Noise Cancellation Method for Robust Speech Recognition

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ABSTRACT
Noise cancellation is the process of removing background noise from speech signal. The degradation of speech due to presence of background noise and several other noises cause difficulties in various signal processing tasks like speech recognition, speaker recognition, speaker verification etc. Many methods have been widely used to eliminate noise from speech signal like linear and nonlinear filtering methods, adaptive noise cancellation, total variation denoising etc. This paper addresses the problem of reducing the impulsive noise in speech signal using compressive sensing approach. The results are compared against three well known speech enhancement methods, spectral subtraction, Total variation denoising and signal dependent rank order mean algorithm.

An automatic speech recognition system for Digits in Malayalam Language is implemented using MFCC and GMM. The impulse noise corrupted speech signal and the enhanced speech signal (the output of the noise cancellation system) are given as input to the classification system. The speech recognition system gives 12.3 % accuracy for noisy signal where as 92.3 % accuracy for the enhanced signal

Objective and subjective quality evaluation are performed for the four speech enhancement scheme. Results show that the signal processed by the compressive sensing based method outperforms the other three methods.

General Terms
Speech Enhancement, Compressive Sensing, and Automatic Speech Recognition.

Keywords
Speech Enhancement, Compressive Sensing, Over complete Dictionary, Quality Evaluation Metrics and Automatic Speech Recognition.

1. INTRODUCTION
Speech enhancement aims in improving the quality of the speech signal by reducing the background noise. Quality of speech signal is weighed by its clarity, intelligibility and pleasantness [1]. Speech enhancement is a preliminary procedure in the speech processing area, including speech recognition, speech synthesis, speech analysis and speech coding.

In communication systems speech signal is sometimes corrupted with short duration noises like impulsive noise [2].

To listeners, these interferences are highly unpleasant and should be suppressed in order to enhance the quality and intelligibility of speech signal. Most of the speech-signal processing algorithms are based on the assumption that the noise follows Gaussian distribution and is additive in nature. But noises like impulsive noise are characterized by non-Gaussian probability distribution. This will reduce the performance of the speech processing systems drastically, in presence of impulsive noise [2]. So we go for impulsive noise cancellation as a pre-processing step.

The classical method for impulsive noise cancellation from speech signal is noise reduction using median filtering method [2]. In this method each window of specific length is processed and the middle sample is replaced by the median of the window. The performance of this method can be improved by introducing adaptive threshold.

In [3] Charu Chandra et al. proposed a method for impulsive noise cancellation in speech based on signal dependent rank order mean (SD-ROM) algorithm. A window of five samples is examined iteratively for impulse sample and if detected within the sampled window, then the corresponding sample is replaced by an estimate based on neighboring samples. This method is very simple but efficient in case of ideal impulse and configurable to the type of impulse.

S. V. Vasighi and P. J. W. Rayner proposed a method for removing impulsive noise from speech and sound signals based on a detection interpolation scheme [2]. A linear prediction based scheme is used in this method. This method transforms the speech into excitation domain of the speech signal where the detectability of noise pulse is high. Samples that are detected as an impulse are replaced by an estimate based on LPC interpolation algorithm. This algorithm is applied to various speech signals and results shows that signal with a periodic structure shows better results.

Based on Discrete Wavelet Transform an impulse noise detection and removal method was reported by Zhiyong He et al [4]. This method uses two steps, impulse detection and noise removal. The first step is to find the difference of energy distribution between noise and impulsive colored noise in frequency domain. Based on this result, a new signal is constructed to detect impulsive colored noise. Evaluation of this method is done by improving signal to noise ratio (SNR). The experiment results show that the output SNR of enhanced speech is better than input SNR and the intelligibility of the enhanced speech is improved.

In [5], Mital A. Gandhi et al. presented a filtering method in time domain for detection and cancellation of impulsive noise in speech. The detection scheme uses the idea of auto regressive model via the Huber M-estimator and iterative expectation maximization (EM) algorithm. This method is computationally less complex than the traditional methods.

