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Compression and Denoising of Speech Signal Using Cellular Automata Concept

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Abstract—Cellular Automata (CA) are made up as of a regular grid of cells, all of which can be in one of a finite set of states. At a particular instance the state of every cell is updated in parallel, and is determined as a function of the value in the neighborhood cell in the preceding time step, i.e. by a set of cell state transition rules. CA is discrete dynamical systems, furthermore their ease coupled by means of their complex behavior has made them popular for simulating complex systems In the previous work, researchers have implemented cellular automata to remove noise from the input signal. But this method does not give the best results in real time noisy environments like airport. We planned to propose a hybrid cellular automata based system which consists of combination of LMS(Least Mean Squares) estimation applied to cellular automata for removing the noise.

By this approach we intend to reduce the noise from the input sound or speech signal and to improve the signal to noise ratio, reducing the time of denoising the speech signal. A cellular automata takes large time as it analyses the system of speech both finegrain and coarse but we can apply coarse denoising based on LMS and finegrain denoising based on cellular automata. Thereby reducing the delay and improving the efficiency of denoising. After denoising using proposed method, compression of the speech signal is also carried out by cellular automata approach which is one of the additional feature of our proposed work. The proposed method shows the SNR is improved and also the computation time became less and also due to compression the size of data is reduced.

Keywords—*Cellular Automata, LMS,Coarse denoising,Finegrain denoising*

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I. INTRODUCTION

Cellular Automata (CA) are made up from a regular grid of cells, each of which can be in one of a finite set of states. At a given time step the state of each cell is updated in parallel, and is determined as a function of the values in the cell's neighborhood during the previous time step, i.e. by a set of cell state transition rules. CA is discrete dynamical systems, and their simplicity coupled with their complex behavior has made them popular for simulating complex systems.

Cellular automaton is a discrete dynamical system. Space, time, and the states of the system are discrete. Each point in a regular spatial lattice, called a cell, can have any one of a finite number of states. The states of the cells in the lattice are updated according to a local rule. That is, the state of a cell at a given time depends only on its own state one time step previously, and the states of its nearby neighbors at the previous time step. All cells on the lattice are updated synchronously.[1]

Speech is fundamental way for humans to communicate information. The major purpose of speech is communication. Speech can be defined as the response of vocal track to one or more excitation signal. Generally speech signals are corrupted by several forms of noise such as competing speakers, background noise and in addition they are subjected to distortion caused by communication channels.

Denoising means removal of noise from signal [2]. Noise reduction of audio signals is a major challenge in speech enhancement, speech recognition and speech communication applications. In all practical situations, the received speech waveform has a some type of noise component. This noise can be a result of the imperfect precision involved in coding the transmitted waveform (quantization noise), or owing to the addition of acoustically coupled background noise [3]. Depending on the quantity and form of noise, the quality of the received waveform can vary as of being a little degraded to being irritating to listen to, and lastly to being entirely meaningless. LMS adaptive filtering has become one of the effective and popular approach for the speech enhancement. Speech enhancement improves the signal quality by suppression of noise and reduction of distortion. Speech enhancement has many applications; for example, mobile communications, robust speech recognition, low quality audio devices and hearing aids etc.

Huge amount of data transmission is not simple both in terms of transmission and storage. Speech compression is a process to convert human speech into an encoded form in such a way that it can later be decoded to get back an original signal. Compression is fundamentally to remove redundancy between neighboring samples and between adjacent cycles. Major purpose of speech compression is to represent signal with lesser number of bits. The reduction of data should be done in such a way that there is acceptable loss of quality [4]. In applications, for example the design of multimedia work stations and high quality audio transmission and storage, the objective is to attain transparent coding of audio and speech signals at the lowest possible data rates. Bandwidth cost money, hence the transmission and storage of information becomes expensive. However, if we can use less data, both transmission and storage become cheaper.

