0.1 To SKO

Hello Søren

It is to our uttermost delight to present this very first worksheet of the season. If it isn't too troubling, could you be a good sport and read the written material so far? We would especially like to hear your opinion about the subjects in the theory. Have we missed any important subjects?

Cherio

Contents

Introduktion 1

1.1 Some Bullshit about sound systems in real life

Hi-Fi audio equipment like amplifiers and loudspeakers, are very popular pieces of household electronics. People are seemingly willing to spend a lot of money on sound systems that is able to accurately reproduce sound. Amongst different types of sound-reproducing devices, the transducers are generally the worst at accurately reproducing the actual sound, especially for loudspeakers[6]. Purely electronic devices, like amplifiers, are generally easier to control.

A key feature of high quality sound systems is their frequency response. Frequency Response is a term describing the variations in amplitude and phase from the original signal as a function of frequency. It is generally desired to have a "flat" frequency response (i.e. the amplitude- and phase response of the output signal behaves the same way regardless of the frequency of the input signal).

This is hard to achieve in real life loudspeakers because of resonances in the physical part of the loudspeaker, which will result in certain frequencies being played louder than others. A given loudspeaker driver will have a set of frequencies

something about the relationship between speaker size and frequency respons.

1.2 Acoustic properties of a room

Even with a perfectly flat frequency response in a sound system, the sound that is actually being perceived by the listener might still be different than the original audio because of the acoustics of the surrounding room.

Room acoustics is a whole subject onto itself, but while this section will not go depth about every physical phenomenon that can influence the sound quality of a room, it will try to briefly cover how the surrounding room can influence the perceived sound quality for a listener.

When a sound wave hits an object in its path, a combination of three things happen. Some of the wave will be reflected off of the surface of the object. The rest of the wave will travel through the object, where some of the waves energy will be absorbed by the material and transformed into heat. Whatever is left of the wave will be transmitted through the object and continue along its direction of propagation. An illustration of this can be seen on figure 1.1
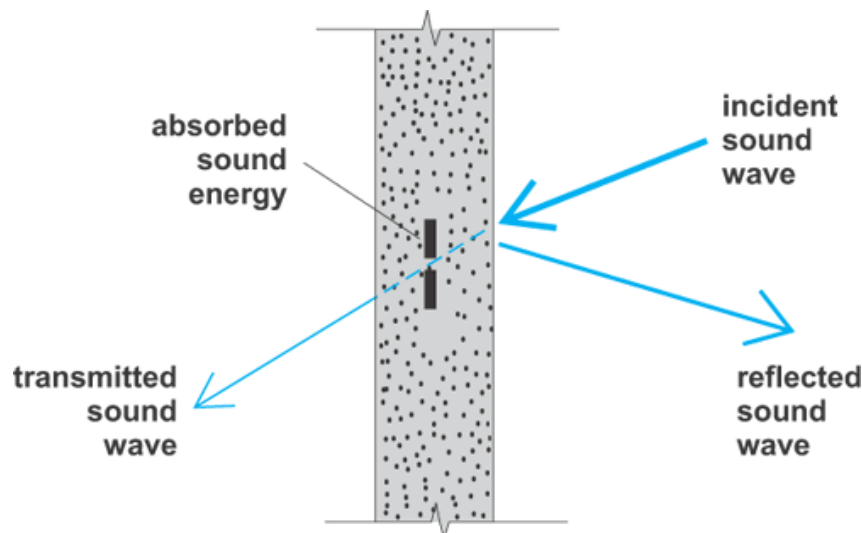
Figure 1.1: sound interacting with a wall. Illustration from [4] should be remade with vector graphics

How much of the sound wave is reflected, absorbed or transmitted through, is dependent on both the material of the object, the angle of incidence, and the frequency of the sound wave. rule of thumb for different building materials If the object being struck by the sound wave is significantly larger in area than the incoming wave, then the reflected sound wave will be reflected at an angle equal to the angle of the incoming wave. For example, a room with large and bare concrete walls will reflect specific frequencies a lot. If the waves hits a surface that is comparable in size to its wavelength, the waves will diffuse and scatter somewhat evenly across the room.

