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Performance on Speech Enhancement Objective Quality Measures Using Hybrid Wavelet Thresholding

S.China Venkateswarlu, A.Karthik, K. Naveen Kumar

Abstract: ***The quality of being easily understandable of the spe ech signals are very importantin communication and other speec h related systems. In order to improve these two in thespeech sign al, Speech improvement sets of computer instructions and devices are used so that itmay be better fully used by other speech proces sing setsof computer instructions. Most of the speech communica tion that requires atleast one microphone and the desired speech signal is usually contaminated by backgroundnoise and echo. As a result, the speech sign must be "cleaned" with advanced sign preparing devices before it is played out, transmitted, or put away. In this venture it has been investigated the required things and degree of upgrades in the field of discourse improvement utilizing discourse de-noising sets of PC directions announced in books with the fundamental intend to concentrate on the utilization of the window shape limits/rules in STSA based Speech Improvement process in which the sign destroyed by commotion is into edges and each part/segment is Windowed and the Windowed Speech pieces/parts zone connected to the Speech Improvement set of PC guidelines and the Improved Speech sign is modified in its time area. In general, the Speech Improvement methods make use theHam ming Window for this purpose. In this work an attempt has been made to study the effect of Window shape on the Speech. The Modified Improved thresholding is proposed by Asser Ghanbari and Mohammad Reza Karami and can be used like a hard thresholding limit with respect to the wavelet coefficients through and through worth progressively conspicuous than limit esteem and resembles an exponential capacity for the wavelet coefficients supreme worth not as much as edge esteem and is characterized.***

Index Terms: speech, windows, wavelet, quality, thresholding, speech enhancement. August

I. INTRODUCTION

The fundamental point of this work is to improve the nature of discourse. Discourse upgrade plans to improve discourse quality by utilizing different calculations.

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* Correspondence Author

Dr. S.China*, Venkateswarlu, Professor in Dept.of ECE, Institute of Aeronautical Engineering (IARE), Dundigal, Hyderabad, T.G, India.

A.Karthik, Assistant Professor in Dept.of ECE, Institute of Aeronautical Engineering (IARE), Dundigal, Hyderabad, T.G, India.

Dr. K. Naveen Kumar, Professor of ECE Department, Marri Laxman Reddy Institute of Technology and Management, Dundigal, Hyderabad, India.

The goal of upgrade is improvement in comprehensibility as well as by and large perceptual nature of debased discourse sign utilizing sound sign handling strategies. This part gives a prologue to discourse upgrade, its applications and normal wellsprings of commotion that corrupt discourse. The principle difficulties and issues identified with single channel upgrade propelled this work and a layout of the proposition is likewise portrayed.

Correspondence by means of discourse is one of the basic elements of people. People have shifted approaches to recover data from the outside world or to speak with one another and the three most significant wellsprings of data are discourse, pictures and composed content. For some reasons, discourse stands out as the most effective and advantageous one. Discourse passes on semantic substance, yet in addition imparts other helpful data like the state of mind of the speaker. Whenever speaker and audience are close to one another in a calm domain, correspondence is commonly simple and precise. In any case, at a separation or in an uproarious foundation, the audience's capacity to get endures (Douglas 2005). In numerous discourse correspondence frameworks the quality and comprehensibility of discourse is of most noteworthy significance for simplicity and precision of data trade. The discourse handling frameworks used to impart or store discourse is typically intended for a clamor free condition yet in a genuine domain, the nearness of foundation obstruction as added substance foundation and channel commotion definitely debases the presentation of these frameworks, causing off base data trade and audience weakness. Throughout the year, scientists have built up various techniques to improve discourse from the corrupted discourse. However, because of complexities of the discourse signal, reestablishing the ideal discourse signal from the blend of discourse and foundation commotion still represents a significant test in discourse preparing and correspondence framework look into.

II. SPEECH ENHANCEMENT

Discourse upgrade manages handling of uproarious discourse signals, going for improving their discernment by human or their right interpreting by machines (Berouti et al 1979).

Discourse upgrade calculations endeavor to improve the exhibition of correspondence frameworks when their information or yield sign are adulterated by commotion. The nearness of foundation clamor makes the quality and understandability of discourse debase. Here, the nature of discourse alludes how a speaker passes on an articulation and incorporates such traits like instinctive nature and speaker obviousness [2]. Understandability is worried about what the speaker had stated, that is, the importance or data content behind the words (Hu and Loizou 2007). In this way, a loud situation diminishes the speaker and audience member's capacity to convey. To decrease the effect of this issue discourse upgrade can be performed. It is normally hard to lessen commotion without twisting discourse and therefore, the exhibition of discourse improvement frameworks is restricted by the tradeoff between discourse contortion and clamor decrease (Boll 1979)[3]. Endeavors to accomplish higher quality as well as coherence of uproarious discourse may viably wind up improving execution of other discourse applications, for example, discourse coding/pressure and discourse acknowledgment, portable amplifiers, voice correspondence frameworks, etc. The objective of discourse upgrade shifts as per explicit applications, for example, to decrease audience exhaustion, to support the general discourse quality, to build comprehensibility and to improve the exhibition of the voice specialized gadget. Henceforth discourse improvement is important to maintain a strategic distance from the debasement of discourse quality and to defeat the impediments of human sound-related frameworks.

Common Sources Of Noise That Degrade Speech

For correspondence frameworks, two general destinations rely upon the idea of the clamor and regularly on the sign to commotion proportion (SNR) of the contorted discourse. With medium to high info SNR, decreasing the clamor level can create an abstractly characteristic discourse signals at a recipient or can acquire solid transmission [4]. For low SNR, the target could be to diminish the commotion level, while holding or expanding the understandability and decreasing the exhaustion brought about by overwhelming clamor for instance engine and road commotion. Figure.

Demonstrates the elements that influence the discourse signal during transmission at different stages by various clamor sources. Sources that corrupt discourse quality are uproarious condition during obtaining, foundation commotion, multi-speaker impact, boisterous transmission channel and flawed discourse generation. In the transmission side the impact of foundation commotion are included with the ideal sign and the sign from different speakers are treated as clamor for the ideal speaker. The sign with foundation commotion is transmitted through the channel where the transmission channel clamor is likewise included with the ideal sign.

The idea of the clamor is a significant factor in settling on a discourse upgrade strategy. Along these lines, a great model of commotion is significant for the presentation of discourse improvement framework and it is essential to examine how well a discourse upgrade calculation/model works with various sorts of clamor (Kamath and Loizou 2002). Clamor can be diverse dependent on different factual, otherworldly or spatial properties. In light of the nature and properties of the commotion sources, clamor can be named added substance foundation commotion, meddling speakers (discourse like commotion), motivation commotion, convolutive commotion, and multiplicative commotion [5]. All in all, it is progressively hard to manage non-stationary clamor, where there is no earlier information accessible about the attributes of commotion. Since non-stationary clamor is time differing, the customary strategy for evaluating the commotion from beginning interims by accepting no discourse sign isn't reasonable for estimation. Commotion types, which are comparative in transient, recurrence or spatial attributes to discourse, are likewise hard to expel or weaken. For example, Multi talker chatter holds a few attributes of discourse and represents an especially troublesome issue for a calculation planned to seclude discourse signal from the foundation commotion.

