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A novel and integrated architecture for identification and cancellation of noise from GSM signal

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ABSTRACT

There are multiple reasons for the evolution as well as the presence of noise over transmitted GSM signal. In spite of various approaches towards noise cancellation techniques, there are less applicable techniques for controlling noise in acoustic GSM signal. Therefore, the proposed manuscript presents an integrated modelling which performs modelling of noise identification that could significantly assist in successful noise cancellation. The proposed system uses three different approach viz. i) stochastic based approach for noise modelling, ii) analytical-based approach where allocated power acts as one of the prominent factors of noise, and iii) wavelet-based approach for effective decomposition of GSM signal for assisting better noise cancellation technique followed by better retention of signal quality. Simulated in MATLAB, the study outcome shows that it offers a cost-effective implementation, A Practical Approach for Noise identification, and Effective Noise Cancellation with Signal quality retention. The proposed system offers approximately 24% of enhancement in noise reduction as compared to any existing digital filters with 1.6 seconds faster in processing speed.

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1. INTRODUCTION

Since last few years, the GSM-based wireless communication has been accounted for emerging growth in the telecom industries. The GSM was introduced as a progression of second-generation cellular technology specified with digital modulation service. At present, the development of the GSM standard has reached the level of meeting daily needs of users and enterprises by providing cost-effective voice services as well as efficient data services which can be accessed 24x7 irrespective of user's location [1]. GSM technology supports various features for its global acceptance and rich popularity [2]. Such features are like it has efficient spectrum, good voice quality service supports low-cost cellular devices, compatible with ISDN and new services and provides roaming services globally. With the evolution of GSM, there are many advances made in digital devices, such as personal digital assistants, PCs, mobile phones, wireless LANs, etc [3]. These devices are enabled with the support of cellular communication module in order to deliver on-demand services and entertainment in various fields of application such as schools, office, healthcare, transport, Industrial area, and many more [4].

In a cellular communication system, the speech and data information transmitted via a radio link communication channel where the quality of transmitted data suffers from many degradation factors such as background noise and channel interferences [5]. The 'term' noise and interference basically refers to unwanted destructive signals introduced into use-full speech and data signals. The sources of noise are varied in nature it can be generated from an environmental factor such as acoustic disturbance form traffic, blowing the engine, wind, loud music, crowed, etc. as well as from mechanical systems such as quantization, channel interference, humming, and handoff. The occurrences of noise in use-full signals affect the quality and intelligibility of the speech and lead to cause call drop factor, instability in voice and data services and making data reception difficult. It has been found that some of the research works have addressed different sources and levels of noise in which some of them are avoidable and some are them are unavoidable [6].

The researchers have presented various techniques to handle such kinds of problem in the cellular system. The avoidable type noise can be recognized and eliminated by utilizing various and different techniques such as speech processing algorithm, noise filtering and classification technique [7-9]. Whereas, unavoidable type noise (signal-fluctuation) can be controlled by bandwidth adjustments technique and signal averaging mechanism. But still, it is very challenging for the researchers and the practitioners to deploy an efficient and robust technique to meet the user satisfaction and quality of experience. It is also been seen that as the numbers of users increases, the demand of quality of voice and data service also increases and therefore due to the constraint of the spectrum and network resource the telecommunication operators also felt a challenge in terms of market competency and user expectation form their services [10].

Therefore, there is need of efficient noise removal technique form the researchers in order to meet the user expectation and upcoming demand of supply because wireless communication market is still growing very fast where different wireless-communication based applications will require divers feature and characteristics. In the future, the 5G will introduce as a new communication technology, which will transform the pattern of existing communication systems in many advanced applications area (healthcare-system, real-time systems, automation industries, etc). Therefore, this paper has presented a solution towards denoising of the noises appearing in GSM signal. Discussion of the literature has been carried out in Section 1.1 followed by brief highlights of identified research problems in Section 1.2. Proposed methodology of denoising GSM signal is briefed in Section 1.3. Section 2 illustrates the proposed algorithm design and its implementation while result discussion is carried out in section 3. Finally, the summary of the paper is given in Section 4.

