0.1 To SKO

remind SKO to send C code for alsa

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? Are we on the right track?

Cherio

Contents

1 Introduction

1.1 Frequency Responses of audio systems

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[1] are generally the worst at accurately reproducing the actual sound, especially for loudspeakers[9]. Purely electronic devices, like amplifiers, are generally easier to control.

A key feature of high quality sound systems is their frequency response.The 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 loudspeakers because of resonances in the physical part of the loudspeaker, which will result in certain frequencies being played louder than others.

The Loudspeaker driver that converts the electric signal to sound waves only has a certain limited range of frequencies where it works optimally and loudspeakers made with only one driver (Full-range drivers) also usually have a hard time reproducing every frequency at a similar amplitude, especially when playing at high levels of power. A lot of loudspeaker systems are therefore made using multiple drivers, that each handles a certain band of frequencies.

There are different design and construction challenges for each type of driver depending on which part of the audio spectrum it should reproduce, but in general drivers for high frequencies are smaller than drivers for lower frequencies. Since drivers for reproducing the lowest part of the frequency spectrum (20 - 200 Hz) are usually quite big and use a lot of power. For this reason, Loudspeaker systems with limited size and power usage (like PC speakers) are usually physically incapable of reproducing these frequencies at the same amplitude as higher frequencies.

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

[1]Devices that convert one type energy into another.

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

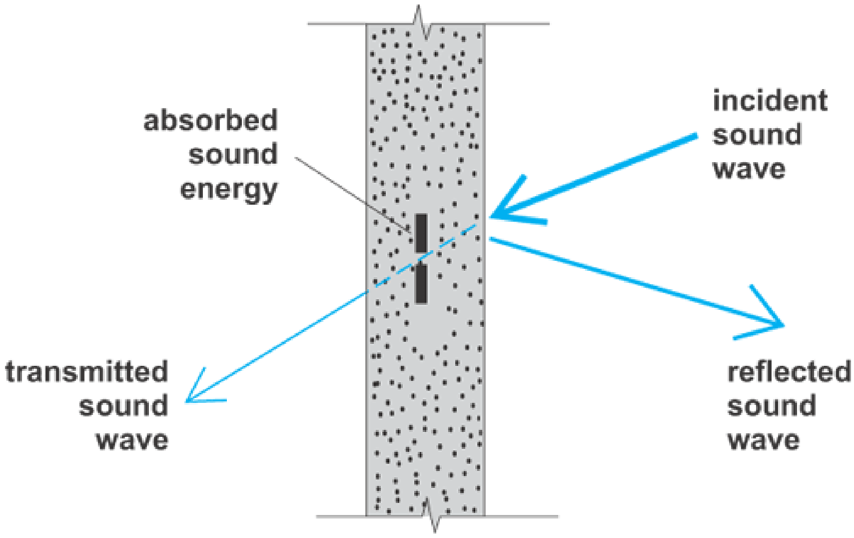

Figure 1.1: sound interacting with a wall. Illustration from [6] should be remade with vector graphics

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

Since a given material absorbs and reflects sound waves differently at different frequencies, and since reflected waves are attenuated a rate given by its frequency according to Stoke's Law of sound attenuation, 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

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

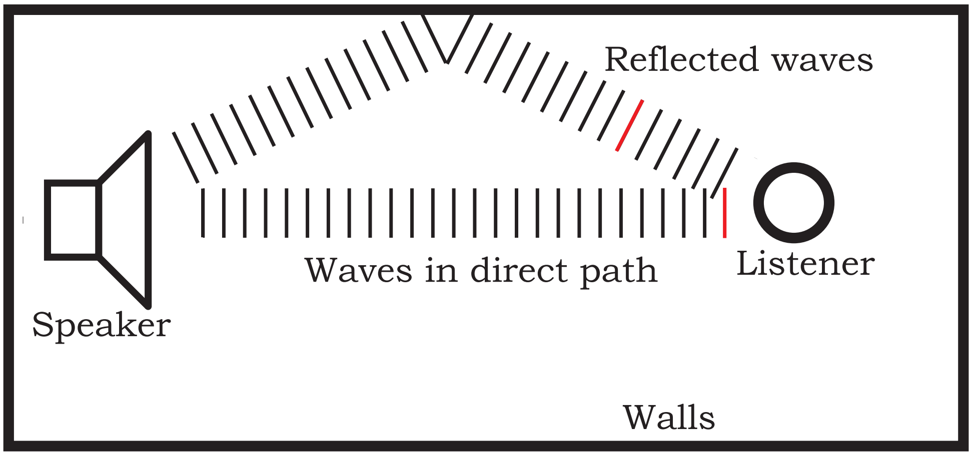

Figure 1.2: Sound traveling towards a listener along different paths[2]

Hz. And since music in general feature a pretty wide spectrum of frequencies [3], 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.

