VERSION 1.0

Dirac Live 2.x Installer Manual

Dirac Live

StormAudio

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Introduction

Hello and welcome to the exciting world of the Dirac Live® 2.x Room Calibration Suite™. StormAudio and Dirac Research are delighted to offer you this software that stems from many years of experience in sound system tuning , extensive research, and development. We understand the listening room is usually the weakest link of any sound system: Loudspeakers have been designed for a certain placement and room, but when used in an imperfect room environment, performance suffers. Even the loudspeaker design itself is often a compromise between aesthetics, sound quality, and cost. Imaging, clarity, and bass tightness are examples of properties typically affected by these shortcomings, with the result that music is not perceived as intended by the artist. Dirac Live® 2.x Calibration Tool™ for StormAudio is our response.

The software suite is tailored and optimized for your Home-Cinema or Music room to offer best possible performance. Version 2.x is built to work with Mac or PC and can be used for multiple installations along with supporting the entire line of StormAudio processors. Now it is easier to measure the acoustic characteristics of your loudspeakers and room, and calibrate them to sound as intended – without unwelcome distortions.

Optimization done on the Dirac Live® 2.x Calibration Tool™ are loaded to the ISP and I.ISP to create Audio Profiles so that it can be used to optimize your sound in real time by means of Listening Presets creation and selection via the Remote Controls.

Quick Guide

The following steps will get you started immediately. Please see the Step-by-step guide if you need more details. We strongly suggest you have the ISP and I.ISP Installer Guide with you during the Dirac Live Calibration process.

1) Visit https://live.dirac.com/register/ and create an account.

2) Login and download Dirac Live® 2.x Calibration Tool™ for your Mac or PC

3) Unzip, Install, and run the Dirac Live® v2.x.x Setup

4) Create your Theater or Audio Zone into ISP & I.ISP Web Configurator.

5) Configure your Theater in terms of bass management and manual speakers' corrections (such as manual parametric EQ or crossovers) only.

6) Group your multi-way channels into one speaker group and subwoofers are put together into its own group, to make sure Dirac Live Calibration Tool sees this as single Speaker.

7) Connect you Speakers to your I.ISP or your Amplifiers and adjust the Master Volume to -40 dB.

8) Press on the DIRAC button (Speakers page still in EDIT mode) which will open a new pop-up.

9) Select Start New Calibration: the ISP / I.ISP will enter the Dirac Calibration mode.

10) Connect your UMIK-1 microphone to the computer's USB port (or your own microphone system and sound card when not using UMIK-1).

11) Start StormAudio Dirac Live® 2.x Calibration Tool™ on your computer and login with your Dirac Live® account.

12) Follow the on-screen instructions to select the device to calibrate, select your microphone input and microphone calibration file, adjust the microphone and output levels, measure and optimize your sound system. Make sure to save the project regularly to be able to resume your work and fine-tune your settings!

13) When you have saved your filters from the Dirac Live® 2.x Calibration Tool™, proceed to the last step of exporting filters to the ISP or I.ISP processor.

14) Exit the Dirac Live® 2.x Calibration Tool™ and go to ISP or I.ISP Web Configurator to finalize the new audio profile creation.

15) Save your Profile and Theater to exit the Speakers Edition mode.

16) Create your Listening Preset to make use of the desired audio profile.

Step-by-Step Guide

This guide will go through each step in more detail to achieve the best possible sound.

Need more thorough explanations? Refer to Advanced Deep Dive (More In-depth) where some important concepts are discussed.

If you have any problems with the Dirac Live® 2.x Calibration Tool™ or the ISP and I.ISP processors, then some possible solutions are addressed in the Troubleshooting section at the end of this manual.

Software Installation

The StormAudio Dirac Integrator Kit comes with a USB microphone that doesn't require specific drivers to be installed manually. This is currently supported natively in Windows 7 and upward, as well as on Mac OSx. If you are using your own microphone system and sound card with your computer for the first time, then please read the installation instructions for your sound card. You may be required to install driver software before plugging the sound card into the computer.

It is also recommended to update any sound card drivers before proceeding.

You will need to create an account on Dirac Live®'s Portal in order to download the Dirac Live® 2.x Calibration Tool™ suite. Point your web browser on your computer to:

https://live.dirac.com/register/

Enter the required information to create your account and click the Register button at the bottom of the form to register. Once you account is created, you will then need to login to your account. You can log into your account by accessing the following URL address on your web browser to:

https://live.dirac.com/login

Once you have logged into the Dirac Live Website, you will need to download Dirac Live® 2.x Calibration Tool™ Software Package to your computer. Click on the DOWNLOAD button in the middle of the page, and then scroll down to the section marked Desktop. Depending on the type of computer you are using you will have a choice between Windows or Mac Download. Click on the correct download that matches your computer. The download will start once you select your option. **Additionally, you can find the tested and approved release at StromAudio.com website in the client portal.**

Run the download installer package (ZIP) file on your computer. Please follow the displayed directions to complete the install.

Please insure you do not run the Dirac Live 2.x® Calibration Tool™ software at this point, as we need to insure the microphone is setup first.

Equipment setup

Using the ISP Installer Guide delivered with your unit or available on the dealer portal, configure the processor from Steps 1 to Step 6:

- System setup and Input definition.
- Theater and Audio Zones creation.
- Speakers connection to the ISP & Amplifiers or to an I.ISP .
- Setup of the Speaker Groups for DIRAC and start DIRAC process in ISP pressing the DIRAC button.

Connect your microphone. If you are using the UMIK-1 USB microphone delivered with the Integrator Calibration Kit, just connect it to a USB port on the computer. Take note of the Serial Number written on the microphone body.

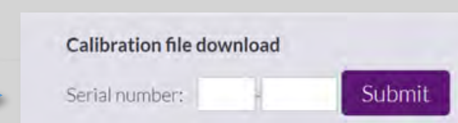
Otherwise, connect the microphone to the sound card input. Some microphones require 48V phantom power. Be sure to follow the microphone usage instructions.

Avoid microphone feedback! If your sound card has a Direct Monitor function, this must be turned off. Some sound cards have a physical volume knob for direct monitor, in this case it has to be turned down to zero level.

Some professional sound cards have sound effects like compression. Double check that any sound effects are turned off.

Download the calibration file of your UMIK-1 microphone using the following URL and by entering your unit UMIK-1 SN# in the calibration file download part:

https://www.minidsp.com/products/acoustic-measurement/umik-1



Two download files will be offered to save. The one with “ 90degree” is to be used in case you measure with the microphone set up vertically/upward, so to use it with a constant directivity profile. This is the typical use case in multichannel systems.

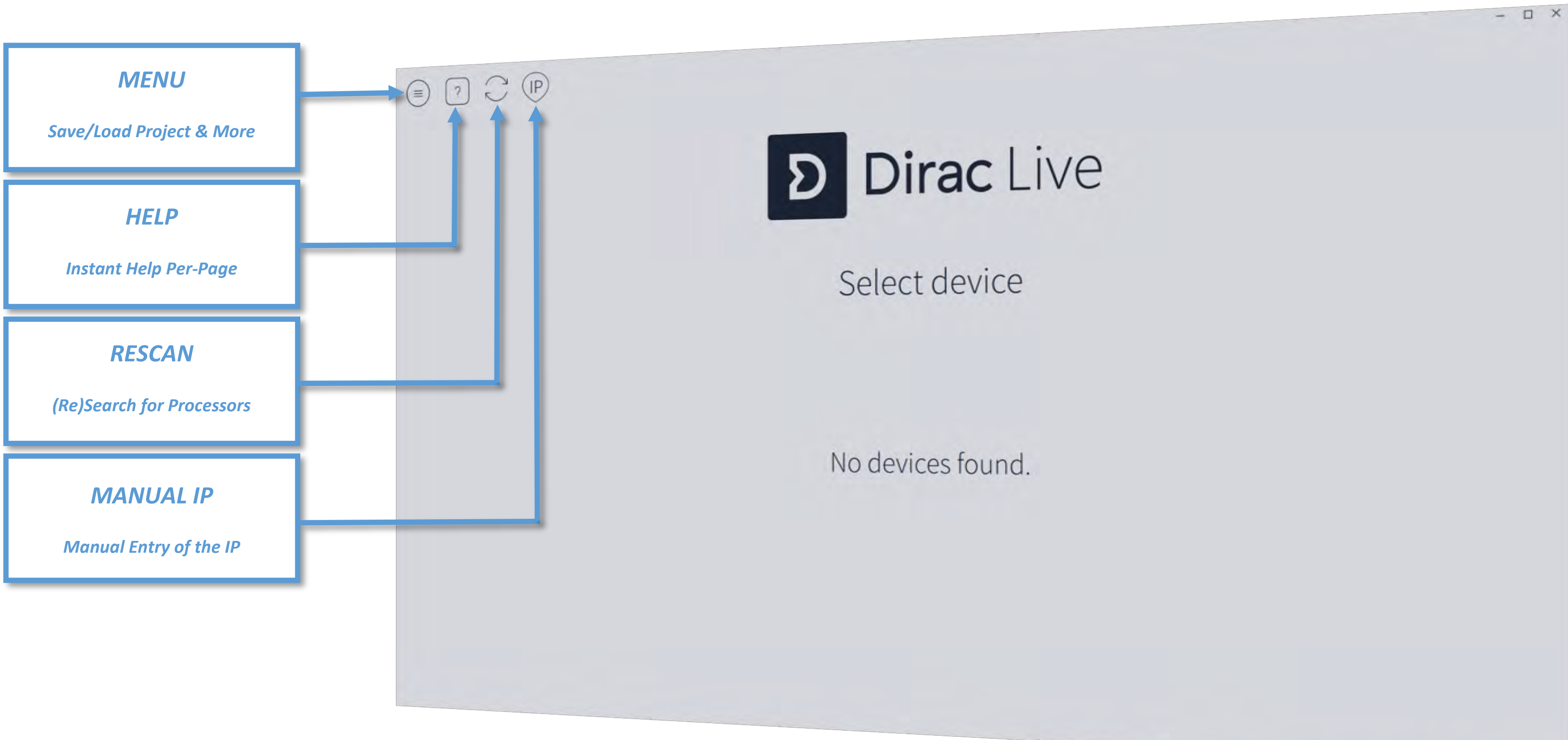
Before continuing, please close all other audio applications on your computer. Double click the Dirac Live icon to run the program if it has not already opened after install. The Dirac Live® 2.x Calibration Tool™ software suite may automatically log you into the software, however if it does not you will need to enter in the credentials that you used to create your account with in the previous step.

