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Review Article

IDENTIFICATION OF INTERFERENCE IN VOCAL CORDS USING MICROPHONES AND VIBRATION SENSOR

B. Vaidianathan[1*], Dr.S. Arul Selvi[2], Dr.B. Karthik[3]

[1*]Research Scholar, Department of ECE, Bharath Institute of Higher Education and Research, Chennai. vaidia5000@gmail.com
[2]Associate Professor, Department of ECE, Bharath Institute of Higher Education and Research, Chennai. arulselvi2003@gmail.com
[3]Associate Professor, Department of ECE, Bharath Institute of Higher Education and Research, Chennai. Karthikguru33@gmail.com

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ABSTRACT

Several person suffer from various disorder like dumb, deaf, blind etc. From these problems we can put forward about the voice disorder. If the person needs to convey the message to another person with the help of voice signal. So for the dumb people who to needs to deliver the message can be predicted by the vibration signal through the movement of the vocal folds by the humans. The vibrations generated by the vocal folds can be measured using the Doppler radar technology. The human vocal folds can provide the information about the phonation and diagnosing voice disorder. In the before method the equipment is place near to the human throat so the essential information is not able to collect because of data loss it happens due to low level frequency. To overcome the above problem in this paper in this paper they proposes the detection of the vibration signal from long distance using radar sensor and the frequency in the range of 96-GHz. Algorithm is used which is the combination of both empirical mode decomposition (EMD) and auto correlation function (ACF). The EMD can neglect the noise in the noise in the radar signal and the ACF is for the extract of the vibration signal from human vocal folds. The ratio of both energy and the entropy can use to measure the voice activity in the radar signal.

Keywords: Signal to Noise Ratio, vibration sensor, vocal folds measurement, Threshold.

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INTRODUCTION

The human speech signal can be emerged from the vocal cord. So by predicting the vocal folds it useful for the speech production and to diagnose the voice disorder. The vibration prediction are highly complex and it is the three dimensional vibration. The vibration generated by the vocal folds can be measured using various devices. For example EGG electro glotto graphs (EGG), Laryngoscopes and high speed measuring equipment are also used for the human vocal folds vibration. The equipments not directly measure the vocal cord motions in non contact to the throat. It can be applied inside the throat it can cause pain and discomfort to the patients. To avoid the problem it can raised to the non contact measure of the vibration signal extraction. The microphone is employed it will monitor the acoustics signal from the human vocal folds. The signals can be easily affected by the noise present in the background which can totally affect the signal strength and the quality. So they decide to use other technology which can totally avoid the noise signal.

The EM electromagnetic radar signal is used in application of the phonation and speech signal. In later part of the year the electromagnetic signal can be applied in speech coding, speech synthesis. After few years later the Holzricher's can applied in the extraction of the speech signal by reducing the intensity of the radar signal. The Glottal electromagnetic micro power sensor can be applied in the tissues of the vocal cord at the time of speech. The electromagnetic sensor signal can monitors the vocal movement from the particular distance which can projected in another way it is non contact. The vibration signal which is monitor and is free from the noise signal. The movement of the vocal cord is measured and potential application in voice signal. Electromagnetic can execute three waves signal form the laryngeal region. The experimental results shows the EM signal can has the highest accuracy in predicting the voice signal generated from the vocal folds of the human. In the further research the continuous emission of the radar signal which can monitors the vibration signal and the acoustics signal. From the result the continuous radar waves signal have much consistency in the acoustics signal.

VIBRATION MEASUREMENT THEORY

The continuous emission of the radar system, transmit a signal from the transmitter. The tone signal can be expressed by the mathematical expression as

$$X_T(t) = P\cos(2\beta f_0 t + \phi(t)) \quad (1)$$

The terms in the above equation can shows about the various parameters f_0 is the frequency of radar signal extraction, P is the amplitude of the signal, $\phi(t)$ is the noise generated at oscillation. d is the signal transmit to the larynx form the certain distance. X(t) is the change in time period. The signal reaches antenna with a time period, the time taken to reach the antenna is determined by the distance and it is measured by the receiver. The received signal can be expressed by the equation

$$XY(t)=QPcos(2\beta f0t-4\mu d0\lambda 0-4\mu x(t)\lambda 0+\phi(t-2d0m))(2)$$

From the above equation the $\varphi(t-2d_0m)$ term shows the noise signal in the receiver antenna, $\lambda_0 = c/f_0$ the frequency is inversely proportional to the λ_0 so the time increases the frequency get decreases. The Q is the depth factor of the wave amplitude. The received signal and the oscillator signal is get mixed.

$$XY(t)=Pcos(2\beta f0t+\phi(t))\cdot[KAcos(2\beta f0t-4\beta d0\lambda 0-4\beta x(t)\lambda 0+\phi(t-2d0c))](3)$$

The combination of both received signal and the oscillator signal is get filtered by the low pass filter. The filter signal can be derived a equation and it is Represented as,

$$X(t)=PQ22cos(4\beta d0\lambda 0+4\beta x(t)\lambda 0+\Delta\phi(t))=KA22cos(4\beta x(t)\lambda 0+\theta(t)(4)$$

The term $\Delta\phi(t)$ can be expressed in full form as $\phi(t)-\phi(t-2d0c)$ which is the phase noise.

The radar signal whose amplitude of several mm, the intensity of the waves can be shown as λ_0, the frequency wavelength is about 96 GHz and the amplitude is 3.17 mm. The time period is less than the frequency so the small angle approximation is invalid.

