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Original Article

An Efficient FPGA-Based Adaptive Filter for ICA Implementation in Adaptive Noise Cancellation

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Abstract - *This study's objective was to examine the theoretical conception and practical application of a digital adaptive filter to improve upon the conventional filters used in digital signal processing. In order to filter real-valued signals, this study will introduce an approach that combines Independent Component Analysis (ICA) with an adaptive filter. The approach is very noise-resistant because it can precisely estimate how much noise the input signal is subjected to. The suggested architecture is implemented using MATLAB. Two audio signals and a mixture of three audio signals are converted to Verilog using HDL coder and then synthesized using a Xilinx virtex-5 FPGA device. Compared to traditional ICA and adaptive filters for FPGA implementation, the suggested design offers a significant area reduction.*

Keywords - *Blind Source Separation(BSS), Independent Component Analysis(ICA), Adaptive filter, Least mean Square(LMS), Field Programable Gate Array(FPGA).*

1. Introduction

Blind source separation(BSS) is also called Independent component analysis (ICA), which aims to recover a set of primary sources from mixtures or observations signals without being aware of mixing details and sources[1-5][11,13,16]. Although these methods were initially put forth in the framework of Neural Network(NN) modelling[7], they have since found a broad variety of applications in fields as disparate as image analysis, biomedical signal processing, telecommunications, and stock market analysis. In recent years, technical papers have started including more BSS applications to structural dynamics. One among them is signals extractions from complex mixing sources for bearing judgment; another is reliable signal extraction from noisy environments for rotating machinery; a fourth is experimental data compression for additional damage identification; and a last is the identifying the specific machine signals from complex machinery systems. By implementing ICA, The sources that are generated won't be exact replicas of the original. This is so that each variable generated will have a unitary variance. Additionally, in some circumstances, stationary coefficient-based filters cannot eliminate the distortion because it is time-dependent. To overcome this drawback, employing adaptive filters[9], which might adjust the coefficients by altering the inputs of the filter and environmental parameters and situations on which it depends. A widely aware adaptive method for filter coefficients updating in dynamic situations and unknowable situations is the Least Mean Square (LMS) [10]algorithm. This approach can use a self-correcting setup to enhance filter response and performance and has been applied in numerous applications. Due to its computing requirements, hardware implementation of adaptive systems is not always easy. Data and information are processed sequentially by digital signal processor (DSP) chips, microprocessors, and microcontrollers by fixed hardware and architecture. Typically, they process a typical instruction word in stages: fetching, decoding, and execution. As a result, they were unable to handle data concurrently.

The FPGA chips can process data continuously for various applications. FPGA applications include communication via the network, video processing technology and communication[12], and cryptography applications. This article proposed an adaptive impulse noise filtering with High-Level Synthesis (HLS) in size, speed, and accuracy on a Xilinx FPGA. The FPGA is well-suited for complicated audio processing[15]. Using the Altera Cyclone II FPGA, a uniform random noise stream was used to make an LMS adaptive FIR filter for active noise control (ANC). On a Virtex II FPGA, [18]constructed the LMS filter using three unusual architectures and contrasted their chip area and performance. Several designs were employed to build multiplication blocks to reduce hardware use and speed up calculation. A FIR adaptive filter using the Normalized Least Mean Square(NLMS) updated algorithm was first simulated using MATLAB SIMULINK and then tested on a DSP board in [20] to assess the hardware and software embedded system's behaviour for various high-frequency and low-frequency noises. For the purpose of eliminating noise from voice signals, Kim and Poularikas[22] devised the conventional Adaptive Noise Cancellation(ANC). They then assessed the effectiveness and computational complexity of each method. However, interest is shifting toward high-performance, low-cost DSP algorithm implementations on system chips.

Additionally, an FPGA requires less clock system speed for the same application than a digital signal or general-purpose processor. This is advantageous for battery-operated devices. This hardware is being used in industrial applications; FPGA implementations are used for various applications and may be the best choice. Other devices, however, might be better appropriate when low sampling frequencies or no low power consumption criteria are needed. The target hardware in this paper is FPGAs. Current FPGAs comprise many resources that support signal processing applications in addition to the typical reprogrammable possessions like lookup table (LUT) slices formed from LUT[14,22,23] registers. These resources include multipliers, Multiply Accumulate(MAC) [5]units, intellectual cores for advanced specific functions, and processors for easier programming. A design tool creates a bit stream from the hardware's VHDL description in the conventional method. High-level synthesis (HLS) tools are being used more frequently these days, allowing designers to design at much higher levels of abstraction. This work emphasises the FPGA realization of an adaptive algorithm created and how it removed the impulsive noise. This approach is cost-effective and frequently used in independent component analysis to provide cheap computing costs. The original approach has undergone some improvements to lower computing costs without sacrificing functionality.

As a result, this paper is structured as follows: The BSS problem overview is detailed in Section 2. Section 3 clarifies the basics of adaptive filters and the concept of the proposed work. ICA with an adjustable filter is explained in Section 4 and Section 5, and results and discussion is given, followed by a conclusion in section 6.

2. Blind Source Separation

An issue statement BSS is a new method for analysing data and processing signals. BSS uses the presumption of their mutual independence to attempt to recover the underlying sources from a set of observable signals. Figure 1 illustrates the BSS's fundamental structure.

This figure includes both the Signal Mixing and Separation processes. The Source Signals are received initially through the receiver; here, V1 and V2 are the two Independent audio Signals.

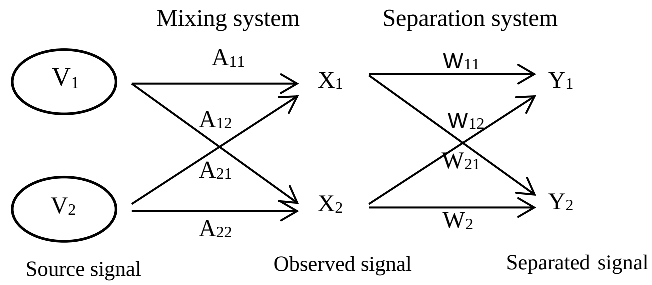


Fig. 1 Basic structure of BSS

These signals will be mixed to obtain the Mixed Signals.X1 and X2 are the Outputs obtained after mixing the Independent Source Signals with different mixing matrices. This mixing matrix depends on signal characteristics, environment, and material used. They are dependent signals, i.e. dependent on each other. Algorithms like the Independent Vector Algorithm (IVA), Principal Component Analysis Algorithm, Binary Masking Algorithm, and ICA will be used to separate these mixed signals. The mixed signals X1 and X2 will be demixed using this algorithm. This algorithm will produce mutually independent signals like Y1 and Y2. In mathematics, it is written as

$$X = \begin{pmatrix} a_{11}v_1 & a_{12}v_2 \\ a_{21}v_1 & a_{22}v_2 \end{pmatrix}$$

$$A = \begin{pmatrix} a_{11} & a_{12} \\ a_{21} & a_{22} \end{pmatrix};\ V = \begin{pmatrix} v_1 \\ v_2 \end{pmatrix}$$

$$X = AV \tag{1}$$

Equation (1) is the ICA algorithm

Where
$X = [x_1, x_2, .., x_m]^T \in R^m$measured scalar signal
$V = [v_1, v_2, .., v_m]^T \in R^n$
$A \in R^{m\times n}$ mixing matrix.
The main aim of this BSS is to find the demixing matrix A^{-1}.
$A^{-1}x$ (2)
Equation (2) recovers the underlying source V using the measured signal x.

