# Improved Noise Suppression Technique for Digital Hearing Aids

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## Abstract
Background noise in digital hearing aid is the most common complaint. Many efforts have been put in recent years aiming to suppress background noise effectively. Adopting wiener filter is one such method in suppressing background noise by considering both the priori and posteriori signal to noise ratio to calculate the wiener gain. This paper presents Improved Noise Suppression Technique (INST) a new method for background noise suppression. The improved noise suppression technique incorporates both the features of wiener filter based on decision directed approach and two step noise reduction technique (TSNR).
Extensive MATLAB simulations, study and analysis of the results shows that Improved Noise Suppression Technique also offers good noise suppression capability.



## Keywords
Digital Hearing Aid, Wiener filter, Noise suppression, Subband domain signal processing. Feedback cancellation, Speech enhancement, Signal to Noise Ratio.


## I. Introduction
Frequency dependent amplification to hearing impaired people is one of the major objectives of the Digital Hearing Aids. As the Digital Hearing Aid has an external speech processor unit, it also performs suppression of background noise, dynamic range compression and feedback cancellation.
Subband domain signal processing for digital hearing aid is a better choice since hearing aid requires frequency dependent amplification.
Use of lapped transform will results in larger rejection of side lobes when compared to block transform [22]. The some of the block transformation techniques are DFT, DCT etc., and some of the lapped transformation techniques are MLT, AMLT etc., The analysis are given in paper by considering 24 subbands.
The feedback leakage signal which interferes with the unamplified sound at the microphones changes the sound quality. An adaptive filter is used to continuously estimate the feedback path and cancel it in the forward path. Siqueira et al., (2000) have studied and presented analytical study state convergence behavior of adaptive algorithms in adaptive feedback cancellation in digital hearing aids. The feedback cancellation in hearing aid can be achieved with LMS or NLMS algorithm. Steady state behavior of NLMS algorithm is better when compared to LMS algorithm, since the variable step size control of NLMS algorithm.
Cyril et al(2006) have studied the limitations of well known Decision Directed (DD) approach for speech enhancement. By using Decision Directed approach, the background noise is efficiently suppressed, but the approach estimated a priori SNR is biased. This approach involves the speech spectrum estimation in the previous frame but not on the current frame and hence it degrades the speech quality. The authors have proposed two improved methods namely Two Step Noise Reduction (TSNR) and Harmonic Regeneration methods for speech enhancement. The proposed two methods perform well against the decision directed approach for speech enhancement and also preserve the benefits of decision directed approach.
Ashutosh et al (2011) have discussed the advantages of modern subband domain digital hearing aid system. These advantages include easy adjustments of gain in every subband, very fast adaptive filter convergence and computational savings. In modern subband domain digital hearing aid, noise suppression is achieved using wiener filter. The feedback cancellation is achieved by the use of normalized least man square (NLMS) algorithm. The use of NLMS algorithm will results in improved active noise control and hence providing added stable gain.
The adaptive filters in the modern subband digital hearing aid, involves in feedback suppression by the use of any of the block transformation like DFT, GDFT etc. LMS and NLMS are the best adaptive algorithms used to estimate the feedback path by adjusting optimum filter co-efficient. These adaptive algorithms works on the principle of least mean square value. If the error signal results in least value, filter co-efficient are said to be optimum and those filter co-efficient are used to continuously estimate the feedback path and cancel it in the forward path. If the level of the enhanced speech crosses the upper thresholds of hearing, there may be cause of damage to the ear, so dynamic range compression should be incorporated into the hearing aid to avoid painful listening. .
Speech enhancement in digital hearing aid involves the primary operation of background noise suppression. The use of improved noise suppression technique will also perform better and provides comfortable listening experience to the hearing aid user.

## II. Method

### A. Modern Subband Domain Digital Hearing Aid
Usually Wiener filter is incorporated in modern subband domain digital hearing aid for noise suppression. The block diagram in Figure 1 depicts the modern subband domain digital hearing aid. It operates with the subbands which are created with block transform. Input to the digital hearing aid is the noisy speech signal added with the feedback leakage signal from Acoustic Feedback (AF) path.

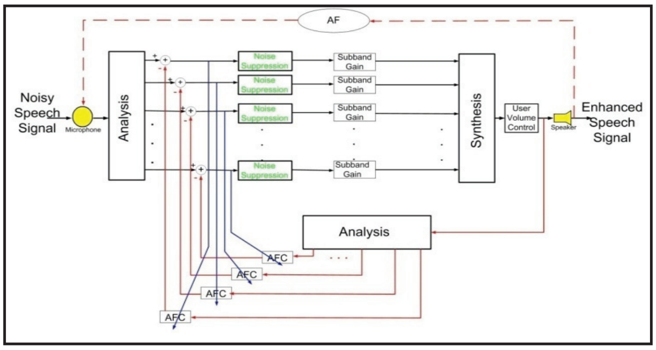


Fig. 1: Block diagram of Modern Subband Domain Digital Hearing Aid.