Using the StormAudio Dirac Live 2.x Calibration Tool™

The Dirac Live® 2.x Calibration Tool™ measures the acoustical properties of the sound system and designs customized Dirac Live® filters for your room and loudspeakers.

Dirac works off a cloud based system, so the computer must be connected to the Internet during the complete use of the Tool this includes the measurements, filter optimization, and export. If the computer is not connected during the measurements, then certain parts of the optimization will be delayed until a connection is available. To save time, it is recommended having a connection during the whole process. Any active firewalls have to allow HTTP.

The work flow is setup in a wizard style format. Below outlines each icon of the Dirac Live® 2.x Calibration Tool™



Select Device

Icons and information shown above may be different based off your installed devices and theater setup.

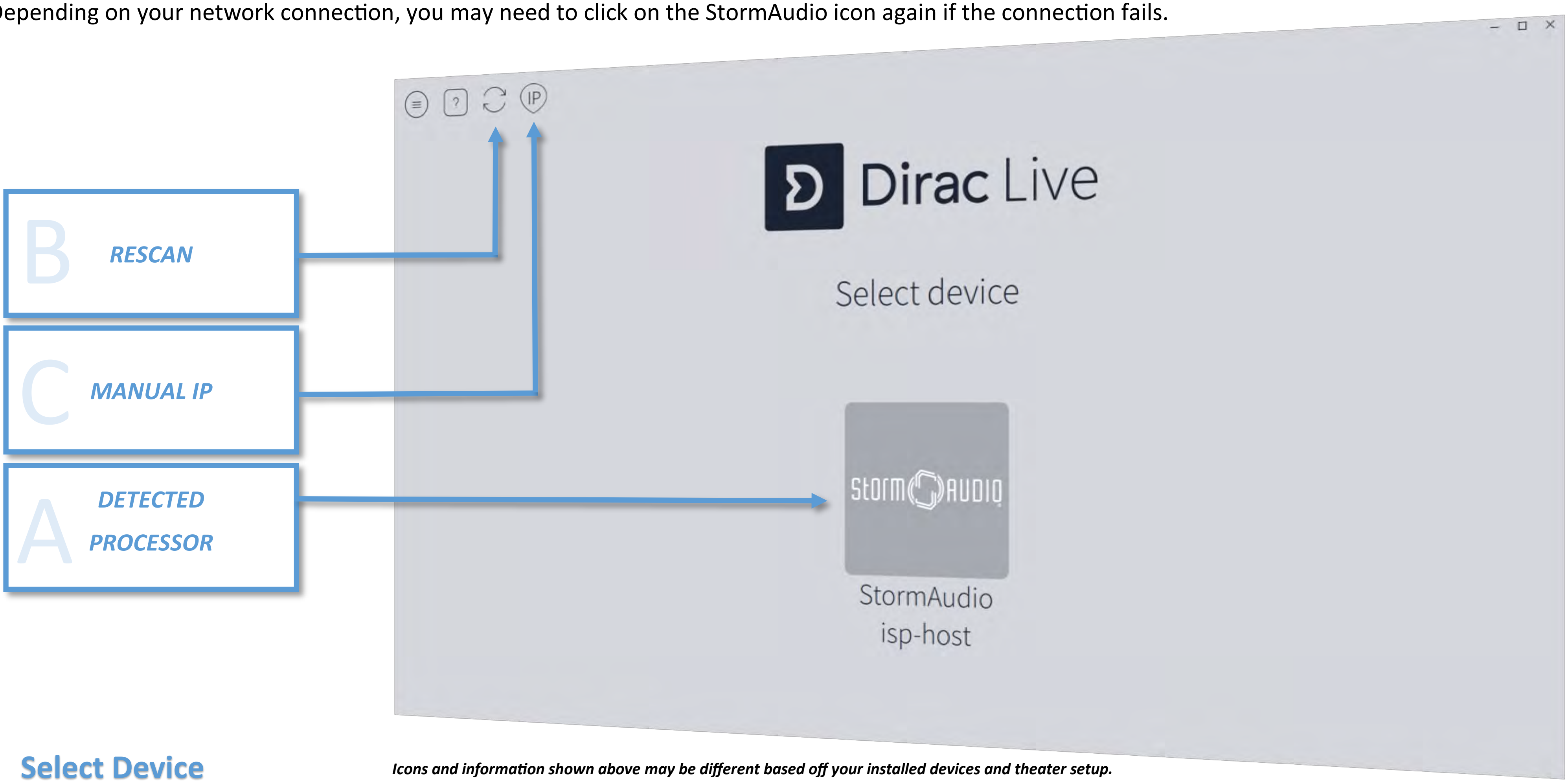
When launching the Dirac Live® 2.x Calibration Tool™, the software will scan the network to check whether an ISP or an I.ISP is in Dirac Calibration mode. When unit(s) have been found, they will be displayed in the lower center part of the application (A) and moving your cursor over the StormAudio logo highlight in black.

Select the device that you want to use for playback in the "Test signal playback device" by clicking on the StormAudio icon (A).

Clicking the "Rescan" icon (B) will check the network again for available units. In some network, due to router settings, the UPNP mechanism used to detect the ISP / I.ISP within the Network might fail. You can override this mechanism by clicking on the manual IP icon (C) and enter the ISP / I.ISP IP address that was used to access the Web User Interface. Click Rescan to initiate the connection and the device will be shown in the "Test signal playback device" list.

Clicking on the StormAudio icon (A) will continue you to the microphone selection.

Depending on your network connection, you may need to click on the StormAudio icon again if the connection fails.