This part of the study is a continuation of our prior review [11]. The work carried out by Norholm et al. [12], have explored noise elimination issue in the time domain and used covariance matrices and optimal filtering approach for single-channel noise minimization. The performance analysis of the presented system is compared with the Wiener filter in terms of SNR. Bertrand et al. [13] adaptive noise minimization algorithm based on an overlap-add mechanism for speech enhancement and delay reduction in the acoustic wireless sensor system. Sayoud et al. [14] focuses on the problem of acoustic noise and presented dual-channel least mean square (LMS) based noise reduction procedure for speech enhancement. The study outcomes result in the superior performance of the presented technique in terms of SNR, MSE, and cepstral distance when compared to traditional similar approaches.

Similarly, the work of Rahima et al. [15] used a joint approach based on blind signal separation and adaptive transversal filtering technique for the reduction of acoustic noise and speech enhancement. Lehmann et al. [16] have designed a modified version of Edge receiver in order to reduce co-channel interference effect for the random linear modulation. The work of Hamida and Amrouche [17] has studied the performance of Echo canceller system in presence of noisy channel. Sadok et al. [18] have introduced improved noise canceller system for single antenna interference using Volterra filters up-to 3rd order. The study of Vihari et al. [19] has carried an assessment of various noise reduction approaches and performed performance analysis with respect to different noise category and SNR.

Kalamani et al. [20] used improved LMS dependent noise minimization technique for speech signal enhancement. Through performance analysis presented mechanism found to be effective in terms of PSNR and MSE. Upadhyay and Jaiswal [21] use an iterative noise assessment procedure with Wiener filtering approach for enhancing single channel speech. Premananda and Uma [22] have introduced a psychoacoustic-based gain regulator for background noise removal and speech signal improvement in mobile telephony communication. Another work carried by Shukla et al. [23] have designed an echo cancellation system based on threshold filter for improving voice signal in a handsfree communication environment.

The work of Afroz et al. [24] and Gupta et al [25] have conducted a comparative analysis of different adaptive filter for evaluating its performance for speech enhancement in terms of PSNR, SNR, and MSE. Gbadamosi et al. [26] designed signal-denoising framework based on Fourier transform and non-parametric modeling for noise elimination in GSM speech signal. Bitzer and Rademacher [27] have introduced two-way approach i.e fingerprint mechanism for detection and interpolation algorithms for cancellation of bumblebee noise. Mahbub and Fattah [28] presented gradient and adaptive LMS based acoustic echo cancellation system for voice signal enhancement.

Wang et al. [29] and Chen [30] have also concentrated on background noise problem and presented an integrated speech enhancement framework based on dual microphone array, H_2 estimator, and speech modeling system for providing a clean signal in a mobile communication device. Shakeeb and

Sayidmarie [31] have presented the configuration of a cellular base station antenna for erratic coverage. Kollem et al. [32] have illustrated a viewpoint which is modified parameter in S-Gradient Histogram protection denoising technique. Awad [33] have demonstrated a novel approach for renovating images deformed by fixed-valued impulse noise.

The significant research problems are as follows:

- Existing research techniques are more focused on developing a sophisticated and new filter which is not only expensive but also effective for a specific set of noises.
- There are fewer research works being carried out considering potential noise problems in GSM signal which is characterized by different forms of noises.
- The inclusion of stochastic characteristics over a noise is something that has never ever been considered in the existing system for which reason the solutions are not applicable for real-time application.
- Existing approaches are more specific to targeted noise only whereas in reality there are various types of noises that affect the GSM signal.

Therefore, the problem statement of the proposed study can be stated as "Developing a noise filtering mechanism that can perform robust identification of the dynamic behavior of noise present in GSM signal and offer a cost-effective solution to mitigate it".

The proposed work is a continuation of our prior implementation [34, 35] where an enhancement has been carried out by constructing an integrated model in order to emphasize on the noise-related problem over GSM signal. The overall architecture of the proposed system is as follows in Figure 1.

