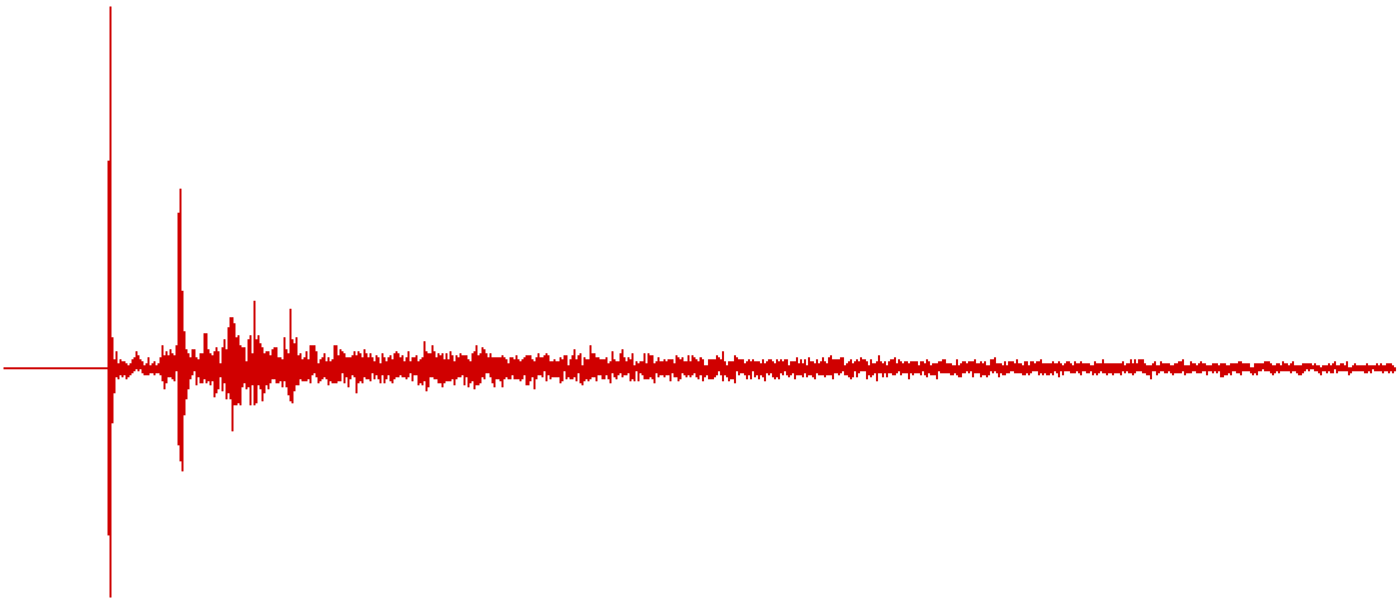
SREV1 Sampling Guide

An Introduction to Impulse-response Sampling with the SREV1 Sampling Reverberator



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1 Introduction

This document provides a basic explanation of sound-field sampling, and a tutorial by which you can actually sample a sound field and audition it on the SREV1.

For more detailed information on the SREV1, IRSampler, or IREdit, please refer to the relevant documentation.

What is Sound Field Sampling?

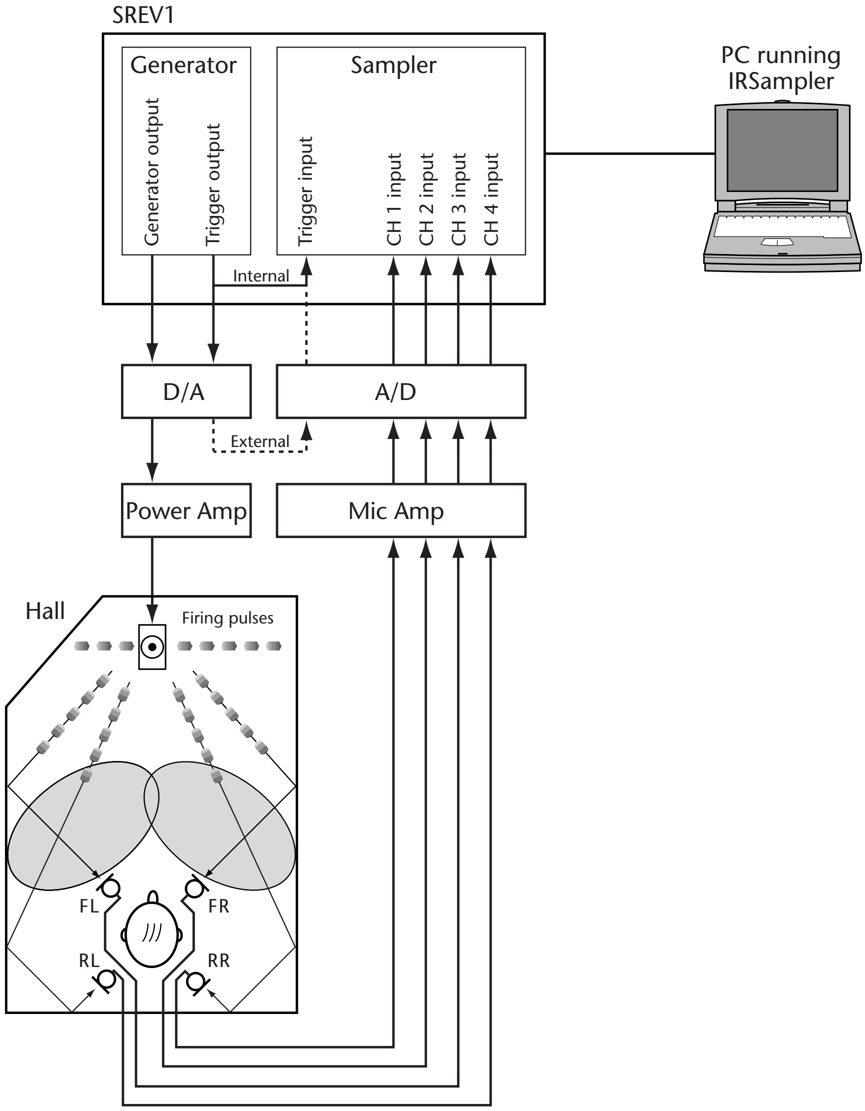
No doubt you are already familiar with the technique of audio sampling, made possible by the ubiquitous sampler? Well, sound-field sampling is similar, except that instead of capturing sounds, we're attempting to capture the unique character of an acoustic space, such as a concert hall or church. When you sample a sound field with the SREV1, you are in fact sampling the reverberant characteristics of that acoustic space. The acquired data can then be loaded into the SREV1 to create a reverb program that reproduces the unique reverberation of the original space.