Based on soft decision and recursion, an impulse noise removal method was proposed by Sina Zahedpour et al [6]. In this method, the location and amplitude of the impulse is given by an adaptive threshold and soft decision. After estimating the position and amplitude of the impulse, an adaptive algorithm is implemented to reduce the noise. Then an approximation of the original signal is obtained using an iterative process. The method is tested using signals created by matlab simulation and it gives good results.

R. C. Nongpiur presented a novel method to remove impulsive type disturbances from speech signals in wavelet transform domain [7]. The method is works on the multi-resolution property of wavelet transform. The wavelet coefficients correspond to impulse noise is identified and removed based on two features, the slow time-varying nature and the Lipschitz regularity of the speech components. The method is tested with speech signals and results show the method is suitable for removing impulsive noise from speech.

In this paper, we propose a robust noise cancellation method for speech signal corrupted by impulsive noise. The method is based on compressive sensing approach and make use of an over complete dictionary that consist of DCT matrix and identity matrix as bases. The method is compared against three well known speech enhancement methods. Section 2, briefly describes the basic theory behind compressive sensing. This section also describes the various quality evaluation metrics used. Section 3 covers discussion of the experimental results and finally the conclusion is provided in section 4.

2. THEORY

2.1 Compressive Sensing

According to Shannon's theorem, a signal can be perfectly reconstructed if and only if the sampling rate is at least twice the maximum frequency present in the signal. This is known as Nyquist rate. Conventional approaches for sampling signals or images are based on Shannon's sampling theorem. Compressive sensing, compressed sensing or compressive sampling is a new method of reconstructing a sparse image or signal (A Signal is said to be sparse if it contains most of the elements as zeros) from fewer samples than the traditional Nyquist rate [8] [9].

Consider a signal x of length Nx1. The real time signals like speech signals are not sparse in time domain. Since compressive sensing is only applicable to sparse signals, we need to convert x into sparse. A dense signal in one domain (e.g. time domain) may be sparse in another domain (e.g. frequency domain). However, for natural signals and images, there exist some bases and dictionaries such that the projection of signal into the dictionary or bases (or some operation) converts our signal of interest to sparse or approximately sparse [10][11]. Let us assume our signal x is sparse in some basis $\psi = \{\psi_i,\ i = 1,2,...,N\}$. Now our signal of interest became sparse. We have $x = \psi\alpha$ where $\alpha = \{\alpha_i,\ i = 1,2,...,N\}$. We project the sparse signal into m bases where m<<n and we get m measurements called y. The bases can be DCT bases, random vectors, wavelet coefficients etc. The Mx1 projection vector y can be written as:

$$y = \phi x = \phi\psi\alpha = \theta\alpha$$

The measurement matrix y is very less in size compared to the original signal x. The original signal x can be reconstructed from measurement matrix y. It is an optimization problem which relies on the compressibility of x in the base ψ. The signal x can be reconstructed by means of standard linear programming algorithms such as L_1 Magic, Orthogonal Basis Pursuit, Orthogonal Matching Pursuit etc. [10].

In the presence of impulse noise, the sparse property of the signal is lost forever (because presence of even one impulse will introduce all the frequency components) and the compressive sensing is no more applicable to the signals corrupted by impulsive noise. Another concept called an over complete dictionary can be applied here. An over complete dictionary D consist of a number of bases or atoms which is more than enough to reconstruct signal. Here some atoms are not unique.

For noise removal purpose we created a dictionary which consists of DCT bases and Identity matrix. Identity matrix in the dictionary has similar characteristics as the impulse noise. If we project our noisy signal x into the dictionary, the identity matrix in the dictionary captures the impulse noise alone from the signal. The actual signal is captured by the DCT bases. The original signal can be reconstructed by using the standard linear programming algorithm L_1 magic[12].

2.2 Quality Evaluation Metrics

In speech enhancement, we need to evaluate the quality of the method based on some metrics. There are objective quality evaluation method and subjective quality evaluation methods.

