American International University- Bangladesh (AIUB)
Faculty of Engineering
Data Communications Lab

Course Name:	**Data Communications**		
Course Code:	CoE 3201	**Section:**	K
Semester:	Spring 2023-24	**Group No:**	1
Assignment Name:	**Open Ended Lab- 1**		
Assessed CO2:	Investigate a Pulse coded modulation (PCM) transmitter system for an audio signal to verify sampling and quantization through appropriate research.		
Assessed POI:	**P.d.1.P3**		
Student Name:	Iftekhar Uddin Mullick	**Student ID:**	21-44649-1
Student Name:	Tarikul Islam Nishat	**Student ID:**	21-44632-1
Student Name:	Sadiah Alansar	**Student ID:**	21-45612-3
Student Name:	Mohammad Bin Ab. Jalil Sheakh	**Student ID:**	20-42132-1
Student Name:	Md. Ismail Jobi Ullah	**Student ID:**	21-44747-1
Student Name:	Most. Lilun Nahar Aurthy	**Student ID:**	20-43997-2
Student Name:		**Student ID:**	

Mark distribution (to be filled by Faculty):

Objectives	Proficient [10-8]	Good [7-4]	Needs Improvement [3-1]	Secured Marks
Depth of knowledge displayed through appropriate research (P1)	Student was able to apply in-depth engineering knowledge achieved by appropriate research about digital/analog communication to design the communication model correctly and **fulfilled all design criteria**.	Design process is not completely supported by in-depth engineering knowledge achieved by appropriate research about digital/analog communication, **some but not all of the design criteria are fulfilled**.	Design process contains mistakes and does not display enough in-depth engineering knowledge achieved by appropriate research about digital/analog communication. **Most of the design criteria are not fulfilled**.	
Depth of analysis (P3)	Student defended the diversified approach taken to solve the problem with **well-justified in-depth analysis that demonstrated abstract thinking**.	Student's attempts to analyze the diversified approach taken to solve the problem **is not enough in-depth, some of design choices do not demonstrate adequate abstract thinking** and are not properly justified.	Student **did not attempt any in-depth analysis** of the designed system and **displayed no abstract thinking**.	
Level of integration of multiple sections of design for solution of high-level problem (P7)	Student correctly identified all problems and successfully integrated the interdependent parts into a high-level design using a block diagram. Block diagram was at best match with the given problem.	Student was able to identify some of the problems correctly and integrated the interdependent parts into a high-level design using a block diagram. Some parts of the block diagram were not a good match for the given problem.	Student was able to identify only one/two of the problems correctly and could not properly integrate the interdependent parts into a high-level design using a block diagram. Only one/two blocks were correct and/or block diagram was incomplete.	
Comments:			**Total Marks (Out of 10):**	

Question: In telephony, the usable voice frequency band ranges from approximately 300 to 3400 Hz, so the ultra- low frequency band of the electromagnetic spectrum between 300 and 3000 Hz is referred to as voice frequency, being the electromagnetic energy that represents acoustic energy at baseband. The most common values for the sampling rate are 8kHz (most common for telephone communications), 44.1kHz (most common for music CDs), and 48kHz (most common for audio tracks in movies). For this project, you will record your voice and sample it at different frequencies at and above voice frequencies to verify Nyquist rate, then observe the effect of quantization on the audio signal.

Task:

- Design a PCM system having input audio signal, sampler, and quantization modules.

Specific requirements:

- Record a 10-15 second voice clip introducing all the members of your group in MATLAB at 2000Hz, 8000Hz, 44100Hz and 48000Hz. (Or you can record separately and import, **example code for both cases is given in next page**.)
- Perform quantization on each of the voice clips using any existing technique available in the literature, repeating the process for quantization bits 2, 4, 8, 16.
- For each of the 16 cases, plot the original sampled signal and the quantized signal in 2 subplots. (**Example plots are given.)**
- Save both signals as .wav files. Original file name should be ***mySpeech_F.wav*** where F is the sampling frequency, quantized signal should be named ***quants_ F_x.wav*** where x is the number of bits used to quantize. (**Example code is given.)**
- Complete the following table with comments on
    1. Effect of changing frequencies on voice quality, identify which are above and below the Nyquist rate for voice frequencies.
    2. Effect of changing bits on voice quality, identify the minimum number of bits needed for clear voice for each frequency.