Let's take a moment to consider the sounds we hear in an acoustic space such as a hall. Vocal or instrument sounds (i.e., the audio source) reach the ears of the listener accompanied by the reverberation of that acoustic space. If we substitute a microphone for the listener's ears and record the sound, we capture the vocal or instrument sound together with the reverberation of that acoustic space. Since the reverberation is unique to that particular space, until now the only way to add it to a vocal or instrument sound was to actually perform in that space and record the resulting sound. With the introduction of the SREV1, however, it's now possible to sample the reverberation characteristics of an acoustic space and apply them to any audio signal.

In conventional recording, changing the location of the singer or instrument (i.e., the audio source), or the position of the microphone (i.e., the pickup point) affects the sound that is recorded. You've no doubt experienced how the same audio source can sound different when heard from a first-floor seat and a balcony seat. In addition, different power amplifiers, speakers, microphones, and other equipment can also have an affect on the recorded sound. These issues also apply to sound-field sampling with the SREV1. Changing the position of the audio source or the pickup point, for example, or changing the equipment, all affect the reverberation that is sampled, even in the same acoustic space. In this respect, sampling a sound field with the SREV1 is similar to normal recording.

Sound-field sampling involves recording the reverberation information that occurs between an audio source located in the space being sampled and a pickup point. Since this consists of the information that occurs between these two points, it's not possible to capture the reverberation character of an entire acoustic space. By experimenting and taking a number of samples at various pickup points and sound source positions, however, you should be able to capture the reverberation character that defines each and every acoustic space.

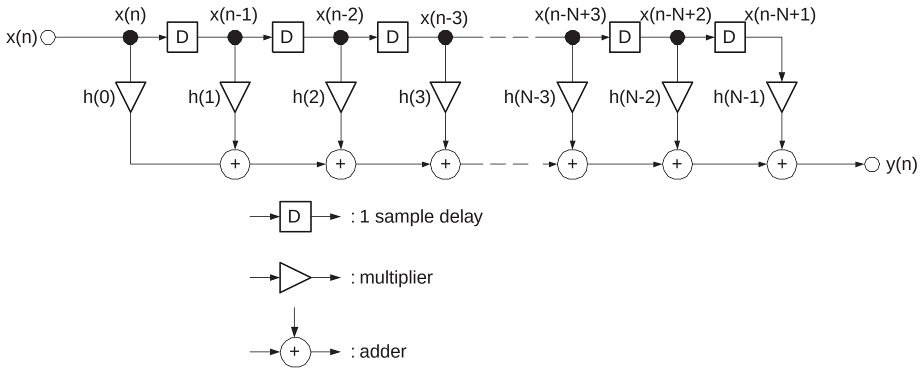
As you can see from the following diagram, sound-field sampling consists of "firing" SREV1 test pulses into an acoustic space, thereby energizing the reverberation in that space, which is then picked up by a number of microphones and returned back to the SREV1 for processing. The acquired data can be saved onto PC Card, edited as necessary using IREdit, and then loaded into the SREV1 to create reverb programs.



What is Convolution?

Convolution is a technique for imposing the characteristics of one signal onto another. What the SREV1 does is to convolve the reverberation characteristics of a previously sampled acoustic space (i.e., its impulse-response) onto another audio source, producing the same overall sound that would have been heard had the audio source actually been heard in that acoustic space.

For the technically minded, the following diagram shows the signal flow and formula for convolution processing.



$$y(n) = \sum_{i=0}^{N-1} h(i) * x(n-i)$$

What is an Impulse Response?

Just as a frequency response shows how an audio circuit responds to a range of frequencies, an "impulse-response" shows how an acoustic space responds to an impulse signal. And since the data acquired by measuring the impulse response of an acoustic space consists not only of the original test signal, but acoustic information about that space as well, we can use an impulse-response to measure reverberation.

No doubt you've clapped your hands in order to check the reverberation characteristics of a hall. Well, since a hand clap is audibly similar to the sound of an impulse, this is akin to hearing an impulse response. What you heard were the reverberation characteristics of that acoustic space, your hands being the audio source, your ears the pickup point. A handclap is nowhere near as accurate as an impulse signal, so it's not strictly the same as what we're talking about here, but it's close enough to give you some idea of what we mean by an impulse response.

Generally speaking, "impulse response" is a response to a linear, time-invariant pulse. "Linear" meaning undistorted, or in other words, the test signal does not affect the reverberation characteristics. By "time-invariant" we mean that the reverberation characteristics do not change over time. Conventionally, "time-invariant" is used to mean a state that does not change, but here we are assuming that the reverberation characteristics do not change from the time the test signal is output to the time sampling is completed.

Impulse signals are ideal for sound-field sampling because they have a very short playback time and a flat frequency response at all frequencies, subject to the limitations of the recording and playback process.

What are Impulse & TSP Signals?

The SREV1 can generate two types of test signal. First we'll take a look at the impulse signal. As explained earlier, an impulse has a very short playback duration and a flat response at all frequencies. You can acquire impulse-response data by using an impulse signal, but you can achieve a better S/N performance by using another type of signal, which we'll talk about later.

If you listen to the impulse signal, you'll hear that it has a relatively low sound pressure (i.e., volume), even when the level meters are virtually at max. And if you turn up your power amps in an attempt to increase the sound pressure, you run the risk of damaging your speakers. In order to acquire impulse-response data with a good S/N ratio, you need the sound pressure (i.e., volume) of the source to be as loud as possible. Obviously, if your microphones or speakers are distorting, you've lost everything.

Now we can introduce the SREV1's secret weapon—the TSP or "Time Stretched Pulse." A TSP has the same flat response as the impulse signal, but differs in that it contains a sweep of all frequencies, providing a relatively high sound pressure (i.e., volume), as you can tell by listening. (Be careful with your volume level settings when using TSPs.) This higher level allows us to capture data with a better S/N ratio than that possible using an impulse signal.

Of course, simply outputting a TSP and miking the result does not produce the necessary data. The SREV1 has to perform various DSP processes to the sampled sound in order to create the impulse-response data necessary for convolution. But that's another story.

What is Averaging?

When sampling in an environment with noise present, such as that of an air conditioning system (i.e., unwanted white noise-type noise), the S/N ratio of the acquired data will be poor. By using a process called "synchronized summing," the SREV1 is able to reduce this noise level by averaging multiple samples, thereby improving the S/N ratio.

As each successive sample is taken, it's added to the previous sample, thereby increasing the level of the useful data. Since any background noise is random (i.e., different for each sample), it increases by only half as much when added.

