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Speech Signal Analysis using FFT and LPC

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***Abstract*— In speech processing there are different signal analysis techniques are used. In this paper two most commonly used signal analysis technique are used. First is Fast Fourier Transform(FFT) and second is Linear predictive Coding(LPC). These techniques are used to extract and compress some features of speech signal for further processing. In this paper five samples of single word is taken by same person. These samples are analyzed using FFT and LPC in matlab and spectra plus software. After analysis different parameters of samples are obtained for FFT and LPC spectrum individually.**

***Index Terms*— Linear Predictive coding, Fast Fourier transform, simulink.**

I. Introduction

In speech processing the speech signal should be first transformed and compressed for further processing. There are many signal analysis techniques are available which are used to extract the important features and compress the signal without losing any important information. Among the most important technique are FFT and LPC. FFT is used to convert the word signal into spectrum but FFT require only complex value. LPC used to compress the signal and we get different spectrum from original signal spectrum. The analysis of word signal can be done by finding the different parameter of FFT and LPC spectrum.

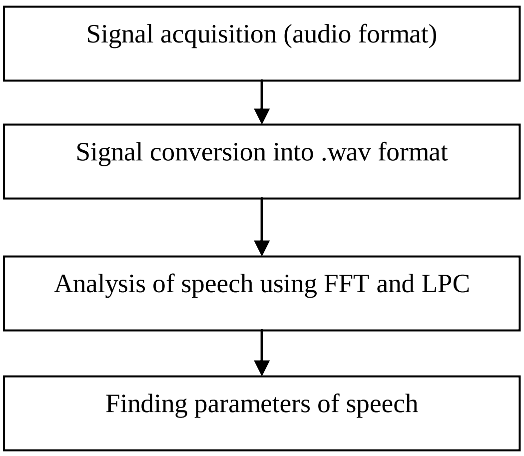

Fig.1 flow diagram of parametric analysis of speech

II. Signal Acquisition

Signal acquisition is the first step for the parametric analysis of speech. Following are the steps for signal acquisition.

1. Record the word signal in the audio format by the recorder.
2. Convert the audio format into .wav format by converter.
3. Covert the sampling frequency to 8 kHz of word signal.
4. Read the .wav file in matlab.
5. This file is used for analysis through FFT and LPC.
6. Spectra plus software is also used for analysis.

III. Signal Analysis

Spectrum analysis is complex process of decomposing the speech signal into similar parts. There are many spectrum analysis technique are available. In this paper mainly two techniques are discuss.

A. Fast Fourier Transform

FFT stands for finite Fourier transform it produce frequency spectrum which contains all the information about original signal, but in different form. It is frequency domain representation of signal. To find the FFT of signal using matlab following algorithm is used:

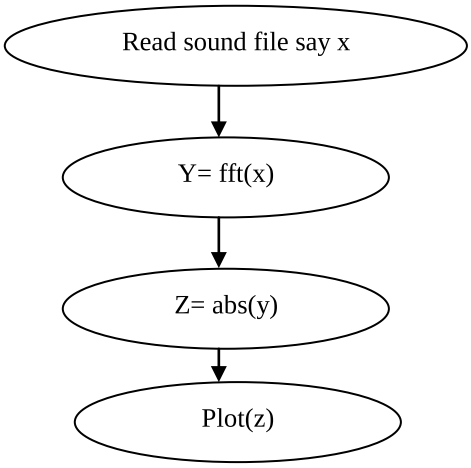

Fig.2 flow diagram of algorithm

Here we take word signal 'orange', the following figure shows the original signal representation.

