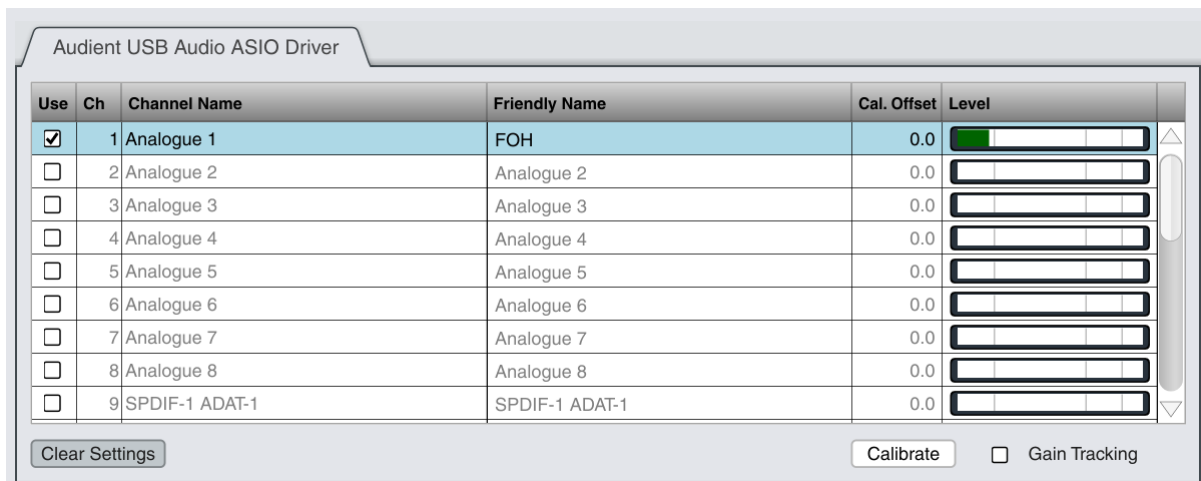


Figure 10 - the Gain Tracking checkbox in the lower right corner enables gain tracking when supported hardware is detected

When the docked meter bridge at the top of the interface is expanded, and a device that supports preamp control is recognized, preamp controls will be shown next to each input's meter.

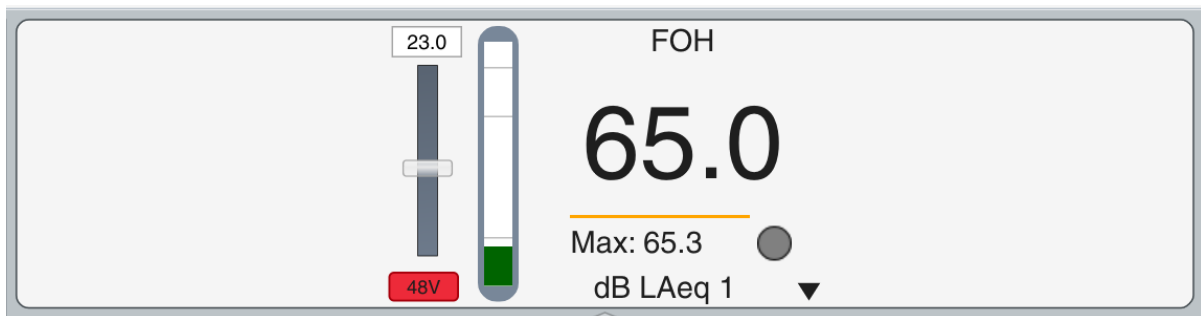


Figure 11 - preamp controls visible in the docked meter bridge at the top of the interface

Additionally, you can open the Input Meters window (*View > Input Meters > Input Meters*) or [Shift] + [E]. A meter strip will be visible for every input selected for use in Input Config. Adjust the gain slider to adjust interface preamp gain.

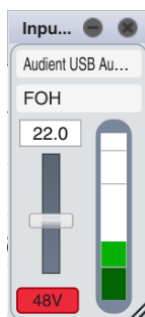


Figure 12 - Input Meter with gain tracking controls

The accuracy of these adjustments is device-dependent but will typically range within a maximum error of about ± 1.5 dB from nominal gain setting, between 0 and 50 dB, when calibrated at a gain setting of 25 dB. To quickly determine the maximum gain adjustment error for your specific device, set input to a gain to 25 dB, then calibrate for sound level measurements with Gain Tracking turned off. After calibrating, with the calibrator still fitted to the microphone and turned on, enable Gain Tracking and assign the input to an SPL meter in Smaart, then observe the measured level as you turn the gain up to 50, and down to 0 dB. The maximum deviation observed from the reference level of the calibrator represents the maximum expected accuracy of your setup.

