Bilkent University

Electrical and Electronics Department

EE321-02 Lab 3 Report:

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Introduction:

In this lab work we applied the concepts of impulse, impulse response, and related filtering to a real-life problem. The task was to record an approximate impulse response of a music auditorium -which in our case is Bilkent ODEON- and then pass music recorded in an anechoic environment through a linear time invariant system to simulate the music that would be actually heard by a person sitting at a certain position while that music is performed on stage.

In this lab, Bilkent ODEON is an LTI system. The anechoic music recording I acquired from the internet is the input of the LTI ODEON system. And finally, the echoing file is the output of the LTI ODEON system.

The audio file “**impuls.mp3**” contains the response of Bilkent ODEON to a popping balloon which is also the approximation of the impulse response of Bilkent ODEON **(Appendix 1)**.

The MATLAB file **“lab3driver.mat”** is the main MATLAB module where the audio data from the anechoic recording and the impulse response is read, the convolution operation took place, and the ODEON simulation output audio file is created. **(Appendix 3)**

The audio file **“opera.mp3”** is the output file where we simulate the ODEON response to the anechoic symphony music. **(Appendix 4)**

Lab Work:

The very first task I was committed was to physically go to ODEON and try to obtain the impulse response of the ODEON system. This was a sensible approach to the problem since an impulse response of an LTI system reveals all the information about that LTI system. Any output signal that ODEON generates can be found with only the input signal being convolved with the impulse response of ODEON.

A very good way of approximating the impulse response of an auditorium is by recording a balloon popping there. Nevertheless, beware that ODEON’s impulse response changes from seat to seat where the recording takes place and where exactly the input is being played on the stage.

My best friend Aleyna and I decided to team up and record one of the specific impulse responses ODEON generates. Here you can see Aleyna, myself and a selfie of us on the day where the experiment took place **(Figures 1.1 to 1.3)**. I popped the balloon on the stage, and she recorded it with her phone. Note that we popped yellow and blue balloons because my favourite sport team is Fenerbahçe.

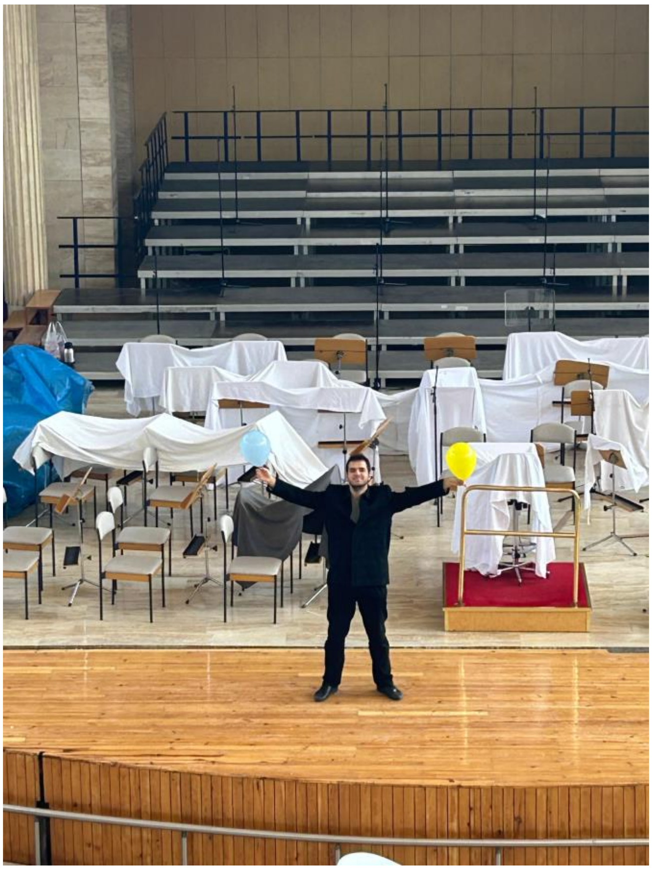

Figure 1.1: Me posing with the Fenerbahçe colored balloons which I will pop seconds later

