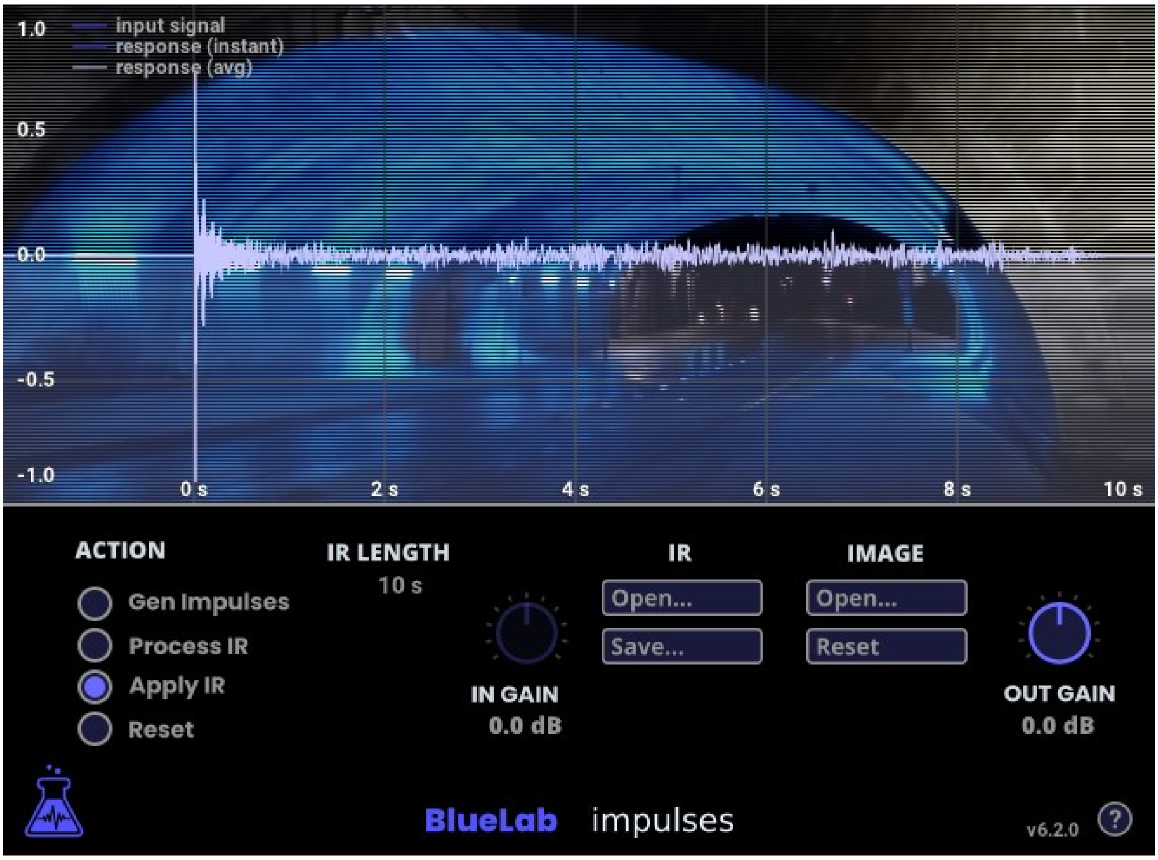
# BlueLab impulses

plugin version: 6.2.0
manual version: 3.0

www.bluelab-plugs.com

# DESCRIPTION

**Impulses** is a plugin that captures the acoustic characteristics of a place, then processes any sound to give the feeling that the sound has been played in that place. For example with **Impulses**, the reverberation of a room can be captured, and then this reverberation can be applied to a voice or a music instrument.
**Impulses** can also be used to capture and render the acoustic characteristics of a couple amplifier speaker/microphone.




# PRINCIPLE

Some **impulses** will be generated in the place we want to capture the acoustic characteristics, and **impulse responses** (IR) will be recorded.

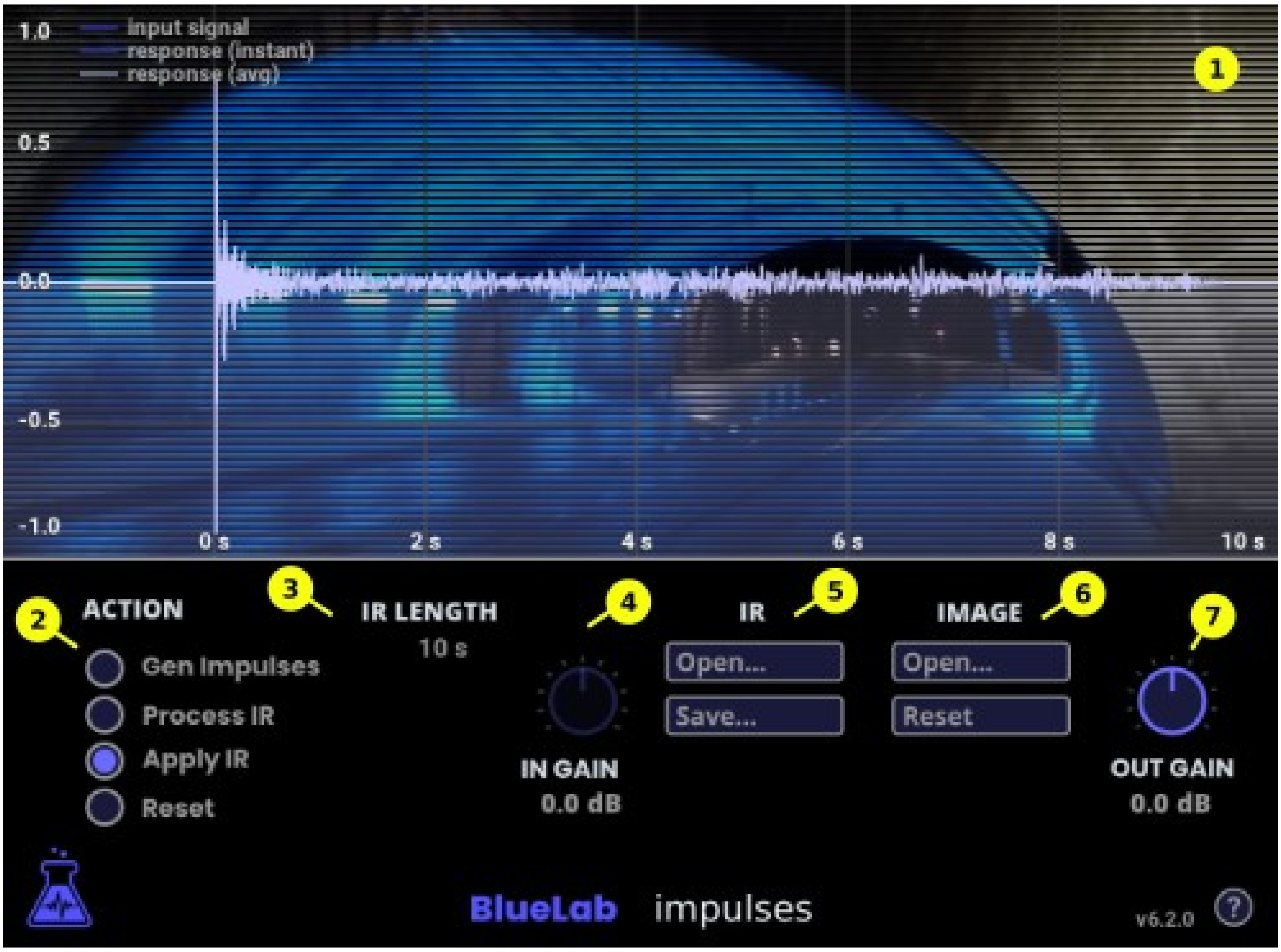
An impulse is a very brief sound, which must be loud enough.

