Speech Analysis of Throat Infected People using DSO

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Abstract*— There are various ways of communication. Two people may communicate with each other through speech, gestures or graphical symbols. Man's most natural way of communication is through speech. Though writing seems to be important means of communication and written words, appear to be more efficient means of transmitting intelligence, the amount of intelligence exchanged by speech is beyond comparison. Considering the importance of speech ,the speech analysis is carried out .Speech analysis is used in innovative way to find out parameters like Amplitude, frequency, energy and power, when throat gets infected because of viral infections or due to any other cause. Speech quality changes which changes the parameters.

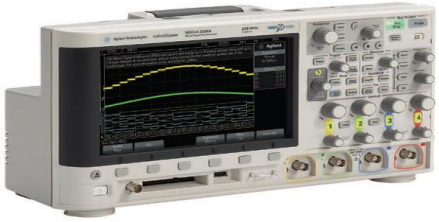
An experiment is carried out with the instrument DSO (Digital signal oscilloscope). Speech analysis of normal and infected throat that is performed without any treatment. The word "Hello" pronounced by 4 different persons in Normal health condition and when person when has infected throat. The main aim of this project is to compare signals of speech of normal throat person and infected throat person by analyzing Amplitude and frequency parameters which are obtained from DSO.The instrument set up consist of Input Module with Microphone (ST2108), DSO(Agilent Infiniti Vision 2000 X-Series oscilloscope). FFT software installed in computer for speech recording.

***Keywords*— DSO (Digital storage oscilloscope), FFT(Fast Fourier Transforms)**

I. EXPERIMENTAL SETUP

Instruments used:-

1. Digital storage oscilloscope
2. Computer
3. Mike
4. Audio power module
5. Sound pressure level meter



1. Digital storage oscilloscope

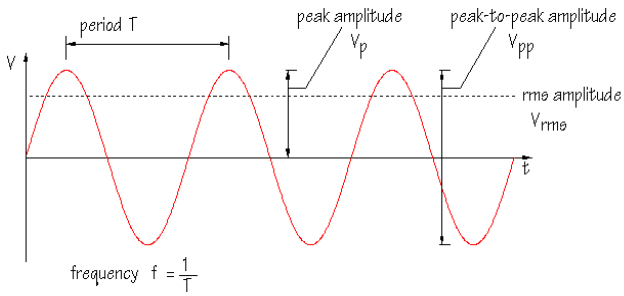
Agilent Infinii Vision 2000 X-Series

2. Audio power module

Audio input module

II. METHODOLOGY

The audio input module accepts the input signal from either a microphone or a cassette recorder which, it amplifies by a gain adjustable voltage amplifier. The input can be selected by pushbutton switch, provided on the front panel. A hand held microphone is provided with the audio input module which can be inserted in a 3.5 mm microphone jack. The module also has a built-in forth order low pass filter with cut-off frequency 3.4 KHz. This filter can be switched into the circuit after amplifier stage with the aid of a push- button. This limits the bandwidth of the modules output signal**.**



A Sine wave signal is defined by the equation:

$x(t) = A\sin(wpt) = A\sin(2\pi fpt) = A\sin(2\pi t/T)$

In signal processing, transformation is used to remove noise or make the signal more meaningful. Transformation is also used later in linear system analysis to make the descriptive equation easier to solve.

Comparisons between the signal of interest and one or more reference signals are often useful in themselves, but also at the heart of many transformations. In fact comparisons form the basis of all the transformation. a signal is compared with reference, or family of reference function. When a family of reference function is used you get a number of comparisons

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III. THEORY

Fast Fourier transforms

A fast Fourier transform is an algorithm to compute the discrete Fourier transform (DFT) and it's inverse. There are many different FFT algorithms involving a wide range of mathematics, from simple complex-number arithmetic to group theory and number theory.

. An FFT is a way to compute the complex result more quickly .FFTs are of great importance to a wide variety of applications, from digital signal processing and solving partial differential equations to algorithms for quick integers In practice, nearly all software and electronic devices that generate frequency spectra apply a fast Fourier transform (FFT), which is a specific mathematical approximation to the full integral solution. Formally stated, the FFT is a method for computing the discrete Fourier transform of a sampled signal.Any process that quantifies the various amounts (e.g. amplitudes, powers, intensities, or phases), versus frequency can be called spectrum analysis. The Fourier series is defined as

$X(t) = A_0/2 + \Sigma\, Am\, Cos(2\pi mt/T-\theta m) \qquad m=1,2,3\ldots..$

Where x(t) is a periodic function with a period of T, and the term,$A_0/2$, account for any nonzero mean value of the signal, this term is also known as DC term.If the signal has zero mean, as is often the case, then this term will be zero.