The number of samples to be taken is set by using the Averaging parameter. When set to "8," for example, the pulse signal is output and sampled eight times. Care must be exercised while multi-sampling is in progress, as any external sounds will have a detrimental affect on the acquired data. If synchronized summing is performed too many times, the accuracy of the high-frequency detail may be affected.

What Equipment is Necessary?

In order to perform sound-field sampling, you'll need microphones and microphone preamps to pick up the sounds, speakers and power amps to output the test pulses, and the necessary connecting cables. Of course, you'll need the SREV1, and depending on the type of I/O being used, you may also require A/D and D/A converters to get signals in and out of the SREV1. You'll also need the CD-ROM containing the SREV1 sampling software, PC Card memory for storing the impulse-response data, a PC running Windows (95, 98, 98SE, 98ME, NT 4.0, or 2000), and a serial cable.

Is the Quality of the Equipment Important?

Since sound-field sampling involves recording the reverberation information that occurs between an audio source located in the space being sampled and a pickup point, the sampling equipment (i.e., the microphones, preamps, speakers, power amplifiers, etc.) does have an effect on the acquired data.

Basically, you need good-quality microphones with a wide frequency response. Those that you use for hall or studio recording should be adequate. The PA system should have a wide, flat range and enough power to energize the reverberation you want to capture. The better the equipment, the more faithful the end results will be.

How about Microphone & Speaker Placement?

Typically you want to set up your speakers where the sound source would otherwise be, and it's probably worth experimenting in order to obtain the desired results. If you are sampling a hall, for example, the center of the stage is a good starting place. Also consider the directionality of the speaker so that it corresponds to the directionality of the instrument whose reverberation effect you are trying to capture. You could use several speakers to mimic an instrument that distributes sound in various directions.

If you are sampling two channels, start by trying the microphone setup that you usually employ for stereo recording. If you are sampling four channels, use the same miking techniques as for 5.1 channel recording.

With your speakers and microphones setup roughly where you think they should go, play some dry instrument sounds or vocals through the system and adjust the microphone position, direction, height, and the left and right spread. When you hear the sound you want, you can proceed to the actual sampling.

2 Sampling Tutorial

Assuming that you've set up your speakers and microphones and connected your PC to the SREV1 (see page 17 for hookup examples), you're now ready to "fire" a few pulses and "grab" some samples.

We're assuming that SREV1 I/O is being taken care of by MY4-AD and MY4-DA cards. If you're using external A/D and D/A converters, your IRSampler I/O assignments will need to be a little different to those specified later.

Before we start, turn on the SREV1, insert the CD-ROM containing the SREV1 sampling software, and insert a PC Card memory card. (The RC-SREV1 Remote Controller is not necessary for sampling.) Finally, start the IRSampler program.

Configuring IRSampler

COM Port (COM Port dialog box)

The COM Port dialog box appears the first time IRSampler is started and is used to select the PC COM port to which the SREV1 is connected. The selected COM port is used automatically the next time IRSampler is started, but can be changed at any time by choosing COM Port from the Setup menu. Choose COM1 or COM2 as appropriate.

Word Clock Source (Word Clock toolbar)

Set the Word Clock source to "INTERNAL" (FS will be "48 kHz")

Channel Mode (Word Clock toolbar)

Set the Channel mode as appropriate (e.g., 1ch, 2ch, 4ch)

Pulse Generator (Generator/Trigger window)

Set the Generator Source to "TSP64k"

Set the Loop Interval to "5000" msec (i.e., 240000 samples)

The Loop Interval must be long enough so that the reverberation energized by each pulse has time to fade away completely before the next pulse is fired. If the next pulse is fired while the previous reverberation can still be heard, the acquired data will be unusable, especially when using TSP signals. Obviously, environments that produce longer reverberation require longer loop intervals.

Start Trigger (Generator/Trigger window)

Set the Trigger Source to "Internal"

Set the Trigger Threshold to "50%"

Set the Trigger Slope to "ABS"

Sampler (Generator/Trigger window)

Set Averaging to "8"

Set the Sampling Time to "4500" msec (i.e., 216000 samples)

Generator Output Assign (Generator/Trigger window)

Assign the Generator Output to "Slot2-1"

Trigger Output Assign (Generator/Trigger window)

Assign the Trigger Output to "Slot2-2"

Input Assign (Generator/Trigger window)

Assign the Sampler Inputs as follows:

Channel 1: Slot 1-1

Channel 2: Slot 1-2

Channel 3: Slot 1-3

Channel 4: Slot 1-4

Input Levels (Main window)

Set the Sampler Input level faders as follows:

Channel 1: 0.0 dB

Channel 2: 0.0 dB

Channel 3: 0.0 dB

Channel 4: 0.0 dB

At this point, try a microphone test and check the input level meters to verify that the microphones are working properly. If no signals are present on the meters, check the connections and input assignments.

Generator Output Level (Main window)

Set the Generator Output level fader to "0.0 dB"

Checking the Generator Output

We're now ready to check the generator output. Click the Start Generator button on the Sampling Control toolbar and verify that the test signal is being output properly from the speaker system. If it's not, check the connections and generator output assignment.

Starting with the level set low, gradually increase the power amp's input attenuators so that the pulses can be clearly heard. (Setting the Generator mode to loop is convenient for outputting pulses continuously while making this adjustment.) The pulse signal should be loud enough to fully energize the reverberant characteristics of the environment you are sampling, but without distortion.

Setting Input Levels Automatically

The Level Adjust function offers a convenient way to set input levels automatically. Click the Adjust button on the main window, and when the Level Adjust window appears, click the Start button. The SREV1 outputs several pulses and IRSampler calculates the optimum input levels. A progress report appears in the Status box.

In addition to setting the input level faders, the Level Adjust function automatically calculates and sets the CNV (Convolution) Bit Shift parameter, which is used to normalize the acquired data to an optimum level for use with the SREV1.

Actual Sampling

Sampling

To begin sampling, click the Start Sampling button on the Sampling Control toolbar. Sampling is performed in accordance with the parameter settings made earlier. When sampling is complete, the Waveform window appears automatically, displaying waveforms of the acquired data.

Saving the Acquired Data

If you are happy with the acquired data, enter titles, and click the Save As button for each waveform. (The title is saved in the file header.) Enter a filename in the Save As dialog box and click Save. By default, data is saved in the "irdata" (c:/yamaha/irmec/irdata) folder on the PC Card in TM4 format. (A ".tm4" file extension is added automatically.)