Need for the Proposed work

From the above survey, it is identified that the factors to be considered during the enhancement process is of residual noise and speech distortion. The quality and intelligibility of the speech signal has to be maintained. In this research, discrete wavelet transform with hybrid thresholding is proposed to enhance the noisy speech signal from additive background noise, which considers the drawbacks of the existing algorithms.

Methodology

A fundamental part of the thesis or thesis is the methodology. This is not the same as the "methods". The methodology describes the broad philosophical basis of the chosen research methods, including whether qualitative or quantitative methods are used or a combination of both, and why

III. METHODOLOGY OF EXISTING SPEECH ENHANCEMENT METHODS

Spectral Subtraction Algorithm

The commotion ruined discourse is handled by the Spectral Subtraction strategy to get prepared or upgraded discourse. Unearthly Subtraction is a well known recurrence area strategy to decrease the impact of added substance uncorrelated commotion in a sign. The commotion range is assessed, and refreshed, from the periods when the sign is missing and just the clamor is available. For rebuilding of the time-space signal, a gauge of the immediate size range is joined with the period of the uproarious sign, and after that changed by means of an Inverse Discrete Fourier Transform (IDFT) to the time area. On the off chance that is the discrete commotion adulterated information signal which is made out of the spotless discourse signal and the uncorrelated added substance clamor signal, at that point the loud sign can be spoken to as:

$$y(n) = s(n) + v(n) \tag{3.1}$$

Since the discourse is definitely not a stationary sign, the preparing is continued brief time premise (outline by edge).

$$y(n,k) = s(n,k) + v(n,k) \tag{3.2}$$

Where is the time record and is the edge file, is the casing. In the recurrence space, with their separate Fourier changes, the power range of the boisterous sign can be spoken to as:

$$P_{yy}(w,k) = P_{ss}(w,k) + P_{vv}(w,k) \tag{3.3}$$

$$|Y(w,k)|^2 = |S(w,k)|^2 + |V(w,k)|^2$$

Type equation here. (3.4)

Here $Y(w,k)$ is the DFT of $y(n,k)$ given by

$$Y(w,k) = \sum_{n=0}^{N-1} y(n,k) e^{-j\frac{2\pi wn}{N}} = |Y(w,k)| e^{j\varphi(w)}$$

Where $\varphi(w)$ the phase of the corrupted noisy signal and N is is the number of samples in the framed speech signal.

Thus from Eq.4.4 the estimation of clean speech signal can be given as

$$|\hat{S}(w,k)|^2 = |Y(w,k)|^2 - |\hat{V}(w,k)|^2 \tag{3.6}$$

Once the estimate of the clean speech is obtained in the spectral domain, the enhanced speech signal is obtained according to:

$$\hat{s}(n,k) = \mathrm{IDFT}\{|\hat{S}(w,k)| e^{j\varphi(w)}\} \tag{3.7}$$

Here, the stage data from the ruined sign is utilized to recreate the time space signal by taking the IDFT. One may sum up the system of otherworldly subtraction by supplanting the greatness squared of the DFT by some intensity of the extent. The example, '2', in Equation.4.6 can be supplanted by 'an' as given underneath:

$$|\hat{S}(w,k)|^a = |Y(w,k)|^a - |\hat{V}(w,k)|^a \tag{3.8}$$

To gauge the clamor, a strategy for exponential averaging is utilized to appraise the commotion. The edge by-outline update plan utilizing the exponential averaging technique is given beneath:

$$|\hat{V}(w,k)|^a = f(x) = \begin{cases} \mu|\hat{V}(w,k-1)|^a + (1-\mu)|\hat{V}(w,k)|^a, & \textit{for noise only} \\ |\hat{V}(w,k)|^a, & \textit{speech and noise} \end{cases}$$

B. Multiband Spectral Subtraction

This present reality clamor is for the most part hued. The hued clamor does not influence the discourse signal consistently over the whole range. The shaded commotion influences the discourse range diversely at different frequencies.

Least difficult strategy for disposing of foundation clamor is otherworldly subtraction. Multiband ghastly subtraction was proposed by Kamath [4](S. Kamath, and P. C. Loizou, "A multi-band unearthly subtraction strategy for upgrading discourse debased by shaded clamor," in Proceedings of Int. Conf. on Acoustics, Speech, and Signal Processing, Orlando, USA, May 2002, vol. 4, pp. 4160 4164.). It is hard for any discourse upgrade calculations to perform homogeneously over all commotion types. Therefore calculations are based on specific presumptions. Otherworldly subtraction calculation of discourse improvement is worked under the suspicion that the commotion is added substance and is uncorrelated with the sign which is valid for couple of sorts of clamor like foundation commotion and meddling clamor. Clamor adulterated discourse sign can be spoken to as:

$$y(n) = s(n) + v(n) \tag{3.10}$$

Where $s(n)$ is clean speech and $v(n)$ is noise. Since speech signal is non-stationary, it is split into smaller frames and is assumed to be quasi stationary which allows applying STFT for signal processing. The power spectrum of noise corrupted speech signal can be approximately estimated as:

$$|Y(k)|^2 \approx |S(k)|^2 + |V(k)|^2 \tag{3.11}$$

Where $S(k)$ and $V(k)$ are magnitude spectra of clean speech and noise respectively. Noise spectrum cannot be obtained directly. An estimate $V(k)$ is calculated from sine tapers spectrum during periods of silence. Estimate of clean speech is obtained as:

$$|\hat{S}(k)|^2 = |Y(k)|^2 - \alpha|\hat{V}(k)|^2 \tag{3.12}$$

Where α is an over-subtraction factor (weights) which is a function of the segmental SNR.

The algorithm re-adjusts the over-subtraction factor in each band based on SSNR. So, the estimate of the clean speech magnitude spectrum in the i^{th} Band is obtained by:

$$|\hat{S}_i(\omega)|^2 = \begin{cases} |Y_i(\omega)|^2 - \alpha_i \delta_i |\hat{D}_i(\omega)|^2, & \text{if } |\hat{S}_i(\omega)|^2 > 0 \; k_i < \omega < k_{i+1} \\ \beta |Y_i(\omega)|^2, & \textit{Otherwise} \end{cases}$$

Where k_i and k_{i+1} are the startand end frequency bins of theith frequency band,αi is the band specific over-subtraction factor of the ith Band, which is the function of SSNR of the ith frequency band. The SSNR of the ith frequency band can be calculated as

$$\mathrm{SSNR}_i(\omega) = \left(\frac{\sum_{\omega=k_i}^{k_{i+1}} |Y_i(\omega)|^2}{\sum_{\omega=k_i}^{k_{i+1}} |\hat{D}_i(\omega)|^2} \right) \tag{3.14}$$

The band specific over-subtractioncan be calculated, as

$$\alpha_i = \begin{cases} 5, & \mathit{if}\ \mathrm{SNR}_i \le -5 \\ 4 - \frac{3}{20}\mathrm{SNR}_i, & \text{if} -5 \le \mathrm{SNR}_i \le 20 \\ 1, & \text{if } \mathrm{SNR}_i > 20 \end{cases} \tag{3.15}$$

The δ_i is an additional band subtraction factor that can be individually set for each frequency band to customize the noise removal process and provide an additional degree of control over the noise subtraction level in each band.