The signal processing is carried out to perform the operations like hearing loss compensation, noise suppression, adaptive feedback cancellation and dynamic range compression. The required subbands are created by the analysis section and same subbands are reconstructed at the synthesis section.

The hearing loss is compensated at the subband gain part and the background noise suppression is achieved at the noise suppression part from the Wiener filter which is given in fig. 2. Feedback cancellation is achieved by LMS or NLMS algorithm and the dynamic range compression is carried out step by step in each stage of noise suppression, subband gain and in user volume control.

### A. Wiener Filter

The Wiener filter is used to suppress the background noise. The mechanism of wiener filter in noise suppression is multiplying the wiener gain to the noisy speech signal. Fig. 2 shows the operations involved in the noise suppression of wiener filtering.

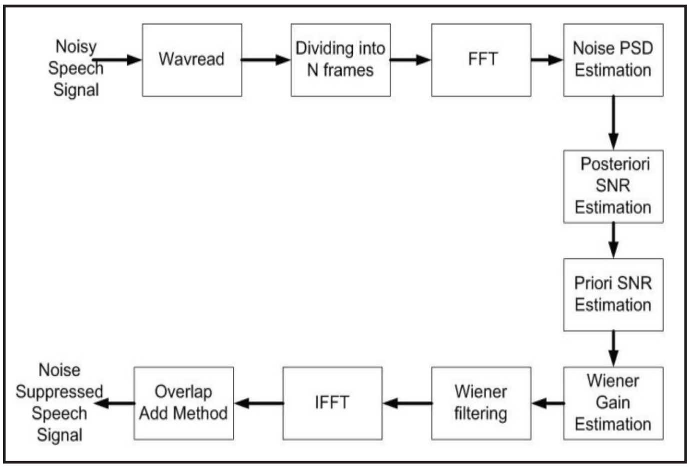


Fig. 2: Noise Suppression by wiener filter in modern Digital Hearing Aid.

The noise characteristics are estimated from the initial silence region of speech signals. Once the wiener gain is calculated, it is multiplied to the noisy speech signal in the frequency domain. Here noise suppression is achieved by decreasing the gain of the overall noisy speech signal. Since the short time energy of noisy signal is less than that of speech signal, the noise signal remains suppressed even after the amplification of noise suppressed speech signal.

### B. Improved Noise Suppression Technique (INST)

The improved noise suppression technique incorporates both the features of wiener filter based on decision directed approach and two step noise reduction technique (TSNR) and it is operated in frequency domain as that of the Wiener filter. Figure 3 shows the operations involved in Improved Noise Suppression Technique (INST) for digital hearing aid. Noisy speech signal is given as input to the INST filter which involves in dividing noisy speech signal into the N number of frames. Noise characteristics are calculated from the few initial frames. Noise Power Spectral Density (PSD) is calculated from noise characteristics. Once the Priori and Posteriori SNR are calculated, initial gain is calculated as in the Wiener filter gain. INST gain is calculated by using the initial gain and Priori SNR. As the hearing aid has to be operated in real time, the computational complexity of the INST is decreased by considering the posteriori and priory signal to noise ratio as that of the wiener filter gain.

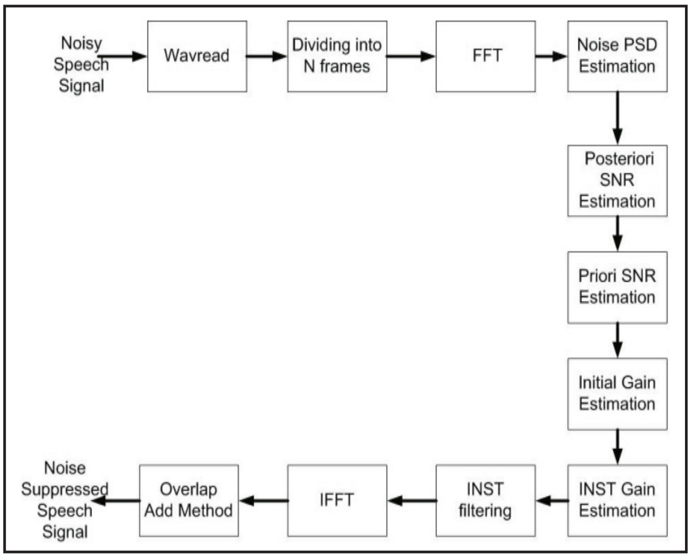


Fig. 3: Noise Suppression by INST filter in modern Digital Hearing Aid.

