"IMPLEMENTATION OF BUTTERWORTH FILTER FOR NOISE DETECTION IN SPEECH SIGNAL"

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ABSTRACT: *Design and implementation of Butterworth filter for noise detection in speech signal removes noise from the audio signal. It is based Upon GUI (graphical user interface) in MATLAB. There are basically three files named first, filter file, filtered sound. Each of them consists of m-file and figure file. These are the programmable files containing the information about the filter and figure files are the way to analyse the given audio and enter the various filter related data. Now open these files in the MATLAB individually in the order as given above. Now run the first file. A graphical window will open as shown below. Upload the sound in the format .wave in the given window. Listen the sound which will appear to be noisy. Study three graphs of the noisy sound namely spectrogram power spectral density and amplitudes time. Analyse these graphs carefully to remove the noise. Click on the filter sound button than a new window will open named filter sound. Now choose the desired filter for de-noising and enter its various parameters namely type order, pass band frequency, stop band frequency, pass band ripple and stop band ripple. Now click on ok than a new window filtered sound will open. This window shows filtered sound along with the graphs and various details about the filters like transfer function, step, and impulse and frequency response. This application of MATLAB can be quite useful in present and future application.*

Keyword: Signal Processing, Butterworth filter

1. INTRODUCTION

Human voice is a commonly useful tool and it is the most important means to pass information to each other. Passing message by voice is the most important and effective way for mankind. Now with the development of the times, mankind has entered the information age, with the modern means of speech signal study, people can generate, transmit, store, access, and apply voice messaging more effectively, which has a very important significance for the promotion of social development. So in this thesis, we built a voice collection system, which can collect voice signal then Analyse the signal, then filter the noise by using various type of filters. MATLAB is a combination of two words matrix & laboratory, released by the U.S. Math works main face of scientific computing, visualization and interactive program designed in high-tech computing environment. Its numerical analysis, matrix computation, scientific data visualization and modelling as well as simulation of nonlinear dynamic systems, and many other powerful features are integrated in an easy way to use Windows environment, for scientific research, engineering design, and the need for effective numerical numerous scientific field provides a comprehensive solution, and largely combines with the traditional non-interactive programming languages (such as C, Fortran) in edit mode, on behalf of the current international advanced level of scientific computing software[1].

2. SAMPLING FREQUENCY

Sampling frequency means that pc collects how many speech samples per second, which not only is describing voice files" pitch, timbre but also can measure the sound card, sound files quality. The frequency is higher while the interval time of sampling is shorter, which means computer gets more sound sample data in a unit time, so the shape of sound wave forms will be more precise. According to the Nyquist theory, only when the frequency of samples is twice higher than the speech signal's peak frequency then we can convert digital signal to the origin sound. This means the sampling frequency is key to sound card collection, record and sound files restoration.

3. SAMPLING RESOLUTION

Sampling resolution is the value of sampling which is a parameter use to measure sound wave change. It is a binary bit when the sound card collecting, playing sounds file. Sampling resolution and sampling frequency are two important indexes to the audio interface which is also a standard to choose an audio interface. No matter how sampling frequency is, theoretically the sampling resolution decides the range of sound intensity. Increase 1 sampling resolution amounts to range increase 6dB. The more resolution has the more accurate the signal collect is.

4. SAMPLING THEORY

In the progress of analog to digital signal convert, when sampling frequency fs.max is twice higher than the highest frequency of the signal f max, fs.max $\geq$ 2 f.max, so the digital signal after sampling will keep the information of the origin signal. In normally application to guarantee the sampling frequency is 5 – 10 times to the signals highest frequency. This is called Nyquist theorem.