3. Adaptive Filter

The output signals of ICA are compared to the input signals, and in most cases, this will not be accurate with the input signal, so it is to be further filtered for better output. The least Mean square(LMS) algorithm is extensively used in statistical analysis and almost every branch of adaptive filtering. In Fig. 2, the general adaptive filter is displayed. The desired signal $d(k)$ is added to the input reference signal $u(k)$ and filtered further to produce the signal $y(k)$. The $y(k)$ will further reduce the power of the erroneous signal e(k). In linear examples, the weight of the filter vector $w(k)$

convoluted with $u(k)$ results in the output of the adaptive filter. At every time step, the adaptive filter weight vector is changed depending on a function of the error signal to generate a new weight vector. Utilizing this adaptive approach, the filter's input signal $u(k)$ is filtered to create an output $y(k)$ similar to the intended signal $d(k)$.

The primary input consists of speech signal s(k) and noise n(k), while the reference input consists of noise nR(k) alone. Noise n(k) is not the same as noise nR(k); it is attenuated and filtered by the noise path. There is also a delay due to acoustic propagation. An adaptive filter is thus needed for maximal cancellation to make nR(k) as similar as possible to the n(k).

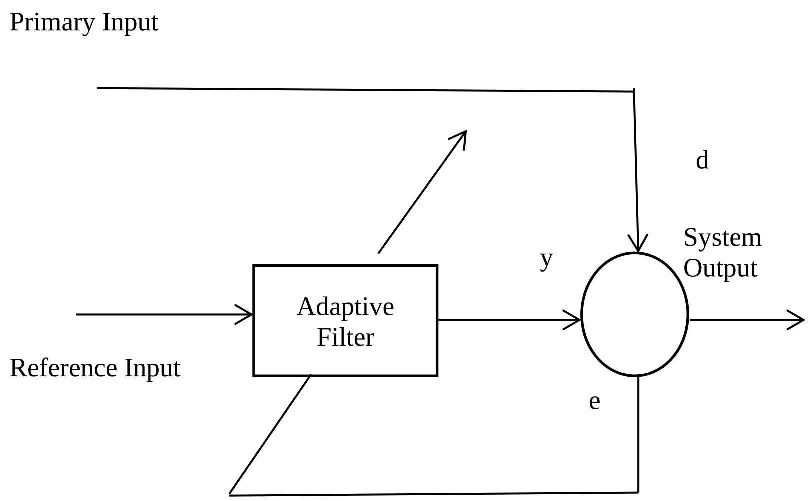


Fig. 2 General structure of Adaptive filter

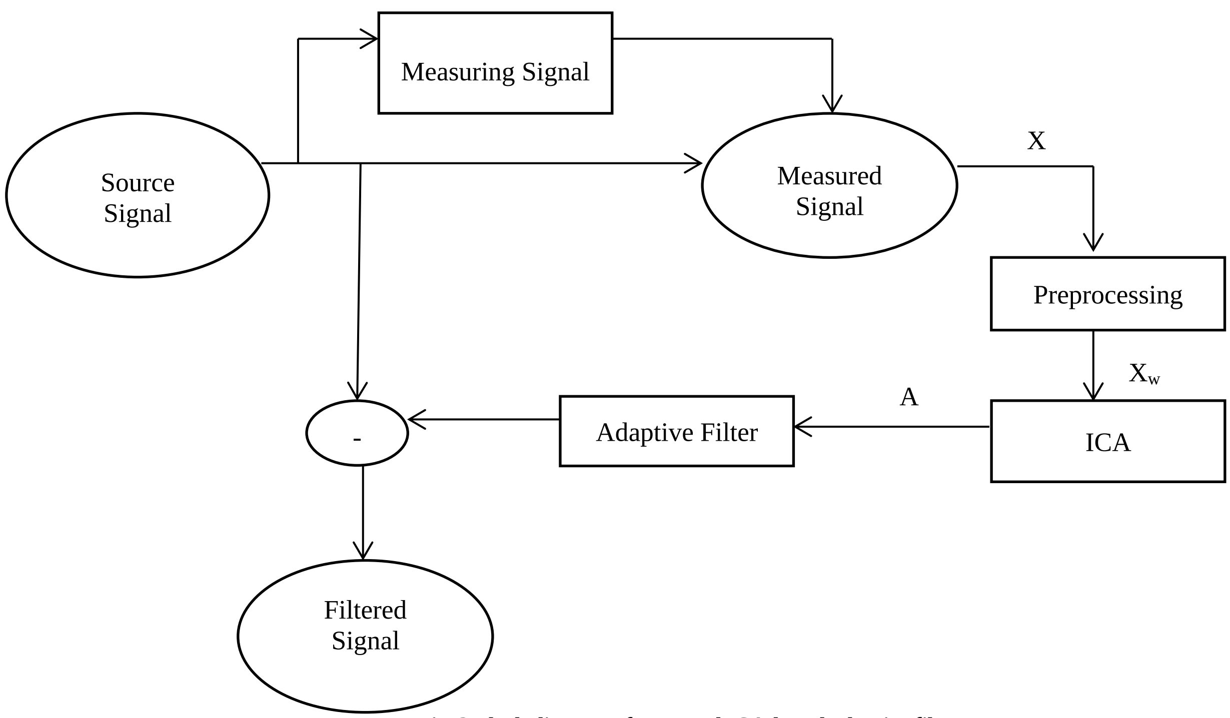


Fig. 3 Block diagram of proposed ICA-based adaptive filter

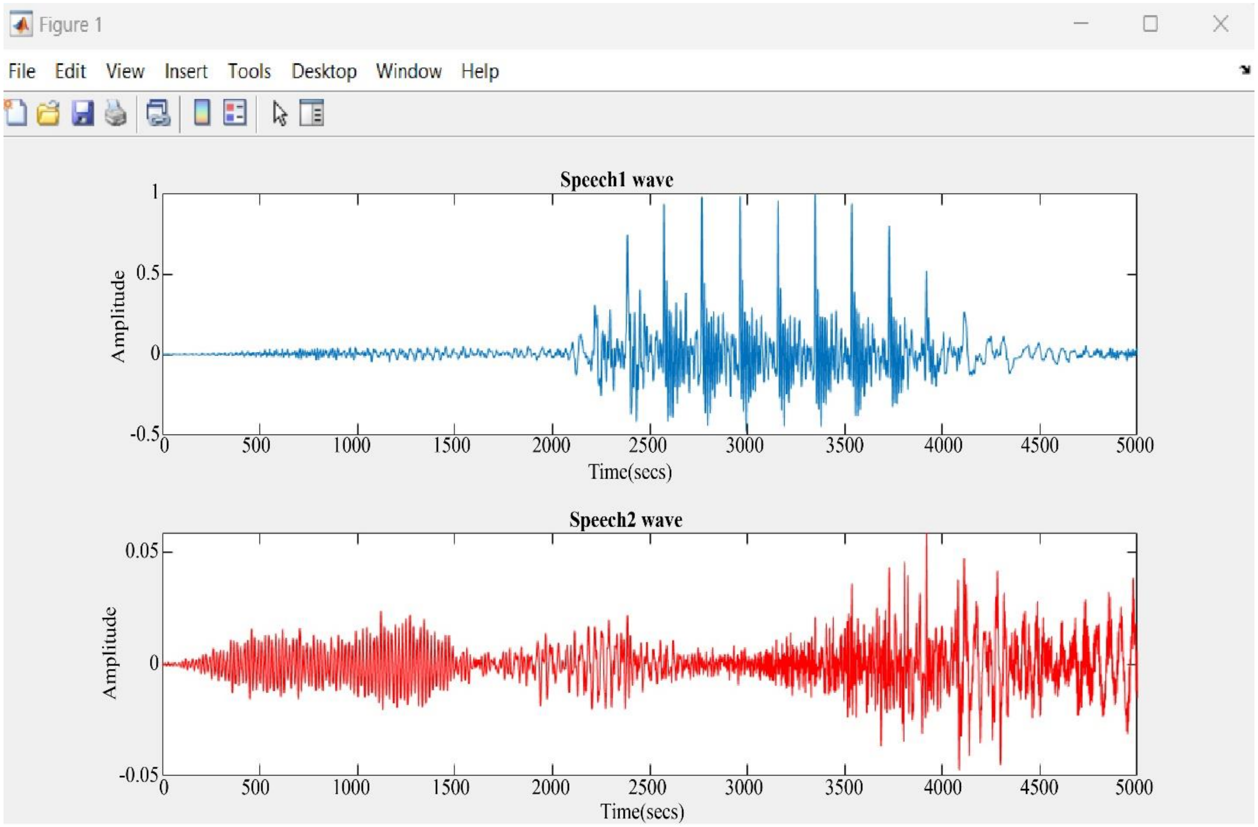


Figure 1
File Edit View Insert Tools Desktop Window Help
Speech1 wave
Amplitude
Time(secs)
Speech2 wave
Amplitude
Time(secs)

(a)