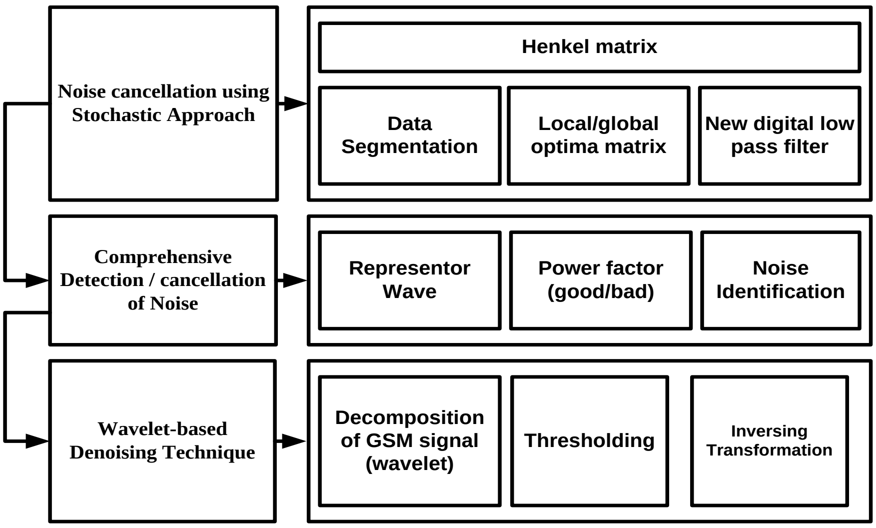


Figure 1. Proposed Architecture of GSM Signal Denoising

The prime design concept of the proposed system is that elimination of noise is one of the most challenging tasks especially if its GSM audio signal as there is always a presence of noise on the progress of time of communication from source to the destination device. Hence, the proposed system offers an integrated architecture that progressively reduces the presence of different types of noises present in the GSM signal. The upper block of the architecture introduces a stochastic modeling approach where the noises are offered a simple feature on the basis of local and global optima in order to assists in identification. The logic behind this is if the identification process is rectified than cancellation process has to precise anyhow. The middle block of architecture uses a novel concept of representing wave where the GSM signal is characterized with respect to power factor computed from its base station. Hence the noise identification is carried out with respect to power feature.

The final / bottom block of the architecture implements wavelet transformation scheme as it can decompose the signal without actually affecting the quality of the signal. The contribution of the proposed scheme is that the upper layer addresses random noise, middle layer addresses Gaussian noise, while the bottom layer addresses TDMA noise. Therefore, the proposed system offers the enhanced capability of identifying different variants of noise in GSM audio signals and is capable enough to eliminate it at the end of the denoising process. The next section discusses algorithm implementation.

2. SYSTEM DESIGN

The core intention of the proposed system design is to explore the extent of possible noise in the GSM signal followed by a successful reduction of the noise without distorting the quality of the received signal.

2.1. Stage-1: Noise cancellation using stochastic approach

This is the first stage of proposed model design that emphasizes the generation of noise followed by a reduction of the noise. For this purpose, a Henkel matrix is constructed which acts as a repository for GSM signals and segmented data. The segmented data is then classified into noisy signals and noise-free signals. The noise-free signals are then transformed to an individual signal retaining local optima matrix which is aggregated for constructing global optima matrix. Finally, the noise-free signal is applied with the stochastic based approach where the designed new low pass filter is applied for the elimination of the noise.

One of the interesting contributions of this operation is that it is capable of identifying the residual noise even after noise cancellation operation has been carried out. According to this concept, a stochastic process is introduced in the proposed system in order to reduce the artifacts of the noisy components. The proposed system also applies a digital filter in order to carry out normalization along with smoothening operation over the data with an aid of the statistic attribute. For this purpose, the proposed system makes use of the digital filters that are also dependent on the size of multiple frames along with degrees of polynomial function. The proposed system, therefore, uses a unique noise cancellation technique as shown below:

Algorithm for Noise Cancellation
Input: A (Audio File)
Output: A_{den} (denoised audio file)
Start
1. *init* A
2. A→$f_1(x, F_s)$
3. Noise→2*rand(x)-1
4. *State* SelectedFilter Case
5. **If** (condition=true)
6. H_d→$gen_{lpf}(F_{pass})|gen_{hpf}(F_{pass})|gen_{bpf}(F_{pass}))$
7. **End**
8. *State* SelectedFilter Case
9. A_{den}→$f_2(H_d)$
End

Algorithm Operation: This algorithm takes the input of GSM file which is in the wave format *A* (Line-1). A specific function $f_1(x)$ is constructed. This further process the input of the GSM signal in order to obtain a sampled data x and sample rate f_s (Line-2) followed by construction of hypothetical random noise over the sampled data (Line-3). A function *rand* () is applied oversampled data x in order to generate this random noise. The next part of the algorithm is about applying different forms of the filter in the form of use case selected filter (Line-4) that offers the user select a mechanism to apply multiple conditions of the filter.