The **Impulses** plugin generates a series of impulses, which can be played in a place that interest us, by using an amplified system.
An impulse can be generated by simply bursting an inflatable balloon in a place that interests us. The sound will be loud, very brief and this method will avoid to use and carry some heavy hardware in the place.
An impulse response is simply the recording of an impulse generated in a given place. Someway it is an “impulse with reverb”.

When an impulse is generated (bursting balloon) in a reverberant place, what we hear (and what we record) is the sound of this impulse colored by the place (the sound of a bursting balloon in a reverberant place). The impulse responses can be recorded simply using a small digital recorder.

The **Impulses** plugin captures and applies impulse responses (IR). It processes a series of impulse responses, by cleaning the defects from it. Then it applies the impulse response captured in a given place, to a track, using convolution. In other words, it “adds the reverberation of the place” to the track.

# USAGE




The **GRAPH (1)** displays the response currently processed in the series, **RESPONSE(INSTANT)**, and the average response, that is to say the result after the processing of the whole series, **RESPONSE(AVG)**. It also displays the input signal during the processing, **INPUT SIGNAL**.

The **ACTION (2)** parameter is the choice of the action to do (processing a series of responses, applying a response to a track…).

The **GEN IMPULSES** action is used to generate a track with impulses for recording it as an audio file and replaying it later in a place we want to capture the acoustic characteristics.
An impulse is generated for example every two seconds (the impulses sound like when hearing numerical audio clicks). This series of impulses will be

recorded on a track, or bounced, to finally get an audio file (WAV or AIFF) containing these impulses.
Later, this audio file will be played back, for example by copying it on a smartphone, then by connecting this smartphone to an amplified system.

The **RESP. LENGTH (3)** parameter is used here to define the spacing between the impulses of the series. For 50 and 100ms (milliseconds), the impulses will be spaced one second. For 1s (second), the impulses will be spaced 2 seconds. For 5s, they will be spaced 6 seconds, and for 10s, they will be spaced 11 seconds.

The **GEN IMPULSES** action is used in the case where the impulses are generated using an amplified system. But they could be generated mechanically (for example by bursting an inflatable balloon).

The **PROCESS RESPONSES** action is used to process the series of impulse responses, captured in a place that interests us, and get the result to apply to a track. The processing removes the incorrect responses from the series, attenuates the external disturbances which may have happened during the recording, and makes an average of the best responses.

The **IN GAIN (4)** parameter adjusts the gain of the input signal to process.
It could be sometimes necessary to increase the input level coming into the **Impulses** plugin, if the impulse responses series has been recorded with a too low level for the plugin. If the level is too low, the responses won't be displayed on the **GRAPH (1)** during the playback.

**Note:** If the impulses responses series was recorded in stereo, or if we want to apply a stereo impulse response, we will have to check that the track goes to stereo after having inserted the **Impulses** plugin. With many DAWs, this is automatic. But with some other DAWs, the track must be created as a stereo track before inserting the plugin.

With the **PROCESS RESPONSES** action, the **RESP. LENGTH (3)** parameter chooses the length of the response to process.
With a small value (for example 1s), the resulting response will be able to render the effect of a short reverberation. With a big value (for example 10s), the resulting response will be able to render some echo effects up to 10 seconds.
We will ensure that the value of the **RESP. LENGTH (3)** parameter matches the spacing between the responses in the recording. For example if in the recording we have a response every 2 seconds, we will choose a **RESP. LENGTH (3)** value until 1 second, but not more (5 or 10s).

To use the **PROCESS RESPONSES** action, insert the **IMPULSES** plugin on the track containing the series of recorded response, and start the playback.

During the playback, the **GRAPH (1)** displays the response which has just been read. If it is displayed in blue, it means that it is correct and will be taken into account for the calculation of the result. If ever the response is displayed in red, it means that it will be removed from the series because it is not correct. The **GRAPH (1)** displays the result too, which is refined during the playback.

At the end of the playback, we get the average impulse response, which could be applied to a track.

At this step it is recommended to select the **APPLY AVG RESPONSE** mode, and save the resulting configuration as a preset in the DAW, so the preset could be recalled later or in another project, and the response can be applied directly.

The **APPLY AVG RESPONSE** action is used to apply the average impulse response (and thus the acoustic characteristics of a place) to a track.

For example if we recorded a series of impulse responses in a tunnel, then processed these responses, and if we want to give the feeling that a music instrument is played in this tunnel, we will apply the corresponding average impulse response on the sound of the music instrument.

When the action is **APPLY AVG RESPONSE**, the **RESP. LENGTH (3)** parameter chooses the length of the response to apply. For example if we have captured an impulse response of length 10 seconds, a smaller value can be chosen for **RESP. LENGTH (3)**.
In this way the impulse response will then be truncated, and the computation will be faster when applying the response.