II. OUR APPROACH

In the previous work, researchers have implemented cellular automata to eliminate noise from the input signal. But this method does not give the best results in real time noisy

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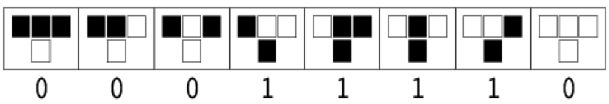
environments like airport. We planned to propose a hybrid cellular automata based system which consists of combination of LMS (Least Mean Squares) estimation applied to cellular automata for removing the noise. By this approach we intend to reduce the noise from the input sound or speech signal and improve the signal to noise ratio, reducing the time of denoising the speech signal. A cellular automata takes large time as it analyses the system of speech both fine grain and coarse but we can apply coarse denoising based on LMS and fine grain denoising based on cellular automata. Thereby reducing the delay and improving the efficiency of denoising. Also due to compressing the data using CA less storage space is required.

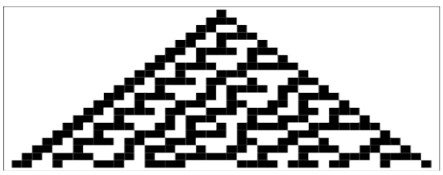
A. *Cellular Automata*

Because of simple arrangement of cellular automata (CA) to form complex behavior system, it has attracted many researchers from various areas. Cellular automata primarily announced by Ulam and Von Neumann in 1950's and also discussed in the book of Wolfram 'A New Kind of Science' with the thought of obtaining models of biological self-reproduction.. In cellular automata state of a cell at the next time step is determined by the current states of a surrounding neighborhood of cells along with its own state and is updated synchronously in discrete time steps. Cellular automaton is a discrete dynamical system. Space, time, and the states of the system are discrete. Each point in a regular spatial lattice, called a cell, can have any one of a finite number of states.

B. *Elementary Cellular Automata*

The simplest class of one-dimensional cellular automata. Elementary cellular automata have two possible values for each cell (0 or 1), and rules depend only on nearest neighbor values. Consequently, the evolution of an elementary cellular automaton can be explained by a table specifying the state a given cell will have in the next generation based on the value of the cell to its left, the value the cell itself, and the value of the cell to its right. Since there are $2x2x2=2^3=8$ possible binary states for the three cells neighboring a given cell, there are total of $2^8=256$ elementary cellular automata, each of which can be indexed with an 8-bit binary number (Wolfram 1983, 2002). For example, the table giving the evolution of rule 30 = $(00011110)_2$ is illustrated. In this diagram, the possible values of the three neighboring cells are shown in the top row of each panel, and the resulting value the central cell takes in the next generation is shown below in the center. n generations of elementary cellular automaton rule r are implemented as CellularAutomaton[r, {{1}, 0}, n].





The evolution of a one-dimensional cellular automaton can be illustrated by starting with the initial state (generation zero) in the first row, the first generation on the second row, and so on. For example, the figure above illustrated the first 20 generations of the rule 30 elementary cellular automaton starting with a single black cell.

C. *LMS*

An adaptive filter is a self-designing system that depends for its operation on a recursive algorithm which makes it possible for the filter to carry out satisfactorily in an surroundings where information of the related statistics is not necessary. The algorithm starts from some predetermined set of initial conditions, representing whatsoever is known regarding the surroundings. The filter updates the filter coefficients at every time instant is Wn+1= wn+Δwn and the adaptive algorithm generates this correction factor based on the input and error signals. The LMS algorithm is a linear adaptive filtering algorithm, which, in general consists of two basic processes: a filtering process and an adaptive process [5].The output of the least mean square adaptive filtering is calculated as follows, the output y(n) is given by w(n)*x(n), then the error signal e(n) is estimated as the difference between desired signal d(n) and the output signal y(n), now using this error signal value, the weight of the filter is adapted as

$$\hat{w}(n+1) = \hat{w}(n) + \mu x(n)e(n)$$

where
e(n) is the error signal,
x(n) is the input signal vector,
μ is the step-size parameter,
ŵ(n) is the tap-weight vector,
d(n) is the desired response

