***Abstract*—**

In an audio speech signal, acoustic noise is a common problem while the speech is processed. Here, we are going to create color noise and add with an audio signal, after that a model are introduced to eliminate that noise.

This paper elaborates a new approach for noise cancellation in speech enhancement using an Adaptive LMS (Least Mean Square) filter and with the help of MATLAB Simulink we get the correct speech signal.

This filter is used to remove the acoustic noise due to its simplicity in computation & robust behavior when implemented in finite-precision hardware.

It provides better communication by suppressing the acoustic noise to a larger extent, since it provides a better balance between complexity & convergence speed.

In spite of various methods, the results obtained in this way of noise cancellation are optimistic.

The basic way of communication in humans to convey information or message is speech. This communication helps the humans to fulfill their basic needs with a bandwidth of 4 KHz.

Noise is an unwanted signal is the major drawback affects the speech signal. There are various sources of noise which affects the speech such as electrical, acoustic, vibration or any other noise components.

Among that, acoustic noise plays a major role since it becomes more noticeable as the number of commercial equipment increases.

Acoustics is a branch related to physics. This noise introduces a masking problem to reduce the distinct nature of speech which affects the communication . During the elimination of this acoustic noise, the change of signal characteristics becomes more common.

There are number of adaptive algorithms to remove the acoustic noise but we use Least Mean Square (LMS) filter to overcome this problem.

The reason behind this is when compared with traditional filter .

A noise is added and using adaptive LMS filter we suppress the acoustic noise leaving the speech signal unchanged.

This method improves the efficiency to a greater extent.