2.2.1 Subjective Quality Evaluations:

Subjective quality evaluations are done by a group of listeners. They are also called as test subjects. The quality of processed speech is expressed using a specific unit, called Mean Opinion Score (MOS). After listening, listeners have to rate that particular enhanced speech signal based on three factors. They are described below.

• The speech signal alone is rated based on signal distortion.

• The background noise is rated based on background disturbances (BAK).

• The overall quality as the mean of SIG and BAK Scale values (OVRL).

The SIG and BAK scale [13] are listed in the Table 1.

Table 1. Description of SIG and BAK Scale

Rating	SIG Scale	BAK Scale
5	Purely Natural, no degradation	Not perceptible
4	Fairly Natural, slight degradation	Somewhat noticeable
3	Somewhat natural, somewhat degraded	Noticeable but not intrusive
2	Fairly unnatural, fairly degraded	Fairly Noticeable, somewhat intrusive
1	Quite unnatural, Highly degraded	Quite Noticeable, Highly Intrusive

2.2.2 Objective Quality Evaluations:

Objective measures are evaluated based on mathematical measures and represents the quality by comparing the original (clean speech) and degraded (enhanced speech) signals. In this study we have chosen four objective measures such as Segmental SNR (SNRseg), Weighted Slope Spectral distance

(WSS), Perceptual Evaluation of Speech Quality (PESQ) and Log Likelihood Ratio (LLR) [14]. A lower value of WSS and higher value of SNRSeg indicates better quality of speech. Usually LLR lies in the range between 0 and 2.

Composite objective measures are derived from basic objective measures to form a new and more accurate measure. Conventional objective measures like SNRseg and LLR, are not correlated highly with speech/noise distortions and overall quality. The composite measures are obtained by using multiple linear regression analysis or by using nonlinear techniques. In this paper, we have chosen a composite measure for signal distortion (CSIG), a composite measure for noise distortion (CBAK), and a composite measure for overall speech quality (COVRL).

3. RESULTS AND DISCUSSIONS

The compressive sensing based noise cancellation method presented in section 2.1 is used for experimenting with a test database of speech signals. Digits (zero to nine) in Malayalam language is recorded with the help of 25 male and female speakers for the training and testing purpose. The speech signal was recorded with 16 KHz sampling rate 16 bit resolution and then the speech was stored as uncompressed .wav format. The impulsive noise is simulated using matlab. The proposed method is compared against the three well known speech enhancement techniques, spectral subtraction [15]-[17], SD-ROM Algorithm [3][18], and Total variation denoising [19][20]. Various parameter values in these four algorithms such as lambda, threshold, tolerance, spectral floor and subtraction factor are adjusted in such a way that the clarity and intelligibility of noisy speech is improved considerably.

Subjective and objective quality measures are evaluated for each test speech signal and the results obtained are compared against the result of the other three methods. The objective quality measures are evaluated in two steps. The first step is to evaluate the objective measures of the clean speech signal and noisy speech signal. This measure gives to what extend the clean speech is degraded by background noise. In second step, the clean speech and the enhanced speech signal is processed. This gives the measure of similarity between enhanced speech signal and clean speech signal. From the results, we found that compressive sensing based methods outperforms the other three algorithms, spectral subtraction, total variation denoising and signal dependent rank order mean algorithm, by its high PESQ scores.

A voice digit recognition system (zero to nine) in Malayalam language is implemented using MFCC and GMM. The output of the noise removal system is given as input to the automatic speech recognition system. A test digit database consisting of 200 speech signals (Malayalam digits by both male and female) is fed as input to the automatic speech recognition system after the speech enhancement. The enhanced signal of compressed sensing method gives 92.3 % accuracy in classification while the noisy speech signal gives only 13.3 % accuracy. The result reveals the importance of impulsive noise cancellation as a pre-processing step in speech processing tasks.

3.1 Compressive sensing based method:

The compressive sensing based method is tested with two types of impulse affected signals, real and ideal impulses. Ideal impulse is one sample long and real impulse is more than one sample long. Ten test speech signals (digits in Malayalam language) are tested with this method. The parameters used in this method, such as lambda and tolerance are adjusted in such a way that the enhanced speech signal quality is as high as possible. The optimum value for lambda and tolerance is found to be 0.005 and 0.001 respectively. Also size of DCT matrix is fixed as 3200.