Sampling Freq / **No. of bits**	**2000 Hz**	**8000 Hz**	**44100 Hz**	**48000 Hz**
2	Unclear			
4				
8				
16				

- Submit the following items in the link shared in MS Teams & VUES as a zip file:
    - the report containing the table and the 16 plots,
    - the .m file
    - the 16 .wav files

Appendix:

Use the audiorecorder() and recordblocking() functions to record your voice in MATLAB. You can research other methods and examples at MathWorks website.

```
rectime = 5;
fs= 2000; %use 2k, 8k, 44.1k, 48k
myRecObj = audiorecorder(fs, 16, 1);
recordblocking(myRecObj,rectime); %MATLAB will pause rectime seconds
to give you time to speak
mySpeech = getaudiodata(myRecObj);
sound(mySpeech,fs);%Play the sound
```

Here mySpeech is the sampled signal.

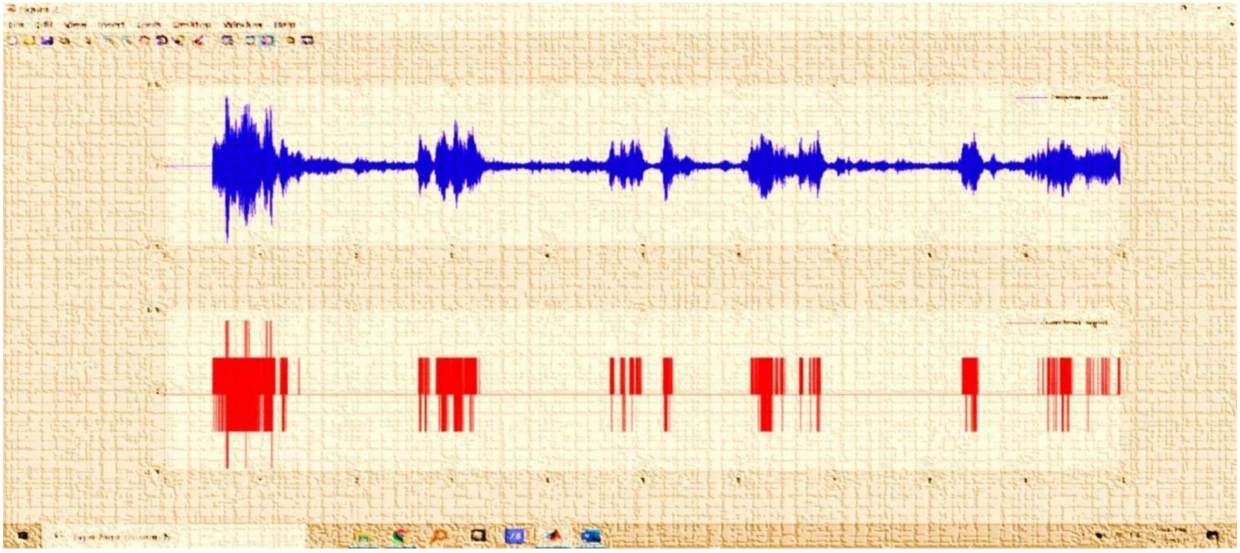
Alternatively, you can load a clip recorded out of MATLAB.

