DIGITAL SIGNAL PROCESSING

AUDIO SPECTRUM ANALYZER

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1. Introduction

Our sense of hearing provides us rich information about our environment with respect to the locations and characteristics of sound producing objects. The human auditory system is able to process the complex sound mixture reaching our ears and form high-level abstractions of the environment by the analysis and grouping of measured sensory inputs. The process of achieving the segregation and identification of sources from the received composite acoustic signal is known as auditory scene analysis. Important technological applications of digital audio signal processing are audio data compression, synthesis of audio erects and audio classification. In this chapter, we review the basic methods for signal processing of audio, mainly from the point of view of audio classification. General properties of audio signals are discussed followed by a description of time-frequency representations for audio.

2. Audio Signal Characteristics

The human auditory system is responsive to sounds in the frequency range of 20 Hz to 20 kHz. The sound captured by a microphone is a time domain waveform of the pressure variation at the location of the microphone in the sound field. A digital audio signal is obtained by the suitable sampling and quantization of the output of the microphone. Although any sampling frequency above 40 kHz would be adequate to capture the full range of audible frequencies, a widely used sampling rate is 44,100 Hz. Now there are various feature of sound and some of are given below .

2.1 Perceptual Features :

Loudness :

Loudness is a sensation of signal strength. As would be expected it is correlated with the sound intensity, but it is also dependent on the duration and the spectrum of the sound. In physiological terms, the perceived loud ness is determined by the sum total of the auditory neural activity elicited by the sound. Loudness scales nonlinearly with sound intensity.

Pitch :

The sensation of a frequency is commonly referred to as the **pitch** of a sound. A high pitch sound corresponds to a high frequency sound wave and a low pitch sound corresponds to a low frequency sound wave. Amazingly, many people, especially those who have been musically trained, are capable of detecting a difference in frequency between two separate sounds that is as little as 2 Hz. When two sounds with a frequency difference of greater than 7 Hz are played simultaneously, most people are capable of detecting the presence of a complex wave pattern resulting from the interference and superposition of the two sound waves.

3. Audio Processing Techniques

Post-processing algorithms are used to suppress the noise and any artifacts created in the first stage of processing. It is primarily focused on echo, distortion removal, and speech enhancement. Equalization and filtering are popular post-processing techniques to add reverberation and noise control.

3.1 Automatic Echo Cancellation (AEC)

Acoustic Echo Canceller plays an important role in audio signal processing. It removes the echo, reverberation and unwanted noise caused by acoustic coupling between the microphone and loudspeaker. Microphones capture the far-end speech due to the acoustic coupling. Suppose you are in a voice call talking with someone over a phone. The speech of the other person you are talking to referred to as far-end speech, which would be played through loudspeaker and your voice, referred to as near-end speech which would be captured by microphone. If the far-end speech gets transmitted back to the other side of the call, the other person would hear their voice after some delay (network + processing delay). AEC blocks the transmission of far-end back to the other party in the call.

3.2 Resampling

Resampling is defined as the total no. of samples generated per second. These samples are measured in kilohertz (kHz), where one unit is equal to 1000 times per second. Different audio systems use different sampling rates and frame rates. It measures the frequency of the audio signals. It works on the principle of oversampling and transcoding which results in less noise and distortion. The higher sampling rate is more advantageous as it gives the more accurate details of rising and falls in the signals, which improve the sound quality.

3.3 Filtering

Filters are considered the most basic circuit in any signal processing used in almost every process. It removes the unwanted noise, echo, distortion, and allows the filtered data to pass through it. We will be discussing pass filters that allow only specific frequencies while rejecting others.

- Low Pass Filters :

  Low-pass filters allow the frequencies below the selected cut-off frequency level and cut the frequencies above the cut-off range.

- High Pass Filters :

  A high-pass filter is the opposite of a low-pass filter. It filters and passes the frequency, which is higher than the cut-off frequency range and attenuates the frequency lower than the cut-off range.

- Band Pass Filters :

After resampling of signals, band pass filter is applied to remove the extra noise and be considered the most ideal filter in signal processing. It attenuates the frequencies which are higher or lower than the cut off frequencies range and only passes the frequencies which fall within the cut-off range.

- Band Stop Filters :

  It is also known as a notch filter and opposite of band-pass filter. It leaves most of the frequencies unaltered and attenuates those within a specified range to very low levels.

3.4 Equalization

Equalizers are used to alter or adjust the frequency so that the sound spectrum frequency at the transmitter should match the sound spectrum's frequency at the receiver. Frequency ranges are being adjusted to high or low using low-pass filter, high-pass filter, band-pass filter. It removes the delay between different frequency components and gets the desired output.

