CHAPTER-IV

Summary

and

Conclusion

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Sound is mechanical energy. It can have wide frequency range depending upon nature of source such as human vocal cords, birdcalls, musical instruments, vibrating bodies, winds etc. This energy can be converted into electrical energy by using suitable transducer. Once sound is converted into electrical form it can be processed as desired. Processing of such electrical signal finds many applications in various fields such as communication, industries, laboratories, quality control, medical instrumentation etc.

Sound is longitudinal wave motion. It is characterized by the parameters like amplitude, frequency, phase, velocity etc. For human being, individual to individual quality of sound and its timbre are different which gives rise to some questions, such as how particular person can speak indifferent tones, what is exactly difference between quality of speech from individual to individual and even from day to day for same person, why particular song gives different effects when sung by different singers. Processing of sound signal may give information regarding above questions.

Thus present work has been proposed for processing of sound signals by microprocessor based systems which are commonly available. Initially it was assumed that human voice has low frequency. Male voice has frequency about 1000 Hz and female voice contains components about 1500 Hz. Hence in telecommunication system only 3.4 kHz band is sufficient for successful transmission of speech. Therefore 8085 based microprocessor system can be successfully used for processing of such signals. With ADC 0804 or ADC 0809 it is possible to digitize and store analog signal with sampling rate 7 kHz. The stored digital sequence can be processed with low level language.

However, for detailed analysis, not only fundamental components but also harmonics and their mixing are required to be studied. The microprocessor 8085 based system with limited memory and speed is not sufficient for such analysis

Hence work is divided into two parts. Low frequency signal analysis using microprocessor-based system and high frequency (within AF region), complex signal analysis using personal computer.

This dissertation consists of four chapters.

In the first chapter information about signals, their classification and characteristics is given. It also includes signal processing and information about sound.

Second chapter deals with μp-based system, PC hardware, software and interfacing. It includes 8085 based system with required interfacing support and ADC, DAC cards with required software. In this chapter information about sound card, computer system and MATLAB software is also given.

Third chapter covers the work actually performed and presented in the dissertation. In the first part, low frequency steady state signal analysis is given It includes sampling, quantizing, storage and retrieval of signal with different frequencies, spike suppression, wave shaping and waveform synthesis. Low frequency standard test signal, the sine wave with amplitude about 5V is used as input to ADC. With sampling rate about 5 kHz, signals of frequency 100 Hz, 200 Hz ---- 500 Hz are successfully digitized and stored using 8085 based system. The signal segments of 1.5 to 2 sec duration are stored.

The stored digital sequence is applied to DAC for reproduction of the signal. It is observed that such low frequency sinusoidal signals are stored and retrieved easily. By changing time delay the output frequency can be changed. The stored digital sequence can be subjected to spike suppression. Using assembly language programming it is possible to suppress high frequency spikes from the stored signal. The wave shaping is also possible with the help of assembly language programming.

Waveform synthesis is an important part in low frequency signal processing. A microprocessor based waveform generator system that generates various types of waveforms is explained in this section. The square, ramp, triangular as well arbitrary waveforms that find applications in designing and testing of industrial control circuits are generated with desired frequency and amplitude.

In second section of third chapter transient signal analysis using personal computer is explained, It includes acquisition of signal segments, digitization of signal at desired sampling rate, display of frequency spectrum, digital filtering, spectrum power estimation etc. For this analysis MATLAB software is used.

MATLAB has no provision for on line recording. Hence using sound card and recording software some speech and musical sound segments of different duration are stored in the form of wave files. The wave files are then read in to MATLAB. In MATLAB these wave files are converted into one-dimensional sequence and processed. Using plotting capability, for a given segment time domain response and frequency domain spectrum are obtained.

Frequency spectra show that for any speech sample the components are observed up to 6500 Hz. Above this frequency the amplitude of components are insignificant. In the band 0-6500 Hz the amplitudes of low frequency components are larger as compared to high frequency components. The low-frequency components up to 1000 Hz contain most of the energy. For male (adult) voice if all the components above 1kHz are suppressed the remaining lower band contains about 85% of total energy. Particularly the band 500 to 600 Hz contains maximum energy. The frequency band 1500 Hz to 2450 Hz is most sensitive for articulation or intelligibility. If all the components above 2450 Hz are suppressed the remaining band gives full articulation. Similarly if frequencies below 200 Hz are suppressed the remaining band gives full articulation. Within the range 200Hz to 2450 Hz suppression of any band significantly affects the articulation. For female voice low frequency range up to1000 Hz contains most of the energy and the frequency band 200 Hz to 3000 Hz is sensitive for articulation.

For musical segment energy is concentrated in still lower frequencies. Low frequency band up to 800 Hz contains about 90% of energy. Particularly the band

200 Hz to 300 Hz contains maximum energy. But these results purely depend on the musical instruments used and are particularly true in case of Indian music. For musical segments frequency range 0 to 7000 Hz is sensitive, and in this range suppression of any band can affect the quality of sound.

Thus it is concluded that for detailed analysis of speech or songs frequency band 200Hz to 3000Hz is sufficient. While for analysis of instrumental music wide audio band must be considered. The analysis is based on Fourier transform that is FFT. It is a powerful tool for spectrum analysis, filtering, energy calculation etc. We can precisely study what are the frequency components in particular signal segments, and what are their relative amplitudes.

However, this is not sufficient to study the quality of sound. It needs to study the fundamental components, harmonics and how these are combined. Hence in the spectral analysis we must consider time factor, that is at what instant and which fundamental and harmonics are combined. It needs further study which can be done, possibly by using the tool like 'wavelet transform'.