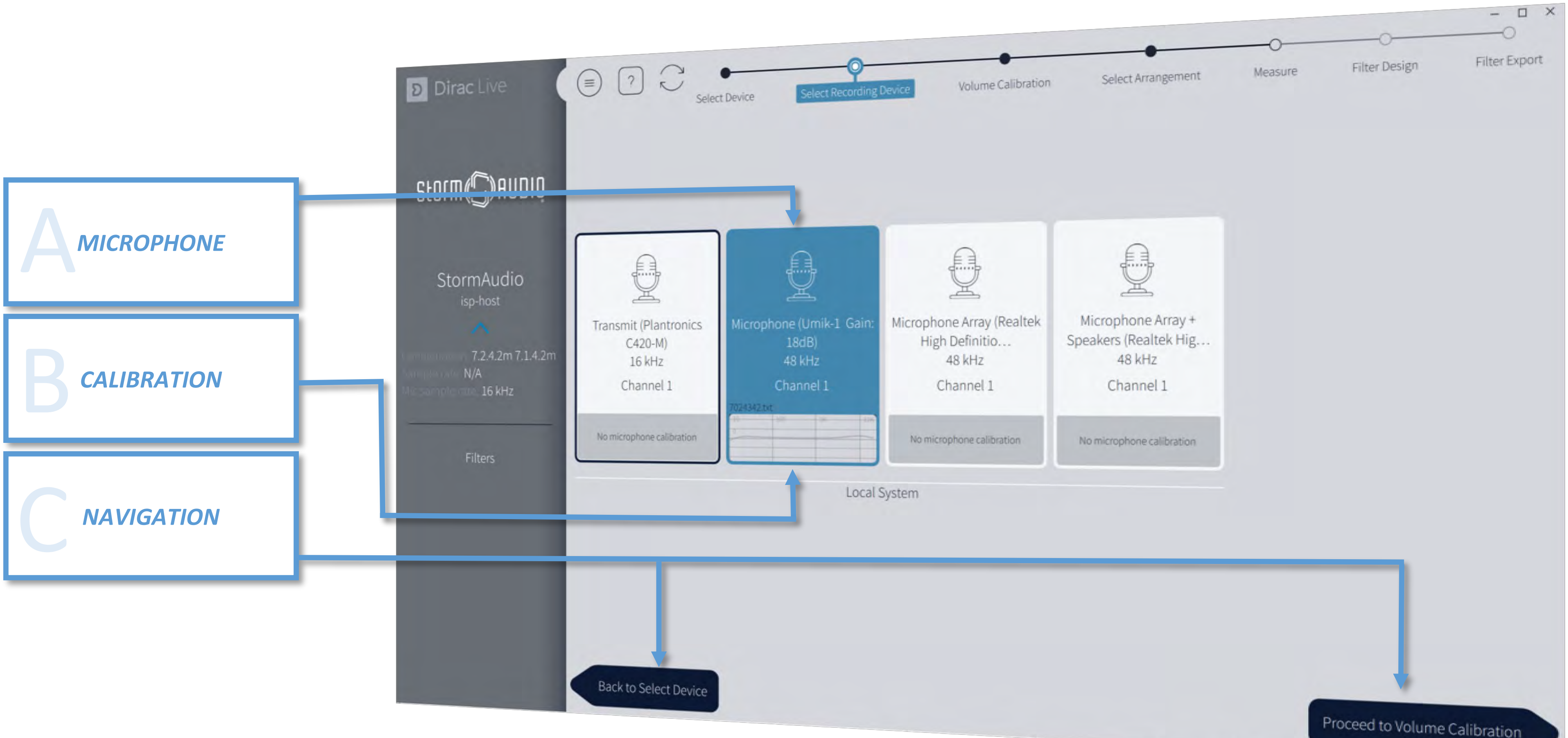
Select Device

Icons and information shown above may be different based off your installed devices and theater setup.

In this section you will select the recording device to take the measurements from the speaker channels. If you are using the Umik-1 Microphone, the calibration file will need to be loaded into Dirac Live® 2.x Calibration Tool™ before selecting the microphone. Click the grey box (B) below the microphone selection icon to load calibration file that was downloaded in the previous step (Page 6). The file name should end with "_90degree.txt" if you are calibrating for an immersive sound stage.

Once you have loaded the calibration file you will see the grey box change from "No microphone calibration" to graph icon representing the file has been loaded (A). Click on the microphone icon itself to select the microphone you will use for the calibration. The image will outline it self in blue or black to show it has been selected.

Use the navigation buttons below (C) to advance (and return) through the calibration process. Click on "Proceed to Volume Calibration" to move to the next step.

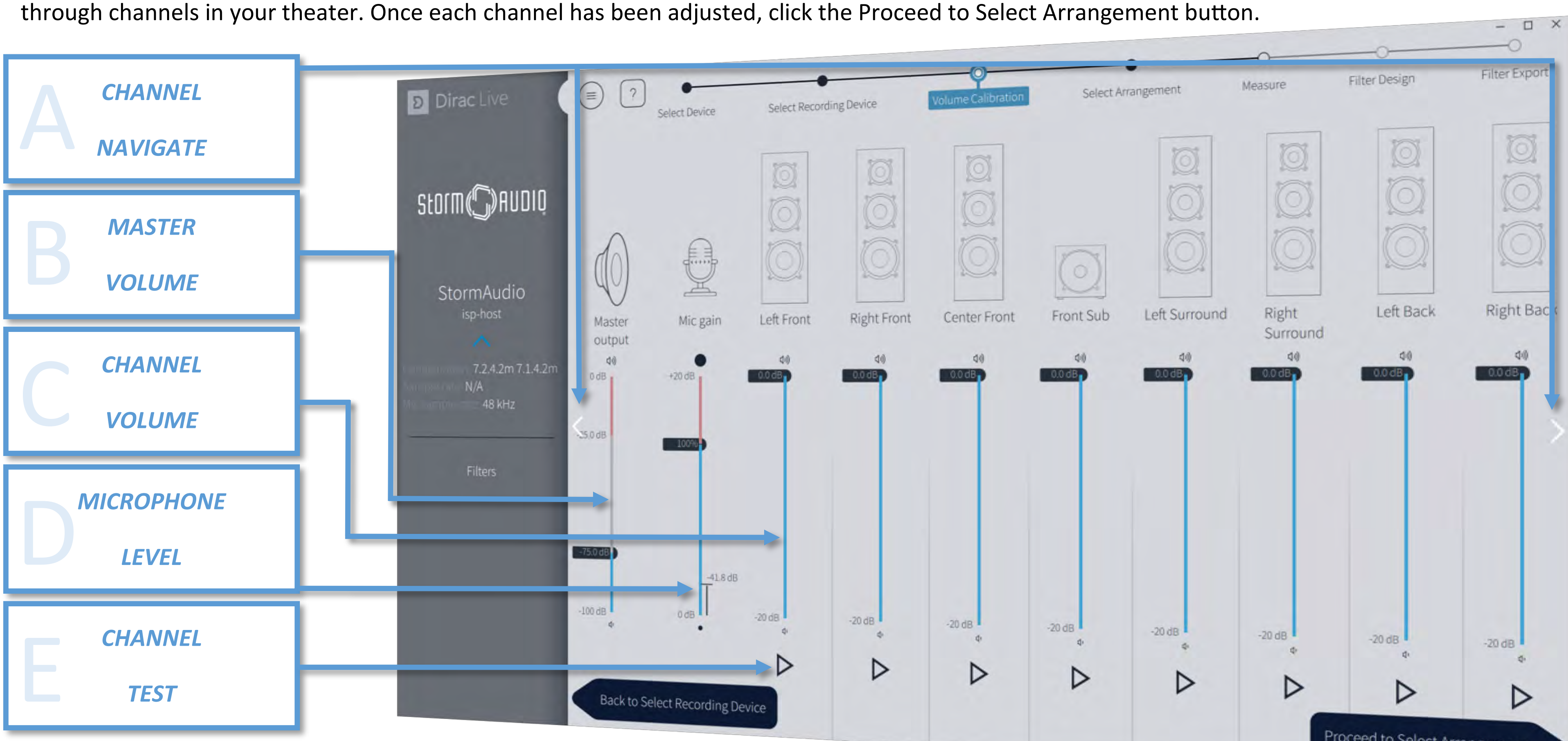


Select Recording Device

Icons and information shown above may be different based off your installed devices and theater setup.

In order to get an accurate measurement of the sound system, some initial adjustments of the playback and recording levels are required. Start by placing the microphone inside the center listening area. Channels as defined in the played theater are listed (C) here.

1. Adjust the Input Gain of the microphone. When using the UMIK-1, adjust its level (D) at 100% (just before the red zone).
2. To protect your ears and equipment: Turn down the volume on the ISP to below -40 dB (B). Start playing a test noise in the selected output channel by pressing the Test button (E), usually start with the channel at the furthest away from the microphone. Gradually increase ISP volume (B) until it channel level is in the green zone. If you can not reach the green zone without the test sound being overly loud, you can increase the mic gain (D).
3. Play the next channel away from the microphone and verify the level. In case level is too high, adjust the played channel using the “Channel volume” trim (C). In case the level is within the green area, keep the level unchanged and test the next channel. Avoid having the level fall in the yellow or red areas as this will cause the sample audio test to clip or produce too soft of a response. It is always preferred the level adjusted slightly lower expected average to avoid clipping, as measurements would fail. Use the right and left navigation arrows/chevrons (A) on the left and right of the page to scroll through channels in your theater. Once each channel has been adjusted, click the Proceed to Select Arrangement button.