DISPLAY CONFIG

You can access display settings by clicking *Config > SPL Config* to open the SPL Config dialog to the *Display* tab. This dialog is home to several settings that control the way that timeline and meter data is displayed in Smaart SPL.

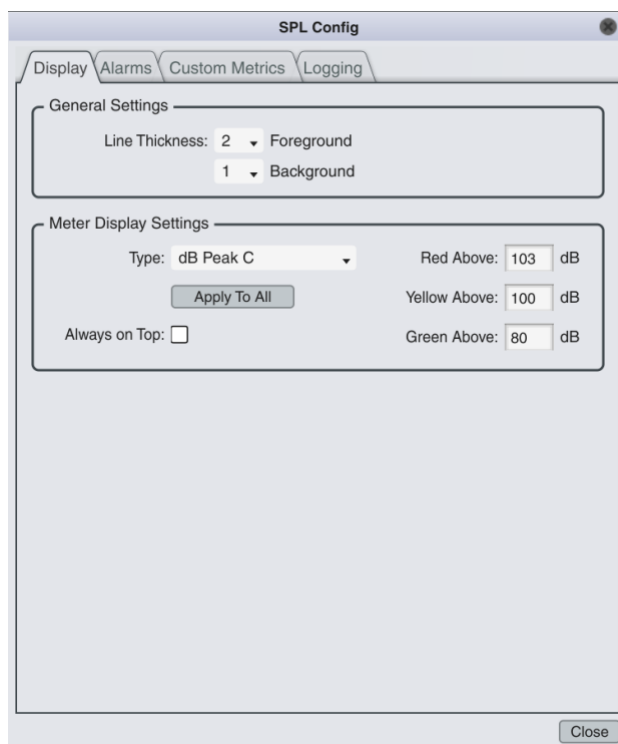


Figure 13 - Display settings

Line Thickness allows you to set the thickness of the history timelines traces. Two dropdowns are available: *Foreground* sets the thickness of the trace on the top of the z order, in pixels. *Background* sets the thickness for all remaining history plot traces, in pixels.

The *Meter Display Settings* field allows you to customize the Red, Yellow and Green color thresholds for Peak, dB SPL, and Leq metrics independently or all at once.

SKINS

Smaart has two built-in color schemes, the *Default Dark* scheme that you see the first time you run the program, and a *Default Light* scheme that uses dark text against a lighter background. *Default Dark* works well indoors, particularly in darkened rooms. *Default Light* might be a better choice in brightly lit environments, or when making screenshots for printed documents or presentations.

You can change between them using the *View > Skins* menu, cycle through all available skins using the hotkey [Ctrl/Cmd] + [Shift] + [X].

In addition to the built-in defaults, Smaart also allows you to define custom color schemes (“skins”) via the *Skins* tab of the *Options > Preferences* dialog.

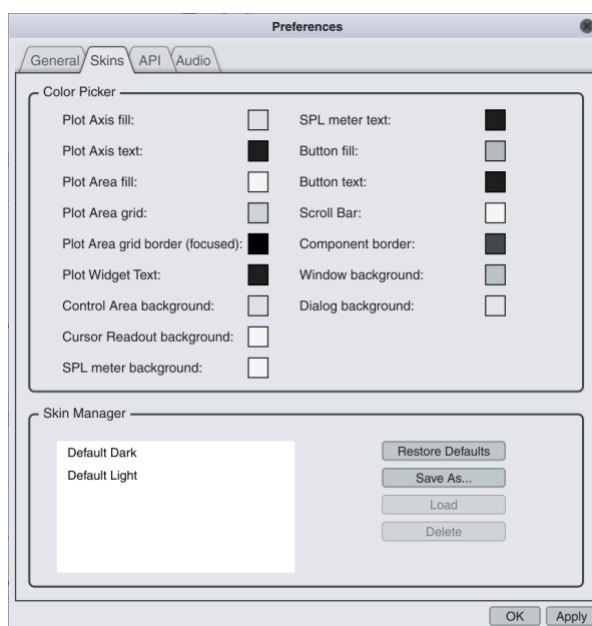


Figure 14 - Skin Manager

The *Skins* tab is divided into two sections, *Color Picker* and *Skin Manager*. The *Color Picker* section consists of color tiles showing the color assignments for various interface elements in the current color scheme. Each color tile is a button that opens a *Color Selection* dialog. Color changes take effect immediately when you click *Apply* in the color selector.