Acoustic treatment of rooms is often done by fitting rooms with object that will diffuse incoming sound ways, a covering surfaces with sound absorbent materials. This is done to reduce reflected waves in a room.

One problem associated with reflections is the emergence of interference patterns between a sound wave in a direct path to the listener the sound wave reflected off a surface. Since the reflected wave has traveled a longer distance than the direct wave, the to waveforms will likely be out of phase with each other and interference will occur. This can cause the resulting wave to either have a larger or smaller amplitude than the original wave. An example of how this can occur can be seen in figure 1.2
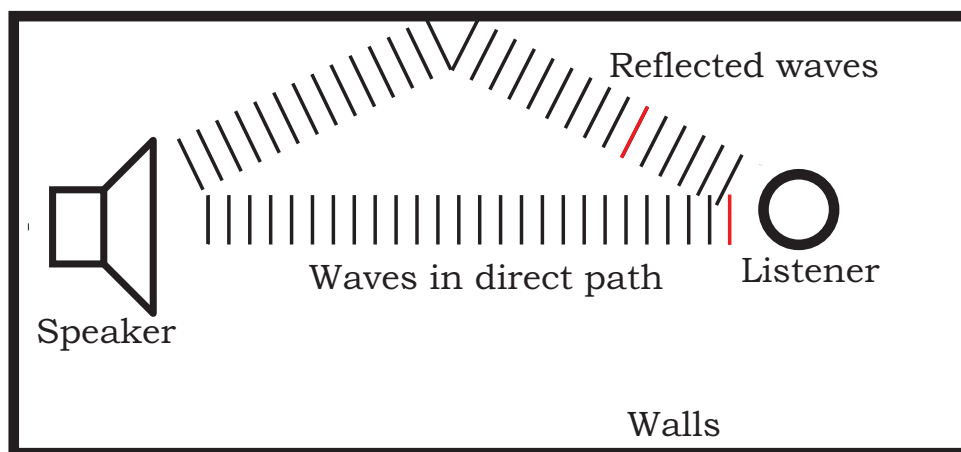

Figure 1.2: Sound traveling towards a listener along different paths[1]

Since a given material absorbs and reflects sound waves differently at different frequencies, and since reflected waves are attenuated a a rate given by its frequency arcording to Stoke's Law of sound attenuation(elaboration?), different frequencies will be affected by interference at the listeners position differently for different frequencies. These phenomena can have a quite significant effect since a wave with a frequency of 50 Hz can behave very differently from one with a frequency of 5000 Hz. And since music in general feature a pretty wide spectrum of frequencies [2], the frequency response of a given piece of music can change a lot by traveling through a room. Furthermore this behavior can vary a lot based on the position of the speaker, the size of the room, the building materials, and the interior decoration of the room, in ways that might not be obvious to the average listener, who just wants to sit at his/her favorite listening spot.

a bit about reverb maybe?

1.3   Initial Problem statement [EARLY DRAFT]

summation of the previous section

Frequencies are often altered by

*which elements go into a se

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[1]The lines representing the sound waves can be seen as tangents to the crests of the wave
[2]20 Hz to 20 kHz

Theory 2

This chapter will contain the analysis of the theories that make the project possible. And maybe some dank memes.

2.1 Equalizer

i've written some questions in red, that you could maybe write about.

why is it relevant to look into equalizers

There are digital and analog equalizers, but in this section the focus will be on the analog equalizers.

In the field of the audio, an equalizer is a device that modifies the response in frequency of the signal, and adapts it to the user. These devices are able to strength or cut the energy of one or many frequency bands.

why is this desired?

The analog filter design is mainly limited by the cost of the different elements, so a selection of better components will lead to a better equalizer.

