VOICE RECORDER USING MATLAB

20ECL52-DIGITAL SIGNAL PROCESSING LABORATORY

A Mini Project Report

Submitted by

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ABSTRACT:

Speech is a form of communication in everyday life. It has existed since human civilizations began. First, signal changes with the time where demonstrates short-time characteristics, which indicates that signal is stable in a very short period of time. Second, spectrum energy of the human's speech signal normally centralized in frequency between 0-4000Hz. [5] It is an analog signal when speak out from human, and it will convert to a digital signal when input into computer, the conversion of this process introduce the most basic theory for signal processing- signal sampling. It provides principles that the time domain speech analog signal X(t) convert into the frequency domain discrete time signal X(n) while keeps characteristics of the original signal in the same time. To use all of the audio recorder object features, the system must have a properly installed and configured sound card with 8- and 16-bit I/O and support for sampling rates of up to 48 kHz.On 32-bit, Windows-based machines with an installed 24-bit audio device, audiorecorder supports 24-bit recording.The FFT function in MATLAB helps in converting the recorded voice signal in time domain to frequency domain.Performing convolution helps in defining a linear time invariant (LTI) system, compute the output of an LTI system for an arbitrary input audio signal given its impulse response. Digital filters are implemented so as to eliminate the unwanted frequencies (noise) of the signal components from the signal being processed. A digital filter uses various functions to compute the input signal. We can classify filters in several different groups but the major types are, FIR (Finite Impulse Response) and IIR (Infinite Impulse Response) filters.

KEYWORDS: Signal sampling , Fourier Domain , Speech processing , LTI.

INTRODUCTION

form of communication in everyday life. It has existed since human civilizations began. Speech is applied to high technological telecommunication systems. Scientists began formulating algorithms which might change the nature of voice signal since the 1970s after the emergence of digital devices. The core of signal processing is a way of looking at the signals in terms of sinusoidal components of various frequencies (the Fourier domain) . The techniques for categorizing signals in their frequency domain, filtering, developed on analog electronics but after the 1970s signal processing has more and more been implemented on computers in the digital domain. In this report, we will be looking more into speech processing with the aid of an interesting technology known as MATLAB. MATLAB (Matrix Laboratory) becomes the efficient tool in digital signal processing. MATLAB is a well-known tool for numerical calculations, this report employs it’s features like 2D plot,audio record as well. This makes MATLAB a perfect tool for the application. Speech processing is one of the fastest growing subjects. Its applications are also expanding very fast. The rapid growth of the computational capabilities of digital machines accelerates its application. This voice recorder and processor project is capable of recording a voice for user desired duration, loading a “.wav file” ,plotting of the recorded signal in frequency domain (FFT). Once the processor receives the file it provides the option of playing it or saving it for later use.. The processor understands the input values such as the duration to record and desired cut off frequency in the filter section as seconds and Hertz respectively. The recorded signal can be convoluted which can be used to study the design of the linear time invarient system (LTI).Pressing the run button will trigger the actions to be performed on the signal based on the values the user fed in.

PROCEDURE

To record data from an audio input device connected to your system:

1. Create an audiorecorder object.
2. Call the record or recordblocking method, where:
    - Record returns immediate control to the calling function or the command prompt even as recording proceeds. Specify the length of the recording in seconds, or end the recording with the stop method. Optionally, call the pause and resume methods. The recording is performed asynchronously.

    - Recordblocking retains control until the recording is complete. Specify the length of the recording in seconds. The recording is performed synchronously.
3. Create a numeric array corresponding to the signal data using the getaudiodata method.
4. FFT of the recorded audio signal is plotted using ‘fft’ function.
5. Convolution of the recorded audio signal is plotted using ‘conv’ function.
6. FIR and IIR filters of the recorded audio signal is plotted by taking in the cut off frequency as the input .
7. Click on the ‘run’ button in the editor tab to view the outputs.