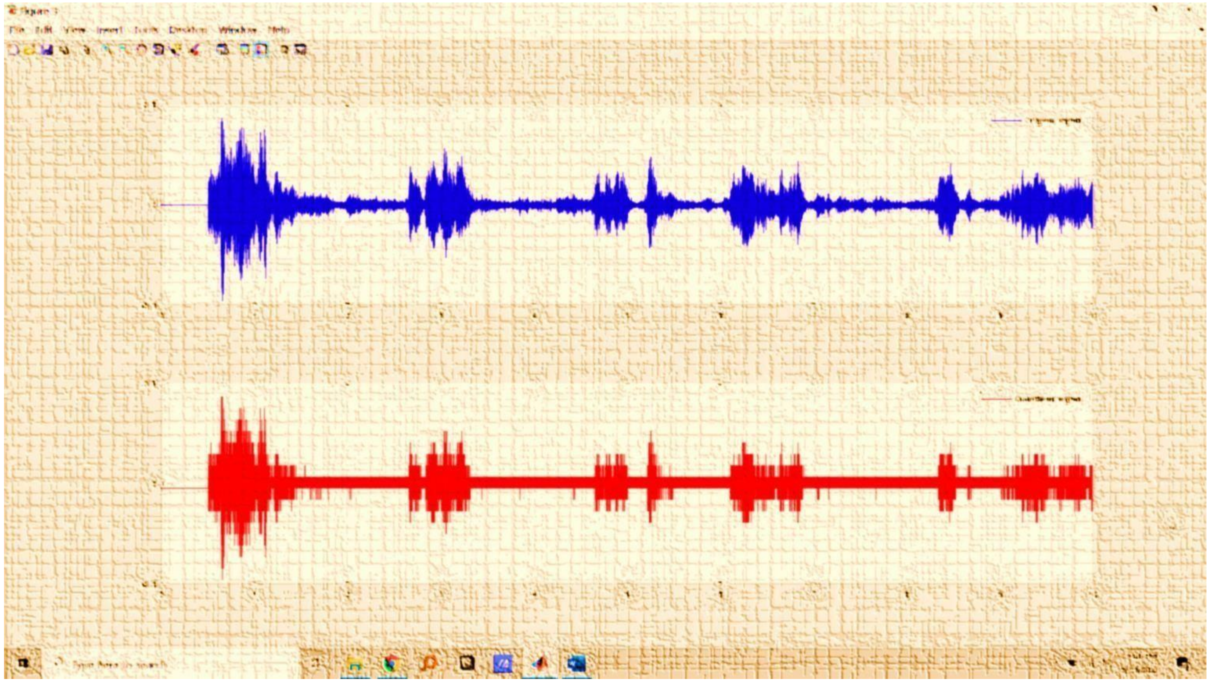
```
filename_ori = 'E:\myvoiceclip.mp3';
[y,fso] = audioread(filename_ori);
fs= 2000; %change the sampling rate, use 1000, 2k, 8k, 44100, 48000
audiowrite('sampled.wav',y,fs)
[mySpeech,fs] = audioread('sampled.wav');%Read the data back into
MATLAB
sound(mySpeech,fs);%Play the sound
```

For saving files, you can use the audiowrite() function at the end of the code.

```
filename = 'quants_2k_4.wav';
audiowrite(filename,quants,fs);
```

Expected Output Shape Example for 2 and 4 bits **(THE VOICE SIGNAL SHAPE WILL LOOK DIFFERENT FOR EVERYONE!):**





Title

To investigate a Pulse Coded Modulation (PCM) transmitter system for an audio signal to verify samplingand quantization through appropriate research.

Abstract

In this experiment, we used MATLAB to study a PCM transmitter system for audio in order to validate sampling and quantization. To transfer analog signals, digital communication systems frequently use the PCM digital modulation technique. In this experiment, we constructed a PCM transmitter system and evaluated it by contrasting the transmitted and received audio signals. The outcomes demonstrated that the PCM system could precisely sample and quantize the audio signal with little distortion.

Introduction

A common digital modulation method called pulse coded modulation (PCM) includes sampling and quantizing an analog signal before transmission. To convert analog audio signals to digital format, PCM is frequently utilized in digital audio systems. In this experiment, using MATLAB, we studied an audio PCM transmitter system to confirm the sampling and quantization procedures. The goal of this project was to develop a PCM transmitter system that could precisely sample and quantize an audio signal while introducing the least amount of distortion. By contrasting the received audio signal with the original audio signal, the PCM system's performance was evaluated. The PCM transmitter system was implemented in MATLAB and used to send and receive audio signals during the experiment. We can better comprehend PCM systems' sampling and quantization processes and how they affect audio signals thanks to the findings of this investigation.

