ACTIVE NOISE CONTROL
: ITS IMPLEMENTATION & PERFORMANCE*

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ABSTRACT

Active noise control systems based on feedforward approach using filtered-x LMS algorithm for single channel (in a duct) and multi-channel (in an enclosure), are considered. The systems are implemented on a *TMS320C30 processor based d_space DSP board* and their performances are evaluated. In the case of duct, for the tonal and white noise, a noise reduction of about 32dB and 12dB, respectively are achieved. For a duct with an axial flow fan, using anti-turbulence probes, 7.66dB noise reduction is obtained. For the multi-channel case, a 2 channel system for a typical aircraft cockpit mockup model is considered and a reduction of about 20dB is obtained for a tonal noise.

1. INTRODUCTION

The noise cancellation with passive absorbers are well suited for the high frequency range of the acoustic spectrum, however their size, weight and the cost increase for the low frequency range (50-500Hz) making them not suitable for mobile vehicles like aircrafts, helicopters, cars and also industrial applications like exhaust and ventilator ducts. This is due to the fact that the passive absorbers reduce the noise by obstructing their propagation and this requires the size of the passive absorbers to be of the order of the wavelength of the noise frequency. At 2000Hz, though the wavelength of sound is 0.18m, at 100Hz, it is 3.4m. Further, a high surface density material passive absorber like lead, is required to achieve higher attenuation. In view of this, the size, weight and hence the cost of the passive noise absorber, is not a viable solution for the mentioned applications.

A propeller engine driven aircraft is 25% more fuel efficient than a jet aircraft, but it is more noisier. The very aircraft structure itself is a source of noise as it has many discontinuities like tail-plane, tailing edge flaps, main wings, leading edge stats, landing gear and engine nacelle spillage and these create turbulence and this gets accumulated as air flows from front to back. The release of landing gear itself increases the noise level by 10dB. The lightweight and 4 stroke automobiles (compared to 2stroke) though fuel efficient are noisier. In many industrial applications the use of passive absorbers generate backpressure and hence calls for the use of increased power. For proper mobile communication, reduction of noise is essential. These necessitate the use of Active Noise Control (ANC) technology as an alternative to passive absorbers.

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The basic principle of ANC is to superimpose an antinoise field of the same magnitude and opposite phase, on the existing/primary noise field. The two noise fields result in a residual noise field, which is supposed to be relatively of small magnitude, compared to the primary noise field. The residual noise is used in the generation of the antinoise. In ANC, unlike in electrical noise cancellation, the cancellation takes place in acoustic domain.

In this paper, the single channel ANC for ducts (with and without flow) and multi-channel ANC for enclosures like aircraft cockpit, implemented on a DSP processor are considered.

2. SINGLE CHANNEL ANC [1,2]

A typical ANC in a duct environment is shown in Fig. 1. The reference noise is picked up from the input microphone. The antinoise is generated and is given out to the acoustic domain through a loud speaker (secondary source). The residual noise is picked up from the error microphone and is used in the generation of the antinoise.

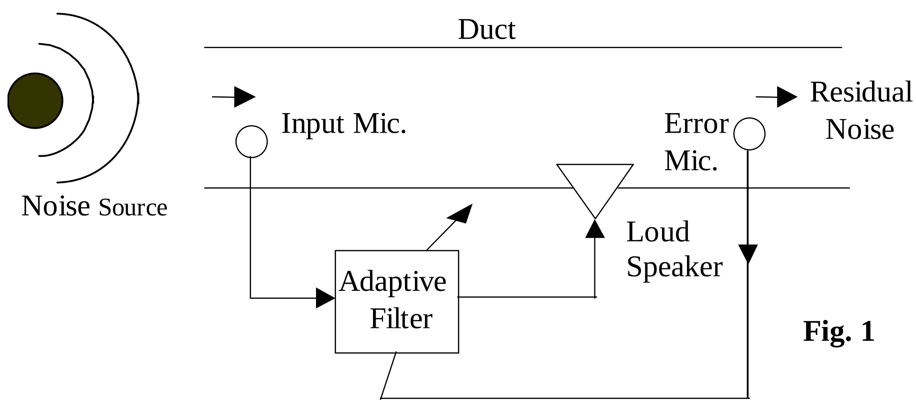

Fig. 1

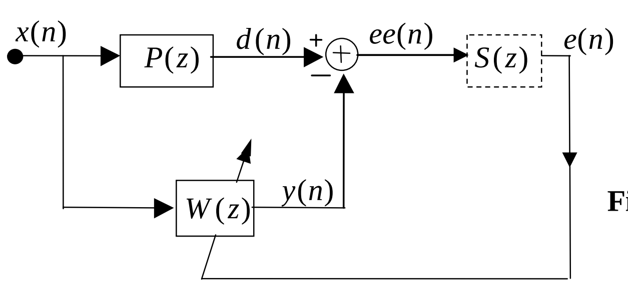

Fig .2

Fig. 1 Typical arrangement for adaptive ANC in a duct.
Fig. 2 ANC block schematic

