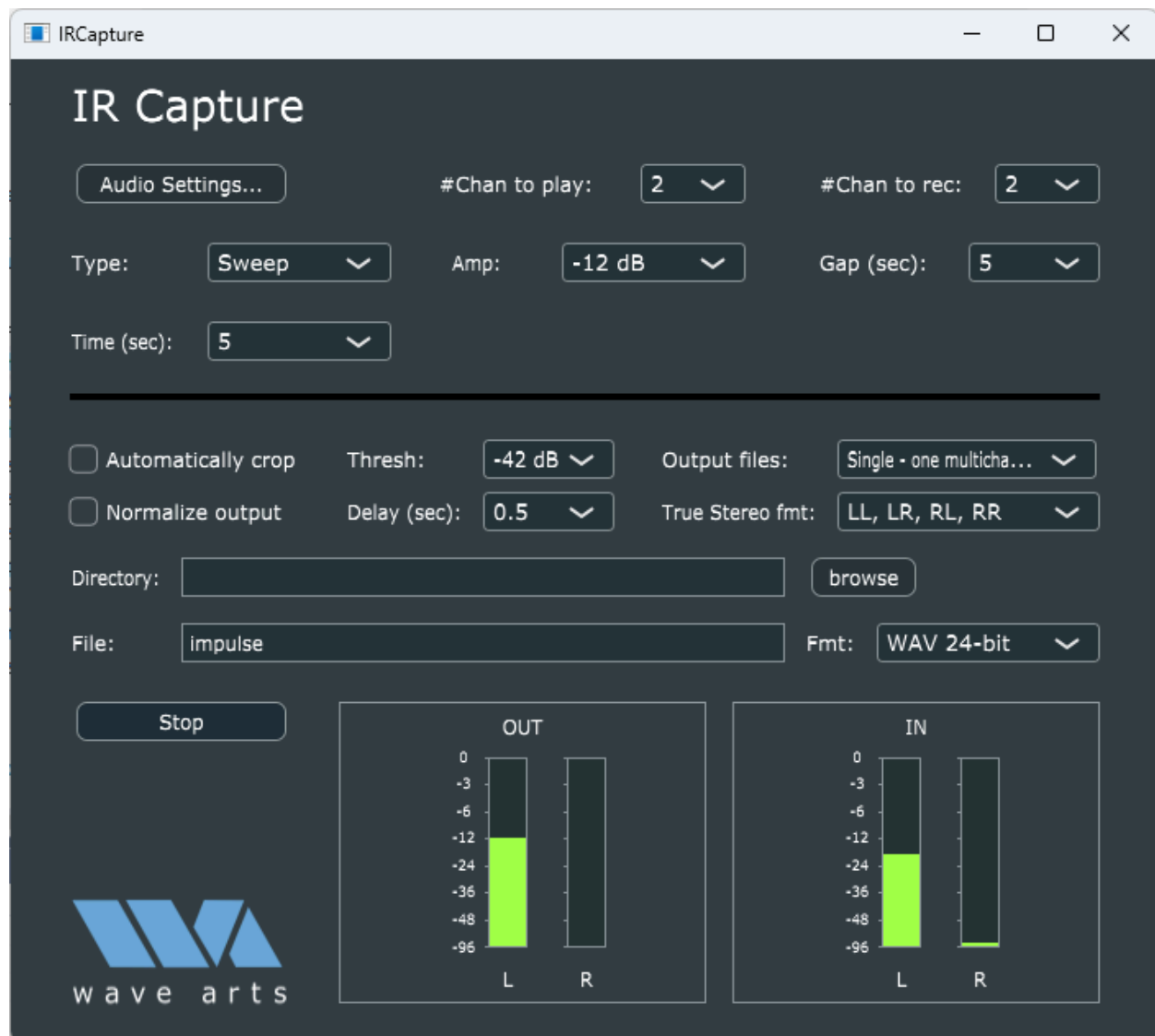


# IR Capture



IRCapture is an application used to measure impulse responses. Many electro-acoustical systems are well modeled by capturing an impulse response of the system. The impulse response can then be loaded into a convolution engine to reproduce the measured system. IRCapture can be used in conjunction with the Convology XT plug-in to reproduce acoustical spaces or digital effects units. You can also analyze the impulse response using mathematical applications such as Matlab or Octave.

## Impulse responses

An impulse response is a mathematical description of a linear, time-invariant system. What systems are linear? Microphones, speakers, amplifiers, and rooms are generally very linear, as are equalizers, delays, echo units, and reverbs. Non-linear systems produce distortion in the form of overtones. Tube amplifiers (when overdriven), distortion effects, limiters, and compressors are examples of non-linear systems.