Taking Several Samples

While your sampling system is setup, it's a good idea to take several samples at various microphone and speaker placements or with different IRSampler settings. Once back in the studio you can easily delete the unwanted data. Going back to a location and setting up all your gear again, however, may not be so easy.

Using Your Data on the SREV1

1 Put your data in the right folder

The SREV1 can only load impulse-response data stored in the "c:\yamaha\srev\data" folder of the PC Card. So the first thing you need to do is to make a new folder called "data" in "x:\yamaha\srev\" and copy your acquired data into it, something you can do on your PC. (Note that "x" refers to the letter assigned to your PC Card while it's inserted in your PC.)

2 Insert the PC card into the SREV1

Insert the PC Card into the SREV1's MEMORY CARD slot.

3 Load your data

On the RC-SREV1, go to the Rev page, select the DATA LOAD button, and press [ENTER] to access the Data Load page. Next, select the PCMCIA button and press the [ENTER] button. Your impulse-response data should appear in the list of files.

Use the DATA wheel or the [-1/DEC] and [+1/INC] buttons to select your file in the list. And then select the button of the channel to which you want to load the data. Click the [ENTER] button to load the data. Repeat this procedure to load data for the other channels.

4 Audition your reverb data

Now you can listen to the reverb generated from your own samples.

5 Adjust the EQ

On the Data Load page, select the BACK button and press the [ENTER] button to return to the Rev page. Go to the EQ pages and try adjusting the pre- and post-EQ parameters.

6 Save your program

If you're happy with your new reverb program, go to the Program page and save it to PC Card now!

3 Data Editing

If you use your samples as they are, you may experience the following:

- A roaring sound at the end of the reverberation.
- An unwanted delay at the beginning of the reverberation.
- Direct sound in the sample is making it unusable.
- The reverberation is too loud.

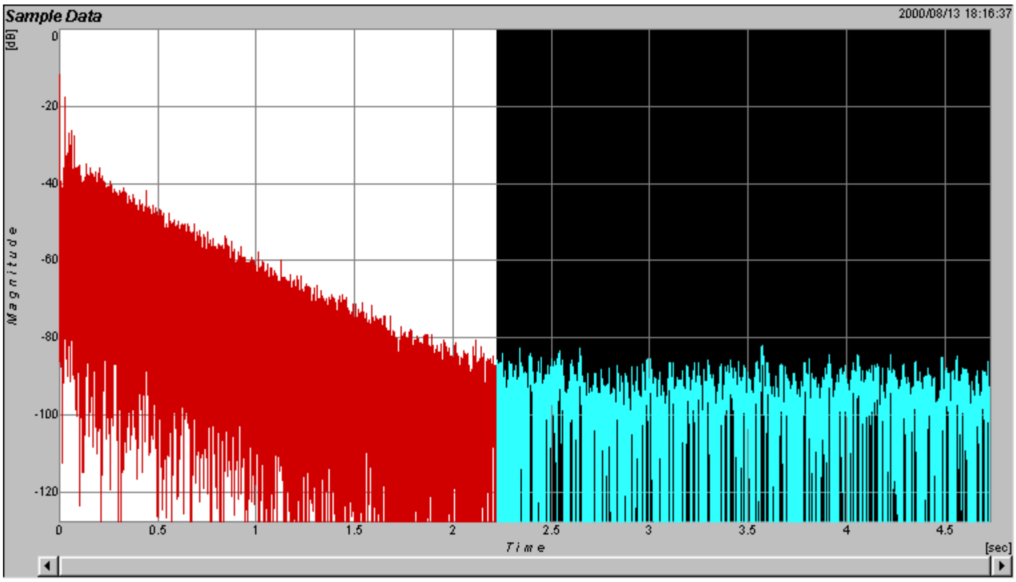
These issues are addressed in the following sections, along with an explanation of how to specify the reverb time and start point that the SREV1 uses to process your impulse-response data.

Fixing the Roar

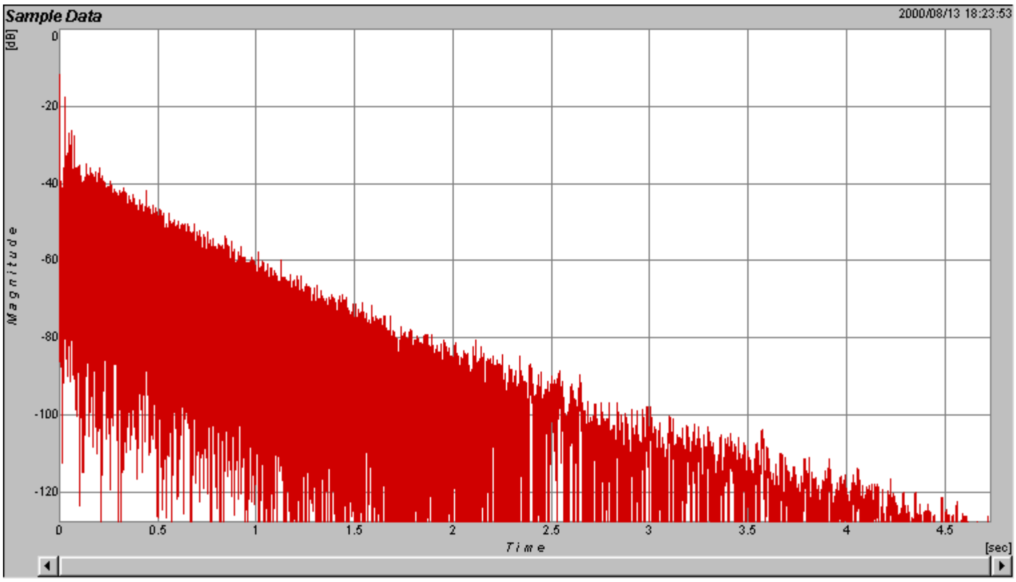
As the reverberation decays, the extraneous noise captured in the impulse-response data becomes relatively louder and can create a roaring sound. This can be seen clearly in IREdit by setting the vertical axis to dB.

This issue can be resolved by fading out the extraneous noise at the end of the sample, so that the reverberation decays naturally. You can do this by using the Fade Out function of IREdit.

The following image shows a region of extraneous noise selected prior to fading.



The following image shows the data after the fade out.

