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RESEARCH ARTICLE

ADAPTIVE SECOND ORDER VOLTERRA SERIES FILTER FOR REMOVING NOISE FROM NONLINEAR SYSTEM

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ABSTRACT

This paper proposes an Adaptive Volterra Series Filter with a lattice structure and delayed element for reducing noise from nonlinear system. Based on the lattice-channel structure, the Adaptive Second Order Volterra Series Filter Least Mean Square (ASVF-LMS) algorithms have been derived. In order to reduce the computational complexity and the nonlinear distortion with no need of solving the nonlinear acoustic equation, the ASVF-LMS algorithm with a lattice structure and least means square algorithm is proposed. MATLAB simulations demonstrate the control performance improvement using the proposed algorithms.

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INTRODUCTION

A system is said to be nonlinear when the output signals are not directly proportional to the inputs of given system, that is, there are no linear combinations that fully represent the system. Then it is known as a Non-Linear Time-Invariant (NLTI) system. Recently, the Volterra series expansion has been applied successfully to the analysis, removal of error and identification of nonlinear systems. It is a multi-dimensional generalization of the impulse response function. Let H_p be the p^{th} order Volterra operator. The output is defined as in equation (1).

$$y(t) = \sum_p H_p\big(x(t)\big) \qquad (1)$$

The Volterra operator is defined as the p-dimensional convolution of the input signal with the p-dimensional Volterra kernel h_p.

$$H_p\big(x(t)\big) = \int_{-\infty}^{+\infty} \dots \int_{-\infty}^{+\infty} h_p(\tau_1, \dots, \tau_p)\, x(t-\tau_1) \dots x(t-\tau_p) d\tau_1 \dots d\tau_p \qquad (2)$$

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If equation 2 is inserted into equation 3 we obtain:

$$\begin{aligned} y(t) = &\int_{-\infty}^{+\infty} h_1(\tau_1)x(t-\tau_1)d\tau_1 \\ &+ \int_{-\infty}^{+\infty}\int_{-\infty}^{+\infty} h_2(\tau_1,\tau_2)x(t-\tau_1)x(t-\tau_2)d\tau_1 d\tau_2 \\ &+ \cdots \\ &+ \int_{-\infty}^{+\infty} \dots \int_{-\infty}^{+\infty} h_p(\tau_1, \dots, \tau_p)x(t-\tau_1)\dots x(t-\tau_p)d\tau_1 \dots d\tau_p \end{aligned} \qquad (3)$$

The first element in equation (3) is recognizing a LTI system. It is the same as for linear system but remaining elements are part of nonlinear system.

Adaptive Second Order Volterra Series Filter LMS (ASVSF-LMS)

The Volterra series is employed to model non-linear systems. Its behavior is similar to the Taylor series.If the output of a nonlinear system depends strictly on the input at that particular time, Taylor series can be used for approximating its response to a given input. In the Volterra series, the output of the nonlinear system depends on the input to the system at all other times. This provides the ability to capture the 'memory' effect of devices like capacitors, inductors. For a discrete-time and causal nonlinear system with memory, with input x[n] and output y[n], the Volterra series expansion is given by:

$$y(n) = \sum_{m_1=0}^{\infty} h_1(m_1)x(n - m_1)$$
$$+\sum_{m_1=0}^{\infty}\sum_{m_2=0}^{\infty} h_2(m_1, m_2)x(n - m_1)\,x(n - m_2) \qquad (4)$$

Where, for simplicity considered a nonlinear model up to the third order. In the above representation equation (4), the functions h_i, i = 1,3, represent the kernels associated to the nonlinear operators H_i[x [n]].The input-output relation can also be written in terms of nonlinear operators as indicated in relation.

$$y[n] = h_0 + h_1[x[n]] + h_2[x[n]] + h_3[x[n]] \qquad (5)$$

The nonlinear model described by these two relations above is called a third order Volterra model. Note that the above representation has the same memory for all nonlinearity orders. In the most general case, the first relation may use different memory for each nonlinearity order. A simple adaptive volterra series filter is shown in Fig.1.

