

A Novel Algorithm for the Reduction of Irregular Noise in Corrupted Speech Signals

ROSHAHLIZA M RAMLI, ALI O. ABID NOOR &
SALINA ABDUL SAMAD

FACULTY OF ENGINEERING & BUILT ENVIRONMENT
UNIVERSITI KEBANGSAAN MALAYSIA
(NATIONAL UNIVERSITY OF MALAYSIA)



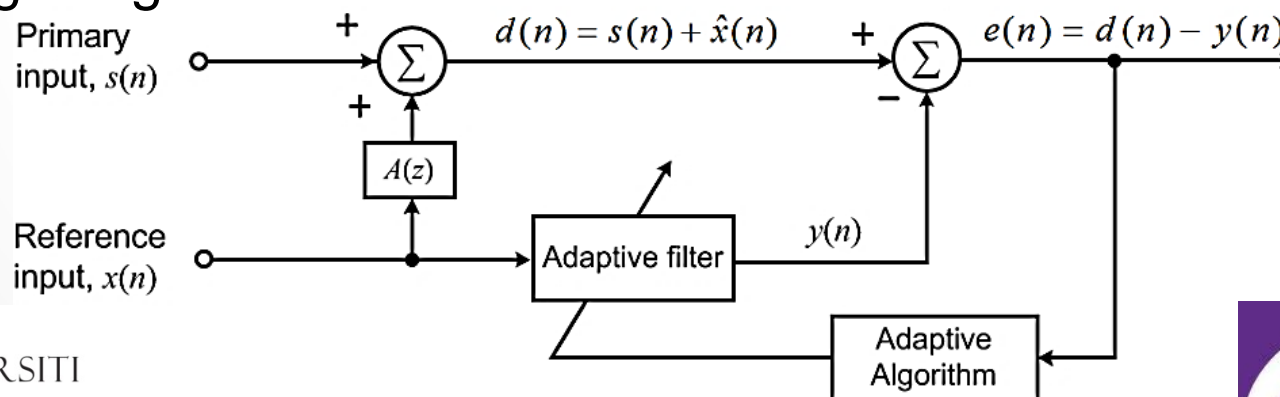
R. M. Ramli, A. O. A. Noor, S. A. Samad

1



INTRODUCTION

- Noise cancellation using an adaptive filter offers lower costs and more practical in noise suppression such as the two-sensor Adaptive Noise Canceller (ANC)
 - One or more sensors of ANC are located at a vicinity of a noisy area where the signal is weak by the reference input sensor(s)
 - Using adaptive algorithm to control the coefficients of a digital filter.
 - The adaptive filter filters out the noise and improves the quality of target signal.



- The choice of an adaptive algorithm is based on its convergence speed and computational power.
- The Least Mean Square (LMS) algorithm is common for most adaptive filters but it becomes very slow for ill conditioned input signals.
- The Recursive Least Square (RLS) and the Affine Projection (AP) algorithms showed best performances in convergence but has increased computational burden.
- Most existing literatures went too complex, merely theoretical and non-applicable in real-time
- Therefore, a **smart noise cancellation system** is proposed based on a **selective mechanism** that can be switched to apply several adaptive algorithms by **measuring the characteristics of the noise** signal



PROPOSED PROCEDURE

...



EIGENVALUE SPREAD

- Calculation of eigenvalue spread is used for the proposed system to select appropriate algorithm in eliminating noise from regular/irregular noisy signals.
- The eigenvalue spread is determined from autocorrelation matrix \mathbf{R} ,

$$\mathbf{R} = E[\mathbf{n}(k)\mathbf{n}^H(k)] = \begin{bmatrix} E[|n_0(k)|^2] & E[n_0(k)n_1^*(k)] & \cdots & E[n_0(k)n_M^*(k)] \\ E[n_1(k)n_0^*(k)] & E[|n_1(k)|^2] & \cdots & E[n_1(k)n_M^*(k)] \\ \vdots & \vdots & \ddots & \vdots \\ E[n_M(k)n_0^*(k)] & E[n_M(k)n_1^*(k)] & \cdots & E[|n_M(k)|^2] \end{bmatrix}$$