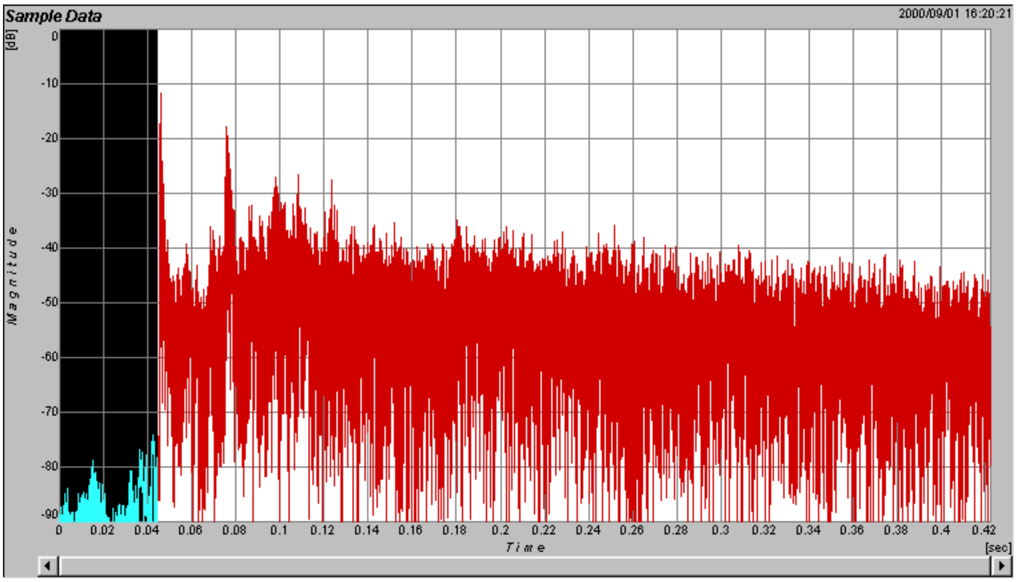


If you fade out the extraneous noise so that its fade rate matches that of the reverb data (i.e., you achieve a linear-looking fade), the reverberation may decay too quickly and sound unnatural. In some cases, a better result can be achieved by fading out the noise less steeply.

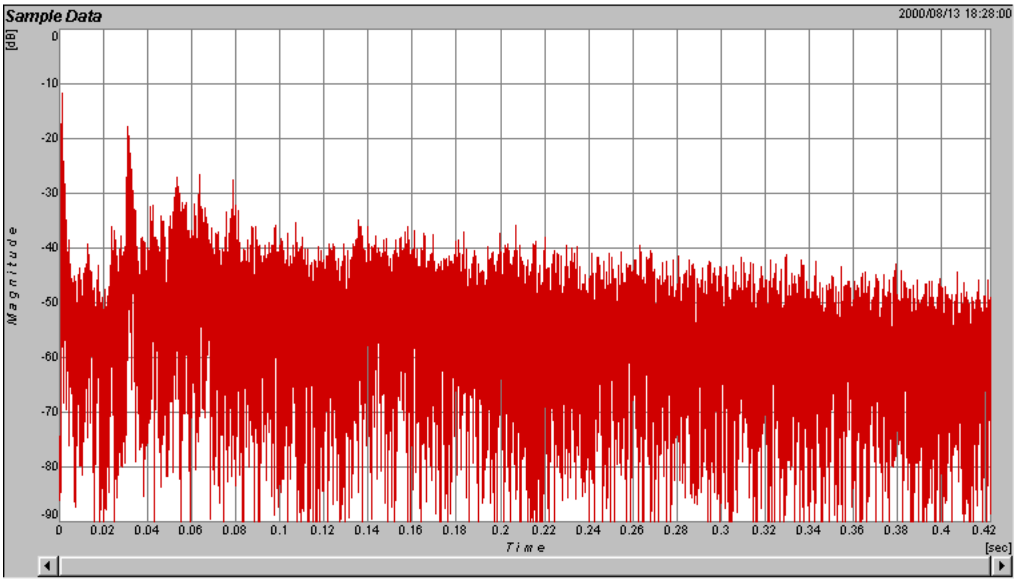
Removing the Delay at the Beginning of the Reverberation

Your impulse-response data may contain a spatial delay at the beginning due to the time it takes the pulse to reach the microphone from the speaker. In some cases this may be a hindrance and can be deleted by using the IREdit Cut function.

The following image shows a spatial delay region selected for cutting.



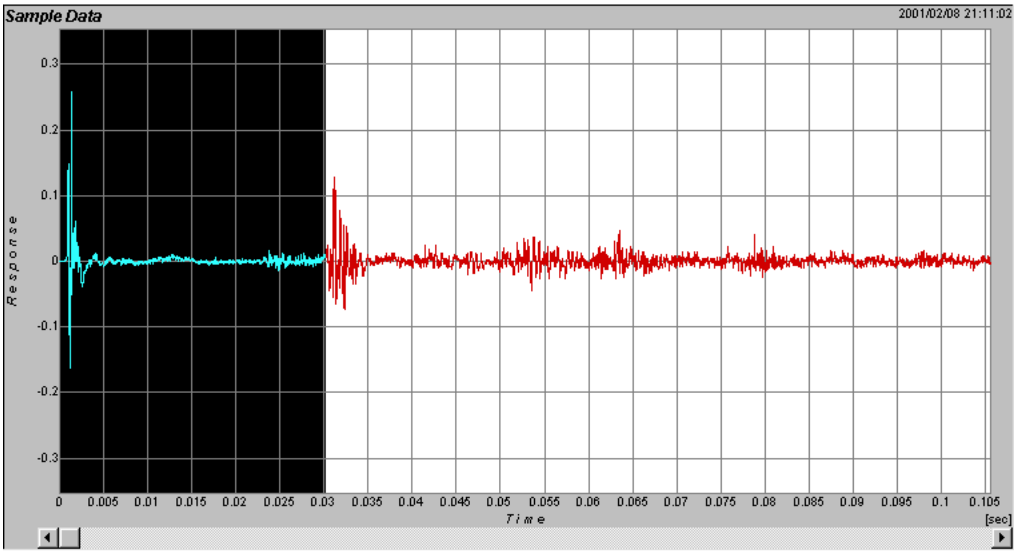
The following image shows the data after the spatial delay out has been cut.



