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SPECTRUM IMPROVEMENT ALGORITHM FOR SPEECH PROCESSING WITH NOISE CANCELLATION

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ABSTRACT

Noise cancellation is the way toward expelling foundation commotion from discourse flag. The corruption of discourse because of essence of foundation clamor and a few different commotions cause troubles in different flag preparing undertakings like discourse acknowledgment, speaker acknowledgment, speaker confirmation and so on. Numerous techniques have been generally used to dispense with clamor from discourse flag like straight and nonlinear separating strategies, versatile commotion cancellation, add up to variety denoising and so on. This paper tends to the issue of lessening the imprudent clamor in discourse flag utilizing compressive detecting approach. The outcomes are looked at against three surely understood discourse upgrade techniques, unearthly subtraction, Total variety denoising and flag subordinate rank request mean calculation. A programmed discourse acknowledgment framework for Digits in Malayalam Language is executed utilizing MFCC and GMM. The motivation clamor debased discourse flag and the improved discourse flag (the yield of the commotion cancellation framework) are given as contribution to the arrangement framework. The discourse acknowledgment framework gives 12.3 % exactness for loud flag where as 92.3 % precision for the improved flag Objective and subjective quality assessment are performed for the four discourse upgrade conspire. Results demonstrate that the flag handled by the compressive detecting based technique beats the other three strategies.

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

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

1. INTRODUCTION

Speech enhancement points in enhancing the nature of the discourse motion by decreasing the foundation clamor. Nature of discourse flag is weighed by its clearness, understandability and agreeableness [1].

Discourse upgrade is a preparatory technique in the discourse handling territory, including discourse acknowledgment, discourse amalgamation, discourse investigation and discourse coding.In

correspondence frameworks discourse flag is here and there debased with brief length commotions like incautious clamor [2]. To audience members, these obstructions are exceptionally unpalatable and ought to be smothered keeping in mind the end goal to improve the quality and clarity of discourse flag. A large portion of the discourse flag handling calculations depend on the presumption that the commotion takes after Gaussian dissemination and is added substance in nature. Be that as it may, commotions like incautious clamor are described by non-Gaussian likelihood dissemination. This will diminish the execution of the discourse handling frameworks definitely, in nearness of rash clamor [2]. So we go for imprudent clamor cancelation as a pre-handling step. The traditional technique for hasty commotion cancelation from discourse flag is commotion lessening utilizing middle separating strategy [2]. In this strategy every window of particular length is handled and the center specimen is supplanted by the middle of the window. The execution of this technique can be enhanced by presenting versatile threshold. In [3] Charu Chandra et al. proposed a strategy for rash commotion cancelation in discourse in light of flag subordinate rank request mean (SD-ROM) calculation. A window of five specimens is analyzed iteratively for motivation test and if distinguished inside the examined window, at that point the relating test is supplanted by a gauge in light of neighboring examples. This strategy is extremely straightforward however productive if there should arise an

occurrence of perfect drive and configurable to the kind of impulse. S. V. Vasighi and P. J. W. Rayner proposed a strategy for expelling rash clamor from discourse and sound signs in view of a location addition plot [2]. A direct forecast based plan is utilized as a part of this strategy. This strategy changes the discourse into excitation space of the discourse flag where the perceptibility of commotion beat is high. Tests that are recognized as a drive are supplanted by a gauge in light of LPC interjection calculation. This calculation is connected to different discourse flags and results demonstrates that flag with an intermittent structure indicates better outcomes. In view of Discrete Wavelet Transform a motivation clamor location and evacuation technique was accounted for by Zhiyong He et al [4]. This technique utilizes two stages, motivation identification and commotion expulsion. The initial step is to discover the distinction of vitality dissemination amongst commotion and indiscreet hue clamor in recurrence space. In view of this outcome, another flag is built to distinguish imprudent hue clamor. Assessment of this technique is finished by enhancing sign to clamor proportion (SNR). The trial comes about demonstrate that the yield SNR of upgraded discourse is superior to enter SNR and the understandability of the upgraded discourse is improved. In [5], Mital A. Gandhi et al. exhibited a separating strategy in time area for identification and cancelation of imprudent clamor in discourse. The discovery plot utilizes the possibility of auto backward model by means of the Huber M-estimator and