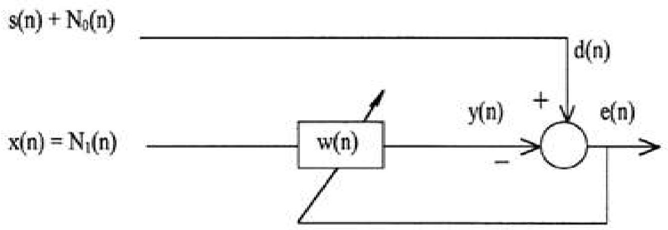

Fig. 1. Concept of Adaptive Filter to Remove Noise

The discrete form of volterra series filteris given as

$$y(n) = H_0 + \sum_{p=1}^{P} H_p x(n) (6)$$

Where,

$$H_p x(n) = \sum_{i_1=a}^{b} \cdots \sum_{i_1=a}^{b} h(i_1 \ldots . i_p) \prod_{j=1}^{p} x(n \quad i_j)$$

Here a, b $\in Z \cup \{-\infty, +\infty\}$ and P $\in N \cup \{+\infty\}$

$h_p(i_1,,,,,i_n)$, h_0 are called Volterra kernels. If P is finite, the series operator is said truncated. If a, and P are finite, the series operator is called doubly finite Volterra series.

System model designing of ASVSF-LMS

An adaptive second order Volterra series filter has been designed for removing noise from nonlinear system using LMS adaptive algorithm. LMS adaptiveVolterra filter help for low computational complexity and less memory space to store result. In comparison with other filters, the proposed Adaptive LMS Volterra Series Filter gives best Error signal (ERS) characteristics.As shown in the block diagram in Fig. 2 of the proposed model there are two paths, primary and secondary. Primary path is called desired signal d(n) path and secondary path called adaptive or learning path. This generates the output (adaptive filter output) y(n) and help to find the error signal e(n) after comparing with the desired output d(n). This process will be repeated until error signal will zero. This process will give a graph between Error signal (ERS) and number of samples called iteration.

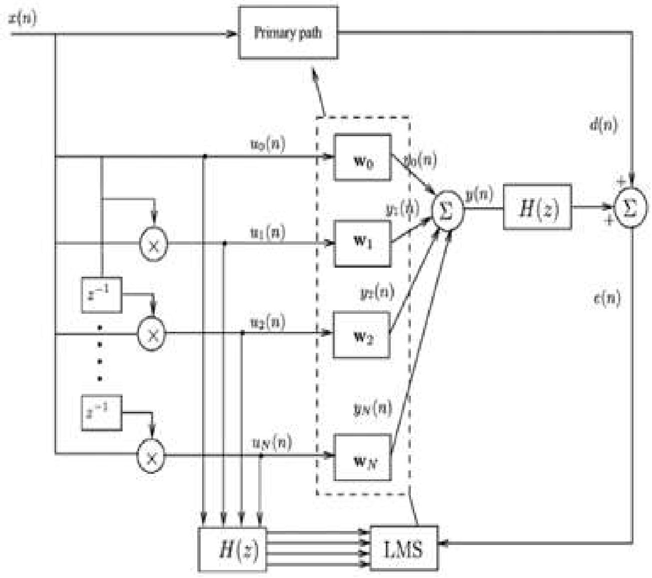

Fig. 2. Block Diagram of Adaptive Filter for Noise Reduction

MATLAB/SIMULINK software has been employed for the simulation purpose of the above model of the Volterra filter.