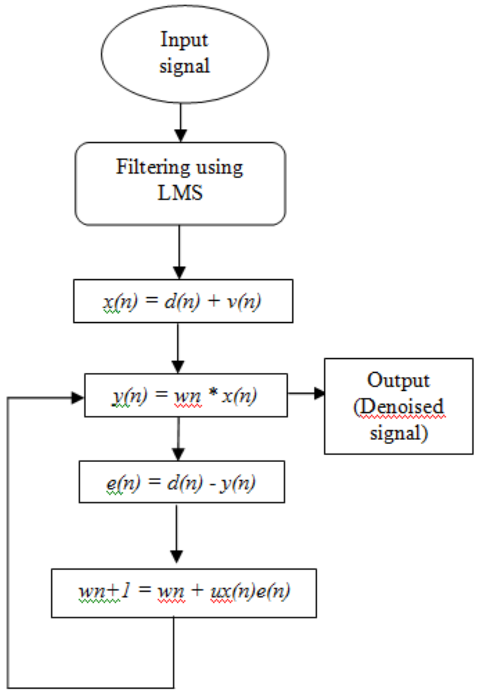


FIGURE 1: FLOWCHART OF DENOISING PROCESS BY LMS ALGORITHM

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D. Cellular Automata Algorithm

In the proposed work, the cellular automata rules are used to compute the rule weights so as to denoise the speech signals. As the computation using the cellular automata is faster as compared to the other techniques, less time is required for denoising. The filtering using cellular automata is applied as shown in the fig. 2 below

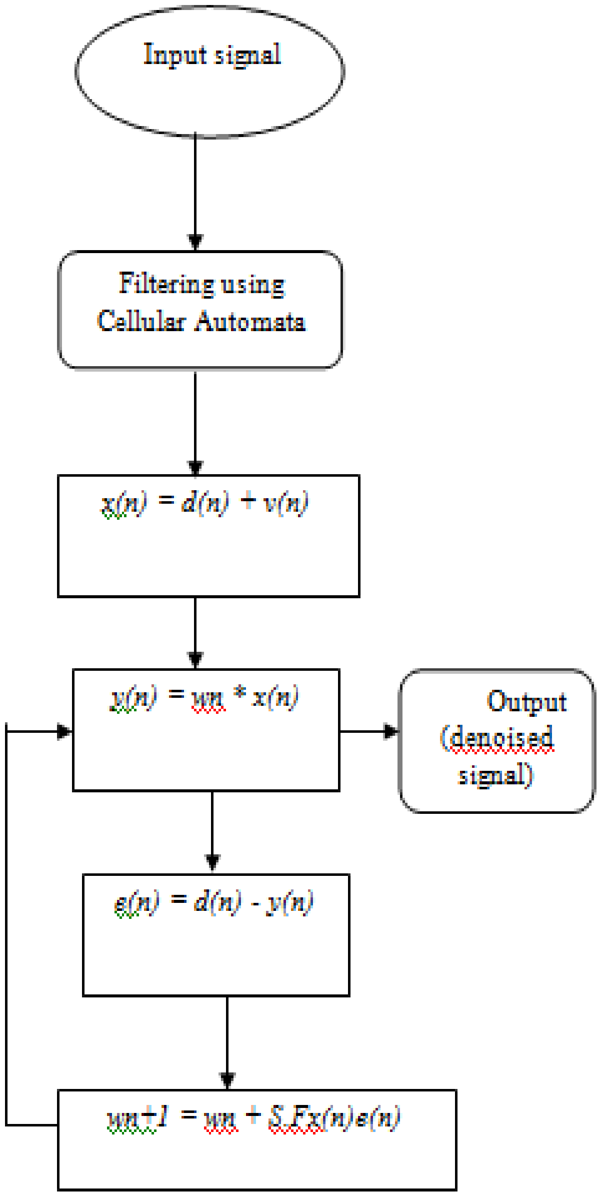

Figure 2: Flowchart of cellular automata Algorithm

E. Complete Denoising using Cellular Automata and LMS

The complete denoising of the speech signal is performed by combining the technique of least mean square adaptive filtering and cellular automata as shown in the fig. 3 to get the complete denoised signal . In this the fuse weight is calculated from the rule weight of the LMS algorithm and the rule weight of the cellular automata algorithm and then the complete denoised signal is obtained.