In the case of the *low-pass filter*, the system checks if the low pass frequency F_{pass} is more than 0 first. If the Fpass value is found to be less than (F_s/2-30) than start and stop frequency is configured. This operation is followed by applying a discrete function $gen_{lpf}(x)$ considering input arguments as start and stop frequency and sample rate F_s (Line-6). A similar method is also applied for *high-pass filter* where the condition applicable for it is if the value of F_{pass} is less than (F_s/2-30) followed by applying a specific function $gen_{hpf}(x)$ while the same process is also carried out to construct a band-pass filter using gen_{bpf} (Line-6). Finally, the process of inversion is initiated in order to generate a noisy signal as follows:

$$A_{noise}=mo\ (1/10pow\ (SNR/10))\ *(Noise-mean)\ *std\_dev(x)/\ std\_dev\ (Noise)$$

In the above expression, the variable Noise is constructed by applying a respective filter over H_d and *Noise* component while the standard-deviation of sampled data and noise is also used for constructing the noisy signal. The final step is to apply another function $f_2(x)$ to the noisy signal in order to obtain denoised signal A_{den} (Line-9). The essential contribution of this algorithm is that it assists in identifying the presence of random noises and it is also capable of minimizing the level of noise. Another significant contribution of this algorithm is that it offers a one-window framework that is capable of normalizing any form of speech/audio file in the GSM signal with customized low frequency as well as it also offers the capability to cancel noise against any specific signal-to-noise ratio value. Hence, a simplified stochastic noise cancellation approach is implemented in this stage of research work.

2.2. Stage-2: Comprehensive detection/cancellation of noise

This stage of work is basically an extension of the prior stage where a random waveform is generated in order to represent the denoised signal obtained from the prior step. It improves the prior operation by considering the power factor associated with the GSM signal that is generated by the moving vehicle. This part of the implementation considers the power factor associated with good GSM signal along with higher feasible power score demand. The proposed system makes use of time as well as frequency-based transformations system in order to implement the power factors. Finally, a revised algorithm is implemented in order to perform detection of noise whose significant steps are as discussed below:

Algorithm for obtaining a transient signal

Input: *s* (source node), f_3(function for adding noise), A_{den} (speech signal)
Output: A_t (transient signal)
Start
1. $s \rightarrow f_3(A_{den})$
2. If $\alpha < T$
3. flag *a*;
4. Else
5. flag *b*;
6. $A_t \rightarrow f_4(\sigma)$
End

Algorithm Operation: The algorithm starts with the baseline of constructing a unique attribute called as *represent* which is a typical amount of GSM signals collected from the source node. The empirical computation of the represent can be carried out by dividing the power quantity of good signal by power quantity of noisy signal. However, for better precision in identifying the GSM signals, the proposed system revised the power factor for a noisy signal with maximum power at a regular time interval. After obtaining the representator pattern, the second-order attributes are extracted too. The proposed system considers primary and secondary attribute as the minimal value of the different variants of the GSM signal and corrupt signal. The proposed system carries out the minima estimation with an aid of a thresholded value where all the numerical values lesser than thresholded value is obtained.

The evaluation of the proposed algorithm is carried out considering a frequency of 2.4 GHz and this configuration represents a window of 1ms slots of time. The proposed algorithm emphasizes over encapsulating the quantity of the transient GSM signal that has the lowest duration due to the frequency of granular sampling factor. The implementation of the algorithm is carried out consider 20ns of frequency. The implication of the algorithm results in initiating communication between the source devices in order to precisely configure the GSM signal. The proposed study makes use of additive white Gaussian noise for the purpose of including the transient noise in the GSM signal that is continued by computing all sorts of the error rates. The system also carries out the identification of a class of the noise in order to perform indexing of the signal that is extracted.