iterative desire boost (EM) calculation. This technique is computationally less mind boggling than the conventional strategies. In view of delicate choice and recursion, a motivation clamor evacuation strategy was proposed by Sina Zahedpour et al [6]. In this strategy, the area and adequacy of the drive is given by a versatile edge and delicate choice. Subsequent to assessing the position and sufficiency of the motivation, a versatile calculation is executed to decrease the clamor. At that point an estimation of the first flag is acquired utilizing an iterative procedure. The technique is tried utilizing signals made by matlab recreation and it gives great outcomes. R. C. Nongpiur introduced a novel technique to expel rash sort unsettling influences from discourse motions in wavelet change area [7]. The strategy is deals with the multi-determination property of wavelet change. The wavelet coefficients relate to motivation clamor is recognized and evacuated in light of two highlights, the moderate time-shifting nature and the Lipschitz consistency of the discourse segments. The technique is tried with discourse flags and results demonstrate the strategy is appropriate for expelling incautious commotion from discourse. In this paper, we propose a hearty clamor cancelation strategy for discourse flag tainted by incautious commotion. The strategy depends on compressive detecting methodology and make utilization of an over total word reference that comprise of DCT lattice and character network as bases. The strategy is analyzed against three surely understood discourse improvement techniques. Area 2, quickly portrays the

fundamental hypothesis behind compressive detecting. This area likewise depicts the different quality assessment measurements utilized. Area 3 covers talk of the trial comes about lastly the conclusion is given in segment 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 $N \times 1$. 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 $\psi = \{\psi_i, i = 1, 2, \dots, N\}$ Now our signal of interest became sparse. We have $x = \psi\alpha$ where $\alpha = \{\alpha_i, i = 1, 2, \dots, N\}$. We project the

sparse signal into m bases where $m \ll n$ and we get m measurements called y . The bases can be DCT bases, random vectors, wavelet coefficients etc. The $M \times 1$ 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. The signal x can be reconstructed by means of standard linear programming algorithms such as L1 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 L1 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 extent 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.

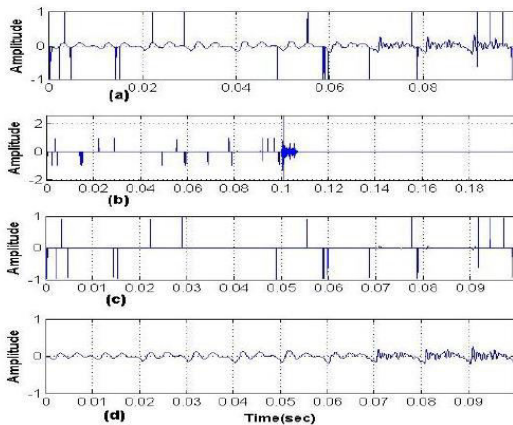


Fig 1: Impulsive noise cancellation using compressive sensing.

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.

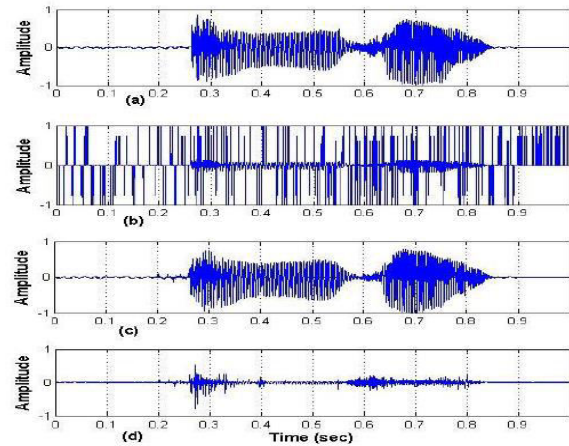


Fig 2: Impulsive noise cancellation using compressive sensing.