Figure 1.2: Aleyna posing from the spot where she took the recording

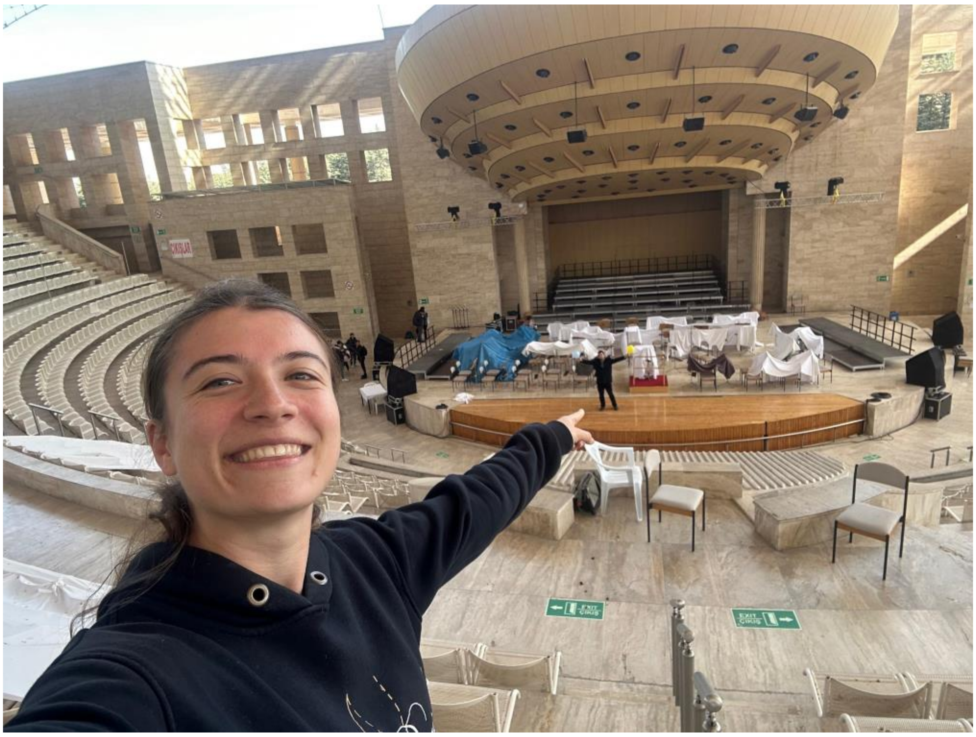

Figure 1.3: Our selfie in Bilkent ODEON

After we recorded the sound into the .mp3 file **"impuls.mp3" (Appendix 1)**, now it was time to acquire the anechoic music from the internet. The reason I was using anechoic music is because I wanted the input to be completely pure in terms of echoes and noises. If the input had echoes in it, then the ODEON system output would not make sense. The ODEON system would put echoes over an already-echoed music, and the experiment would have failed.

There is a project where the researchers have recorded anechoic symphony music for auralization and concert hall acoustics studies. The samples can be found on internet and be used free for academic research. The full documentation of recordings is presented in a journal article published in Acta Acustica united with Acustica. Please, also check this article if you want more information on how they acquired the anechoic music **(Appendix 2)**.

I downloaded and stored an anechoic music in a local directory in my computer system. Now it was time to simulate in MATLAB. Firstly, I extracted both the impulse response and the anechoic music into MATLAB as vectors with different lengths. Then I have shortened both vectors in order to achieve two things: In the impulse response recording, the balloon popped a few seconds after the recording has started, I adjusted the start of **impuls.mp3** to the point where the audio started just as the balloon popped. Also, the anechoic music files were 227 seconds long and consisted of different .mp3 files in which each of them contains a different music instrument sound – drums, violin, vocal etc. I have shortened the vectors where the sound data only 10^{th} to 30^{th} seconds were present in the vector. Then I added these multiple vectors which correspond to different instruments in order to create a symphony of different instruments.

Here you can see the impulse response of ODEON both clipped and unclipped which are sampled at a rate of 48000 Hz [bits/second] **(Figures 2.1&2.2)**.

