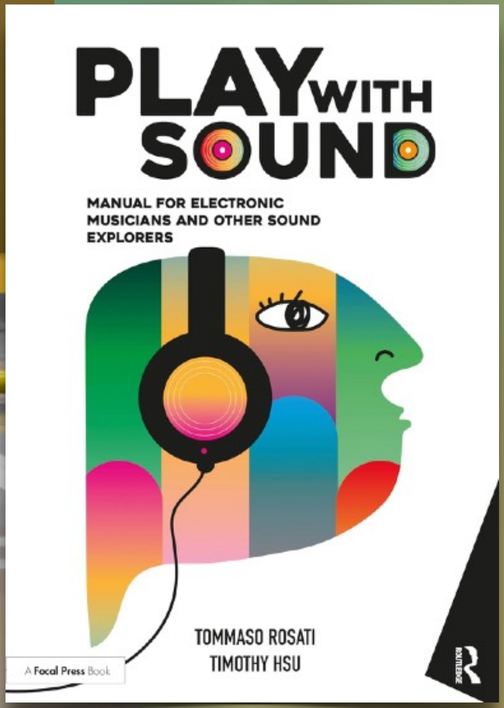
REVERB

NATURAL REVERB
CLASSIC DIGITAL REVERB
CONVOLUTION REVERB

TOMMASO ROSATI
SOUND ART

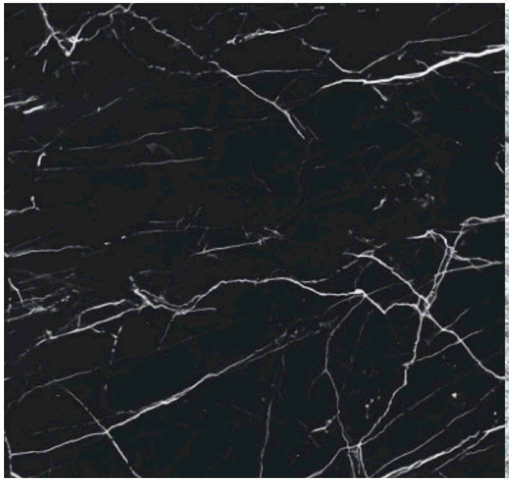
THE BOOK IS NOW AVAILABLE!



WHAT IS A REVERB?

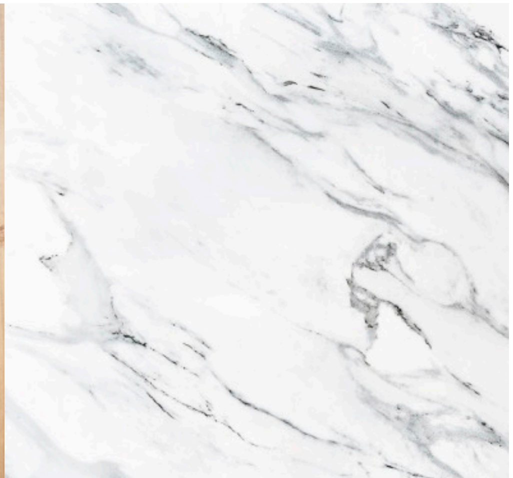
Reverb is the natural acoustic phenomenon that results from sound waves interacting with and bouncing off surfaces in a room.

Different surfaces, different reverberations




Reverberation is affected by the size of the room, the material of the surfaces, and the overall geometry of the room, all contributing to create the unique “sound” or “color” of the room.

When sound reaches a surface, some energy is absorbed, and some energy is reflected, depending on the material of the surface.

Dimensions

The room's size or volume greatly affects the reverberation type, length, and "color."

If the materials are the same, the larger the room volume, the longer the reverberation.

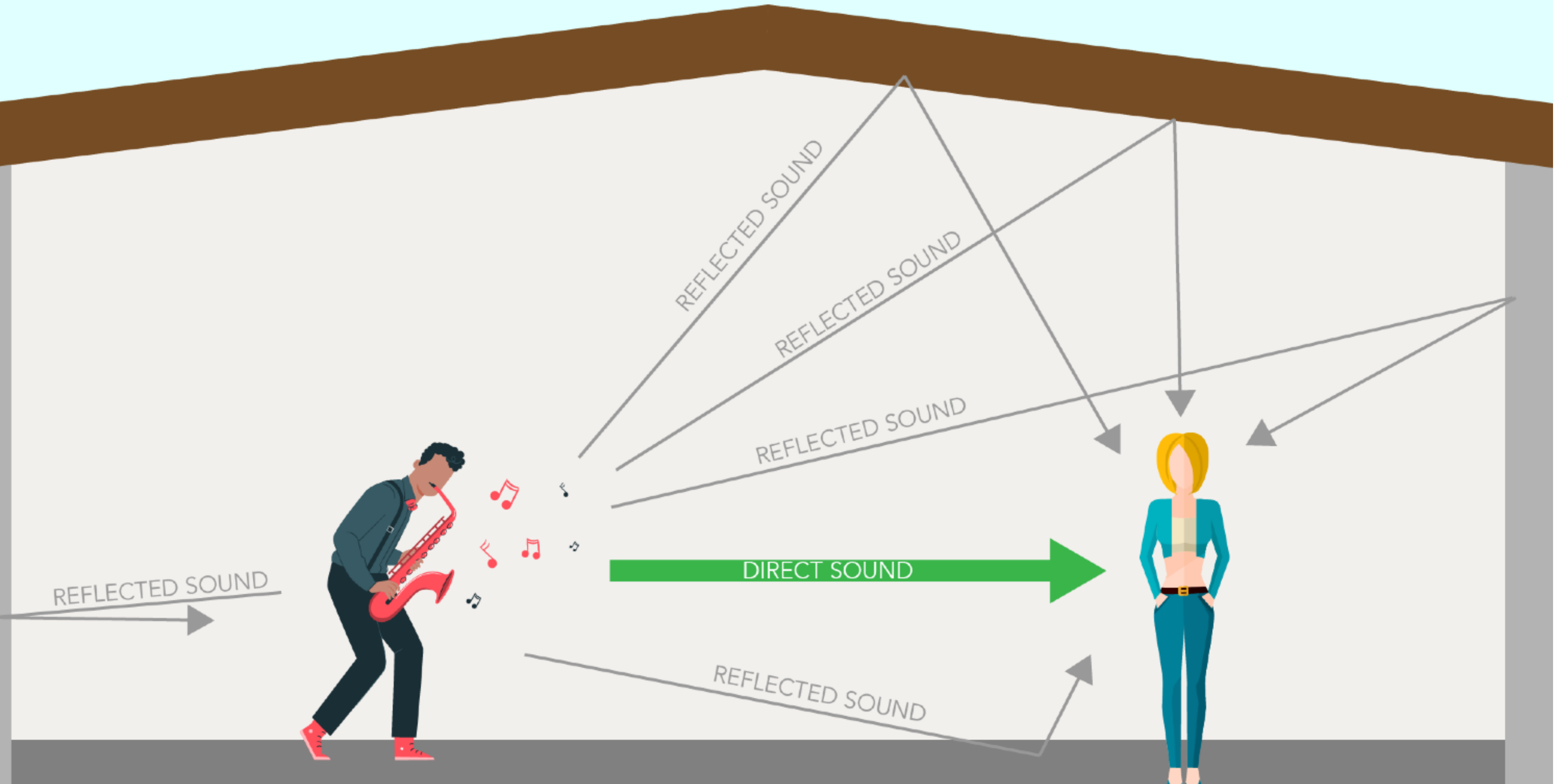
Open space

If surfaces are absent, such as playing in the middle of a meadow in open air, we generally will not experience significant reverberation of our sound.

Direct sound, Reflected sound

In the case of enclosed spaces, the sound reaching the listener is of two types:

1. **Direct sound**
2. **Reflected sound**



Direct sound

Direct sound is defined as the sound from a source that reaches the listener without encountering any obstacles or reflective surfaces of any kind.