Volume Calibration

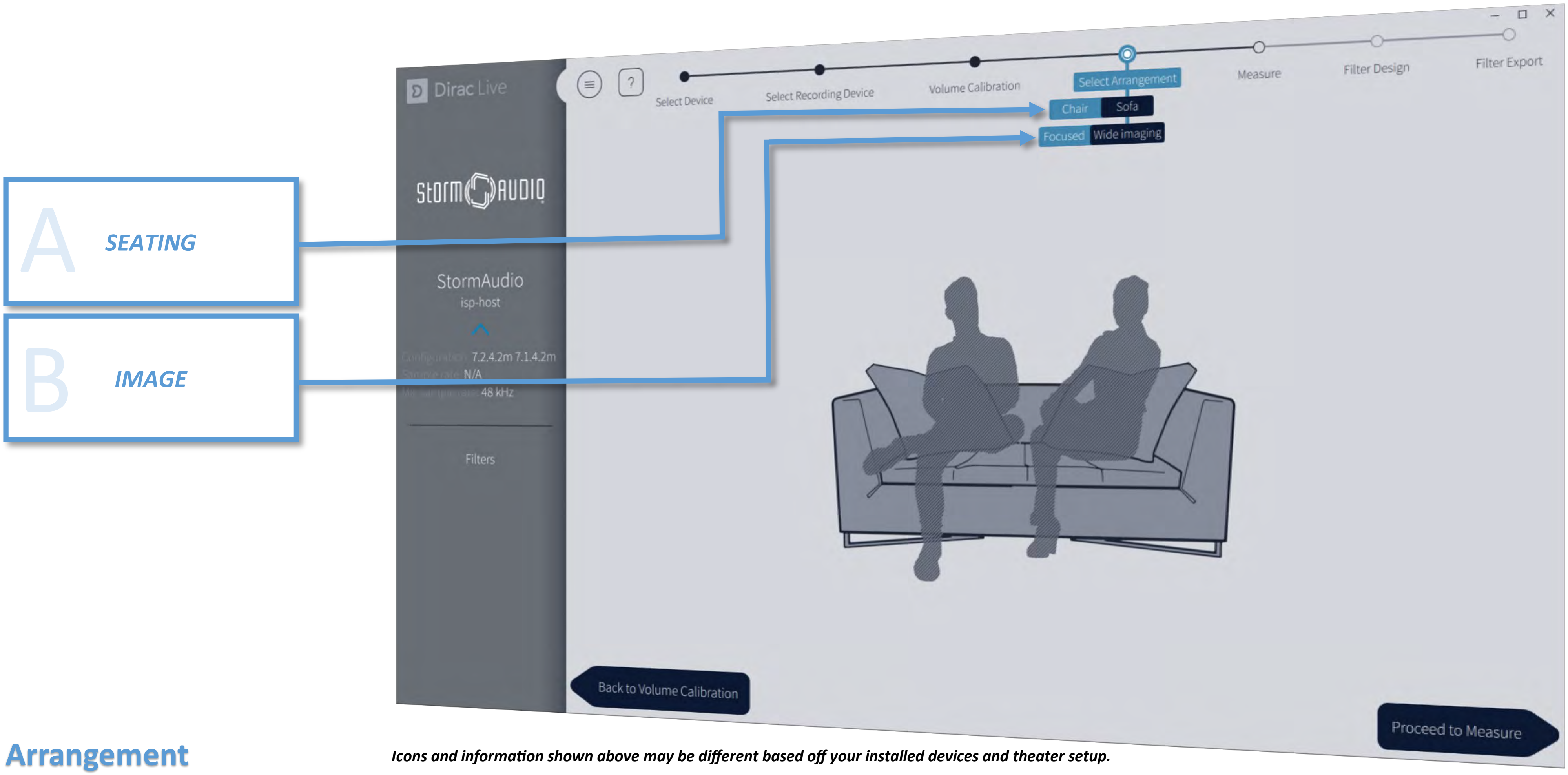
Icons and information shown above may be different based off your installed devices and theater setup.

It is now time to acquire the acoustical models by measuring the sound system in the listening room. The accuracy of the models is very important for designing optimal room correction filters. On this screen, the correct seating arrangement needs to be selected.

Select the most suitable listening environment preset by clicking on the corresponding choice (A). If the listening environment comprised mostly of theater seating, it is suggested to select, Sofa and Wide Imaging (B) as shown below. In general, it is important to collect measurements in the most likely "listener's head" positions (sitting, standing, leaning forward, etc.).

Now would also be a good time to save your project before proceeding with the measurements. You can save the project at any time by clicking the menu icon in the upper left corner (see page 7) and then "Save Project".

Once you save the project, click Proceed to Measure in the lower right corner.



Icons and information shown above may be different based off your installed devices and theater setup.

Arrangement

Use a microphone stand to firmly place the microphone in the indicated positions found in the next step.

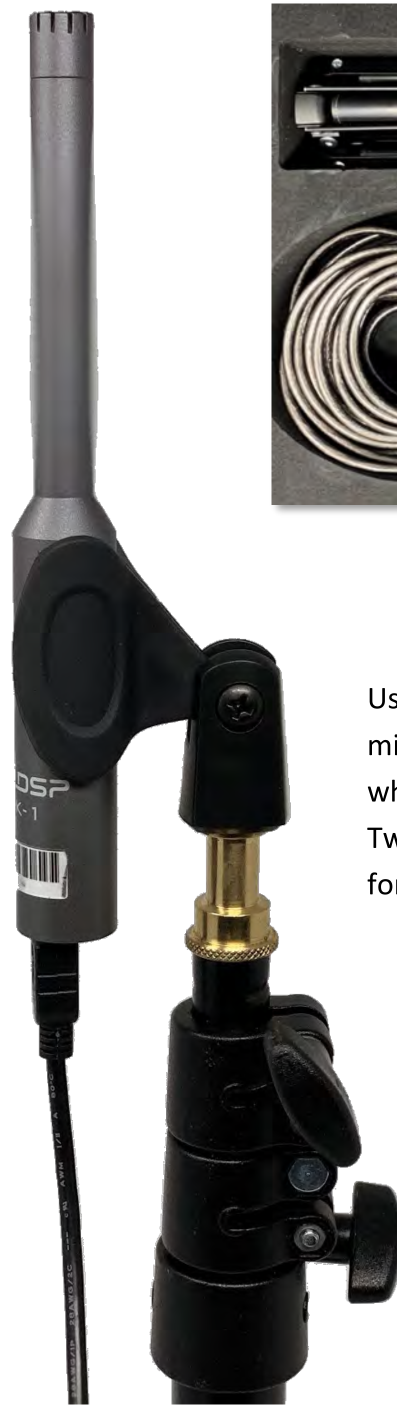
In an immersive audio calibration, the height of the tip of the microphone should be relative to average human head position at ear level.

Direct the microphone upward, pointing to the ceiling, to get the most omnidirectional recording of the room response, or towards the speakers depending on the particular microphone and its calibration file (if any). Please follow the instructions from your supplier and make sure to use the correct calibration file in case of UMIK-1 (upward = xxx_90degree.txt" file, toward "xxx.txt" file, xxx being the serial number of the UMIK-1).

Minimize external noise such as talking, opening or closing doors or windows, air conditioning systems, refrigerators, ceiling fans, and playback of sounds during the measurements. This is to avoid interruption in the measurement process and corruption of acoustical model data.

Turn off any computer programs that may make any noise, such as Skype or your email client.

Turn off any sound card effects such as "low-cut" or compression.