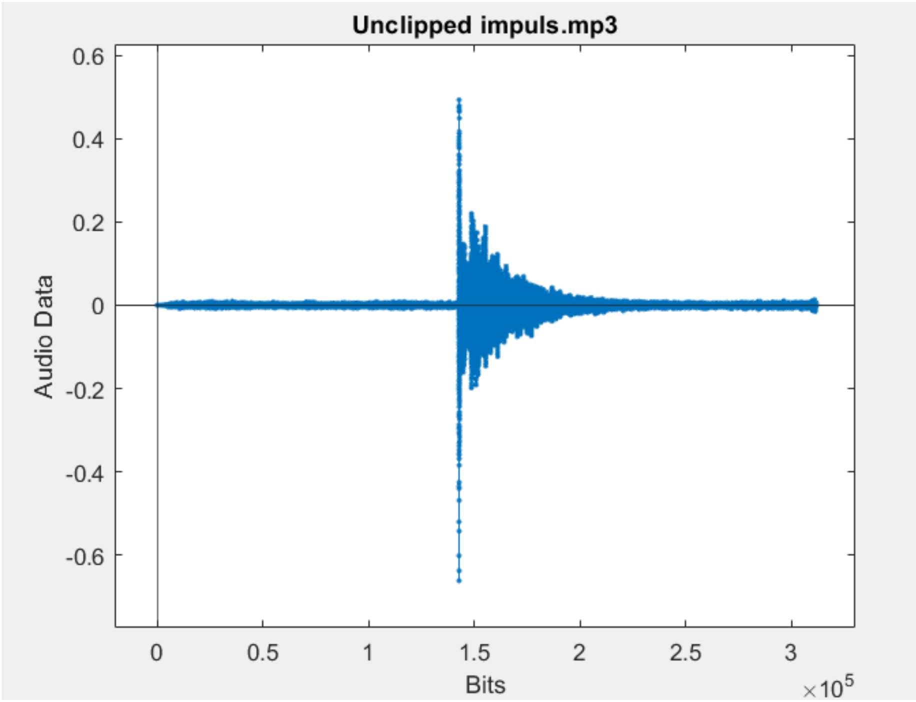

Figure 2.1: Original audio data of the impulse response of ODEON

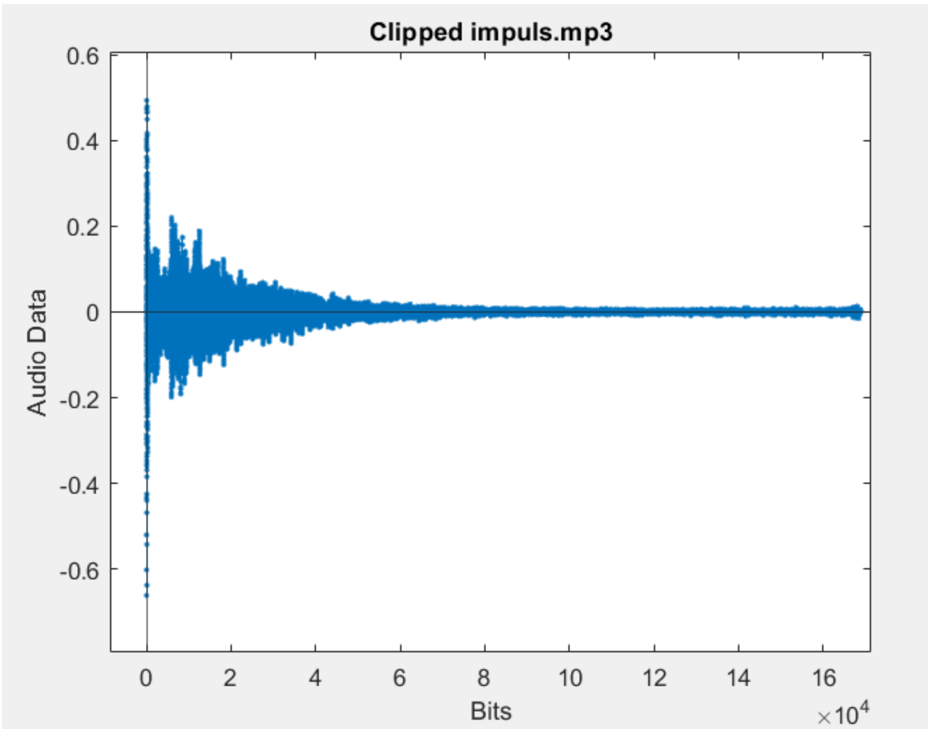

Figure 2.2: The clipped audio data of the impulse response of ODEON

Here you can see both the 227 seconds long and the clipped 20 seconds long anechoic symphony music data which are sampled at a rate of 48000 Hz [bits/second] **(Figures 2.3&2.4)**.

