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Hardware Realization of Adaptive Processing for Noise Elimination using TMS320C6713 DSP

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***Abstract*— Every day, we knowledge the property of acoustic noise whereas communicating on portable phones be it a moving vehicles. In reality, it is there even having conversation on a noisy telephone control. This noise pollutes the unique information carrying signal with noise starting its nearby atmosphere. Allowing for, a classification has been developed to discriminate a original signal from other noise sources or intrusive signals in an audio noisy surroundings. The hardware design to implement such a system is a tough task in real-time realistic noise deletion applications. Thus, this Article describe the real-time Architecture design to implement Adaptive Noise deletion method which considers the Least Mean Square (LMS) algorithm as a bench mark on TMS320C6713 DSP, to eliminate unwanted noise from the receiver for different audio associated applications.**

For the research affirmation, we used Texas Instruments' integrated development environment (IDE), and Code Composer Studio (CCS) for TMS320C6713 Digital Signal Processing Starter Kit (DSK) as the objective board. Three different cases are conceded by considering different audio inputs to test the effectiveness of the considered ANC arrangement. The Code Composer Studio is used to implement Least Means Square Algorithm , and realize with the DSP C6713. A 300, 500, 800Hz,1, 3KHz sound signals and male- vocalization signals are taken as the bench mark inputs, to map out the noise of the signal till it is disappeared. Hence, the preferred signal can be prevail. outcome of this research specify that the adaptive noise eliminator can eliminate the noise from signal expediently and in point of fact. The concert of ANC system is calculated in terms of different parameters like 1) convergence speed, and the2) order of the filter and 3) S/N ratio. The intended system expresses a note worthy level of enhancement in S/N Ratio (SNR). Hence, the S/N ratio of the processed signal using LMS adaptive filter progressed by 3 to 9 dB. The results are the visual proof of the Architecture and LMS algorithm performance under different cases.

***Index Terms*— Adaptive Noise Cancellation (ANC), Digital Signal Processor (DSP), Mean Squared Error (MSE), Least mean square algorithm (LMS), TMS320C6713 DSK, Code Composer Studio (CCS), Signal to noise ratio (SNR).**

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I. INTRODUCTION

Adaptive processing of signals are Mostly preferred in conditions where signal & system parameters change steadily, and the processing mechanism has to be familiar to compensate in favor of change. A uncomplicated, variable but controlling FIR filter is the linear adaptive combiner. The LMS algorithms has been applied to present the tactic for modifying the filter coefficients (Haykin, 2002).
The LMS algorithm (Haykin,2002) is certainly the familiar algorithm for adapting the impulse response $W = [W^0 \ldots\ldots W^{N-1}]$ of an Finite impulse response filter so as to reduce the mean-square error (MSE) between its productive signal $y(k)$ and a preferred signal d(k) response .The filter coefficients are updated as

$$e(k) = d(k) - W_{k-1}\, u(k) \quad \ldots.. (1)$$

$$W_k = W_{k-1} + 2*\mu*e(k)*u(k) \quad \ldots..(2)$$

u(k)represent is the input signal and *μ represent* the variation gain . The LMS algorithm is a non-disappearing step size category of a stochastic gradient algorithm (SGA). It is accepted to a maximum scope owed to its computational correspondence. Also, its implementation is moderately straightforward to recognize and the algorithm shows to be moderately similar in temperament beside execution errors. This article principally visional on an imperative features of an adaptive processing i.e. S/N ratio in unwanted sound annulment. Audio signal filtering implementation tolerates due to the occurrence of a range of different types of noises next to the unique signal. The work is meant to execute adaptive Noise Cancellation System by means of hardware i.e. DSP Processor, for elimination of unwanted sound which is expected signal in different audio associated applications. LMS algorithm is implemented to eliminate the adaptive noise for its good quality tracking abilities and in addition, as it does not need mathematical operations like that of finding square ,mean ,differentiation. It gives an substitute technique for shaping the best possible filter coefficients without clearly calculating the matrix inversion (Haykin, 2002). And also, the LMS adaptive algorithm is easy in accomplishment giving extremely proficient, attractive and precise outcome (Widrow, 1975). It has a range of real-time applications such as conversation systems, AC ducts, plains , robots, cars, machine tool developed and automations (Rosado-Muñoz, 2011).

