RESTORATION OF PULSE NOISE DETERIORATED AUDIO SIGNALS

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Audio signals are subject to pulse noise in a wide range of applications - from scratched LP records to mobile systems. Various software packages for pulse noise cancellation have been developed up to now. They generally perform well over seriously damaged audio signals. But due to the complex algorithms they utilize the processing speed is low. This paper presents an examination of a fast and efficient method for pulse noise cancellation. It is based on using median filters to remove the pulses. Applying median filtration to the whole signal leads to attenuation of its high-frequency components in the undamaged sections. This may be avoided by adaptively implementing filtration only where pulse noise is detected.

Keywords - *adaptive filtering, audio signal, median filter, pulse noise, pulse detection, signal restoration.*

I. Introduction

Audio signals are subject to pulse noise in a wide range of applications. Whether it is a scratch noise of an old LP record, or propagation effects in mobile systems, the deterioration in the audio signal may be severe [1,2]. This calls for an efficient method for signal restoration as close as possible to its original shape.

II. Existing Software

In search of finding efficient ways to signal restoration, different restoration techniques and algorithms have been developed.

One possible approach to pulse noise cancellation is to develop a noise profile for the processed signal through spectral analysis of the noise and the signal. This technique is suitable for background noise cancellation. The pulse noise is registered during the spectral analysis and consequently removed. Although the processing gives good results, the high computational complexity slows down processing and that is why it is worth using it for audio signals with comparatively low signal-to-noise ratio (SNR).

Another way to tackle the problem is through graphical processing of the deteriorated signal. If the signal is "unfolded", the damaged section can easily be spotted and erased. The original signal is restored through suitable graphical interpolation in the gap left after the erasure. This algorithm is, however, suitable for removing single scratches.

It is possible to cancel pulse noise by its detection and filtration. For example a fast compressor may react to high-amplitude peaks and suppress them. The efficiency of this approach is, however, low, since it sometimes distorts signals with similar form - e.g. produced by drums.

More complex processing algorithms may also be used. The algorithms may be optimized for certain features of the damaged signal - e.g. very poor SNR. Such techniques often affect and actually damage some good sections of the signal. Thus, it is required that an additional examination of the processed signal be performed. Then the unnecessary changes can be cancelled.

To sum up, the different approaches mentioned here do remove pulse noise quite efficiently, but at the expense of sophisticated algorithms and low processing speed.

III. Specific Research

A. General Considerations

Our research uses a proposed adaptive method, based on linear prediction and conformed filtering [3]. It provides pulse noise cancellation and a considerable improvement in signal quality.

The method has "borrowed" the median filter from image processing [4,5]. The filter is generally described by the following equation:

$$m_n = med\left\{x_{n-k},\ldots,x_n,\ldots,x_{n+k}\right\} \tag{1}$$

where m_n is the median value of the signal x_n in a window of length N=2k+1 samples.

When applied to the whole signal, the median filter has smoothing effect in noiseless regions, which leads to loss of high-frequency details [6]. A better solution would be to apply filtering only to the sections where pulse noise is detected. The detection is carried out by means of a high-pass filter approximating the second derivative of the signal, described by:

$$x_n = D^2\left\{x_n\right\} = x_{n-1} - 2x_n + x_{n+1} \tag{2}$$

A strobe can hence be obtained from the wrapping of the high-pass filter response. Thus, the median pulse cancellation filter will be active only in damaged sections of the signal. For strobe generation a recursive median filter is preferred, since it can suppress a sequence of scratches close to each other. Such a filter is defined by:

$$r_n = med\left\{r_{n-k},\ldots,r_{n-1},x_{n-k},\ldots,x_n,\ldots,x_{n+k}\right\} \tag{3}$$

where r_n is the output of the recursive median filter.