The main features will be as follows:

- Identify the parameters of ASVSF for designing the model.
- Scripting the LMS algorithm.
- Design and simulation of adaptive Volterra model.
- Implementation of LMS algorithm into Voltera model. Analysis of the result is obtained as performance of ERS v/s sample time

Adaptive Volterr Series Filter with LMS algorithm has been designed and simulated for removing the noise of nonlinear system as shown in Fig. 3. A discrete sine waveform has been considered as desired signal d(n) with noise (Band Limit White Noise) called N_1, has been applied in ASVSF-LMS filter through the error signal (ERS), ERS error is a difference of desire and ASVSF output signal. ASVSF output comes from designed filter. ASVSF model have two input name error signal (ERS) and Band Limit White Noise signal named as N2, which is used as reference signal for ASVSF. For minimize the error signal (ERS) noise N1 and N2 should be equal or nearby equal.The internal structure of the ASVSF-LMS contains delayed elements, multipliers and summers for performing operations.Signal required for performing operations are reference signal and ERS signal. These two signal followed by LMS algorithm, which helps to updates kernel's parameters. For optimum performance step size (practically 0.5) is used.

Simulation result and analysis

The input to the ASVSF-LMS is the sinusoidal waveform, which is nonlinear signal, with the noise added to it. Fig. 4 shows the desired signal, noise and the desired signal added with noise.

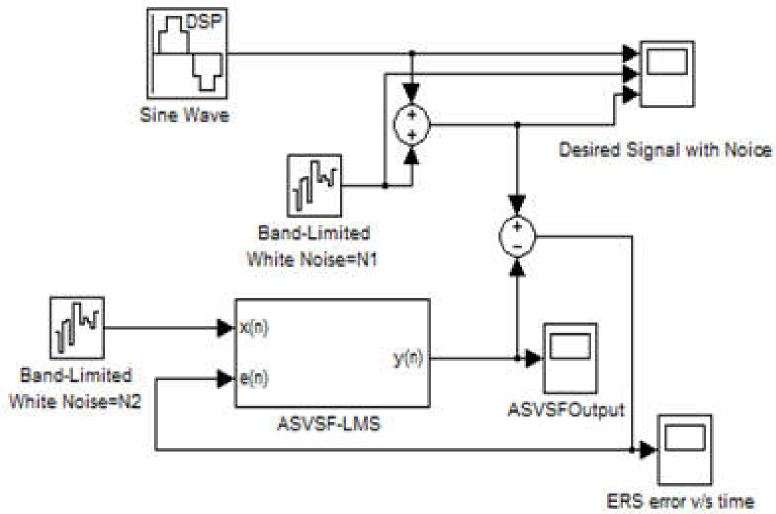

Fig.3. SIMULINK Model of ASVSF-LMS and Measurementof ERS

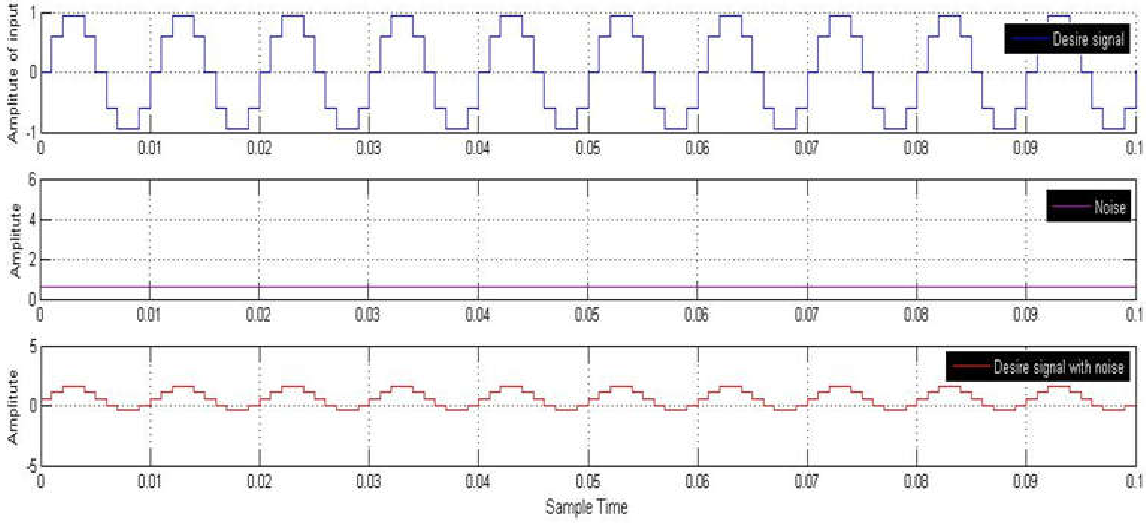

Fig.4. Input Waveforms for the ASVSF-LMS as Desired Signal without Noise and with the Noise