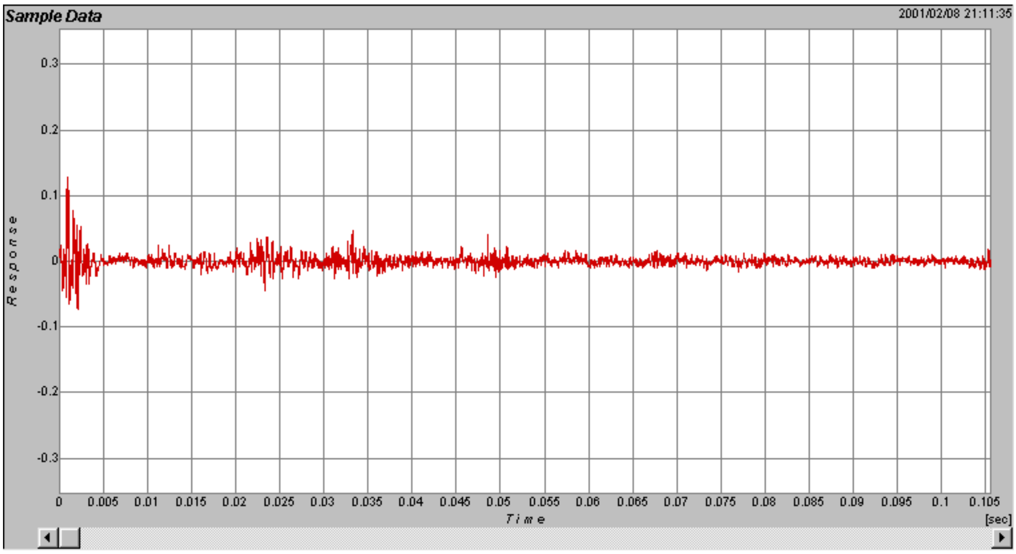
Deleting the Direct Sound

In addition to reverberant sound, the acquired data typically contains direct sound that reaches the microphones directly from the speakers. In order to use the Reverb Balance parameter, or to create a mix of dry and wet signals on your mixer, it's best to remove the direct sound component, which you can do by using the Cut function of IREdit.

The following image shows a direct sound region selected for cutting.



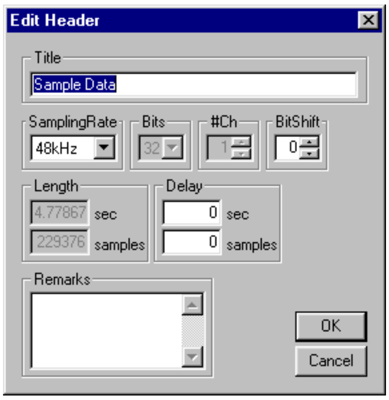
The following image shows the data after the direct sound has been cut.



Adjusting the Reverberation Level

If you use your samples as they are, you may find that the reverberation produced is too loud relative to the input signal. This can be resolved by editing the header information with IREdit.

In IREdit, choose Header from the Tools menu. The Edit Header window appears, displaying various information about the data, as shown below.



One of the values you can adjust is “BitShift,” which you can use to reduce or increase the level of the sample data. If the reverberation is too loud, try various negative bit shift values and chose the one that provides the best results.

Specifying the Reverb Time & Start Point

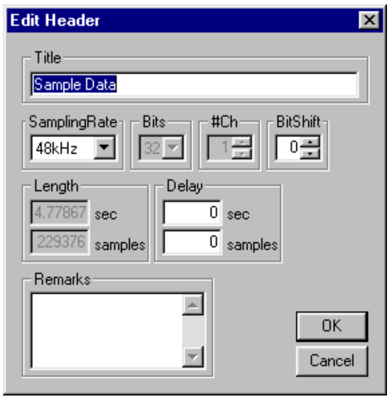
Impulse-response data files (i.e., TM4 format files) do not specify a reverb time value, so the SREV1 automatically calculates values for the reverb time parameter that is used by the RC-SREV1. This is based on the length of the sampled data, not the length of the file. You can, however, specify a reverb time (RT) by editing the file header information with IREdit. The SREV1 will then base it calculations on the value you specify.

Say, for example, that you specify a reverb time of 2 seconds (e.g., RT=2.0) for a file that the SREV1 would have automatically calculated a reverb time of 2.1 seconds. Based on your value, the SREV1 recalculates the impulse-response data and passes on the results to the convolution processors. Basically, the SREV1 calculations affect the way in which the reverberation fades out.

You can also specify the point at which reverberation fade-out processing begins. This is called the reverb start (RS) point. Technically speaking, this is point at which the SREV1 begins exponentially attenuating the reverb data.

Essentially, reverberation consists of two components: the early reflections and the main reverb which follows. Early reflections define the character of an environment, and if the SREV1 recalculates the early reflection information in addition to the main reverb, then that character may be lost. To preserve the character, you can instruct the SREV1 to begin its calculations after the early reflection information.

Take, for example, an impulse-response data file where samples 0 to 1,000 represent the early reflections, and samples 1,001 to the end of the file represent the main reverb. By specifying a reverb start point of 1,001 (e.g., RS=1001), the SREV1 will begin recalculations after the early reflection information, thereby preserving the character of the original reverberation.



To specify the RT and RS values in IREdit, choose Header from the Tools menu. The Edit Header window appears, as shown here.

Enter a reverb time in the Remarks box in the following format:

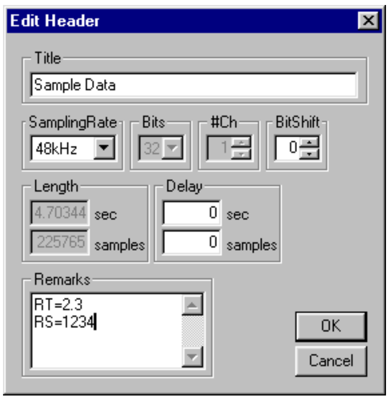
RT=xx (e.g., RT=2.0)

Values can be entered in 0.1 second steps.

On a new line, enter a reverb start point in the following format:

RS=xx (e.g., RS=1001)

Value can be entered in single sample steps.