In the duct, sound wave propagates from the noise source end to the termination of the duct, where the noise is to be attenuated. As the noise travels from source end, it undergoes both magnitude and phase changes depending upon the acoustic path before it reaches the other end. The adaptive filter builds this magnitude and phase changes for the reference signal picked up by the input microphone and generates the antinoise, which is superimposed by the loud speaker. The residual noise forms the error signal for the adaptive filter and adjusts its coefficients so that the error energy is minimized. The superposition of the antinoise and the primary noise fields is over a physical space and not at one point. In view of this, the error signal can be considered to be effectively available only after it passes through a transfer function corresponding to the acoustic path between the secondary source and the error microphone which is generally referred to as secondary path/ error path (Fig. 2).

In the ANC schematic (Fig. 2), $P(z)$ is the main path, $S(z)$ is the secondary path, $W(z)$ is the adaptive filter to model $P(z)$. $x(n)$ is the input noise, $y(n)$ is the antinoise, $ee(n)$ is the error signal prior to secondary path and $e(n)$ is the error microphone pickup.

In the absence of $S(z)$, the secondary path, the duct ANC reduces to a system identification problem and is solved by the famous LMS algorithm which adapts the weight vector $W(z)$ by

$$w_i(n+1) = w_i(n) + \mu(n)\ \underline{ee(n)\ x(n-i)} \qquad 0 < i < L-1 \qquad (1)$$

where $$\mu\ (n) = \frac{\mu}{L * P(n)}$$

$$P(n) = \beta\ P(n-1) + (1-\beta)\, x^2(n), \qquad 0 < \beta < 1$$

where L is the length of the adaptive filter and $P(n)$ is the power of the input signal.

This requires a *stringent time alignment* between the input $x(n)$ and the error $ee(n)$. Therefore, the presence of $S(z)$ affects this time alignment and requires modification of LMS algorithm for applying to ANC system.

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2.1 Filtered – X- LMS algorithm

The transfer function $S(z)$ can be shifted prior to the summing junction (Fig. 3b). Further $S(z)$ can be combined with the plant $P(z)$ to form a new plant $P'(z)$ and also, $S(z)$ can be shifted prior to *W(z)* (Fig.3c) [3]. For the part of the Fig. 3c shown inside the dotted box, the error signal *e(n)* is *directly* available and hence the LMS algorithm can be applied. The adaptation rule is given by

$$w_i(n+1) = w_i(n) + \mu\ (n)\ \underline{e(n)\ x'(n-i)} \qquad 0 < i < L-1 \qquad (2)$$

where $$\mu(n) = \frac{\mu}{(L+M)\ P'(n)}$$

where M is the secondary path length and

$$P(n) = \beta\ P(n-1) + (1-\beta)\, x'^2(n)$$

Here $x'(n) = s(n) * x(n)$ which is the filtered version of the input $x(n)$, the filter being $S(z)$. This means, to adapt $W(z)$ with the error *e(n)* which is available only after it propagates through the secondary path *S(z)*, the input signal to be used is a filtered version *x'(n)* of the actual input *x(n)*. This requires *S(z)* to be shifted prior to the summing point to form a new plant *P'(z)*. However, this is practically impossible. But the effect of *S(z)* can still be overcome, by using the filtered input $x'(n)$ only for the adaptation of $W(z)$ and the input for it is retained as $x(n)$ itself, which does not call for any modification in the practical setup. Since the filtered input is used for the adaptation, this algorithm is known as the *filtered-X LMS (FXLMS)* algorithm.

