**INSTITUTO TECNOLÓGICO Y DE ESTUDIOS SUPERIORES DE OCCIDENTE**

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**MAESTRÍA EN CIENCIA DE DATOS**

**R1:**

**Problem description and contextualization**

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In any enclosed environment or space, as a room, when we listen sounds, they got altered by the reproduction system and the space characteristics like its dimensions and materials

In order to compensate this, there is a specific area which is called digital room equalization, which consists of measuring the room’s response in both: frequency and phase using a specialized measuring microphone (typically omnidirectional) and applying an equalization phase to the loudspeakers in order to compensate measured errors.

There is to note that this prior procedure applies equalization to one specific point of the room, but for the rest of the points in the room this can be counterproductive and make things worse in a subjective manner.

Also, it is important to keep in mind that not all the defects or errors can be cleaned or equalized, so the impossible shall not be expected even while using the best technology available.

Talking about what can be equalized, graphic and parametric equalizers have been previously used with not much gain and lots of uncertain results, this caused mainly because the presence of time domain problems with early reflections and reverberation.

Graphic equalizers are suitable for frequency correction but not for time domain problems.

The basic room equalization shall compromise minimum phase type, but aiming for the best results there are more exceptions to take into consideration.

**Pre-responses and Minimum Phase Equalization**

The concept or principle of causality states that an effect can not be seen before its cause, which applies to real-world filters whose output is expected after its inputs, but for discrete time and digital domain acausal filters can be generated. The causal filter is minimum phase if its inverse filter also implies causality, but even though filters in the majority of graphic and parametric equalizers are minimum phase, the impulse response measured is almost never minimum phase which traduces into acausal room equalizers whose input arrives after its pre-response.

Room response equalization shall be minimum phase in order to avoid problems with pre-responses in ambients and rooms that are not correctly acoustically treated, which is the case of the vast majority. There are a lot of ways to ensure a REF is minimum phase

**Hilbert Transform**

Overall frequency and phase response of a filter are computed by the Hilbert transform knowing the amplitude frequency response. Knowing.

Spectral power response at several listening points can also be measured, and then used for equalize the average of all these points instead of equalizing for just one single-point.

**Room Measurement**

The loudspeakers are feed with an impulse and then the impulse response is measured the microphone, then the room’s response is derived from the impulse response by mathematical means. But for this, a problem arises: power in an impulse that can be fed to the speakers is limited, so any noise in the room will degrade the measurement accuracy. This is the reason that test signals with much higher power than an impulse are used, as a chirp test signal. Then by deconvolution, the impulse response can be derived from the output of the measuring microphone after the chirp tone was fed and played by the speaker.

**System’s Impulse Response**

A system can be modeled by its impulse response (**IR**) and its output can be determined by the convolution of its impulse response and input signal :

**Complex plane**

In the complex plane its transfer function is given by the Laplace transform of its impulse response:

In the complex plane the impulse response is given by the product of the transfer function and the **Laplace transform** of the input signal:

**Frequency Response of a System**

The frequency response of a system is given by the Fourier Transform of the impulse response:

Where is the angular frequency.

In the frequency domain the convolution is given by the product of the system’s frequency response and the **Fourier transform** of the input signal:

**Unit Impulse Function**

This function is also known as Dirac delta ranges in real numbers

**Inverse Transfer Function**

The aim of this work is to find , which is the inverse of the room’s transfer function

Then it can be applied to the audio signal in order to get a “plain frequency” output signal which is attenuated on certain frequencies prior of its reproduction on the loudspeakers.

So one of the system’s impulse response, transfer function or frequency response shall be obtained and thus the others can be calculated mathematically.

**For the digital domain or discrete time:**

Using the Z-transform the discret-time signal is converted to the frequency domain, which is the analog of the Laplace transformation for discret-time.

System’s behaviour in complex plain domain, its transfer function is given also by the Z transform:

In the complex plain domain the convolution is given by the product of the system’s transfer function and the **Z- transform** of the input signal:

Finally, the true aim of this work is to find , which is the inverse of the room’s transfer function

**Bibliografía**

[1] Gerzon, M. (1991). Digital room equalization.