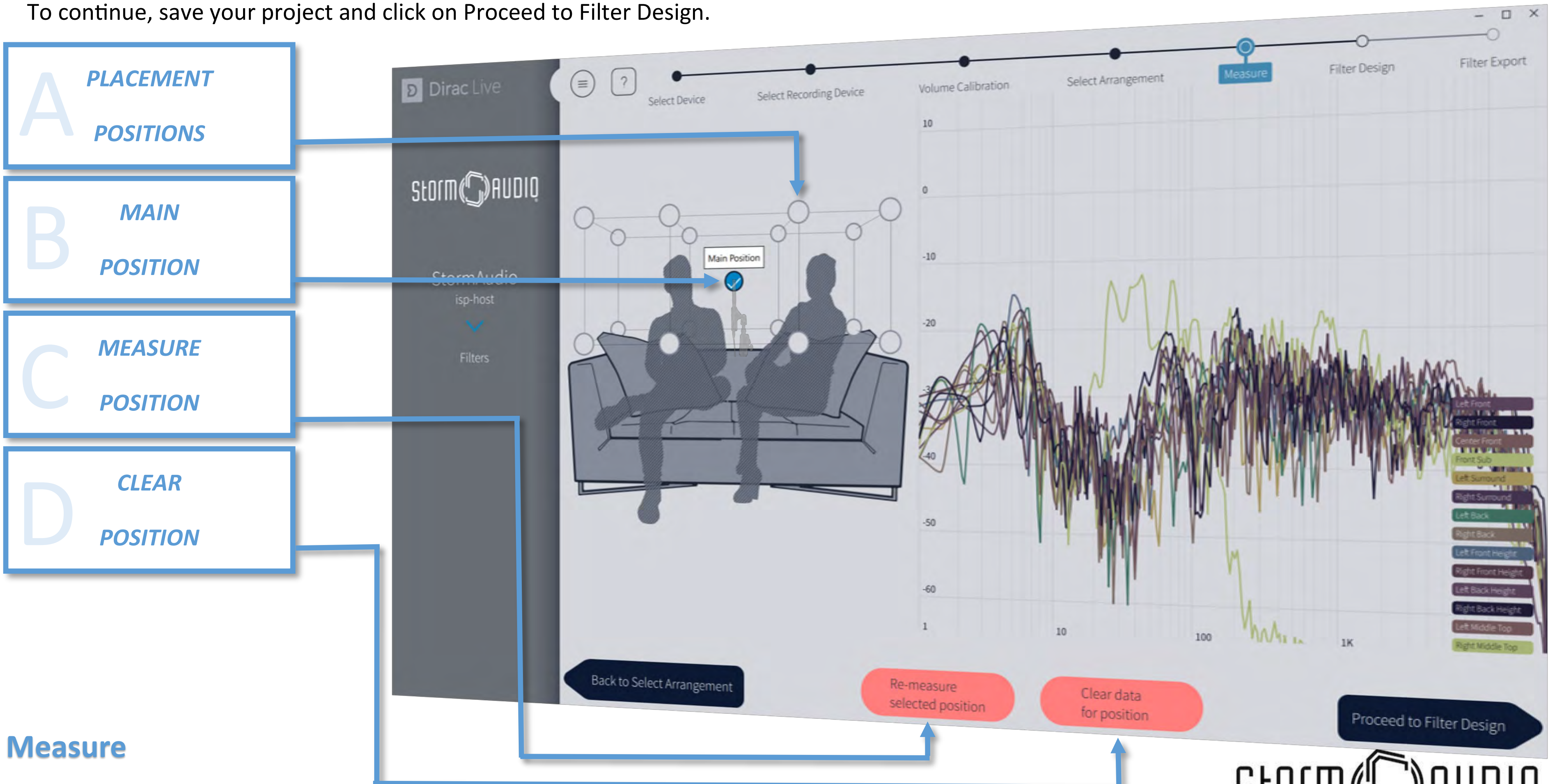


Use a stand that allows you to vary the height of the microphone position within the listening volume, which is important to get an accurate acoustical model. Two types of stand are provided in the integrator kit for maximum flexibility.

Avoid making measurements in a too small space. Even for the "Chair" listening environment, it is important to spread out the microphone positions in a sphere of at least 1 meter of diameter. A too small space will result in over-compensation that will sound very dry and dull.

1. The first measurement should always be taken in the center of the listening region (B), in the desired "sweet spot", as this will be used for alignment of levels and delays between loudspeakers.

2. Click on Measure selected position (C) to collect a set of measurements. This will play a sweep in each loudspeaker and one final sweep in the first loudspeaker again. If the measurement was successful, then the position indicator will have a check (or tick) mark in the circle. Repeat the procedure until at least nine (9) of the seventeen (17) positions have been covered (A). The more measurements taken the better the results. If your sound card or microphone pre-amplifier has a clip indicator, then keep an eye on the microphone input. If it clips during measurement, then re-take that measurement. Usually, the clipping will be detected by the Dirac Live® 2.x Calibration Tool™, but depending on level settings, the clipping event may fail to be detected.) If the measurement failed, then usually an error message will be given. Follow the instructions in the error message, or refer to the troubleshooting section Recording problems on page 23. You can always retake the measurement by click on Clear Data for Position (C) at anytime. To continue, save your project and click on Proceed to Filter Design.



Measure

Icons and information shown above may be different based off your installed devices and theater setup.

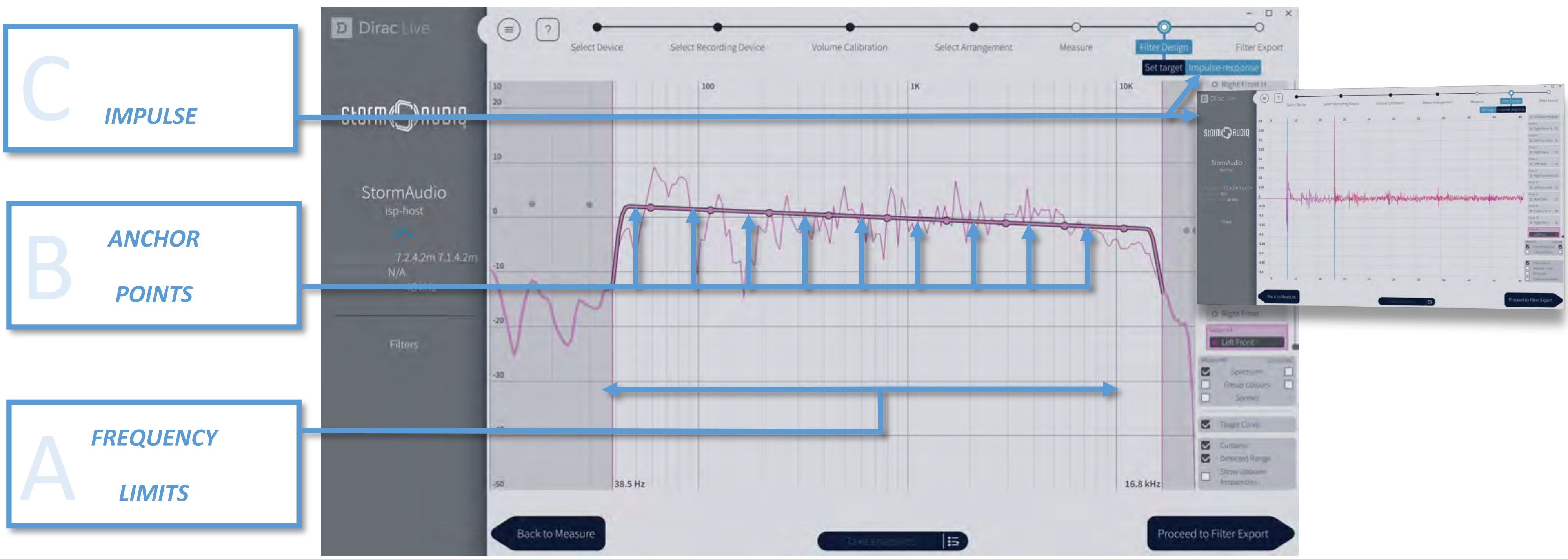
The filter design algorithm uses the measured data together with a target frequency response to calculate filters that optimize the frequency response and the impulse response of the sound system. Finding the best target frequency response may require a little experimenting from your side. After completing the measurements, the Dirac Live® 2.x Calibration Tool™ will automatically generate a suggested target curve that matches the measurements. This, however, can often be tweaked for even better performance.

There are two ways to modify the frequency response:

1. Dragging the target frequency curve at the anchor points (B). You can add (or delete) anchor points by right clicking on the target curve (or on the points) (B).
2. Dragging the frequency limits on either side of the frequency range (A). The resulting filter will leave the audio signal unmodified in the shaded frequency regions (A) or curtains. This can be useful if you, for example, are perfectly happy with the high frequency performance, but want to address room resonances in the bass.

Refer to Target curve design guidelines on page 20 for advice on how to design a good target response. To zoom in, draw a box around the area that you would like to zoom into. To zoom back to default level, double-click in the graph.

To view the impulse response of the system, before and after correction, click on the "Impulse response" toggle button (C). In the impulse response view, this button will highlight and "Set target" can be clicked on it will take you back to the frequency view.



Filter Design

Icons and information shown above may be different based off your installed devices and theater setup.

Dirac Live® 2.x Calibration Tool™ supports speaker grouping. Speaker grouping allows each group to have a signal common target curve for all speakers in that group. Many times in a theater there are speakers that are the same model numbers and share the same characteristics. For example the Front Left and Front Right speakers may be identical model numbers, and so will have the same manufacture specifications. These two speakers can now be grouped in order to take advantage of collectively sharing the same target curve and phase matching. On the far right of the page, you will see the list of all the channels in the Dirac filter design (B) . Each channel will have its own group number. To move a channel to another channel's group, you simply drag and drop the channel you want to move into the channel group you want it to be part of. In the example below, you can see that group number three (3) has the front left and front right speaker together (A). Also you can see all the top channels have been grouped into group number four (4) (B).