Table 2: Objective Quality Measures for Impulse Noise Cancellation Using Compressed Sensing

Compressive Sensing	Digit	Csig	Cbak	Covr1	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.

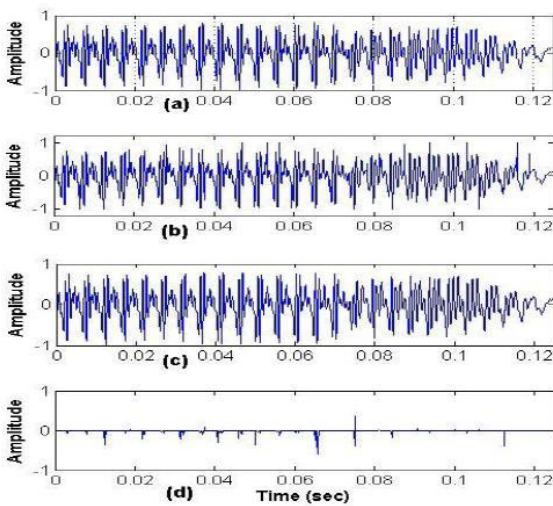


Fig 3: Impulsive noise cancellation using SDRM Algorithm.

Table 3: Objective Quality Measures for SDRM 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.

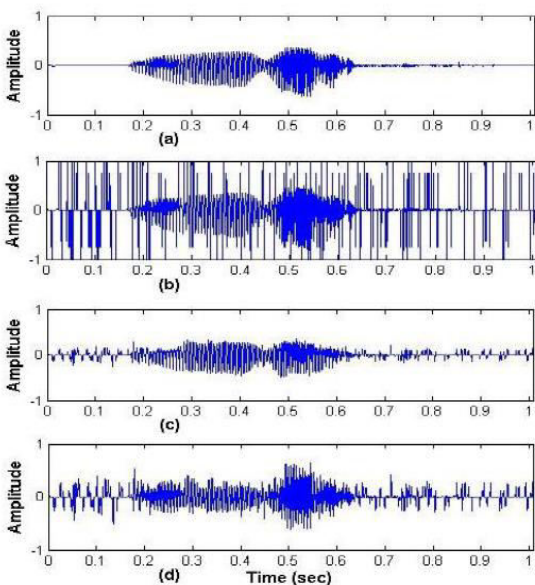


Fig 5: Total variation denoising.

Table 4: Objective Quality Measures for Spectral subtraction method

Spectral Subtraction	Digit	Csig	Chak	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

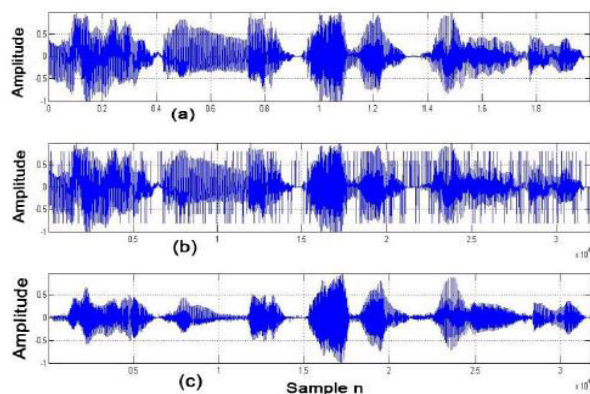


Fig 4: Spectral subtraction method.

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	SNReeg	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

5. REFERENCES

[1] S.China Venkateswarlu, K.Satya Prasad and SubbaRami Reddy, "Improve Speech Enhancement Using Weiner Filtering", Global Journal of Computer Science and Technology, Vol. 11, Iss. 7, Ver 1.0, May 2011.

[2] S. V. Vasighi and P. J. W. Rayner, "Detection and suppression of impulsive noise in speech communication systems, IEE Proc. of Communications, Speech and

Vision, vol. 137, Pt. 1, no. 12, pp. 38-46, February 1990.