It is:
- **Attenuated in amplitude due to the inverse square law**

Reflected sound

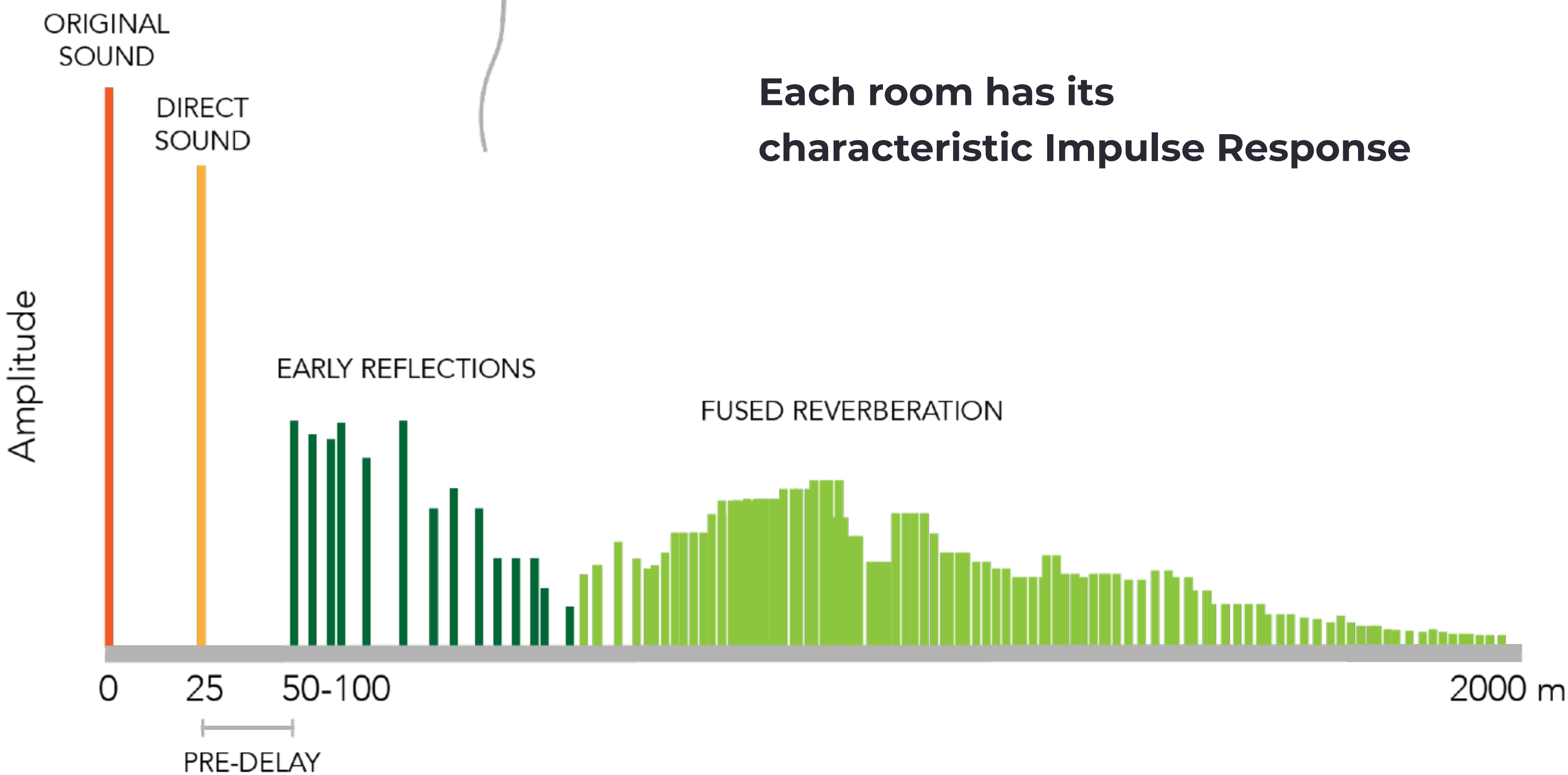
Reflected sound is defined as all the sound waves that reach the listener after reflecting off the room surfaces.

It is:
- **Filtered**
- **Attenuated in amplitude**
- **Time Delayed**

Impulse response

If we create a rich, impulsive sound in a room, such as a balloon pop or a loud bang with very fast attack and decay, and measure the amplitude of the sound over time, we get the room's Impulse Response.

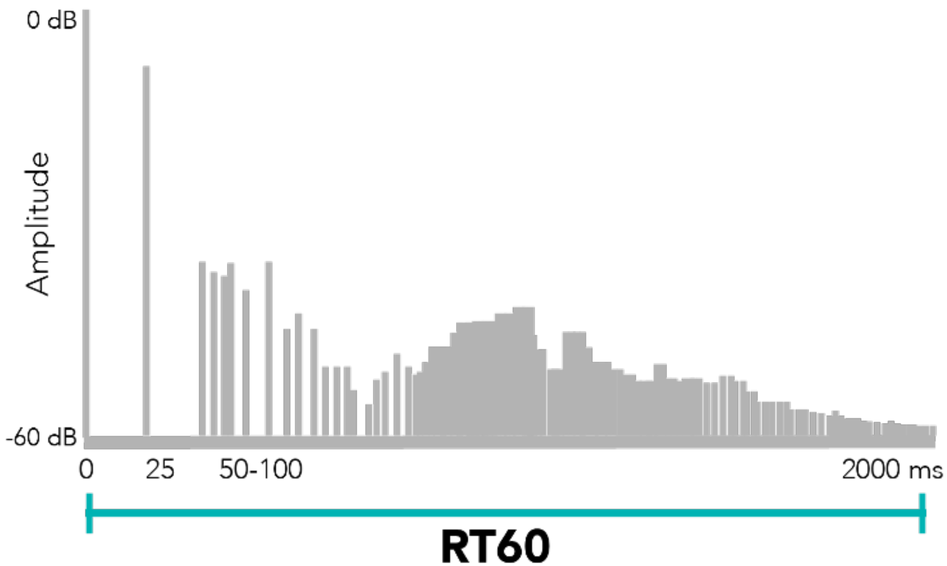
Each room has its characteristic Impulse Response



Reverberation time: RT60

RT60 reverberation time is a measure of the length of reverberation.

It's the amount of time it takes a sound to attenuate by 60 dB

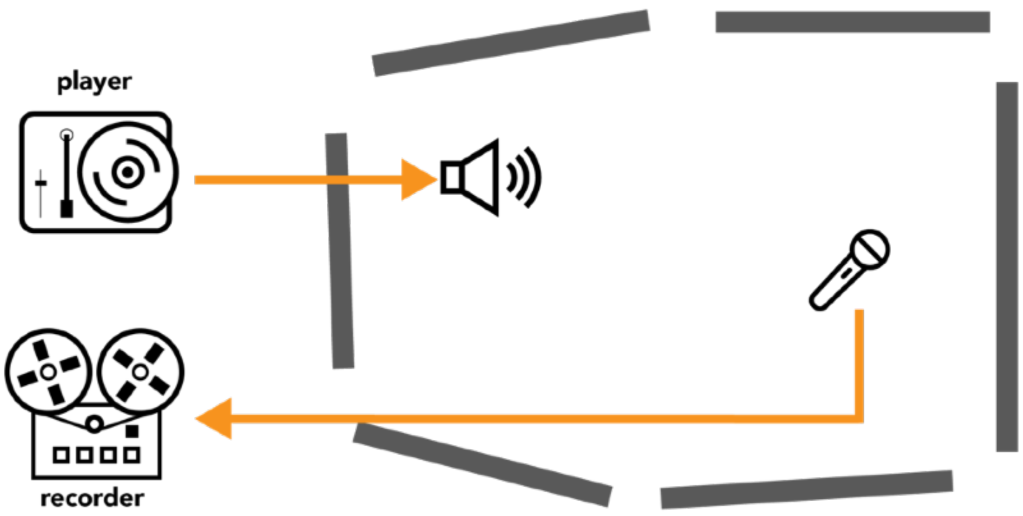


	RT60 (average values)
Small bedroom	**0.1 - 0.5 s**
Recording studio	**0.3 - 0.7 s**
Concert hall	**1.5 - 3 s**
Cathedral	**3 - 10 s**

Artificial acoustic reverbs

Acoustic echo chambers

Acoustic echo chambers are real rooms where sound is played back from a speaker into a room. The sound then propagates in the room where a microphone and a recording system re-records the resultant sound.



