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EEE321-02

SIGNALS AND SYSTEMS LAB 3 REPORT

Theory of the Experiment:

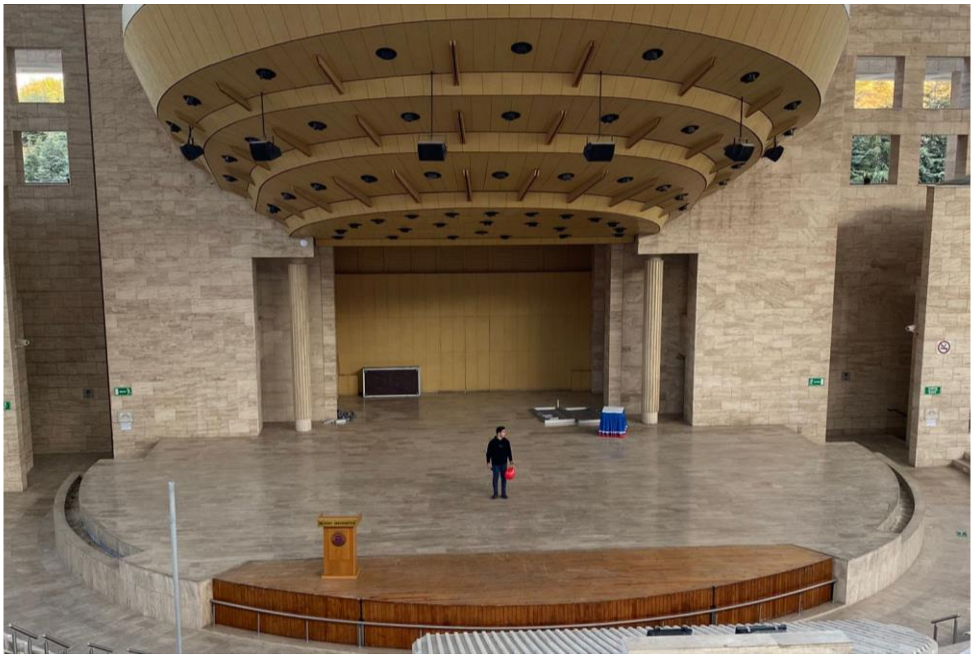
In this experiment, the main objective is to find the output of an LTI system by using its impulse response. Since MATLAB and the device used for sound recording record the continuous sound signal as a discrete signal by sampling, the convolution operation will be written in the discrete form. For LTI systems, the discrete impulse response can be found with the following equation.

$$y[n] = h[n] * x[n] = \sum_{k=-\infty}^{\infty} h[k] \cdot x[n-k]$$

For this experiment, the sound recording of the balloon blown up with a needle represents the desired seat's response to the impulse generated at the stage of the Odeon. The anechoic music signal represents the input to the system, and the music heard at the Odeon's given seat represents the system's output to the given input. It can be summarized with the following:

1. The system's input (x[n]) = An anechoic clear music recording.
2. Impulse Response of the LTI System (h[n]) = Sound recording of the blown-up balloon from the seat.
3. The output of the system to the Given Input (y[n]) = Music supposed to be heard from the seat when the music is played on the stage.

