



Achieving Effective Noise Reduction by Using SSBLFNRA Algorithm

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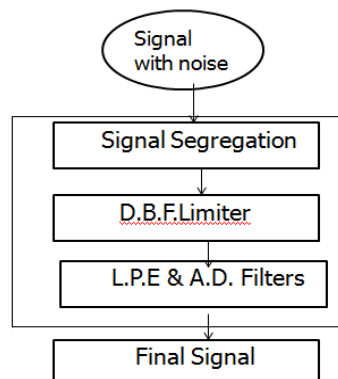
Abstract: In wireless sensor networks there are one major issue, that is channel noise. Now a day's Noise is the major problem while working with wireless sensor Networks. In this algorithm introducing new noise detection mechanism based on RF energy duration. If noise is detected in this go through SSBLFNRA (Source Segregation through Bandwidth frequency domain limiter and Linear Filters for Noise Reduction Algorithm) to reduce the detected noise. The idea of this is to examine how different aspects of sound—noise, speech privacy, speech intelligibility, and music—impact patient and staff outcomes in healthcare settings and the specific environmental design strategies that can be used to improve the acoustical environment of healthcare settings. The term "noise" is commonly used to describe a phenomenon of interference or disturbance that occurs within both wireless sensor and wired communication systems. In practical scenarios, noise is almost unavoidable and can never be eliminated completely from a wireless sensor communication channel. Also, there are multiple types of noises originating from different sources known to be contributing in making the process of wireless sensor communication difficult and highly unreliable. Experimental Results and discussions noise can be reduced by applying SSBLFNRA, in this reduce that noise effectively than previous methods. This algorithm is enhancement of DUET and SAFIA. We presented first an algorithm for estimating the mixing matrix, which can be seen as an extension of the DUET and the SAFIA algorithms but requires less stringent condition them. We then confirmed the validity of the algorithm by performing several simulations and by comparing these results with those obtained using standard clustering algorithms. After the mixing matrix was estimated, we were also able to recover the source matrix using a standard linear programming algorithm. This method improved the SNR.

Keywords:

I. Introduction SSBLFNRA:

Noise is any unwanted energy. Noise can actually affect the entire communications system, but it is in the channel where it can do the most damage since this is where the signal is weakest.

SSBLFNRA segregates target Data by using the characteristic that a suitable frequency resolution concentrates the Data signal point this spectrum on specific frequency components. In this review SSBLFNRA briefly below the original SSBLFNRA wireless sensor uses both inter channel amplitude difference and inter channel phase difference. This ever, to reduce computational load, in this use only inter channel amplitude difference. Assume that destination is router which is indicated by the IP address in this algorithm; describe the application of SSBLFNRA networks to reduce noise in wireless sensor networks. Since the noise step is very high at the time of broadcasting data .There are two requirements, in this need to improve the SNR under conditions where the input signals SNR is negative and keep the speech and data (in terms of signals) quality .To meet these requirements, in this made three improvements to SSBLFNRA networks, In this added a frequency domain limiter, a PE filters and AD filters.



II. Various steps to reduce the noise:

- a. Signal Segregation
- b. Dynamic Bandwidth frequency domain limiter
- c. Prediction and additive filters

a. Signal Segregation

The purpose of this is to monitor how different situations of sound- noise, speech intelligibility and speech-privacy patient and staff outcomes in healthcare setting Keywords are used to search for articles included noise, speech intelligibility, and speech and information security. The term "noise" is commonly used for describe a event of interference that occurs within both wired and wireless sensor communication systems.

This interference and disturbance is usually described by the fluctuation that occurs within wireless sensor communication signals due to many external and internal sources associated with or surrounding the wireless sensor communication systems and media. In practical scenarios, noise is almost unavoidable and can never be eliminated completely from a wireless sensor communication channel. Also, there are multiple types of noises originating from different sources known to be contributing in making the process of wireless sensor communication difficult and highly unreliable. A few common sources of noise have been identified.