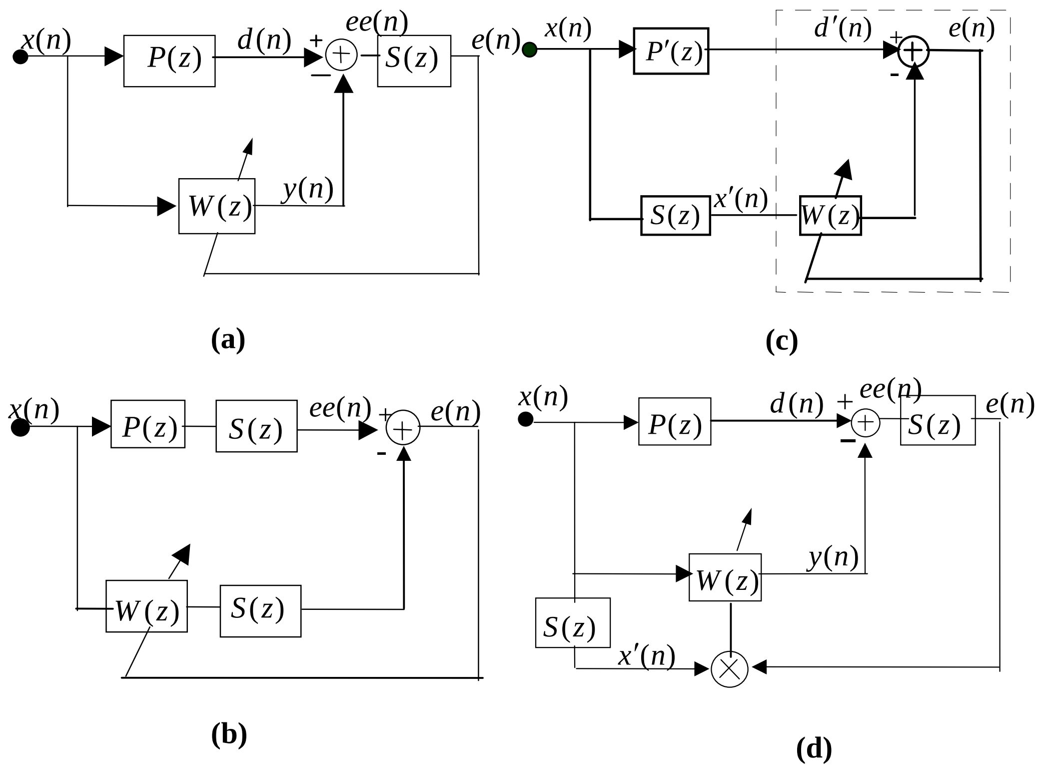

Fig.3 Development of FXLMS algorithm

Even in Eqn. 2, it very important to maintain the time alignment between $e(n)$ and $x'(n)$ to ensure proper convergence. If $S(z)$ is known, the time alignment is not a problem. However it has to be estimated and its accuracy decides the performance of the ANC system. For convergence, the time alignment has been quantified and it has been found that phase error between $S(z)$ and its estimate $\hat{S}(z)$ should be between $\pm 90^0$ and as it reaches the limits the ANC performance deteriorates [4]. The effect of the delay introduced by $S(z)$ also limits the step-size value [5]. To take into the account of changes in the secondary path due to changes in temperature, humidity, flow conditions and aging of the loudspeaker; secondary path is identified on-line basis (in the presence of primary noise) rather on offline (in the absence of primary noise) with white noise input, $v(n)$ (Fig. 4) [6].

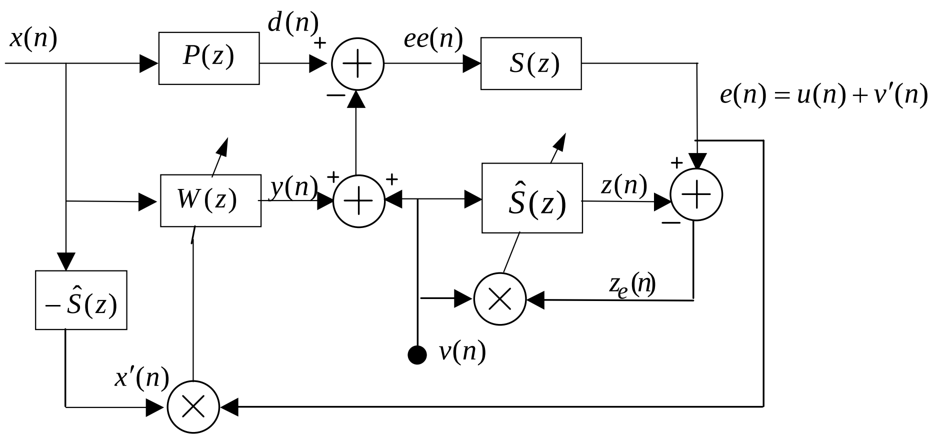

Fig. 4 FXLMS ANC with on-line secondary path modeling

The online identification has to be done in the presence of a strong observation noise $u(n)$, as its major component is due to the primary noise.