The Fourier transform the time period of the signal can be viewed which is x(t). It is the combination of the vibration signal. The vibration signal can be generated by the vocal folds of the human. The signal can be consider as the acoustic signal it is expressed $x(t) = V\sin wt$.

The $\theta(t)$ is the total phase angle. $K_a(x)$ where k is the phase band of the function. The phase band can be further derived into several frequency harmonics. The frequency which is get multiple by the reference signal.

In the first stage of the equation which shows the DC term so it is get omitted. The first part does not provide the proper detection. From the equation 4, θ(t) can be expressed into two ways when θ(t) as the odd multiples of 2π the frequency is recovered maximum, when the θ(t) as the even multiple of 2π the frequency is fully get vanished. In the Doppler Effect the conditions often takes place which is said to be zero-point problem. The zero-point problem can takes in every quarter of the radar wavelength. In the mixture of the signal whose frequency is about 96 GHz in the receiver antenna the quadrature signal is to solve the problem. The output of the radar mixture signal is derived as

$$AI(t)=PI\cos(4\beta x(t)\lambda 0+\theta(t) \quad (5)$$

The final equation which can provide the vibration signal measured form the vocal track which can shows the accurate value.

RESEARCH METHODOLOGY

The frequency of the radar signal is configured to range 96-GHz. The non contact microphone and the contact vibration sensor can be used in the study which can detect the vibration signal generated from the vocal folds. The 16 bit data channel power lab is connected to the personal computer. Power Lab can be produce by the AD Instrument pvt., ltd which is in sydeny. The communication between the Power Lab and personal computer takes place by the USB cable. The power Lab is designed for the signal sampling the baseband signal is of 16 bit resolution. The ADC is used for the signal conversion from analog to digital. The above functions are takes place in the Lab chart software which is installed in the personal computer.

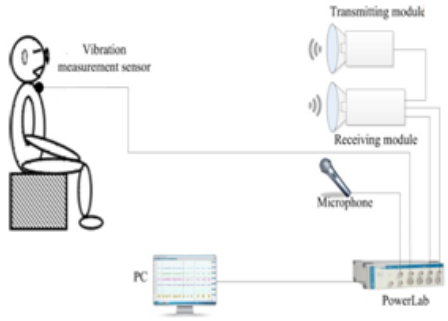

Fig. 1: Measurement of Vibration in the Vocal Track Using RADAR

The signal is get sampled at the frequency of about 16 kHz. The sampled signal is got saved in the lab view as WAV file format. The radar system is powered at the voltage of 12 V. The real time experiment is performed by consider a volunteer sitting in front of the RADAR system. The vocal track is maintained as the same height as the RADAR. The microphone is placed which is non contact and the vibration sensor can be placed contact to the vocal to the throat. The microphone and the radar can be separated to the distance in the range of about 1 to 12 m. The vibration sensors are fixed to the skin in the larynx portion. Laser is used to focus the beam reflected from the larynx portion.

RESULTS AND DISCUSSIONS

The human vocal fold vibration can be measured using the non contact radar signal. The microphone and the vibration sensor are employed and the comparison is made between them predict the vibration. The text is provided in the peaceful environment as the input signal. Several parameters have been measured such as the time domain, frequency spectra and spectrogram it can evaluate the noise in the background and the distribution of the frequency. To perform the operation two English sound is provided the vowel e and the "hello". The quality of the signal can be determined by the distance separation. For the high quality signal the distance is about 3m between the microphone and the vibration measurement sensor.

Fig. 2: Experiment Using RADAR

The experiment is performed into two parts. In the first part the experiment is takes place in the dark room and in the second part the experiment took place in the open ventilation. The microphone is placed which is non contact to the throat and the vibration sensor is placed in contact to the throat. The 25 old year male is considered to locate in front of the radar maintained at the particular height. The vocal track is placed at the position parallel to the RADAR system. The vibration sensor can be placed in the skin over the larynx region. The vibration sensor can acquire the signal generated at the time human vocal folds.

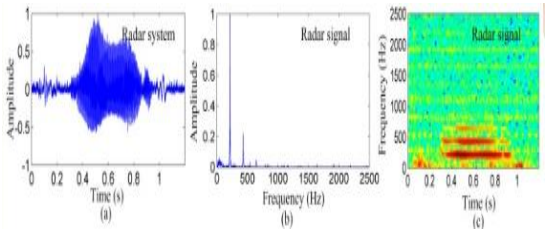

Fig. 3: Waveform of RADAR with Respect to Amplitude and frequency

The above graph shows the RADAR wave signal at time of generation of vibration signal when the human vocal folds. The waves are respect to the change in amplitude, time and frequency.

CONCLUSION

The vibration generated by the vocal folds of the human can be measured using the radar system. It consists of the microphone and the vibration sensor. The microphone is non contact to the skin and the vibration sensor is placed contact to the skin in the larynx portion. Comparing the vibration signal parallel measured using the microphone and the vibration sensor using this signal measured by the radar is determined. The frequency of the signal is about 96 GHz and the distance separation is about 3 m between the radar system and the person. It can measure the

time domain, frequency domain, spectrogram. The waves can be varied with respect to the above parameters. The parameters are needed in the case of the vocal fold vibration in the second and third harmonics. For the higher frequency it is get avoided. This method has the major advantage in the measurement of the vocal vibration because the vocal folds can be determined by the separated distance. It can avoid the noise generated in the background, ensured high directivity. The signal which is get measured during the vocal folds by the radar can get distributed in the frequency range less than the normal frequency.

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