The values of δ_i is empirically calculated and set to

$$\delta_i = \begin{cases} 1, & f_i \le 1\,\mathrm{kHz} \\ 2.5, & 1\mathrm{kHz} < f_i \le \frac{f_s}{2} - 2\mathrm{kHz} \\ 1.5, & f_i > \frac{f_s}{2} - 2\mathrm{kHz} \end{cases} \tag{3.16}$$

Here is the upper bound recurrence of the Band and is the examining recurrence. The inspiration for utilizing littler estimations of for the low recurrence groups is to limit discourse contortion, since the majority of the discourse vitality is available in the lower frequencies. The two elements, alphai and can be balanced for each band for various discourse conditions to show signs of improvement discourse quality. As this present reality clamor is exceptionally irregular in nature, improvement in the MBSS calculation for decrease of WGN is important. The MBSS calculation is found to perform superior to other subtractive-type calculations.

C. Wiener filter Algorithm

The Wiener channel, named after Nobert Wiener, targets assessing an obscure irregular sign by sifting a loud perception of the sign. It has a wide assortment of uses in commotion decrease, framework distinguishing proof, de-convolution and sign recognition. For example, the Wiener channel can be utilized to de-clamor sound sign, similar to discourse, or to expel commotion from an image. The coefficients of a Wiener channel are determined to limit the normal squared separation between the channel yield and an ideal sign. In its fundamental structure, the Wiener hypotheses accept that the sign are stationary procedures. Be that as it may, if the channel coefficients are occasionally recalculated for each square of N sign examples then the channel adjusts to the normal attributes of the sign inside the squares and progresses toward becoming square versatile. A square versatile (or portion versatile) channel can be utilized for sign, for example, discourse and picture that might be considered practically stationary over a generally little square of tests.

D. Wiener Filter for Additive Noise Reduction

The Wiener filter gives the MMSE estimate of the short-time Fourier transform (STFT) whereas the spectral subtraction obtains the MMSE estimate of the short-time spectral magnitude without changing the phase. Thus, the condition for minimum mean square error is,

$$H(\omega) = \frac{P_{ss}(\omega)}{P_{yy}(\omega)} = \frac{P_{ss}(\omega)}{P_{ss}(\omega) + P_{dd}(\omega)} \quad (3.17)$$

Here $P_{ss}(\omega)$ and $P_{dd}(\omega)$ are the signal and noise power spectrum, respectively

$$H_{wiener}(\omega) = \frac{P_{ss}(\omega)}{P_{yy}(\omega)} \quad (3.18)$$

$$= \frac{P_{yy}(\omega) - P_{dd}(\omega)}{}$$

The enhanced signal is estimated as,

$$\widehat{P}_s(w,k) = H_{wiener}(w) P_y(w,k) \quad (3.19)$$

$$|\hat{s}[n]| = \mathrm{IFFT}\left[\sqrt{\widehat{P}_s(w,k)}\right] \quad (3.20)$$

The estimate of the clean speech spectral magnitude is recombined with the noisy phase giving an estimate of the ehanced speech signal as

$$\hat{s}[n] = |\hat{s}[n]| \angle y[n] \quad (3.21)$$

IV. METHODOLOGY OF PROPOSED SPEECH ENHANCEMENT METHOD

A. Wavelet Hybrid Thresholding

In customary de-noising systems, channels and Short time Fourier change are not all that great for discourse signal de-noising. Wavelet thresholding de-noising systems give another approach to decrease foundation commotion in discourse signal.

In the course of recent decades there is gigantic increment in the degree of encompassing ecological clamor. This has been because of development of innovation. Commotion is included by different elements like loud motors, overwhelming machines, siphons, vehicles, over boisterous phone channel or utilizing radio specialized gadget in an air ship cockpit. For example in hands free discourse correspondence conditions circumstance happens that discourse is superposed by foundation clamor. The wavelet change has turned into a useful asset of sign examination and is broadly utilized in numerous applications which incorporate sign discovery and de-noising. The wavelet de-noising procedure is called thresholding and it is a nonlinear calculation. The primary center is to improve wavelet edge technique which is improved for discourse signal de-noising. De-noising of any discourse sign can be ordered in to two sections: (i)Traditional filtering method

(ii) Wavelet de-noising method

B. Wavelet Thresholding Techniques

The reason for the thresholding strategy is to dispose of or stifle little worth wavelet coefficients which primarily speak to the clamor content in this manner the decision of edge levels and thresholding systems are basic to the presentation of wavelet de-noising procedures. A few techniques have been proposed in the writing for thresholding the wavelet coefficients [20-24]. The thresholding methods analyzed in this paper are depicted beneath

. Let $\{Z_{j,k}\}$ be the wavelet coefficients of (w) , let $\{S_{j,k}\}$ be the wavelet coefficients of logS(w) and let $\{n_{j,k}\}$ be the wavelet coefficients of η(w). Then by the linearity of the discrete wavelet transform, (8) is transformed to:

$$Z_{j,k} = S_{j,k} + \eta_{j,k} \quad (3.26)$$

Where the subscript j shows the scale and the subscript k demonstrates the wavelet coefficient. Since the clamor is about Gaussian, the standard thresholding capacities utilized in the wavelet based upgrade frameworks are hard and delicate thresholding capacities which we survey before presenting another thresholding capacity that offers improved execution for discourse signal.

Hard thresholding sets to zero any component whose supreme worth is lower than the limit. In Soft thresholding, the components whose total qualities are lower than the edge are first set to zero and afterward contract the nonzero coefficients towards zero. In these procedures, T is the limit worth and δ is the thresholding capacity.

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Hard Thresholding (HT)

The hard thresholding function is defined for threshold T as:

$$\delta_H(z,T)=\begin{cases} z, & \textit{if } |z|\geq T \\ 0, & \textit{otherwise}\end{cases} \qquad (3.27)$$

Soft Thresholding (ST)

The soft thresholding function is defined as:

$$\delta_S(z,T)=\begin{cases} z-T, & \text{if } z\geq T \\ 0, & \text{if } |z|<T \\ z+T, & \text{if } z\leq -T\end{cases} \qquad (3.28)$$

The Modified Improved Thresholding (MIT)

The Modified Improved thresholding [13] is proposed by Asser Ghanbari and Mohammad Reza Karami and can be utilized like a hard thresholding capacity for the wavelet coefficients outright worth more prominent than limit esteem and resembles an exponential capacity for the wavelet coefficients supreme worth not as much as edge esteem and is characterized [13] as:

$$\delta_{MIT}(z,T)=\begin{cases} z=z & \text{if } |z|\geq T \\ \operatorname{sign}(z)\dfrac{T\left\{\exp\left(\gamma_1\frac{|z|}{T}\right)-1\right\}}{\exp(\gamma_1)-1} & \text{if } |z|<T\end{cases} \qquad (3.29)$$

Here, the factor γ_1 is important and for this work $\gamma_1=3$ is used in order to have better performance.