Here, $\mathbf{n}^H(k)$ is the Hermitian transpose of input $\mathbf{n}(k)$

- The eigenvalues are calculated from the characteristic equation of \mathbf{R}

$$\det(\mathbf{R} - \lambda_j \mathbf{I}) = 0$$

- Here, \mathbf{I} is the identity matrix and the eigenvalues λ_j is given by

$$\lambda_j = \begin{bmatrix} \lambda_1 & & & 0 \\ & \lambda_2 & & \\ & & \ddots & \\ 0 & & & \lambda_M \end{bmatrix}$$

where, $\lambda_1, \lambda_2, \dots, \lambda_M$ are the eigenvalue elements of \mathbf{R}

- The eigenvalue spread of \mathbf{R} is then calculated as

$$s(\mathbf{R}) = \frac{\max(\lambda_j)}{\min(\lambda_j)} \leftarrow \begin{array}{l} \text{maximum eigenvalue of } \mathbf{R} \\ \text{minimum eigenvalue of } \mathbf{R} \end{array}$$

- Using the measurement of $s(\mathbf{R})$, the selection mechanism of appropriate adaptive algorithm is set

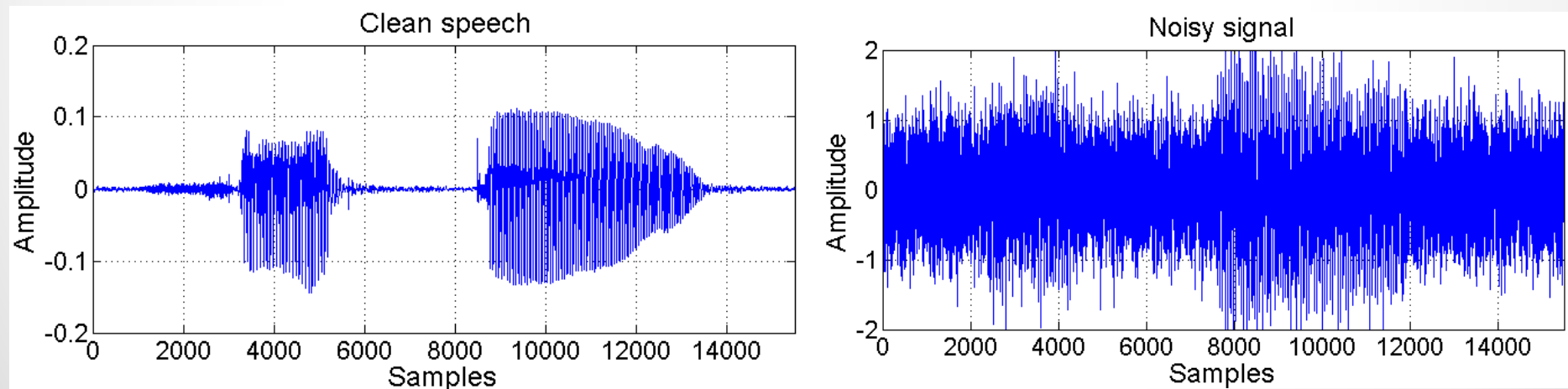
SELECTION MECHANISM

- The adaptive algorithms used are the Least Mean Square (LMS), the Recursive Least Square (RLS) and the Affine Projection (AP) algorithms.
- Algorithm application conditions :
 - LMS – low eigenvalue spread
 - RLS – large eigenvalue spread is very large
 - AP – between 2 conditions above
- The AP would reduce colored noise with low projection order, similar to the Normalized LMS with mild complexity
- SNC selects an adaptive algorithm intelligently based on a flag setting.