The *Skin Manager* section lists all available color schemes. Since the *Default Light* and *Default Dark* skins can’t be altered, making a change while one of them is the active skin will create a *New Custom Skin* using either the *Default Light* or *Default Dark* skin as a starting point. You can rename a custom skin by double-clicking it in the list and typing a new name. Custom skins update with any changes in real time. The *Restore Defaults* button

will restore all colors in the custom skin to match either the *Default Light* or *Default Dark* skin on which it was based. *Save As* will save a copy of the skin with a new name. *Load* will select a skin for use, and *Delete* will delete the custom skin.

Custom skins are stored in the *Documents\Smaart SPL\Skins* directory as .xml files, so you can move them to other computers or share them with friends by placing the files in this folder.

CONFIGURING METRICS

By default, Smaart SPL offers 15 sound level metrics in addition to the digital full-scale signal level (dB FS Peak). Ten of these are hard-coded and always enabled: SPL Fast, SPL Slow and Leq 1, each in three weightings (A,C and unweighted), and Peak C. In addition, the *Custom Metrics* table is populated by default with Leq 10 in the same three weightings, plus both variants of sound exposure (Exposure O and Exposure N).

The Custom Metrics table is accessible via *Config > SPL Config* and then clicking the Custom Metrics tab. Alternatively you can click the Log Config button in the Control Bar and then click over to the Custom Metrics tab.

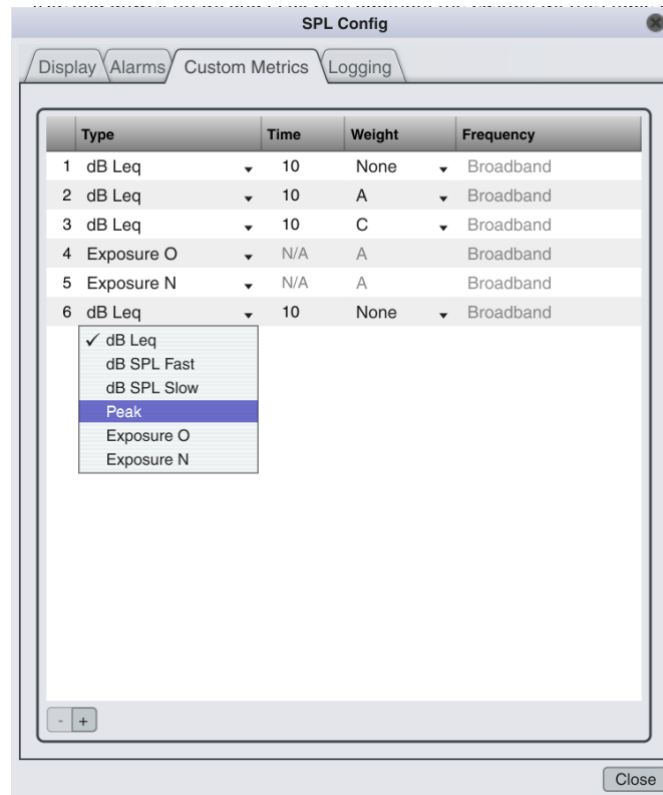


Figure 15 - adding a new custom metric via the Custom Metrics table

Add a new custom metric to the table by clicking the [+] button in the lower left corner of the dialog. You can add an unlimited number of custom metrics – if you accidentally duplicate an already-configured metric, Smaart will simply remove the duplicate for you when you close the dialog.

Custom metrics can be *dB Leq*, *dB SPL Fast*, *dB SPL Slow*, *Peak*, *Exposure O* or *Exposure N*.

CUSTOM LEQ METRICS

For Leq metrics, enter a time period in minutes in the *Time* field. You can also use “s” for seconds or “h” for hours. For example, enter “10s” for a ten second Leq. Smaart SPL will allow you to create Leq metrics in any length from 1s to 24h. Use the dropdown in the *Weight* column to select None, A, C, C-A, or Octave weighting. Selecting Octave will enable the *Frequency* dropdown for you to select octave bands on centers from 16 Hz to 16 kHz.

CUSTOM SPL FAST AND SPL SLOW METRICS

Since all broadband (A/C/unweighted) variants of dB SPL Fast and Slow are already permanently configured as options, custom dB SPL Fast and dB SPL Slow metrics are automatically set to Octave weighting. Use the *Frequency* dropdown to select an octave band from 16 Hz to 16 kHz.