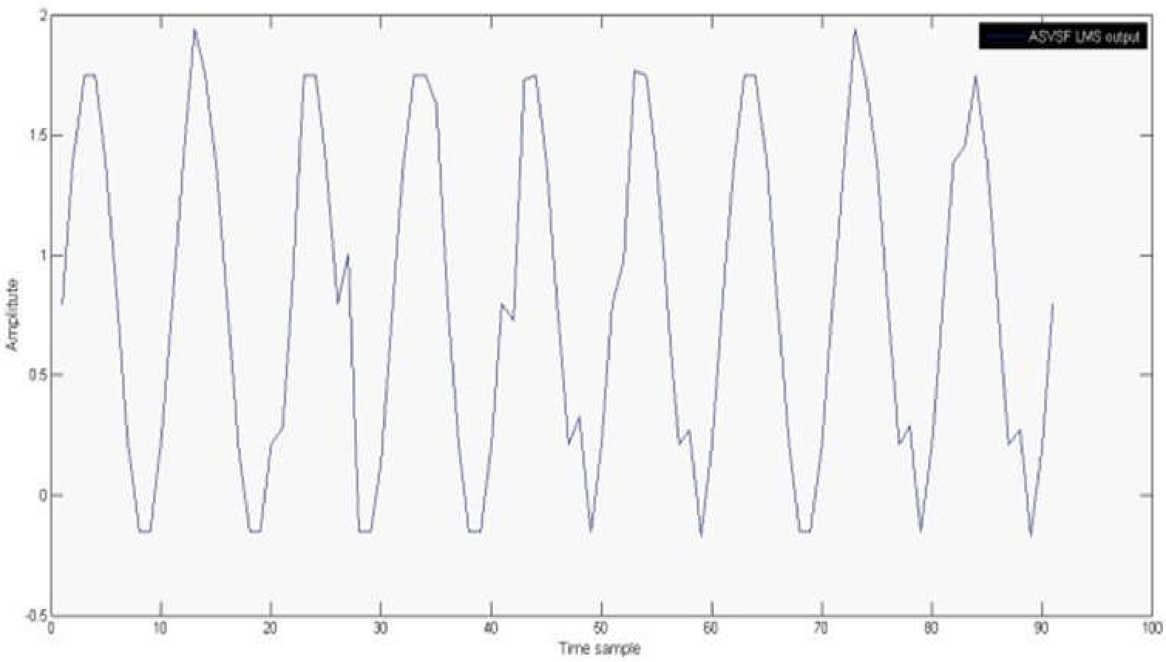

Fig. 5. ASVSF-LMS Output Waveform

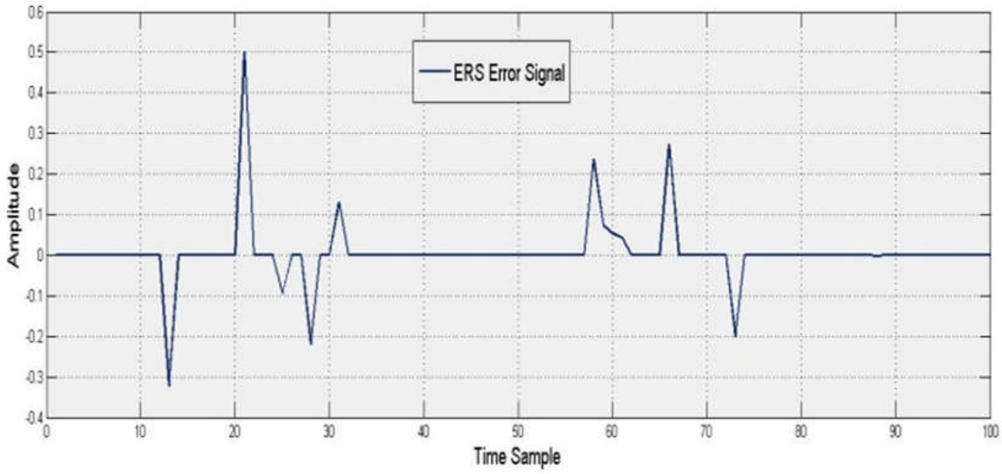

Fig.6. ERS Error of the ASVSF-LMS Model with Time

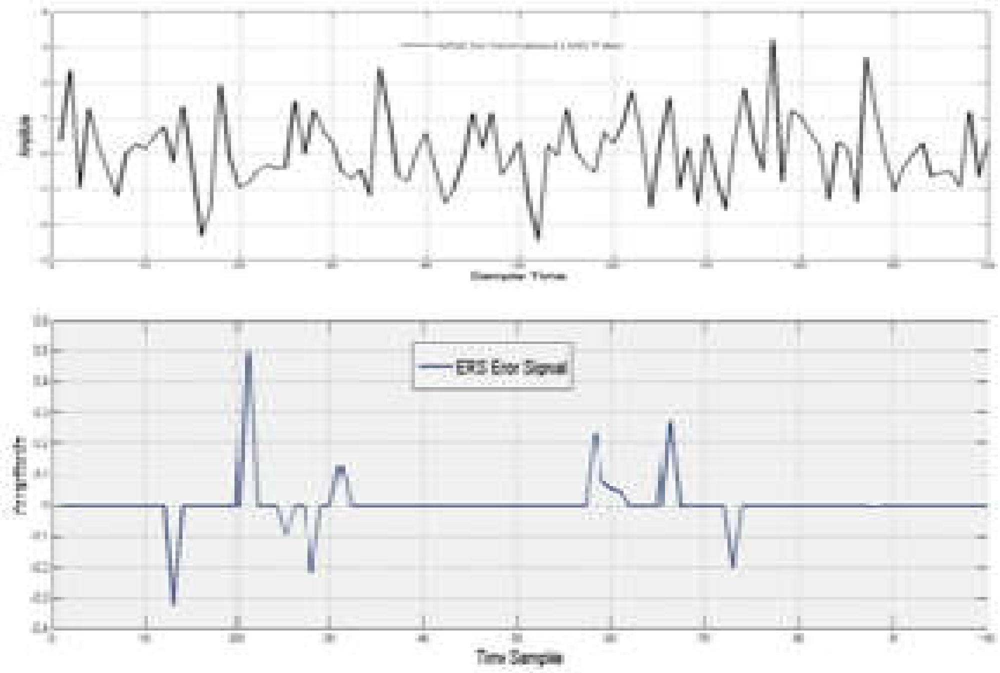

Fig. 7. Comparison of error signal (ERS) between the Normalized LMS Filter and ASVSF Filter

The output of the ASVSF-LMS filter y(n) is obtained as shown in Fig.5. Designed filter ASVSF-LMS at MATLAB/SIMULINK gives these output waveforms and then it used for compare with desired signal and it comparison give ERS error.The ERS is the difference of ASVSF output and desired signal (with noise), its measured from the system model and results of ERS is decreasing or near to zero with time increases as shown in Fig 6.For show the better performance of designed Adaptive Volterra Series filter, we compare the resultant ERS error signal of ASVSF with the normalized LMS filter's error signal and find comparatively better result.

Conclusion

This paper presented an Adaptive Volterra Series Filter (ASVSF) structure with Least Mean Square (LMS) algorithm for removing noise from nonlinear system. Simulation and with that of a Normalized LMS filter or conventional LMS filter at a much-reduced complexity. Furthermore, this new structure is able to produce a more-accurate with minimum ERS error, proposed system model give advantages compared with Normalized-LMS or equality models for computational complexity, speed of response, less ERS error, linearity and the results showed in this paper. The ability of the proposed structure to exploit the time and frequency-response nature of the system nonlinearity, allowing for additional computational reductions.

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