The computational complexity of the INST is similar to the wiener filter for noise suppression and hence the proposed INST method has the ability to operate in real time and hence providing the improved and comfortable listening experience to the hearing impaired people.

## III. Results and Discussion

For the sake of comparison we simulated subband domain modern digital hearing aid in MATLA performing hearing loss compensation, dynamic range compression, noise suppression and feedback cancellation.

The two sets of subband domain signal processing is carried out involving 4(S1) and 8(S2) subbands respectively. The experimental work was carried out for different types of background noisy speech signal taken from NOIZEUS database. The 8 sets of noisy speech signals at various SNR like 0dB, 5 dB, 10dB and 15 dB are taken as the test signals, for calculating performance of digital hearing aid.

The figure 4 shows the performance of INST and wiener filtering technique for noise suppression in hearing aid application. For the sake of performance testing we tested two methods by considering same type of noisy speech signals for both the methods.

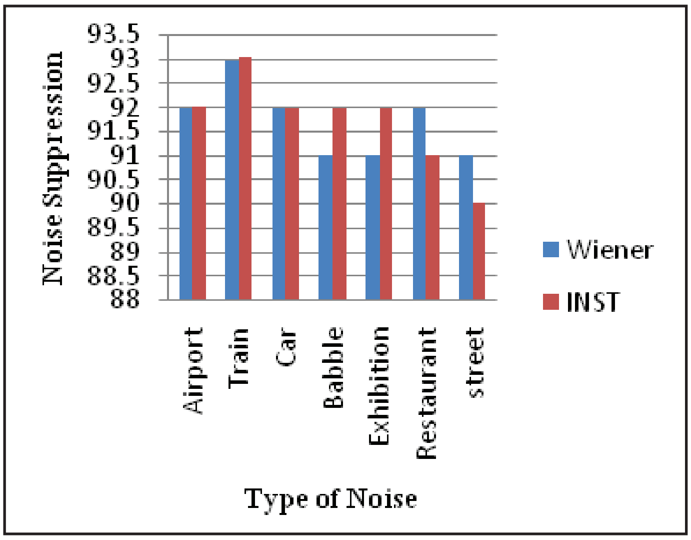


Fig. 4: Noise Suppression capability of Wiener and INST Filters

 ISSN : 2230-7109 (Online) | ISSN : 2230-9543 (Print)

The performance measure is calculated by summing up the absolute amplitude of enhanced speech signal at the output of the simulated digital hearing aid in time domain.

The analysis of the fig. 4 shows that the INST offers some increase in noise suppression capability when compared to the wiener filter in some trails.

The Fig. 5 shows the plot that will illustrates the noise suppression capability of INST over wiener filtering in modern subband domain digital hearing aid.

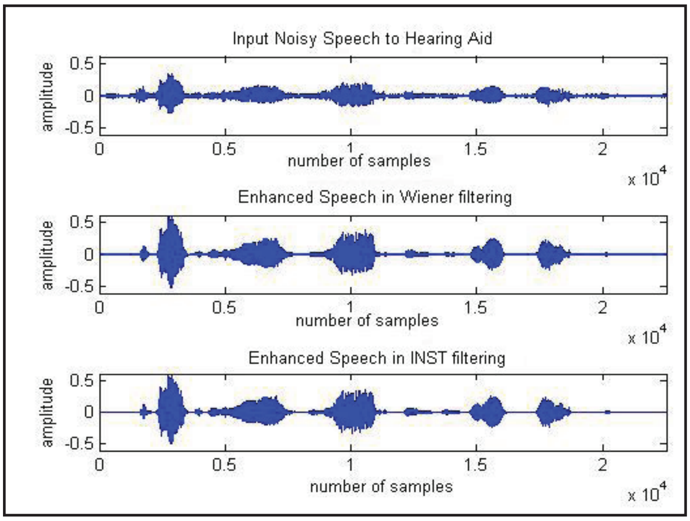


Fig. 5: Comparison of wiener and INST Filtering Technique for Digital Hearing Aid.

In Fig. 5, the first signal is input noisy speech signal to the digital hearing aid involving wiener and INST filter and second signal is the output of digital hearing aid involving wiener filter and third signal is the output of digital hearing aid involving INST filter.

## IV. Conclusion

In this paper we presented a new method for signal processing in hearing aids to suppress background noise. The performance of INST is compared with wiener filter in MATLAB simulations.

By observing MATLAB simulations and subjective evaluations of the obtained results, the INST can be used in the modern subband domain digital hearing aid for noise suppression to provide better and comfortable listening experience to the hearing aid user.

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