Multi-Channel Active Noise Control System Designs with Fuzzy Logic Stabilized Algorithms

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Abstract. In active noise control filter, IIR filter structure which used for control filter assures the stability property. The stability characteristics of IIR filter structure is mainly determined by pole location of control filter within unit disc, so stable selection of the value of control filter coefficient is very important. In this paper, we proposed novel adaptive stabilized Filtered_U LMS algorithms with IIR filter structure which has better convergence speed and less computational burden than conventional FIR structures, for multi-channel active noise control with vehicle enclosure signal case. For better convergence speed in adaptive algorithms, fuzzy LMS algorithms where convergence coefficient computed by a fuzzy PI type controller was proposed.

Multi-channel Active Noise control systems

Unwanted acoustic noise problems become more and more evident as increased numbers of large industrial equipment such as engines, blowers, fans, transformers, and compressors are in use. Mechanical vibration is another related type of noise that commonly creates problems in all areas of transportation and manufacturing, as well as many household appliances.

The traditional approach to acoustic noise control use passive techniques such as enclosures, barriers, and silencers to attenuate the undesired noise. These passive silencers are valued for their high attenuation over broad frequency range, but they are relatively large, costly, and ineffective at low frequencies. The attenuation of passive silencers is low when the acoustic wavelength is large compared to the silencer dimensions, such as body length or lining thickness.

ANC systems utilizing a microphone and an electronically driven loud speaker to generate a canceling sound was first proposed in a 1936 patent by Lueg[1]. Recently, ANC systems needs highly precise control, temporal stability, and reliability performances. It is therefore desirable for noise canceller to be digital, where signals from electroacoustic or electromechanical transducers are sampled and processed using DSP systems with enough speed and precision to execute sophisticated mathematical functions in real time[2~3].

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In this paper, we proposed novel adaptive stabilized fuzzy Filtered_U LMS algorithms with IIR filter structure which has better convergence speed and less computational burden than conventional FIR structures, for multi-channel active noise control with vehicle enclosure signal case.

Fig 1. Shows that multi-channel ANC systems using J reference sensors, K parallel adaptive filters and corresponding secondary sources, and M error sensors[4]. Fig1. Can be simplified to block diagram shown in fig. 2. The elements of the vector d(n) are the primary noise components of the error sensor outputs. The matrix P(z) represents M x J primary path transfer function, the matrix S(z) represents M x K secondary path transfer functions.
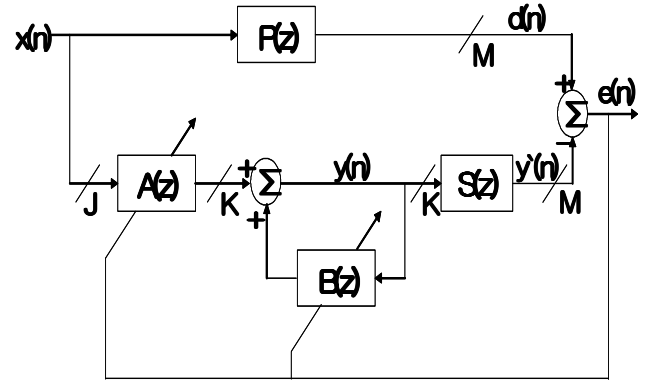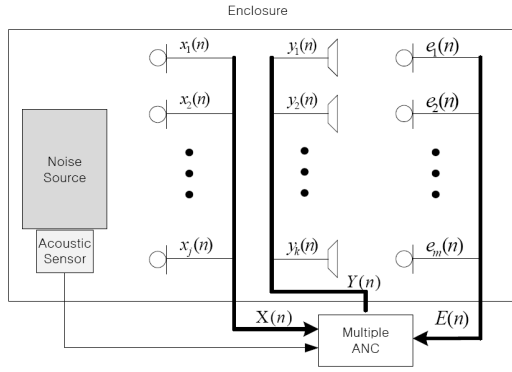
Fig. 1. 3-Dimensional Active Noise Cancelling System   Fig. 2. Block diagram of multi-channel ANC System

The output of k-th secondary transfer function as follows.

$$y_k(n) = \sum_{j=1}^{J} a_{kj}^{T}(n) x_j(n) + \sum_{i=1}^{K} b_{ki}^{T}(n) y_i(n-1) \tag{1}$$

for $k = 1, \quad 2, \quad \cdots \quad, K$, where

$$a_{kj}(z) \equiv a_{kj,0}(n) + a_{kj,1}(n)z^{-1} + a_{kj,2}z^{-2} + \cdots + a_{kj,L-1}(n)z^{L-1} \tag{2}$$

$$b_{ki}(z) = b_{ki,1}(n)z^{-1} + b_{ki,2}(n)z^{-2} + b_{ki,3}(n)z^{-3} + \cdots + b_{ki,I}(n)z^{I} \tag{3}$$

