The existence of noise is inevitable in real applications of speech processing. In fact, background noise is one of the major factors that adversely affect the perceived grade of service in speech communication systems. It is well known that the additive noise affects mainly the performance of the system and reduces the Signal to Noise Ratio (SNR) and the speech intelligibility. A noise reduction scheme, capable of handling a wide variety of noise situations with varying characteristics and noise levels, becomes necessary.

The traditional approach to noise cancellation lay in utilizing standalone noise cancellation modules on the near-side or transmit path. This approach works well under constant conditions, but as environment changes, the performance gets degraded and the system struggles to adapt. Illiev and Kasabov (1999) proposed the noise suppression system using adaptive notch filter based on Gray-Markel lattice filter under comparable SNR improvement.

Boll and Pulsipher (1980) showed that for the application of adaptive noise cancellation, large filter lengths were required to account for a highly reverberant recording environment and that there was a direct relation between filter misadjustment and induced echo in the output speech. He had also shows that it was feasible to reduce ambient noise power by at least
20 dB with minimal speech distortion and thus to be potentially powerful as noise suppression preprocessors for voice communication in severe noise environments.

Vitterli and Gilloire (1988) and Usevitch and Orchard (1996) separately proposed a sub band filtering scheme where the expected signal and the error signal were divided into sub bands using a filterbank and each sub band was then processed independently. The conditions for the filterbank were researched by Petraglia and Alves (1997). Alves et al studied the convergence properties of the sub band adaptive filter structure. The filterbank method is more suitable for subband feedback ANC as the processing can be done on a per sample basis.

Sergiy A. Vorobyov et al (2001), proposed the adaptive noise cancellation for multi sensory signals based on a linear combination of outputs of all noise cancellation systems and proved that every channel is optimum in the adjustment of the weight parameters.

Joachim Thiemann and Peter Kabal (2002) described a noise suppression algorithm only for narrow band signals. But in wideband signals, the quality needs to be improved. Kumar Swaminathan and Srinivas Nandkumar (2003) dealt with the concept of spectral amplitude enhancement. It performs spectral filtering by using gain function. This method introduces minimal distortion.

Dong-hai Zhai et al (2003) proposed a non-linear noise canceller based on additive-multiplicative fuzzy neural networks which are used to approximate noise, and the noise approximated is cancelled from the measuring signal to obtain useful signal. This neural network model describes the learning algorithm and the universal approximation used in the model.
Israel Cohen (2004) presented a multi channel post filtering for minimizing the log-spectral amplitude distortion, in nonstationary noise environments. He proposed that a desired signal component is stronger than the reference noise signal. Shing-Chow Chan and Yue-Xian Zou (2004) derived an algorithm for robust adaptive filtering in impulsive noise. It is suitable only when the desired and input signals are corrupted by impulsive noise.

Suthikshn Kumar (2004) described the fuzzy volume controller by intelligently adjusting the volume level to improve the quality of service for both stationary and non-stationary background noise in mobile environments. The controller needs many fuzzy rules for single noise type to optimize its performance.

Zhang Yan Zhao et al (2004) mooted the adaptive noise cancellation technique using back propagation algorithm and genetic algorithm. The back propagation network adopts gradient searching. But the convergence rate was slow and the genetic algorithm used only optimized the initial weight value in the network.

Juarez-Hernandez et al (2005) discussed the concept of converting linear adaptive noise cancellation into a nonlinear technique using ANFIS. The best that ANFIS can do is to minimize the error component, attributable to desired component. The estimation error is more pronounced at certain values. Hence improvement is needed for training data from a longer interval.

Banovic et al (2005) explained FPGA based digital signal processing based on hardware logic that does not suffer from any of the software based processor performance problems. FPGAs allow applications to run in parallel and pipelined so that filtering, correlation and many other applications could be run simultaneously.
Greg Eslinger and John Dixon (2006) discussed the integration of ANC with AEC where the control signals from the noise canceller and AEC together created an environment for more accurate voice activity detection and better convergence. Without this interaction, the system may inadvertently try to cancel voice instead of noise.

Ersoy Kelebekler and Melih İnal (2006) presented White and Color Noise Cancellation of Speech Signal by Adaptive Filtering and Soft Computing Algorithms in which Gaussian white noise and color noise of speech signal are reduced by using adaptive filter and soft computing algorithms. The main target is noise reduction of speech signal in a car, ambient noise recorded.

Lin et al (2003) proffered a fast noise estimation algorithm for speech enhancement using a perceptual Wiener filter and showed that noise estimation technique gives accurate results even at very low signal-to-noise ratios, and works continuously, even in the presence of speech. It is effective for both non-stationary and colored noise.

Oscar and Patricia (2004) described the application of type-2 fuzzy logic for achieving adaptive noise cancellation to filter out an interference component by identifying a model between a measurable noise source and the corresponding un-measurable interference. The use of type-2 fuzzy logic is justified due to the high level of uncertainty of the process, which makes it difficult to find appropriate parameter values for the membership functions.