Electromechanical reverb

Electromechanical reverb uses the interaction between the electrical signal and mechanical parts to introduce a reverberation effect to a given sound.

The most commonly used mediums are springs and plates, resulting in **Spring Reverb** and **Plate Reverb**, respectively.

SPRING REVERB

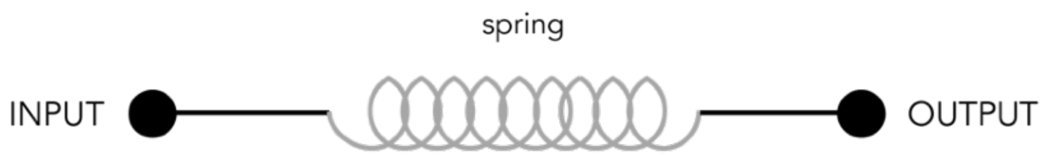
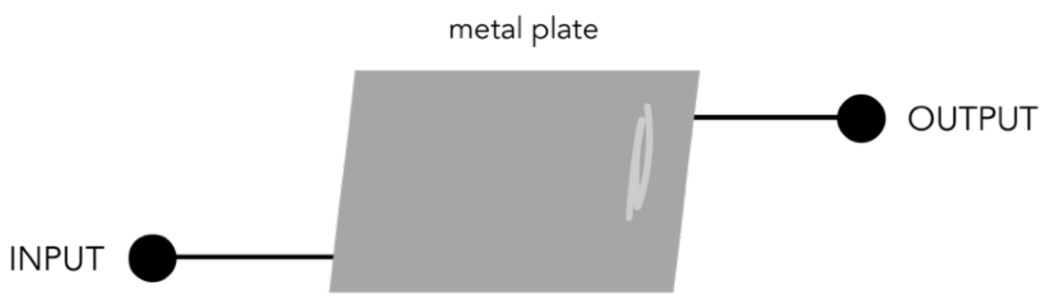


PLATE REVERB



Classic digital reverb

In digital reverb, an algorithm simulates the various components of natural acoustic reverberation.

The first person to create digital reverberation algorithms was **Manfred Schroeder** in 1961

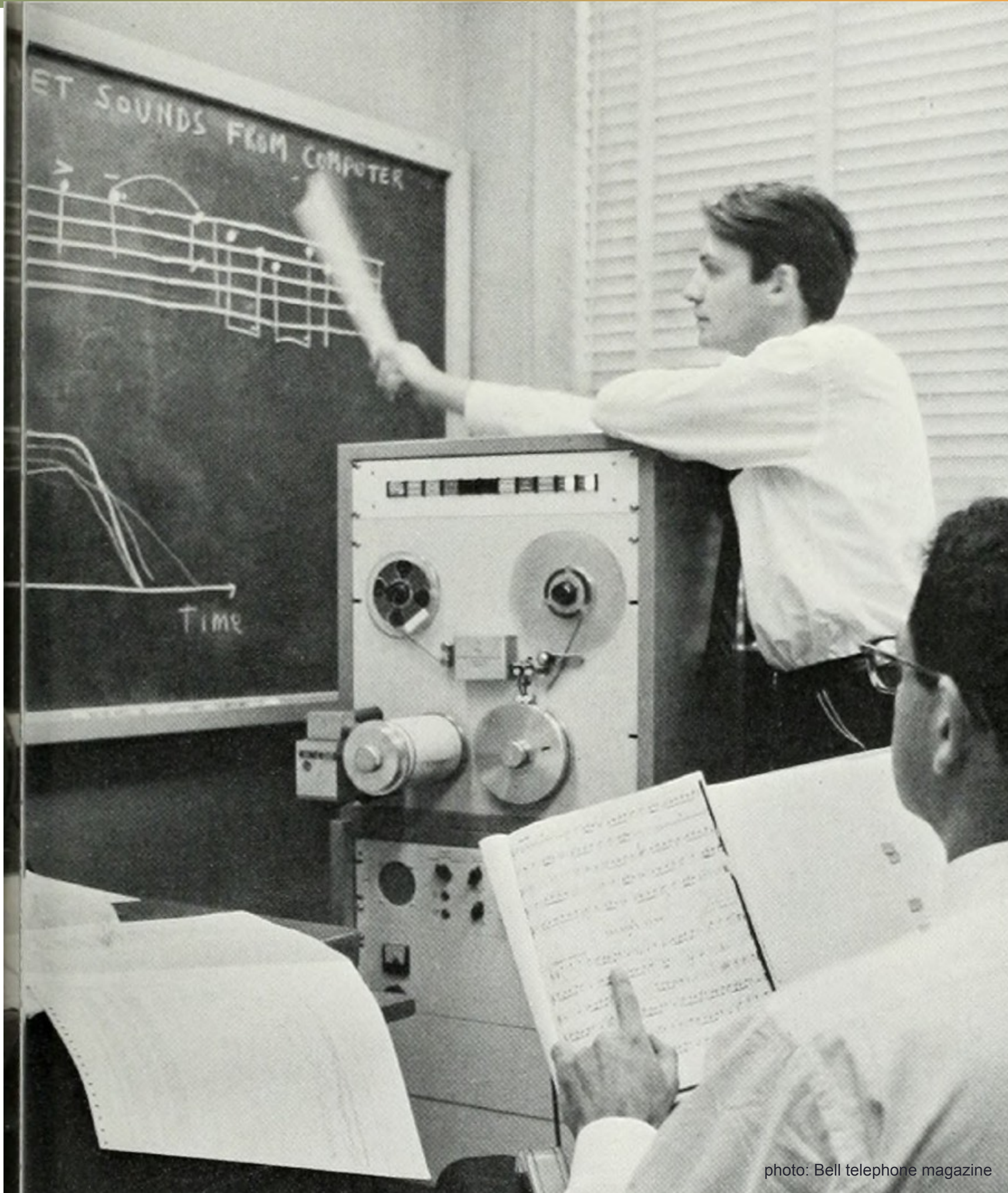
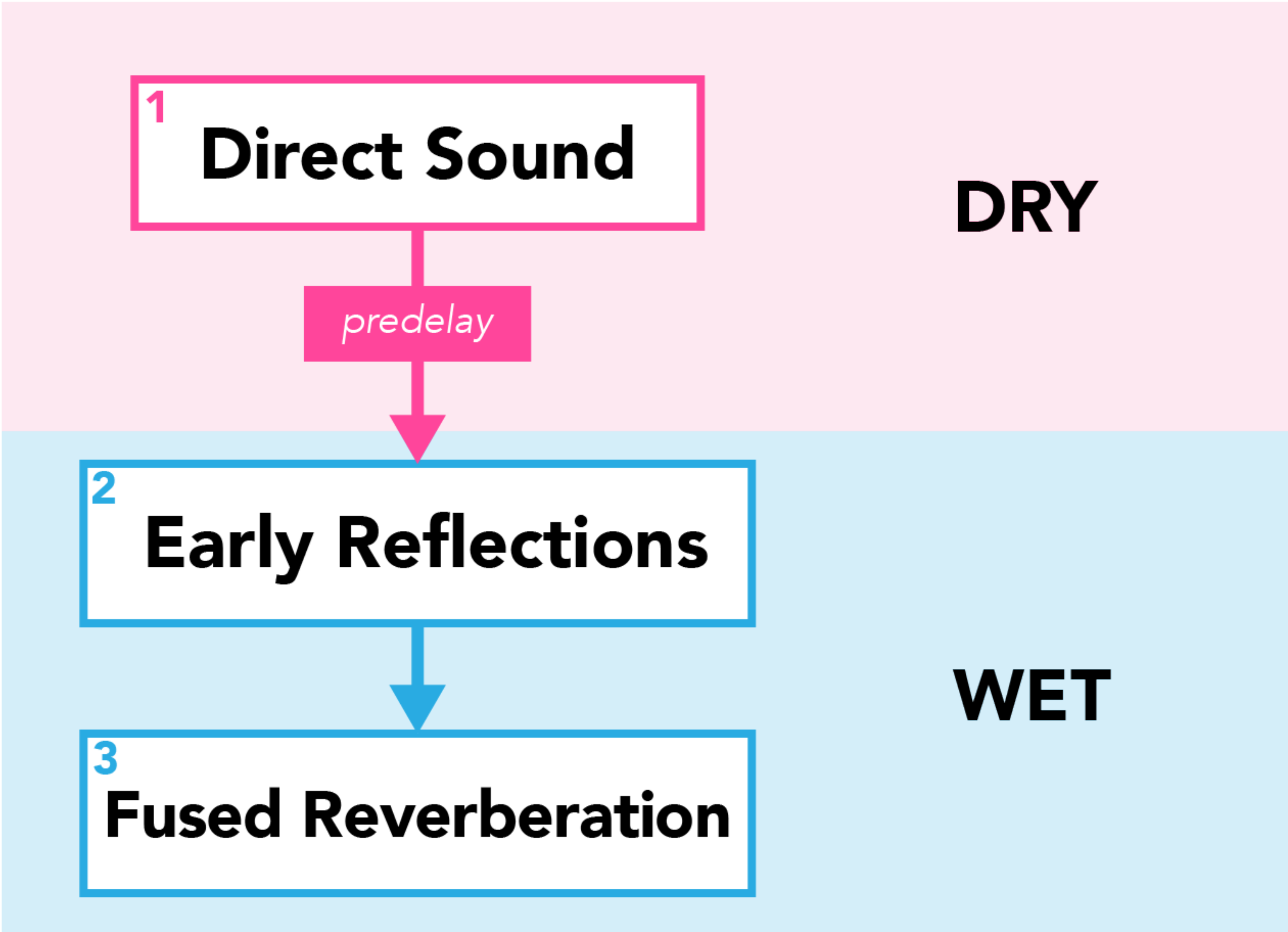
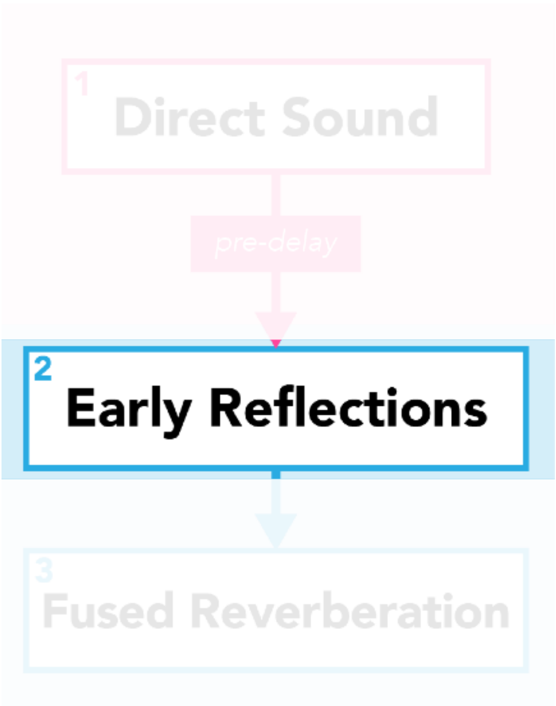
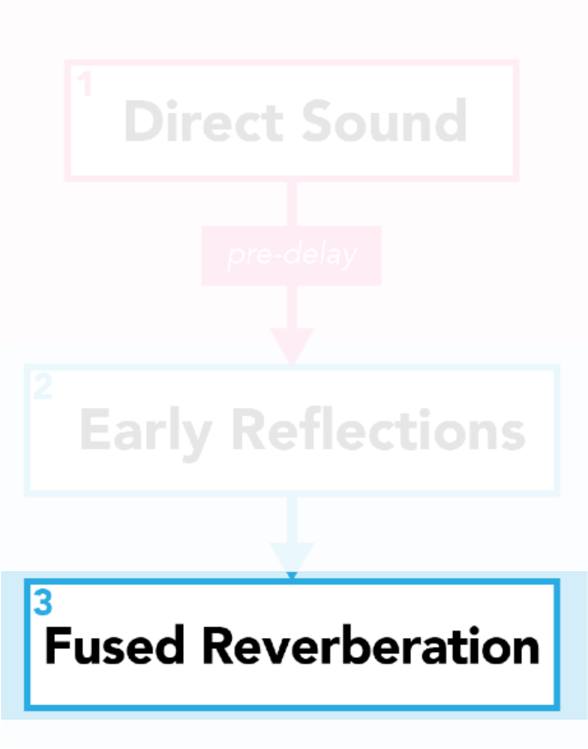