Short -Time Fourier Transform (STFT) is

$$y = \frac{1}{\sqrt{2\pi}} \int_{-a}^b xh(t) e^{-j\omega\tau} d\tau$$

General signal equation:

The noise ratio would be calculated by the standard formula

The complex ratio of $A(\omega, i) = 20 \log_{10} \frac{S1(\omega, i)}{S2(\omega, i)}$

Here s1, s2 are the signals delivered by the sender, in this consider the some particular number of signals, through this in this can estimate a finite value of the noise, let us consider for the two signals

$$S1 = \sum_{k=0}^n a1mSm(t) \quad , \quad S2 = \sum_{k=0}^n a2mSm(t)$$

Here the matter of reducing the noise beyond that main aim of the SS is segregating the signal. Hence here it can be possible with following Basic case: two mixtures of two sources 2.1. Preliminary approach: temporal analysis

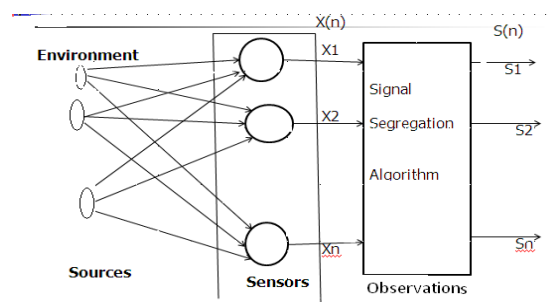


Fig: Representation of signal segregation

The separation is called blind to emphasize that the source signals are not observed and that the mixture is a black box to the observer.

b. Dynamic Bandwidth frequency Domain limiter

The bandwidth of a signal refers to the width of the band (or range) of frequencies occupied by a particular signal. Bandwidth is very important in designing communication systems since it dictates the type of channel suitable for the transmission of a particular signal.

If the signal is composed of several sinusoidal signal components, then

$$BW = \text{highest frequency} - \text{lowest frequency}$$

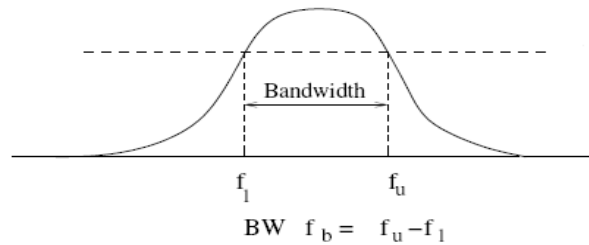


Figure 1: Bandwidth of a signal

In this develop an efficient bandwidth allocation algorithm that takes explicitly into account traffic statistics to increase the users' benefit and the network revenue simultaneously. In the past 10 years, these systems have slowly replaced many of the sound-processing devices used in part-time and professional configurations. Now days, with the considerable increase in the computing on the desktop, the audio community is an important change away from these systems, This is required for specialized hardware, towards general purpose desktop computing systems. The signals can be processed within these domains and each process in one domain has a corollary in the other Signals can be represented in many different ways.

Implementation

The SS object stores time-domain signals as buffers of samples upon which the SS analysis is done. For the purpose of discussion, the examples given in this algorithm make use of buffers of 1024 samples. Unlike time-domain signals, a frequency-domain signal is represented by a succession of spectral "frames". Like frames in a movie, the frames of SS and PEF data represent a "snapshot" of a brief segment of an audio signal. A frame consists of a certain number of equally spaced frequency bands called "bins". The number of bins is equal to the size of the SS buffer, thus the frames of data have 1024 bins. Each bin describes the energy in a specific part of the audio signal's frequency range. Bin-by-bin, using three sample streams running at the sampling rate. Thus, each bin is represented by three samples consisting of "real" and "imaginary" values, and the bin number (index). Note that unlike a classic frequency converter, the FFT-based filter can only shift frequencies by multiples of an FFT bin width. The longer the FFT, the finer the step width for this kind of frequency conversion.