The Modified Sigmoid function Thresholding (MST):

Modified Sigmoid function thresholding proposed by Ting-Hua.Yi et al. [14] attempts to address the deficiency of hard and soft thresholding de-noising methods and is given as:

$$\delta_{MST}(z,T)=\begin{cases} (z-T)-T\left[\dfrac{2}{1+e^{\beta\frac{(z-T)}{T}}}-1\right] & \text{if } z\geq T \\ 0 & \text{if } |z|<T \\ (z+T)-T\left[\dfrac{2}{1+e^{\beta\frac{(z+T)}{T}}}-1\right] & \text{if } z\leq -T\end{cases} \qquad (3.30)$$

In this function, the factor β is important and for this work $\beta=4.5$ is used in order to have better performance.

C. Wavelet Threshold Methods

The Universal Threshold Method

The fundamental issue in wavelet de-noising is deciding a suitable edge level T. For zero-mean, regularly disseminated commotion with difference, Donoho and Johnstone proposed the purported all inclusive limit given by $T=\sigma\sqrt{2\log N}$. It is straightforward since it doesn't rely upon the information, yet on the clamor difference, and functions admirably for uncorrelated commotion. On the off chance that the commotion is stationary and hued, the fluctuation of the clamor wavelet coefficients will be diverse for each scale in the wavelet disintegration [24]. Subsequently, scale-subordinate thresholding was proposed to represent the various changes of the commotion wavelet coefficients in each scale. In this way the level-subordinate changes of the commotion wavelet coefficients were evaluated by [25].

$$\operatorname{var}(n_{j,k})=\sigma_j^2\equiv\frac{1}{N}\sum_{k=0}^{N-1}S(k)\left|H_j(k)\right|^2 \qquad (3.31)$$

Where $H_j(k)$ is the frequency response of the length N periodized wavelet filter of level j, and S(k) is the Fourier transform of the autocorrelation function $r_{\eta\eta}$ of the noise □(w) which is approximated by [25]

$$r_{\eta\eta}=\begin{cases} \sigma_\eta^2\left(1-\dfrac{|i|}{L+1}\right), & \text{if } |i|\leq L+1 \\ 0, & \text{otherwise}\end{cases} \qquad (3.32)$$

Where L is the number of tapers, and σ_η^2 is the variance of (w).

In scale-dependent universal thresholding, the threshold T at each scale j is selected as$T=\sigma_j\sqrt{2\log N}$. The wavelet coefficients $z_{j,k}$, can be thresholded at each level j using either $\delta_H(\dots)$ or $\delta_S(\dots)$ etc.

$$\hat{z}_{j,k}=\begin{cases}\delta(z_{j,k},T), & \text{if } 1\leq j\leq q_0 \\ z_{j,k}, & j>q_0\end{cases} \text{-------} \qquad (3.33)$$

Where q_0 is, some specified coarse resolution level.

D. SURE Method

The all-inclusive thresholding strategy will in general utilize a high limit level, and as a rule it over smooths the boisterous sign [30]. In this work likewise the creators saw that the general thresholding strategy shows extremely mediocre execution when connected to assess the discourse quality target measures. In [30], Donoho and Johnstone demonstrated that the Stein's fair hazard estimator (SURE) could be utilized as the fair-minded gauge of the MSE for the delicate thresholding plan. Johnstone and Silverman [31] later summed up this plan to the instance of hued clamor, and demonstrated that the SURE strategy can likewise be utilized within the sight of associated commotion and the present work is kept just SURE edge technique.The Stein's unbiased estimate of the risk for a specific threshold T and input signal $x=\{x_i\}_{i=1}^{N}$ using the soft thresholding function is given by [31]

$$\hat{U}(x,T)=\widehat{\sigma^2}N+\sum_{i=1}^{N}\{\min(x_i^2,T^2)-$$
2σ2Ixi≤T (3.34) (3.34)

Where σ^2 is the variance of the noise and I is the indicator function $I(.)=1$ if $|x_i|\leq T$ and $I(.)=0$ if $|x_i|>T$.

$$T=\arg\min_{0\leq T\leq\sigma\sqrt{2\log N}}\hat{U}(x,T)\text{--------} \qquad (3.35)$$

For level-dependent thresholding, the noise variance σ_j^2 for level j can be obtained using the median absolute deviation (MAD) from zero estimates [31]:

$$\hat{\sigma}_j = \frac{MAD(Z_{j,k})}{0.6745} \text{-----------} \quad (3.36)$$

Where 0.6745 is a normalization factor. The operator MAD picks out the median of the absolute values of all the wavelet coefficients $Z_{j,k}$, at resolution level j. As the universal threshold method gives high threshold value [5], in this study the SURE method is considered only.

E. Advantage of De-noising using Wavelet vs Other Methods

The principle favorable position of wavelet premise is that they regardless of having sporadic shape can splendidly remake capacities with direct and higher request polynomial shapes, for example, rect, triangle, second request polynomials, and so on. Note that Fourier premise neglect to do as such, as if there should arise an occurrence of popular case of rect work at the edges. Thus, wavelets can de-commotion the specific flag obviously better than regular channels that depend on Fourier change structure and that don't pursue the arithmetical principles obeyed by the wavelets.

Wavelet is best for non-stationary sign examination, here unique sign join with mother wavelet sign and after that makes performance, implies Mother wavelet sign make Zoom of essential sign, at that point examination, so result is coming best in contrast with other strategy.

V. SIMULATION RESULTS

In this section the tool used as an approach for the proposed idea is discussed in brief, also the input to the project is mentioned apart from that the output obtained is also shown.

A. Softwares used

We are using MATLAB R2014a programming. MATLAB (network research focus) is a multi-perspective numerical figuring condition and selective programming language made by MathWorks. Cleve Moler, the head of the product designing division at the University of New Mexico, started making MATLAB in the late 1970s. He arranged it to give his understudies access to LINPACK and EISPACK without them learning FORTRAN. It after a short time spread to various universities and found a strong gathering of observers inside the associated number juggling system. Jack Little, a modeler, was exhibited to it during a visit Moler made to Stanford University in 1983. Seeing its business potential, he joined with Moler and Steve Bangert. They patched up MATLAB in C and set up MathWorks in 1984 to continue with its headway. These revamped libraries were known as JACKPAC. In 2000, MATLAB was altered to use a more forward-thinking set of libraries for system control, LAPACK. MATLAB was first gotten by investigators and specialists in control fabricating, Little's quality, yet quickly spread to various spaces. It is as of now furthermore used in guidance, explicitly the teaching of straight factor based math, numerical examination, and is standard among analysts related with picture taking care of. MATLAB is a programming stage arranged unequivocally for creators and scientists.

The core of MATLAB is the MATLAB language, a network based language permitting the most common articulation of computational arithmetic. Utilizing MATLAB, we can examine information, create calculations, make models and applications. The language, applications, and implicit math capacities empowers us to rapidly investigate various ways to deal with land at an answer. MATLAB gives us a chance to take our thoughts from research to creation by conveying to big business applications and installed gadgets, just as incorporating with Simulink and Model-Based Design.