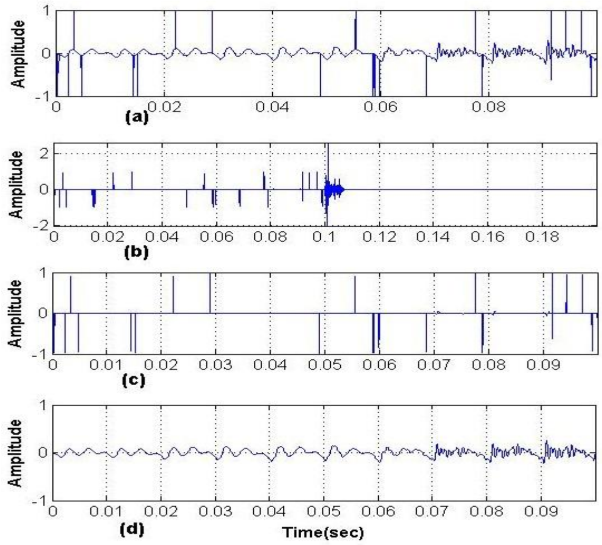

Fig 1: Impulsive noise cancellation using compressive sensing.

Figure 1(a) shows impulsive noise signal, 1(b) the reconstructed signal which is a combination of low frequency signal regions and high frequency signal regions. The first half contains high frequency portion (impulse) which is captured by the identity matrix. Figure 1(c) shows impulse alone and 1(d) shows the enhanced speech.

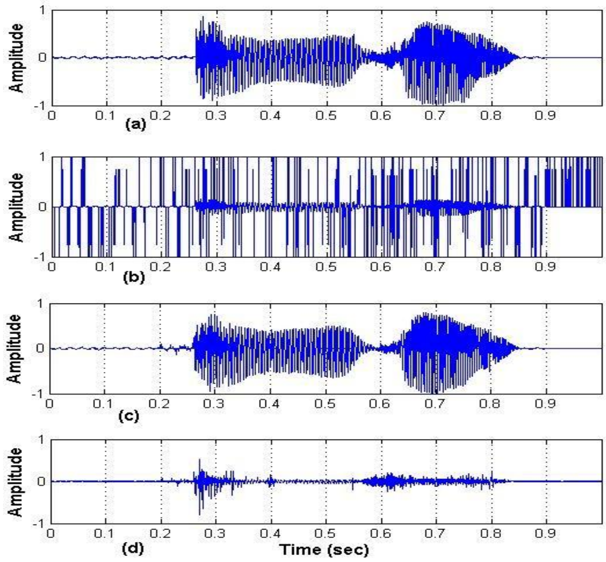

Fig 2: Impulsive noise cancellation using compressive sensing.

Table 2: Objective Quality Measures for Impulse Noiseancellation Using Compressed Sensing

Compressive Sensing	Digit	Csig	Cbak	Covrl	LLR	SNRseg	WSS	PESQ
Original & Noisy	1	2.2249	2.1225	2.0317	1.6582	-2.8649	33.4468	1.8893
Original & Enhanced	1	3.1576	3.4561	3.2697	1.7796	5.0447	11.1551	3.3105
Original & Noisy	2	2.4138	2.3053	2.109	1.4791	-0.0113	29.3542	1.8358
Original & Enhanced	2	3.2995	3.8767	3.3241	1.6141	12.0732	12.6545	3.2858
Original & Noisy	3	2.6844	2.5364	2.4552	1.4866	0.1639	25.0589	2.2332
Original & Enhanced	3	3.3467	4.0469	3.4193	1.6472	13.7519	13.3908	3.4315
Original & Noisy	4	2.8146	2.7301	2.5578	1.4082	2.6163	24.3282	2.3044
Original & Enhanced	4	3.7281	3.7709	3.4644	1.1201	11.4018	11.4551	3.1355
Original & Noisy	5	2.3463	2.3865	2.0765	1.544	1.2872	29.6876	1.8395
Original & Enhanced	5	3.363	3.8271	3.3318	1.53	11.5792	11.7494	3.2341

Figure 2 (a) shows the clean speech, 2(b) the noisy speech, 2(c) the enhanced speech and 2(d) shows the error signal. In figure 2(c), we can see that the enhanced signal is almost equivalent to the original signal. The objective quality measures are evaluated and shown in Table 2.