5. SPECTROGRAM

A spectrogram is a time-varying spectral representation (forming an image) that shows how the spectral density of a signal varies with time. Also known as spectral waterfalls, sonograms, voiceprints, or voice grams, spectrograms are used to identify phonetic sounds, to analyse the cries of animals; they were also used in many other fields including music, sonar/radar, processing, seismology, etc. The instrument that generates a spectrogram is called a spectrograph and is equivalent to a solography. The most common format is a graph with two geometric dimensions: the horizontal axis represents time, the vertical axis is frequency; a third dimension indicating the amplitude of a particular frequency at a particular time is represented by the intensity or colour of each point in the image. There are many variations of format: sometimes the vertical and horizontal axes are switched, so time runs up and down; sometimes the amplitude is represented as the height of a 3D surface instead of colour or intensity. The frequency and amplitude axes can be either linear or logarithmic, depending on what the graph is being used for. Audio would usually be represented with a logarithmic amplitude axis (probably in decibels, or dB), and frequency would be linear to emphasize harmonic relationships, or logarithmic to emphasize musical, tonal relationships.
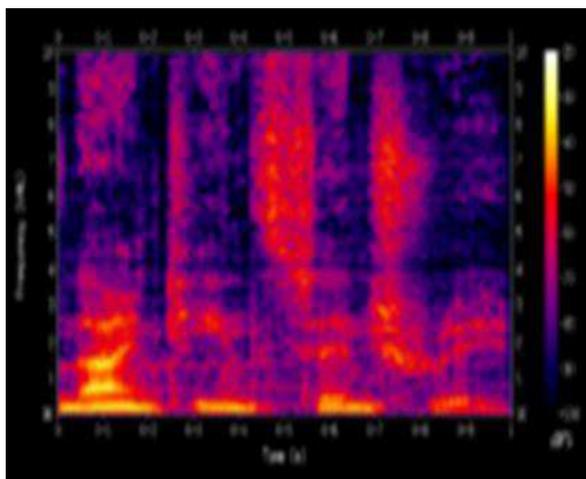

Figure 1: Spectrogram of Male Voice Saying "Tata" And "Ninetieth" Century

Spectrograms are usually created in one of two ways: approximated as a filter bank that results from a series of band-pass filters (this was the only way before the advent of modern digital signal processing), or calculated from the time signal using the short-time Fourier transform (STFT). These two methods actually form two different quadratic Time-Frequency Distributions, but are equivalent under some conditions.
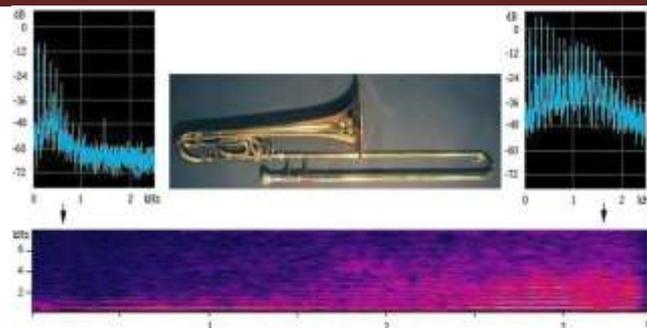

Figure 2: Sound Spectra of Musical Instrument

6. AUDIO FILE FORMAT

An audio file format is a file format for storing digital audio data on a computer system. This data can be stored uncompressed, or compressed to reduce the file size. It can be a raw bit stream, but it is usually a container format or an audio data format with defined storage layer.

7. TYPES OF FORMATS

It is important to distinguish between a file format and an audio codec. A codec performs the encoding and decoding of the raw audio data while the data itself is stored in a file with a specific audio file format. Although most audio file formats support only one type of audio data (created with an audio coder), a multimedia container format (as Matroska or AVI) may support multiple types of audio and video data.

There are three major groups of audio file formats:

Uncompressed audio formats, such as WAV, AIFF, AU or raw header-less PCM;

Formats with lossless compression, such as FLAC, Monkey's Audio (filename extension APE), WavPack (filename extension WV), TTA, ATRAC Advanced Lossless, Apple Lossless (filename extension m4a), MPEG-4 SLS, MPEG-4 ALS, MPEG-4 DST, Windows Media Audio Lossless (WMA Lossless), and Shorten (SHN).