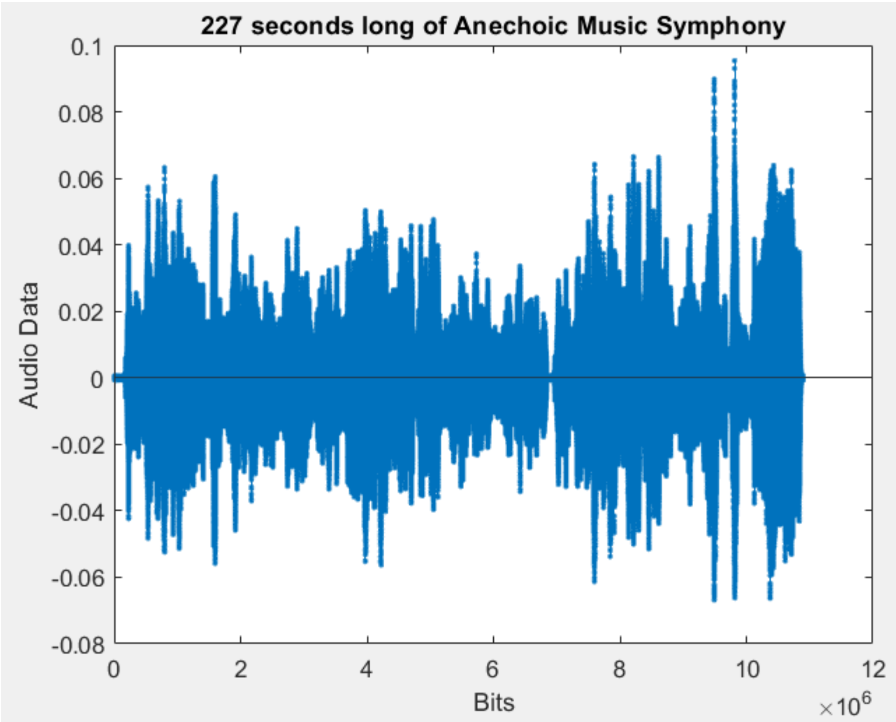

Figure 2.3: The entire audio data of the anechoic symphony music

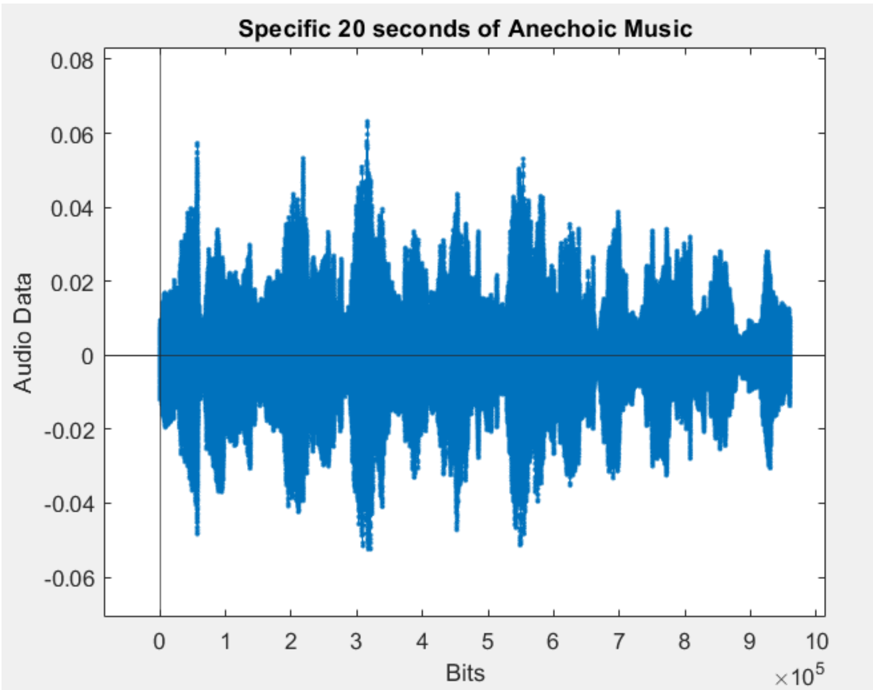

Figure 2.4: The clipped 20 seconds long audio data of the anechoic symphony music