The proposed system performs an evaluation of the noise content within the duration of the 1 millisecond that is further followed by drafting a conditional statement for both the category of noise for the purpose of performing the superior form indexing of the measurement signals associated with the GSM noise connected with the transient noise. The initial conditional statement offers a suggestion that in case the rate of error is found to be lower than the thresholded value than it is assumed to be a superior form of the GSM signal. On the other hand, if the rate of error *e* is found to be higher than the thresholded value than it is still considered as an error-prone signal. The rate of error as a specific form of the rate that has surfaced owing to the other types of errors that are corrected. Once the attribute of the minima, as well as a summation of attributes of second minima, are obtained via normal unit evaluation, the proposed system ultimately evaluates all these values.

Finally, the algorithm generates a transient GSM signal using function f_4(x) that uses input arguments as the duration of time, amplitude, unit step function, rinse time, frequency. One of the interesting usages of this algorithm is its threshold value *T*, which can be altered as per different speech-based application. Therefore, this algorithm can be suitably used for identifying different types and levels of noise in the GSM signal and it can also differentiate good signal from the bad signal. The proposed study also assumes that there is two distinguished form of GSM signal of transient characteristics that are considered to be extracted from the fast-moving vehicle. Exactly, a similar form of the time interval is used that is reported to be used in GSM signal as the orientation base is considered for the proposed study. The proposed algorithm also carries out an equivalent computational operation that is also reported to be applied towards the deviation of the signals to noise ratio. Therefore, the proposed system is capable to identify and eliminate the noise elements from the overall contents of the noisy GSM signal in a much cost-effective manner. The algorithm is applicable for identifying and eliminating noises from all GSM signals static or transient.

2.3. Stage-3: Wavelet-based denoising technique

The prime objective of this stage of implementation is applied to a wavelet-based transformation technique for assisting in the elimination of noise from one dimensional GSM signal to reconstructed signal free from noise. It is a fact that there is a presence of real-time noise over all the ranges of frequencies and hence it is quite challenging to eliminate the noise from the GSM signal. One of the advantages of using wavelet-based transformation scheme is that it is capable of addressing denoising problems of almost all types of noises e.g. electronic noise, electromagnetic noise, acoustic noise, and electrostatic noise. In order to perform denoising on GSM audio signal, the proposed algorithm is subjected to series of mechanism viz. decomposition of GSM signal, applying different variants of thresholding mechanism, and finally reconstruction of the original signal by an elimination of noise. The flow of the algorithm is to consider the original signal which is further corrupted by standard white noise followed by decomposition technique and thresholding. The reconstruction is obtained after performing an inverting operation on wavelet transformation. The significant steps of the proposed algorithm are as follows:

Algorithm for wavelet-based denoising
Input: A_t (transient audio signal)
Output: A_{densig}(denoised signal)
Start
1. $[t_{sig}\ F_s\ n_b] \rightarrow f_1(A_t)$
2. $t_{sigN} \rightarrow$ noise($amp*t_{sig}$, SNR)
3. Select $f_4(x) \rightarrow$ wt
4. $[a_1\ a_2\ a_3\ a_4] \rightarrow f_5(\mathrm{wt})$
5. $[b_1\ b_2] \rightarrow f_6(t_{sigN}, lev, a_1, a_2)$
6. $[c] \rightarrow f_7(b_1, b_2, \mathrm{wt}, lev)$
7. $[d_1\ d_2\ d_3] \rightarrow f_8(b_1, b_2)$
8. $[E_1, E_2, E_3, E_4] \rightarrow f_9(c, b_1, b_2, a_3, a_4, n)$
9. Select $t_{selRule}$
10. $Th_1 \rightarrow th_{Sel}(E_2, E_3, E_4, [t_{selRule}])$
11. $th_n \rightarrow f_{10}(E, th_{op}, th_1)$
12. $A_{densig} \rightarrow E_1 + th_n$
13. $err \rightarrow \arg_{max}(|t_{signN} - A_{densig}|)$