In Figure 2, (a) shows the clean speech, (b) the noisy speech, (c) the enhanced speech and (d) shows the error signal. In (c), we can see that the enhanced signal is almost equivalent to the original signal. The objective quality measures are evaluated and shown in Table 2.

3.2 Signal Dependent Rank Order Mean method:

The SD-ROM algorithm is first tested with ideal impulsive noise. The noise is perfectly removed and enhanced signal possess high perceptual quality. The algorithm uses two threshold values T1 and T2. The threshold is adjusted in such a way that the enhanced signal quality is maximum. The optimum threshold is found to be T1=0.05 and T2 =0.2 for a normalized signal.

Figure 3 shows the output of SD-ROM algorithm, 3(a) clean speech, 3(b) Noisy speech, 3(c) enhanced speech and 3(d) error signal.

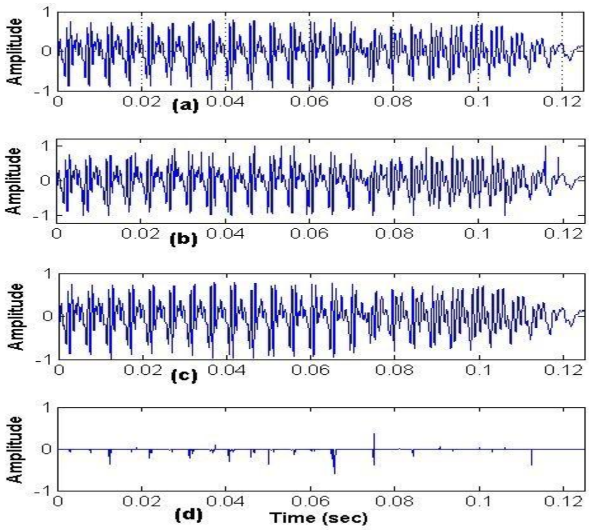

Fig 3: Impulsive noise cancellation using SDROM Algorithm.

Table 3: Objective Quality Measures for SDROM algorithm

SDROM	Digit	Csig	Cbak	Covrl	LLR	SNRseg	WSS	PESQ
Original & Noise	1	2.388	2.223	2.201	1.602	-2.5976	33.3928	2.0628
Original & Enhanced	1	2.599	2.299	2.341	1.468	-2.3041	28.0871	2.1057
Original & Noise	2	2.437	2.235	2.087	1.426	-0.714	27.9655	1.7618
Original & Enhanced	2	2.69	2.479	2.329	1.332	1.1751	24.8353	1.9757
Original & Noise	3	2.551	2.486	2.362	1.581	-0.1724	25.7758	2.1836
Original & Enhanced	3	2.83	2.716	2.555	1.397	2.361	21.7396	2.2716
Original & Noise	4	2.989	2.688	2.663	1.264	1.6372	23.7504	2.338
Original & Enhanced	4	3.175	2.944	2.83	1.186	4.3574	20.8671	2.4709
Original & Noise	5	2.346	2.387	2.077	1.544	1.2872	29.6876	1.8395
Original & Enhanced	5	2.611	2.663	2.325	1.46	3.4433	23.2627	2.0389

The method is then tested with real impulses which are speech signals corrupted by triangular impulse. A triangular impulsive speech signal is the one in which three consecutive samples are impulses.

In figure 3(c), we can see that the amplitude (effect) of the noise is reduced to one half. By increasing the window size we can improve the result to an extent. But increasing window size increases the computational complexity and running time of the algorithm. The objective quality measures are evaluated and shown in Table 3.