For example, with **RESP. LENGTH (3)** set to 50ms, 100ms, or even 1s, the impulse response can be applied in real time. Otherwise, for bigger values, the computation will be too long to be applied in real time. The DAW will have then to be used in “offline” mode (bounce…).

It is recommended to choose 10s for the value of **RESP. LENGTH (3)** for rendering a quite long echo effect.

The **BOUNCE AVG RESPONSE** action makes a bounce of the processed average impulse response, on the current track. This option is useful in order to import the impulse response in other software or plugins if possible. We bounce the response as a sound file, then read the sound file in another software.

The **RESET** action resets the previously processed impulse response. It is used if we want to erase the current response to process a new series of impulse responses.

The **IR (5)** parameter loads and saves impulse responses directly to disk. Either to transfer IRs between different DAWs for example, or to use them with an IR plugin from another brand.

Finally, the **IMAGE (6)** parameter loads an image corresponding to an impulse response and displays it in the background. The image will be saved in the setting as the same time as the impulse response. To remove a background image, use the image **RESET** button.

The **OUT GAIN (7)** parameter adjusts the output gain.

### *SUMMARY*

- Capture (and possibly clean) a series of impulse responses
- Import this series to a track, and insert the **Impulses** plugin on this track
- Choose the **PROCESS RESPONSES** action, and a length for the **RESP. LENGTH (3)** parameter (for example 10s)
- Launch the playback or make a bounce
- Choose the **APPLY AVG RESPONSE** mode, and then save the processed impulse response in a preset of the DAW

- Import the sound to process into a track
- Insert the **Impulses** plugin to this same track
- Recall the preset containing an impulse response previously processed
- Choose the **APPLY AVG RESPONSE** mode and a length with the **RESP. LENGTH** parameter
- Launch the playback of the track or make a bounce

**Note:** the plugin is relatively generic. Two examples are described below but one can find other usages!

# FAQ

### When the mode is PROCESS RESPONSES, the processing stops before the end and the DAW warns about a lack of resources

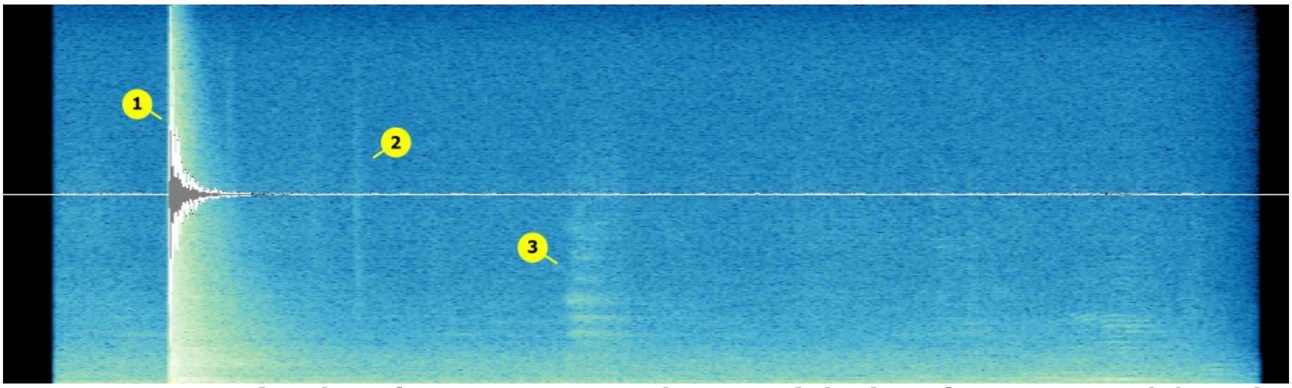
The processing of the series of impulse responses can require a large amount of resources with long responses lengths. To bypass this drawback, make a bounce with the DAW in **PROCESS RESPONSES** mode instead of starting the playback. At the end of the bounce, the series of impulse responses is processed!

### When an impulse response is applied to a track, there is saturation in the resulting sound

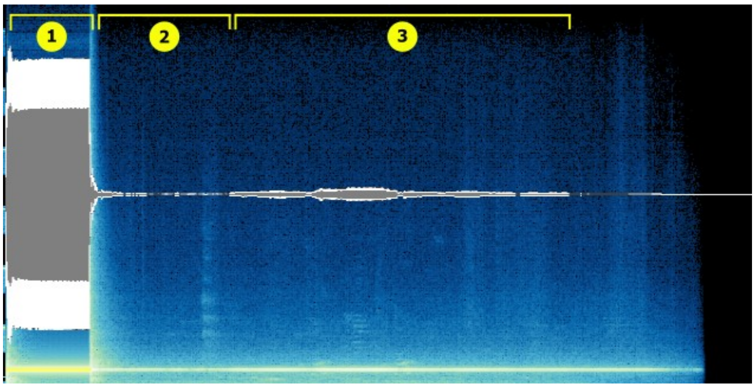
It can happen if the gain of the signal is very big, but doesn’t necessarily saturate when it is played as is. In this case, the gain of the original signal will have to be decreased to avoid the saturation of the result.

### When the impulse response is applied to a track, the sound level of the echoes increases after a while, instead of decreasing

If the sound level of the result increases whereas it should decrease (for example the echoes become louder and louder), it means that external sounds have been recorded while recording the impulse responses. And these external sounds are processed as if they were some acoustic responses of the place.




**Spectrogram of an impulse response – Main part of the impulse response (1), and external sounds on the recording (2) and (3)**




**Spectrogram of the result – Original brief sound (1), echo and echo decrease (2), sound level of the echo increasing abnormally after a while (3)**

To avoid this drawback, it may be necessary to clean the series of captured impulse response. We can suppress the impulse responses that has been disturbed by external sounds. Depending on the tools used for cleaning, we can then, suppress or attenuate the external sounds while keeping the captured impulse responses.

As a last resort, the length of the impulse responses can be diminished before applying it, by choosing a smaller value for the **RESP. LENGTH (3)** parameter.

### When the impulse response is applied to a track, the sound level of echoes doesn’t disappear totally or even much too slowly compared to what it should

If the sound of the echoes in the result never decreases, it could be because of a background noise (hiss) that is too loud in the recording of the series of impulse responses (or even wind noises if the recording has been done outdoors). This background noise may comes from the recording device. The **Impulses** plugin manages the background noise removal until a given level (-54dB). But above this level, it is recommended to process the recording "by hand" with a third party software to remove the background noise.

