**Historical Overview:**

The first recognized work on eliminating noise from speech signal was documented in 1934 when Paul Lueg was granted the first patent for a noise control system. In his work, Paul Lueg explained his way to cancel sinusoidal tones in ducts by phase-advancing the wave and then by inverting the polarity to cancel random sounds in the area nearby an amplifier. Basically he developed a process of silencing disturbing sound oscillations or noise which was able to work independently of the main source of oscillation or speech. [1]

After this work of Paul Lueg, a lot of people started working on improving noise cancelation technique used by Paul and later in 1950, Lawrence J. Fogel designed a systems where he successfully canceled the noise in helicopter and airplane cockpits communication. This work of Lawrence J. Fogel was a milestone in the noise cancellation system development where he designed a system for supplementing a standard noise suppressor by eliminating the peaks of energy as well as the nulls of energy. He claimed his design to be able to increase the lucidity of speech in noisy environment significantly over the works done previously. [2]

Meanwhile Willard Meeker was working to implement active noise control to a circumaural earmuff and in 1957 he successfully developed a working model of his headset with an active attenuation bandwidth of approximately 50–500 Hz and a maximum attenuation of approximately 20 dB. [3]

During 1980’s due to the technological boom, travelling less and doing the meetings over telephone became an efficient way to reduce the pressure on our environment and do business more economically. At that time technology present to reduce the noise from the speech signal was not efficient enough to make a perfect teleconference without any noise interference issue. As a consequence, more private researchers and companies were interested to develop more advanced working design to cancel noise efficiently to have a completely noise-free environment for teleconference or other communication systems.

In 1980, one highly recognized work on this field was that of S. Boll of University of Utah, who was successful to efficiently suppress the acoustic noise in speech using two microphone adaptive noise cancellation system. He used least mean square (LMS) and the lattice gradient algorithms to suppressed acoustic noise with energy greater or equal to the speech by adaptively filtering a separately recorded correlated version of the noise signal and then subtracting it from the speech waveform. Both methods were able to decrease ambient noise power by at least 20 dB with insignificant speech distortion and therefore was powerful enough to work as noise suppression preprocessors for voice communication in severe noise environments. [4]

Another work of 1980’s which is still one of the best techniques used for speech enhancement is the work of Y. Ephraim and D. Malah, which was published April 1985. They have developed a highly efficient Short-Time Spectral Amplitude (STSA) estimator for speech signals to minimize the mean square error of the log-spectra. Actually this work was an enhancement of their previous work titled “Speech Enhancement Using a- Minimum Mean-Square Error Short-Time Spectral Amplitude Estimator” published in December 1984. They formulated the STSA by estimating the amplitude of each Fourier expansion coefficient of the speech signal, given the noisy process in the interval [0, T]. Their Gaussian statistical model was motivated by the central limit theorem, as each Fourier expansion coefficient is basically a weighted sum (or integral) of random variables resulting from the process samples. Eventually they were able to have lower residual noise level without hampering the speech itself taking the signal presence uncertainty into account. [5] [6]

Since then there has been lot of work going on reducing noise in speech signal. Later on 1992, G. Mirchandani, R. Zinser Jr. and J.B. Evans developed a structure for joint process estimation for noise canceling, when the reference signal is contaminated with crosstalk. They have done the simulation as per their design with the transversal and the lattice-based CTRANCs, and they have found out that for 1 to 9 dB levels of crosstalk, their design shows 3 to 11 dB improvement in the mean-square error over that for the standard adaptive noise canceller. [7]

After going through several works of various scholar of different time and research field including the works mentioned above as well, we have decided to study more on this topic and develop a design to appropriately apply noise cancellation in such a way which enables the system to extract only the voice signal removing as much noise as possible. After discussing with our project supervisor Mr. Per Zetterberg we have decided to go ahead with this project with our previous knowledge gained from the Adaptive Signal Processing course. In the adaptive signal processing course, different adaptive signal processing algorithms were unmasked to us for extracting relevant information from noisy signals where the emphasis was mainly on recursive, model based estimation methods for time-varying systems.

Xavi if you can put few key notes on adaptive signal processing here I believe we are done with the historical overview! ☺

**References:**

[1] [Online]. Available: https://www.google.com/patents/US2043416. [Accessed 25 May 2015].

[2] [Online]. Available: https://www.google.com/patents/US2866848. [Accessed 25 May 2015].

[3] [Online]. Available: http://scholar.lib.vt.edu/theses/available/etd-04222002-143554/unrestricted/02rudissertation.pdf. [Accessed 26 May 2015].

[4] S. Boll and D. Pulsipher, “Suppression of acoustic noise in speech using two microphone adaptive noise cancellation,” IEEE Trans. Acoustic Speech, Signal Processing, Vol. 28, pp 752-753, Dec. 1980.

[5] Y. Ephraim and D. Malah, “Speech enhancement using a minimum mean-square error short-time spectral amplitude estimator,” IEEE Trans. Acoustic Speech, Signal Processing, Vol. ASSP-32, pp 1109-1121, Dec. 1984.

[6] Y. Ephraim and D. Malah, “Speech enhancement using a minimum mean-square error log-spectral amplitude estimator,” IEEE Trans. Acoustic Speech, Signal Processing, Vol. ASSP-33, pp 443-445, Apr. 1985.

[7] G. Mirchandani, R. Zinser Jr. and J.B. Evans, “A new adaptive noise cancellation scheme in the presence of crosstalk,” IEEE Trans. On Circuits and Systems-II: Analog and Digital Signal Processing, Vol. 39, pp 681-694, Oct. 1992.