$A_0=2/T \int x\,(t)\,dt$

Digital Signal Processing

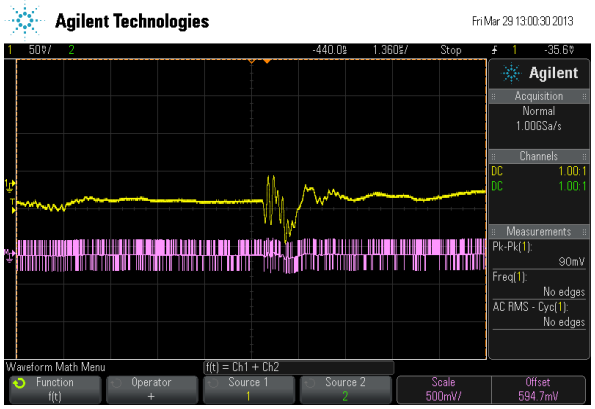
A signal is a physical quantity that is usually a function of time, position, pressure, etc. For example, the voltage output from a microphone represents sound pressure as a function of time.

The objective of signal processing is to transmit or store signals, to enhance desired signal components, and to extract useful information carried by the signals.

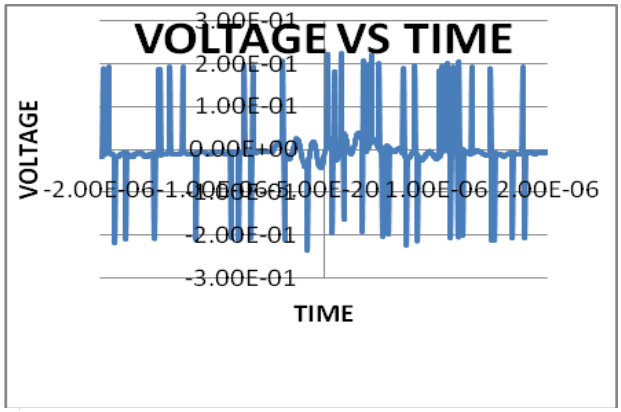
Signal Processing is a method of extracting information from the signal which in turn depends on the type of signal and the nature of information it carries

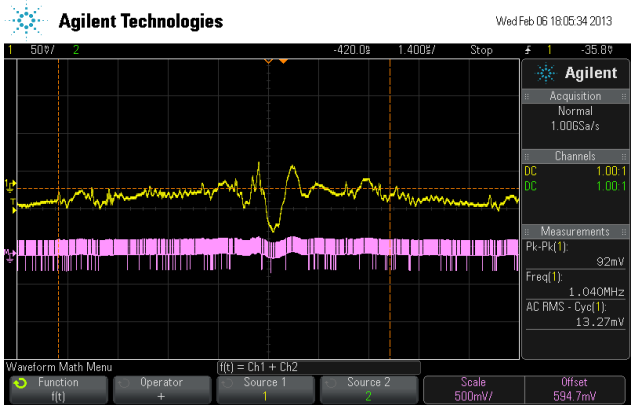
IV. OBSERVATIONS

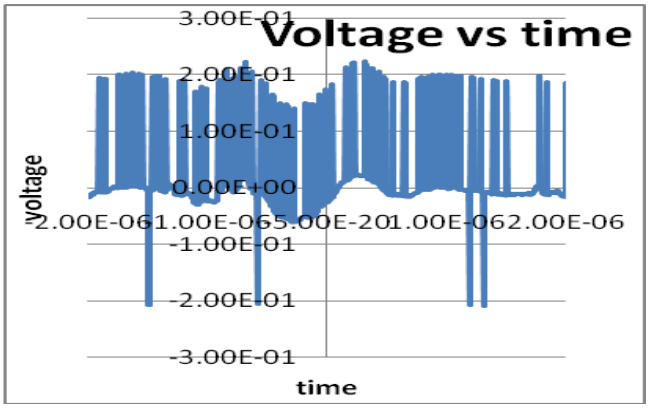
Normal Recording of speech



Speech recording when throat infected







V. OBSERVATION TABLE

Table I: Normal data for Person 1

Data collection of people	Normal Throat			
	Inte (dB)	PkPk (mV)	Freq (MHz)	Amp (mV)
Person 1	49.5	109	8.21	109
Person 2	53.7	94	743	57
Person 3	50.1	78	6.23	47
Person 4	60.4	92	2.191	92

Table II : Data for infected throat (Person 1)

Data collection of people	Infected Throat			
	Inte (dB)	PkPk (mV)	Freq (MHz)	Amp (mV)
Person1	67.7	101	3.02	59
Person2	76.8	157	2.35	46
Person3	60.4	92	1.04	35
Person4	52.3	181	1.04	16

Table III: Normal data for Person 2

Data collection of people	Normal Throat			
	Inte (dB)	PkPk (mV)	Freq (MHz)	Amp (mV)
Person 1	56.6	107	8.82	60
Person 2	51.3	115	7.63	65
Person 3	57.8	80	6.45	16
Person 4	51.5	86	8.06	20

Table IV : Data for infected throat (Person2)

data collection of person	Infected Throat			
	Inte (dB)	PkPk (mV)	Freq (MHz)	Amp (mV)
person 1	76.9	155	5.06	22
person 2	68.6	306	4.23	21
person 3	58.3	94	3.45	18
person 4	57.8	82	1.94	16

VI. CONCLUSIONS

- Frequency under normal condition and under throat infected condition show variations. It decreases with respect to previous condition.
- FFT of peak to peak amplitude decrease in magnitude. For individual case it show variations for normal condition and Throat infected condition.
- The major component of spectrum analysis is amplitude .it show sudden decrease in magnitude.
- In some cases variation in result occurs which show recording drawbacks

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