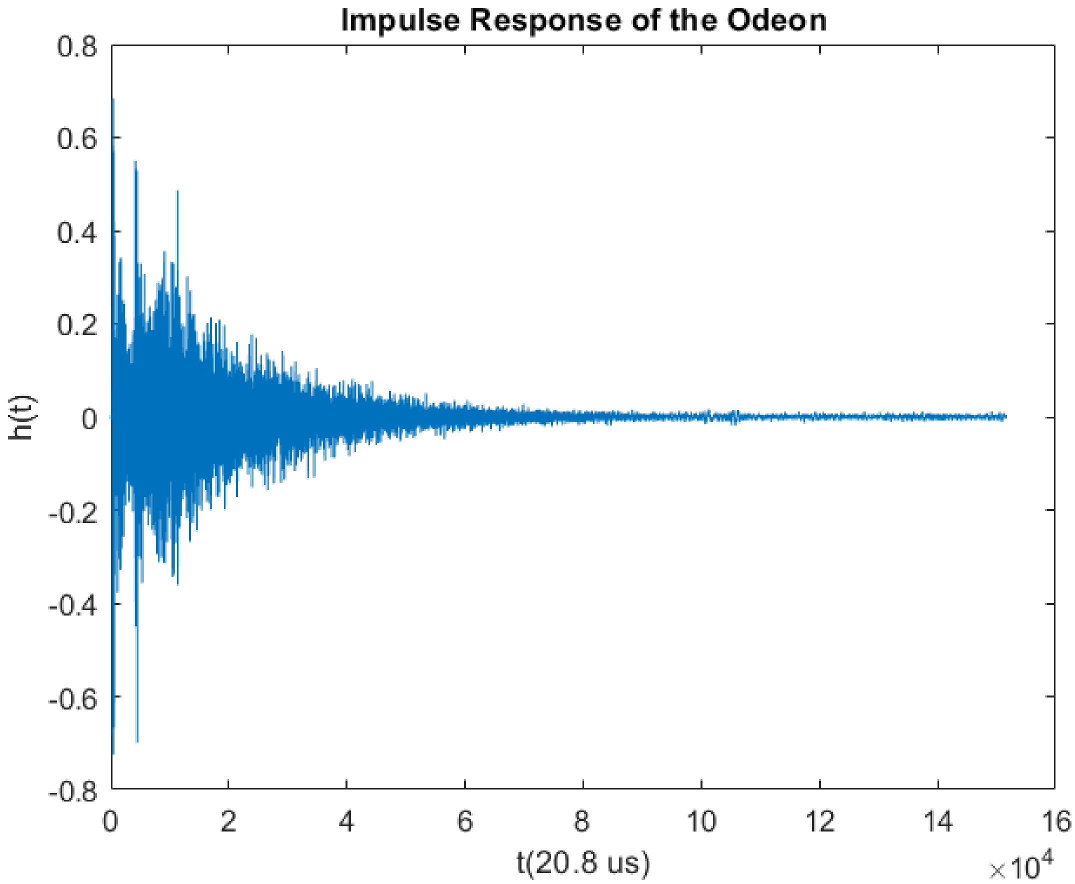
After understanding these concepts, I blew up a balloon at the center of the Odeon's stage (Fig.1), and my friend Gökay recorded the sound from the seat we had chosen (Fig.2). This sound recording represents the LTI systems response to the impulse signal.



(Fig.1: The center located at the stage where the balloon was blown up with a needle)

(Fig.2: Location of my friend, Gökay, at a seat that we have chosen to record the impulse)

After the sound was recorded at the Odeon, the rest of the work was done using MATLAB. The input sound is an anechoic file from the 2nd violin's recording from Beethoven's 7th symphony [1]. The sound of the blown-up balloon, which represents the system's impulse response, and the anechoic sound, which represents the input signal, were read using MATLAB and converted into discrete time arrays with a predefined sampling rate equal to 48kHz, which means 1 sample every 20.8 microseconds. The plot of a segment of the input (Fig.3) and the plot of the system's impulse response (Fig.4) can be analyzed to understand the properties of both signals.