A. Adaptive Noise Elimination

It is experimental that in elevated noise surroundings, powerful background noise frequently alter speech and reduces the quality of communication systems. Even though many single channel noise elimination methods like that of Wiener filtering, Kalman filtering available in standard literature but dual channel which use adaptive filter and implement LMS algorithm for noise cancelation signal chosen up by the principal sensor (Widrow,1975), is broadly used. The associated research, (Hong-Yuan, 2011), alters an execution of noise cancellation by means of adaptive Wiener filter which is based on FPGA design and has shown the efficiency on noise deletion by means of adaptive Wiener filter. Figure 1 Describes the Algorithmic path of the adaptive noise cancellation system.
The fundamental notion of adaptive noise deletion is to filter signal from two sensors and to decrease the intensity of unwanted noise by means of adaptive filtering methods. The signal is rooted over a channel to a sensor which collects the signal and the unwanted noise, *n*. The mutual signal with noise, *d+n*, shape the 'principal input' to the system. A subsequent sensor receives a noise *n*' which is no way concern with the signal but concern in some unidentified way with the noise signal *n*. This sensor supply the 'standard input' to the eliminator. The noise *n* ' can approach from the identical spring as *n* but can be customized by the surroundings. The noise signal n' is processed to create an output *y* that is a close copy of *n* i.e. the adaptive filter's outcome *y* is modified to the noise signal *n*. next, the processed output is subtracted from the primary input, hence giving rise to the output which is free from the signal. Now, the obtained output *y* is subtracted from the major input to create the classification amount produced expected to be free from noise.
The consequential signal is said to be an error signal *e*, and that comes out of the design. The consequential error signal would be the preferred segment of the main signal . This error signal is fed reverse to the adaptive filter and it is recursive to revise the adaptive filter's coefficients till the error signal is equal to original signal e = *d*. At this stage the adaptation process comes to end and the error signal e is precisely equal to the desired Audio signal d Therefore, on the whole outcome is the error signal e and not the adaptive filter's output *y*. If *d* is unconcerned with *n*, then the approach is to reduce $E(e^2)$. therefore, in the classification, the allusion effort is filtered by an adaptive filter that mechanically models its own desired IR by means of(LSA) algorithm like that of LMS that responds to an Audio error signal which dependent amongst additional belongings, on the processed outcome. Consequently, by means of effective algorithm,

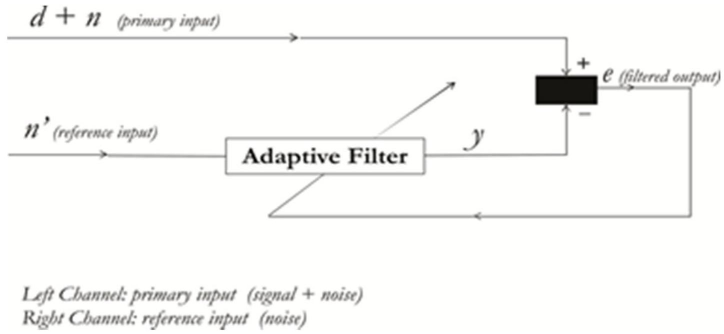


Figure 1 fundamental concept of adaptive noise removing

the processing mechanism can function beneath different circumstances and can remodel itself constantly to reduce the Audio error of the signal. imagine that d is no way concern with n', and let that n is connected with n'. then the output will be $e = d + n' - y$ ……… *(3)* Squaring on two sides of equ (3) we get $e^2 = d^2 + (n' - y)^2 + 2d(n' - y)$ ….. *(4)* expected values eq. 4. $E[e^2] = E[d^2] + E[(n' - y)^2] + 2E[d(n' - y)]$ and realizing so as to d is uncorrelated with n' and y, we obtain, $= E[d^2] + E[(n' - y)^2]$ ……………… *(5)* The signal power $E[d^2]$ will be unchanged as the mechanism is attuned to reduce $E[e^2]$, thus the least amount produced power is given by $E_{min}[e^2] = E[d^2] + E_{min}[(n' - y)^2]$ ……… (6). When the mechanism is attuned so that $E[e^2]$ is least produced, $E[(n' - y)^2]$ is, therefore, also reduced. which gives the LSE of the primary noise n'. In totaling to the given , while $E[(n' - y)^2]$ is reduced , $E[(e - d)^2]$ is also reduced, because from eq. 2, $(e - d) = (n' - y)$ ………………*(7)*

The output e will usually contain original Audio signal and Audio noise signal. From eq. 1, the outcome Audio noise is specified by $(n' - y)$. As reducing $E[e^2]$ reduces $E[(n' - y)^2]$, reduces the total outcome power reduces the outcome audio noise power and, As the Audio signal in the outcome is stationary, the sum of audio outcome power would cause in enhancing the output Audio S/N ratio. From eq. 3, the least probable outcome Audio power is as $E_{min}[e^2] = E[d^2]$. When this is attainable, $E[(n' - y)^2] = 0$. Consequently, $y = n'$ and $e = d$.