What systems are time-invariant? Time-invariance means the system's response does not change over time. Systems that change with time would include any effect that modulates amplitude, frequency, or timbre. Effects such as tremolo, vibrato, leslie, wah wah, filter sweeps, flanging, phasing, chorusing, etc. all incorporate considerable time variation which is essential for the character of the effect, and hence are not well characterized by an impulse response. Digital reverbs may also incorporate some time variation to achieve a smoother decay sound, but in this case the variation is subtle enough that an impulse response effectively captures the character of the reverb. Similarly, acoustical reverberation may exhibit some slight time variation due to air currents.

## Sinusoid Sweeps

IRCapture by default will use sinusoid sweeps for a measurement signal. Sinusoid sweeps are good signals for measuring electro acoustical systems such as loudspeakers and microphones. The sweep is a sinusoid that starts at low frequency (20 Hz) and increases in frequency up to the Nyquist frequency (half the sampling rate). The frequency increases exponentially, spanning each octave of frequency range in the same amount of time.

Sweep measurements are good at ignoring harmonic distortion products, which are commonly produced by loudspeakers. Sweep measurements are also relatively immune to the use of different sample playback and recording clocks, and internal sample rate conversion.

When measuring a reverberant space, a good rule of thumb is to set the sweep time at least as long as the reverb decay time of the space, and follow this with an equally long gap delay to allow the response to decay fully before the next channel measurement. The impulse response length computed by IR Capture will equal the sweep time plus the gap delay.

## Maximum Length Sequences (MLS)

IRCapture can optionally use maximum length sequences (MLS) as a measurement signal. An MLS sounds like white noise. ML sequences have ideal correlation properties that make them attractive as a measurement signal. However, care must be taken to avoid non-linear and time-varying systems. When MLS is used to measure a non-linear system, the resulting impulse

response will contain parasitic echoes that are not in the true response. When MLS is used to measure a time-varying system, the resulting impulse response will contain extra noise.

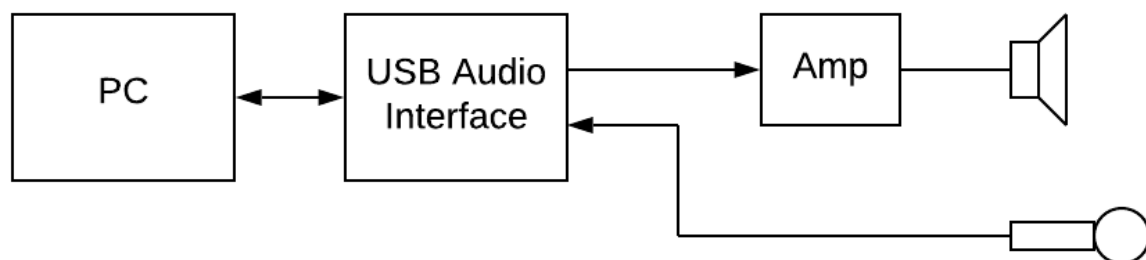
With MLS, it's very important to use the same audio device for both playback and recording, so that the sampling clocks for D/A and A/D conversion are perfectly synchronized. If different devices are used, the differing sampling clocks create time variation and the resulting impulse response will contain excessive noise. The same is true if sample rate conversion is applied to the playback or recording signal.

On Windows, using the "DirectSound" device type may apply internal sample rate conversion, hence it is preferred to use the "Windows Audio" device type. This will force you to use a sampling rate that is supported by the audio device hardware. You can also select "ASIO" for low latency, but check that your ASIO device is not set to do sample rate conversion.

## Measuring an acoustical system

It's recommended to use a good quality external USB audio device for both playback and recording. Make sure the USB device doesn't mix any microphone signal into the output, i.e. set any monitor mix control to minimum position.

To measure an acoustical system, such as a room, connect the audio output to a loudspeaker and connect a microphone. This will make a monophonic measurement of the acoustical space. Typically one would use two microphones to obtain a stereo measurement from a single speaker, and one can also use two speakers to create a "true stereo" response which contains four impulse responses: left->left, left->right, right->left, and right->right.

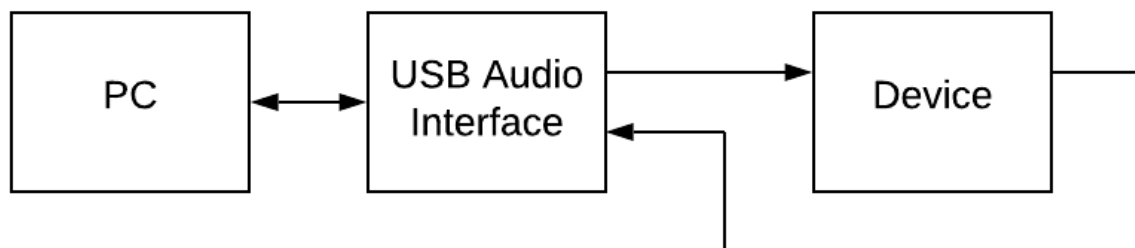


There are many techniques to record rooms, varying in the types, spacing and orientation of the microphones, and the positioning of the microphones with respect to the sound sources. A full description of how to record room reverberation is beyond the scope of this guide.