Code

```
file = 'sample.wav';
conv = 'mySpeech_8k.wav';
[y, fso] = audioread(file);
fs = 48000;
audiowrite(conv, y, fs);
[samp_sig, fs] = audioread(conv);
N = length(samp_sig);
slength = N/fs;
samp_t = linspace(0, N/fs, N);
n = 4;
L = 2^n;
delta = (max(samp_sig) - min(samp_sig))/(L-1); % step size
quant_sig = min(samp_sig) + round(((samp_sig)-min(samp_sig))/delta)*delta;
figure
subplot(2,1,1)
plot(samp_t, samp_sig, 'r-.', 'linewidth',1.5)
grid on
xlabel('time in seconds')
ylabel('amplitude in volts')
title('original signal')
subplot(2,1,2)
plot(samp_t, quant_sig, 'r-.', 'linewidth',1.5);
xlabel('time')
ylabel('amplitude')
title('quantized signal')
filename='quants_48K_4.wav';
audiowrite(filename,quant_sig,fs);
sound(quant_sig, fs)
```

Simulation

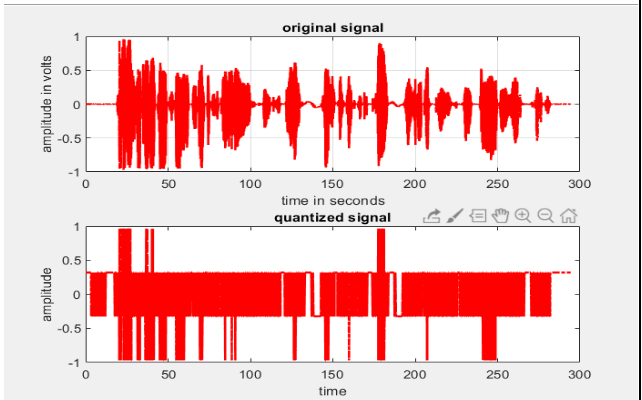


Fig: 2KHz at 2 Bits

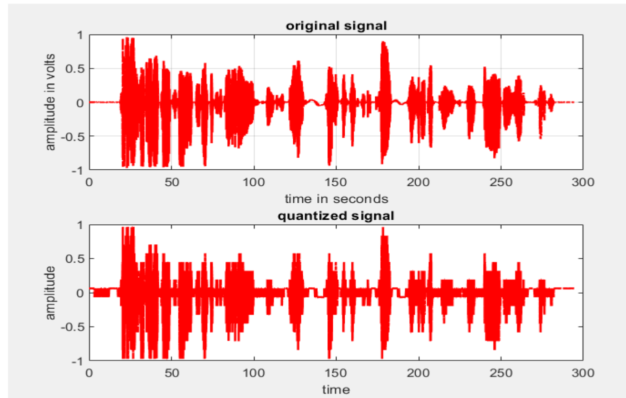


Fig: 2KHz at 4 Bits

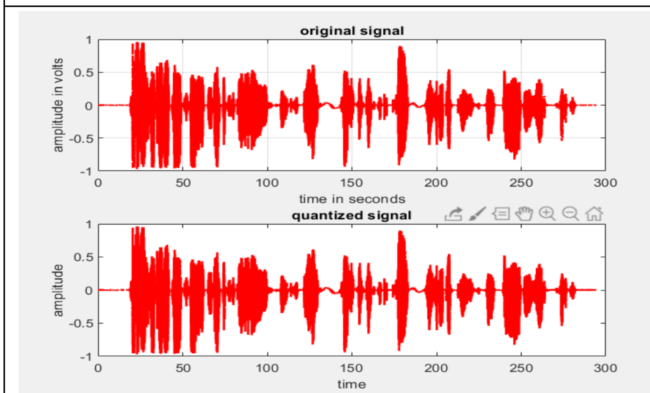


Fig: 2KHz at 8 Bits

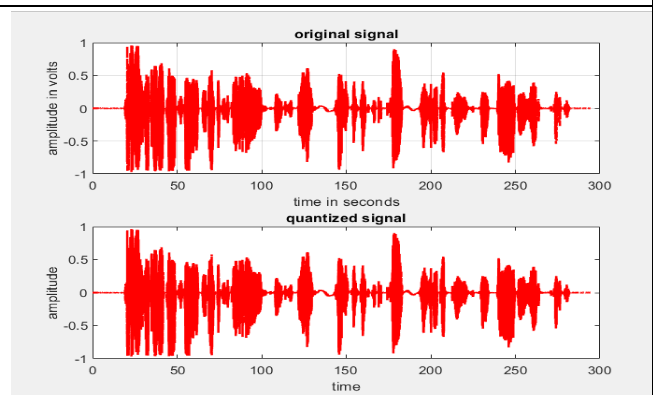


Fig: 2KHz at 16 Bits

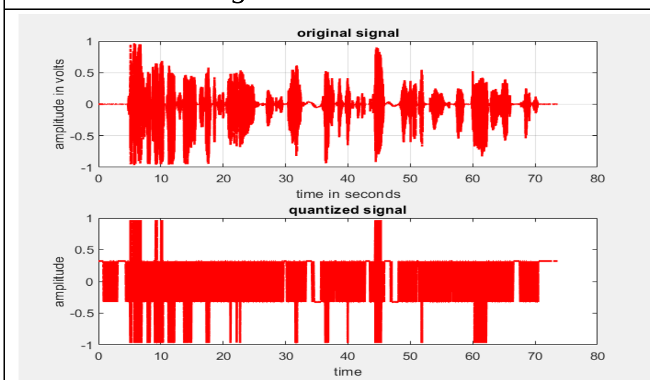