For unknown or time varying equations of adaptive LMS algorithms is as follows.

$$a_{kj}(n+1) = a_{kj}(n) + \mu_a \sum_{m=1}^{M} x'_{jkm}(n) e_m(n) \tag{4}$$

$$b_{ki}(n+1) = b_{ki}(n) + \mu_b \sum_{m=1}^{M} y'_{ikm}(n) e_m(n) \tag{5}$$

where

$$x'_{jkm}(n) \equiv \hat{S}_{mk}(n) * x_j(n) \tag{6}$$

$$y'_{ikm}(n) \equiv \hat{S}_{mk}(n) * y_i(n) \tag{7}$$

are respectively, the filtered reference and output signal vectors.   In order to get optimal control filter coefficients, must be consider the effect of secondary path transfer functions $\hat{S}_{mk}(n)$, and then Fig. 2 modified as Fig. 3(J=1, M=2, K=2).   These methods are called by Filtered_U LMS algorithms.
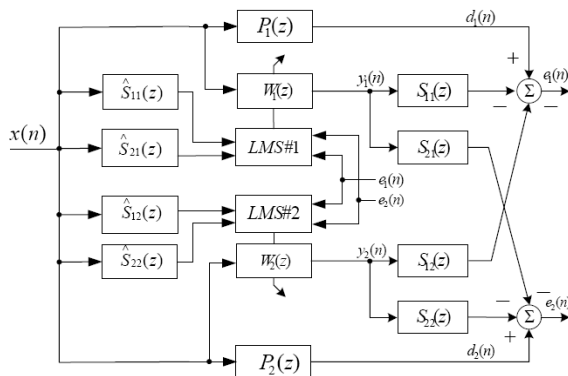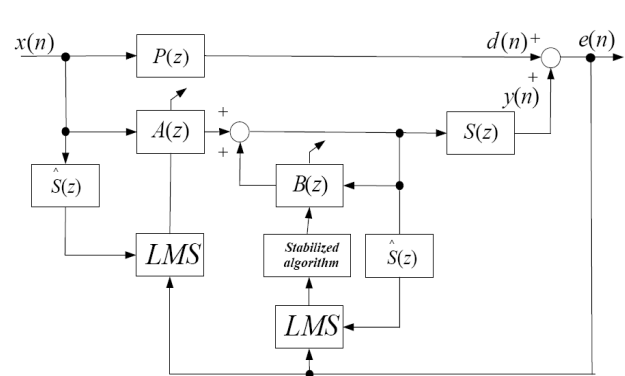

Fig. 3. Multi-channel(1x2x2) ANC System   Fig. 4. Stabilized Filtered_U ANC System

Stabilized IIR filter structure

An adaptive IIR filter structure was considered the feedback as a part of the overall plant, it allows the system to dynamically track changes in the secondary and feedback paths during cancellation operations.   But Eq. 6 implies the adaptive IIR filter would be try to generate poles corresponding to the ideal response , and since the trajectory of the filter poles during convergence is rather unpredictable, one or several poles could be more outside the unit circle, leading to instability.   The problem with unstable filter poles can be solved such as leaky LMS algorithm[5].   However, This methods has slow convergence characteristics of adaptive algorithms.

In this paper, we proposed the stabilized algorithms for adaptive IIR filter structure[6~7] with multi-channel ANC systems which shows in Fig. 4.   Eq. 7 represent proposed algorithms that pre-filter polynomial matrix F(z) which multiply by control filter polynomial matrix B(z).   Pre-filter polynomial f(z) has a role that moving the unstable filter poles in transient state, and then return originally in steady state.

$$\mathbf{F}(z) = \begin{bmatrix} f_{11}(z) & f_{12}(z) & \cdots & f_{1K}(z) \\ f_{21}(z) & \cdots & \cdots & f_{2K}(z) \\ \vdots & \ddots & \ddots & \vdots \\ f_{K1}(z) & \cdots & \cdots & f_{KK}(z) \end{bmatrix} = \mathbf{I}_K - k\mathbf{B}(z) \tag{8}$$

where $\mathbf{I}_k$ is K x K unit matrix, and then pre-filter polynomial is

$$f_{kk}(z) = 1 + k(n)b_{kk,1}(n)z^{-1} + k^2(n)b_{kk,2}(n)z^{-2} + \cdots + k^I(n)b_{kk,I}(n)z^{-I} \tag{9}$$

where $0 \le k \le 1$, calculation of coefficient k is as follows

$$k(n+1) = \lambda k(n) + (1-\lambda) \quad , \quad k(0) = 0, \quad 0 \le \lambda \le 1 \tag{10}$$