PROGRAM

```
clc

clear all

close all

warning off

Fs=8000;%Sampling frequency in hertz

ch=1;%Number of channels--2 options--1 (mono) or 2 (stereo)

datatype='uint8';

nbits=16;%8,16,or 24

Nseconds=input("Enter the duration to record ");% to record audio data from an input device such as a microphone for processing in MATLAB

recorder=audiorecorder(Fs,nbits,ch);

disp('Start speaking..') %Record audio to audiorecorder object hold control until recording gets completed

recordblocking(recorder,Nseconds);

disp('End of Recording.');

%Store recorded audio signal in numeric array

x=getaudiodata(recorder,datatype);

%Write audio file

audiowrite('test.wav',x,Fs);

[x,Fs] = audioread('test.wav');

[m,n]=size(x);

dt=1/Fs;

t=dt*(0:m-1);

idx = (t>=1.030) & (t<1.032);

selected_t = t(idx);

selected_x = x(idx,:);

subplot(3,1,1)

plot(selected_t, selected_x)

title("Waveform of the recorded signal")

xlabel("Time in sec")

ylabel("Amplitude")
```

% FFT OF THE RECORDER SIGNAL

```
[data, fs] = audioread('test.wav');
data_fft = fft(data);
subplot(3,1,2)
plot(abs(data_fft(:,1)))
xlim([0,10000])
title("FFT of the recorded signal")
xlabel("Time in sec")
ylabel("Amplitude")
```

% CONVOLUTION OF THE RECORDED SIGNAL

```
out=conv(selected_t,selected_x);
subplot(3,1,3)
plot(abs(out(:,1)))
title("Convolution of the recorded signal")
xlabel("Time in sec")
ylabel("Amplitude")
```

% FIR BAND PASS FILTER DESIGN OF THE RECORDED SIGNAL

```
[y,Fs] = audioread('test.wav');
Fl = 300;
Fh = 3400;
Wc = [Fl Fh]./Fs;
bbp = fir1(256, Wc, 'band');
figure(1)
plot (1:length(bbp), bbp)
figure(2)
freqz(bbp,1)
outbp = filter(bbp,1,y);
audiowrite('filtered_audio.wav',outbp,Fs);
sound(outbp,Fs)
title("FIR Band pass filter design of the filtered signal")
```

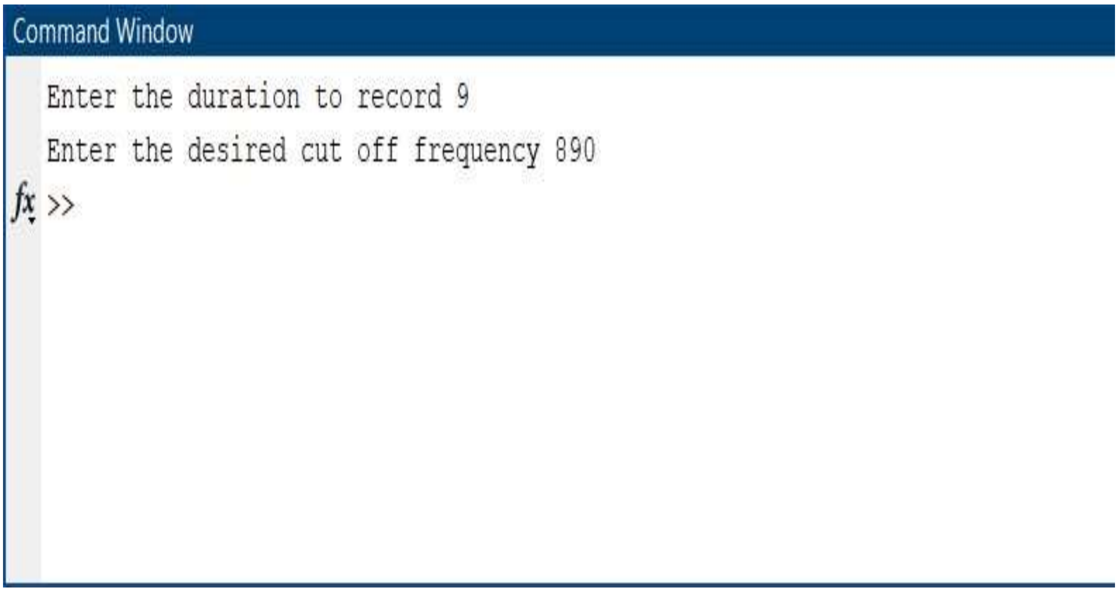
```
xlabel("Normalised frequency")
ylabel("Magnitude")
```