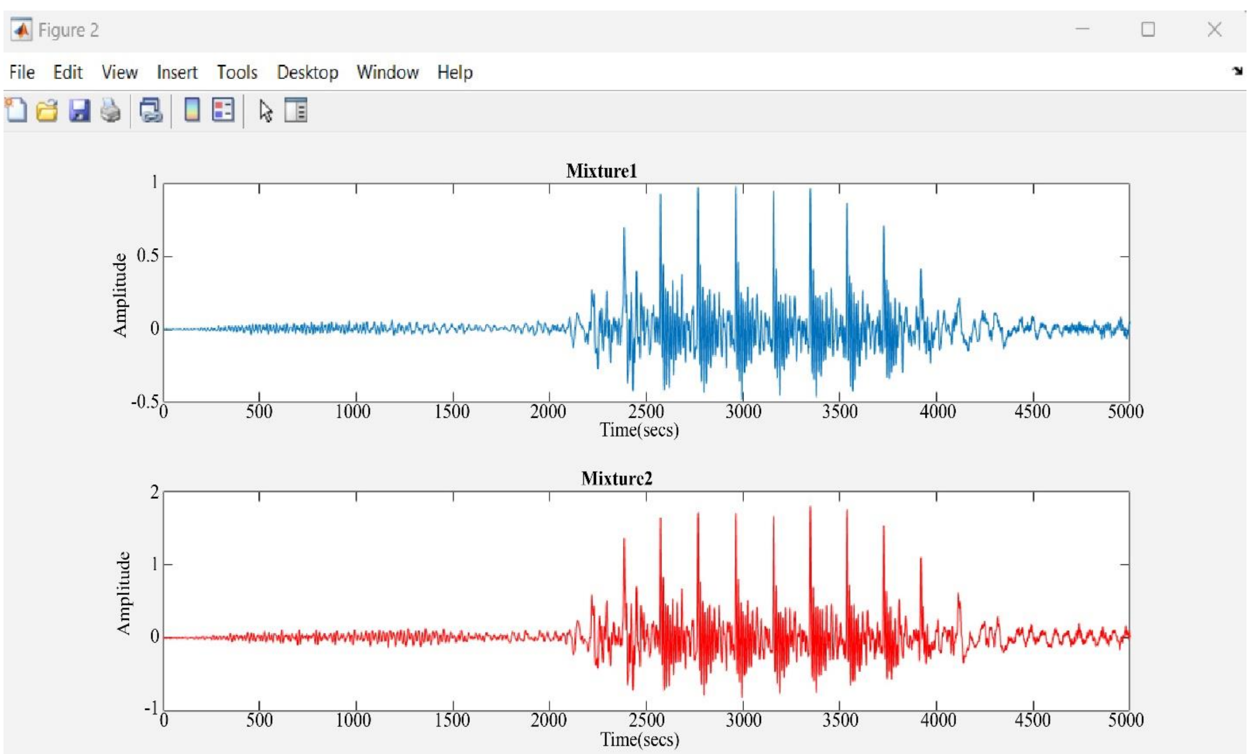
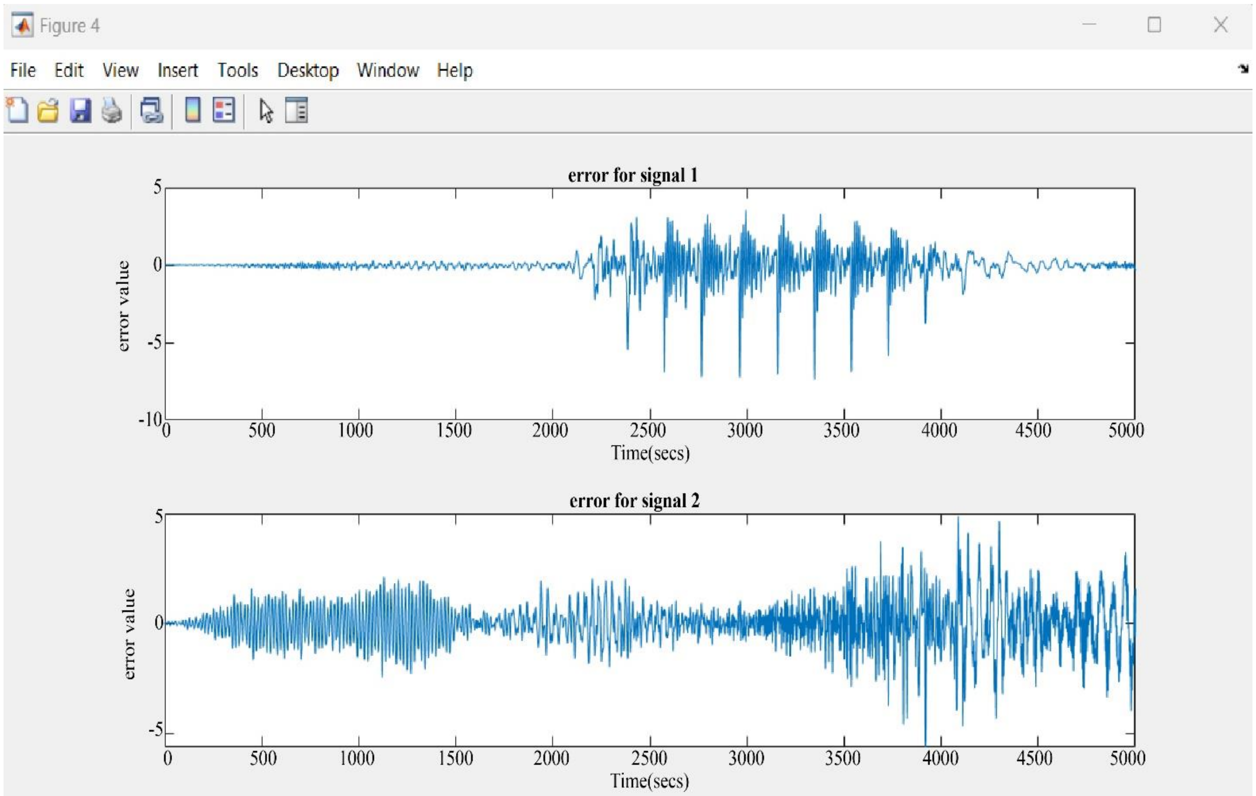
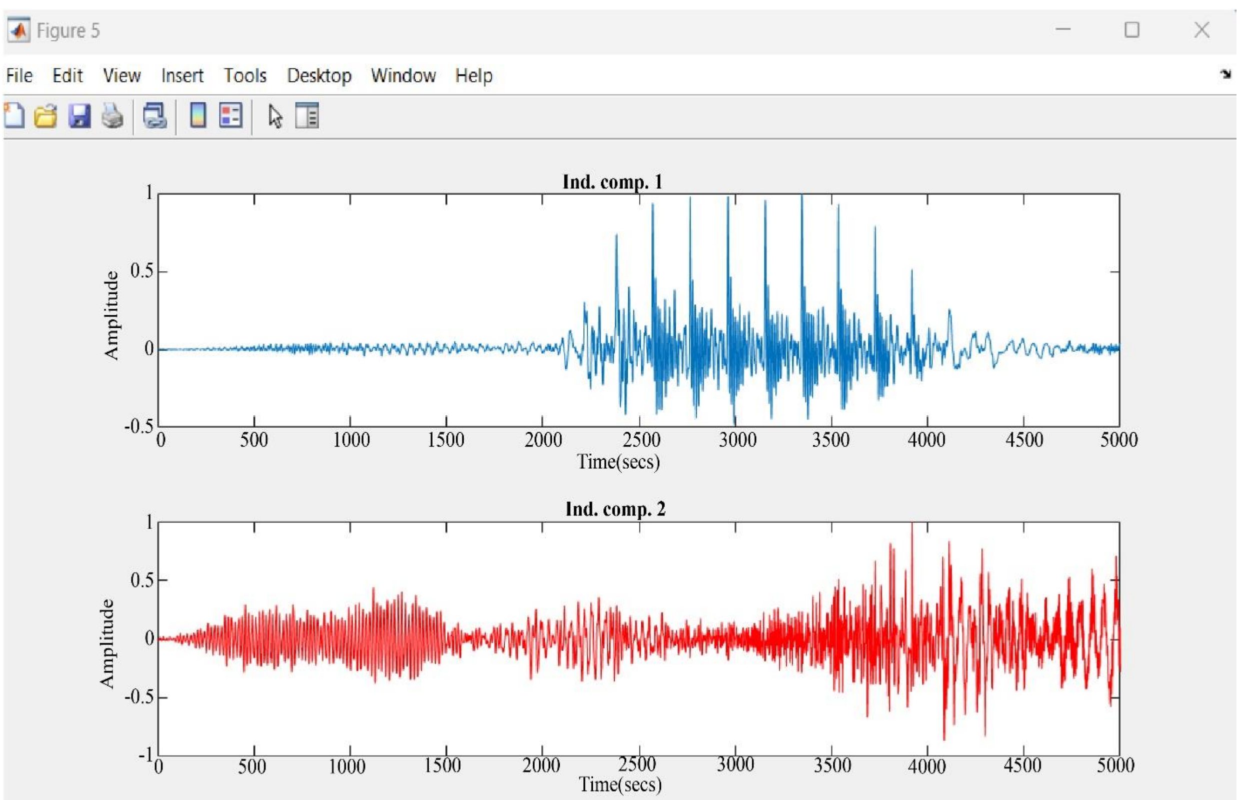


Figure 2
File Edit View Insert Tools Desktop Window Help
Mixture1
Amplitude
Time(secs)
Mixture2
Amplitude
Time(secs)

(b)



(c)



(d)