CUSTOM PEAK METRICS

In addition to the permanently configured *Peak C* metric, you can add additional Peak metrics with A Weighting, no weighting, or Octave weighting. If you select Octave weighting, the *Frequency* dropdown will allow you to select an octave band from 16 Hz to 16 kHz.

CUSTOM EXPOSURE METRICS

By default, both exposure metrics (O and N) are configured as custom metrics in the table, however if you removed one and want to recreate it, you can do so by selecting either Exposure O or Exposure N from the dropdown. Exposure metrics are simple dosimeters and have no user-configurable options.

CONFIGURING METERS

In Smaart SPL, there are two banks of sound level meters. The first is the docked meter bridge that appears across the top of the main program window and displays a single meter for each enabled, calibrated input. Each meter can be set to display any configured metric. The docked meter bridge can be hidden by clicking the disclosure arrow below the meter bridge. See Figure 11.

Additionally, Smaart SPL features a floating meter grid in a separate, resizable window that can contain a user-configurable number of meters. Each meter can display any configured metric for any calibrated input and be given a custom user-assigned name. Show the meter grid by clicking the *Show Meter Grid* button in the Control Bar, or by pressing [E] on your keyboard.

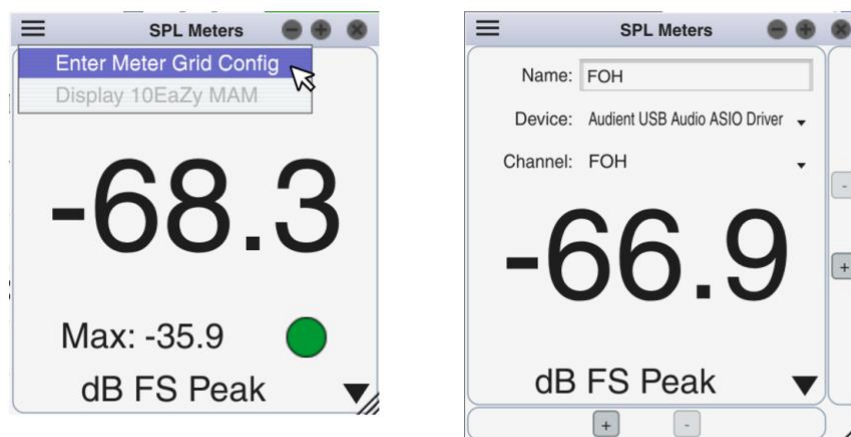


Figure 16 - Meter grid in running mode (left) and config mode (right)

Click the hamburger (three-line) icon in the top corner of the meter grid window to open the SPL meter hamburger menu, and choose *Enter Meter Grid Config* to put the meters into Config mode. In Config mode, small borders appear along the right and bottom edges of the meter grid containing [+] and [-] buttons that control the number of meters in the rig.

Clicking the [+] on the right edge will grow the meter grid horizontally by adding another column of meters, and the [-] will shrink it by removing a column. Clicking the [+] on the bottom edge will grow the meter grid vertically by adding another row of meters, and the [-] will shrink it by removing a row.

Type a name for each meter in the *Name* field, and use the Device and Channel dropdowns to assign any calibrated input to the meter.

When you are done configuring the meters, choose *Exit Meter Grid Config* from the hamburger menu. The dropdown metric selection menu at the bottom of each meter can be adjusted even if the meters are not in Config Mode. You can also resize the entire meter grid window by dragging the resize handle in the bottom right corner, whether or not the meters are in config mode.

CONFIGURING ALARMS

Alarms notify the operator visually when a specified metric exceeds a specified level, and the alarm occurrence is logged in the log file. Alarms are configured in *Config > SPL Config* in the *Alarms* tab. Each entry in the table represents an alarm. Select the *Use* checkbox to enable the alarm. Select from configured inputs with the *Input* dropdown, or *All Calibrated Inputs* for a global alarm, and then choose the metric that you would like to use for the alarm trigger in the *Type* dropdown, which lists all configured metrics. The *Level (dB)* field sets the alarm threshold, and *Duration* sets the amount of time, in seconds, that the associated meter will flash a visual warning when the alarm is triggered.