$$e(n) = [\{d(n) - y(n)\} - v(n)] * s(n)$$
$$= u(n) + v'(n),$$

Where $u(n) = \{d(n) - y(n)\} * s(n)$, $v'(n) = -v(n) * s(n)$

Further, the input excitation $v(n)$ level has to be relatively quite small compared to the primary noise field, as it decides the residual noise floor level. Any disturbance picked up by the error microphone also adds to the observation noise.

$$\hat{S}(n+1) = \hat{S}(n) + \mu_s(n)V(n)z_e(n)$$
$$= \hat{S}(n) + \mu_s(n)\, V(n)[v'(n) - z(n)] + \mu_e\, \underline{u(n)v(n)}$$

Here, only *the mean value* of $\hat{S}(n)$ converges to the optimum solution and hence its instantaneous value will be varying due to the term $\mu_s(n)u(n)v(n)$ and the gradient step-size and the observation noise level decide its variance.

Also, a large step size is required as the error path information is required in advance for computing filtered input, its adaptation should be faster and in a fixed-point processor the use of a very small step-size leads to adaptation stalling.

In view of these, an secondary path estimate *as exact as possible*, is very much essential and in an ANC system, the secondary path identification should: have a large step-size, tolerate large observation noise including interference; and at the same time the errors introduced by these inevitabilities, have to be reduced. To overcome the observation noise problem, a noise canceller [7] or a predictor [8] for the secondary path identification has been used.

The performance of the ANC system is decided by coherence between $x(n)$ and $e(n)$, $C_{ex}(\omega)$ [1,2] which is a measure of linearity between them is given by,

$$S_{ee}(\omega) = \left[1 - \left|C_{ex}(\omega)\right|\right] S_{xx}(\omega)$$

Where ω is the frequency of interest, $S_{xx}(\omega)$ and $S_{ee}(\omega)$ are the auto spectral densities of $x(n)$ and $e(n)$, respectively.

It is necessary to have a high coherence $\left[C_{ex}(\omega) \approx 1\right]$ to get a better noise reduction, given by

$$-10\log_{10}[1 - \left|C_{ex}(\omega)\right|]$$

If $C_{ex}(\omega) = 0.99$, the maximum attenuation that can be achieved is 20dB. Any noise added at the error microphone that is not correlated to input $x(n)$ will bring down the coherence and hence the attenuation of ANC. In the case of exhaust and ventilator ducts, there will be turbulence due to airflow. These local turbulence effects will be picked up by the microphones act as noise for the sound signal and reduces the coherence between input and error microphone signals. This effect can be reduced by using anti-turbulence probe [9], which is nothing but a microphone housed at one end of a perforated tube and the other end has a conical shaped smooth termination. Further, the perforated tube is wrapped with a cloth to attenuate the turbulence and to allow only the sound signal to be picked up by the microphone. The smooth conical termination avoids building up of additional turbulence due to discontinuities.

3. MULTICHANNEL ACTIVE NOISE CONTROL [1,2]

The acoustic noise field is characterized by normal modes and to cancel the noise, for each mode one channel is required. The number of modes N is given by

$$\boldsymbol{N} = \frac{\boldsymbol{4}\pi}{3}\,\frac{\boldsymbol{f}^{\boldsymbol{3}}\,V}{\boldsymbol{c}^{\boldsymbol{3}}}$$

where V *– Volume,* c *- velocity of sound and* f *- the frequency.*

The increase in the volume calls for more number of channels. Hence, the single channel FXLMS algorithm can be extended further to develop a multi-channel algorithm. Here, the cross couplings (between microphones and loud speakers) have to be taken into consideration.

In a $I \times J \times K$ ANC system, the weight vector $\boldsymbol{W}(z)$ is adapted as

$$\boldsymbol{W}(n+1) = \boldsymbol{W}(n) + \mu\ \boldsymbol{X}'(n)\ \boldsymbol{e}(n)$$

$$\boldsymbol{w}_{ij}(n+1) = \boldsymbol{w}_{ij}(n) + \mu \sum_{k=1}^{K} X'_{ijk}(n) e_k(n)$$

where $i = 1,2,....,\boldsymbol{I}$, no. of input microphones
$j = 1,2,....,\boldsymbol{J}$, no. of secondary sources (loudspeakers)
$k = 1,2,....,\boldsymbol{K}$, no. of error microphones
$X'_{ijk}(n)$ - Input $X_i(n)$, filtered by the secondary path $S_{kj}(z)$
i.e, j^{th} loudspeaker to k^{th} microphone.