It's important not to drive the loudspeaker too loudly, this will force it into non-linear behavior. It's better to keep the levels gentle and use either a longer sweep signal or more MLS repetitions to get a good signal to noise level in the response.

## Measuring an electronic device

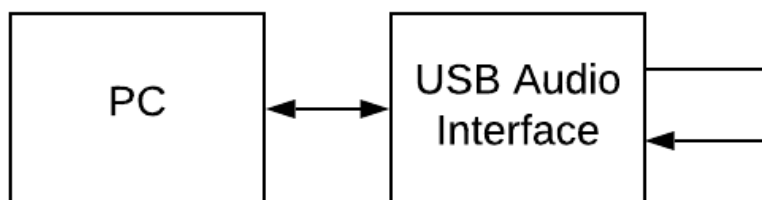
To measure an electronic device, connect the outputs of the audio interface to the inputs of the device, and connect the device outputs to the audio inputs.



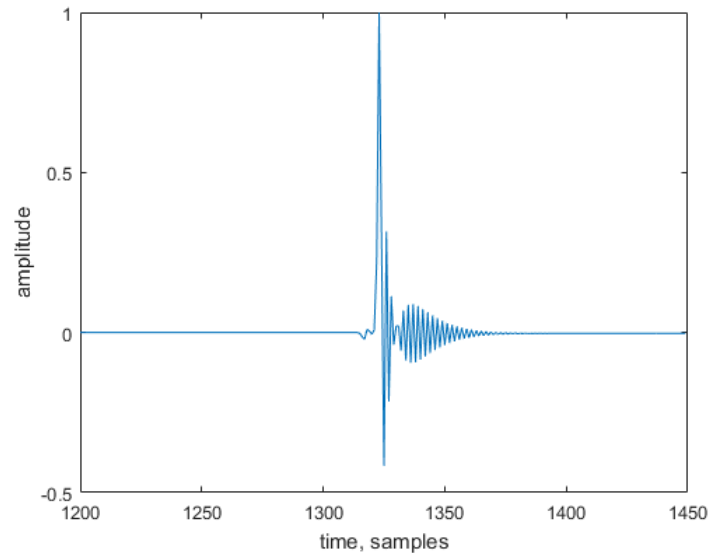
If you want a true stereo response, make stereo connections to and from the device.

## Loopback connection

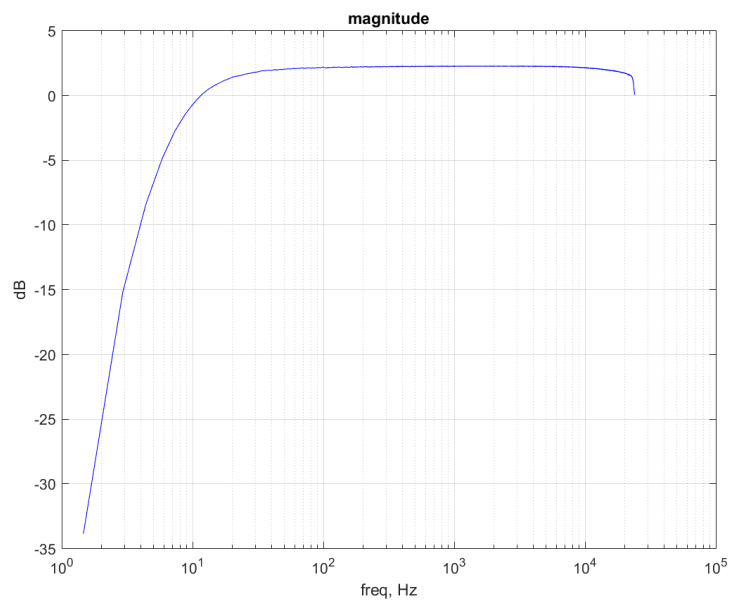
One can connect the audio output directly to the input. A measurement of this system will consist of the delay in the audio buffering and the response of the analog to digital conversion in the audio interface. These are always part of the measurement system. Measuring a mono or stereo loopback connection is an excellent way to test your setup and make sure that IRCapture is working properly.



An example response from a loopback measurement is shown below. This was from a Presonus AudioBox connected to a Mac running Catalina, measured using MLS. The squiggly spike is a typical time response of the D/A and A/D converters in the audio interface due to the anti-alias filters and the A/C-coupling (all MLS and sweep measurements are A/C-coupled, i.e. have -inf dB response at DC). The response is delayed about 28 msec, which is the total buffering delay in the system.