Formats with lossy compression, such as MP3,Muse pack, AAC, ATRAC and Windows Media Audio Lossy (WMA lossy)).

List of format

Comparison of audio formats and List of codecs 3gp - multimedia container format can contain proprietary formats as AMR, AMR-WB or AMR-WB+, but also some open formats

ACT - ACT is a lossy ADPCM 8 Kbit/s compressed audio format recorded by most Chinese MP3 and MP4 players with a recording function, and voice recorders

AIFF – standard audio file format used by Apple. It could be considered the Apple equivalent of wav.

AAC– the Advanced Audio Coding format is based on the MPEG2 and MPEG4 standards. AAC files are usually ADTS or ADIF containers.

M4A – A proprietary version of AAC in MP4 with Digital Rights Management developed by Apple for use in music downloaded from their iTunes Music Store.

MMF - A Samsung audio format that is used in ringtones.

MP3 - MPEG Layer III Audio. Is the most common sound file format used today.

MPC - Muse pack or MPC (formerly known as MPEG plus,MPEG+ or MP+) is an open source lossy audio codec, specifically optimized for transparent compression of stereo audio at bitrates of 160–180 Kbit/s.

MSV – a Sony proprietary format for Memory Stick compressed voice files.

MXP4 – a Mucinous proprietary format allowing play of different versions (or skins) of the same song. It allows various interactivity scenarios between the artist and the end user.

WAV – Standard audio file container format used mainly in Windows PCs. Commonly used for storing uncompressed (PCM), CD-quality sound files, which means that they can be large in size—around 10 MB per minute. Wave files can also contain data encoded with a variety of (lossy) codecs to reduce the file size (for example the GSM or MP3 formats). Wav files use a RIFF structure.

WMA – The popular Windows Media Audio format owned by Microsoft. Designed with Digital Rights Management (DRM) abilities for copy protection.
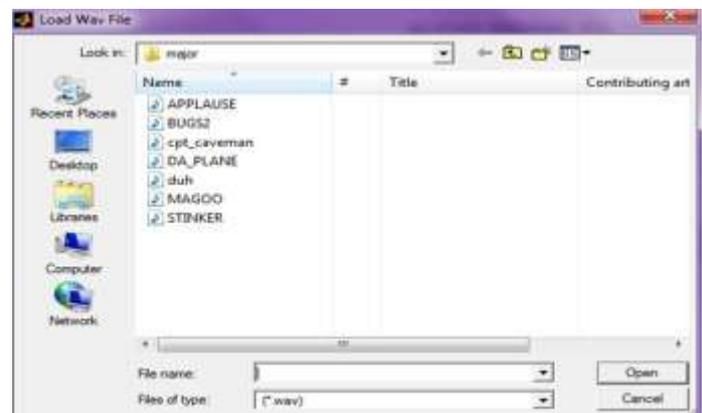
8. WORKING

1. There are basically three files named first, filter file, filtered sound. Each of them consists of m-file and figure file.
2. Now open these files in the MATLAB individually in the order as given above
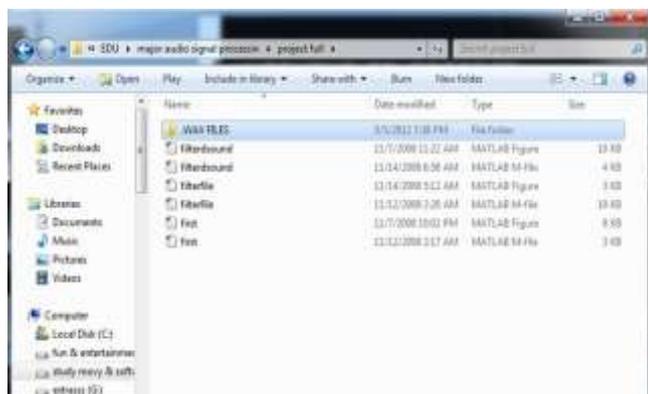
3. Now run the first file. A graphical window will open as shown below.
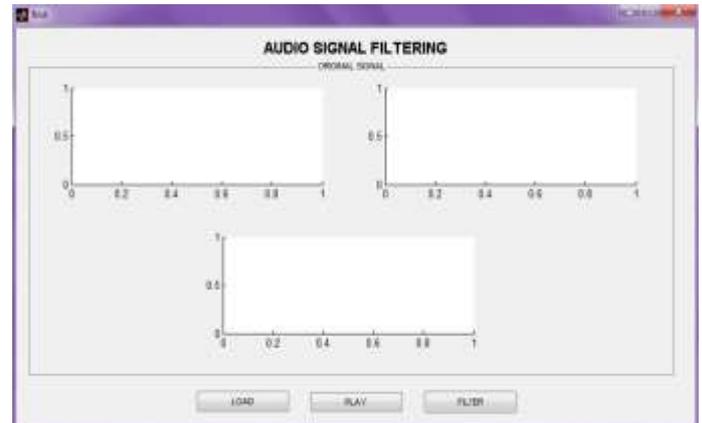
4) Upload the sound in the format .wav in the given window.