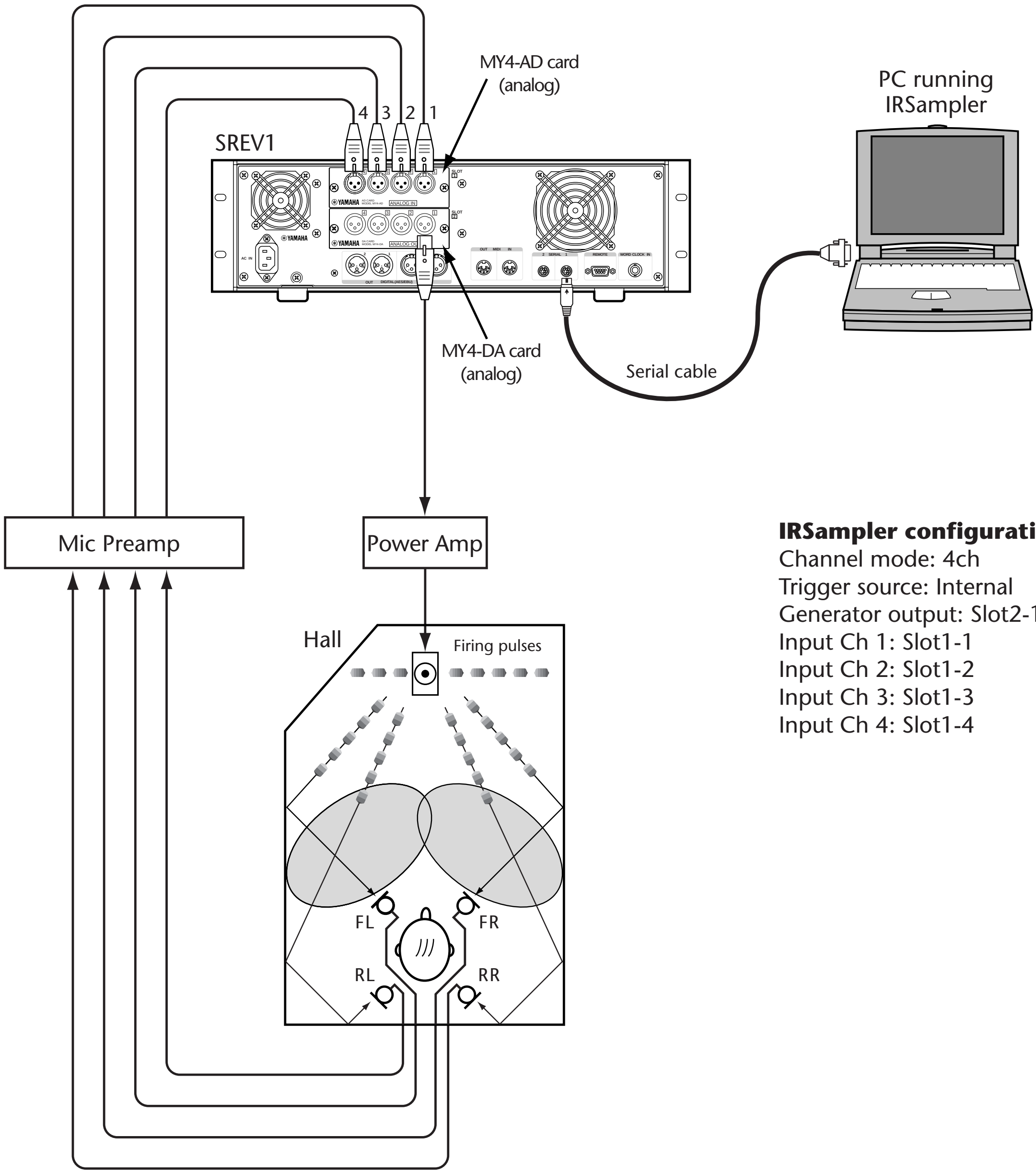


This Remarks box contains both RT and RS values.

4 Hookup Examples

MY4-AD/DA I/O System

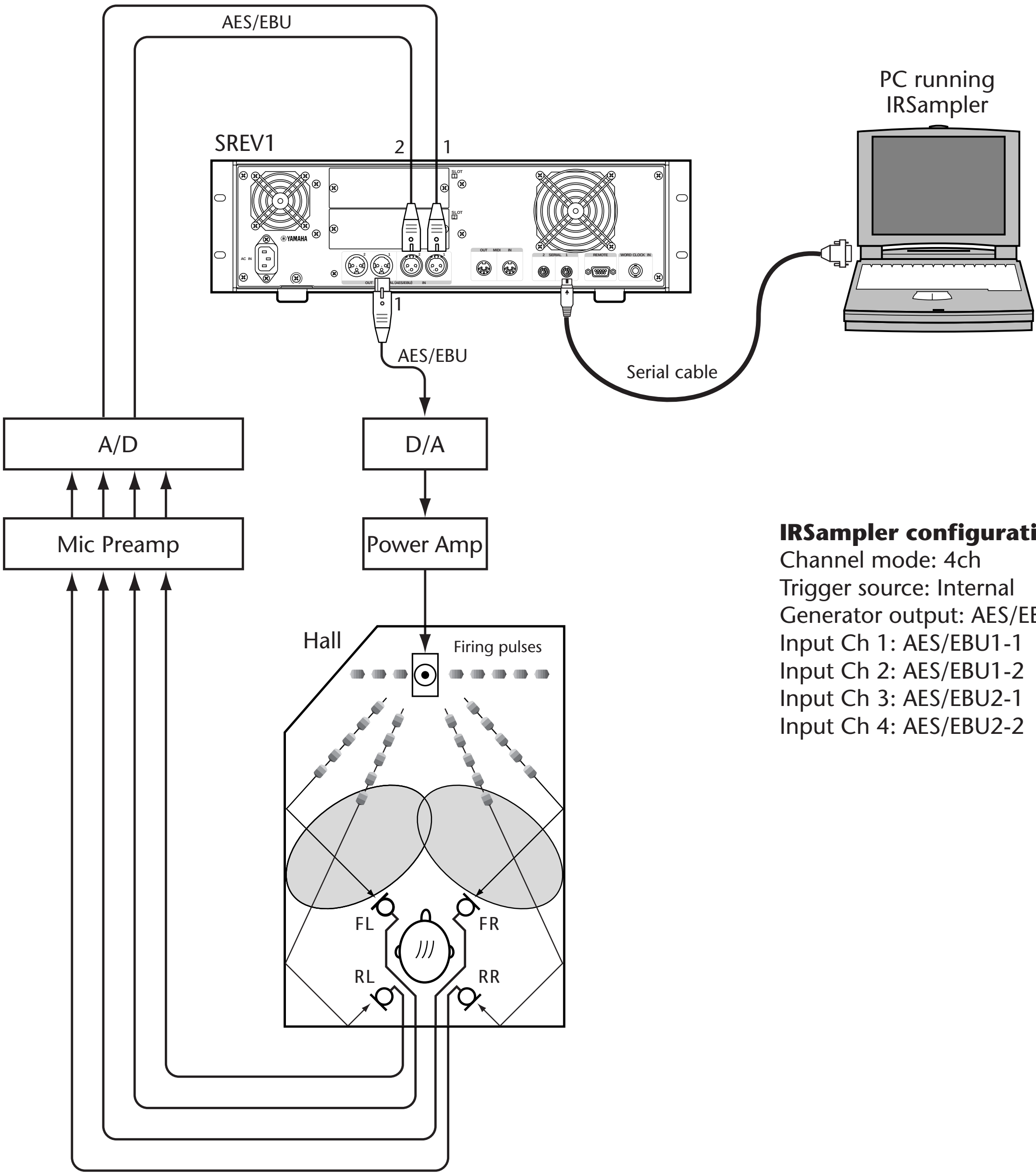
The following example shows a typical SREV1 setup for sampling a hall. Analog I/O on the SREV1 is provided by MY4-AD and MY4-DA cards. The trigger source is set to internal. The generator output is assigned to SREV1 Slot2-1. From there it's fed via a power amplifier to the speaker system. Carefully placed microphones pick up the reverberating sound, which is fed back to the analog SREV1 inputs via the mic preamp.