Below is the corresponding frequency response, showing the dropoff at very low frequencies and at the Nyquist frequency.



## IRCapture controls

**Audio Settings.** This opens a window where you can select the audio device type (OS driver), the playback and record devices, the sampling rate, and the buffer size. You must have enough

stereo pairs active to accommodate the selected number of playback and record channels, otherwise an error will appear when you start the measurement.

**#Chan to play.** This is the number of output channels to measure. IRCapture supports from 1 to 16 playback channels, depending on the capabilities of your audio interface.

**#Chan to record.** This is the number of input channels to measure. IRCapture supports from 1 to 16 playback channels, depending on the capabilities of your audio interface.

After selecting the number of channels you wish to measure, open the Audio Settings and select the audio device and if necessary, activate additional stereo pairs if you want to go beyond stereo.

**Type.** The measurement signal type, either Sweep or MLS.

**Amp.** The amplitude of the measurement signal.

**Gap.** This is the delay in seconds between successive channel measurements. When measuring a reverberant room or a reverb, this should be as long as the reverberation time to allow the room to decay fully between measurements.

**Time.** The length of the sinusoid sweep. A good rule of thumb is to use a sweep at least as long as the decay time of the system being measured. The resulting impulse response time will be equal to the sweep time plus the gap time.

**MLS len.** The length of the ML sequence in samples, which equals the length of the computed impulse response. At least two copies of the MLS are played and recorded (one can increase the number of repetitions beyond two to do averaging). The impulse response time is the length divided by the device sampling rate. When measuring a reverberant room or a reverb device, the MLS length should be as long as the reverberation time. For example, if the 60 dB RT of the room is 1.5 seconds and you are measuring at 48 kHz, this is 72000 samples, and you should use 131071 for the MLS length. When the MLS is too short, the response will time alias and hence the reverb tail will wrap around to the start of the response.

**#Reps.** The number of repetitions of the ML sequence. The recorded repetitions are averaged together before computing the impulse response; doing so is equivalent to averaging multiple measurements. This increases the signal to noise ratio of the impulse response measurement. Use this option when there is a lot of background noise. If increasing the number of repetitions does not decrease the noise in the impulse response, this is an indication that there is time variation in the system, perhaps sample rate conversion or different playback and recording sampling clocks.

**Automatic crop.** This option enables automatic cropping of the impulse response. After computing the impulse response, IRCapture looks at the start and end of the impulse response and crops leading and trailing samples that are below threshold. All channels of a multi-channel response are cropped the same.

**Thresh.** This is the threshold in dB for the auto-crop function. The threshold is measured with respect to the maximum sample across all channels.

**Normalize output.** Enable this to normalize the maximum sample across all channels to 0 dB. The same normalization gain is applied to all channels.

**Delay.** This sets the initial delay before the first channel measurement. Setting a delay here may be useful to ignore keyboard click noises, or give you time to leave the room.

**Output files.** This option determines how the impulse responses are stored into one or more output files. It has the following options:

- **Single.** All responses are stored in a single multi-channel file containing  $P * R$  channels where  $P$  is the number of playback channels and  $R$  is the number of recording channels. The first  $R$  channels correspond to the response between playback channel 1 and record channels 1 through  $R$ , the next  $R$  channels are for playback channel 2, and so on. For the special case of True Stereo (2-out, 2-in), the channel layout can be controlled via the True Stereo Format option.
- **Per output.** This creates one multi-channel file per measured playback channel.  $P$  files will be created, where each file has  $R$  channels, corresponding to the responses from that playback channel to each recording channel. Each file will have a suffix describing the playback channel, either L, R, or a numerical channel index from 0.
- **Per response.** This creates a single monophonic response file for every combination of playback and recording channel, hence  $P * R$  files are created. Each file will have a combination suffix describing the channels, e.g., \_LR means left playback channel to right recording channel. If more than 2 channels are active, numerical channel indices from 0 are used.

**True Stereo Format.** If measuring a stereo recording from a stereo playback, a 4-channel “true stereo” response will be created. This option specifies the order of channels in the file. Convology XT expects format (LL, LR, RL, RR).

**Directory.** This specifies the directory where impulse response files are written. If not specified than the current working directory is used, typically the same directory where the application is run.

**File.** The filename for the impulse response, without directory or filename extension.

**Fmt.** This option selects the output format. The user can select either 16-bit WAV format, 24-bit WAV format, 32-bit float WAV format, or txt (ASCII) format.

**Start.** This starts a measurement. You should hear the stimulus (either a sinusoid sweep or white noise) emitted and the level meters should show both output signal levels and recorded signal levels. When the measurement is in progress you can click Stop to abort the measurement. If the record level is clipping (pinned at 0 dB) you should stop and reduce the input gain. Similarly if the record level is really low, you should increase the input gain.

## License

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