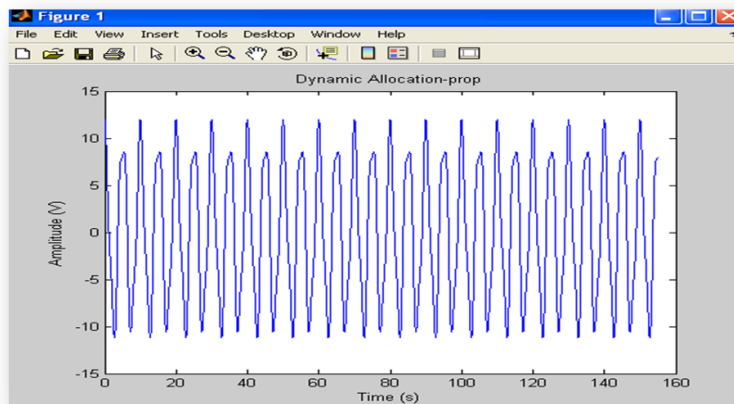


Fig: Dynamic Bandwidth Frequency Limiter

C. Linear Prediction Error & Adaptive

Digital Filters: PEF stands for prediction error filter; ADF stands for adaptive digital filter. Prediction is a technique for signal source modeling, it leads in speech signal processing and having wide appliance in other areas. Preliminary with an expression of the relationship between this linear prediction and the common difference equation for linear prediction systems, the component shows the process of formulating linear prediction equations and solving. The component discusses the use of linear prediction for modeling, the resource of a signal and signal spectrum. When you have worked through this component, you should be conscious of how linear prediction can able to afford a model of a signal source to state the operational limitations of the modeling in qualitative conditions How these equations are solved on a computer system and when to use linear prediction as an alternative for spectral analysis with the Fourier transform Simulation.

The intention of linear prediction is to form a model of a linear time-invariant digital system during examination of input and output sequences. in particular linear prediction calculates a set of coefficients and these provides an a prediction - for a informative output. Sample $y[n]$ given knowledge of previous input $(x[])$,previous output $(y[])$ samples:

$$y'[n] = \sum_{k=0}^p a_k x[n-k] - \sum_{k=1}^q b_k y[n-k]$$

Here a, b are called predictor coefficients. The common form of linear prediction used in signal processing. This is one in which the coefficients are zero, hence that the output estimate is made completely based on the previous output samples:

$$y'[n] = -\sum_{k=1}^q b_k y[n-k]$$

This model is called an all-pole model; it may be seen by analogy with the frequency response equations of LTI systems. The use of an all pole model is motivated by the following reasons:

- (i) In this frequently do not have access to the input sequence,
- (ii) Many easy signal sources are closely modeled by an all-pole model,
- (iii) This model gives the equations of system which may be capably solved.

The relation of the linear predictor equation with the LTI system equation may be more willingly understood by introducing a predictor error signal $e[n]$ the difference between the real output and the prediction: of the prediction and adaptive digital filter is achieve the noiseless information from the wireless sensor networks. Here the signals which re collected by the process of SS, in this the segregate sources will be support in order of predicting error filter. When the noise signal travel through the prediction error filter the noise may trapped here with the huge techniques of error filters. Linear prediction is a technique for signal source modeling dominant in speech signal processing and having wide application in other areas. Starting with a demonstration of the relationship between this linear prediction and the general difference equation for linear systems, the unit shows how the linear prediction equations are formulated and solved. The unit then discusses the use of linear prediction for modeling the source of a signal and the signal spectrum.

Comparison:

Properties	DUET	SAFIA	SSBLFNRA
Signal segregation	Low	Medium	High
Bandwidth limitation	No	Not sufficient	sufficient
Filtering	No	Musical signals	All Data signals
Reduction capacity	Low	Medium	high
Objective results	Partially	Partially	Fully
After filter Noise Ratio in given signal	29.096Db	20.42356Db	0.335998Db

III. Conclusions

Finally in this algorithm noise can be reduced by applying SSBLFNRA, in this reduce that noise effectively than previous methods. These enhance the algorithms called DUET and SAFIA in formula1 race car. We presented first an algorithm for estimating the mixing matrix, which can be seen as an extension of the DUET and the SAFIA algorithms but requires less stringent condition them. We then confirmed the validity of the algorithm by performing several simulations and by comparing these results with those obtained using standard clustering algorithms. After the mixing matrix was estimated, we were also able to recover the source matrix using a standard linear programming algorithm. This method improved the SNR.

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