Dirac Live® 2.x Calibration Tool™ pre-optimizes and corrects the target curve at this point. To view the difference between what was measured and what was corrected, click on the check (tick) box on the right side of Spectrum (C). Click the check (tick) box on the left of Spectrum to toggle in and out of measured channel (D).

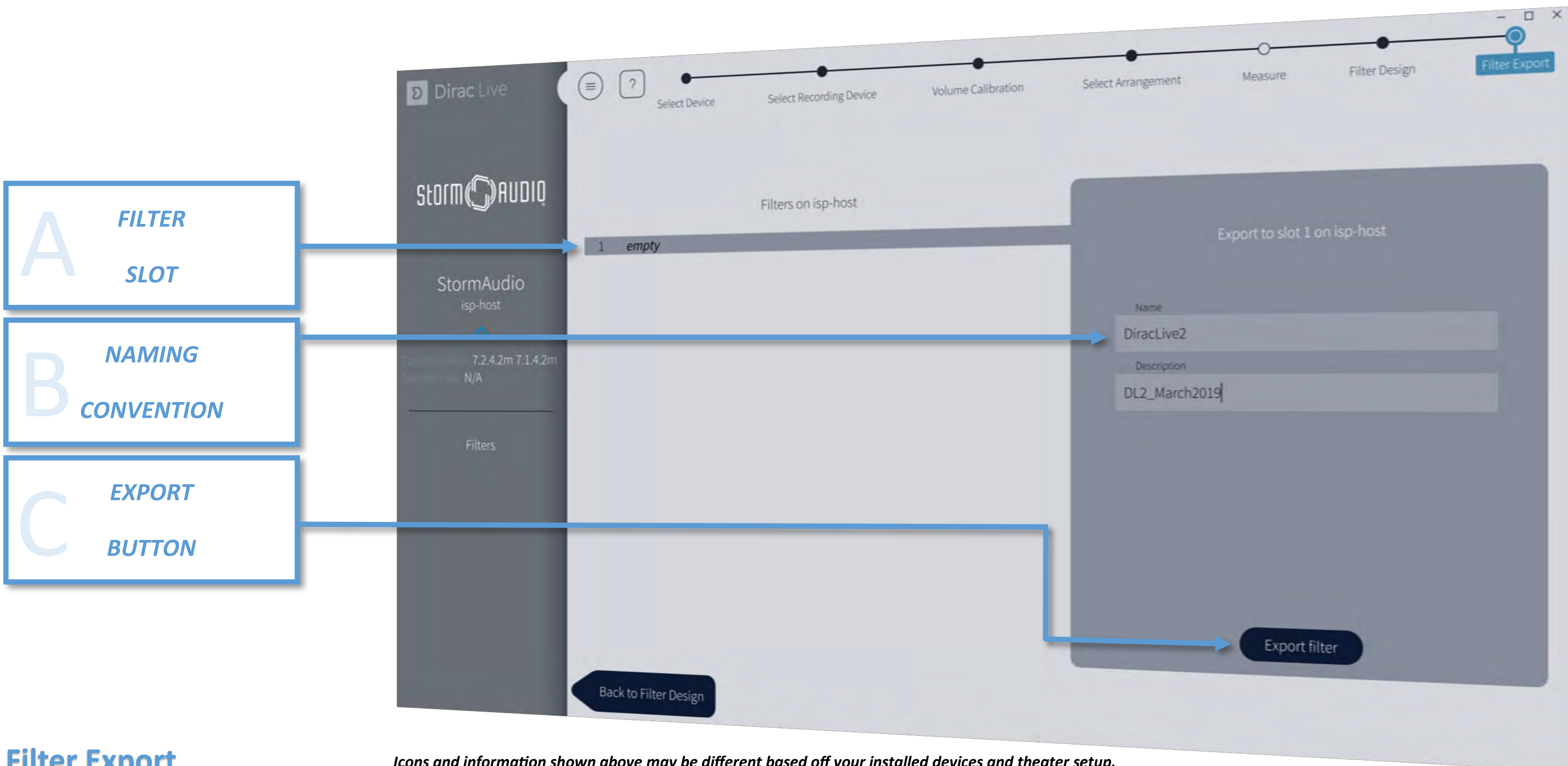
It would be strongly suggested to save your project at this time. Click Proceed to Filter Export to continue.

Icons and information shown above may be different based off your installed devices and theater setup.

To Export the newly created filter to the StormAudio processor, a name and description needs to be created before it can be exported to the processor.

Click on the empty slot (1) in the middle of the page to expand the naming convention dialog boxes (A). There will be two fields to enter in the desired name and description that will carried over and stored in the ISP and I.ISP profile (B). The StormAudio processor supports a certain amount of characters per field. The name field can be a maximum of 13 characters, and will be the name of the profile. The description box can be a maximum 15 characters, and is used to easily identify what the calibration is. Dirac Live® 2.x Calibration Tool™ will allow you to enter more than 15 characters in each field, however the StormAudio processor will truncate any characters after the 13 and 15. Try to avoid special characters, as these characters may fail to transfer over to the ISP. Once you have inputted your naming selection, click the Export filer button below to send the filter to your processor (C).

At this stage, the design phase is completed and you can exit with the button Exit [X] to complete the New Dirac Audio Profile creation on the ISP & I.ISP configuration page.

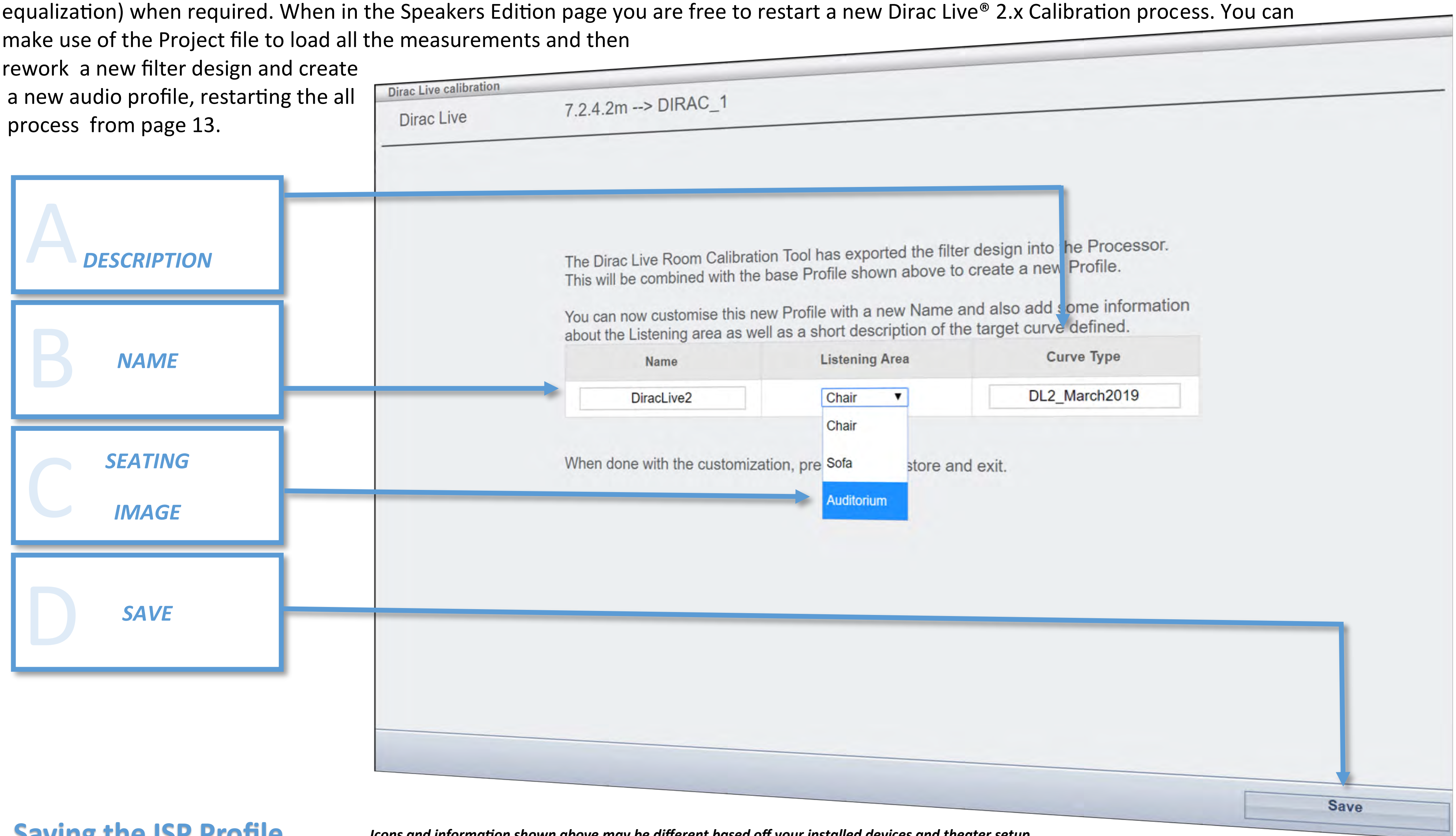


Filter Export

Icons and information shown above may be different based off your installed devices and theater setup.