In this scenario reducing the output power forces the outcome signal to be free from audio noise signal (Haykin, 2002), (Widrow, 1975).

II. Realization Of Lms Adaptive Noise Eliminator On The Dsk Tms320c6713

The execution of Real-time digital signal filtering by means of a DSP processor is a tough job. Its effective realization be capable of lead to effectual design of a range of signal filtering algorithms for actual world applications [5]. The comprehensive formation of hardware execution of the adaptive Audio noise eliminator is shown in figure 2. For successful implementation of unsighted processing, the adaptive processing needs the parameter values of standard Audio noise and the principal input Audio signal. The Audio noise can be successfully utilize as a standard just if it is calculated in a field where the supply is scrawny. consequently, the microphone that account the standard Audio noise and the one that account the principal input be supposed to be reserved at a definite least amount of distance. The two speakers, who are participating in the Audio noise, have to be in the similar locate so that, to bring into the row the sound, all over the place in the surroundings set-up [6].

The over view of hardware execution of ANC eliminator is shown in figure 3.The 6713 DSK hardware is a less-cost enlargement display place that supports assessment as well as progress of applications for the TI C6713 DSP unit. The stereo line input accepts two diverse signals as its input - the affected principal signal in the left sided channel, and the standard noise signal in the right sided channel. The standard noise, collectively with the predictable processor output and the step size *mu*, estimates how rapid weights are rationalized. Analogous work is consider on noise decline (Martínez, 2011) by performing adaptive processing on Field Programmable Gate Arrays (FPGA) chip and verified it on the hardware. In (Farzad Nekouei, 2012), projected different ways i.e. hardware execution of adaptive least mean square (LMS) filter with respect to unchanging step size and self correcting adaptive filter (SCAF) architecture on spartan3 XC3S400 Field Programmable Gate Arrays (FPGA) chip and analysed the effectiveness of both ways.