Fig. 4 Two input samples analyzed using ICA (a) speech signal, (b) mixture signal, and (c) and (d) are the separated error signal and independent signals

3.1. ICA-based Adaptive filter algorithm

Any real-time signals will be a combination of many sinewaves with different frequencies and amplitudes. And one can assert that a real signal will be described as a linear combination of sines to eliminate the starting phase from the argument. Possessing distinct frequencies are sine and cosine. Therefore, if AWGN were to damage this real signal, removing the noise is merely required to identify the important frequencies and put all sines and cosines that possess these frequencies into an ICA processing block along with the input signal. Matlab simulations have been run to assess the filtering abilities of this approach and contrast them with the outcomes of a traditional FIR filter. Figure 3 shows the processing steps that this approach uses. With certain limitations to aid with verification, the input signals are assumed to be the sum of many sines with randomly created amplitudes, various frequencies, and initial phases. The source signal is combined with the unknown signal or mixture signal and generates the mixed signal; this mixed signal is modulated, and scaling is carried out. The mixed signal is processed with the ICA algorithm, where the mixed signals are separated into individual source signals. Even if separated as individual sources, the output will not be accurate to rectify that; the output signal is given to the adaptive filter through a multiplexer, and the estimated output is taken.

4. Result and Discussion

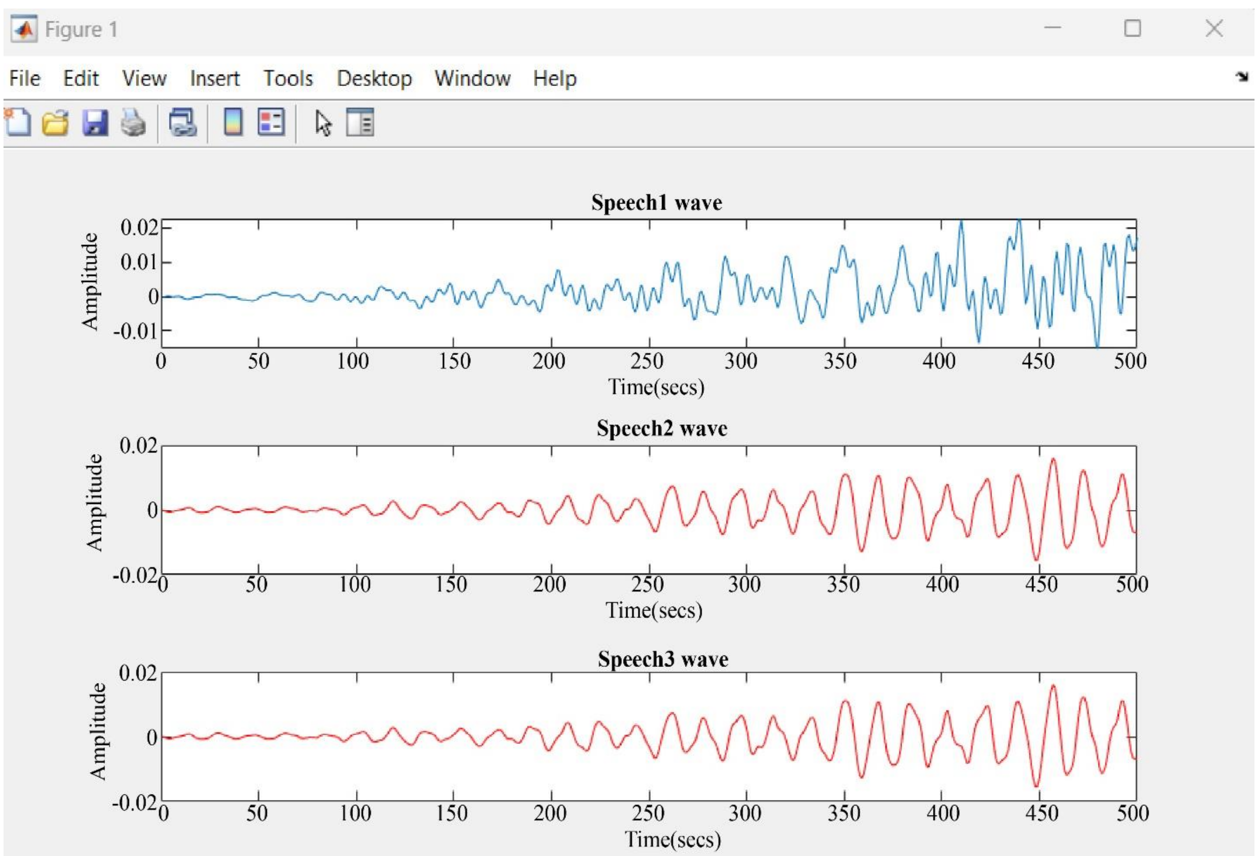
ICA is implemented for two sets of signals with 2 inputs and 3 inputs, respectively

4.1. Two Input Signals

ICA is implemented for two sets of audio signals. In 1st stage, two audio signals are taken and separated through the ICA algorithm. Their outputs are shown in figure 4, where 4(a) speech signal, 4(b) mixture signal and 4(c) and 4(d) are the separated error signal and independent signals.

4.2. Three Input Signals

ICA is implemented for two sets of audio signals. In 1st stage, three audio signals are taken and separated through the ICA algorithm, and their outputs are shown in figure 5. where 5(a) speech signal, 5(b) mixture signal and 5(c) and 5(d) are the separated error signal and independent signals.



(a)