As a last resort, the length of the impulse responses can be diminished before applying, by choosing a smaller value for the **RESP. LENGTH (3)** parameter.

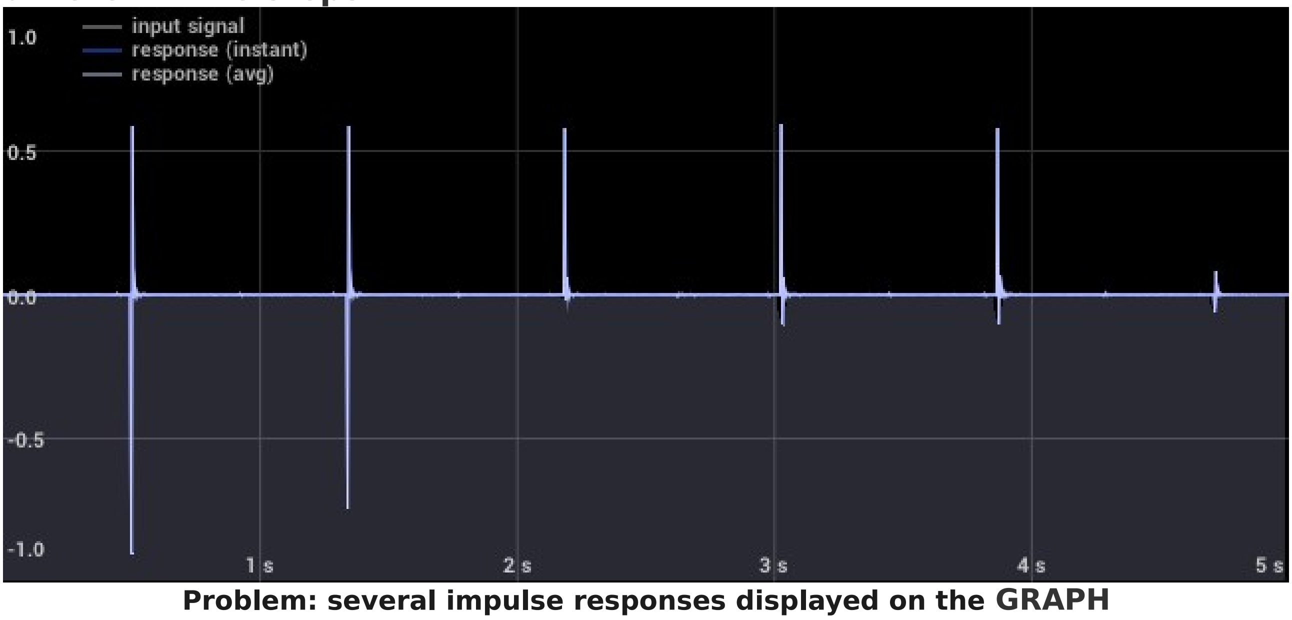
**When the impulse response is applied to a track, if the RESPONSE LENGTH (3) parameter is set to 10s, the DAW slows down when launching the playback**

This is not abnormal. When applying and impulse response of 10s, it requires a lot of calculations and therefore a lot of resources to make the processing. For these value, the impulse response must be applied by making a bounce instead of starting the playback.

**The plugin is in PROCESS RESPONSES mode, the playback is started, we can hear the sound of each response in the series, but no curve is displayed in the plugin**

This can happen if the level of the recording of the impulse responses series is too low. In this case, we have to increase the input gain with the **IN GAIN (4)** parameter so that the responses are detected even with a low recording volume.

**When in PROCESS RESPONSES mode, after having launched the playback, several impulse responses are displayed on the GRAPH at different time steps**




**Problem: several impulse responses displayed on the GRAPH**

In the case above, we recorded one impulse response every second, but we set the **RESP. LENGTH** time to 5s. The plugin then tries to process responses of length 5 seconds. But it processes in fact 5 impulse responses at the same time. That is not the result we want at all. There must be a single impulse response on the **GRAPH** at a given time. This means that the **RESP. LENGTH** time is too big, compared to the “rhythm” of the responses recorded in the series. In this case, the **RESP. LENGTH** parameter must be set to 1s.

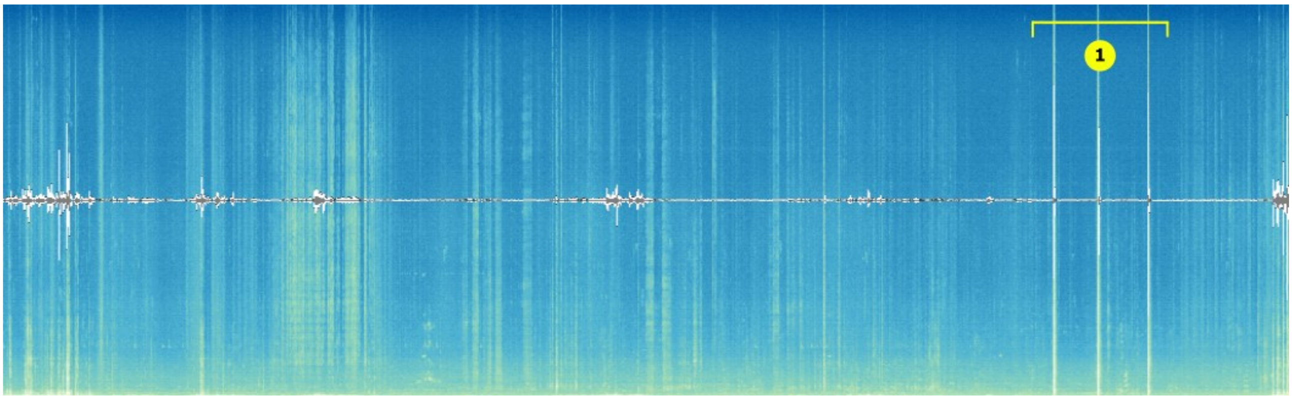
### I have a stereo impulse response, but when applying it, the sound seems to stay in mono

This can happen depending on which DAW is used. Sometimes the track doesn’t turn to stereo automatically when inserting the plugin. In this case the track must be created as a stereo track in the DAW, to be able to use a stereo impulse response in the best conditions. We can meet the same problem when processing the impulse response series.