photo: Bell telephone magazine


1
Direct Sound
predelay
DRY
2
Early Reflections
3
Fused Reverberation
WET



Multitap delays are used to simulate these first early reflections.

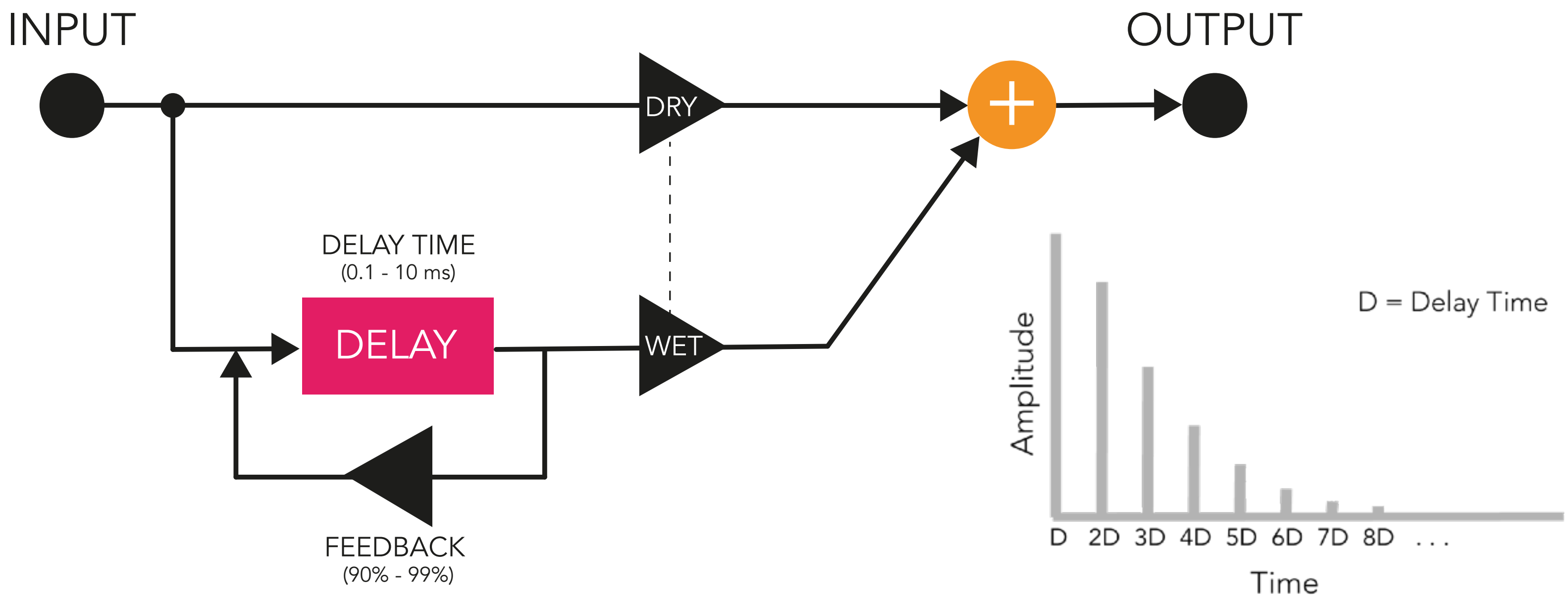


To achieve this, we utilize, connect, and duplicate two processes:

Comb filters and All-pass filters

Comb filters

The Comb Filter is a delay with feedback that has a very short delay time.