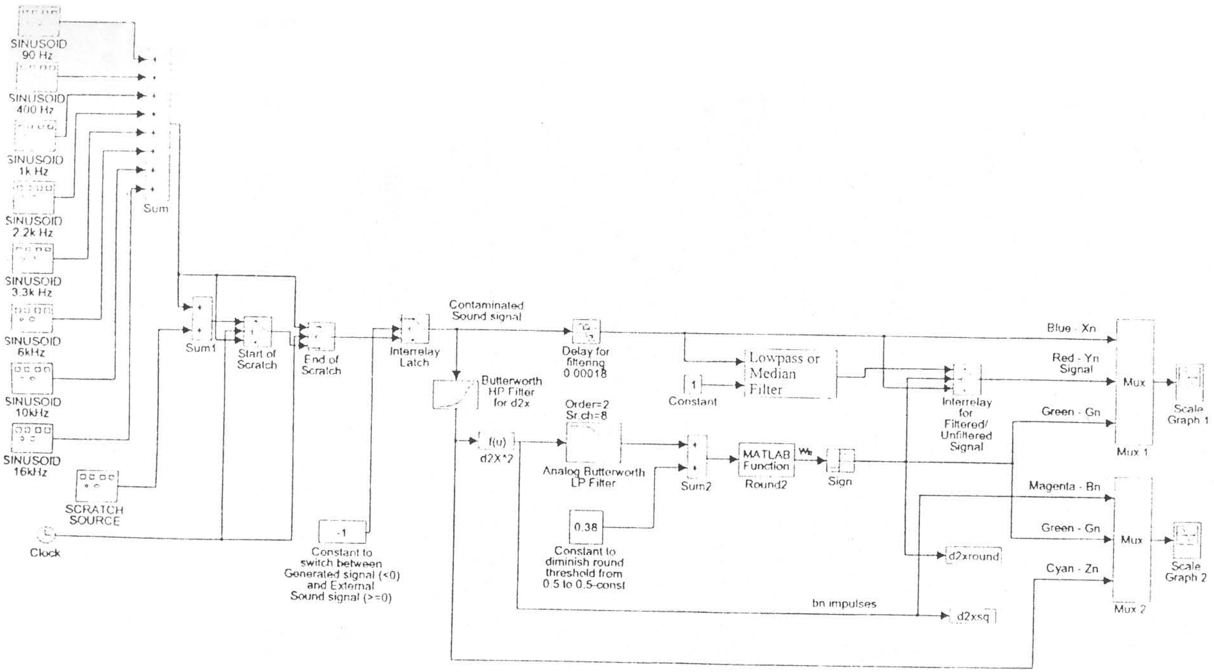


Fig.1. Block diagram of the filtering algorithm.

B. Experiments

In order to compare the efficiencies of low-pass and median filtering in pulse noise cancellation, a software simulation utilizing **MatLab Simulink** was prepared. The block diagram of the suggested filtering mechanism is shown in Fig. 1.

A number of **sinusoid** generators were used for audio signal simulation. A summation of those sinusoids allows for complicated signal shapes to be modelled. The pulse noise is generated by a **scratch source** and added in to the audio signal. The initial and final scratch moments are controlled by means of switches **Start of scratch** and **End of scratch**. The **Interrelay Latch** enables the input of real damaged audio signal from an outside source. The switching is carried out by the **Constant to switch** block.

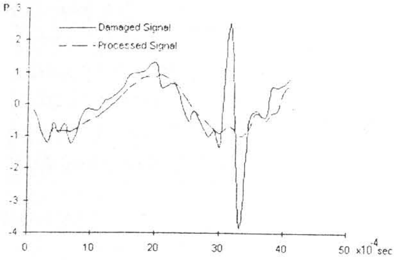
The signal is then applied to a combination of blocks, carrying out the noise cancellation. The **Interrelay for Filtered/Unfiltered Signal** block switches between filtered and unfiltered signal on the basis of strobe pulses. An artificial delay is introduced in the signal by the **Delay for filtering** block , to allow for a possible delay during strobe generation. The contaminated signal x_n, the filtered signal y_n and the strobe signal are visualized on a **Scale graph 1** block.