# APPENDIX I – CLEANING THE RECORDING OF A SERIES OF IMPULSE RESPONSES

Before processing a series of impulse responses in the plugin, and even if we can get good results by using the rough recording of the series of impulse responses, it can be necessary to check and make a cleaning of the recording.

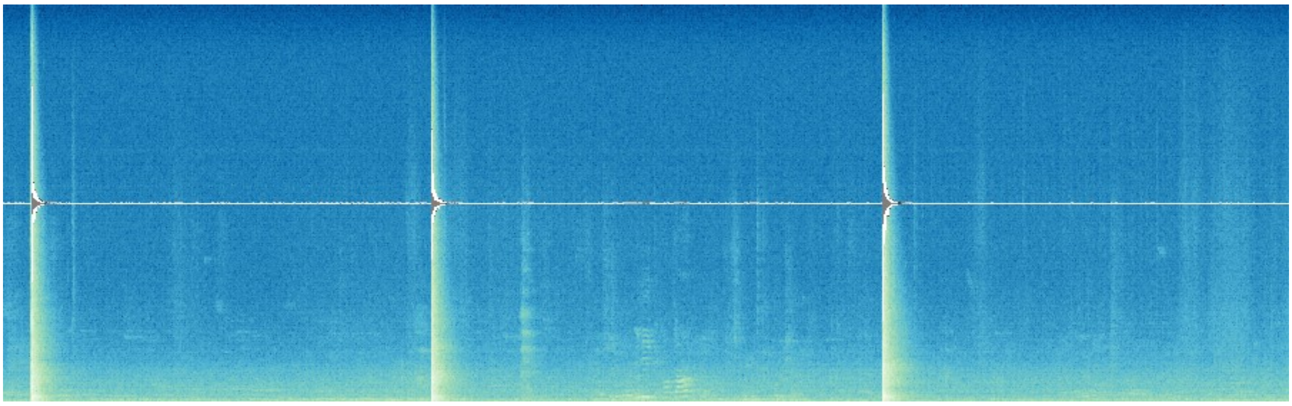
Especially to check that there are not too many external sounds on the recording (cars passing, steps noises, wind sounds…). In these cases, a cleaning of the sound file avoids future drawbacks.




**Spectrogram of the raw recording of a series of 3 impulse responses (the 3 impulse responses are at the end) (1)**

# EXTRACT THE RESPONSES

We will begin by extracting the impulse responses, by removing the parts of the file before and after the impulse responses series.



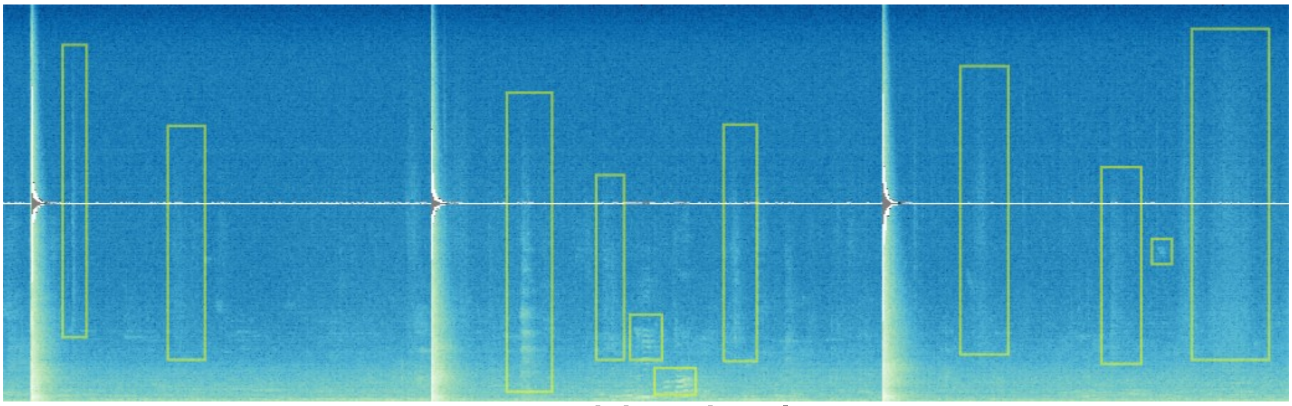
**Spectrogram of the 3 impulse response, after extraction**

This will avoid processing some sounds that are totally external to impulse responses. This will also avoid a too long processing by the plugin.

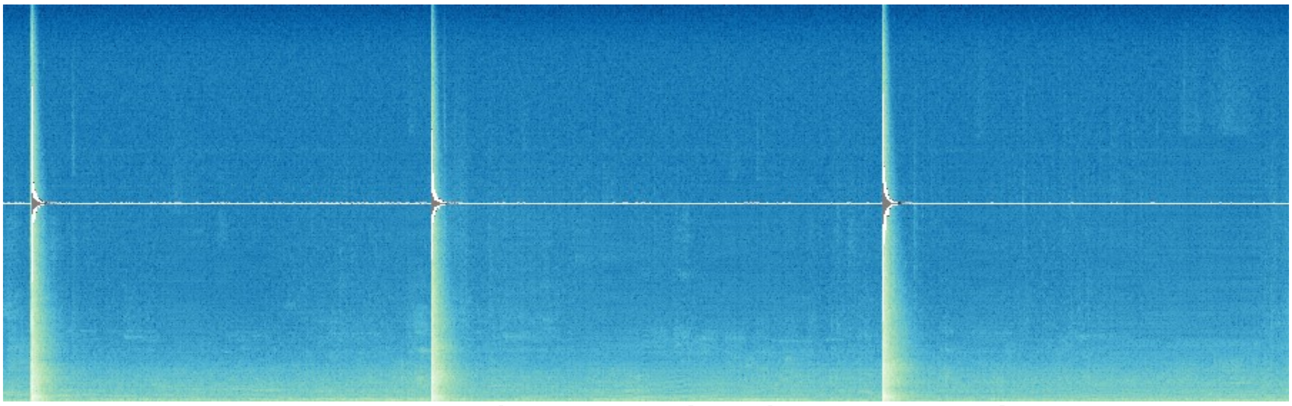
Be careful to keep enough time after the last response! In the example above we try to keep at least 10 seconds after the last impulse peak.
At this step, it is also possible to suppress an impulse response if it is too disturbed.