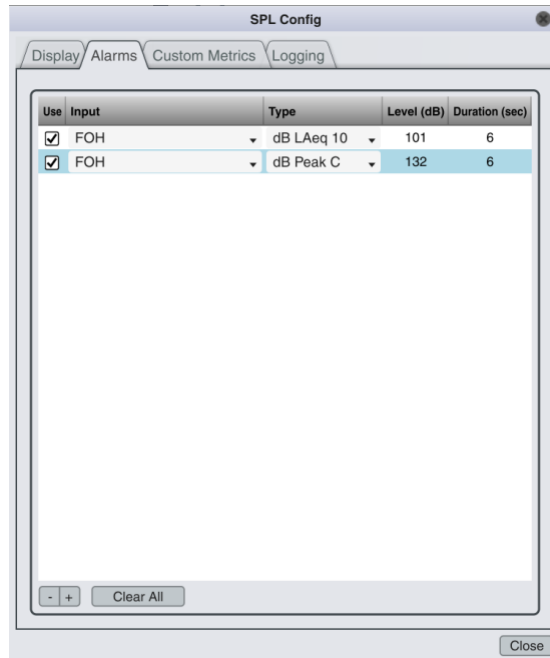


Figure 17 - Alarms tab

Click the [+] button in the lower left corner to add a new alarm, or the [-] button to remove the selected alarm from the list.

LOGGING

When logging is enabled, the values of all configured sound level metrics are written to a timestamped logfile every logging interval. The default logging interval is three seconds, and can be adjusted in *Logging* tab of SPL Config, which is accessible directly by clicking the Log Config button in the Control Bar. For “short term” metrics (Peak, SPL Fast and SPL Slow), the highest level reached during the logging interval is recorded so that no maximum values are missed. “Long term” metrics (Leq and Exposure) are logged at their present value.

Engage logging by clicking the Start Logging button in the Control Bar, which will turn red. Logfile data will begin to populate the SPL History Timeline. Logging will continue until it is stopped by clicking the Stop Logging button. Once logging is stopped, the logfile will appear in the SPL Data Bar along the left side of the screen.

The logging status is persistent, such that if you close Smaart SPL while logging is running, the program will automatically begin logging again the next time it is opened.

At any point during logging, you can click the Add Note button in the Control Bar to add a timestamped lollipop marker to the timeline at the current time, with a customizable text note. This is helpful to mark certain events such as set changes, pyro cues, and so forth, for later review.

LOLLIPOP MARKERS

Smaart SPL uses color-coded “lollipop” markers in the SPL History Timeline to indicate certain events.

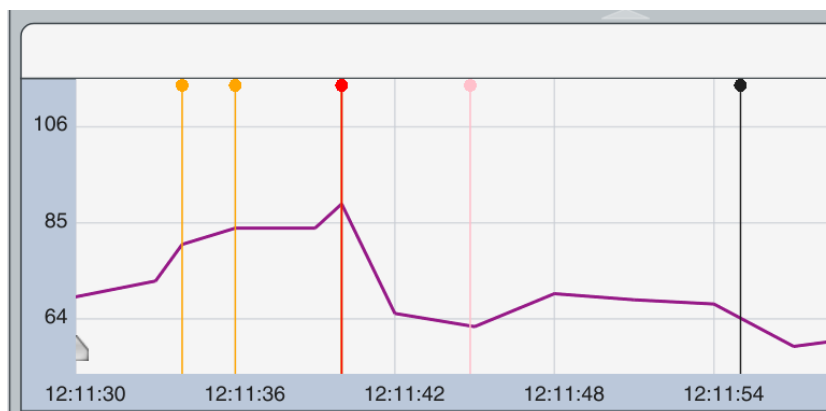


Figure 18 - a portion of SPL History Timeline data showing two alarm markers (orange), an overload marker (red), a note marker (pink) and an Leq reset marker (black)

NOTES (PINK)

Notes added to the timeline via the *Add Note* button in the Control Bar are indicated as pink lollipop markers at the corresponding timestamp. Hover your cursor over the marker to see the associated note text.

LEQ RESET (WHITE / BLACK)

When Leq buffers are manually reset (either by using the *Reset* button in the Control Bar or the SPL Meter Grid right click menu, the reset operation is indicated by a white lollipop marker in the timeline when using the dark interface color scheme, and a black lollipop marker when using the light interface color scheme.

ALARMS (ORANGE)

Whenever a metric's logged value exceeds the corresponding alarm threshold for that metric, an orange lollipop will indicate that the alarm has been triggered. The alarm state will persist until the metric's measured value drops below the alarm threshold. Hover your cursor over the marker to see the metric associated with the alarm.

OVERLOADS (RED)

When Smaart detects an overload condition in an incoming signal (three consecutive samples at 0 dBFS), all averaging buffers will be flushed to remove the undefined values, and will begin to refill as soon as the overload condition passes. Overload events are logged into the logfile and indicated in the SPL history timeline as a red lollipop marker, and on the meter grid by a red status indicator and the word "OVERLOAD".