3.3 Spectral Subtraction method:

The spectral subtraction method is applied for impulse noise removal. The parameter values such as spectral floor and subtraction factor are adjusted in such a way that the impulse noise is reduced. The value of subtraction factor varies according to the SNR value in each frame. So the maximum subtraction factor is set as 10 and the spectral floor parameter is set as 0.005. The performance of this method is not satisfactory in presence of impulsive noise. Figure 4(a) shows the clean speech, 4(b) input noisy speech and 4(c) is the enhanced speech signal. We can see that in enhanced speech signal, some of the impulses are not removed. This algorithm degrades the perceptual quality of the speech signal. The remaining impulses in the enhanced speech will affect the intelligibility of the speech signal. The objective quality measures are evaluated and shown in Table 4.

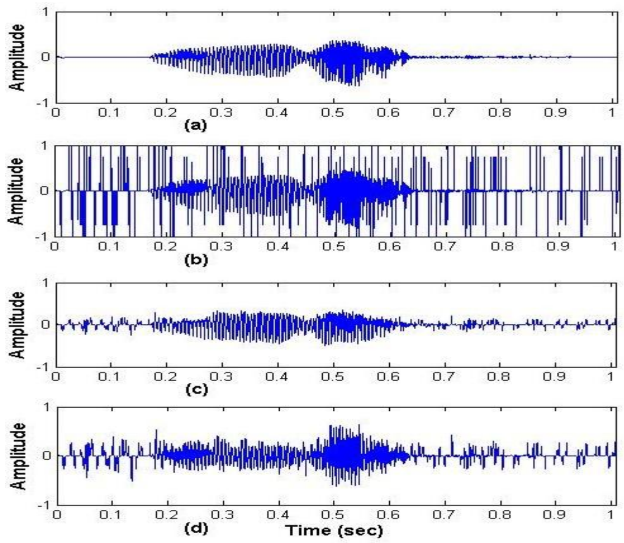

Fig 5: Total variation denoising.

Table 4: Objective Quality Measures for Spectral subtraction method

Spectral Subtraction	Digit	Csig	Cbak	Covrl	LLR	SNRseg	WSS	PESQ
Original & Noise	1	2.3883	2.2226	2.2008	1.6016	-2.5976	33.3928	2.0628
Original & Enhanced	1	1.9685	2.0062	1.8245	1.6677	-1.6616	58.7455	1.8579
Original & Noise	2	2.4365	2.2354	2.0865	1.4259	-0.714	27.9655	1.7618
Original & Enhanced	2	2.5739	2.3955	2.271	1.2939	1.7579	50.8491	2.1061
Original & Noise	3	2.6844	2.5364	2.4552	1.4866	0.1639	25.0589	2.2332
Original & Enhanced	3	1.9067	2.3646	1.9028	1.9089	1.703	49.3749	2.0271
Original & Noise	4	2.9885	2.6884	2.6627	1.2638	1.6372	23.7504	2.338
Original & Enhanced	4	2.802	2.5414	2.4381	1.158	2.9717	46.7948	2.192
Original & Noise	5	2.3463	2.3865	2.0765	1.544	1.2872	29.6876	1.8395
Original & Enhanced	5	1.8157	2.2498	1.7336	1.7904	2.5368	60.215	1.8357

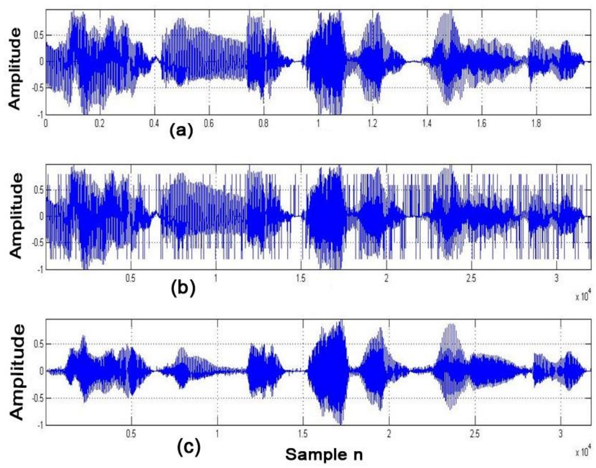

Fig 4: Spectral subtraction method.