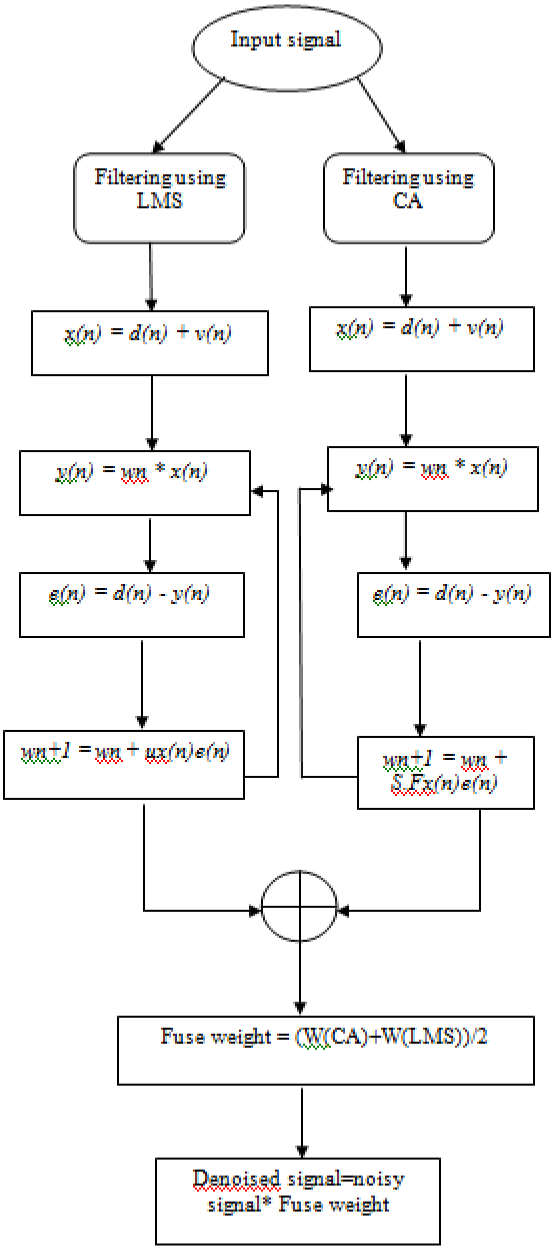

Figure 3: Flowchart of Hybrid Cellular automata approach

F. Compression using Cellular Automata

In our proposed work, the speech signal has been compressed using the cellular automat as it takes less time to compress the signal which is an additional feature of our proposed work.

Suppose if there are 1000 samples then the average has been calculated of first two samples, now this value is compared with the samples and it is replaced by that sample which gives the minimum difference. In this way for every two samples we are having one output sample. The computations are performed using the cellular automata rules. Thus the compression of the speech signal has been achieved using cellular automata.

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III. RESULT

The proposed work is implemented on Matlab tool. We have analyzed the proposed system at different signal to noise ratio (SNR). The result obtained at different signal to noise ratio are presented in this section. Fig. 4 shows denoised and compressed speech signal with SNR of 4dB. At SNR of 4 dB, PSNR of 26.0658 dB, MMSE of 0.0025 is found at the output of proposed system. The time required to complete denoising is 0.0151 s. After denoising, compression of same speech signal is carried. Percentage compression achieved by proposed system is 49.9975% and time required to do compression operation by cellular automata is found to be 0.56957425 s.

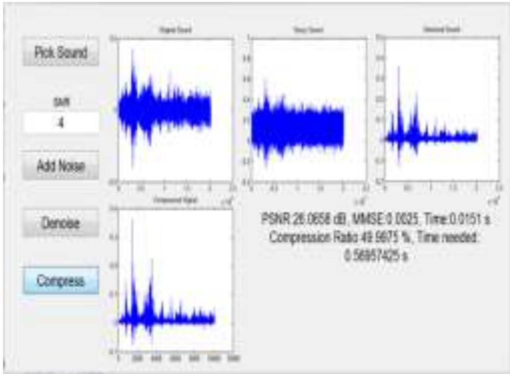

Figure 4: Denoised and compressed output of proposed system at SNR of 4 dB

Fig. 5 shows denoised and compressed speech signal with SNR of 6. At SNR of 6 dB, PSNR of 26.8283 dB, MMSE of 0.0021 is found at the output of proposed system. The time required to complete denoising is 0.0266 s. After denoising, compression of same speech signal is carried. Percentage compression achieved by proposed system is 49.9975% and time required to do compression operation by cellular automata is found to be 0.57253733 s.