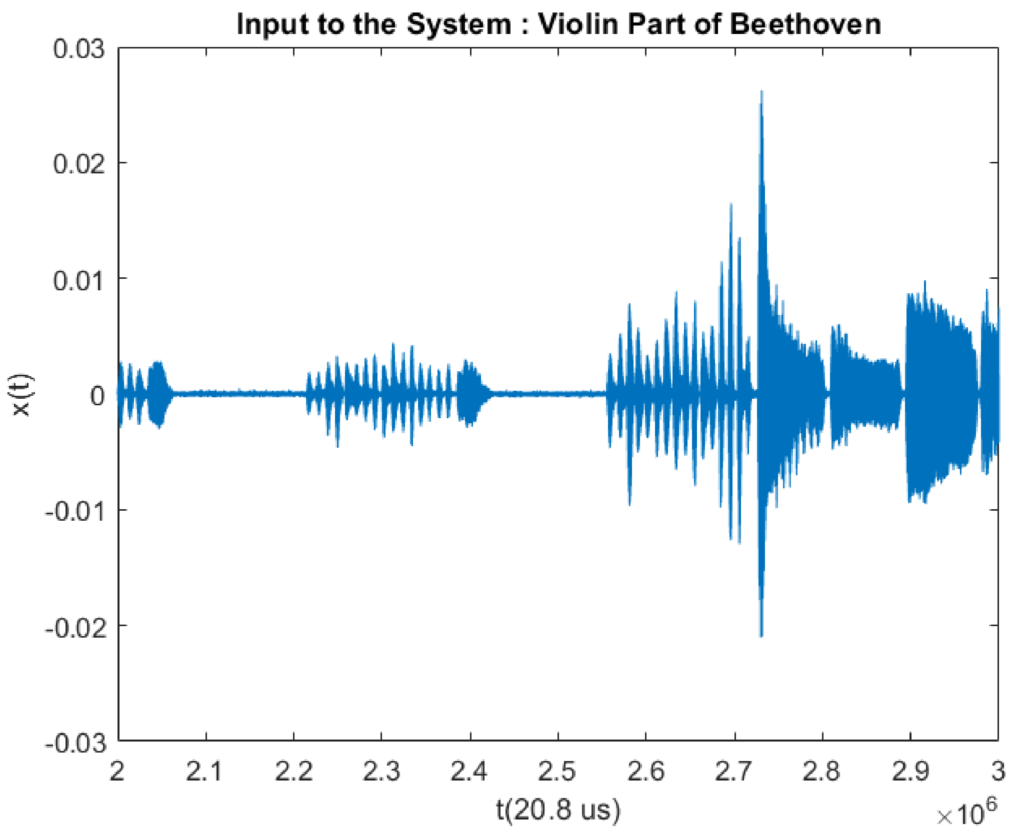


(Fig.3: Sound recording of the balloon blown in the Odeon)

The sound recording of the balloon has been sampled starting from the point where the balloon actually blows to avoid any mistake related to the system's causality since the system's impulse response should satisfy the following inequality to be a causal system, which is a necessity for a physical system.

$$h[n] < 0 \; if \; n < 0$$

Another method that can be applied is that a threshold signal value can be set and any value smaller than that can be written as zero. This method is called simple thresholding. But for the sake of simplicity and also to have an unshifted output, the signal has been cropped manually.



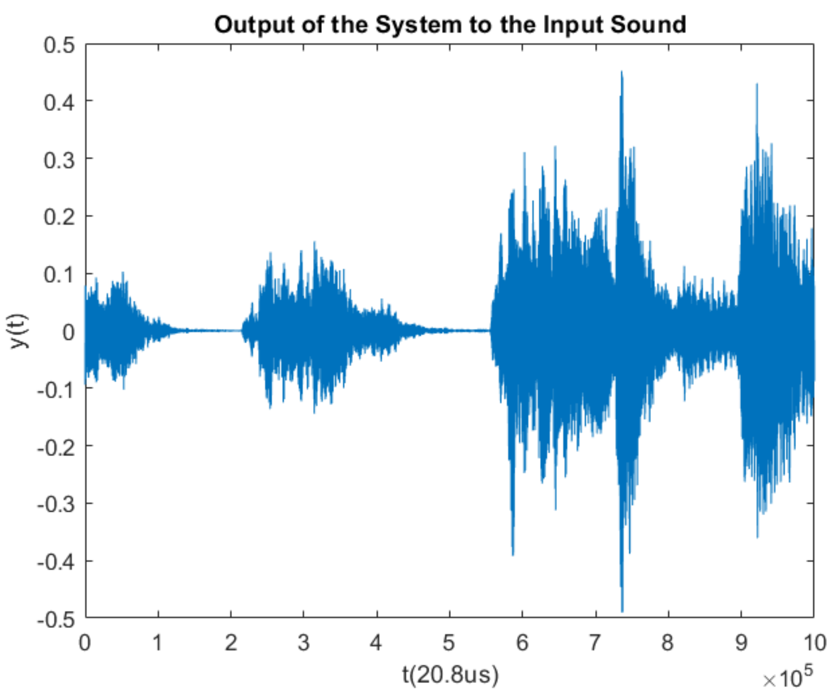
(Fig.4: Plot of the 20.83-second-long segment of the input signal)