The pulse noise, as mentioned above, is detected through its second derivative by a **Butterworth HP filter for d2x** block. Thus the presence of pulse noise is emphasized due to its high frequency components. The obtained signal z_n is then squared in the **d2x^2** block for the wrapping b_n of z_n to be determined. Afterwards the wrapping is smoothed in the **Analogue Butterworth LP Filter**, which actually approximates a recursive median filter. The signal is then rounded in the **Round2** block, but the round threshold can be controlled by means of **Constant to diminish** block. The correction constant is in this case chosen to be 0.38, which leads to a threshold of 0.12. The threshold can be adjusted to obtain best performance of the algorithm. The signal is finally processed in **Sign** block, which transforms the pulse into a standard pulse with an amplitude of +1. The signals second derivative z_n its wrapping b_n and the final strobe pulse g_n are shown by means of **Scale graph 2**.

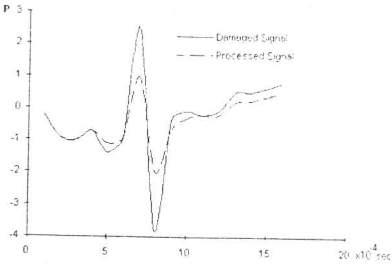
IV. Experimental results

The performance of the median filter is greatly affected by its windowing size.(L). If the windowing size is large, median filter's behaviour is close to that of a low-pass filter and vice versa, as shown in fig. 2a, resp. 2b.

Two series of experiments were carried out - with low-pass and median filter (fig. 3 and 4 respectively). The input signal is made up of a mix of frequencies to modell real-life applications. Fig. 3a and 4a depict the system performance under contamination with single pulse, whereas 2b and 3b show the response to series of pulses.

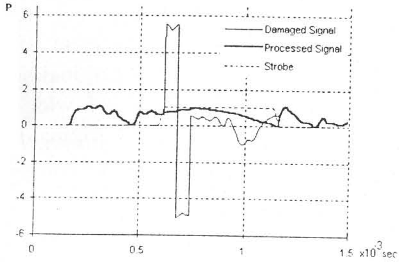


a) L=170

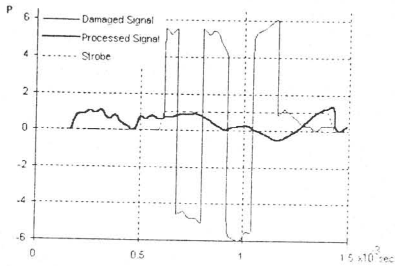


b) L=30

Fig. 2 Median smoothing algorithm

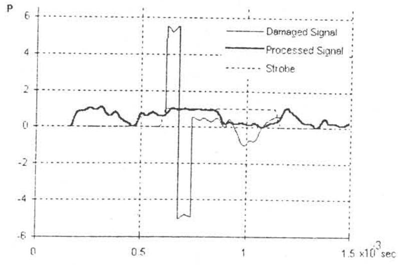


a) Frequency mix contaminated with single pulse

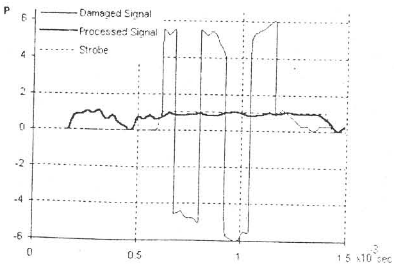


b) Frequency mix contaminated with multiple pulses

Fig. 3. Pulse noise cancellation implementing low-pass filter



a) Frequency mix contaminated with single pulse



b) Frequency mix contaminated with multiple pulses

Fig. 4. Pulse noise cancellation implementing median filter

As can be seen, the low-pass filter performs well enough for low frequency components under the influence of single pulses. The median filter obviously offers improvement as far as low-frequency components of the signal are concerned. The high peaks due to pulse noise are cancelled, while the trend of the original signal is preserved to a considerable extent. Moreover, its behaviour towards high-frequency components even under multiple pulse contamination is much better than that of the low-pass filter.

V. Conclusion

The importance of the windowing size of the median filter should be mentioned. The filter seems to perform poorly when the window is large in size, as its behaviour is close to that of a low-pass filter. The experiments have shown the best results can be achieved with as narrow window as possible, thus preserving the reference points next to the damaged area.

The results show a low-pass filter is suitable for narrow-band signals deteriorated with single scratches. It restores to a decent extent the original signal. Under more rigid conditions, such as long sequences of pulses and wide-band signals, he median filter performs considerably better. It preserves the original shape and should be the filter of choice if high quality noise cancellation is required.

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