3.5 Automatic Gain Control

It gives a constant output despite having various input signals. It shows the amount of gain or attenuation applied to the input signals to get the target input signal. If the input signal is higher than the target input, then AGC subtracts the gain, and if it is lower than the target input level then AGC adds the gain. Gain shows the loudness of the input of the channel, which controls the tone.

4 . Algorithm of the Code

4.1 Initialize the parameters :

- ‘Fs’ : Sampling Frequency (e.g. 8kHz)
- ‘channel’ : Number of audio channel (e.g. 1 for mono)
- ‘bits’ : Number of bits per sample (e.g. 16-bits CD-quality audio)
- ‘ID’ : Device ID audio input (e.g. -1 for default device)
- ‘duration’ : Duration of audio recording (e.g. 23 seconds)
- ‘filename’ : Name of the output WAV file (e.g. ‘myvoice.wav’)

4.2 Record Audio :

- Create an audio recorder object (‘recaudio’) with specified paramters (Fs,bits,channel).
- Start Recording .
- Record audio for the specified duration using ‘recordblocking’.
- Stop Recording.

4.3 Playback Audio :

- Play the recorded audio using the ‘play’ function.

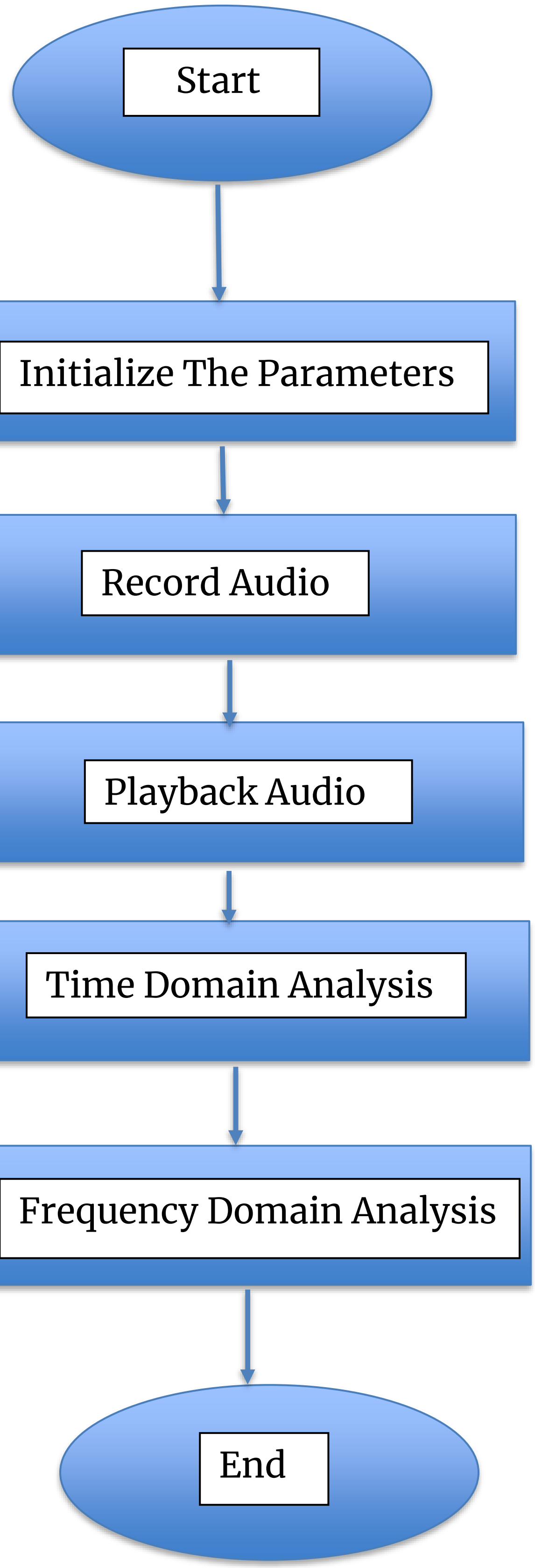
4.4 Time Domain Analysis :

- Generate a time vector ‘t’ corresponding to the duration of the recorded audio .
- Plot the recorded audio signal in the time domain using ‘plot’.

4.5 Frequency Domain Analysis :

- Calculate the FFT (‘fft’) of the recorded audio signal.
- Generate the frequency vector(‘f_0’) to centered at 0 Hz using the FFT shift (‘fftshift’) function.
- Plot the magnitude of the FFT result (‘abs(y_0)’) against the frequency vector ‘f_0’.