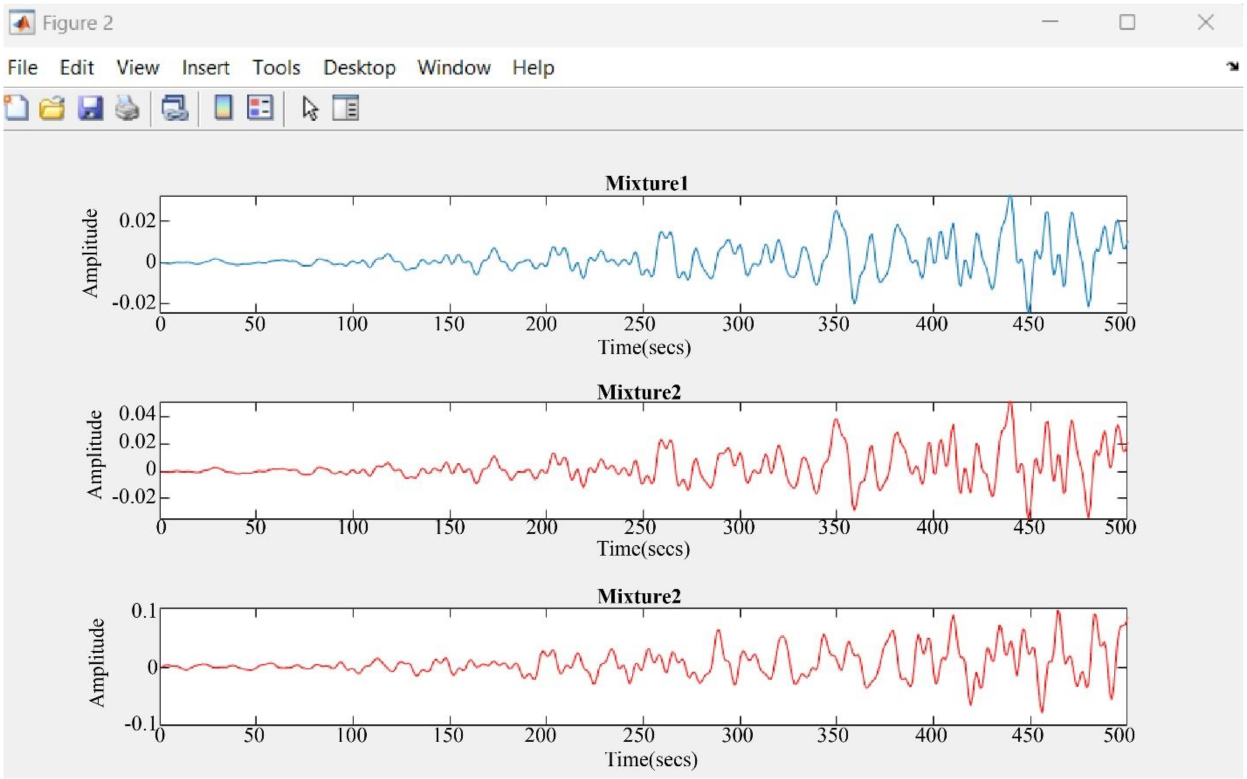


Figure 2
File Edit View Insert Tools Desktop Window Help
Mixture1
Amplitude
Time(secs)
Mixture2