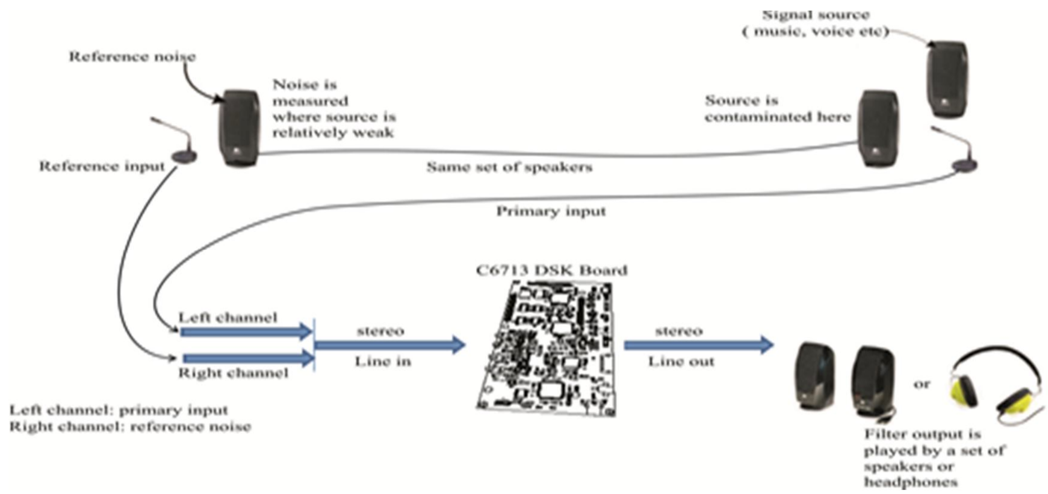


Figure 2 experimental arrangement of ANC system with DSP processor

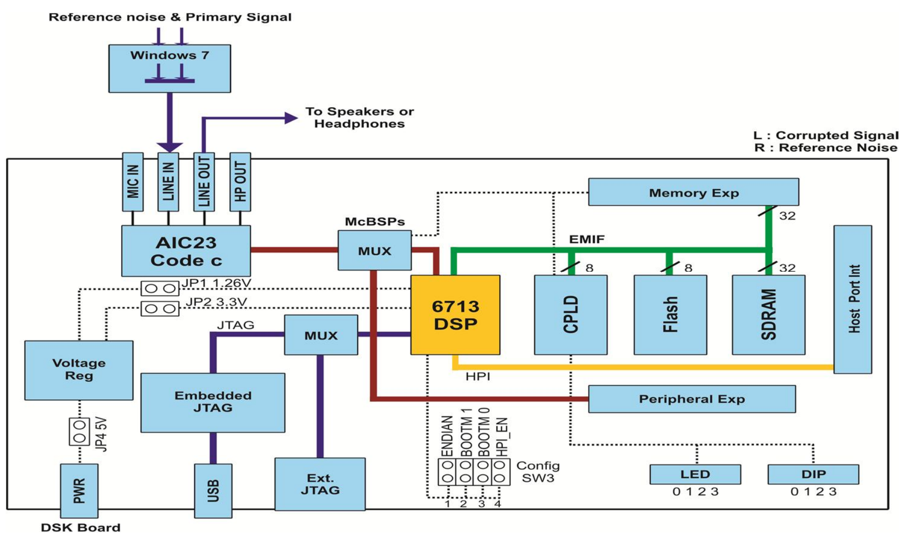


Figure 3 Entire hardware structure of the TMS320C6713 DSK hardware design

III. Research Work Carried Out

Preliminary research is carried on MATLAB software tool to realize ANC system as shown in figure 4. next , adaptive filter is used to realize stationary signals on DSK 6713. The execution of ANC arrangement for recorded non-stationary signals is conceded out on DSK 6713. Supplementary, the effective execution of real time realization of ANC on DSK 6713 DSP processor was approved out for Acoustic signals.

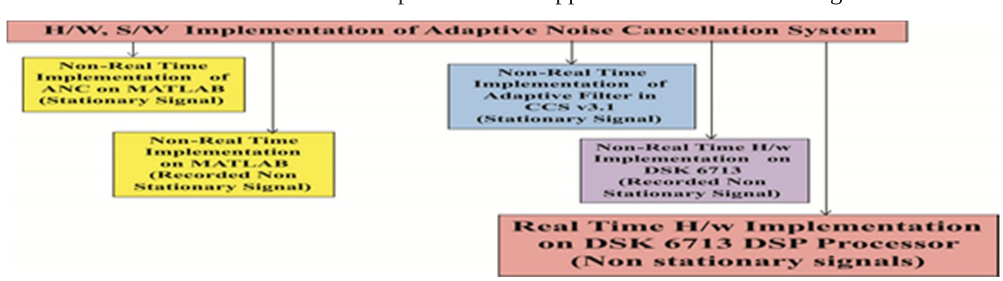


Figure 4 S/w and h/w realization of ANC system

A. Algorithm of ANC System:

The Block diagram for ANC system is in figure 5.