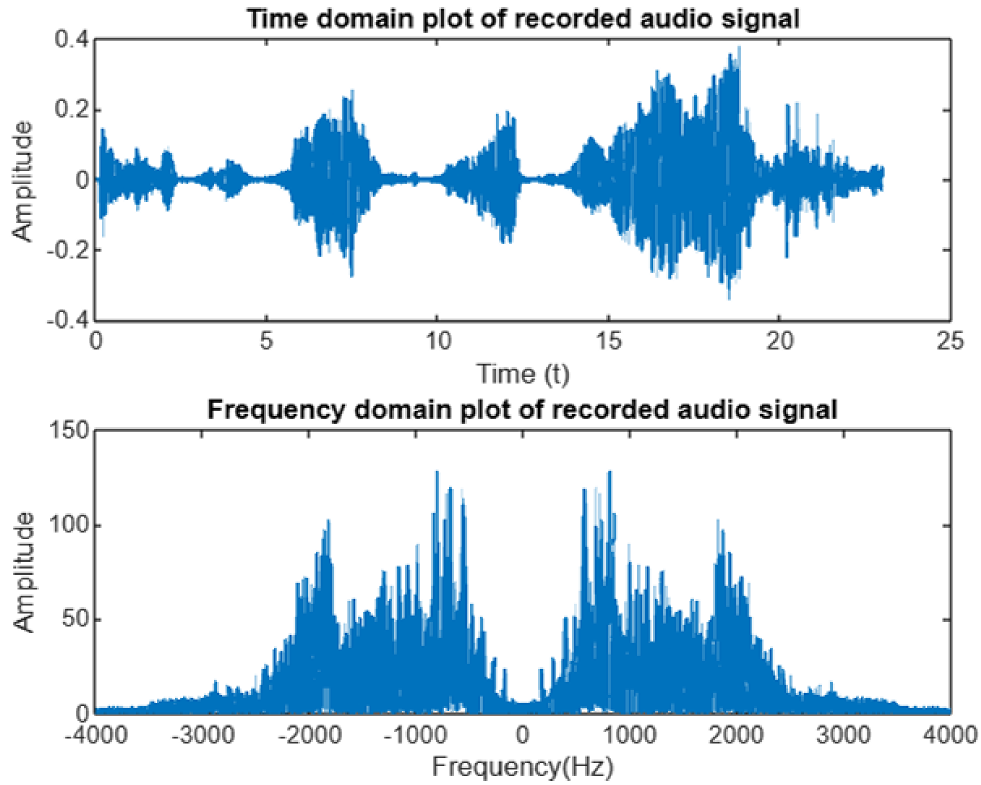
5. Flowchart


Start
Initialize The Parameters
Record Audio
Playback Audio
Time Domain Analysis
Frequency Domain Analysis
End

6. Matlab Code

```
clc;
close all;

Fs = 8e3;
channel = 1;
bits = 16;
ID = -1;

% Audio Recording Section
recaudio = audiorecorder(Fs,bits,channel); % data of audio processing
(object file)
disp('Recording Started');
duration = 23; % duration of audio
recordblocking(recaudio,duration); % it will record audio
disp('Recording Stopped');
p = play(recaudio);


mySpeech = getaudiodata(recaudio); % audio convert to data(numeric array)
sound(mySpeech,Fs,bits); % it will convert array to sound(electrical
signals)

% audio file save and write operation
filename = 'myvoice.wav'; % file banayi
audiowrite(filename,mySpeech,Fs); % from now recorded audio will store in
this file


% Time Domain Analysis
t = 0:1/Fs:(length(mySpeech)-1)/Fs; % duration of signal
subplot(2,1,1);
plot(t,mySpeech,'LineWidth',1.5);
xlabel('Time (t)');
ylabel('Amplitude');
title('Time domain plot of recorded audio signal');

% Frequency Domain Analysis

n = length(mySpeech);
f = 0:(n-1)*Fs/n; % k/n = f/fs;
y = fft(mySpeech,n);
f_0 = (-n/2:n/2-1).*(Fs/n); % frequency not centered at 0 so it will
centered it to 0
y_0 = fftshift(y); % shifting of fft
ay_0=abs(y_0);
subplot(2,1,2);
plot(f_0,ay_0,'LineWidth',1.5);
xlabel('Frequency(Hz)');
ylabel('Amplitude');
title('Frequency domain plot of recorded audio signal');
```

7. Experimental Results



8. Conclusion

From this project we have learned about the basics of audio signal processing and techniques of audio signal processing. Advancements in digital audio technology have propelled us to have very efficient and high-quality speech processing algorithms in place. These algorithms are applied in the process of recording, storing, and transmitting the audio content. Audio content brings lots of unwanted echo, interference and distortions that need to be removed to get the desired results in audio quality. It works on the principle of converting the audio signals between analog and digital formats, adjusting the frequency ranges, removal of unwanted noise and adding audio effects to get the smooth and flawless speech quality.