All-pass filters

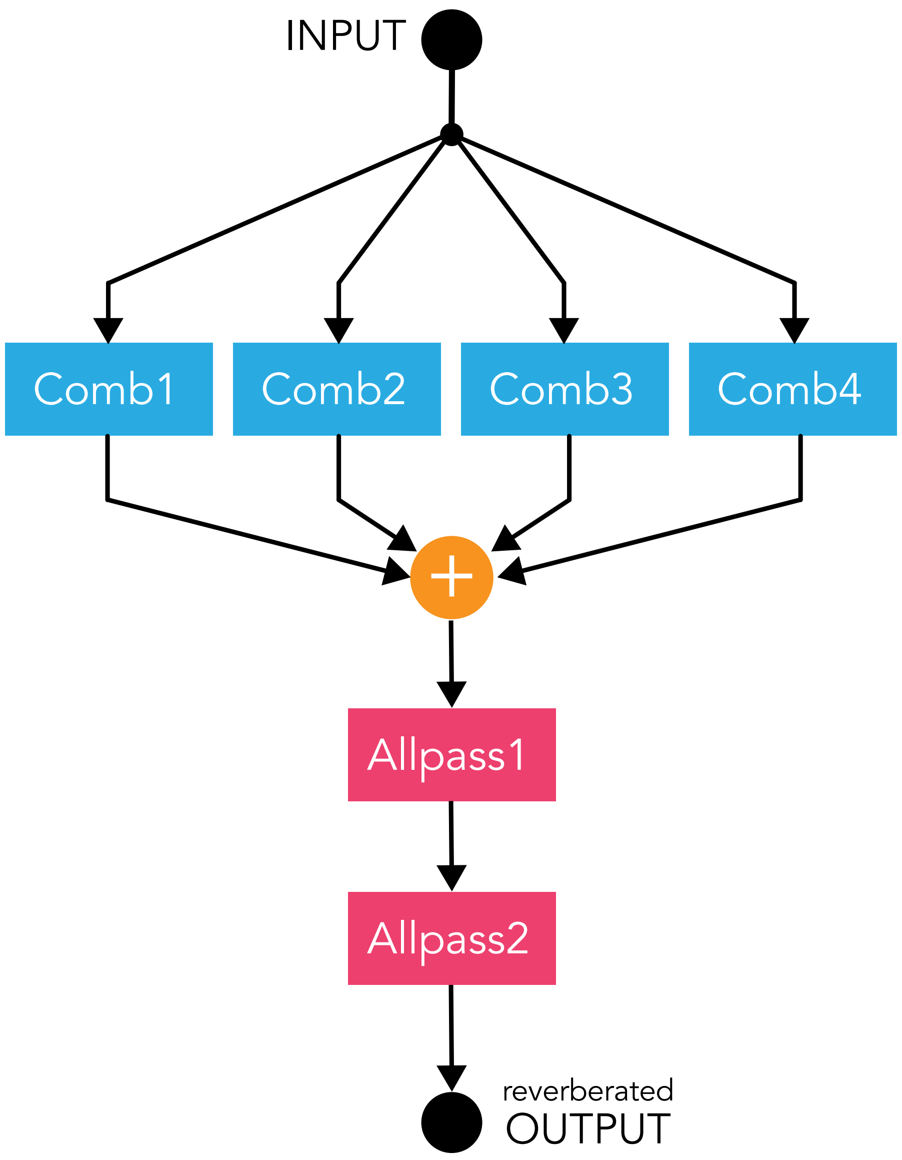
The All-pass filter will pass through amplitudes unchanged but will alter the phase depending on frequency.

Combining comb and all-pass

To achieve sufficient density to perceive a natural reverberation, it is necessary to combine multiple comb and all-pass filters together.

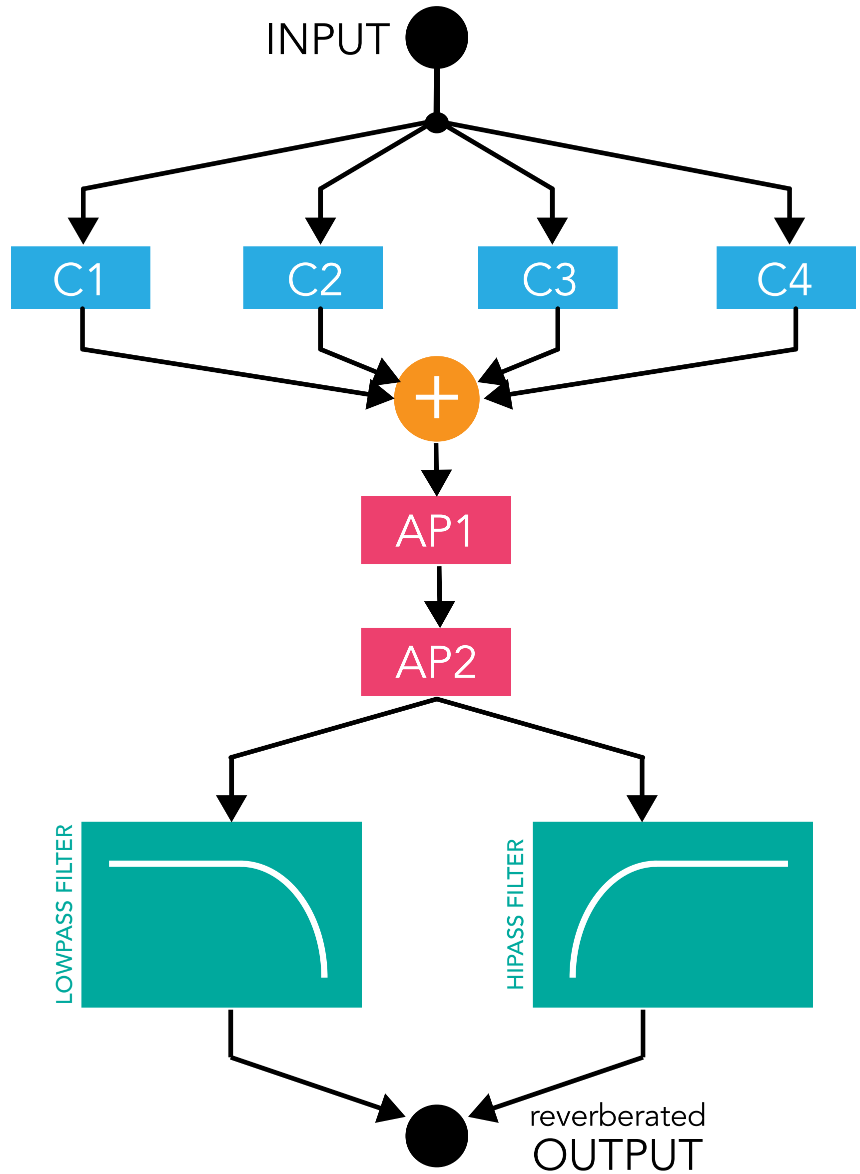
In parallel fewer processes are needed because you will add up the reflections given by each comb or all-pass filter. This is generally good for comb filters because you minimize any spectral problems.

In series more density in processes occur because each reflection is multiplied by subsequent reflections.