Now it was time to perform the main part of the experiment where we calculated the mathematical convolution of the clipped impulse response **“impuls.mp3”** and the clipped 20 seconds long input signal symphony music and then created an output. After convolving both vectors with the built-in MATLAB function for convolution, I have created an output vector and have stored it in an .mp3 file **“opera.mp3”** in a local directory in my computer. Here you can see the output music **(Figure 2.5)**:

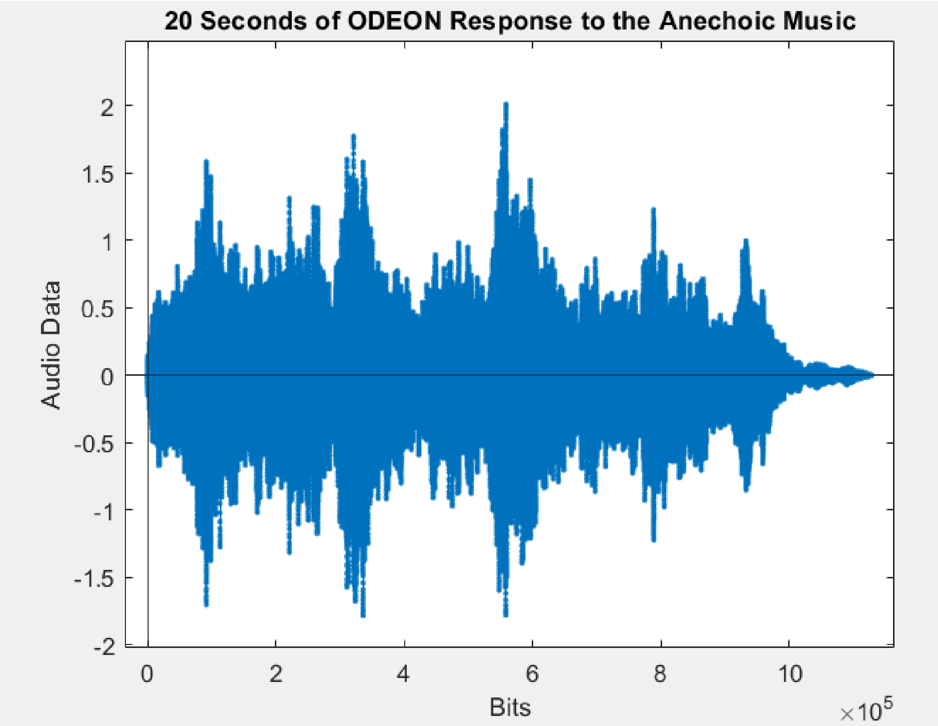

Figure 2.5: ODEON Response to the Anechoic Music Symphony

At this point, the lab work was completed and now it was time to comment on the results.

Conclusion & Comments:

I-) Comments on the quality of the recorded impulse:

The recorded impulse audio is just an approximation of the actual impulse response of ODEON. Going even further, the recorded audio is not even the actual sound of the balloon popping. A mobile phone cannot perfectly record the sound of an environment due to limitations in its hardware and processing capabilities. Here are some examples to the reasons why a mobile phone is not 100% accurate in recording audio.

Mobile phone microphones are typically small and optimized for speech frequencies (roughly 300 Hz to 3000 Hz), so they may not capture the full range of audible frequencies

(20 Hz to 20,000 Hz). This can lead to loss of detail, particularly for high or low-frequency sounds.

Phones use specific sampling rates and bit depths to capture audio. Typical recording settings (e.g., 44.1 kHz, 16-bit) may lose some subtle details, especially in high-fidelity environments. Also, many mobile phones use lossy compression (AAC, MP3) to reduce file size, which can introduce artifacts and reduce audio quality.

Finally, many mobile devices have AGC to automatically adjust the recording volume. While this helps prevent distortion in very loud or very quiet recordings, it can also cause unintended volume fluctuations.

II-) Comments on the nature about the impulse response:

Even though we perfectly captured the balloon sound and filtered all the unwanted noises, there is still an issue with the balloon popping being approximated as the impulse response.