IRSampler configuration
Channel mode: 4ch
Trigger source: Internal
Generator output: Slot2-1
Input Ch 1: Slot1-1
Input Ch 2: Slot1-2
Input Ch 3: Slot1-3
Input Ch 4: Slot1-4

AES/EBU I/O System

The following example is essentially the same as the previous system, but with separate A/D and D/A converters connected to the SREV1's onboard AES/EBU I/O.



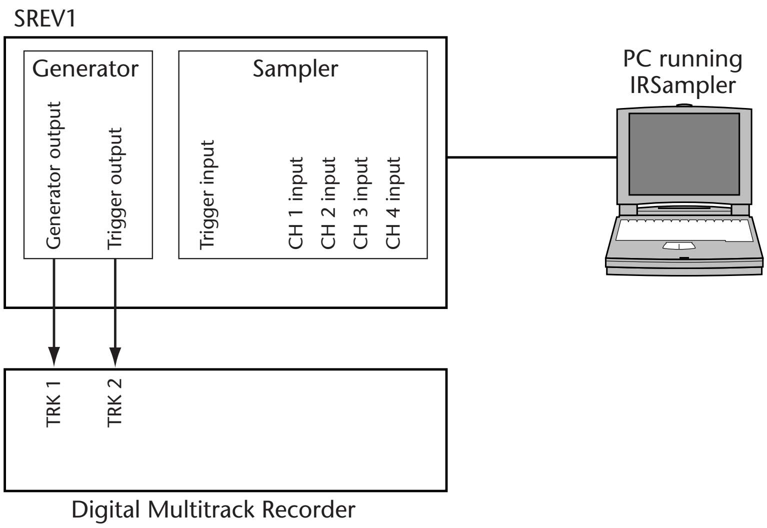
IRSampler configuration
Channel mode: 4ch
Trigger source: Internal
Generator output: AES/EBU1-1
Input Ch 1: AES/EBU1-1
Input Ch 2: AES/EBU1-2
Input Ch 3: AES/EBU2-1
Input Ch 4: AES/EBU2-2

Sampling with a Digital Recorder

Instead of sampling directly to an SREV1, a digital recorder can be used. This may be more convenient than using multiple SREV1s to sample four or more channels. Digital recorders are ideal for recording fluctuating waveforms, such as acoustic reverberation, since they provide the kind of simultaneousness across tracks necessary for accurate averaging.

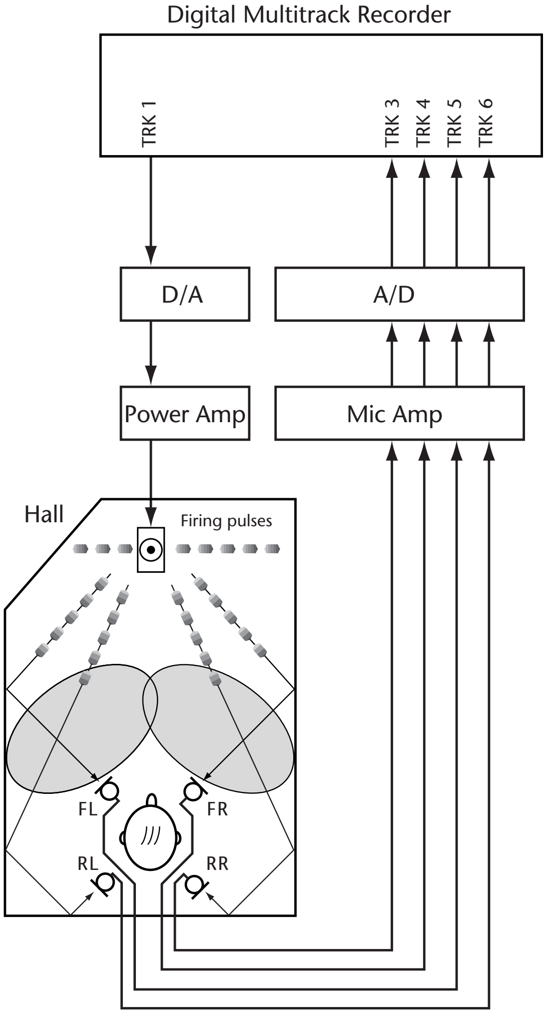
Step 1. Recording Pulses on the Digital Recorder

The first step is to record some pulses on track 1 and the corresponding trigger signals on track 2, as shown below.



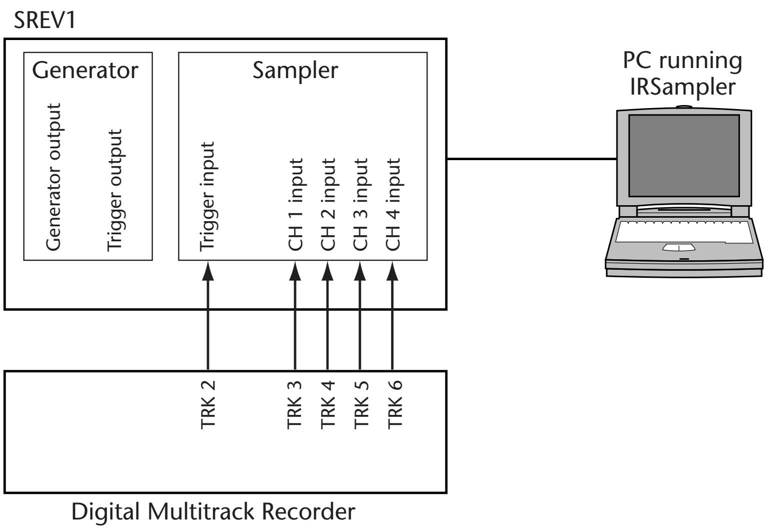
Step 2. Recording on Location

At the location the pulses previously recorded on track 1 are played back, while the responses picked up by the microphones are recorded onto tracks 3, 4, and so on.



Step 3. Sampling into the SREV1

The final step is to sample the recorded responses into the SREV1, using the trigger signal previously recorded on track 2 to trigger sampling, as shown below.



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