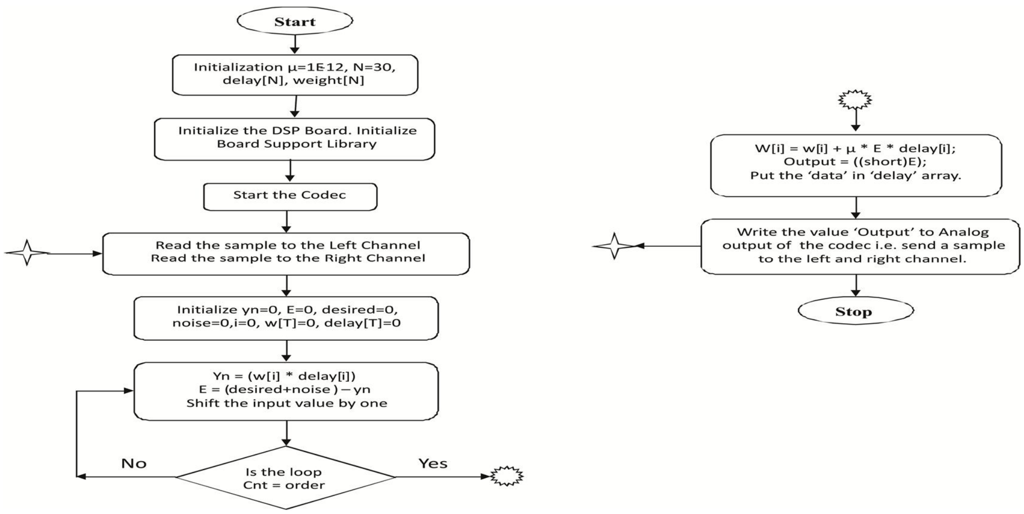


Figure 5 Flowchart for realization of software

1 Methodology for the revision procedure by means of the adaptive setting is as follows:

- Find out novel samples of the principal signal *d* and the standard input to the adaptive processor, which gives a noise signal.
- Compute the adaptive FIR filter's output *y*, applying eq.1 and 2 with an FIR filter. In the construction of Figure 1, the in general output *e* is the similar as the preferred signal *d*. compute the fault gesture by applying eq.1.
- Revise every coefficient or load by applying eq.2.
- Another time iterate all the way through the adaptive procedure until the subsequently output is obtained [9], [10].

Figure 5 shows a systematic implementation of the adaptive LMS algorithm intended for the noise deletion relevance. The adaptation rate, filter order, and amount of samples are measured as 1E-12, 30, and 60, and *Fs* = 44.1 kHz correspondingly. The on the whole output was the adaptive filter's output *y that* updates to the preferred signal *d*.

Figure 6.a & .b shown below are the output of both of the preferred Acoustic signal and the unwanted noise signal. Figure 6.c shows the output i.e. error signal E after the alteration procedure is converge to the preferred Acoustic signal. All these signals are plotted using CCS.By varying the alteration or meeting rate μ as 0.0001, 0.001, 0.01, 0.1, 0.2, 0.5, 1 are used and experimental the higher rate of alteration.

Fig. 6 (a) principal Input: Acoustic signal + noise signal

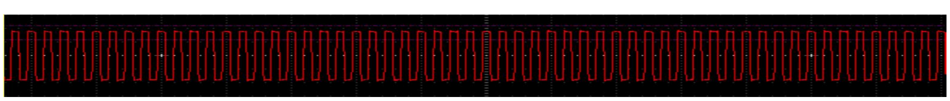


Fig. 6 (b) Standard Input: Audio signal

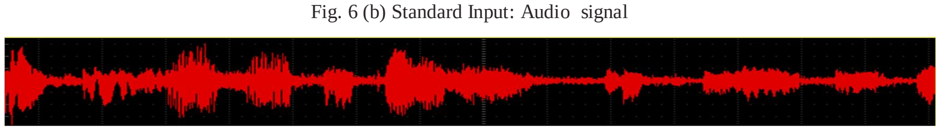


Fig. 6 (c) processed output: Acoustic signal (unique information signal)

B. Methods realized for Experiments

The realization of Algorithm is being tested three times on DSK6713 to scrutinize the potential of the intended ANC system as specified in figure 7

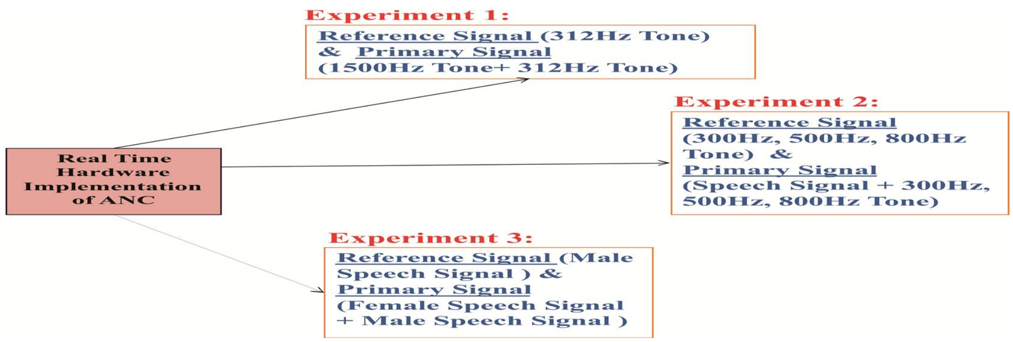


Figure 7 Algorithm implemented on TMS320C6713 DSK

The system arrangement is experimented with a existent Acoustic signal having different frequency. Three types of experiments are conducted for Acoustic speech and Noise signal to observe the competence of the intended system as revealed in figure 7.