WORKING WITH THE METER GRID

Whenever sound level data is being logged and/or displayed on a meter, Smaart calculates data for all configured metrics, so you can switch between any of them without having to wait for averaging buffers to refill for a newly displayed metric.

Meters with Leq metrics assigned to them display a horizontal bar that reflects the state of the Leq buffer. The bar is yellow when the buffer is partially full and gradually extends across the face of the meter, turning green when the buffer is full. Hover your mouse cursor over the bar to see the buffer status.

The *Max* indicator shows the highest value reached by that metric. Click the colored circle icon to the right to reset the Max value. The circle icon turns green in the presence of signal and indicates a “healthy” meter, and turns red in the case of overload (see above).

Right click in the center of a meter to open a pop-up menu with two options: Reset Leq buffers for this input, and Reset Exposure for this input.

You can also choose to globally reset max levels, Leq buffers, or exposure by using the *Reset* button in the Control Bar.

WORKING WITH LOGFILE DATA

In both the Navigation pane and the Detail pane, the Friendly Name of the input generating the data is displayed in the upper right corner of the plot. Immediately below is the name of the currently displayed metric. Click the metric name, or the white arrow next to it, to open the metric selection droplist. The droplist displays all metrics contained in the current logfile. Check the box next to a metric to select it for display in the timeline.

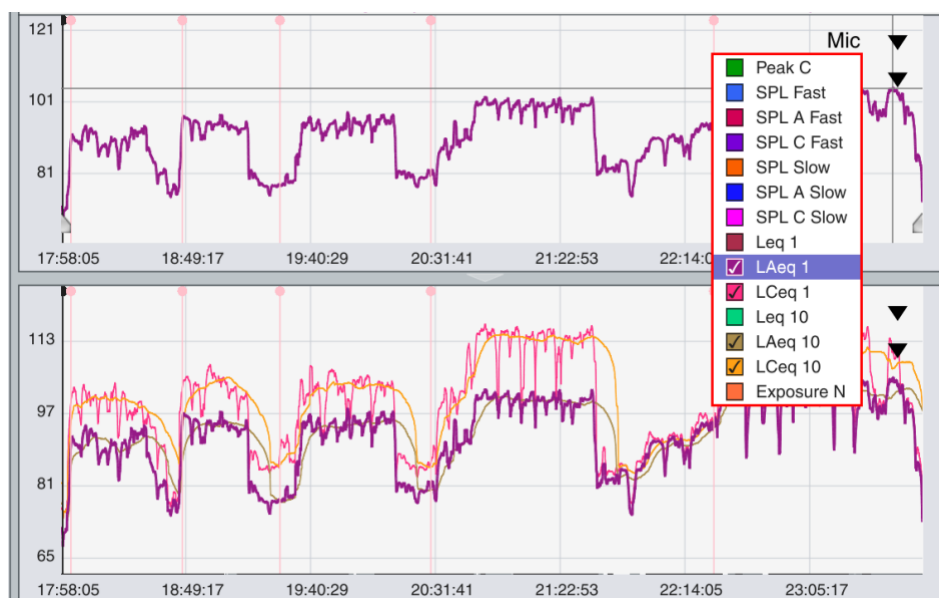


Figure 19 - Metric selection droplist

You can click the name of a metric to bring it to the top of the z order or use the [Z] key on your keyboard to cycle metrics to the top of the z order. The metric that is currently at the top of the z order is also the metric displayed in the top (Navigation) pane.

The navigation pane will always automatically scale itself in both the horizontal and vertical dimensions to display the entire logfile. The detail pane will also automatically set its vertical scale as metrics are shown and hidden to properly accommodate the data being displayed and can be zoomed in or out vertically by pressing the [+] and [-] keys on your keyboard. Return the vertical scaling to default (auto-fit) by clicking anywhere in the plot border.

SELECTING A TIME RANGE

You can focus the detail plot in on a specific time range by dragging the arrowhead-shaped widget handles that appear in the lower edge of the navigation plot. The time range that you have bound by the widgets in the navigation plot will fill the detail plot. By default, they appear in the bottom left and bottom right corners to indicate that the entire time range is shown in the detail plot. Dragging them inwards selects a shorter time range. The portions of the timeline data that lie outside the selected time range will be shaded to indicate they are not displayed in the detail plot. You can also enter a time range direction into the Time Range field in the lower right corner, below the detail plot. For example, type 32:00 to set the time range to 32 minutes.