End

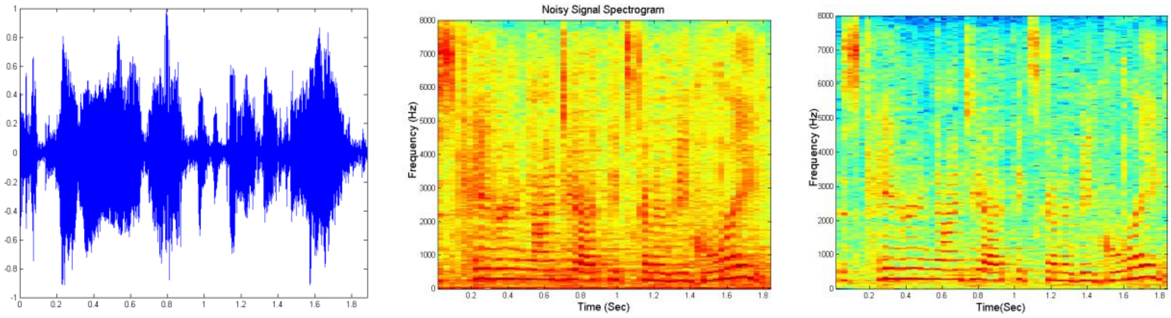
Algorithm Operation: This above-shown steps of algorithm perform the application of discrete wavelet transforms in order to perform denoising operation on the GSM speech signal. Similar function $f_1(x)$ is applied to the transient audio signal A_t (Line-1) in order to obtain true signal t_{sig}, sample rate F_s, and a number of bits n_b. The next step of this algorithm is to add up an additive white Gaussian noise over the true signal that is obtained from product of initialized amplitude *amp*, signal-to-noise ratio SNR, and t_{sig} value (Line-2). Hence, a noisy signal t_{sigN} is now obtained. The next process is to apply wavelet decomposition using a discrete function $f_4(x)$, which can be any form of transformation in order to obtain different forms of wavelets *wt* (Line-4). This process is further followed up by applying a function $f_5(x)$ which performs a filtering process for the given wave and leads to the generation of four different forms of coefficients decomposed low and high pass filter (a_1 and a_2) while reconstructed low and high pass filter (a_3 and a_4) (Line-4). Decomposition of the wavelet is carried out using function $f_6(x)$ on noisy signal t_{sigN} and decomposed coefficients of filters a_1 and a_2 with respect to the given range of level of decomposition *lev* (Line-5).

This operation leads to the generation of decomposition and bookkeeping vector b_1 and b_2 (Line-5). Approximation of the coefficient is then carried out using function $f_7(x)$ over b_1 and b_2 as well as with wavelet *wt* with respect to the defined level of decomposition *lev* (Line-6). Extraction of the detail coefficient is then carried out using a function with respect to b_1 and b_2 in order to generate multiple coefficients c (Line-7). Finally, the reconstruction process is carried out over b_1, b_2, and reconstructed coefficient filter a_3 and a_4 using function $f_9(x)$. The proposed system also applies a discrete selection of threshold strategy $t_{selRule}$ which is applied over E_2, E_3, and E_4 (all reconstructed coefficient) (Line-10-11). Finally, the proposed algorithm applies two distinct forms of thresholding i.e. hard and soft thresholding using function $f_{10}(x)$ (Line-11). Although, hard thresholding is one of the easiest methods of implementation for controlling threshold factor still the proposed system advocates the use of soft thresholding mechanism as it offers better compatibilities with mathematical properties. The acquisition of denoised signal A_{densig} is obtained by summing up a reconstructed coefficient filter at the 3rd level of approximation and the newly obtained threshold value (Line-12). The proposed system also computes the error value *err* by obtaining the difference of noisy signal t_{sigN} with denoised signal A_{densig} (Line-13).

3. RESULT ANALYSIS

The implementation of the proposed study has been assessed over normal windows system where MATLAB is used for scripting the algorithm. A call recording app was used for extracting the voice signals over the 4G network whose size is 60 kilobytes and bitrate is 256 kilobyte per seconds. The overall GSM signal content of audio is around 3 seconds, which is basically of wave file format which can be easily processed by MATLAB using its inbuilt function.