- Initial experiment is completed on 1500Hz of preferred sinusoidal signal and 312Hz of unwanted signal. The unwanted 312Hz sinusoidal signal is being steadily abridged, at the same time the 1500Hz preferred signal is unchanged
- Subsequent experiment is completed on 300Hz of noise effected signals at the same time the additional tests are measured for 500Hz, 800Hz, 1000Hz and 3000Hz of noise effected signals having dissimilar huge amount of noise powers.
- The third experiment is completed on male Acoustic signal as a standard signal and principal signal is the female Acoustic signal Effected by male Acoustic signal having different frequency.

This considered classification is experienced under different tests with a extensive variety of processing parameters. The decrease of audio noise has been examined with steady development in S/N R with elevated precision. Subsequent are the different tests conceded out by allowing for unusual acoustic inputs. Many audio signals be effectively implemented to examine the concert of ANC.

Experiment 1

312Hz Reference Signal with 1500Hz Primary Signal : Primary Input: 1500Hz information signal + 312Hz noise signal, Reference Input: 312Hz noise signal, Filtered output: 1500Hz primary signal

3.3.2 Experiment 2:

300Hz Reference Tone Signal with Primary Speech Signal: Primary Input: speech signal + noise signal Reference Input: tone signal, Filtered output: speech signal

500Hz Reference Tone Signal with Primary Speech Signal:

Primary Input: speech signal + noise signal ,Reference Input: noise signal, Filtered output: speech signal

800Hz Reference Tone Signal with Primary Speech Signal:

Primary Input: speech signal + noise signal ,Reference Input: noise signal , Filtered output: speech signal

3.3.3 Experiment 3:

Reference Male Speech Signal with Primary Female Speech Signal: Primary Input: female speech signal + male speech signal, Reference Input: male speech signal,Filtered output: female speech signal

C. Experimental Setup

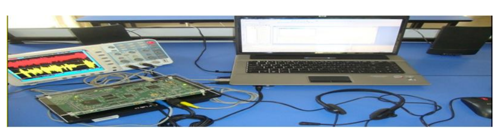


Figure 8 Hardware testing arrangement and the outcome for the anticipated ANC

The test set-up in figure 8, shows the two different WAV files which are used to get unaffected speech signal *d*, the corrupted signal *n*, the principal input signal i.e. *d+n* and the analogous input signal *n'*. The principal signal which is noise effected is generated. The generated output signal is given as input to the DSK (DSP Starter Kit) line_in a stereo jack,the principal Acoustic signal is given as in put ito the left channel and the signal with which acoustic signal is compared into the right channel. Table I shows study of the projected parameter for successful noise deletion. The finest parameters, that are measured for efficient execution of ANC on DSK, are the length of the Adaptive filter which is considered to be 30 taps (M=30). The sampling frequency which is considered to be as 44.1 KHz and the convergence aspect intended for the LMS algorithm is 1E-12. Consequently, different parameters measured for successful noise deletion were effectively tested as per the planned design (Wallace, 1992).

TABLE I. ANALYSIS OF PROJECTED PARAMETERS AND THEIR RECITAL

Sr. No.	Parameters	Used
1	Order of filter	30
2	Sampling rate	44.1KHz
3	Convergence Factor	1E-12

IV. EXPERIMENTAL RESULTS

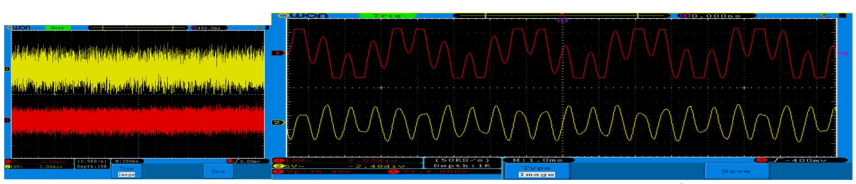
A. Experiment 1

Required: 1500Hz voice Signal, Unwanted: 312Hz voice Signal
A Required voice signal with 1500 Hz and with an additive unwanted voice signal of 312 Hz constitutes one among the two different inputs to the adaptive processing arrangement. A suggested Audio signal, with a frequency of 312 Hz, is the input to a 30 - coefficient adaptive FIR filter. The 312 Hz suggested Audio signal is associated with the 312 Hz additive unwanted voice signal but not with the 1500 Hz required signal. At every sampling moment, the output of the adaptive FIR filter is intended and the 30 weights or coefficients are reorganized by the side of the delay samples. The error signal E is the overall wanted output of the adaptive arrangement. The difference of desired and adaptive noise gives error E signal(d-n).