Figure 20 - Using the time range sliders in the navigation pane to view 52 minutes and 24 seconds of log data in the detail pane

When the selected time range is shorter than the total logfile duration, hovering the mouse cursor over the navigation pane between the two time range selection widgets will turn the cursor into a hand icon. Click and drag left or right to move the selected time range. For example, if you have a five minute time range selected, you can slide that 5 minute selection left or right to see a different five minutes' worth of data in the detail pane. Sliding the selection all the way to the right side of the navigation plot (most current data) will force Smart to always show the most recent data as logging continues.

STATISTICAL ANALYSIS (DYNAMIC RANGE STATISTICS)

Below the detail pane, across the bottom of the main program window, is the statistics bar. This bar displays a statistical analysis of the data appearing within the selected time range, for the metric on the top of the z order. For most metrics, the statistics bar displays Max, L10, L50 and L90 data for the selected time range.

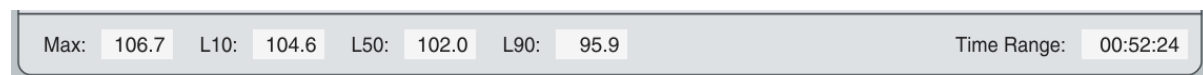


Figure 21 - Time Range statistics bar

Max is the highest value logged for the metric within the selected time range.

L10 is the level that was exceeded by 10% of the data points within the selected time range.

L50 is the level that was exceeded by 50% of the data points within the selected time range. In other words, the median level.

L90 is the level that was exceeded by 90% of the data points and is sometimes regarded as a useful indicator of background or ambient noise levels.

When *Exposure O* or *Exposure N* data is the selected metric (on top of the z order), the statistics bar instead changes to display Max, Min and Delta.



Figure 22 - Exposure Statistics

Max is the highest exposure value logged within the selected time range, in percent.

Min is the lowest exposure value logged within the selected time range, in percent.

Delta is the difference between *Max* and *Min*, or otherwise, the sound exposure dose accumulated in the selected time range, in percent.

INCOMPLETE FILES

While logging is in progress, the data is held in a temporary text file located in the same directory as your session folder. Once logging is completed, the logfile data is finalized, column headers, statistical analysis data and other information are added, and then the finalized logfile is saved with an automatically generated name in your session folder.

If logging is interrupted without properly closing Smaart (a crash or the computer is shut down improperly etc), this process will be interrupted. Although the temporary logfile is not directly Smaart-readable as a logfile, it does contain the log data so no data loss occurs. To manually recover log data in the event of an interrupted logging session, it is recommended to start logging again, allow it to run for a few seconds, then stop it, which will generate a proper logfile reflecting the current program configuration, and including accurate column headers for the configured metrics. The temporary data can then be copied into this logfile using a text editing program and it should line up with the proper header format. For help reformatting orphaned log data in the event of a failure, please contact Rational Acoustics support.

GENERATING REPORTS

The Generate Report feature allows you to export a PDF report for the selected time range, including a variety of user-editable details, statistical data for configured metrics, and optionally history thumbnail graphics. Click the *Create Report* button in the Control Bar to open the Generate SPL Report dialog. The left side of the dialog

contains user-editable text fields for *Operator*, *Company*, *Venue*, *Event*, *Engineer*, *Production Company*, and a general *Notes* field.

Figure 23 - Generate SPL Report dialog

The right side of the dialog is the History Thumbnails menu. The *Include History Thumbnails* checkbox is selected by default – un-check it if you wish to exclude the history thumbnail graphics from the report. All metrics included in the logfile data are available in the list; select the checkboxes for those which you wish to include thumbnails for in the report. By default, the selected metrics will match the metrics that were visible in the SPL History Timeline when the Create Report button was clicked. You can also quickly *Select All* or *Select None* using the buttons below the metric list.

Click the *Create* button to create the SPL PDF report and export it to the SPLReports directory located in Document\Smaart SPL\Data\SPL Logs\SPLReports. Selecting “Open file location when done” before creating the report will open your operating system’s file browser to the report directory so you can easily access the report. Selecting “Email Report” will change the *Create* button to read *Create and Email*. Once you click it, Smaart will attempt to launch your operating system’s default mail browser with the PDF report attached to a new email draft.

SPL WEBVIEWER (SMAART API)

Smaart SPL allows realtime remote viewing of the SPL history timeline data, meters, and alarms via a web browser on a computer or mobile device that shares a network connection with the computer running Smaart SPL .