5. Listen the sound which will appear to be noisy.
6. Study three graphs of the noisy sound namely spectrogram power spectral density and amplitudes time. Analyse these graphs carefully to remove the noise.
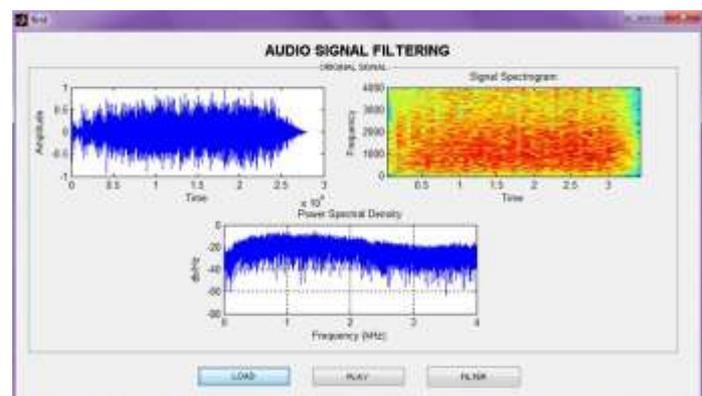
7. Click on the filter sound button than a new window will open named filter sound.
8. Now choose the desired filter for de-noising and enter its various parameters namely type , order, pass-band frequency, stop-band frequency, pass-band ripple and stop-band ripple.
9. Now click on ok than a new window filtered sound will open.

9. This window shows filtered sound along with the graphs and various details about the filters like transfer function, step, and impulse and frequency response.
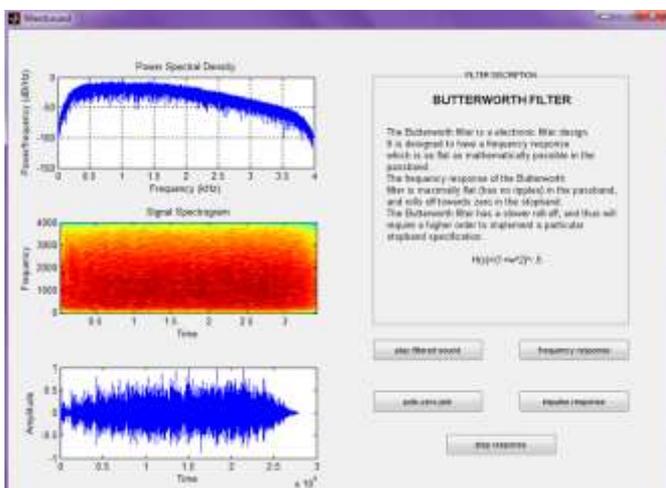
9. CONCLUSION

1) Using voice card and microphone in laptop which can be a sensor and collector in data collection system.
2) Using the strong data processing function in MATLAB, various voice signal analysis and process can be down well.
3) Using the filter in MATLAB, noise can be filtered and original sound will reappear.

9. REFERENCES

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