## SUPPRESS EXTERNAL SOUNDS

The next cleaning step aims to attenuate or suppress the external sounds from the series of responses (cars passing, steps, voices, wind sounds…). To achieve that, a third party software providing spectrogram editing should be used. Spectrogram editing is a feature that is more and more widespread and one can even find this feature natively included in the DAW.



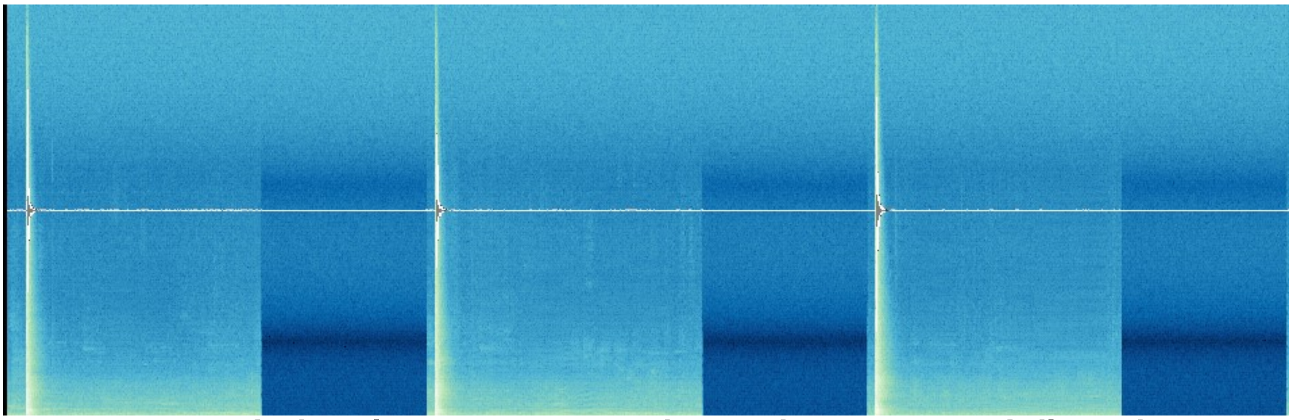
**Spectrogram containing 3 impulse responses**
**Many external sounds are visible**



**Spectrogram after attenuation of external sounds**

## ADJUST THE RESPONSES IN TIME

If the impulse responses are not adjusted in time as we want, some parts of the file between the responses can be cut, or some zones of silence can be inserted.



**Spectrogram of 3 impulse responses - We inserted some zones of silence between the responses, because there were too close to each other during the recording**

In the example above, we inserted some zones of silence between the response. Indeed, during the recording, the impulses were spaced by about 8 seconds.
By keeping the spacing of 8 seconds, we could have chosen in the **Impulses** plugin a value of **RESP. LENGTH (3)** at a maximum of 5 seconds.

By inserting zones of silence, we get a spacing of 15 seconds between the responses, and we will be able to choose a value of 10s for the **RESP. LENGTH (3)** parameter. We can then manage the entire length of 8 seconds of each response in the plugin, and then avoid losing the last recorded 3 seconds.

But we could have avoided this correction during the recording if we had spaced each impulse by 10 seconds or more.

# APPENDIX II – EXAMPLE OF CAPTURING THE ACOUSTIC CHARACTERISTICS OF A PLACE

## INTRODUCTION

We want to capture the acoustic characteristics of a tunnel, and apply them to a recording of a singing voice we have.

**The place from which we want to capture the acoustic characteristics**

**Some practical details:**

This is an outdoors place, so we will make the recordings by night to avoid as many ambient noises as possible.

As we don’t want to carry some heavy and expensive hardware on the spot, we will use some inflatable balloons that we will make burst to generate the impulses. The recording of three responses should be enough.

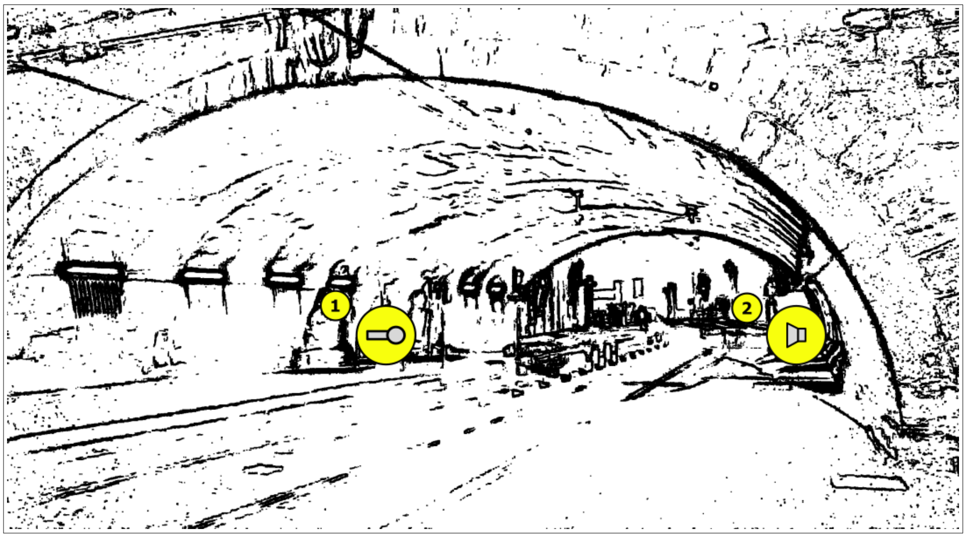
We will use a small stereo digital recorder. A stereo recorder captures some pairs of impulse responses, and provides the possibility to finally make a stereo sound rendering.

This is an outdoors place, so we will use a microphone windscreen (a basic one will suffice).

We will bring a little microphone stand to stare the recorder and turn the microphone like we want.

# SETUP

The whole setup is positioned about in the middle of the tunnel. The digital recorder is positioned at one of the edges of the tunnel, on a little stand, turned towards the opposite edge of the tunnel.




**Setup on the spot (almost in the middle of the tunnel)**
**The recorder (1) is put on the left, on a little microphone stand**
**We are positioned on the right, in front of the recorder (2) to generate the impulses**

We are positioned in front of the recorder, at the opposite edge of the tunnel, to generate the impulses.

In the final rendering, the apparent position of the listener will be the position of the digital recorder, and the apparent position of the sound source will be the point where the inflatable balloons burst.