SIMULATION PROCEDURE

- Target signal is a Malay utterance “SATU” sampled at 16 kHz
- Noisy speech subjected to several types of noises e.g. white, car, voice babble and pink noise
- The eigenvalue spread of the input noise signals are calculated using 125 data/frame with 60 frames each signal and is repeated to observe the changes in the noise signals



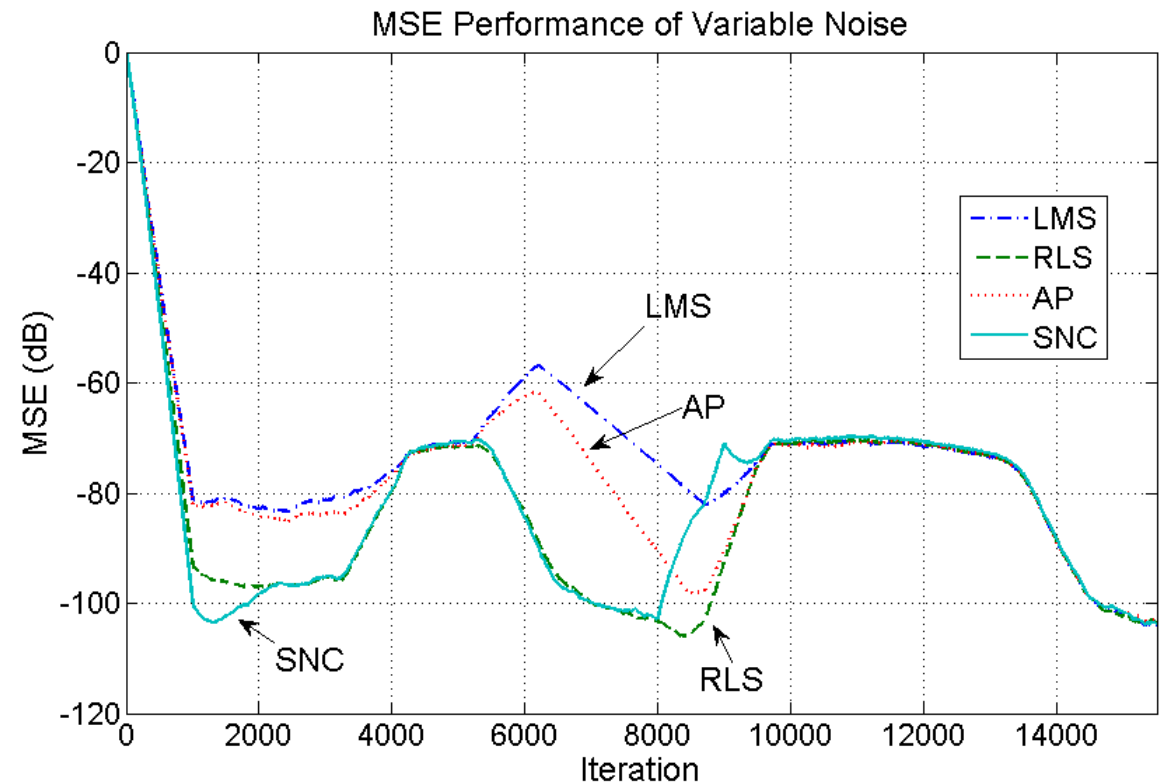
RESULTS & DISCUSSIONS

...