There is an important phenomenon that provides a general idea about the behaviour of a room and it is independent from the listening point, the reverberation. The reverberation is the greater or the less persistence of the sound when a sound source stops emitting, and it depends on the intensity of the sound source and the threshold of hearing.

1.3 Initial problem statement [EARLY DRAFT; please give response]

In the previous sections it has been shown that a piece of audio can often be altered during playback by different frequencies being represented differently. Furthermore, the processes which alters the characteristics of the sound might not be immediately obvious or even fixable for the user of the sound system. Since this is a 5th semester project in Electronics and IT, the project has been is meant to revolve around an electronic system that can interact with its surroundings by measuring some form of signal, processing this data, and generating a new signal based on this processing. For this reason It has been decided to work with a system that can measure the frequency response of a sound system, and correct the frequency response based on these measurements. It is therefore possible to formulate an initial problem statement based on this, as the following:

Which elements are needed to make a system that corrects the frequency response of a sound system?

Which technologies will need to described to specify a system that corrects the frequency response of a sound system?

[3] 20 Hz to 20 kHz

2 Theory

This chapter will contain the analysis of the theories that make the project possible.

2.1 Equalizer

Citation needed

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

In the field of audio, an equalizer is a device that modifies the response in frequency of the signal, and adapts it to the user, solving all kinds of problems. These devices are able to either amplify or attenuate one or more frequency bands.

There are different types of equalizers: High-pass and low-pass filters, shelving filters, graphic equalizers and parametric equalizers.

A high-pass filter is an electronic circuit that passes high frequencies and attenuates low ones. The same for the low-pass filters, that allows low frequencies to pass but attenuate the high ones.[4]

On the other hand, shelving filters (figure 2.1) are used as tone controls, as they can attenuate or amplify a signal above or below a certain frequency. [4]

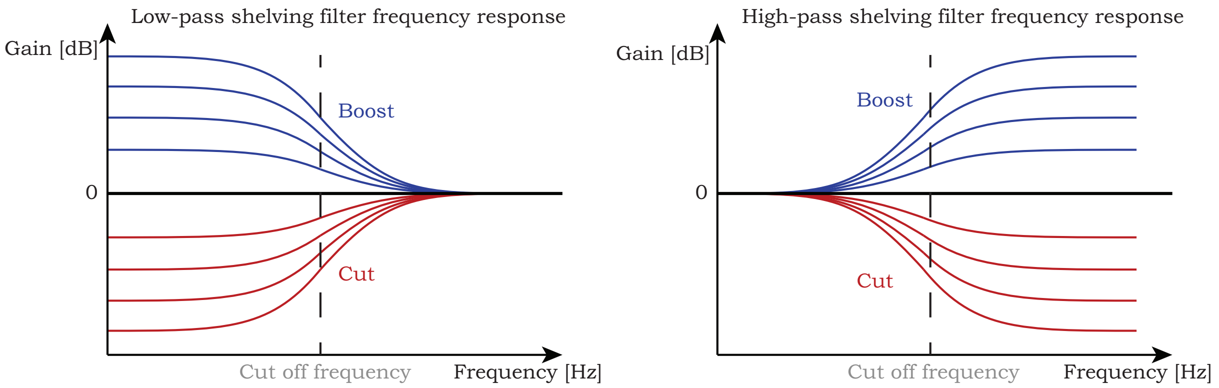

Figure 2.1: Shelving filter

[10]

A graphic equalizer (figure 2.2, however, is composed by some band-pass filters. It takes the name from the physical position that the faders are placed, making possible to see in a easy way the changes made. [4]

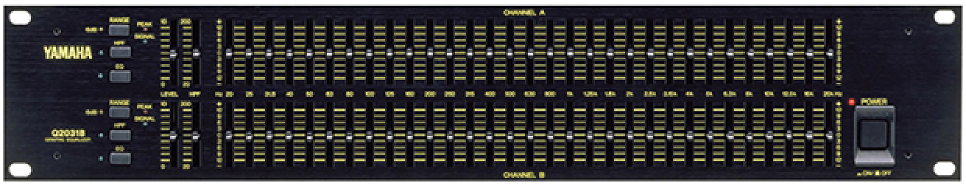

Figure 2.2: Graphic equalizer

[11]

Finally, the parametric equalizer (figure 2.3), allows the individual control of three main parameters of the filters that compose the equalizer: amplitude, center frequency and bandwidth. [4]