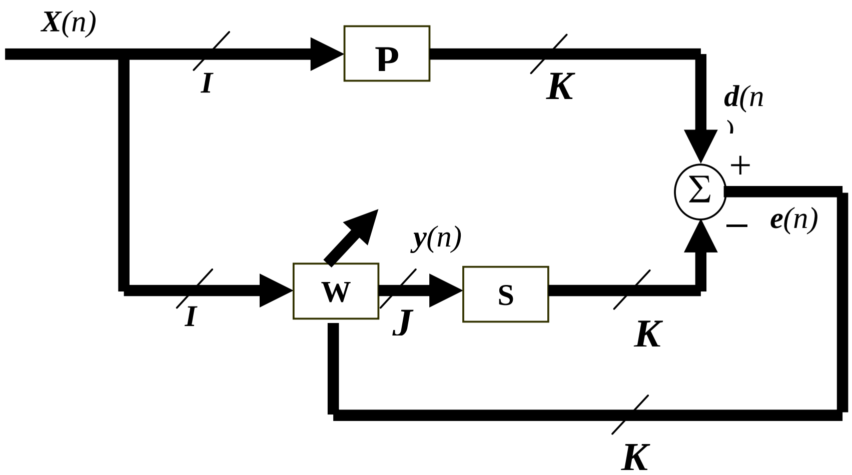

Fig. 5 Block schematic of multi-channel ANC system

4. ALGORITHM COMPLEXITY AND REAL TIME IMPLEMENTATION

The required sampling rate is 1KHz/ sampling time is 1ms, since the operating range of ANC is 50-500Hz. FXLMS algorithm involves multiply and accumulate operations, it is well suited for the implementation on the DSP processor. But the computations required increase with increase in the filter length. This calls for the highly optimized assembly code to satisfy the real time constraint of 1ms computation time.

Further in the multi-channel ANC, with the increase in the no. of channels, the secondary paths increase, due to the existence of the cross paths. The complexity comparison is given in the table. For L = 128 and M = 64, the computational complexity of the multi-channel ANC systems are computed in-terms of multiply and accumulate (MAC) operations.

Computational complexity of multi-channel FXLMS algorithm

$I \times J \times K$	$1\times 2 \times 2$	$3 \times 3 \times 3$	$8\times 8 \times 8$
No. of main path adaptive filters, $I \times J$	2	9	64
No. of error paths, $J \times K$	4	9	64
No. of filtered inputs $I \times J \times K$	4	27	512
$MAC_{FXLMS} = I\,J\,K(L+M) + I\,J\,L + K$	1026	6339	106504

Table indicates that, computations and memory required for the filtered inputs is very large compared to the coefficient adaptation itself. This filtered input increases the computational burden and the memory requirement. This necessitates the DSP code to be highly efficient both in terms of computations and memory consumption. To reduce the computations a re-formulation of FXLMS algorithm which avoids direct computation of filtered inputs has been proposed [10].

The algorithms are implemented on a TMS320C31 (33.3 MFLOPS, 40MHz) floating point DSP processor present on a d_space DS1102 board. This has 12-bit Analog to Digital Converters (4 nos.) and 16-bit Digital to Analog Converters (4 nos.) [11]. To make efficient use of features like circular buffering and hardware branching, the software is implemented in assembly language [12].

5. EXPERIMENTS AND RESULTS

5.1 Single channel ANC

The single channel ANC algorithm is implemented for a duct of dimensions $1' \times 1' \times 12'$ as shown in Fig. 5. The normal audio microphones and loud speakers do not satisfy the frequency requirements of ANC viz., flat frequency response over 50-500 Hz. Hence special low frequency loudspeakers and low frequency unidirectional