Filters

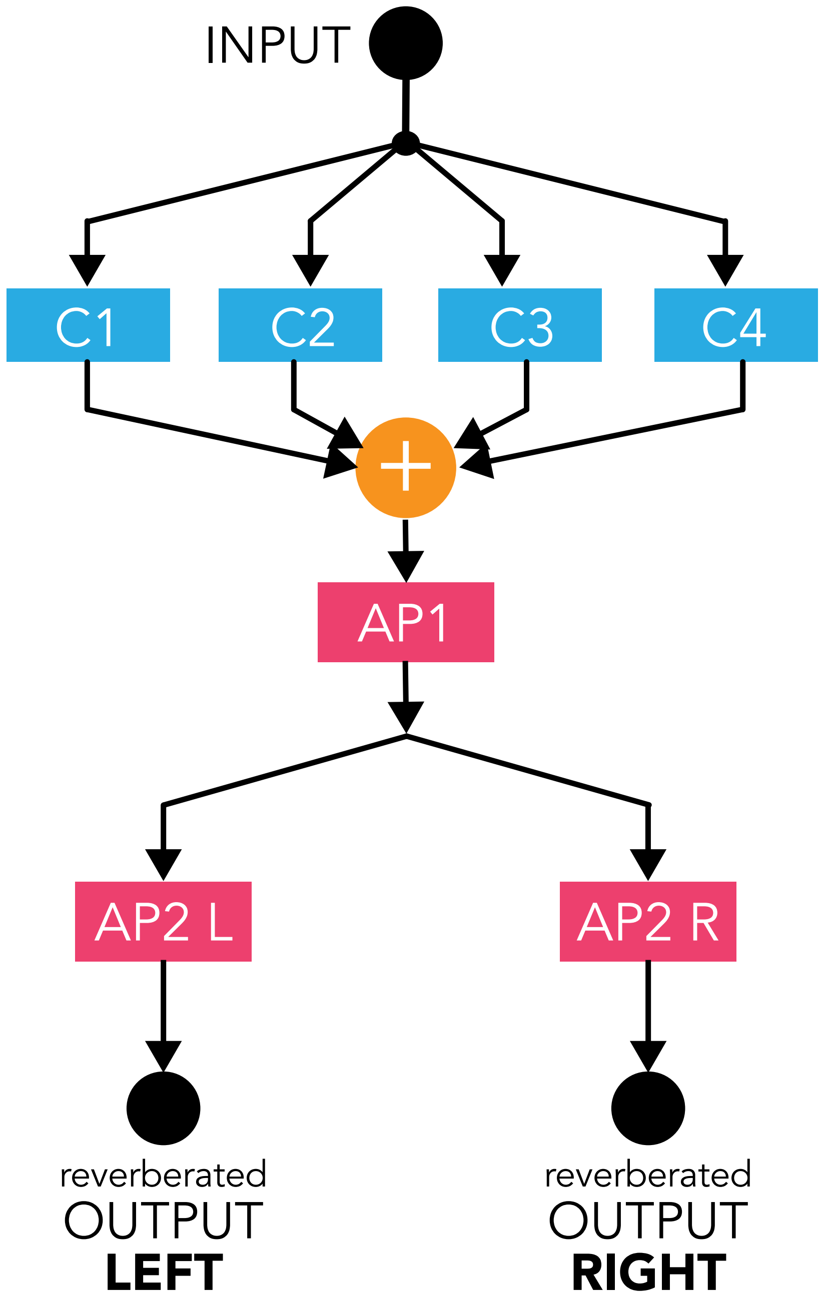
To further refine our digital reverberation, we can use Low-pass or High-pass filters at the end of the algorithm to simulate the natural absorption



Stereo reverb

The easiest way to ensure that the left and right channel are slightly unique is to apply a different all-pass filter to the each channel.

One all-pass filter per channel in the final process is sufficient to create a perceptually acceptable stereo reverb.



Reverb settings

The **parameters** we see in reverb are the controls that allow us to set the characteristics of our desired reverb.

In most cases, these are macro-parameters settings that work on multiple parameters within the reverb units.

PREDELAY — sets the delay time between the original sound and the first reflections

SIZE — creates the delay times of the various reverberating units, simulating the feeling of room size.

REVERBERATION TIME — controls the overall duration of the reverberation

DIFFUSION — determines the density of reflections that our reverberation creates

FILTER or DAMPING

simulates the absorption of certain frequencies caused by absorptive properties of the walls

REVERB TYPE

allow for presets that simulate particular types or analog reverbs by setting various parameters to specific values. Examples: Plate, Spring, Chamber, Hall, Gated

MIX or DRY/WET

controls the ratio between the amplitude of the original sound to the amplitude of the reverberated sound.

Digital convolution reverb

Convolution reverb uses a mathematical operation called convolution to transform our original sound to reverberate as if it were in a simulated room.