When exiting the Dirac Live® 2.x Calibration Tool™, you will go back to the web user interface of the ISP & I.ISP opened in your internet browser. Since the filter design has been loaded to the ISP & I.ISP, you will be able to finalize the new audio Profile creation. For a Theater, if the name convention failed to copy over correctly you can rename the Profile under name (B) and enter a description (A) for your curve type. You can select which Listening Area was selected for the measurements between Chair, Sofa or Auditorium (C). Use Auditorium if your seating selection was Sofa with Wide Imaging.

When everything is defined, you can then SAVE (D) to go back to the Profile Speakers Edition page where you will be able to do listening tests and potentially make a duplication of this new Profile for additional manual modifications (Bass Management tuning, delay fine tuning or even manual parametric equalization) when required. When in the Speakers Edition page you are free to restart a new Dirac Live® 2.x Calibration process. You can make use of the Project file to load all the measurements and then rework a new filter design and create a new audio profile, restarting the all process from page 13.



Saving the ISP Profile

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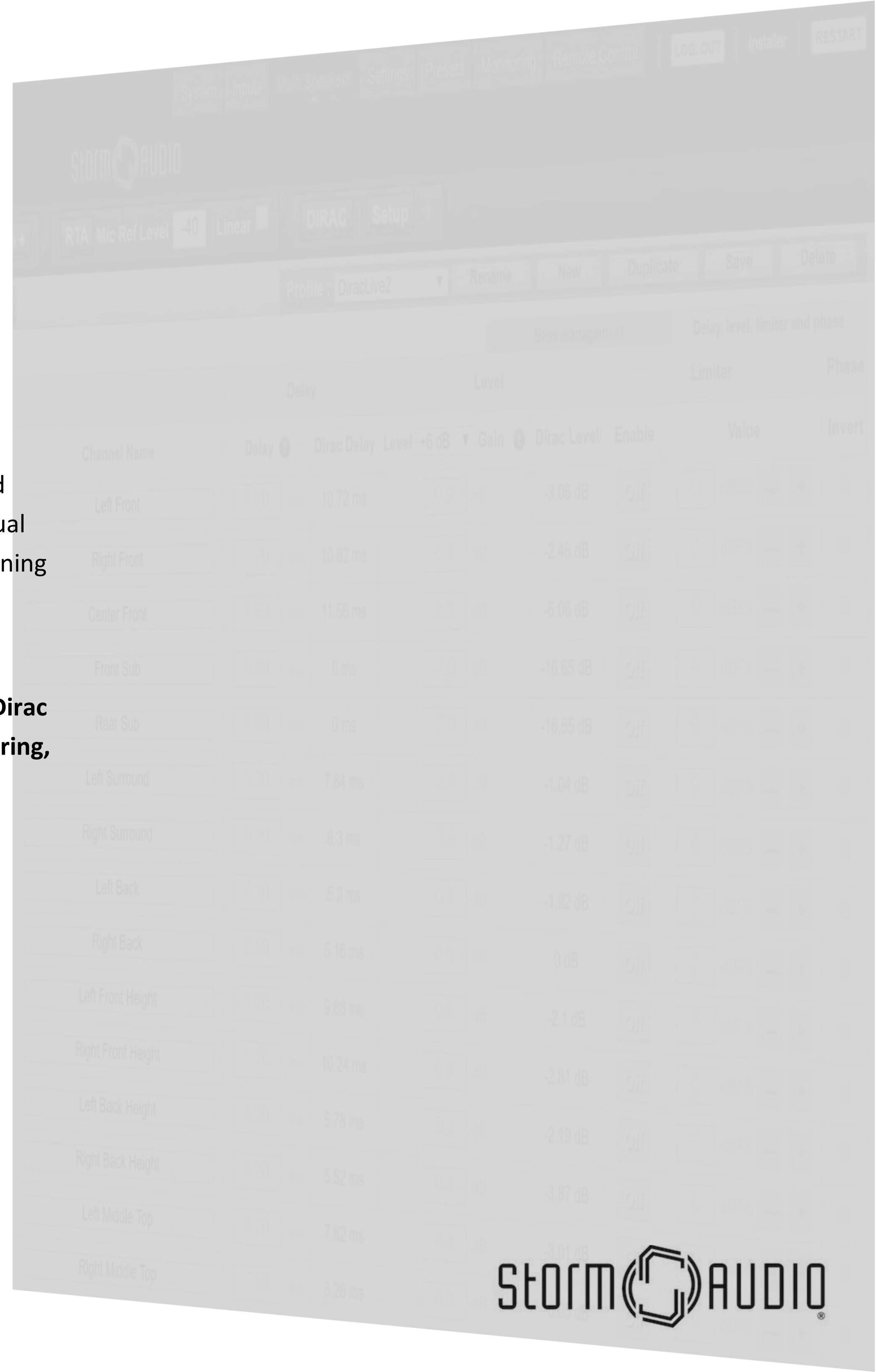

Dirac Live® 2.x Calibration Tool™ will design filters that play on magnitude and impulse speakers response. It is a complex mechanism where additional manual correction can degrade the overall listening experience or provide a poor listening experience, if not done cautiously.

Please see the next section on Advanced Deep Dive (More In-depth) about Dirac and StormAudio. It will cover more details and better understanding on filtering, target curves, impulse response, and more.

This concludes the full step-by-step for Dirac Live® 2.x Calibration Tool™ .

Advanced Deep Dive

This section will provide a more in-depth understanding of Dirac Live 2.x and Calibration.

Impulse response

The impulse response measures how a transient such as a drum beat is reproduced in the room. Reflections, diffraction, resonances, misaligned drivers, etc., combine to smear out the transient. As the name indicates, the impulse response shows how the system and room responds to an impulse input signal. An ideal speaker adds no ringing, coloring, or time smearing to the recording, so its impulse response should look just like the input impulse.

The similarity between the different loudspeakers' impulse responses at the listening position affects the perception of stage and image. The more similar the impulse responses, the easier it is to trick the brain that the loudspeakers are not really there, and that the sound emanates from a virtual stage spanned by the physical speakers.

The performance of Dirac Live® 2.x is quite unique in that it is so successful in making the system response resemble that of an ideal speaker. This is thanks to robust design of so-called mixed-phase filters. The result of applying Dirac Live® 2.x can be viewed in the Dirac Live® 2.x Calibration Tool™, by clicking the IMPULSE button in the Filter Design page.

Magnitude response

The magnitude response (a commonly used but imprecise term is "frequency response") measures how much each frequency component, or tone, is attenuated. A good magnitude response should be as smooth as possible. Otherwise, different tones are reproduced with unequal strength, and the music is subjected to undesired coloration.

Mixed-phase filters

Infinitely many different filters can be designed to have the exact same magnitude response. They differ only in their impulse response. Therefore, it is useful to classify filters according to how their impulse responses behave.

Two commonly used filter classes in audio applications are minimum-phase filters and linear-phase filters. They are two special cases that are relatively easy to design, but that come with tightly constrained impulse response characteristics. A minimum-phase filter, by definition, is constrained to apply only the smallest possible delay to the signal given a desired magnitude response. A linear-phase filter, by definition, applies a delay which is constant across the whole frequency range. Therefore, neither of these two filter designs can make a desired change to the phase or impulse response, unless the desired change is exactly the particular change they make by definition. Minimum-phase and linear-phase filters may even worsen both the impulse response and the magnitude response of a system, simply by applying their magnitude response corrections at the wrong time.

A more difficult design task is to make a mixed-phase filter that matches a desired magnitude response while also having a customized impulse response.

A properly designed mixed-phase filter can make significant improvements to the impulse response of a sound system at the listening position:

- Misaligned drivers in multi-way loudspeakers can be corrected by automatically applying different delays to different frequency ranges.
- Energy from direct wave and early reflections can be optimally combined to arrive as a single wave-front to the listener.

Dirac Live® uses mixed-phase filters to optimize the impulse response while applying a desired magnitude response.