1. Primary input: sine (1500 Hz) + sine (312 Hz) ---- from Left Channel
2. Reference input: sine (312 Hz) ---- from Right Channel

B. Realization of the adaptive filter noise eliminator in CCS:

The adaptive noise eliminator is executed in C language as it has exceptional opening for a rapid and specialized means to understand the entire process of the structure. The outcome is tested and subsequent outputs are visualized, in figure 9. The unwanted 312 Hz signal is reduced steadily, while the required 1500 - Hz signal leftovers unaffected. The output desired is the error signal E. The faster rate of deletion was experiential with a better value of μ. And if the value µ is exceptionally large, the adjustment becomes unbalanced and consequently wants to be optimized.
The output signals of the implemented hardware are pictured.



(a) (b)

Figure 10: processed voice signal at less noise

S/N Ratio:

TABLE II. S / N RATIO ANALYSIS

Experiment 1	Input /Output	Avg. Signal Power (dB)	Avg. Noise Power (dB)	S/N (dB)
1500Hz Audio Signal + 312hz voice Signal	processed Signal	11.861	3.111	8.75
	principal Signal	12.087	8.285	3.802
	S/N enhanced			7.076

soon as the intended system is experienced with sinusoidal voice signal as revealed in the above table, it reveals SNR enhancement of 7dB at 312Hz of noise. The ANC system results are supplementary experienced by adjusting variations in the amplitude and frequency of the unwanted input voice signal in experiment 2 that shows the enhanced S/NR of the ANC (Mollaei, 2009).

C. Experiment 2: Adaptive FIR Filter for Noise Cancellation Using Desired Signal as a Speech Signal.

The subsequent test extends the preceding one to terminate unwanted sinusoidal voice signal while by means of preferred signal as a voice signal. The 16-bit preferred signal is input from the left channel and the unwanted 16-bit noise signal from the right channel.

Required: voice Signal (Shree Gajanan Vijay Granth, Adhyaya 10 (it's in a female's voice))
Unwanted: 300Hz, 500Hz, 800Hz, 1000Hz, 3000Hz of voice Signal

A preferred voice signal with a adjustable frequency, tainted with above stated noise into the left channel and on the other hand unwanted noise signal of 300Hz into the right channel is provided as input. After the execution of Algorithm, it is pragmatic that the 300Hz of noise signal gets compact steadily. The frequency of input sinusoidal outside voice intrusion gets considerably augmented from 500Hz to 3000Hz to experiment the projected system. It is experimental that the concert of ANC system enhance by variable most favorable value of µ. The figures 10 -14 shown express the output band of the error signal E subsequent to the adaptation procedure when converged to the preferred signal.

Primary Input: preferred voice Signal + 300Hz Unwanted voice Signal Reference Input: 300Hz Unwanted voice Signal

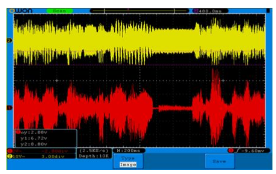


Figure10. processed voice signal at 300Hz of noise

Primary Input: preferred voice Signal + 500Hz Unwanted voice Signal Reference Input: 500Hz Unwanted voice Signal

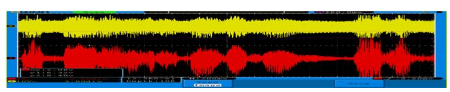


Figure 11 processed voice signal at 500Hz of noise

Primary Input: preferred voice Signal + 800Hz Unwanted voice Signal Reference Input: 800Hz Unwanted voice Signal

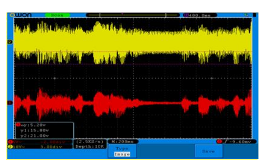


Figure 12 Filtered tone signal at 800Hz of noise

Primary Input: preferred voice Signal + 1000Hz Unwanted noise Signal Reference Input: 1000Hz Unwanted noise Signal

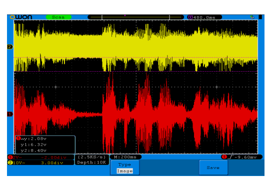


Figure 13 processed voice signal at 1000Hz of noise

Primary Input: preferred Audio Signal + 3000Hz Unwanted noise Signal Reference Input: 3000Hz Unwanted noise Signal

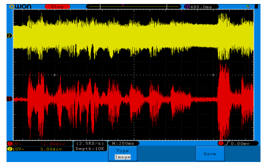


Figure 14 processed tone signal at 3000Hz of noise

S/ N Ratio:

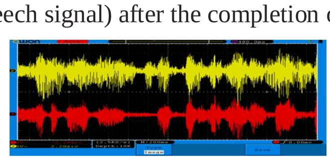
TABLE III . S/N R ANALYSIS

Experiment 2	Input /Output	Avg. Signal Power (dB)	Avg. Noise Power (dB)	SNR (dB)
Audio Signal + 300Hz voice Signal	processed Signal	12.105	3.283	8.822
	principal Signal	11.829	8.156	3.673
	S/NR Enhanced			7.238
Audio Signal + 500Hz voice Signal	Processed Signal	11.731	2.068	9.663
	principal Signal	12.425	9.493	2.932
	S/N R enhanced			8.627
Audio Signal + 800Hz noise Signal	Processed Signal	10.511	4.183	6.328
	principal Signal	11.855	9.242	2.613
	S/NR enhanced			3.923
Audio Signal + 1000Hz noise Signal	processed Signal	9.863	1.303	8.559
	principal Signal	8.331	5.352	2.979
	S/NR enhanced			5.58
Audio Signal + 3000Hz noise Signal	processed Signal	11.6405	2.329	9.311
	principal Signal	8.195	5.403	2.792
	S/NR enhanced			7.153

The outcomes obtained described in the above table is a proof of doing well and noteworthy contribution for the system vis-a-vis S / N ratio.

D. Experiment 3

Desired: Female Speech Signal (Shree Gajanan Vijay Granth, Adhyaya No.10 (in a female's voice))
Undesired: Male Speech Signal (Shree Gajanan Vijay Granth, Adhyaya No.11 (in a male's voice))
The last trial guides the principal Acoustic signal i.e. female voice signal as the wanted signal is polluted with the male Acoustic signal given as the input to the left channel and the and the signal with which Acoustic signal which is nothing but standard signal i.e. and the undesired signal which is the male speech signal as the input to the right channel. On realization of ANC system, it's experiential that the output error signal e which is converged to the preferred female speech signal, illustrate the male speech signal which is being reduced steadily. Figure 15 shows the principal signal (female + male Acoustic signal) and the preferred processed output (female speech signal) after the completion of adaptation process.



(a) & (b)

Figure 15 tentative results (a) Incoming primary signal (b) processed female speech signal

Signal to Noise Ratio:

TABLE IV .SNR ANALYSIS

Experiment 3	Input/ Output	Avg. Signal Power (dB)	Avg. Noise Power (dB)	SNR (dB)
Female + Male Speech Signals	Processed Signal	10.888	0.644	10.244
	principal Signal	10.606	6.794	3.8125
	S/N R enhanced			9.122

Thus, the structure is experienced for many voice signals. Three experiments are conducted and individual outcome are shown. The transmitted and Received Signal stay unaffected. The Signal to noise ratio at the input and output was calculated by taking most select parameters for examination and is shown in Table II, III and in Table IV. Simulations and real-time hardware execution outcome observed that both realizations are most precise.

V. CONCLUSION

By means of the LMS algorithm, that can successfully carry out actual time noise decline lacking previous information of noise figures, ANC has been analysed on Texas Instruments TMS320C6713 DSK board, productively.The real time processor consequences are analyzed for two types of standard signals i.e. noise signal and male voice signal with diverse frequecies.These signals are experienced and concluded by means of Digital Storage Oscilloscope, and the filter efficency is calculated in terms of S/N R enhancement. Thus, noise canceller might process out noise signal and male voice signal successfully in actual time.

Audio signal is observed at a variety of frequencies and voltage levels to ensure the consequence of noise, and analysed that the consequence of noise gets more important when the frequency of noise and voice signal is highly correlated otherwise noise signal has large amplitudes.It is experiential that the Acoustic signal can still be improved absolutely. It needs the order of the filter to be effective and µ value also then we can obtain the steady-state presentation, consequently. Thus accomplished that as the step size value increases within the solidity region, it improves convergence rate and also the misadjustment. And as the step size value is indirectly proportaional to decreases, the convergence period and the misadjustments. on the other hand, the filter order is directly proportional to the convergence period and also the misadjustments. In short and average power noises, the hardware architecture gives S/N R enhancement up to 7dB and at more power noise shows a 5 dB of S/N R enhancement. Depending on these outcome using the intended model in TMS320C6713 DSP system, LMS algorithm of adaptive noise eliminator can successfully determine the difficulty of removing noise from a preferred signal such as speech signal or any different type of Acoustic signal and show to be efficetively.

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