An ideal impulse has an infinitesimally short duration, theoretically a single point in time. However, a balloon pop has a finite, though short, duration. An ideal impulse contains all frequencies equally and has white noise. While a balloon pop does produce a broad range of frequencies, it may not contain all frequencies with equal energy or distribution.

Also, a balloon popping produces a significant sound pressure level, but in some environments as large as ODEON, the sound pressure level may not be high enough to excite all acoustic reflections and resonances.

III-) Comments on the validity of the linear time invariant system assumption of the acoustic environment

An acoustic environment such as ODEON can be approximated as a linear and time-invariant system, especially in controlled or stable environments where conditions remain constant. Nevertheless, both linearity and time invariance may break down in breakdown in certain extreme conditions.

For linearity, superposition and scaling principles should be satisfied. In general, acoustic wave propagation through air is linear, as sound waves combine additively. However, linearity may break down at very high sound pressure levels, such as in cases of extreme loudness – explosions, high pressures etc.- where nonlinear effects like harmonic distortion can occur. For everyday sound environments, the linearity assumption usually holds well.

An acoustic environment is time-invariant if its characteristics do not change over time. The time-invariance can hold over short periods, such as in a stable room where environmental factors remain consistent. However, in dynamic environments - wind, moving objects, temperature or humidity differences- the time-invariance assumption may not hold due to changing conditions.

To sum up, an acoustic environment can be approximated as a linear and time-invariant system, especially in controlled or stable environments where conditions remain constant. Linearity holds under normal sound pressure levels but may break down under extreme conditions. Time-invariance holds over short durations in steady environments but may fail in dynamic or changing environments.

IV-) Comments on the distortions and their reasons

The distortions in mobile phone recordings can arise from several factors, primarily related to hardware limitations and digital processing. Some mobile phones apply digital filters that can alter phase relationships, resulting in phase distortion. This can affect the recordings and cause certain sounds to seem shifted or spatially inaccurate. Also, high sound levels can introduce nonlinear behavior in the microphone, leading to harmonic distortion. Finally, limited bit depth in the ADC of the microphone causes quantization error, which is especially noticeable in quiet recordings where subtle details are lost.

V-) Comments on the noise during the recording and its effects

Even though Aleyna and I tried to record the balloon popping in a very quiet time, there is always some unwanted noise behind while we are recording. This could be people talking with each other outside of ODEON, cats meowing near ODEON, Ezan sound coming from nearby mosques, winds or even the seismic activities coming from the ground. Of course, Aleyna’s phone which we took the recording cannot detect all these unwanted noises, but it definitely can’t only record the balloon sound as well. All these types of noises affected the quality of the recorded impulse response.

VI-) Comments on the effects of acoustical structure of the listening environment to the listening quality

The structure of the acoustical environment is extremely significant on the listening quality of the audience. Every single change in the environment changes the impulse response of the system such as the shape and size of the environment, where the singer is on the stage, where the individual listener sits, the temperature and humidity of the environment, or even the presence of an audience. The audience acts as an absorber and reflector to the input signal and therefore, the output that system produces to the same input changes according to whether the ODEON is empty or full.

Conclusions:

I personally prefer acoustic music over the anechoic one because it feels more natural. The anechoic music is very robotic, and I think the very good thing about music is that it feeds the human soul. How can a robotic sound feed a human's soul? Natural is better in my opinion.

I think this lab was very entertaining and helpful for us both in understanding the essence of the convolution operation and the joy of doing some fun work in a hard department as EEE. I will never forget this lab; it will always put a happy face on me whenever I think of what I have done in this lab. Also, the concept of anechoic music was completely new to me, and I am very amazed on what I learned about music and echoes.

Appendices:

1. https://github.com/fmcetin7/Bilkent-EEE-321/blob/main/lab%203/impuls.mp3
2. 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. [https://www.ingentaconnect.com/content/dav/aaua/2008/00000094/00000006/art00004;jsessionid=k8fcjppmbkmp.x-ic-live-02].
3. https://github.com/fmcetin7/Bilkent-EEE-321/blob/main/lab%203/lab3driver.mat
4. https://github.com/fmcetin7/Bilkent-EEE-321/blob/main/lab%203/opera.mp3