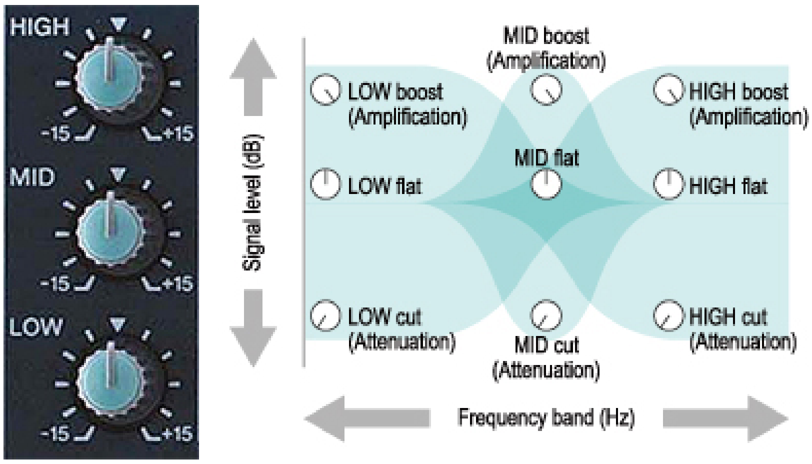

Figure 2.3: Parametric equalizer

[11]

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 for example heat and electrostatic noise.

2.2 Digital sound

When sound have to be transported over longer distances, or stored for later use, it might be necessary to convert it into a digital signal. This can be done by converting the sound to an analog signal trough a microphone and then converting it to a digital signal trough a analog to ADC (analog to digital converter). The digital signal can then be stored as a string of bits, which can be read by computers and for example be filtered ore shared via the internet. Furthermore it is possible to make calculations based on the digital signal such as equalization or addition of multiple signals.

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 determines the number of samples, there has to be played pr second. According to the Nyquist sampling theorem[15] 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 2.4.

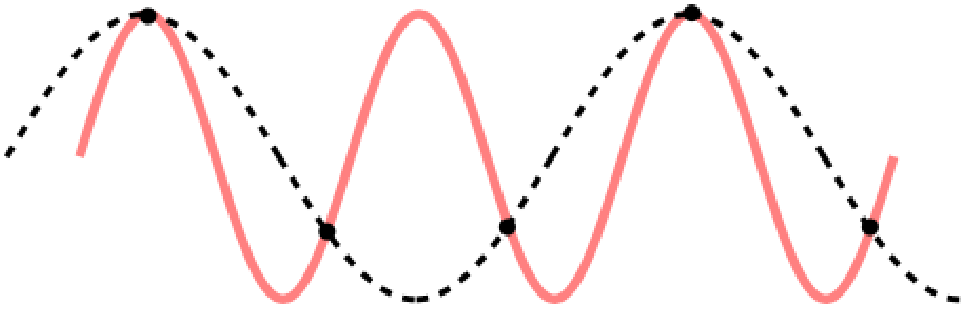

Figure 2.4: Falsely represented signal, due to the sampling frequency being to low. Figure from source [14]

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[3] the sampling frequency of sound should be 44,1 kHz. This way audio signals can be recreated up to 22,05 kHz.

File resolution

When storing audio in a file it is defined by a string of bits that matches the amplitude of a specific sample. Here the amount of bits used to determine the sample, also known as bit debt, is a big factor in the quality of the sound. 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 2.5.

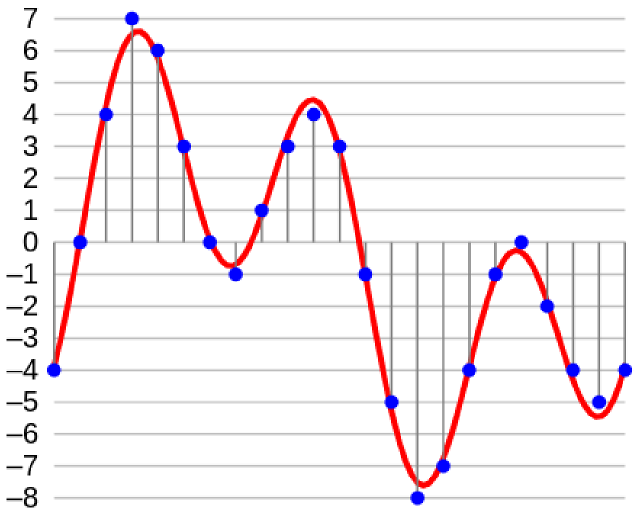

Figure 2.5: Audio signal and it is representation in bits. Figure from source [13]

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 as the analog value. This error is referred to as quantization. The quantization error can be ± 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.