% IIR FILTER DESIGN OF THE RECORDER SIGNAL

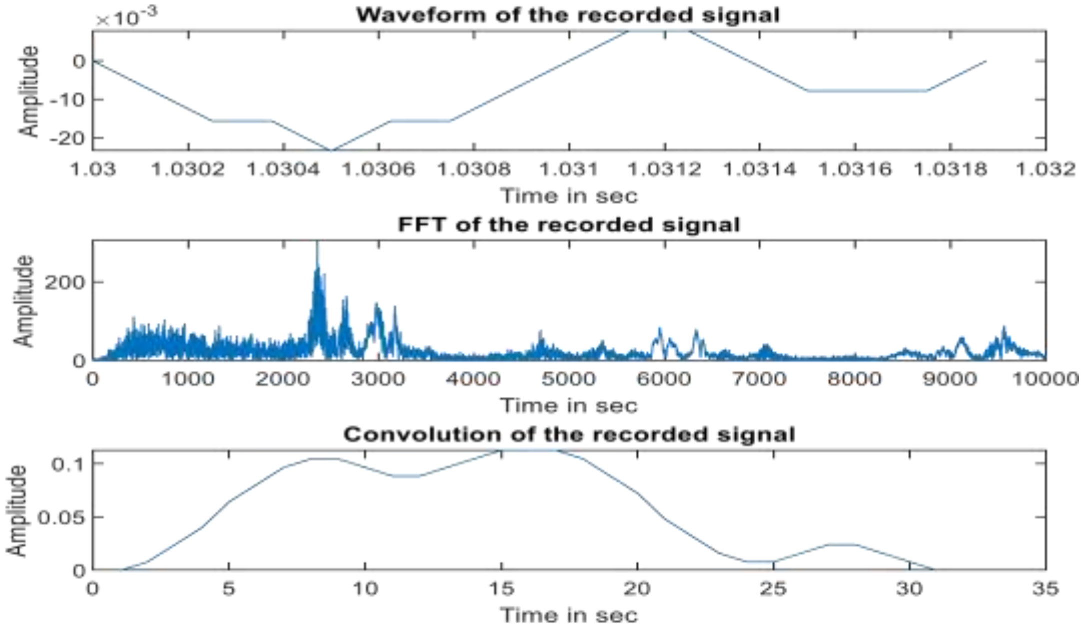
```
[dataIn, Fs] = audioread('test.wav');
% Filtering the signal
fc = input("Enter the desired cut off frequency "); % Make higher to hear higher frequencies.
% Design a Butterworth filter.
[b, a] = butter(6,fc/(Fs/2));
freqz(b,a)
% Application of the Butterworth filter
title("IIR Butterworth Filter design of the recorded signal")
filteredSignal = filter(b, a, dataIn);
% Play the sound.
player = audioplayer(filteredSignal, Fs);
play(player);
```

OUTPUT

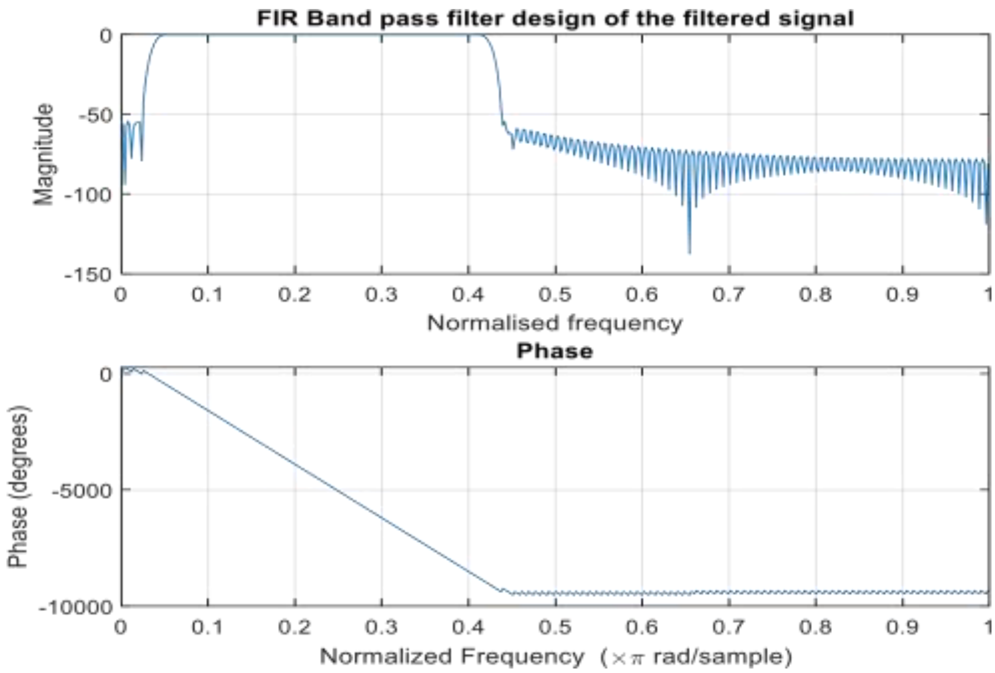
SETTING UP OF PARAMETERS REQUIRED:



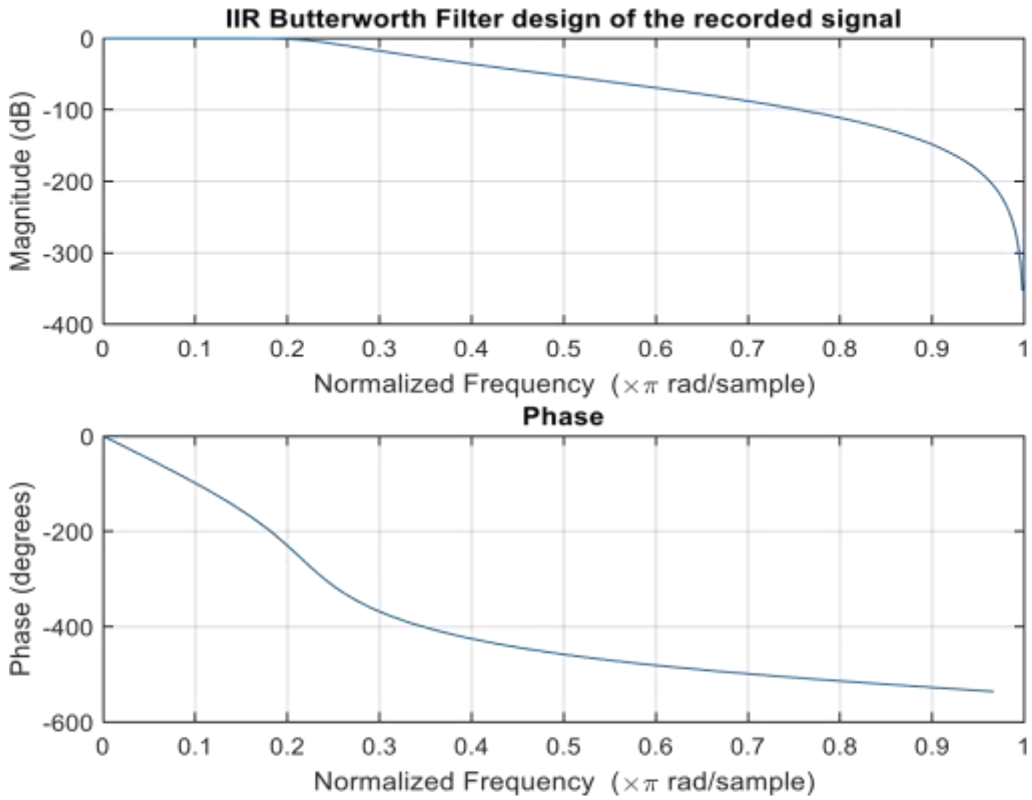
WAVEFORM,FFT & CONVOLUTION PLOT OF THE RECORDED AUDIO SIGNAL:



BAND PASS FIR FILTER PLOT FOR THE RECORDED AUDIO SIGNAL:



BUTTERWORTH IIR FILTER PLOT FOR THE RECORDED AUDIO SIGNAL:



RECORDED AUDIO SIGNAL:

test.wav

CONCLUSION

Voice processing is a very deep subject. It is rooted deeply in advanced mathematics and signal processing techniques. This project can be seen as a gateway or entry level for speech processing applications. Yet, it is challenging enough if someone does not have a concrete knowledge on the speech processing subject. The future of voice processing seems reasonably bright. During the process of this thesis, many issues have been found to be potential topics for further research work. For that reason, the following issues raised for further developments.

- A Speech recognition system: Speech to text conversion is the process which plays a very important role specially for hearing impaired individuals.
- It is also applicable in subtitling, automatic translation, court reporting, telematics, military and others.
- A Speaker recognition system: This process of identifying the speaker from its speech signal has applications in authenticating the identity of a speaker as part of the security process.

Practical Components	**Excellent** **(80 – 100%)**	**Good** **(50-79%)**	**Satisfactory** **(< 50%)**	**Marks**
Report (20)				
Presentation (15)				
Result (10)				
Viva Voce (15)				
TOTAL (60)				

RESULT:

Thus the recording of an audio signal, plotting it's waveform along with it's FFT, convolution , FIR Band pass,IIR filters using MATLAB was done successfully.