B. Praat

Praat is a free PC programming group for the legitimate assessment of talk in phonetics.[1] It was arranged, and continues being made, by Paul Boersma and David Weenink of the University of Amsterdam. It can continue running on a wide extent of working structures, including various variations of Unix, Linux, Mac and Microsoft Windows (2000, XP, Vista, 7, 8, 10). The program underpins discourse amalgamation, including articulatory blend. 'Praat' is a PC program with which phoneticians can investigate, combine, and control discourse, and make top notch pictures for articles and proposals. It has capacities for discourse examination, discourse amalgamation, learning calculations, marking and division, discourse control, listening tests, and the sky is the limit from there.

C. Audacity

Dauntlessness is a free and open-source propelled sound article supervisor and recording application programming, available for Windows, mac OS/OS X and Unix-like working structures. Dauntlessness was started in the fall of 1999 by Dominic Mazzoni and Roger Dannenberg at Carnegie Mellon University and was released on May 28, 2000 as variation 0.8. As of October 10, 2011, it was the eleventh most noticeable download from Source Forge, with 76.5 million downloads. Daringness won the Source Forge 2007 and 2009 Community Choice Award for Best Project for Multimedia. In March 2015 encouraging was moved to Foss Hub.Daringness is a free, simple to-utilize, multi-track sound proofreader and recorder for Windows, Mac OS X, GNU/Linux and other working frameworks. The interface is converted into numerous dialects. We can utilize Audacity to:

1. Record live sound.
2. Record PC playback on any Windows Vista or later machine.
3. Convert tapes and records into advanced chronicles or CDs.
4. Edit WAV, AIFF, FLAC, MP2, MP3 or Ogg Vorbis sound documents.
5. AC3, M4A/M4R (AAC), WMA and different arrangements upheld utilizing discretionary libraries.
6. Cut, duplicate, graft or combine sounds.
7. Numerous impacts including change the speed or pitch of an account.
8. Write your very own module impacts with Nyquist.

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And the sky is the limit from there!

D. Inputs to the peoject module

There are two inputs to our project module:
Clean Speech
Noisy Speech
Clean Speech

The spotless discourse has been gathered from Speech Corpus. A discourse corpus (or spoken corpus) is a database of discourse sound documents and content interpretations. In Speech innovation, discourse corpora are utilized, in addition to other things, to make acoustic models (which would then be able to be utilized with a discourse acknowledgment motor). In Linguistics, verbally expressed corpora are utilized to do examination into Phonetic, Conversation performance, Dialectology and different fields. There are aggregate of 30 clean discourse sign removed from the discourse corpus information base out of which the venture has been tried with few clean discourse signals.

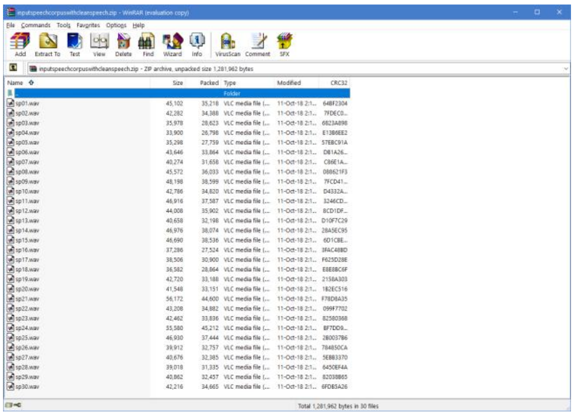

Figure 5.1: Clean Speech Database from Speech Corpus

The overall analysis of all of these signals has been carried out and presented below. The analysis has been carried out in the field of the signal's intensity, formants, pitch, pulses, and harmonics by using software called "praat". Below is the analysis of one of the clean speech signal "sp01" taken from the speech corpus database using the "praat" software:

We have carried out the entire analysis for all the clean speech signals obtained from speech corpus database, in terms of their spectrum, pitch, intensity, formants, and pulses using "Praat" software as shown below:

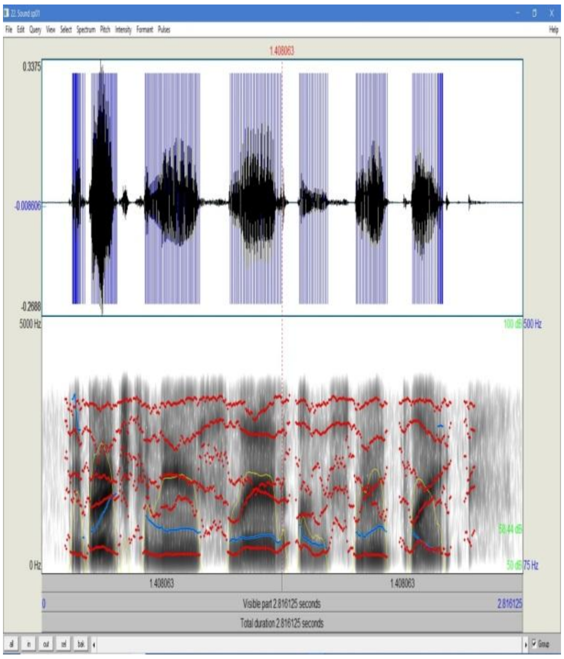

Figure 5.2: The Complete Analysis of the Clean Speech Signal Sp01 In Terms of Its Spectrum, Pitch, Intensity, Formants, Pulses

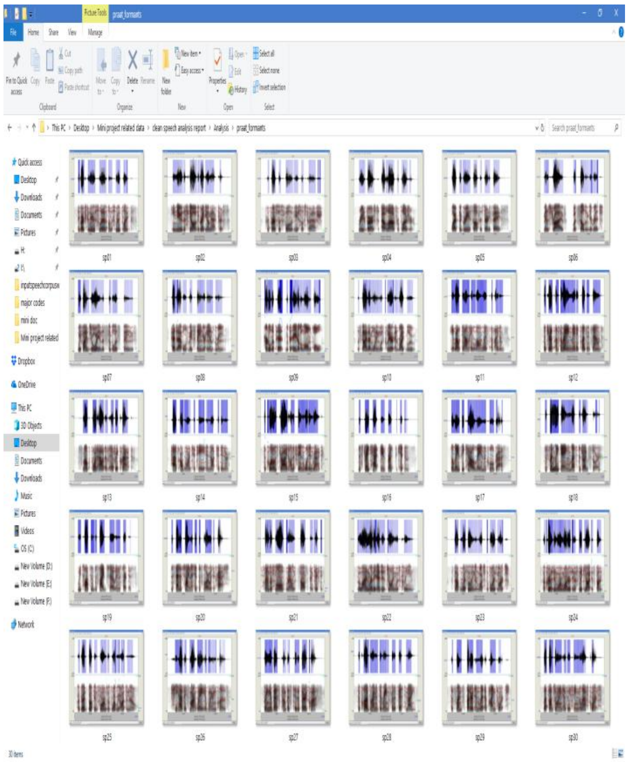

Figure 5.3: The Analysis of All the 30 Clean Speech Signals Taken From Speech Corpus Database