It is well known that most of the signals contain noise, but analog filters have the disadvantage that they add noise to the signal through heat and electrostatic noise, for example.

http://www.xowave.com/doc/effect/eq/analogvsdigital.shtml

https://ledgernote.com/columns/gear-reviews/best-studio-equalizers/

2.2 Digital sound

Mike to digital

Sampling frequency

To ensure that the audio signal can be recreated as an AC-signal, the sampling frequency has to be taken into account. The sampling frequency determents the number of samples, there has to be played pr second. According to the Nyquist sampling theorem[9] the sampling frequency should be above two times the highest frequency of the signal. If this is not the case, some frequencies could be represented as a different frequency, as seen in figure 2.1.
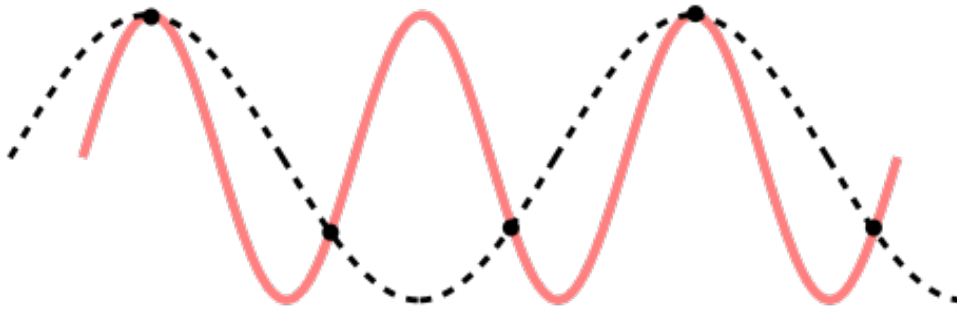
Figure 2.1: Falsely represented signal, due to the sampling frequency being to low. Figure from source [8]

Since the Nyquist sampling theorem states that the sampling frequency should be at least twice the highest frequency of the signal, the sampling frequency should be at least 40 kHz, when sampling audio signals. According to the standard IEC-60908[2] the sampling frequency of sound should be 44,1 kHz. This way audio signals can be recreated up to 22,5 kHz.

File resolution

When storing files the bit debt is a big factor in the quality of the sound. Each sample of the audio file have to be defined by a number of bits. This means that the amount of bits used per sample determents how well the sound will be when converted to an analog signal. For a 4 bit signal, there would therefore be 16 different values for a sample, and for a 16 bit signal there would be 65536. An illustration of the correlation between an analog signal and its representation in bits, can be seen in figure 2.2.
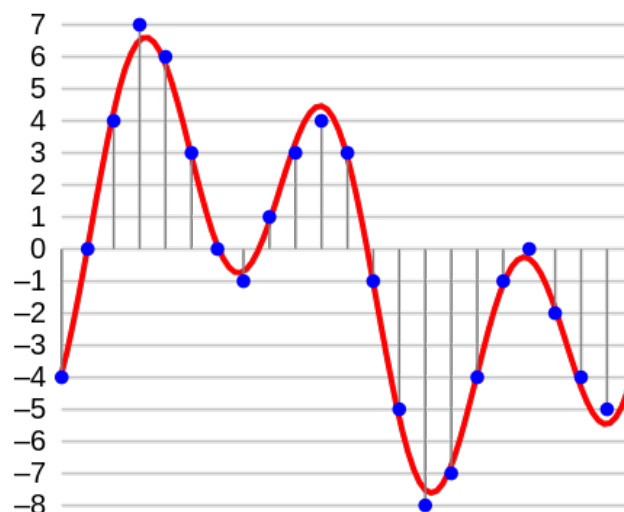

Figure 2.2: Audio signal and its representation in bits. Figure from source [7]

CD quality sound is usually stored in a 16 bit resolution. In sound studios they use a bit debt of 24 or above to ensure a proper D/A conversion. If there is a need for adding two 16 bit signals together, 17 bits are needed to avoid clipping.

When audio is converted from analog to digital, there can occur an error in the sampling process. This happens because the digital value might not be exactly the same at the analog value. This error is referred to as quantization. The quantization error can be $\pm$ 0,5 times the value of the

least significant bit(LSB). The signal-to-quantization-noise ratio (SQNR) can be calculated as seen in equation 2.2.1.

$$\text{SQNR} = 20 \cdot \log_{10}(2^Q) \quad [\text{dB}] \qquad (2.2.1)$$

where:

Q　　= Signal bit-dept　　　[·]

Using the equation, a 1 bit signal would have an SQNR of 6,02 dB. For every bit added to the bit dept the SQNR goes up another 6,02 dB.