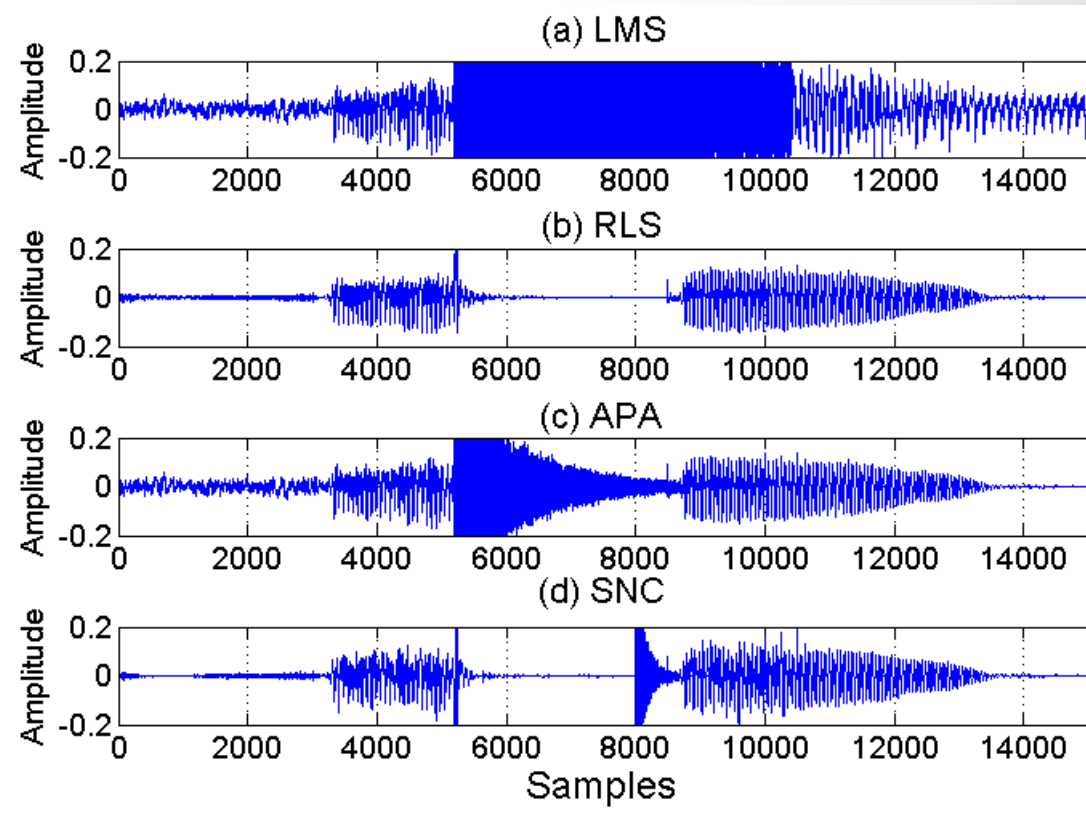
MSE PERFORMANCE

- MSE performance compared to other single algorithms
- At the beginning, the SNC convergence shows **a similar behavior to that of the RLS**
- Then, **converges faster than others** at the middle of the operation



OUTPUT SIGNAL

- The figure shows the processed speech using different algorithms to control the adaptation process
- SNC showed capability of computing different algorithm when the noise properties changed



COMPUTATIONAL COMPLEXITY

- Calculations are made using parameters in the simulations, filter length $N = 32$ and projection order for AP, $M = 4$
- The proposed system has nearly 65% reduction to that of the RLS

Algorithm	Computational Complexity		
	Additions	Multiplication	Calculation
LMS	$2N + 1$	$2N + 1$	65
RLS	$3N^2 + 11N + 9$	$3N^2 + 7N + 9$	3305
AP	$(M^2 + 2M)N + M^3 + M^2 - M$	$(M^2 + 2M)N + M^3 + M^2$	848
SNC	$\frac{3N^2 + 15N + 12}{3}$	$\frac{3N^2 + 11N + 11}{3}$	1145

CONCLUSION

- The paper proposed a novel noise canceller based on measurement of eigenvalue spread
- Capable to remove regular and irregular noise by applying an appropriate algorithm
- The convergence performance of proposed system outperformed that of other algorithms
- Computational complexity is reduced almost 65% of the RLS algorithm
- This study can be extended to include more variants of existing algorithms