The figures obtained from these analyses have been noted down and are tabulated neatly as shown below:

Table 5.1: Detailed Analysis of All the

WAV FILES	PITCH (Hz)				INTENSITY (dB)			PULSES	FORMANTS				Harmonicity of voiced parts only		
	Maximum	Minimum	Mean	Median	Maximum	Minimum	Mean		F1(mean)	F2(mean)	F3(mean)	F4(mean)	Mean autocorrelation	Mean noise-to-harmonics ratio	Mean harmonics-to-noise ratio
sp01	373.845	111.929	54.211	142.33	75.409	19.695	64.78	213	533.694	1403.833	1983.089	2761.354	0.879353	0.166354	12.661dB
sp02	151.583	98.476	125.318	124.513	74.632	21.491	66.774	179	536.926	1497.828	2200.139	2750.693	0.872384	0.171842	11.095dB
sp03	462.348	106.886	150.944	130.973	77.473	20.145	66.046	161	648.251	1427.973	2104.188	2770.385	0.880781	0.169085	10.933dB
sp04	162.28	77.434	135.758	135.957	74.511	16.36	67.359	147	555.152	1403.686	2102.97	2820.524	0.82964	0.240524	8.497dB
sp05	330.137	114.812	143.548	133.258	76.109	16.823	66.895	148	567.572	1534.061	2152.411	2770.084	0.854816	0.199533	9.192dB
sp06	442.878	90.811	144.97	120.937	77.27	14.505	67.355	205	583.858	1485.167	2259.227	2860.938	0.791201	0.318406	7.406dB
sp07	388.922	87.641	134.218	121.15	76.012	14.63	66.019	200	515.783	1455.488	2245.556	2807.473	0.867438	0.180662	10.472dB
sp08	439.872	90.169	133.489	122.409	78.626	16.021	66.596	216	500.907	1505.36	2195.954	2857.935	0.847949	0.211311	9.339dB
sp09	504.569	100.215	194.495	138.782	74.598	11.094	66.836	378	495.164	1350.878	2236.481	3001.419	0.832223	0.253735	9.378dB
sp10	445.171	91.515	142.848	119.195	67.124	16.923	67.124	235	503.954	1421.706	2142.299	2752.934	0.824801	0.260739	8.588dB
sp11	313.016	74.746	234.227	239.93	79.944	14.918	67.761	413	604.145	1331.04	1969.766	2814.953	0.932469	0.087514	14.544dB
sp12	314.43	195.427	233.562	222.896	75.921	20.14	67.016	390	572.549	1357.882	2063.144	2832.232	0.927304	0.090914	13.579dB
sp13	336.639	76.675	232.333	223.313	75.727	19.738	67.731	251	676.192	1375.982	1976.723	2803.144	0.872869	0.190235	11.294dB
sp14	484.697	95.765	230.687	229.605	80.75	18.632	67.548	381	641.338	1323.959	2130.657	2799.671	0.91899	0.112517	14.015dB
sp15	492.403	205.908	250.209	234.622	74.648	20.513	67.13	453	626.387	1276.801	2011.165	2815.299	0.934513	0.092041	15.361dB
sp16	359.905	87.387	275.713	274.085	78.138	18.482	66.898	212	648.419	1461.451	2270.056	2898.68	0.903129	0.124859	12.563dB
sp17	373.278	81.721	271.029	267.295	75.044	17.587	66.411	367	628.778	1373.065	2061.145	2894.27	0.934547	0.08267	14.731dB
sp18	394.253	101.212	274.85	256.815	74.166	17.176	66.135	336	606.799	1421.062	2149.111	2877.869	0.882629	0.178886	13.162dB
sp19	383.906	88.25	269.416	263.318	80.625	16.862	67.77	328	663.418	1413.479	2101.286	2885.964	0.917253	0.115637	14.885dB
sp20	355.365	86.18	269.13	277.132	78.514	16.827	67.514	300	740.326	1458.624	2173.42	2899.752	0.841824	0.231122	9.253dB
sp21	391.6	104.804	145.704	126.829	75.66	23.964	66.594	269	585.185	1329.626	2091.461	2798.405	0.855645	0.219521	10.983dB
sp22	445.084	95.468	143.533	130.129	72.542	20.54	65.426	279	488.419	1386.789	2053.427	2766.504	0.88528	0.155547	11.284dB
sp23	454.331	102.281	151.256	126.68	74.901	19.602	66.149	215	562.969	1334.884	2085.671	2786.705	0.816799	0.279512	8.540dB
sp24	453.205	94.711	163.648	136.576	75.149	18.628	65.9	356	517.176	1361.152	1970.144	2667.348	0.872983	0.177726	10.744dB
sp25	490.861	90.097	138.092	118.722	77.195	19.127	66.548	225	519.803	1291.086	2022.564	2754.721	0.821842	0.254041	8.266dB
sp26	275.982	77.243	193.346	219.96	76.57	23.965	66.649	262	614.565	1383.587	2143.857	2789.119	0.863273	0.190046	10.355dB
sp27	424.732	90.713	212.682	216.033	73.48	20.673	66.168	246	653.434	1428.716	2086.682	2775.529	0.880959	0.182602	12.226dB
sp28	304.396	189.52	223.048	219.362	78.818	23.121	66.932	295	607.347	1380.088	2081.332	2798.293	0.902755	0.134734	12.556dB
sp29	291.304	81.42	221.292	222.114	78.386	22.474	67.087	230	645.6	1405.656	2086.053	2821.985	0.877556	0.191735	12.667dB
sp30	273.077	97.452	213.322	215.303	77.26	25.254	66.852	296	655.096	1335.242	2071.102	2810.429	0.907227	0.124779	12.868dB

Apart from these analysis there are two analysis namely cepstral, spectral analysis that are carried out by subjecting the signals to windows like Hanning window, Hamming window, Gaussian window $\alpha = 2.5, 3.5, 4.5$ by using "Audacity"software.

The following are the cepstral and spectral response when clean speech sp01 is subjected to the windows:

Cepstral Response of clean speech sp01 when subjected to various windows:

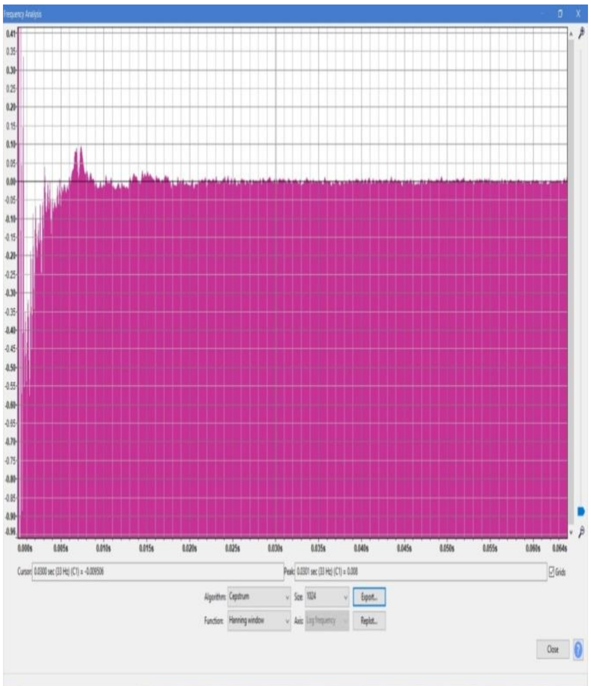

Figure 5.4: Cepstral Response of sp01 When Subjected to Hanning Window

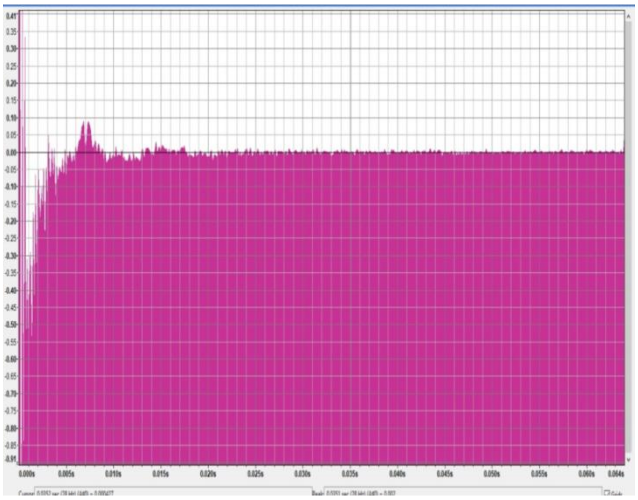

Figure 5.5: Cepstral Response of sp01 When Subjected to Gaussian Window With $\alpha = 4.5$

Spectral Response of clean speech sp01 when subjected to various windows:

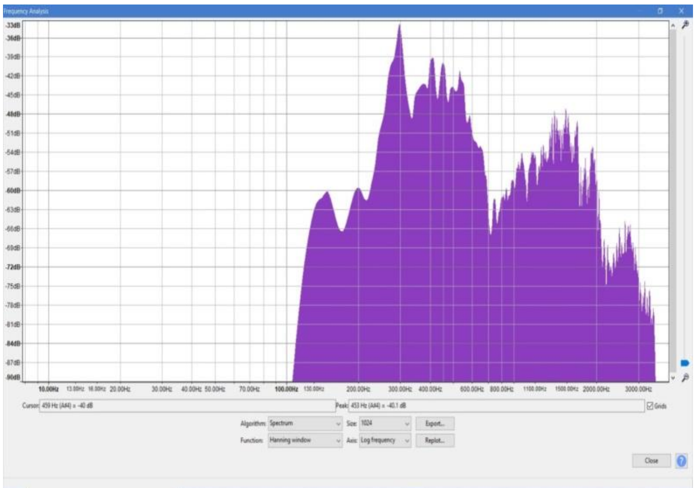

Figure 5.6: Spectral Response of sp01 When Subjected to Hanning Window

These were the cepstral and spectral analysis of only a clean speech signal named "sp01", like this these analysis are carried out with all the 30 clean speech signals. All of this analysis has also been repeated with Noisy Speech signals which are discussed in the following section.

Noisy Speech

As we know that clean speech has to be corrupted and then it has to be de-noised with the proposed methodology in the project, we are corrupting the clean speech signals by the noisy ones.

Corrupted speech = Clean speech + Noisy speech

The noisy signals has not been collected from any standard speech database, rather it has been recorded by us. In our project we have recorded the noisy speech under different environments. It contains the voice of the people both male and female under certain age groups ranging from 6 years to 45 years. Below there is a table mentioned which gives brief details about the recorded Noisy Speech:

The collected Noisy Speech is processed using "Audacity" software and they are converted into following formats:

16-bit PCM
24-bit PCM
32-bit PCM

The analysis of Noisy Speech is carried out in the same way as we have carried out the analysis of the Clean Speech, i.e., in terms of intensity, pitch, spectrum, formants, pulses. The other two analysis i.e., the cepstral and spectral analysis has also been carried out by subjecting the noisy speech signals to Hanning window, Hamming window, Gaussian windows with $\alpha = 2.5, 3.5, 4.5$.

There are a total of 18 Noisy speeches which have been recorded and they are in 3 formats 16-bit PCM, 24-bit PCM, and 32-bit PCM. But the analysis has been carried out only for the 32-bit PCM format of each age group of signal. Below is the collection of noisy speech which belongs to different gender (both male and female) and age groups:

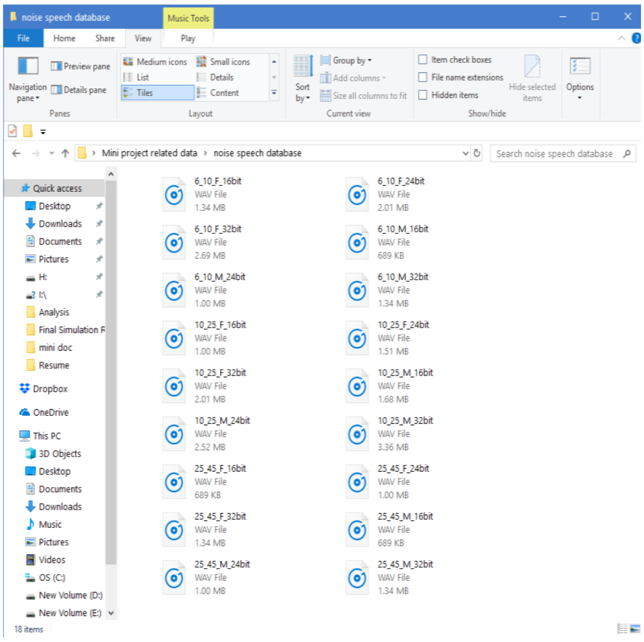

Figure 5.7: Noisy Speech Database

The analysis of noisy speech 6_10_F_32bit has been carried in terms of its intensity, pulses, pitch, formants, and spectrum as follows:

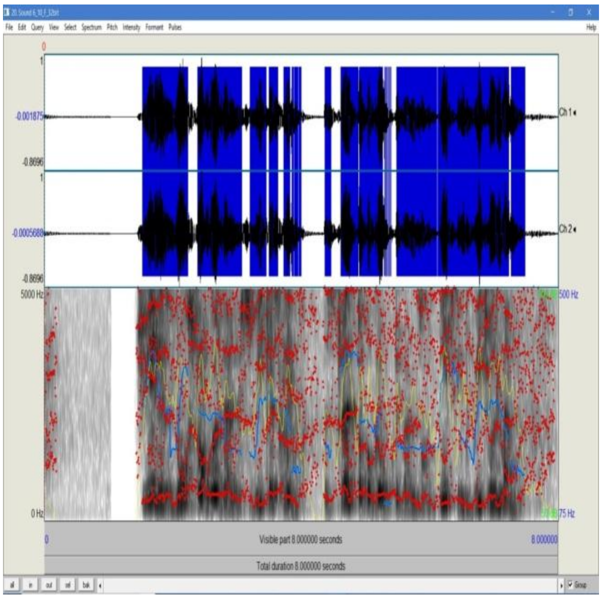

Figure 5.8: The Complete Analysis of the Noisy Speech Signal _10_F_32bit In Terms of Its Spectrum, Pitch, Intensity, Formants, Pulses