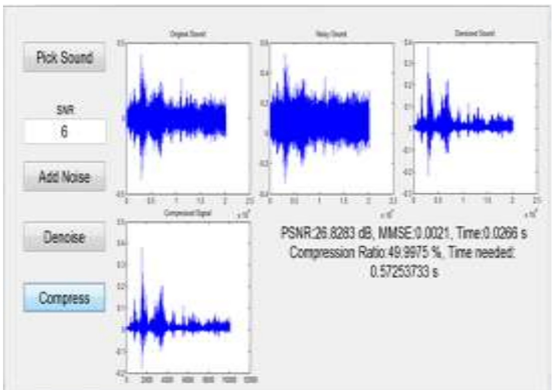

Figure 5 Denoised and compressed output of proposed system at SNR of 6 dB

Fig. 6 shows denoised and compressed speech signal with SNR of 8dB. At SNR of 8 dB, PSNR of 27.6388 dB, MMSE of 0.0017 is found at the output of proposed system. The time required to complete denoising is 0.0166 s. After denoising, compression of same speech signal is carried. Percentage compression achieved by proposed system is 49.9975% and time required to do compression operation by cellular automata is found to be 0.56360508 s.

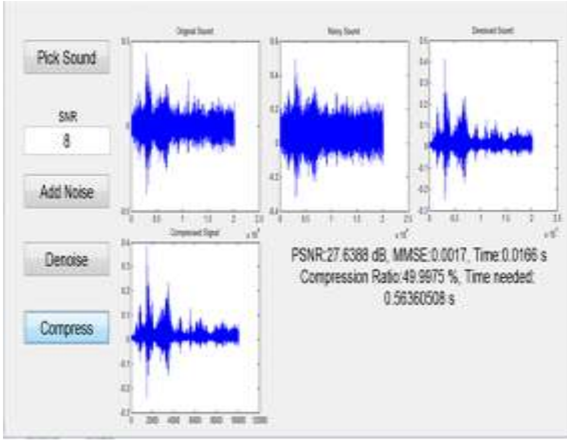

Figure 6: Denoised and compressed output of proposed system at SNR of 8 dB

Fig. 7 shows denoised and compressed speech signal with SNR of 10. At SNR of 10 dB, PSNR of 28.4554 dB, MMSE of 0.0014 is found at the output of proposed system. The time required to complete denoising is 0.0263 s. After denoising, compression of same speech signal is carried. Percentage compression achieved by proposed system is 49.9975% and time required to do compression operation by cellular automata is found to be 0.56417150 s.

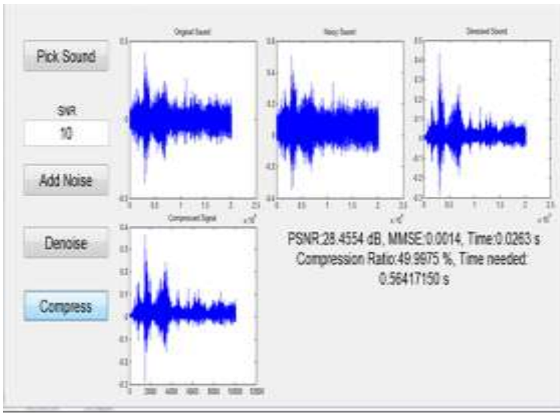

Figure 7 Denoised and compressed output of proposed system at SNR of 10 dB

Table.2. Comparison of PSNR, MMSE, Time needed and Compression at different SNR

Sr No.	SNR (dB)	PSNR (dB)	MMSE	Compression (%)	Time Needed (seconds)
1	4	26.0658	0.0025	49.9975	0.5695743
2	6	26.8283	0.0021	49.9975	0.5725373
3	8	27.6388	0.0017	49.9975	0.5636051
4	10	28.4554	0.0014	49.9975	0.5641715

IV. CONCLUSION

The purpose of this work is to combine two different denoising techniques based on Cellular Automata and least mean square to obtain best result and noise free speech signal. In this work, different methods of denoising and compression are discussed. It is found that estimation of rules is fast in cellular automata as compared to least mean square algorithm. At different signal to noise ratio (SNR), generated outputs are presented in this work and it is find that delay gets reduced and compression of the input speech is also achieved. With signal to noise ratio, peak signal to noise ratio also varied. It is concluded that denoising of noisy speech signal and compression of same input signal can be done by proposed method.

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