Figure 2(a) highlights the spectrum of the original GSM signal, which when applied to algorithm-1 yields an outcome of the denoised image. The spectrometer of the GSM signal corrupted with noise is shown in Figure 2(b), while the denoised signal in the form of the spectrometer is showcased in Figure 2(c). The spectrometer visualization exhibits that maximum i.e. 85% of noises are actually removed in this phase itself. This output of the denoised signal when subjected to the algorithm-2 yields further better noise reduction performance. Algorithm-1 was focused more on random noise elimination while algorithm-2 was more focused on the elimination of white Gaussian noise. This can be seen in the outcome of Figure 3 which is basically a scattered plot to show that the density of the noise-related components is significantly getting reduced with the progress of the sum of minima. Considering a thresholded value of 1ms duration, the second algorithm offer around 92% of elimination of the noisy components from the GSM signals. Apart from this, the processing of the proposed algorithm is also found to be yielding a result in much faster track irrespective of any forms of cut-off values being used in the analysis.



(a) (b) (c)

Figure 2. Visual outcomes of stage-1 implementation

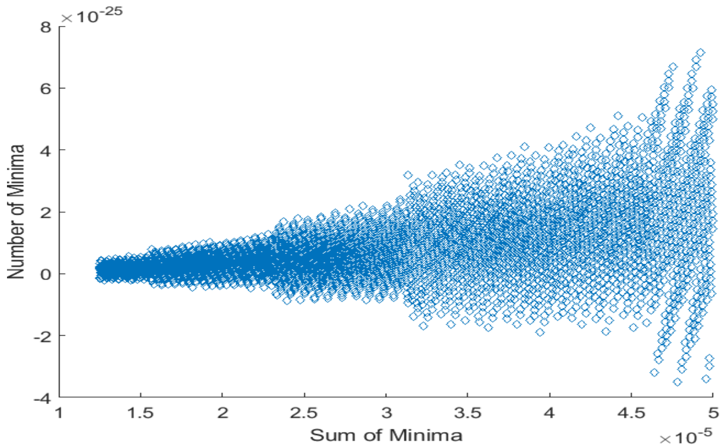


Figure 3. Visual outcomes of stage-2 implementation

The prime reason behind this is the usage of the represent that offers the better capability to distinguish between signals. Another interesting fact about this outcome is that complete outcome can be analyzed just from the signals received from the base station without having any dependencies to capture any GSM signals right from the moving vehicle. This operation makes a faster analysis of the noise in the GSM signal with more preciseness to detect the spurious signal.

Figure 4 highlights the visual outcome of the visual outcome of the algorithm-3where it is shown that the original GSM signal is further subjected to TDMA noise. This part of the analysis considers TDMA noise as it includes various other forms of noises too within a GSM signal. The visual outcome also shows the outcome waveform as well as spectrogram to be free from TDMA noise. For an effective analysis, the proposed system is also subjected to comparative analysis with an existing system.