After converting both sound files to discrete signal arrays, convolution was done on the data with the following code (Fig.5).

```
[h,fsh] = audioread("C:\Users\efeta\OneDrive\Desktop\impresp.m4a");
[violin,fviolin] = audioread("C:\Users\efeta\OneDrive\Desktop\beethoven_mp3\beethoven_vl2a_6.mp3");
violin2 = violin*1.5;
h = h(149000:size(h,1));

%out = conv(violin,h);
%save h.mat h ;
%save out.mat out ;
%save violin.mat violin2 ;
%audiowrite('output_violin.m4a',out,fsh);
%audiowrite('input_violin.m4a',violin2,fsh);
%audiowrite('input_response.m4a',h,fsh);
load out.mat out
load h.mat h
load violin.mat violin2


%plot(h);
%title('Impulse Response of the Odeon');
%xlabel('t(20.8 us)');
%ylabel('h(t)');


plot([2000000:2999999],violin2([2000000:2999999]));
title('Input to the System : Violin Part of Beethoven');
xlabel('t(20.8 us)');
ylabel('x(t)');


%plot([1:1000000],out([2000000:2999999]));
%title('Output of the System to the Input Sound');
%xlabel('t(20.8us)');
%ylabel('y(t)');
```

(Fig.5: Code for the convolution and creating the plots of the signals)

After convolving two signals, the resultant signal, the LTI system's output for the input signal, is obtained and converted to a song file to listen to.



(Fig.6: Plot of the 20.83-second-long segment of the output signal)

With these results, the experiment is completed. There are three different sound files, including the input song, impulse response which is the sound of the blown balloon, and the resultant output signal of the convolution operation.

Comments on the Results of the Experiment:

i) Quality of the Impulse

The quality of the impulse signal highly depends on the action of blowing up the balloon. Because impulse must be concentrated at the t = 0 point, the physical dynamics of the blowing up effect must also occur rapidly and be concentrated at t = 0. Also, the impulse response can not be obtained by additional mechanical properties because musical instruments create the sound by vibrating the air particles. Therefore, dropping an object to the ground is not a good way to create that impulse sound for several reasons. First, the object will definitely jump a small or a large amount from the ground and create several collisions with the ground until equilibrium. By that, it creates several impulse-like signals with decreasing amplitudes, reducing the impulse's quality. Second, it is not a good impulse because it creates the impulse by smashing it to the ground, which vibrates the solid wood particles first and then the air. But since most musical instruments are being played by directly vibrating the air rather than the floor of the stage, the initial vibration medium is better to be air.

In conclusion, there are several important properties of the physical modeling of an impulse for an experiment like this that blowing a balloon satisfies. Firstly, the impulse must occur in a concentrated time interval around 0, just like the defined impulse signal. Secondly, there should not be following signals created by the source of the impulse; only the effects of this impulse are acceptable. And lastly, the medium of the initial signal must be the same as the medium of the instruments, air. The balloon explosion sound is created by the elasticity of the material and the high air pressure inside the balloon, so it satisfies the medium condition.

ii) Nature About the Impulse Response

The system's impulse response in this experiment is the sound signals that come from the stage with the effect of exploding a balloon with a needle. So, assuming that the balloon explosion is a good representation of the impulse, all the echoes that the acoustic structure of the Odeon creates at the specified seat are considered the system's response to an impulse on the stage. Odeon's impulse response includes every detail that the materials around create. The amount of openings at the ceiling of the Odeon, materials and shapes of the floor, seats, stage and architectural structure of the Odeon, air quality, weather conditions, heat, and many other factors are a part of the features that affect the impulse response of the system. Some of these factors have a larger effect, like the architectural structure of the building, and some of the features have less impact, such as the existence of a seat at the far back corner of the Odeon. But suppose the experiment's goal would be to filter the anechoic sound with the impulse response with the recorded impulse when Odeon is empty. In that case, it somehow leads to a large error since, during a concert, many people are also located at different seats, which changes the impulse response.