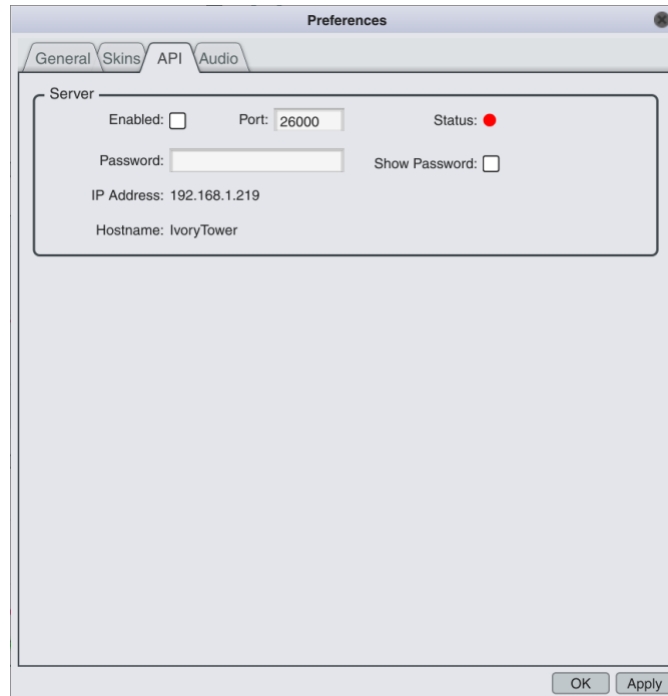


Figure 24 - API settings for SPL webviewer

Logging must be running for the API to be enabled. Once logging has begun and the SPL History Timeline starts to populate with log data, open the *Options* menu and choose *API*. If desired, you can enter a password. Client machines will be prompted to enter the password when connecting to the webserver. Then select the “Enabled” checkbox to start the webserver running. The status indicator circle will turn green.

The IP address field displays the host computer’s IP address. If the machine has multiple valid IP addresses or network adapters, all of them can be used to connect to the Smart API. Hovering your cursor over the displayed IP address to show a popup displaying all valid IP addresses for that machine.

Connect to the web viewer from client devices by opening a web browser and entering either the IP or hostname, followed by a colon and the port number, as displayed in the API options tab, for example “192.168.1.117:26000”

You can test the web server by connecting to it from a web browser on the host machine by entering the hostname “localhost” and the port (localhost:26000).

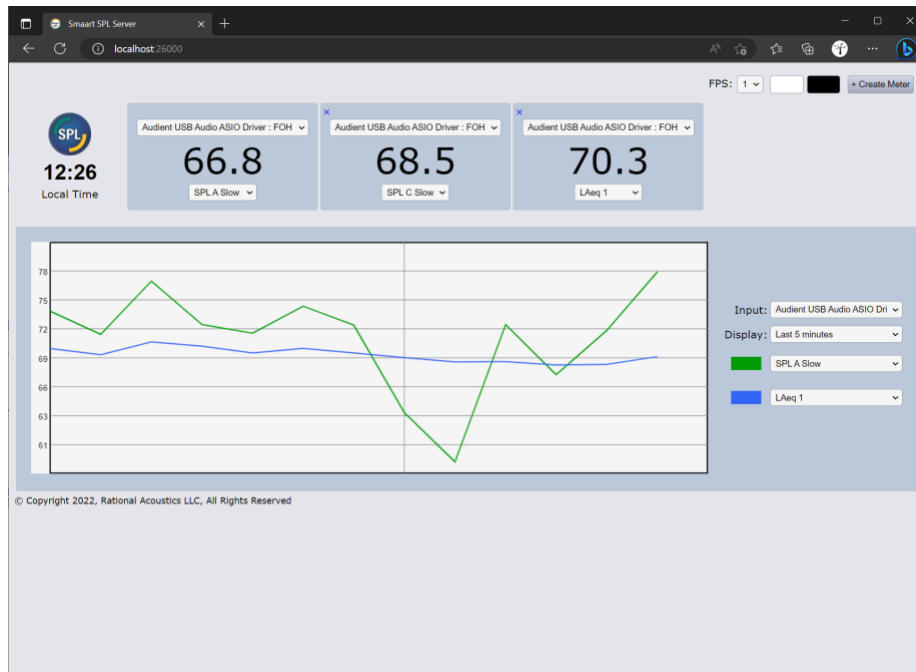


Figure 25- SPL Webviewer running in a web browser

We recommend using your web browser's private browsing / incognito mode, as it disables scripts and plugins that can interfere with the API's functioning.

Multiple client devices can connect to a single instance of the Smaart SPL webserver, and client devices can connect multiple browser tabs or windows to multiple instances of Smaart SPL, for example, to view levels on multiple stages at a music festival.