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

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 because it is not limited to passive and active components, but to basic arithmetic, can therefore do a lot more[12]. Additionally there are no outside noise effecting a digital signal, which therefore does not have to be taken into account.

Digital filters

Finish this when lectures about digital filters are done.

Filters are used to selectively choose some wanted frequencies to be allowed through, and cut out other frequencies. Active filters can also be used to amplify or attenuate the specified frequencies.

A digital filter takes the input sequence of numbers from sampling the signal and computes a new sequence of numbers based on the filters transfer function. However digital filters cannot amplify a signal, only scale it up to the maximum bit value determined by the bit debt. Digital filters can only do basic operations such as: addition, subtraction, multiplication and division, more complex functions must be a combination of these [12] [8].

With digital filters the accuracy of the calculations of the functions can be controlled much more precisely than analog filters. The more complex a function and the closer the more accurate the function needs to be to the ideal, the longer it takes to compute. High computation times limits digital filters usability for real time applications. For real time applications either the filters complexity and accuracy needs to be lowered to decrease the computation time, or a fast digital signal processor is needed [8].

Due to the fact digital filters are calculated in a processor, they can be changed easily compared to analog filters, where the passive or active components would have to be changed.

2.4 The amazing world of sound cards

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.

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.

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.[7][16]

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

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 used 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.[2]

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 real time digital signal processing the highly specialized digital signal processors (DSP) was created, these had a lot less functionality than normal CPU's but were faster for processing digital signals. Today however CPU's have gotten so fast that they can be used for most real time digital 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[5].

A microcontroller will typically be set to run a single program indefinitely. This makes it possible to use the CPU[1] quite effectively. It also removes the overhead of trying to effectively schedule the CPU usage between different processes.

[1] which is often somewhat slow compared to CPU's used in PC's

A very widely used series of microcontrollers are the Arduinos. The most common Arduinos are equipped with an ARM CPU with a clock speed in the range of about 8-80 MHz.

Computer

Another option for a platform for digital signal processing, could be a fully fledged computer. Besides a CPU and RAM, a complete computer will often have some form of permanent memory, network capabilities, a dedicated graphical processing unit (GPU), and multiple ports for communication with peripherals. A complete computer is also easily implemented with a operating system, which makes it easier for the CPU to handle multiple simultaneous processes with the help of a kernel.

While you normally think of a desktop PC or a laptop, when you hear the term "computer", There has recently been an emergence of small single board computers, where you have a complete computer implemented on a single PCB. The most well known example of a single board computer is the Rasberry Pi.

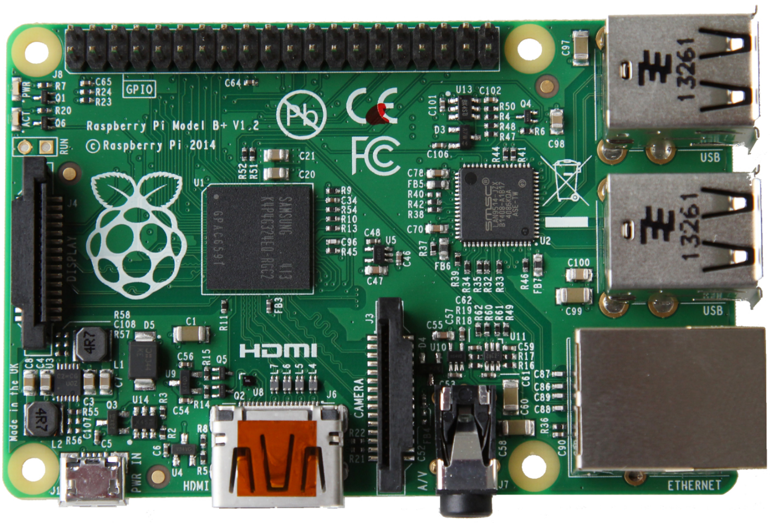

Figure 2.6: Rasberry Pi model B+. Image from [1]

There are multiple versions of the Raspberry Pi, each version with slightly different hardware. Each type has an ARM processor with a clock speed of at least 700 MHz which is several times faster than even the fastest Arduino. This fact might help with negating the extra overhead associated with running a program in an operating system instead of an embedded system. Another benefit of using a computer, is that it might be possible to use already existing drivers to communicate with a sound card.

Glossary

Equalizer An electronic device for adjusting the frequency response of a system.. 4

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

Rasberry Pi Series of single board computers, often used for small projects.. 9

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