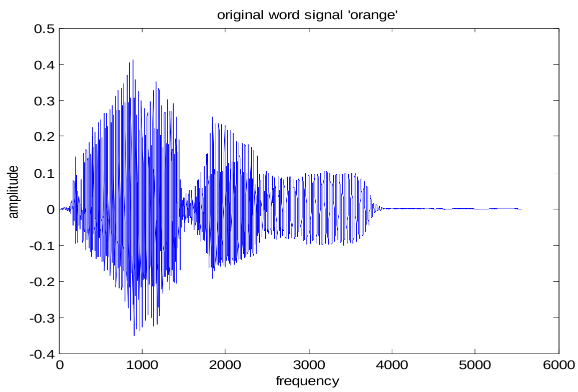

Fig.3 Original signal

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Using the above FFT algorithm we can find the FFT of original signal. Following figure shows the FFT spectrum of original signal.

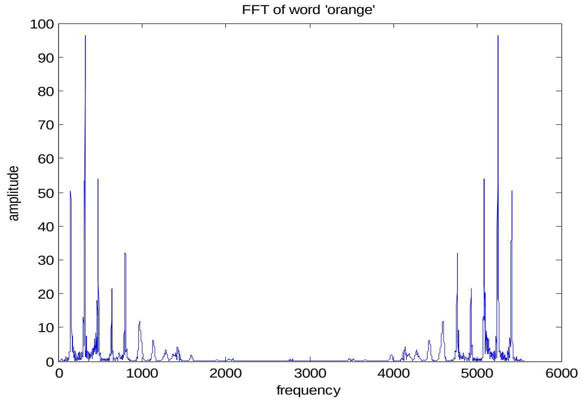

Fig. 4 FFT of original signal

B. LPC analysis

LPC [1], [2], [3], [4] is powerful feature extraction technique. It is used to compress the signal without any loss of information. To find the LPC spectrum of original signal simulink model in matlab is used. First the word signal is recorded in .wav format and its sampling frequency is converted into 8 kHz using sampling converter. This signal is given to simulink model. This model gives reflection coefficient, residual signal and LPC spectrum of signal. We use LPC spectrum for further analysis of speech signal. Following figure shows the simulink model for speech analysis in matlab.

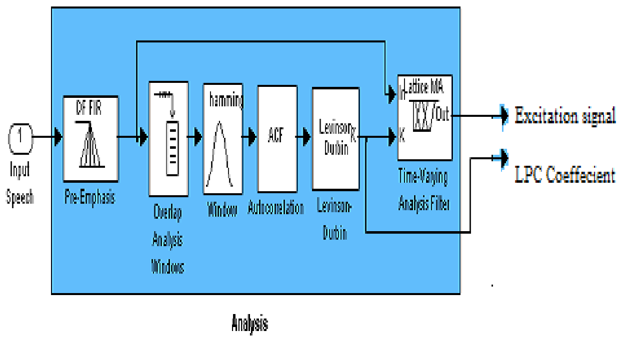

Fig. 5 simulink model of LPC

Following figure shows the LPC spectrum of word 'orange'.

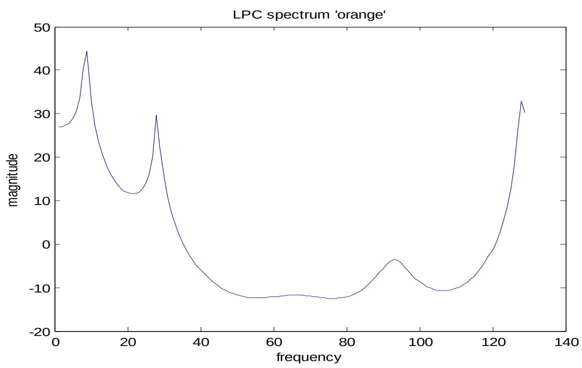

Fig. 6 LPC spectrum

IV. PARAMETERS OF SPEECH SIGNAL

After finding the FFT and LPC spectrum of signal next we find different parameters of signal using matlab and spectra plus software. For calculating the parameter of signal we take five sample of single signal word say s1, s2, s3, s4, s5. Here we take word 'orange' and the following table represents the various parameter of signal using spectrum analysis.