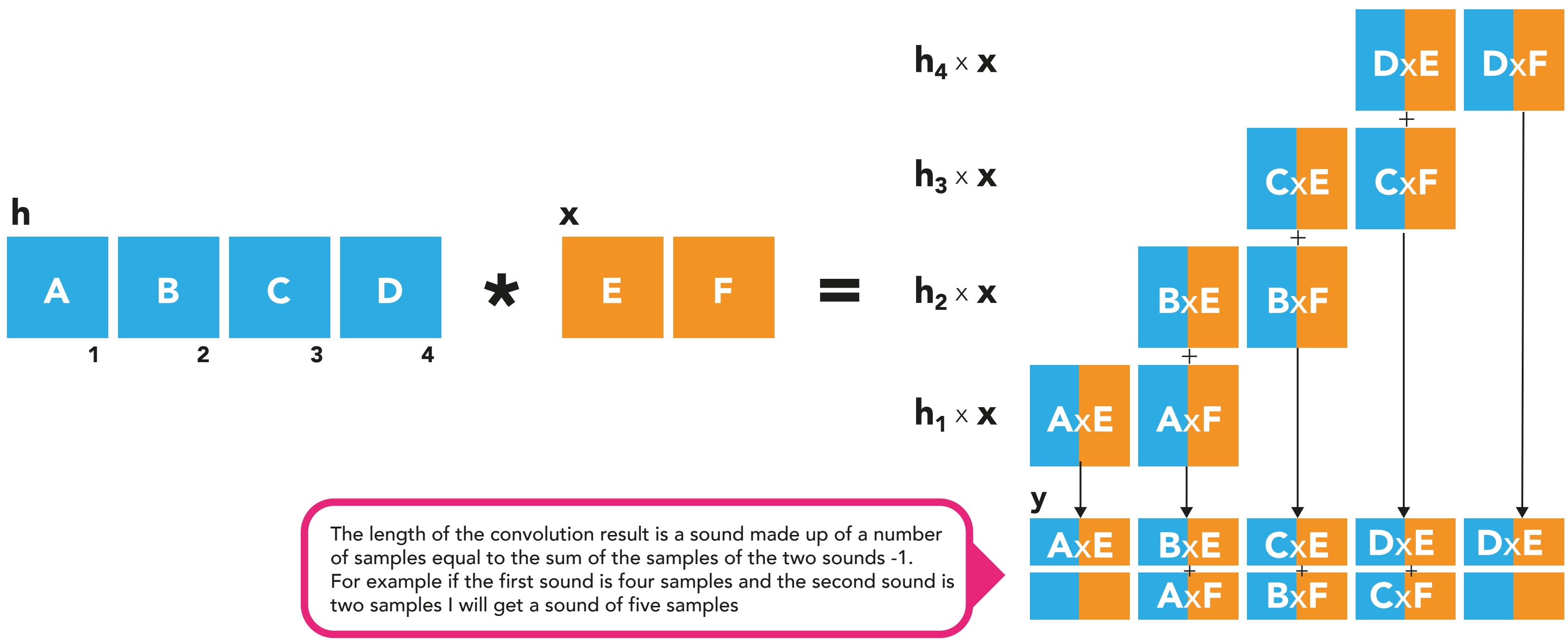
Symbol of convolution operation

$\ast$

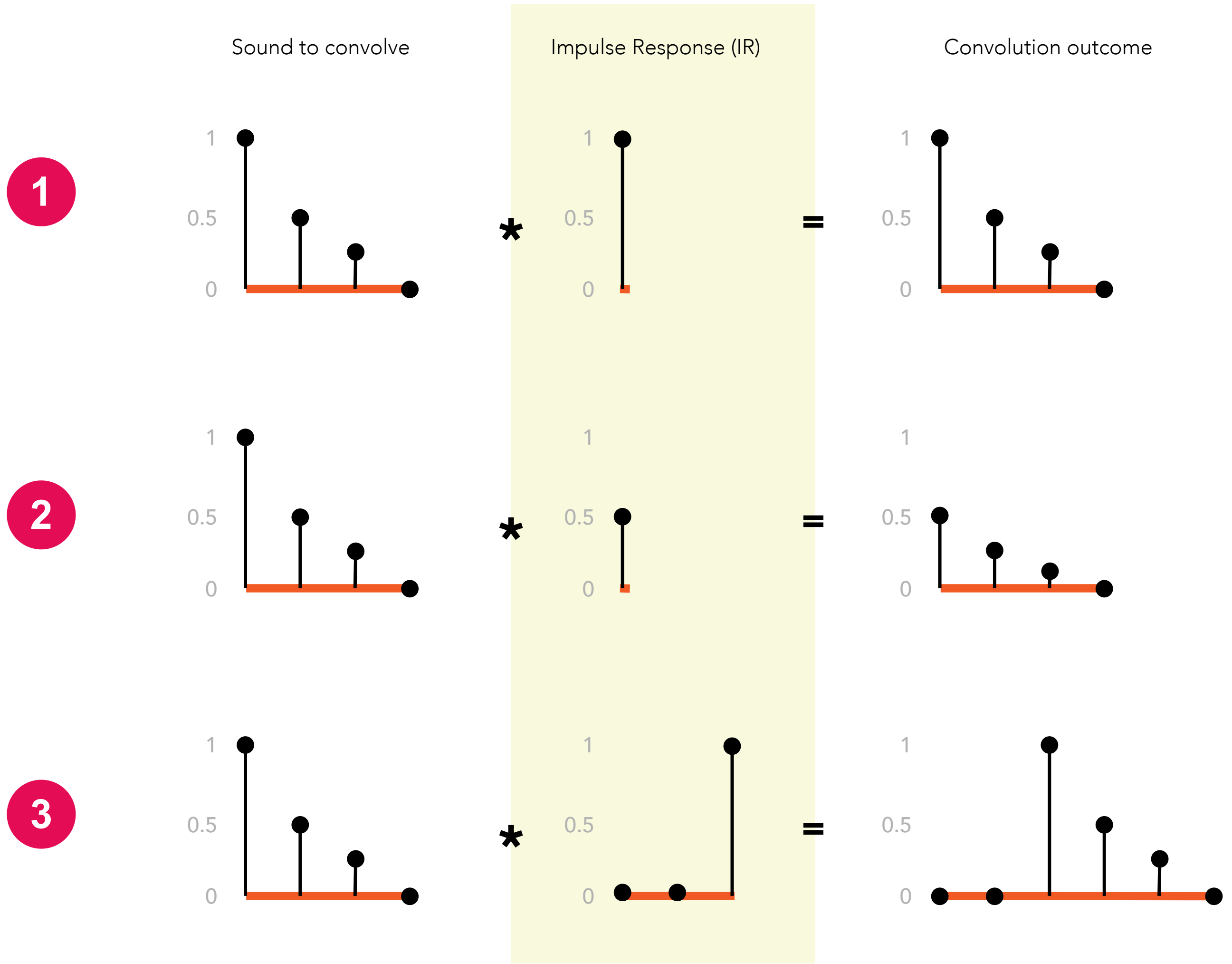
To perform convolution, we need the impulse response of the room that we want to simulate and the original musical sound that we want to apply reverb to.

If **h** is the room impulse response and **x** is the dry musical sound, I need to:

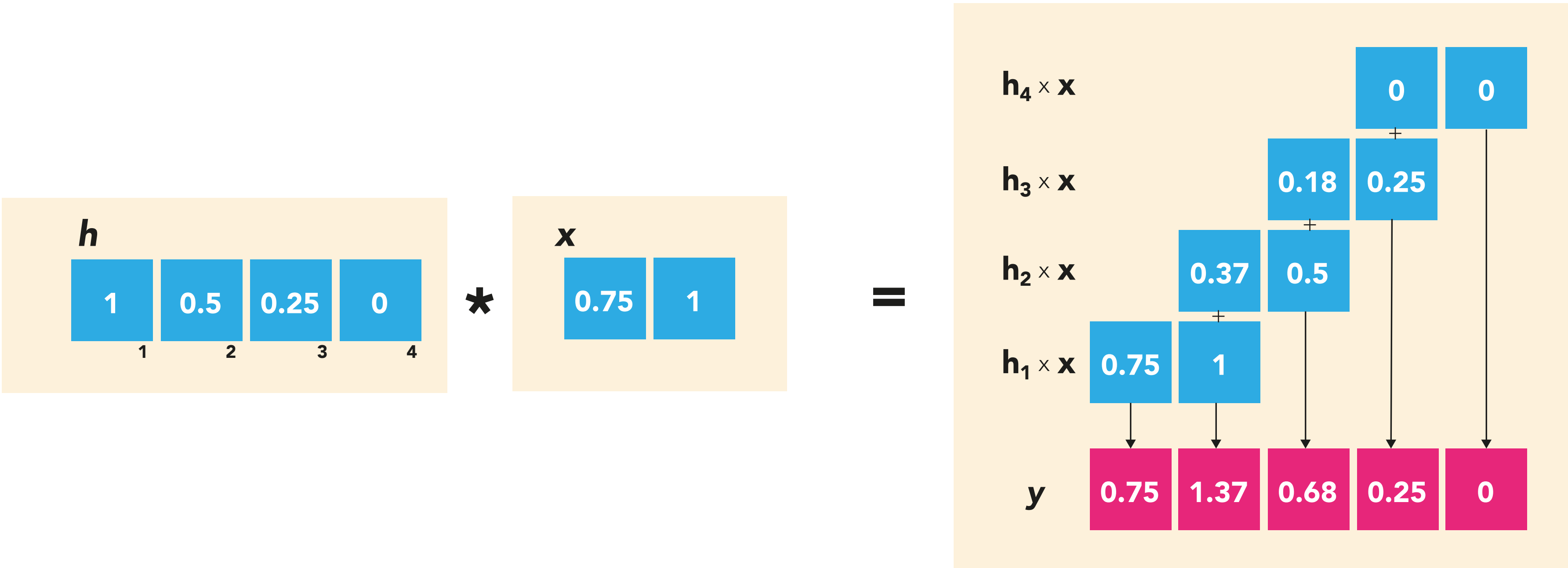
- **Multiply** h_n by x_n
- **Place** each resultant in a table from the bottom up, offsetting each new row to the right by one column
- **Add** each column up vertically to get the output **y**

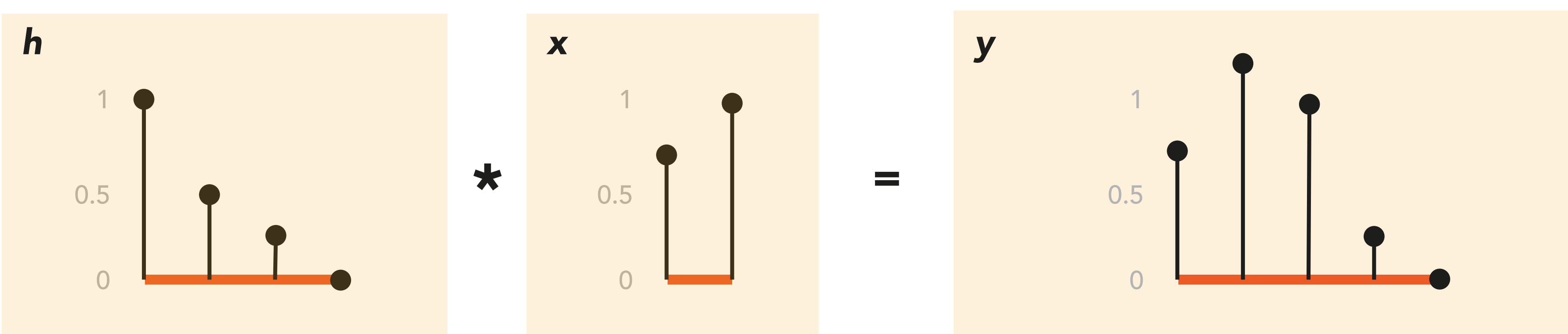


Let's take 3 simple cases to illustrate this mathematical operation better:

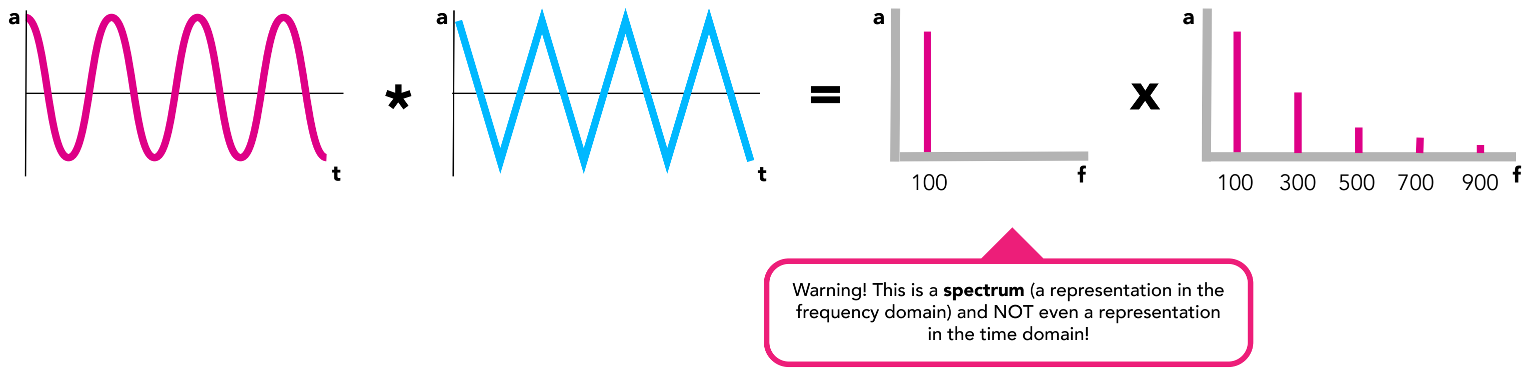


As we can tell from the graphical result of a more complex case, our final sound will be a sound with characteristics of the first and second sounds.





The direct convolution in the time domain of two sound signals is equivalent to the multiplication of their relative spectra!



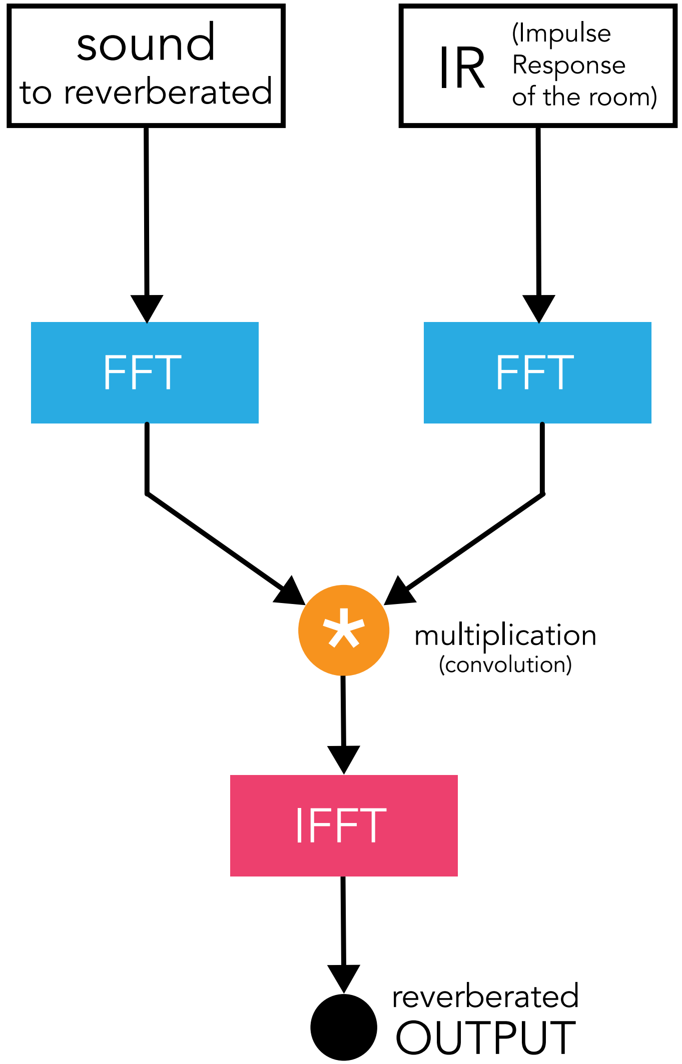
Why is this very important?

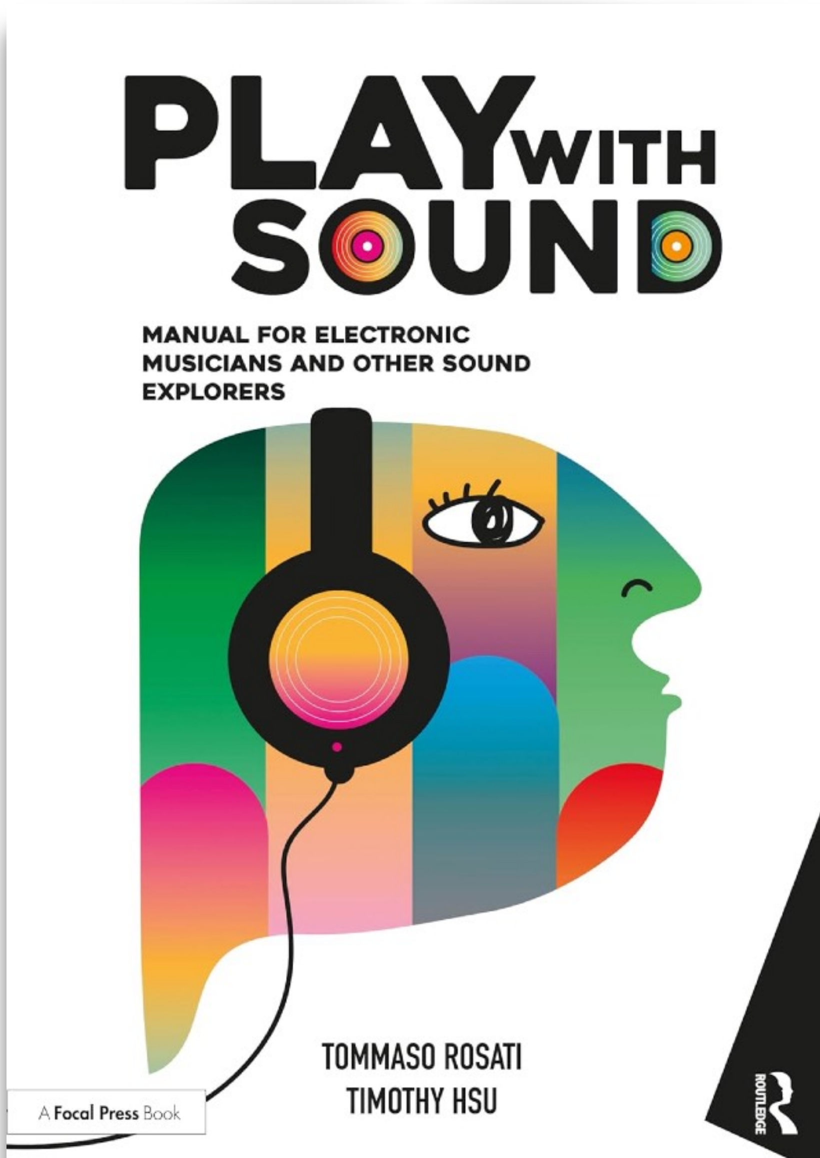
Because doing convolution between two sounds in the time domain (Direct Convolution) is very tiring by hand and computationally expensive.

Doing a convolution with the spectrum is much more agile and reduces the time and load on our processor. This process is called Fast Convolution.

Fast convolution is the multiplication of the two spectra. To get the spectra of our sounds, we do an operation called the Fourier Transform. The Fourier Transform converts time domain information into frequency domain information. We can also use an optimized discrete version of it, the **Fast Fourier Transform (FFT)**.

After multiplying the spectra, we need to convert the audio signal back to the time domain for playback. To do this, we perform an Inverse Fourier Transform (IFT) or **Inverse Fast Fourier Transform (IFFT)**.





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