iii) Validity of the LTI System Assumption of the Acoustic Environment

The time invariance of a system in a physical manner means that the system must give the same results for the experiment when the experiment is conducted at different times or temperatures. Hence it is true that the Odeon as a system technically is not a time-invariant system since the heat, air density, amount of people, and type of different noises outside that effect the Odeon might change; therefore, the true finding of the experiment is the output of the system to a given input at a specific time but since those features are not as much effect on the impulse response like the structure of the building they can be regarded for this part of the experiment. So, the basic assumptions are no object enters or leaves the Odeon when the time changes, and temperature, air pressure, and weather conditions are also the same through time.

Secondly, we must consider linearity. Luckily linearity is not something that requires lots of features to be ignored. The air medium has good linear features for sound waves with normal amplitudes that humans can safely hear, enabling us to use the convolution formula. Air medium is not linear when sound signals like shock waves appear, which have high amplitude and pressure that change the speed of the sound and create problems when adding two signals.

Signals with small amplitude and pressure have less effect on their speed, so the system is a good linear approximation.

The LTI assumption of this system is correct when those features are accepted to have a minor effect on the system's response and, therefore, can be ignored. This concludes that the system described throughout this report is an LTI system, and its output can be found by convolving its input response with a given input.

iv) Distortions and Their Reasons

The first reason for distortion is the physical setup, which implies it is impossible to create a pure impulse with no o other signal input. There are many sound sources around the Odeon, even though they are minor. That distorts the impulse and, therefore, the output. The second reason may be the properties of the recording device. We intentionally chose to use Gökay's phone, which was bought one day ago, implying there is little dirt or dust around the microphone openings. But that is not sufficient for eliminating all the distortion that microphone causes. A microphone is a converter that converts sound signals to physical vibrations and converts these to electrical signals by using induction. Throughout this process, many materials and components may create distortions and errors. The phone also samples this electrical signal to store it as an audio file, where sampling loses some information and might create distortion. The input signal and impulse response are convolved using MATLAB, where there is a limit for the bytes of a single number. Therefore, this can also lead to small distortions during the computational process. To conclude, it is proper to say that the major distortion reason is the microphone that we recorded the balloon explosion sound.

v) Noise During the Recording and Its Effects

There are many noise sources in physical systems, especially highly non-isolated ones like the ones we have used in this experiment. Odeon can capture sounds generated inside the building and create undesired echoes, so sound sources around us, like wind, weather, car sounds, etc., are some noise sources during recording. The noise is also related to the atomic nature of the air, where it can behave differently since the atoms are loosely interacting. This noise enters into the experiment from different sources, like the recorder when we recorded the balloon explosion to use it as the system's impulse response. The result of this can be expressed with the following equation by pursuing the LTI assumption:

$$h[n] = h'[n] + \mathrm{N[n]}$$

$$\textit{where } h'[n] \textit{ is the actual impulse response and } N[n] \textit{ represents the noise sources}$$

Then assuming the anechoic sound file does not introduce any additional noise,

$$y[n] = h[n] * x[n] = h'[n] * x[n] + N[n] * x[n] \; = y[n] + N'[n]$$

The signal $N'[n]$ is the effect of the noise at the output, which creates undesired distortions for the sound and damages the harmony.

I think the acoustic structures are created to amplify, protect, and create the harmonics of the played notes on the stage, which are probably the echoed sounds. Those additional voices from around the Odeon create a "surrounding" music source around our heads. I think this is also the main motivation behind home cinema systems, where the sound is desired to come from every corner of the room. It is more pleasant to hear the echoing and filtered music than the unechoed one because of the harmonics. There is also a trade-off; since we are not robots, we choose the introduction of distortion for good music quality since the harmonics, and their echoes increase the experience.

References:

[1] Pätynen, J., Pulkki, V., and Lokki, T., "Anechoic recording system for symphony orchestra," *Acta Acustica united with Acustica*, vol. 94, nr. 6, pp. 856-865, November/December 2008. [Online IngentaConnect]