[3] Charu Chandra, Michael S. Moore and Sanjit K. Mitra, "An efficient method for the removal of impulse noise from speech and audio signals, IEEE Proc. on Circuits and Systems, vol. 4, no. 8, pp. 206-208, June 1998.

[4] Zhiyong He, Xuhong Guo, Maoqing Zhang, "Detection and Removal of Impulsive Colored Noise for Speech Enhancement," IEEE Proc. on Information and Automation, pp. 2320-2324, June 2010.

[5] Mital A. Gandhi, Christelle Ledoux, and Lamine Mili, "Robust Estimation Methods for Impulsive Noise Suppression in Speech," IEEE Proc. on Signal Processing and Information Technology, pp. 755-760, December 2005.

[6] Sina Zahedpour, Soheil Feizi, Arash Amini, Mahmoud Ferdosizadeh, and Faroh Marvasti, "Impulsive Noise Cancellation Based on Soft Decision and Recursion," IEEE Trans. on Instrumentation and Measurement, vol. 58, no. 8, pp. 2780-2789, August 2009.

[7] R. C. Nongpiur, "Impulse Noise Removal in Speech Using Wavelets," IEEE ICASSP, pp. 1593-1596, 2008.

[8] Massimo Fornasier and Holger Rauhut, "Compressive Sensing," April 18, 2010.

[9] Emmanuel J. Cands, and Michael B. Wakin, "An Introduction To Compressive Sampling," IEEE signal processing magazine, March 2008

[10] K. P. Soman and R. Ramanathan, 2012, "Digital Signal and Image Processing- The



sparse way", ELSEVIER Science and Technology book.

[11] Xiuzhi Guan, Yulong Gao, Jian Chang and Zhongzhao Zhang, "Advances in Theory of Compressive Sensing and Applications in Communication", 2011, First International Conference on Instrumentation, Measurement, Computer, Communication and Control, pp.662-665, 21-23, October 2011.

[12] Emmanuel Candès and Justin Romberg "L1 magic: Recovery of Sparse Signals via Convex Programming", Caltech, October 2005.

[13] Monica FIRA, Liviu GORA, Constantin BARABASA, Nicolae CLEJU, "On ECG Compressed Sensing using Specific Overcomplete Dictionaries," Advances in Electrical and Computer Engineering Volume 10, Number 4, 2010.

[14] Hu, Y., Loizou, P. C., "Evaluation of Objective Quality Measures for Speech Enhancement". IEEE Trans. on audio, speech and language processing, Vol. 16, No. 1, pp. 229-238, January 2008.

[15] Kamil K. Wójcicki, Benjamin J. Shannon and Kuldip K. Paliwal, "Spectral Subtraction with Variance Reduced Noise Spectrum Estimates," Signal Processing Laboratory Griffith University, Nathan Q4111, Australia, March 1984.

[16] Upadhyay, Navneet, Karmakar and Abhijit, "The spectral subtractive-type algorithms for enhancing speech in

noisy environments," 2012, 1st International Conference on Recent Advances in Information Technology (RAIT) , pp.841-847, 15-17, March 2012.

[17] Miyazaki. R, Saruwatari. H, Inoue. T, Takahashi Y, Shikano K and Kondo. K, "Musical-Noise-Free Speech Enhancement Based on Optimized Iterative Spectral Subtraction," IEEE Transactions on Audio, Speech, and Language Processing, 2012.

[18] Michael S. Moore and Sanjit K. Mitra, "Statistical Threshold Design for the Two-State Signal Dependent Rank Order Mean Filter", Department of Electrical and Computer Engineering, University of California , Santa Barbara.

[19] Ivan W. Selesnick and Ilker Bayram, "Total Variation Filtering," February 4, 2010.

[20] G. R Vogel and M. E. Oman, "Iterative methods for total variation denoising", SIAM J. Sci. Comput.