	S1	S2	S3	S4	S5	avg	std
mean	1.78	1.84	1.98	2.12	2.63	2.070	0.339
Median	0.25	0.31	0.57	0.48	0.98	0.51	0.288
Std	5.69	5.36	4.69	6.99	5.93	5.73	0.84
Max	96.36	90.39	50.49	130.64	85.5	90.67	28.59
Min	1.73	2.62	3.29	4.42	1.02	2.61	1.32
Rms	5.97	5.6	5.10	7.30	6.49	6.09	0.84
P. freq (Hz)	218.750	218.752	218.756	218.750	218.753	218.75	0.0025
P.amp (dB)	-54.28	-56.49	-58.01	-56.45	-50.08	-55.0	3.08
T. Pwr(dB)	-53.94	-55.71	-56.42	-55.19	-46.09	-53.47	4.22
THD (%)	11.64	16.39	41.69	35.40	18.86	24.79	13.00
THD +N (%)	8.85	14.72	15.97	8.64	8.54	11.34	3.68
IMD (%)	100.3	100.3	100.7	100.5	51.82	90.72	21.7
SNR (dB)	21.06	16.64	15.93	21.27	1.06	15.19	8.27

Table 1. Parameters value for FFT spectrum

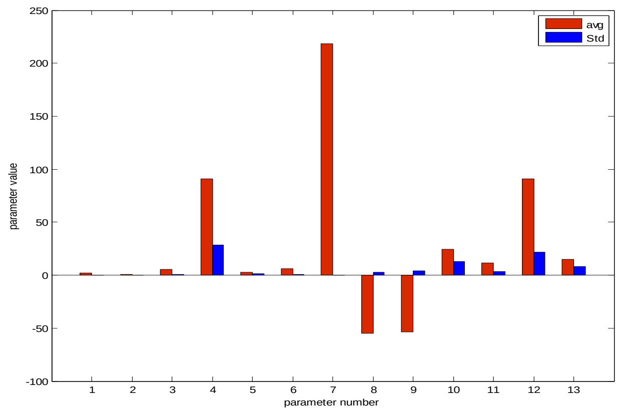

Fig. 7 graphical representation of parameters value

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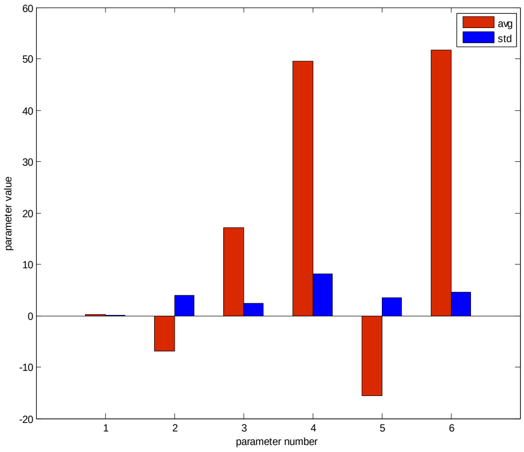
	S1	S2	S3	S4	S5	Avg	std
mean	0.211	0.15	0.21	0.23	0.15	0.19	0.03
Median	-7.16	-7.80	-9.6	-10.02	-0.12	-6.94	3.99
Std	14.91	15.34	18.30	20.80	16.4	17.15	2.42
Max	44.2	48.12	55.6	59.98	39.96	49.57	8.18
Min	-12.49	-13.3	-14.19	-16.67	-21.00	-15.53	3.43
Rms	48.12	50.16	53.78	58.6	47.81	51.69	4.53

Table 2. Parameters values using LPC Spectrum



Graph 2. Graphical representation of parameter value

V. Conclusion

In this paper we have use two signal analysis techniques FFT and LPC. Using these techniques the speech signal first analyzed and different parameter of speech signal is obtained. If we find parameters for different words then, these parameters values can be used in many software for word recognition.

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