Fuzzy logic LMS algorithm

LMS algorithms which were used for control filters, assure the convergence property, and computational burden of these algorithms are proportionate to the number of filter taps.   However, slow convergence speed of the LMS algorithm is a major disadvantage.   The convergence speed of the LMS algorithm is mainly determined by the value of the convergence coefficient, so optimal selection of the value of the convergence coefficient is very important.   Convergence speed of the normalized LMS algorithm is relatively faster than the conventional LMS algorithm[8].

In this paper, a fuzzy LMS algorithm where the convergence coefficient is computed by a fuzzy proportional controller was proposed[9] in Fig. 5.

The estimation errors and mis-adaptation degrees were used for input of the fuzzy logic LMS algorithm in Eq. (11).   Proposed fuzzy logic LMS algorithms is

$$W(n+1) = W(n) + \frac{v(n)}{(L+1)}e(n)x(n) \tag{11}$$

where $v(n)$ is output of fuzzy inference engine, and L is order of control filter.

Calculation procedures of $v(n)$ is as follows.
1) Calculate the estimation error and change of error as follows in Eq. 12~13.

$$ue(n) = \alpha_1 ue(n-1) + (1-\alpha_1)\frac{\| e(n)x(n) \|}{(L+1)} \tag{12}$$

$$uce(n) = \alpha_1 uce(n-1) + (1-\alpha_1)\| ue(n)) - ue(n-1) \| \tag{13}$$

where α_1 is forgetting factor of $ue(n)$, $ue(n)$ is quantitative measurement of misadaptation which obtained by norm of cross correlations of input signal and estimation error signal vector, and $uce(n)$ is result of smoothing filter which norm of change of error signal ue(n).

2) Define the membership function of fuzzy input/output variable $ue(n), uce(n), v(n)$, respectively, as shown Fig. 6.
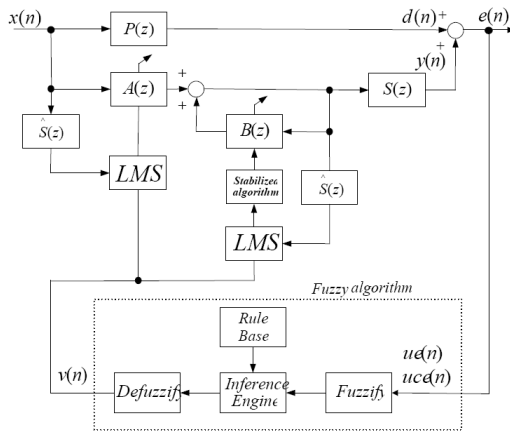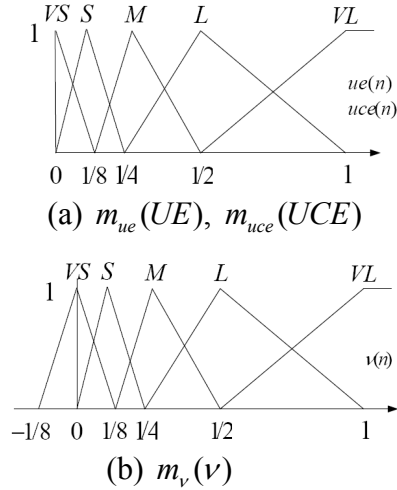
Fig. 5 Stabilized adaptive fuzzy LMS ANC algorithm

Fig. 6 Membership function of fuzzy input/output variables

3) Define the linguistic control rule for decision of convergence coefficient that fast convergence characteristics as Table 1.

Table 1 Linguistic control rules for decision of convergence coefficient

uce＼ue	VS	S	M	L	XL
VS	VL	L	M	S	VS
S	VL	VL	M	M	S
M	VL	VL	L	L	M
L	VL	VL	VL	VL	L
XL	VL	VL	VL	VL	VL

where VS : very small, S : small, M : medium L : large, XL : very large, respectively.