Amplitude
Time(secs)
Mixture2
Amplitude
Time(secs)

(b)

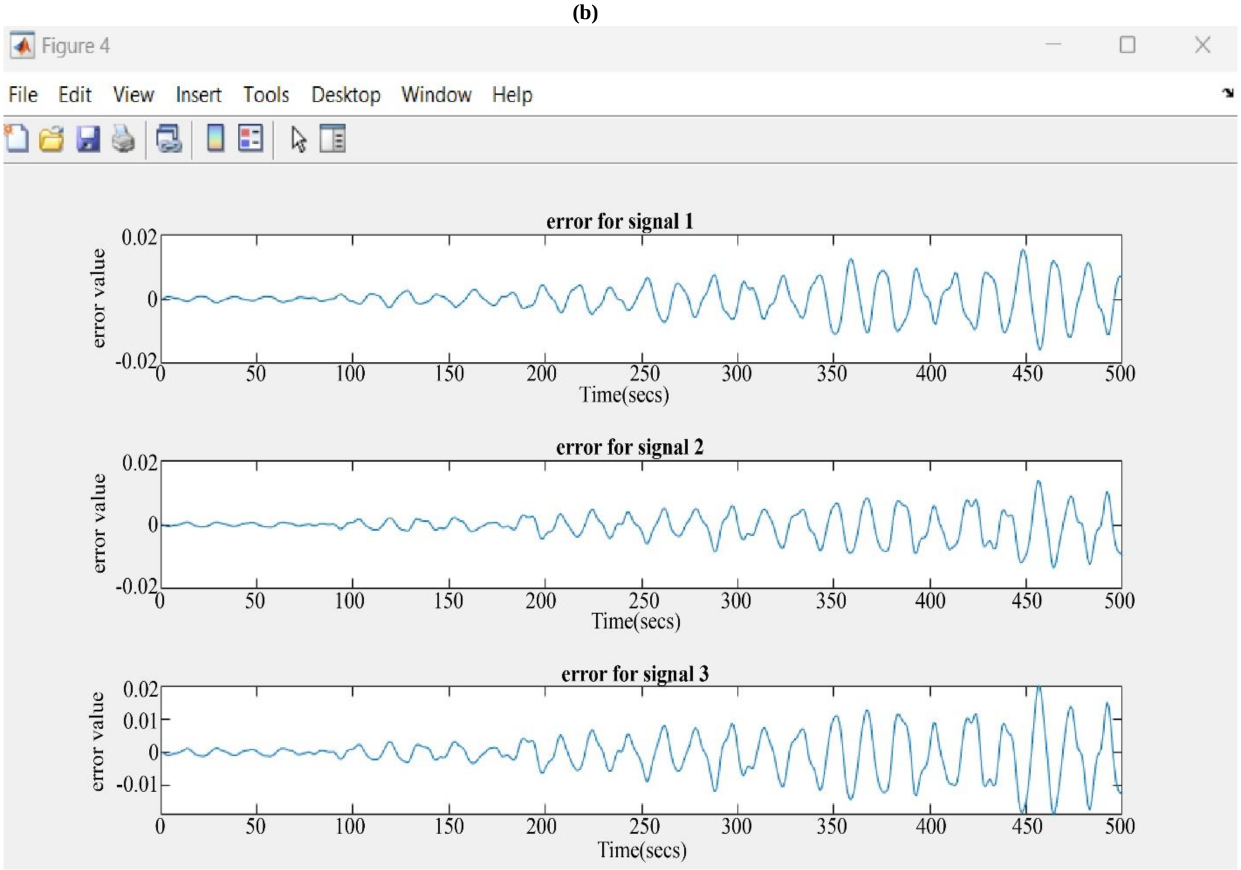
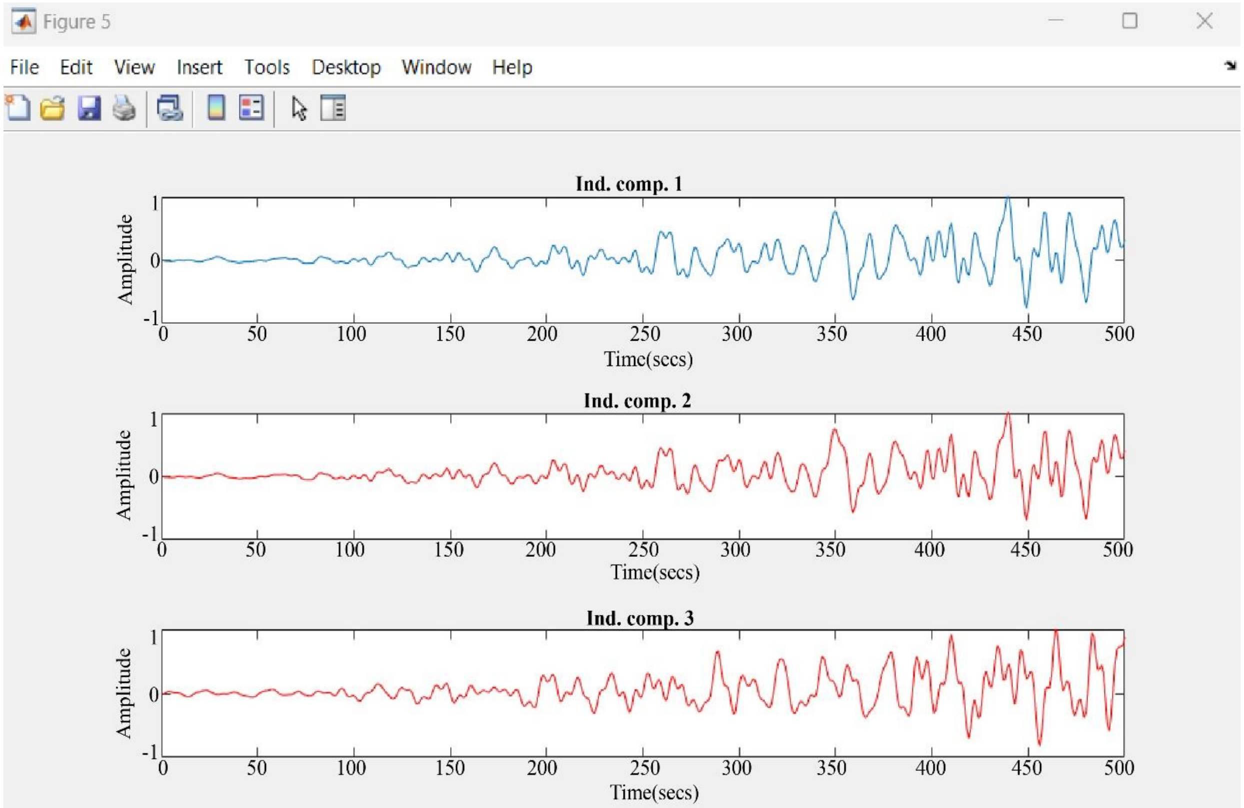