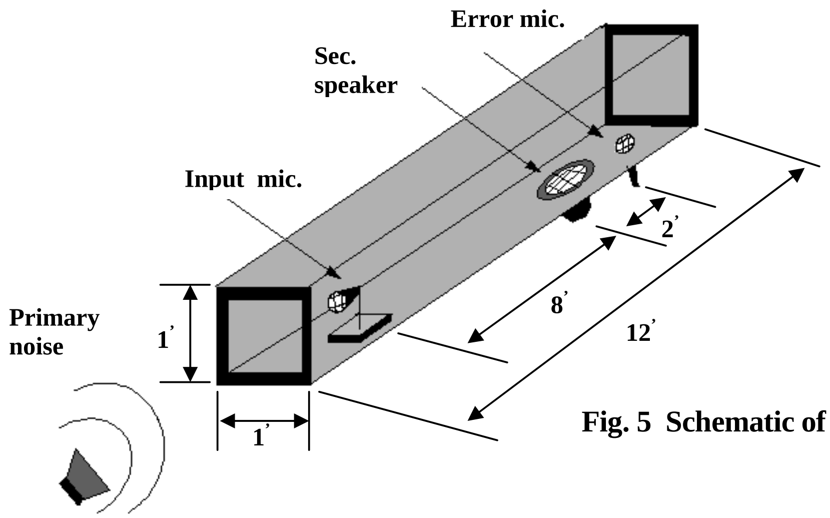

Fig. 5 Schematic of a duct

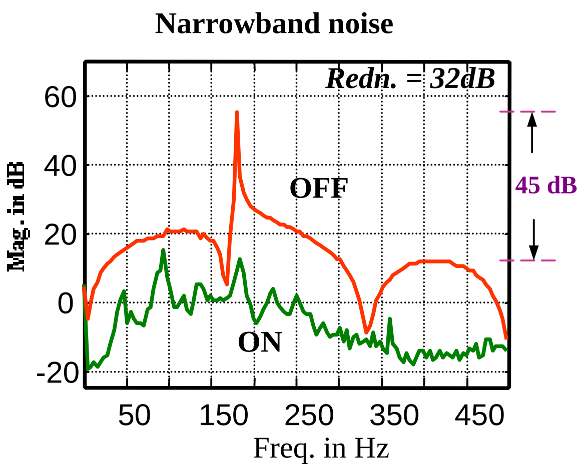

Fig. 6a Power spectral densities of signals with ANC OFF and ON

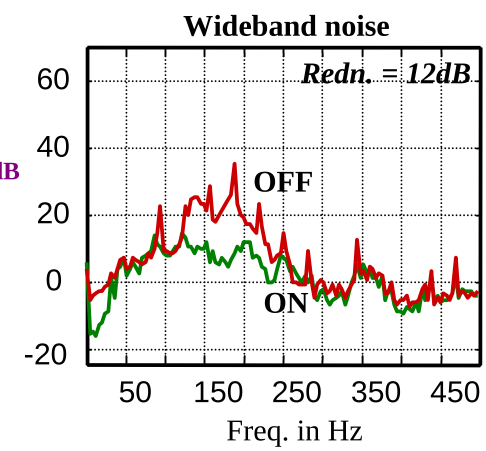

Fig. 6b Power Spectral densities of signals with ANC OFF and ON

microphones (to avoid feedback from secondary source to input microphone) are used for the in this study. The microphone is positioned sufficiently away $2'$ (4 times the diameter of the loudspeaker (5″) from the loudspeaker so that the evanescent modes are allowed to die out and it will pickup only the study sound field.

The secondary path is identification is done on on-line using uniform white noise as input, which is about 35dB below the primary noise level. A narrow band noise characterized by a single tone at 180Hz at 93dB was generated by an electronic oscillator, power amplifier and a loudspeaker (50Watt). The parameters chosen are : adaptive filter lengths for mainpath = 128, secondary path = 64 and for the noise canceller for the identification of the secondary path (NC)= 128; the step-size for the mainpath = 0.003, errorpath = 0.5 and for NC=0.7. Forgetting factors of 0.999,0.99 and 0.99 were used for power updation of input for mainpath, errorpath and for NC, respectively. Fig. 6a shows the power spectral densities of the error with ANC ON and OFF and a noise power reduction of 32dB was achieved.