We have carried out the entire analysis for all the Noisy speech signals belonging to different age groups having 32bit format, and have noted down thein terms of their spectrum, pitch, intensity, formants, and pulses using "Praat" software as shown below:

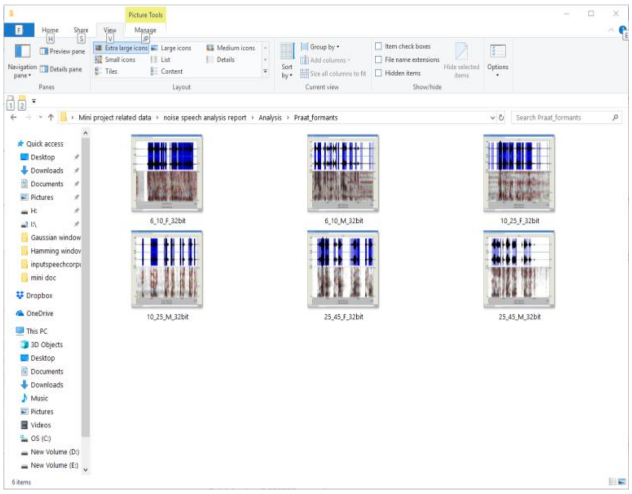

Figure 5.9: The Analysis of All the Noisy Speech Signals belonging to different age group (both male and female) having 32bit format.

The figures obtained from these analyses have been noted down and are tabulated neatly as shown below: Apart from these analysis there are two analysis namely cepstral, spectral analysis that are carried out by subjecting the signals to windows like Hanning window, Hamming window, and Gaussian windows with $\alpha = 2.5, 3.5, 4.5$ by using "Audacity" software. The following are the cepstral and spectral response of all the noisy speeches having 32bit PCM when they are subjected to the Hanning window, Hamming window and Gaussian windows with $=$ 2.5,3.5, 4.5.

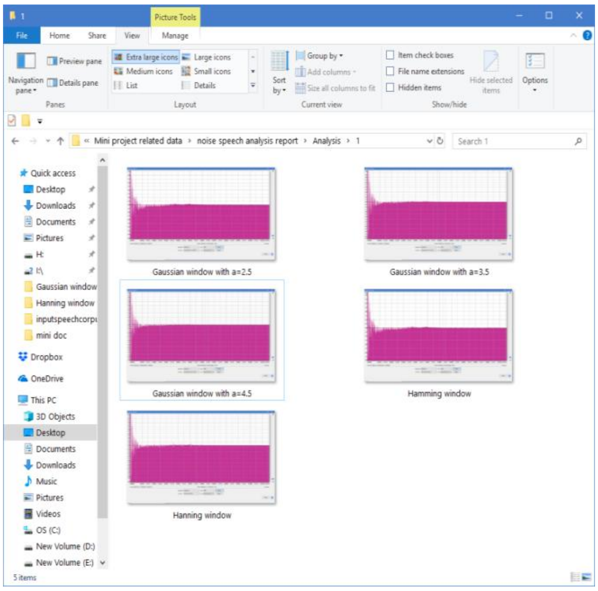

Figure 5.10: Cepstral Analysis of the 6_10_F_32bit PCM Noisy Speech When Subjected to Different Windows.

TABLE 6.1: FWSEG-SNR OBJECTIVE MEASURES WITH DIFFERENT WAVELET

Wavelet family	Input SNR	FWSEG Measure Using Different Threshold Methods					
		T_H	T_S	T_I	T_{MI}	T_{HI}	T_{H2}
Db4	0	7.111	6.164	6.702	5.978	6.209	6.270
	5	8.586	6.976	7.570	8.085	7.854	8.189
	10	10.269	7.688	8.298	10.992	9.398	9.957
	15	11.887	8.361	8.964	14.090	10.559	11.237
Db6	0	7.170	6.233	6.818	5.930	6.241	6.277
	5	8.699	6.987	7.629	8.104	7.923	8.246
	10	10.399	7.633	8.271	11.060	9.445	10.012
	15	12.010	8.318	8.934	14.172	10.580	11.259
Sym5	0	7.152	6.176	6.776	5.921	6.147	6.204
	5	8.640	6.968	7.577	8.089	7.851	8.193
	10	10.342	7.646	8.253	11.064	9.472	10.012
	15	11.870	8.247	8.897	14.161	10.560	11.228
Sym7	0	7.108	6.335	6.833	5.936	6.219	6.274
	5	8.632	7.061	7.670	8.114	7.880	8.242
	10	10.387	7.689	8.353	11.065	9.419	9.997
	15	12.011	7.746	9.034	14.199	10.564	11.265

VI . IMPLEMENTATION

The source code has been implemented by giving a clean speech and a noisy speech as an input, then the clean speech is mixed with the noisy speech and the result is corrupted speech. This corrupted speech is then de-noised by using our proposed method de-noising using wavelet thresholding.

VII. SIMULATION RESULTS CONCLUSION

In this task, a relative performance of Hard, Soft, Improved, Modified Improved and the proposed Hybrid Thresholding techniques utilizing Daubechies and Symlet wavelet families have been made to Enhance Telugu discourse signals. This performance gives the decision of Threshold capacity to utilize Wavelet de-noising for Telugu Speech. The consequences for the five Proverbs have been inspected. The estimations of the extricated parameters are likewise introduced. From the outcomes, Db4 and Sym5perform superior to anything different wavelets chose for this examination. The Proposed calculation will be tried with the Real commotions like Babble, Car, Airport and so on as a component of their future work.

VIII. ACKNOWLEDGMENT

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AUTHORS PROFILE

Dr. S. China Venkateswarlu, B.Tech, M.Tech, Ph.D.,MISTE, CSI, IAENG, SDIWC, ISRD, IFERP.,working as a PROFESSOR in Dept. Of ECE at Institute of Aeronautical Engineering, Dundigal, Hyderabad. INDIA, Worked in various Engineering College and have 21 years of Teaching, R&D and Industrial experience. Research in the area of Signal Processing / Speech Processing.Image Video processing. He has presented 13 National, 12 International Conference Papers are Presented throw EDAs. He has published 15 research papers in National and International Journals. He has reveived text book on DSP.He has written text book on Speech Enhancement Techniqes.

A.Karthik, Research scholar in department of ECE-Veltech Dr.RR & Dr.Sr University,Chennai. M.Tech (ECE) in Mallareddy institute of technology & science (JNTUH), Obtained B.Tech (ECE) in Vaageswari College of engineering (JNTUH) .Worked as Assistant professor in Institute of Aeronautical Engineering, Dundigal, Hyderabad.Areas of interest Digital Signal Processing, Speech processing, Embedded Systems,karthik011190@gmail.com.

Dr. Kommu Naveen, B.Tech, M.Tech, PhD.,MISTE, working as a PROFESSOR in Dept. Of ECE at MLR IT M, Dundigal, Hyderabad. INDIA, Worked in various Engineering College and have 12 years of Teaching, experience. Research in the area of Digital Signal Processing and Wireless communications He has published 9 research papers in National and International Journals.