Figure 4
File Edit View Insert Tools Desktop Window Help
error for signal 1
error value
Time(secs)
error for signal 2
error value
Time(secs)
error for signal 3
error value
Time(secs)

(c)



(d)

Fig. 5 Three input samples were analyzed using ICA (a) speech signal, (b) mixture signal, and (c) and (d) are the separated error signal and independent signals

The output of the ICA signal will be the individual independent signals. These inputs are applied to the adaptive filter through a multiplexer in FPGA virtex II. HDL convert is used here to convert the MATLAB codes to Verilog codes to implement in FPGA. The schematic view of the adaptive filter is multiplexer is shown in figure 6, which is generated using the Model sim tool.

The synthesis report of the adaptive filter is shown in figure 7. Which number of slice and slice register utilization is 3% which is very less compared to the conventional architectures. By this, the hardware requirements for implementation will also reduce. This reflects in area optimization. The simulation result of the adaptive filter is shown in figure 8. In this input, signals are x_in, and x_val, e_out and y_out are output. It is clear from the output signals the frequency of the signal is increased by this, and the processing speed will also increase. The output is taken using the model sim simulation tool.

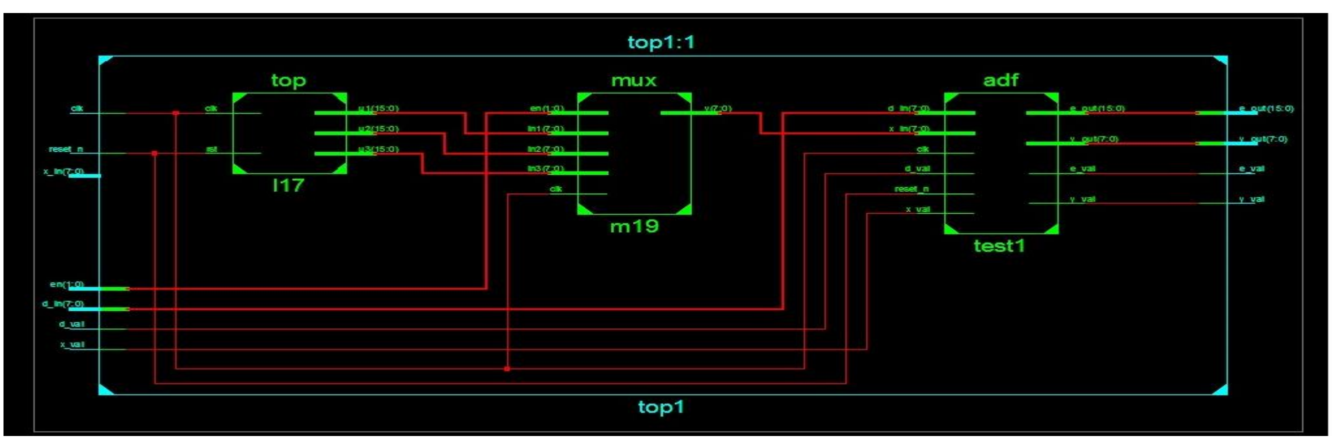


Fig. 6 Schematic view of adaptive filter

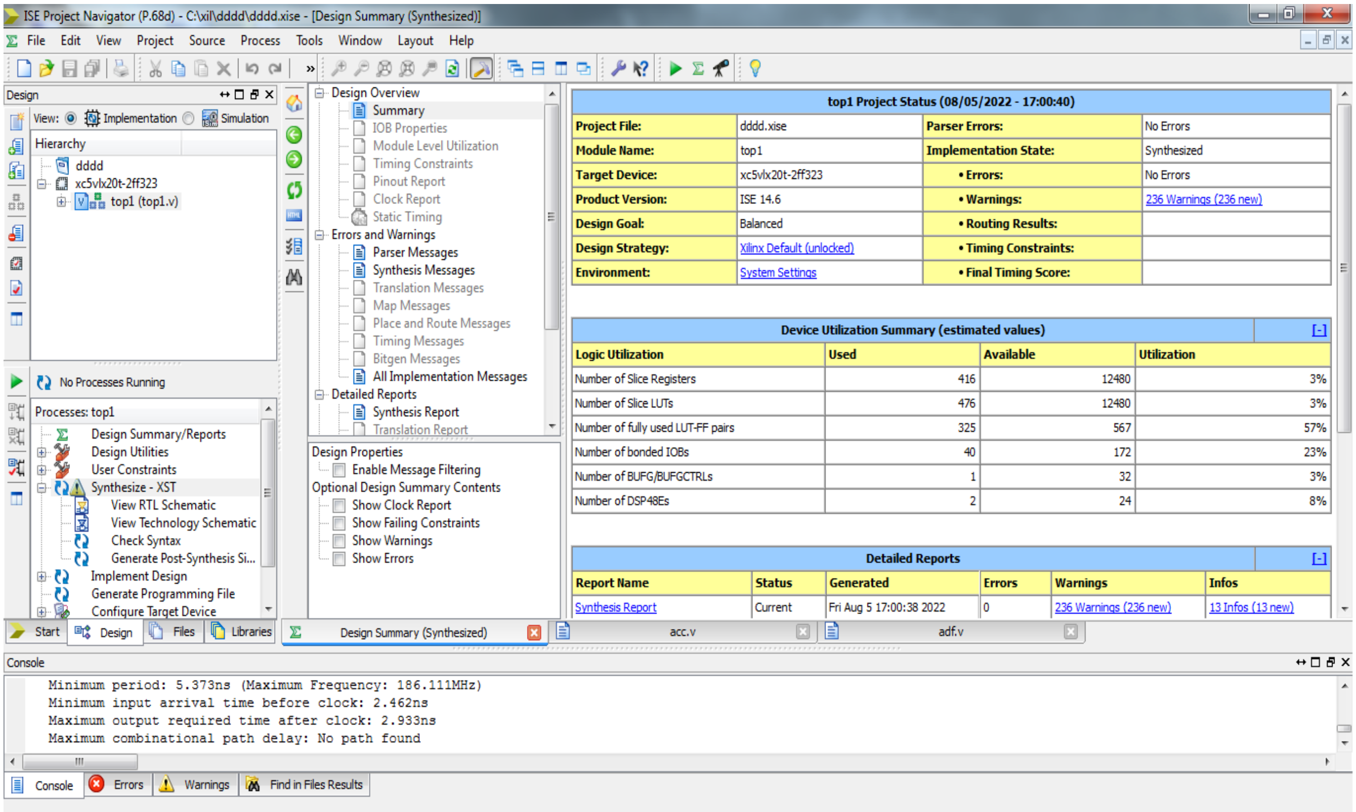


Fig. 7 Synthesis report of adaptive filter

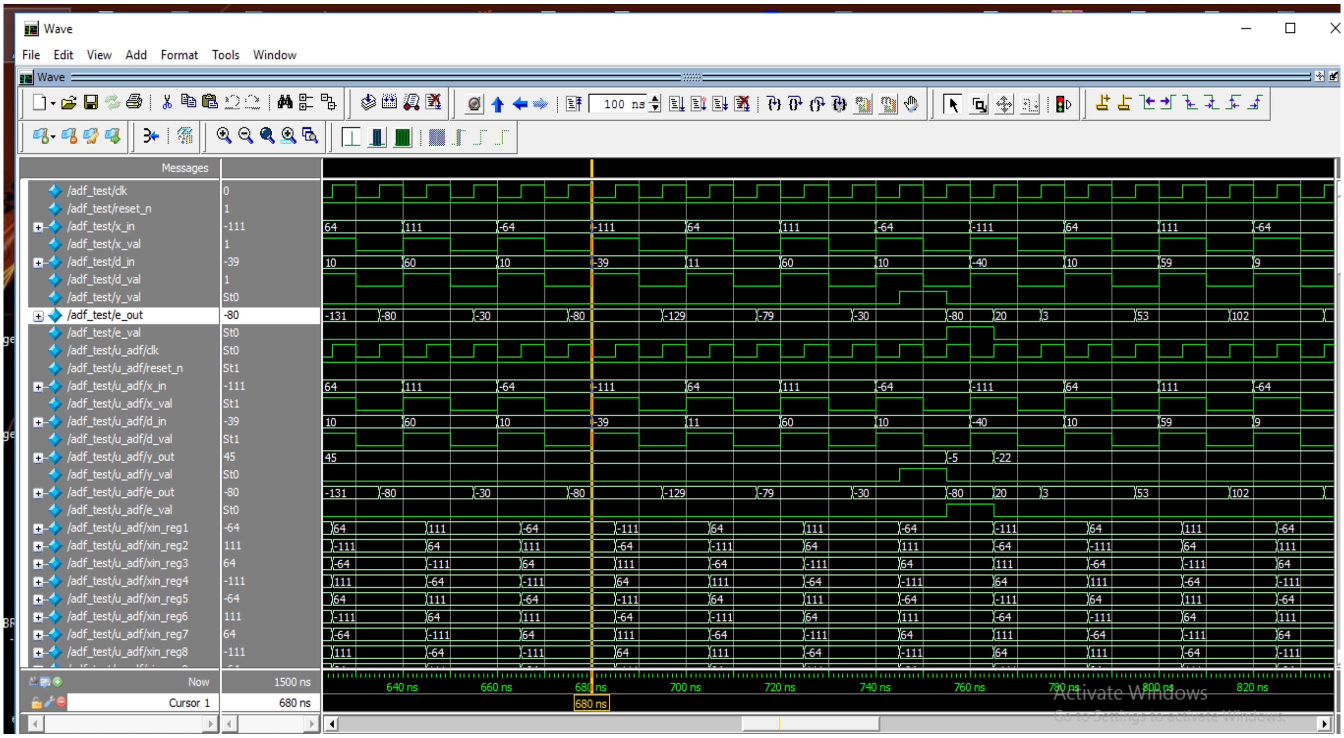


Fig. 8 Model sim simulation waveform of the adaptive filter

5. Conclusion

The objective of this work is to develop the leading adaptive filter with ICA. It produces the best estimation of the desired signal from the noisy signal and achieves adaptive noise cancellation without any reference signal. The LMS is used in this study's audio signal processing for analysis. Matlab is the main tool used in this paper for signal reading, and model sim is also used to analyse the signals in Verilog, which is further implemented in FPGA. The current approach fails to recover signals adequately when error signals seldom occur for minutes error signals and will take up a lot of space when implemented in hardware. This implemented technique proves that the proposed system provides superior signal processing output, around 43.3% improvement in the area over existing and less delay, around 31.3%.

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