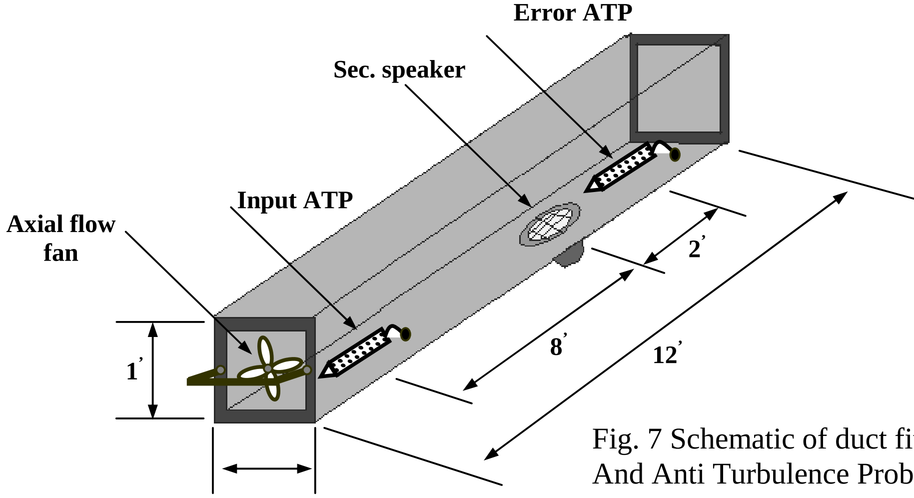

Fig. 7 Schematic of duct fitted with fan And Anti Turbulence Probe (ATP)

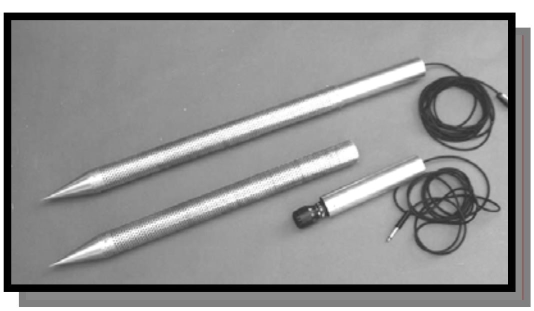

Fig. 7b Anti turbulence tube (ATP)

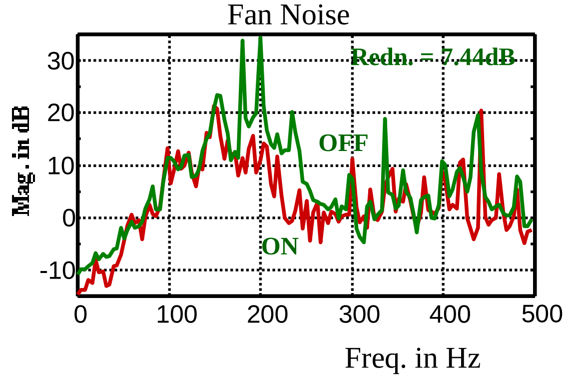

Fig. 7c Power Spectral density for noise at 80dB

For a white noise at 78dB, the parameters selected are: Mainpath length=128, Errorpath length = 64 and for NC= 128; Step-sizes for mainpath =0.01 errorpath = 0.1 and for NC=0.5; forgetting factors for both mainpath errorpath were chosen to be 0.999 and for NC=0.99. This gives a noise attenuation of 12dB (Fig. 6b).

For an axial flow fan fitted at one end of the duct to simulate an exhaust pipe (Fig. 7a) to improve the coherence, anti-turbulence are used (Fig. 7b). In this case, at 80dB noise level, a reduction of 7.66dB is achieved. (Fig. 7c).

5.2 Multi-channel ANC

The cockpit mock-up model(Fig. 8) of volume of about $3m^3$ is used for the feed forward multi-channel ANC system. A $1\times2\times2$ (one reference mic., two secondary speakers and two error mics.) ANC system was used for a narrowband noise field (180Hz tone) at 80dB.