2.3　Digital signal processing

Digital signal processing and analog signal processing both is a description of a process to modulate an input. Digital signal processing utilizes digital filters to achieve this modulation, this makes digital signal processing much more versatile than analog signal processing because digital filters can both emulate analog filters, and do a lot more.

Digital filters

A digital filter takes the input sequence of numbers and computes a new sequence of numbers based on its transfer function. Only basic operations such as: addition, subtraction, multiplication and division can be done, more complex functions must be a combination of these.

with digital filters the accuracy of the calculations of the functions can be control. The more complex a function and the closer the function needs to be to the ideal, the longer it takes to computate. High computation times limits digital filters usability for real time applications. This means for real time applications either the filters complexity and accuracy needs to be handled, or a fast digital signal processor is needed.

Digital filters are more flexible than analog, and can be changed due to the fact they are programmable

2.4　The amazing world of sound cards

What is a sound card In order for a computer to play and record audio, it needs a device that can convert analog sound to digital, and vise versa. A sound card also allows the user to perform some degree of digital signal manipulation, where the most common would be to manipulate the audio with an equalizer.

Interface A sound card has a lot of interfaces, depending on the type of sound card. One of these could be a digital input or output from a CD-ROM drive or a MIDI(Musical Instrument Digital Interface). Depending on the type of sound card and the price range, the interfaces of the sound card differs, but the most common is analog input to a microphone or analog outputs to loudspeakers.

Internal components Since a sound card performs its operations digitally, all analog signal signal needs to be converted to digital in order to make recordings, and digital signal must be converted to analog for audio playback. This is done with an ADC(Analog to Digital Converter) and a

DAC(Digital to Analog Converter). The resolution of these components depends on the price range, but a resolution of 16 bits is needed for audio in CD quality.[5][10]

For the digital audio processing there are mainly two options. Either use to computers CPU or to use a Digital Signal Processor(DSP).

Drivers In order to control and issue commands to the sound card, some form of device driver is needed. The driver handles the data connection between the sound card and an operating system. The driver needed depends mostly on the users OS. For Windows users the drivers will generally be written by the manufactures and will therefore be licensed. Linux users on the other hand, have a bit more freedom. The most wildly uses driver is called ALSA(Advanced Linux Sound Architecture). One of the bigger advantages with the ALSA-driver compared to most Windows drivers, is that it allows the user to directly interact with the kernel and therefore the hardware.[1]

2.5 Platforms for digital signal processing

In order to correct the soundlevels of the loudspeaker compared to the measured sound, a platform that can process and compare the audio is needed.

For digital signal processing a digital signal processor (DSP) would be ideal, but a regular general purpose central processing unit (CPU) is today almost as fast as a specialized signal processor, for signal processing.

Microcontroller

A microcontroller is a circuit created for and used to control embedded systems. A microcontroller usually contains a CPU and some random access memory (RAM), besides that it has input output ports, so that it can be used in a embedded system [3].

Computer

2.6 Wireless communication

Commonly used wireless systems

Because there is lot of different types of communication systems, some which is not relevant for the scope of this project, we will try to list some of the more commonly used systems, with an actual chance of being useful in the project.

Wireless Local area network Most people with wireless access to the internet are using either cellular or Wireless Local area network (WLAN for short). Cellular communication will discuss later. The most common implementation of WLAN is through WIFI, which is based around the IEEE 802.11 standard.

- Broadcasting (radio) return to this after antenna theory

loooooong antannas one way communication

- WIFI (IEEE 802.11)

advanced well documented

- Bluetooth and PAN in general (IEEE 802.15)

(maybe not as well documented as WIFI) not as advanced ideal for two or few devices low range up 250m for business grade, but more commonly 0.5-10m

- G3/G4/LTE

Glossary

Frequency response The way a system output behaves in terms of magnitude and phase, as a function of the frequency of the input. 1

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