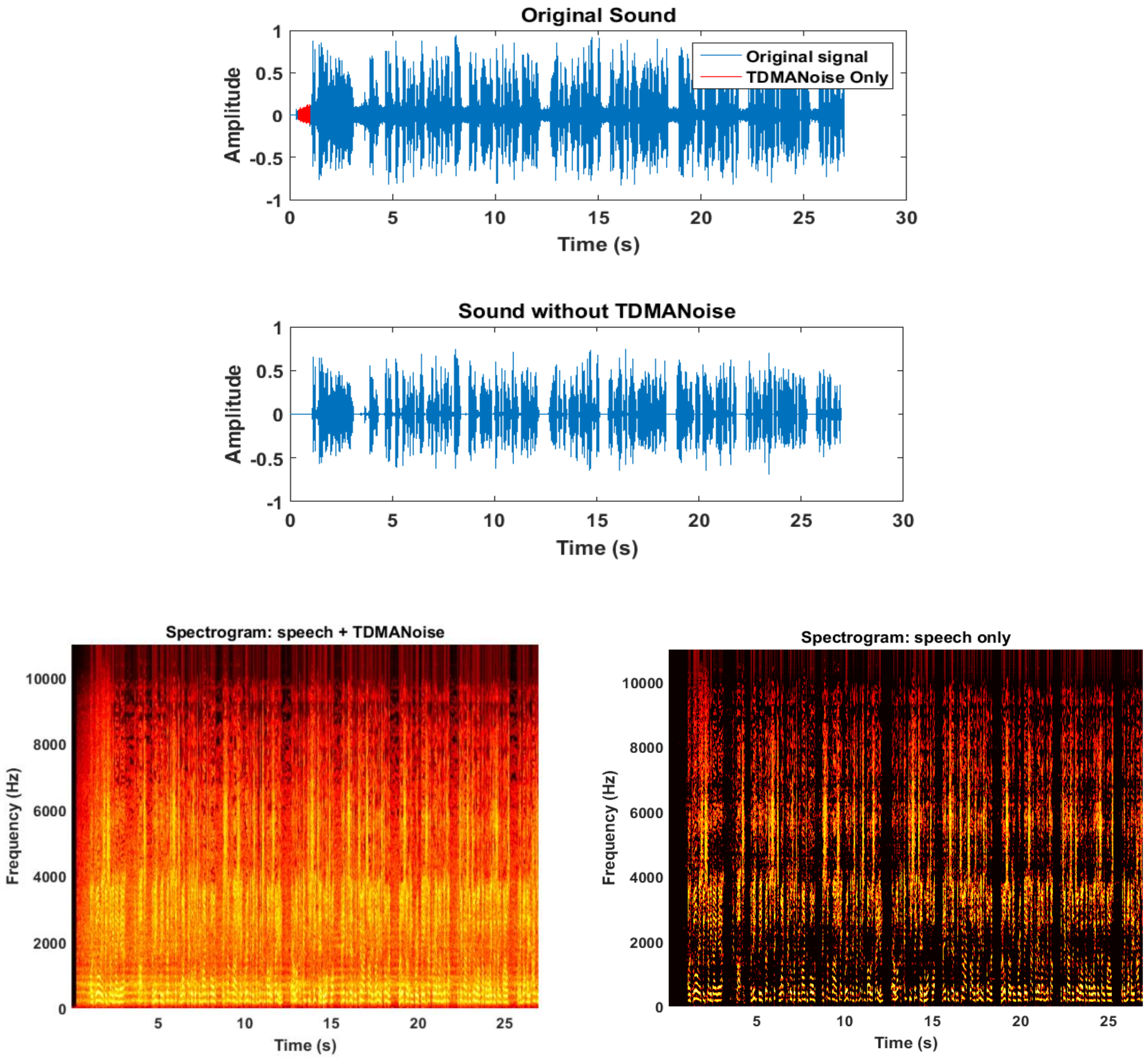


Figure 4. Visual outcomes of Stage-3 implementation

Table 1 highlight that the proposed system offers better noise reduction performance as compared to all the major denoising approaches in existing approaches. Apart from percentile of noise reduction parameters, the proposed system is also witnessed to offer faster processing time in core i7 process. Even in lower processor version, then the difference in noise reduction in terms of SNR is very much negligible. Hence, the proposed system can be claimed to offer highly extensible, cost-effective, and practical denoising approach using one window operation in order to address the noise problems in GSM signals.

Table 1. Comparative analysis

Sl. No.	Denoising Approaches	% of Noise Reduction	Processing Time (s)
1	Butterworth Filter	75.21%	1.982
2	Chebyshev filter	67.98%	2.5611
3	Elliptical filter	73%	1.4882
4	Weiner filter	71%	0.9672
5	Adaptive Noise cancellation	82%	2.6771
6	Proposed System	98%	0.3287

4. CONCLUSION

Noise cancellation from the acoustic GSM signal is definitely not an easy task. At present, basically, the hardware-based approaches are used for ensuring that there is no noise embedded with the transmitted GSM signal. However, there are less effective modeling found for this reason. Therefore, this paper presents an integrated modeling where the noise cancellation is done in a progressive manner. The novelty of this paper are i) multiple forms of noise e.g. random noise, white noise, TDMA noise are possible to be identified and canceled, 2) the implementation of proposed system is not based on any sophisticated filter with higher resource dependencies and hence it is cost effective, 3) apart from noise cancellation, proposed system offers higher retention of signal quality too, and 4) proposed system offers a highly practical approach where power allocation over transmitting device is considered as one prominent point in noise cancellation.

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