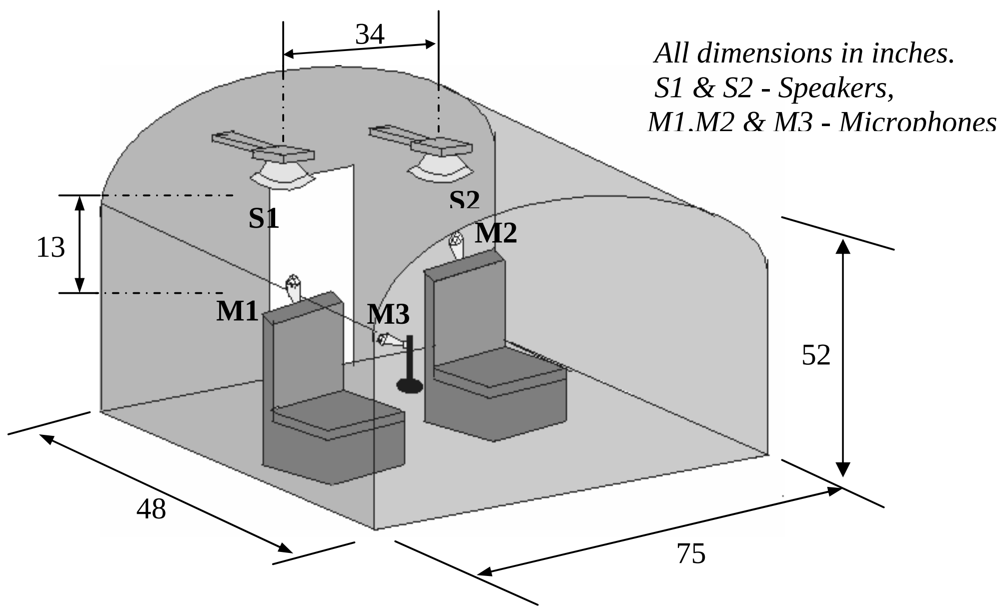

Fig. 8 Schematic of Multi-channel ANC in a cockpit mockup model

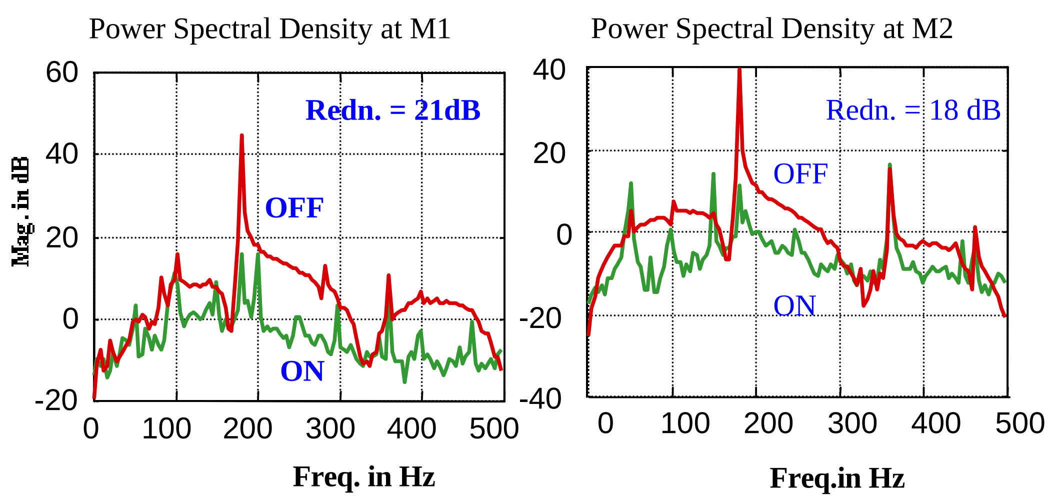

Fig. 9 Multichannel ANC for narrowband noise at 80dB

The secondary sources S1 and S2 are located just above the two seats and the error microphones M1 and M2 are located just behind the head position. The input noise, which is generated outside, is picked up by an input microphone M3 placed at the entrance of the cockpit. The parameters chosen were : a length of 128 is chosen for the main paths (M3 to S1 and M3 to S2), secondary paths(M1 to S1, M1 to S2, M2 to S1 and M2 to S2) and NC (from M3 to M1 and M3 to M2). The secondary paths are identified with two uncorrelated uniform white noises, which are about 25dB below the primary noise level. Step-sizes for the main paths, secondary paths and for NCs are 0.002, 0.02 and 0.05, respectively. Forgetting factors for main paths, secondary paths and for NCs are chosen to be 0.99. A reduction of 21dB and 18dB are observed at M1 and M2, respectively (Fig 9). In addition to the input spectral peak, there are other spectral peaks due to different modes resulting from the reverberation of the enclosure and are dependent on the enclosure geometry and size

6. CONCLUSION

Single and multi-channel ANC systems based on feed forward approach using FXLMS algorithm for a duct (with and without airflow) and for a aircraft cockpit model were considered. The systems were implemented on a TMS320C30 processor based d_space board. Significant noise reduction was achieved : 32dB, 12dB and 7.66dB for a duct with tonal noise, white noise and airflow, respectively. For the cockpit model, for a tonal noise, a reduction of about 20dB was achieved at the two seats.

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