3.4 Total variation denoising

The total variation denoising method is then applied for impulse noise removal.

The parameter value lambda, is adjusted in such a way that the impulse noise is reduced. The optimum value of lambda is found to be 10. This method is not suited for impulse noise removal. Figure 5(a) shows the clean speech, 5(b) input noisy speech and 5(c) is the enhanced speech signal, 5(d) is the error signal. From the results, we can observe that some of the impulses are still present in the signal and signal quality is degraded significantly. The objective quality measures are evaluated and shown in Table 5.

Subjective quality measures are done using ten test subjects for the three methods. Results are shown in Table 6. Compressed sensing based method gives better subjective measures than the other three methods.

4. CONCLUSION

This paper presents a novel and simple algorithm for automatically removing impulsive noise from speech signals. In this work, we have introduced compressive sensing based methods for noise cancellation. This method removes the impulsive noise from speech signal with the help of an over complete dictionary, which consist of an Identity matrix and DCT bases. It gives better results compared to the traditional methods like total variation and median filtering techniques.

The proposed algorithm is evaluated using different objective and subjective tests like LLR, SNRSeg, PESQ etc. The result shows that the quality of speech has been improved considerably. Also the output of the enhanced speech signal is shows 92.3 % accuracy in automatic speech recognition of Malayalam digits whereas the noisy speech signal shows only 12.3 % accuracy. This shows the importance of impulsive noise removal in speech processing algorithms as a pre-processing step.

Table 5: Objective Quality Measures for Total Variation Denoising

Total Variation	Digits	CSIG	CBAK	COVRL	LLR	SNRseg	WSS	PESQ
Original & Noisy	1	2.0838	1.9417	2.0051	1.799	-5.7974	41.5178	2.0159
Original & Enhanced	1	1.5266	2.0098	1.7667	2.2836	-4.018	59.0481	2.1804
Original & Noisy	2	2.0818	2.024	1.8545	1.6765	-2.8623	33.0581	1.6774
Original & Enhanced	2	1.4052	2.2647	1.8278	2.6123	-2.6755	47.0482	2.3611
Original & Noisy	3	2.1724	2.2038	2.0238	1.7517	-2.1158	29.7617	1.9068
Original & Enhanced	3	1.9683	2.3986	2.1997	2.1857	-2.1069	44.6115	2.5306
Original & Noisy	4	2.686	2.3987	2.4442	1.4345	-1.3808	31.3923	2.2415
Original & Enhanced	4	1.6299	2.3737	2.0241	2.4711	-1.9505	50.6451	2.5463
Original & Noisy	5	2.0023	2.1234	1.8576	1.7937	-1.8054	34.7147	1.7702
Original & Enhanced	5	1.6233	2.3568	1.9678	2.421	-1.4866	49.7773	2.437

Table 6: Subjective Quality Measures for Impulse Noise Cancellation.

Test Subject	CS - SIG	CS-BAK	CS-OVRL	SS-SIG	SS-BAK	SS-OVRL	SD-SIG	SD-BAK	SD-OVRL	TV-SIG	TV-BAK	TV-OVRL
1	4	4	4	3	2	3	3	2	3	2	2	2
2	4	4	4	3	2	3	3	2	3	3	3	3
3	3	3	4	3	2	3	3	2	3	2	3	2
4	3	3	4	4	3	3	3	2	3	2	2	2
5	4	3	4	3	3	3	3	1	2	2	2	2
6	3	3	4	3	2	3	3	2	3	3	3	3
7	4	3	4	3	3	3	3	2	3	3	3	3
8	4	4	4	3	2	3	3	2	3	3	2	2
9	4	4	4	3	2	3	3	2	3	2	2	2
10	3	4	4	3	2	3	3	1	2	3	2	2

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