# RECORDING

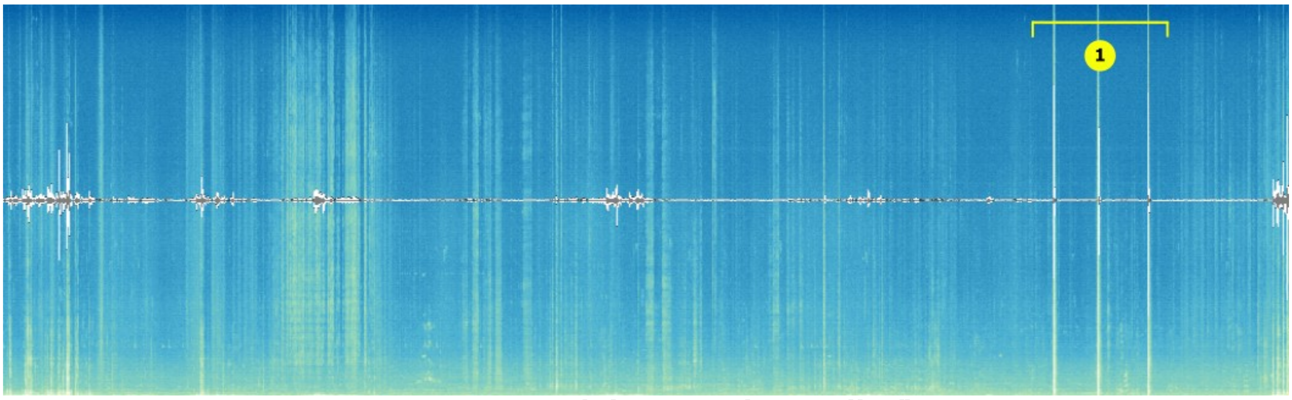
We setup the input level of the recorder, we check that the recorder is set to record using a non-compressed file format (WAV, AIFF…), and we launch the recording.

We go in front of the digital recorder, at the opposite edge of the tunnel, and we inflate three balloons in anticipation. The balloons will be strongly inflated, to have them burst briefly and with a strong sound volume. One by one, we place the balloons at about 1.5m from the ground, and we make them burst using a pin. We take care to wait at least 10 seconds between each burst.

We stop the recording.

This is all for the recording of the impulse responses!

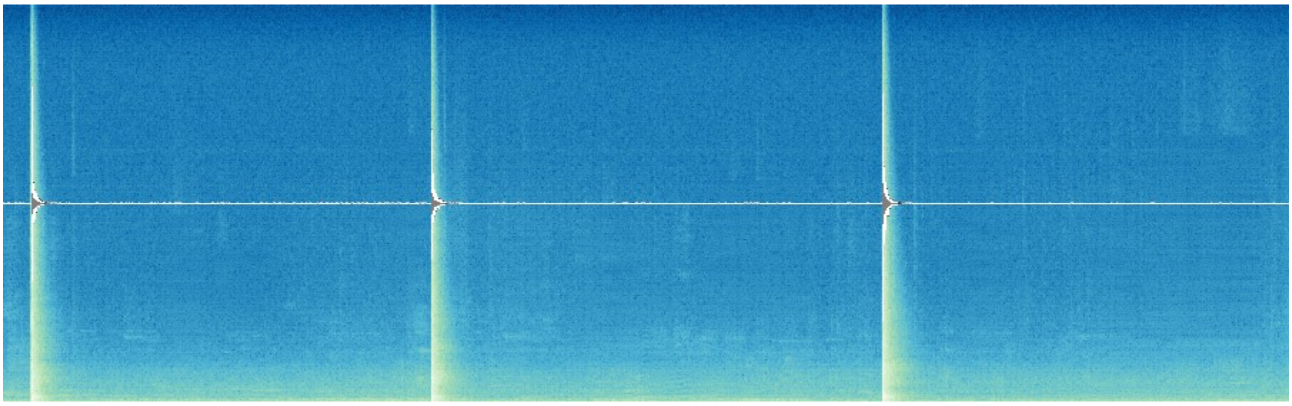
We get the following audio file:




**Spectrogram of the resulting audio file**
**The 3 impulse responses corresponding to the balloons bursts are in (1)**

## CLEAN THE RECORDING

We couldn’t have avoided recording some car noises, a voice far off, and some other noises. So we will clean the recording.



**Spectrogram of the audio file after cleaning**

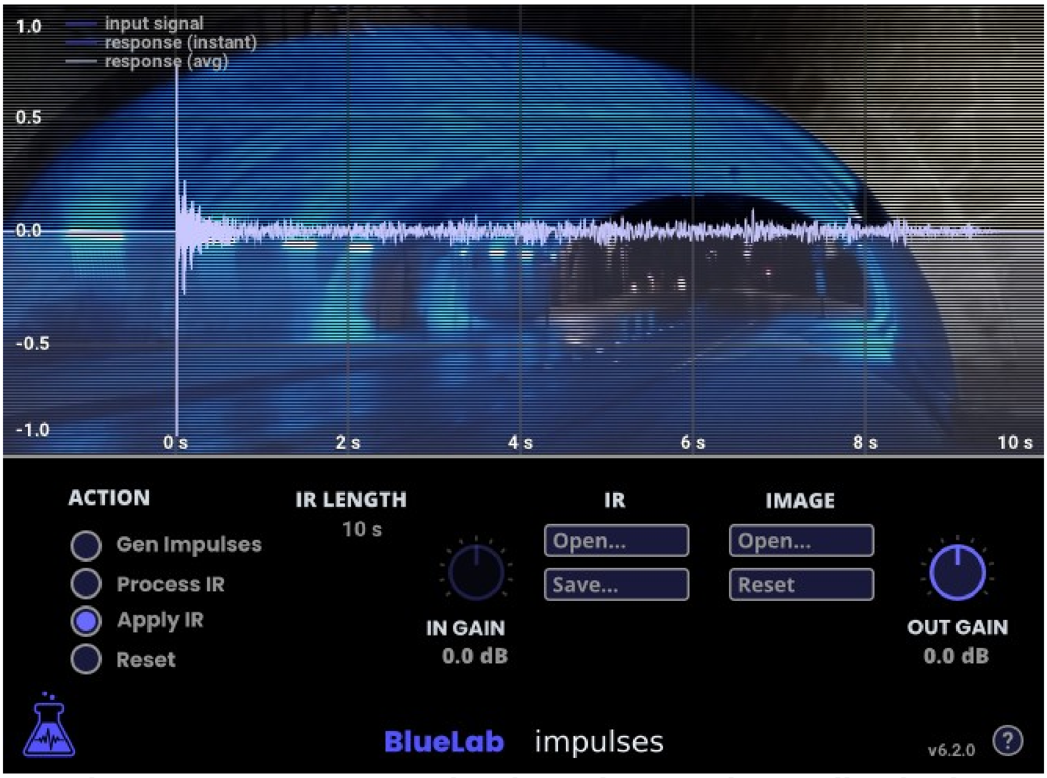
## PROCESS THE SERIES OF RESPONSES

We then import the audio file in the DAW, and process the series of impulse responses with the **Impulses** plugin.