4) Calculate max-min composition operation

$$v^0(n) = \max[\sum_{i=1}^{N} \min[m_{UEi}(ue_i), m_{UCEi}(ue_i), m_{vi}(v_i)]] \tag{14}$$

where $v^0(n)$ is fuzzy output variable.

5) Calculate the crisp output $v(n)$, using discrete center of gravity methods (defuzzify methods)

$$v(n) = \frac{\sum_{i=1}^{N}(v^0(n)m_v(v_i))}{\sum_{i=1}^{N}(m_v(v_i))} \tag{15}$$

Computer simulation

 To show effectiveness of proposed systems, computer simulations was performed by under such condition as Table 2 in given common parameters.   Simulation model is applied by 1x2x2 multi-channel models[4].

Table 2. Common simulation parameters

Item	Data	
Transfer Function	1x2x2 multi model(order : 25)	
Sound speed	340[m/s]	
Sampling frequency	2[kHz]	
Calculation step	10,000step(5 sec.)	
Reference signal	Gasoline Engine muffler	
Estimated secondary transfer function	FIR structure, 192tap	
Control filter tap(conventional)	FIR structure, 192tap	
Control filter tap(proposed)	Numerator : 96	Denumerator : 6
Convergence coefficient	0.000015	

 To reduce the computation burden in fuzzy inference engine that overcome real time implementation problem, we make the look-up table which composed of convergence coefficient that calculation result of fuzzy inference engine.
 Fig. 7 is represented by pole-zero location plotting of transfer function of primary source and secondary sources path(system order : 25).
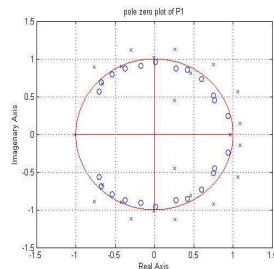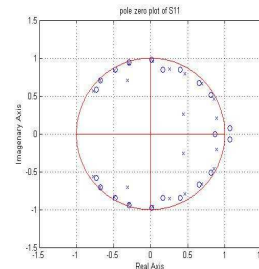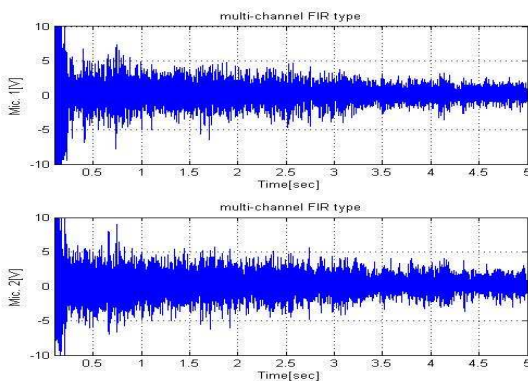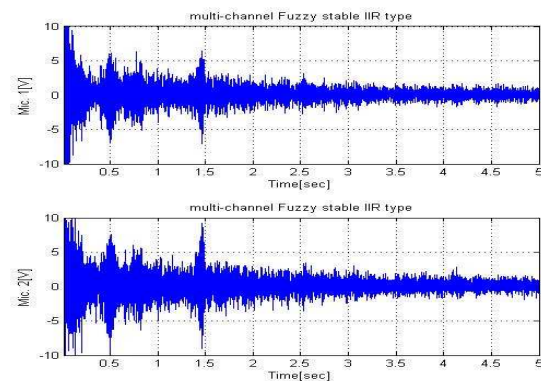

(a) primary source (P_1)   (b) secondary source(S_{11})

Fig. 7 Pole-zero location plotting of transfer function

 Fig. 9 is simulation result of conventional algorithm and proposed algorithms, respectively. Proposed algorithm is shown that better convergence speed and computational burden than conventional algorithm.

(a) conventional algorithms(FIR structure)   (b) proposed algorithms(Fuzzy Stable IIR)

Fig. 9 Comparison of conventional and proposed algorithms

Conclusion

In active noise control filter, IIR filter structure which used for control filter assures the stability property. The stability characteristics of IIR filter structure is mainly determined by pole location of control filter within unit disc, so stable selection of the value of control filter coefficient is very important. In this paper, we proposed novel adaptive stabilized Filtered_U LMS algorithms with IIR filter structure which has better convergence speed and less computational burden than conventional FIR structures, for multi-channel active noise control with vehicle enclosure signal case. For better convergence speed in adaptive algorithms, fuzzy LMS algorithms was proposed where convergence coefficient computed by a fuzzy PI type controller was proposed. Computer simulations were performed to show the effectiveness of proposed algorithms with 3-dimensional model problem.

Acknowledgement

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