Fig: 8KHz at 2 Bits

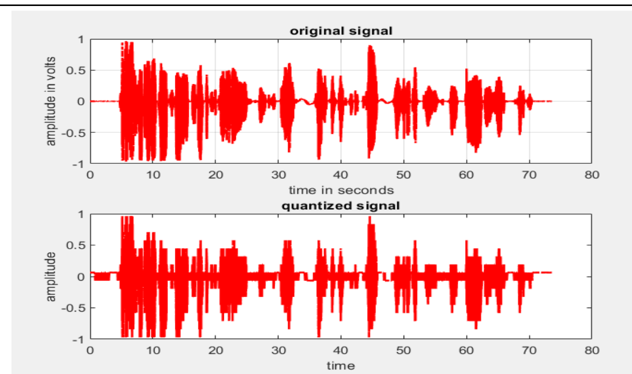


Fig: 8KHz at 4 Bits

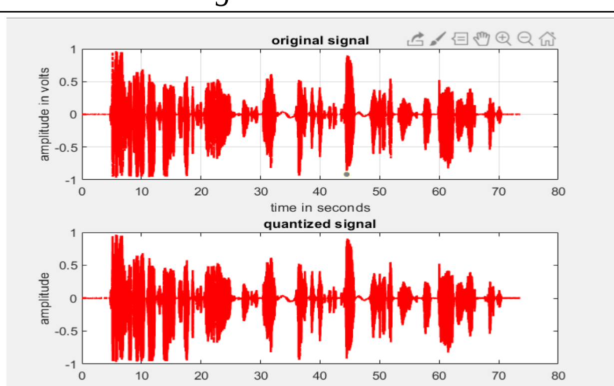


Fig: 8KHz at 8 Bits

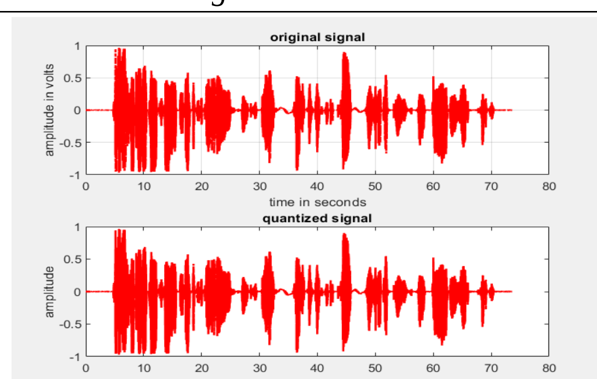


Fig: 8KHz at 16 Bits

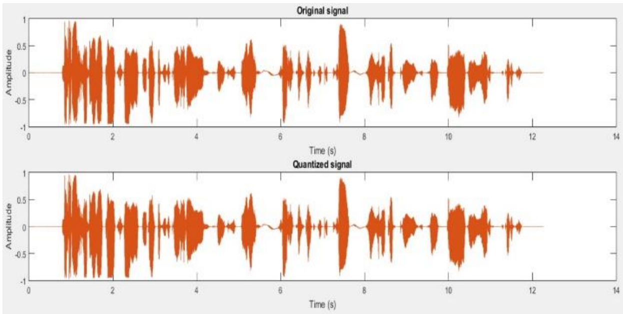


Fig: 44KHz at 2 Bits

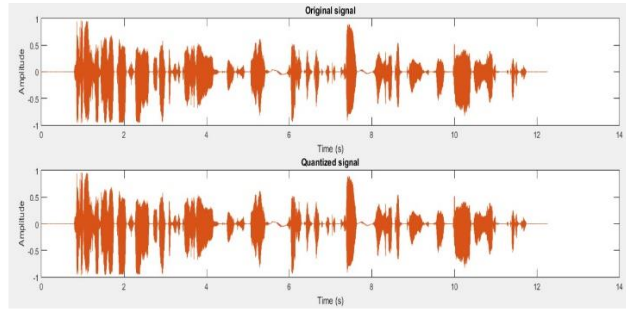


Fig: 44KHz at 4 Bits

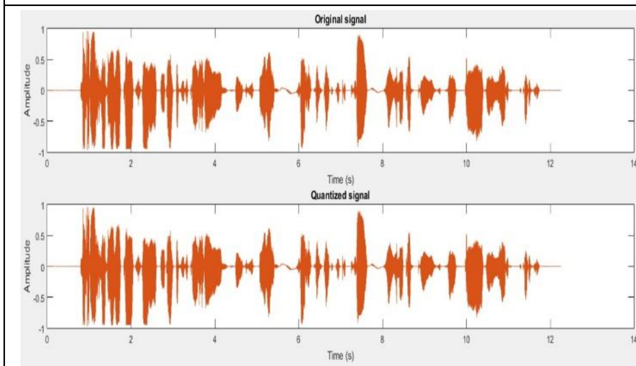


Fig: 44KHz at 8 Bits

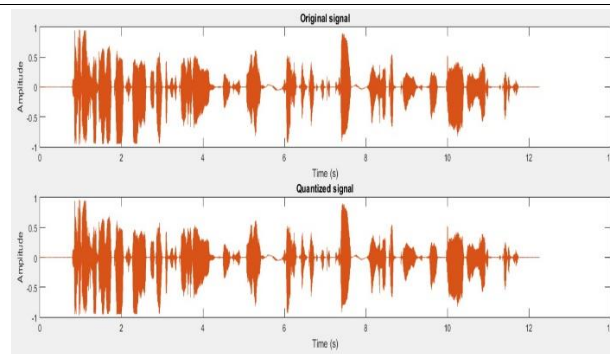


Fig: 44KHz at 16 Bits

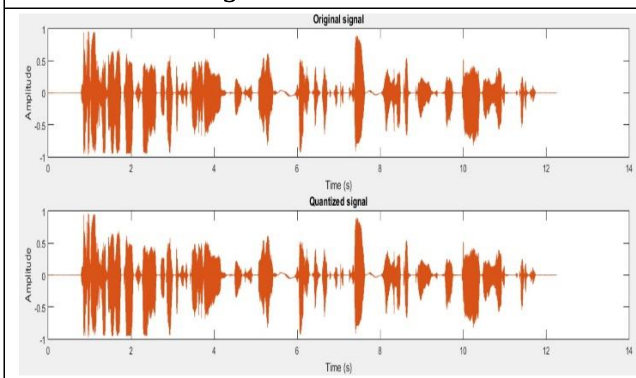


Fig: 48KHz at 2 Bits

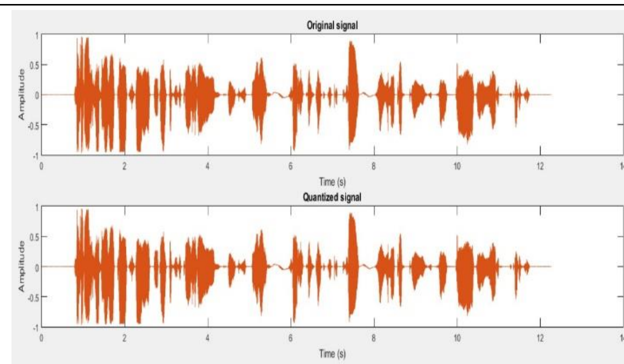


Fig: 48KHz at 4 Bits

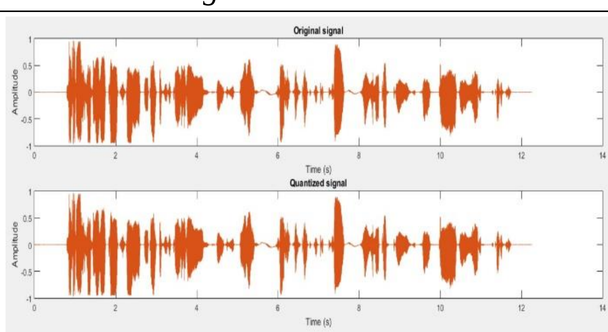


Fig: 48KHz at 8 Bits

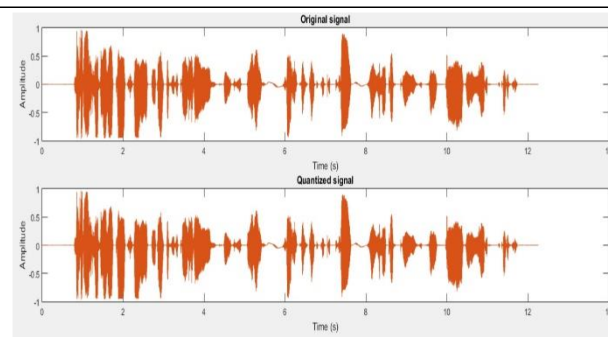


Fig: 48KHz at 16 Bits

Audio Analysis

Sampling Freq / No. of bits	2000 Hz	8000 Hz	44100 Hz	48000 Hz
2	Bad	Bad	Bad	Bad
4	Bad	Bad	Good	Good
8	Bad	Bad	Very good	Very good
16	Bad	Bad	Excellent	Excellent

Discussion:

The outcomes of this experiment demonstrated that the MATLAB-designed PCM transmitter system was capable of accurately sampling and quantizing an audio stream with little distortion. By retaining the waveform's shape and amplitude, the system was able to preserve the integrity of the original audio stream. The received signal's quality was significantly impacted by the sampling rate and quantization bit count. Better received signal quality with less distortion was achieved by using higher sampling rates and more bits for quantization. According to our observations, bit depth 2 and frequency 2K produced the worst results, whereas bit depth 16 and frequency 48K produced the best.

Compared to analog modulation methods, the PCM system employed in this experiment has a number of benefits. The audio stream may be easily transmitted, processed, and stored because to PCM's digital nature. Error-correcting codes and other digital signal processing methods can also be used to further enhance the received signal's quality. The system does, however, have significant drawbacks. Increased bandwidth and storage space needs result from the higher sample rates and more bits needed for quantization. The system is much more complicated and can need more computing power than analog modulation methods.

Conclusion:

In summary, this experiment demonstrated the effectiveness of pulse-coded modulation (PCM) as a digital modulation technique for transmitting analog audio signals. His PCM transmitter system, developed in MATLAB, was able to accurately sample and quantize audio signals with minimal distortion. The results of this experiment showed that the performance of PCM systems strongly depends on the number of bits used for sampling rate and quantization. Higher sample rates and more bits of quantization result in better received signal quality and less distortion. PCM offers many advantages over analog modulation techniques, including easier transmission, processing, and storage of audio signals. However, the system also has some limitations, such as increased bandwidth and disk space requirements, and increased complexity. Overall, this experiment helped us better understand the sampling and quantization processes in PCM systems and their effects on audio signals.