Target curve design guidelines

In the Dirac Live® 2.x Calibration Tool™, the desired magnitude response for each loudspeaker can be drawn manually. The reason for providing this feature, instead of just optimizing for a flat magnitude response, is that different loudspeakers and different rooms may require slightly different target curves for the best result.

Normally you want to have the same target curve for identical pairs of channels like the left and right front speakers. Therefore, these similar speakers should be grouped together in the Dirac Live® 2.x Calibration Tool™. Correcting for differences between the channels will often give a substantial improvement in the stereo image.

The shaded regions in the target editor illustrate regions where the magnitude response will be left untouched. These regions are adjustable by dragging in the handles on the shaded regions. Normally you want to correct the whole spectrum that is within the frequency limits of the loudspeaker. In case of high quality speakers, it might sometime be recommended to limit the correction window below 1 or 2 kHz to maintain the signature and timber of the speakers to the closest possible as originally, considering the room has limited effect in the high frequency range.

All loudspeakers tend to have a low frequency limit, below which the response is dropping rapidly. It is important to not boost the bass too much under this low frequency limit, because it can easily overdrive the loudspeakers and lead to distortion. Dirac Live® 2.x Calibration Tool™ tries to identify the frequency where the response starts to drop and the automatically generated target curve and correction frequency limits are adjusted accordingly.

If there is noise present during the measurements, this will often be visible in the measured frequency response as increasing low frequency levels below the loudspeaker low frequency limit. This is not part of the loudspeaker response so this frequency region is better left untouched.

Some guidelines to adjusting the target curve follows:

- Even small changes in the target magnitude response give more or less audible results. It is worthwhile to tweak the target to address even the smallest nuance in the magnitude response.
- Avoid sharp peaks in the target curve, as they may result in annoying tonal components in the frequency response. Generally, you want to have a smooth target curve.
- A target curve that is slightly tilting down towards high frequencies (like the auto-target) is often preferable - a flat target often sounds too bright. A loudspeaker with a flat on-axis response will usually have a slightly tilting in-room response. You can experiment with the amount of tilt, especially in the treble region, to get a good timbral balance in your room.
- If you have strong room-resonances in the low frequency region, completely eliminating these may make the sound thin in comparison. You may want to apply a slightly increasing low-end below 100 Hz.
- If you have dips in the magnitude response, sometimes it sounds better to not fill these in completely.

Loudspeakers in rooms

The topic of sound reproduction in small rooms, such as a living room as opposed to a concert hall, is huge. The room will define how reproduced music is perceived. Roughly, the effect of the room on the sound can be divided into two parts - the effect on frequency response (timbre, coloration) and the effect on spatial aspects of sound (spaciousness, envelopment, reverberation). Both parts are affected by speaker placement and listener position.

Loudspeaker placement

Correct placement of the loudspeakers in your room is vital to get the best possible sound. Therefore it is often worthwhile to experiment a bit with the loudspeaker placement. Always use the speaker manufacturer recommendations as a starting point. Using Dirac Live® 2.x Calibration Tool™, you have more freedom in loudspeaker placement, although the best result will still be achieved with the optimal loudspeaker placement in combination with Dirac Live®.

Sweet spot and dispersion

The loudspeaker response varies with position in a room. The sweet spot is the region where the sound is perceived as best and with very little variations. Outside of the sweet spot, the sound character deteriorates and is quite different from that within the sweet spot. Room correction systems cannot extend the size of the sweet spot! The dispersion of the speaker cannot be changed, unless several speakers are used together as a ”super speaker”. However, Dirac Live® 2.x Calibration Tool™ can still improve the sound quality outside the sweet spot.

Reflections – Good and bad

The sound that is radiated from the loudspeakers will be reflected by the walls of the room and eventually reach the listener at different times and at different angles, depending on how many times the sound wave has been reflected off the walls.

Early reflections from angles close to the loudspeaker and diffraction reflections from the loudspeaker box itself are detrimental to perceived sound quality and lead to undesirable coloration of the timbre of the sound.

Side reflections and late reflections are generally not as detrimental and can even help in improving perceived sound quality. The reason for this difference lies in our hearing:

- In a reasonably sized room, we can tell a side reflection from a direct wave because of the time and direction differences which our brain has learned naturally to separate. These reflections often contribute to a larger, more enveloping sound stage. A small amount of reverberation is generally considered desirable in a room for sound reproduction. Dirac Live® 2.x Calibration Tool™ leaves the late reflections almost untouched. It only corrects for their spectral coloration.

- Early reflections and loudspeaker diffraction, which are very close in both time and angle to the direct wave, make the sound more difficult for the brain to process properly. However, Dirac Live® 2.x Calibration Tool™ corrects for these shortcomings by optimizing the early arriving sound at the listener position.

At bass frequencies, room reflections may build up and create what is known as standing waves or room resonances. Characteristic of these is that at certain frequencies, the bass is strong in some parts of the room and weak in other parts. Dirac Live® 2.x Calibration Tool™ mitigates the effect of standing waves inside the listening area but will not correct for poor subwoofer alignment and strong standing waves. We always recommend to first get your subwoofers well positioned and aligned in time domain so that the response in the listening area is good while your subwoofers play together. Indeed, Dirac Live® 2.x Calibration Tool™ is treating each subwoofer individually and not globally, limiting the correction of their interaction in the room. Therefore, aligning them first and then group them in the ISP Speakers Group configuration prior to going through Dirac Live® calibration process is highly recommended. In such case, Dirac Live® 2.x Calibration Tool™ would see all the subwoofers as one and will not alter the inter-subwoofers alignment, just making sure it well integrates with all the other speakers in the room.

Troubleshooting

If you run into any trouble, then please first refer to the common errors below.

If the problem remains, we would appreciate it if you would send us a description of the problem, perhaps including a supporting screenshot, and also provide information of the Dirac software version, computer type and OS version, and ISP/I.ISP software version.

You can contact StormAudio in several ways:

Send an email to support@stormaudio.com,

Or visit us at **http://www.stormaudio.com/support/** for more help.

Troubleshooting the Dirac Live® 2.x Calibration Tool™

Sound system setup problems

Problem	Possible Cause	Fix
ISP / I.ISP Processor is not detected	ISP / I.ISP is not in Dirac mode	In the Speakers Edition mode, launch DIRAC and Start New Calibration
	ISP / I.ISP and the computer running Dirac are not in same network.	Ensure that you run the complete system (ISP/I.ISP and computer) in the same network and that ISP/I.ISP is in Dirac mode.
	UPNP not supported in the network	Use the tool with the Manual IP address insertion to connect to the ISP / I.ISP.
Dirac shows a different processor than the StormAudio ISP-HOST	ISP / I.ISP has out dated software.	Insure the ISP / I.ISP is updated with the latest firmware. Dirac Live 2.0 requires version 3.4r2 or above to function correctly.

Microphone config.

Problem	Possible Cause	Fix
No options or missing options in the 'Recording device' In icon list.	Sound card driver is not compatible with DLCT.	Update the sound card driver to the latest version. Or use the driver which is provided by OS.
	The USB microphone didn't connect well with computer.	Check if the microphone connects to USB port properly. Or reconnect the USB mic. Then press 'Rescan' in the 'Sound system' step.

Playback problems

Problem	Possible Cause	Fix
No sound from level test or measurement playback ⚠ **Caution: To protect ears and equipment, never switch on or connect devices at a high volume setting!**	Amplifier is OFF	Switch on amplifier
	Volume too low	Increase volume on ISP / I.ISP. If this does not help, then reduce the volume to a reasonable level before attempting any other fixes!
	Loudspeakers are distorting	Decrease the playback volume on ISP / I.ISP

Recording problems

Problem	Possible Cause	Fix
Level test meter does not react **Switch off sound playback while attempting to fix this problem. Instead, clap your hands or make some other sound to detect a reaction on the level meter.**	Microphone disconnected	Connect microphone to microphone preamplifier and connect microphone preamplifier to sound card or to computer.
	Microphone needs phantom power	Enable the 48 V phantom power on the microphone preamplifier input. Depending on the microphone preamplifier, this could be a physical button or a software setting.
	Microphone preamplifier is OFF	Switch on microphone preamplifier
Measurement does not complete successfully	Error message	Follow instructions given in the error message.

Filter design problems

Problem	Possible Cause	Fix
Filter optimization does not complete	Client is offline	Connect the client computer to the web
	Firewall is blocking access	Disable firewall or set it to allow HTTP traffic

Resources

Recommended reading

For the interested reader, a good book for learning more about how loudspeakers and rooms interact and how to achieve good sound is "Sound Reproduction, Loudspeakers and Rooms" by Floyd E. Toole (2008).

Credits

Thank You and Credits to bring this documentation to our valued dealers at StormAudio.

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