We import the audio file in a new track, and insert the **Impulses** plugin.

Our recording is stereo, so we will check that the track is a stereo track, and that the **Impulses** plugin is inserted to operate in stereo. So the final result will sound better.

We choose the **PROCESS RESPONSES** mode, with a value for **RESP. LENGTH(3)** of 10s, and we launch the playback of the track. At the end of the playback, the average response has been computed.
We choose the **APPLY RESPONSE** mode, open an image to be displayed in the background, and save all this as a preset in the DAW.




**Average impulse response resulting from the sound recording in the tunnel**
**The environment was very reverberant, so**
**the impulse response is quite long (about 10s)**

# APPLY THE RESPONSE TO A TRACK

We can finally apply the impulse response we got, and so apply the acoustic characteristics of the tunnel, to the singing voice recording we have.

For that, we create a new project in the DAW, we import the singing voice recording on a new track, and we insert the **Impulses** plugin on the same track.

We recall the preset corresponding to our impulse response that we previously saved.
And we make a bounce of the track to get the result.

Instead of making a bounce, we could also have decreased the **RESP. LENGTH (3)** parameter in order to make a real-time processing (set to 1s for example if possible). But the result for this very reverberant place sounds better if we keep the parameter set to 10s.

We will be able to apply this impulse response to any other audio track later.
So we get the sound of our singing voice, with the specific reverberation of the tunnel we were interested in!

# APPENDIX III – EXAMPLE OF CAPTURING THE ACOUSTIC CHARACTERISTICS OF A GUITAR AMPLIFIER SPEAKER AND A MICROPHONE

## INTRODUCTION

We have a guitar amplifier and a microphone for the sound recording in front of the amplifier. We want to capture the acoustic characteristics of this speaker/microphone couple, in order to later apply these characteristics to various direct guitar recordings (without amplifier).

When we often have to take sound shots of guitars, we sometimes have only temporarily access to a particular model of interesting amplifier, or to a model of particularly efficient microphone. We would like to render the sound this hardware provides on future guitar recordings, when we will no longer have the hardware available.

Or else we have a sound shot of guitar which has been recorded in direct mode, and we want to test it by passing it through many guitar amplifier models, using various models of microphones, arranged in front of the amplifiers at different positions and orientation.

With the **Impulses** plugin, it is possible to test many configurations in some minutes, if we have previously captured the acoustic characteristics of the systems.

In general, using impulse responses can simplify the setup when using the re-amping
technique.

## SETUP

To be able to acquire the acoustic characteristics of both the speaker and the microphone together, we set the amplifier and the microphone as if we had to make a standard sound recording of a guitar. So the microphone is placed on a stand in front of the speaker.

The microphone is connected to a recording system (sound card + computer, mixing console + digital recording…).

We plug a guitar to the input of the amplifier, and setup the amplifier knobs to get the sound we want (volume, EQ/tone...).

Once the amplifier is adjusted as desired, we unplug the guitar and make arrive a series of impulses instead. For that, a numeric audio player containing a track of impulses can be plugged to the jack input of the amplifier.

# RECORDING

For the sound shot, we make enter a series of impulses by the guitar amplifier input. This series has previously been generated by the **Impulses** plugin with the **ACTION (2) GEN IMPULSES** and we have the corresponding WAV or AIFF file.

With one impulse every second or every two seconds in the series, we are sure to acquire well the full speaker response. For this purpose, we choose the **ACTION (2) GEN IMPULSES** and a value of **RESP. LENGTH (3)** smaller than 5s.

We may want to set a long reverberation in the amplifier. For this, we will need a bigger value for the **RESP. LENGTH (3)** parameter to acquire the reverb.

The impulses audio file could be played back from a computer or another device connected to the input of the amplifier.

It is also possible to send the impulses directly from the plugin, without using an audio file. To do that, send the audio output of the computer to the input of the amplifier, launch the DAW, load the plugin and choose the **ACTION (2) GEN IMPULSES**.

We then launch the recording of the impulse responses, coming from the impulses entering in the amplifier, and going outside through the microphone.

We finally get an audio file containing the series of impulse responses.

# PROCESS THE SERIES OF RESPONSES

We then import the responses audio file in the DAW, and process the series of impulse responses with the **Impulses** plugin.

We import the audio file in a new track, and we insert the **Impulses** plugin.

We choose the **PROCESS RESPONSES** mode, choosing 1s for the **RESP. LENGTH(3)** parameter, and we launch the playback of the whole track.

At the end of the playback, the average response has been computed. We choose the **APPLY AVG RESPONSE** mode, and we save all as a preset in the DAW.

# APPLY THE RESPONSE TO A TRACK

We can finally apply the impulse response we got, and so apply the acoustic characteristics of the guitar amplifier speaker and the microphone to a guitar recorded in direct mode.

For that we create a new project in the DAW, we import a direct guitar recording on a new track, and we insert the **Impulses** plugin on the same track.

We recall the preset that we previously saved, corresponding to our impulse response.

We finally launch the track playback or make a bounce of the track to get the result.

The **RESP. LENGTH (3)** parameter could be decreased to 100ms for example in order to make a real-time processing using less resources.

The impulse responses corresponding to sound shots in front of an amplifier are often much shorter than one second!

This impulse response can be applied later to any other direct guitar track.

So we get the sound of a guitar as if it was played with a guitar amplifier, from the sound of a guitar recorded without amplifier!

**Note:** We can get the response of the speaker and the microphone, some amplifier EQ characteristics, some amplifier reverb characteristics, but it is not possible to render distortion, crunch, and other effects.