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Design and Analysis of Sub-band Coding of Speech Signal under Noisy Condition using Multirate Signal Processing

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Abstract - *Multirate systems are most commonly used in Digital Signal Processing applications. Systems that use more than one sampling rates in the digital signal processing are called multirate DSP. The objective is to design sub band coding system. Sub band coding is a method where the speech signal is sub divided into several frequency bands and each band is digitally encoded separately. The input speech signal spectrum is divided into frequency sub bands using a bank of finite impulse response (FIR) filter. In this project low pass and high pass finite impulse response filters are designed and implemented using different windowing techniques such as Hamming, Hanning, Blackman, Rectangular and Kaiser and the responses of the FIR filters are obtained. Finally performances are evaluated based on Signal to Noise Ratio. Comparing the results of different windows it is observed that the Blackman window provides maximum Signal to Noise Ratio values when compared to other windowing techniques. This shows that performance of the Blackman window is better and it is suitable for deigning the FIR filters.*

Key Words: FIR Filter, Signal to Noise Ratio, Multirate, Windowing.

1.INTRODUCTION

Multirate signal processing is the processing of signals at more than one sampling rate by the discrete time systems. Various structures in digital audio signal processing frequently function at different sampling rates. The connection of such systems requires a conversion of sampling rate. The process of converting a signal from one sampling rate into another sampling rate is called sampling rate conversion. There are two sampling rate conversion techniques in multirate digital signal processing which are decimation and interpolation. The process of decimation and interpolation are done by using up sampler and down sampler. Decimation is the process of reducing the sampling rate by a factor and Interpolation is the process of increasing the sampling rate by a factor. The combined operation of filtering and down sampling is called Decimation. Up sampling is the manner of placing zero valued samples among original samples to increase the sampling rate. Multirate techniques are used in the acquisition of high quality data, spectral analysis with high resolution, design and implementation of narrow band digital filtering and sub-band coding of speech signal.

Window techniques and filters are widely used in speech processing and digital signal processing. A mathematical function that is zero-valued outside the selected interval is considered as the window function. Windowing technique is very popular, effective method because this technique is simple, convenient, fast and easy to understand. A filter is used to remove unwanted signal and allows only the desired signal. Different types of filters are available based on the applications. Low Pass Filter (LPF) allows only low frequency band and attenuates high frequency band whereas High Pass Filter (HPF) allows only high frequency band and attenuates low frequency band. A filter which has finite impulse response is considered as Finite Impulse Response (FIR) filters. During the implementation of FIR filters, it has no feedback and it is also called as non-recursive filter.

2. LITERATURE SURVEY

Ashraf M. Aziz (2009) proposed a structure of two channel quadrature mirror filter with the low pass filter, high pass filter, decimators and interpolators to perform sub-band coding of speech signals in digital domain. The proposed structure decomposes a signal into low frequency and high frequency components. The results show that the proposed structure reduces the error and achieves considerable performance improvement compared to delta-modulation encoding systems. The limitation of this paper is higher computational complexity.Maurya A.K. and Dr.Deepak Nagaria (2011) presented decimation and interpolation techniques of multirate signal processing which are rate conversion techniques. The advantage is interpolation can change the sampling rate of the signal without changing its original content. The disadvantage of this paper is decimation without filtering can cause degradation in the signal due to aliasing.

Saurabh Singh Rajput and Dr.S.S. Bhadauria (2012) developed low pass finite impulse response filter using an efficient adjustable window function based on Blackman window function. Signal comparison between original speech and the filtered speech signal shows that the high frequency component of the speech signal has significantly removed by using this low-pass FIR filter. The disadvantage of this paper is that Blackman window provides higher side lobe attenuation comparison to Hamming window and the width of the main-lobe is slightly greater than Hamming window function. Vijayakumar Majjagi (2013) used rate conversion techniques of multirate signal processing to design and implement a sub-band coding system of an optimum four channel Quadrature Mirror Filter bank. The reconstructed signal is compared with the original speech signal. The main drawback is higher computational complexity.

Dolly Agrawal and Divya Kumud (2014) presented the multirate signal processing techniques such as decimation and interpolation and then demonstrated how the two processes can be combined to obtain sampling rate conversion by any rational factor. The limitation of this paper is that the effects of aliasing for decimation and pseudo images for interpolation are created while designing the multirate systems. Prajoy Podder, Tanvir Zaman Khan, Mamdudul Haque Khan and M.Muktadia Rahman (2014) proposed windowing techniques for the comparison of performance of Hamming, Hanning and Blackman window based on their magnitude response, phase response and equivalent noise bandwidth in time and frequency domain using matlab simulation.. Comparing simulation results of different window, Blackman window has best performance among them and the response of the Blackman window are more smooth and perfect when compared to Hamming and Hanning windows. The main drawback is that the Blackman window has higher equivalent noise bandwidth.

Jagriti Saini and Rajesh Mehra (2015) presented comparative analysis of speech signal using different windowing techniques such as Hamming, Hanning and Blackman window. It can be obtained from the simulated results that Blackman window contains almost double power as compared to Hamming and Hanning window. So for long distance communication Blackman window is used. The major disadvantage of this paper is that the losses occur while travel long distance. Suresh Babu, Dr.D.Srinivasulu Reddy and Dr.P.V.N.Reddy (2015) used windowing techniques and the performance of Hamming, Hanning and Blackman windows are mainly compared based on their magnitude response, phase response for designing the FIR low pass and high pass filters using matlab. Comparing the simulation results using different windows, it is observed that Blackman window produces better results among them.

Lalima Singh (2015) developed a new speech signal analysis technique based on Fast Fourier Transform (FFT) and Linear Predictive Coding (LPC). These techniques are used for extracting and compressing some features of speech signal for further processing. The main limitation of this paper is that the spectrum analysis is complex process of decomposing the speech signal into similar parts. Lalitha R. Naik and Devaraja Naik R L (2015) presented a very low rate speech coder based on sub-band coding method. This paper mainly concentrating the comparison of correlation values for different clean speech signals and correlation values for after adding high amplitude noise to the same speech signals. Taking correlation tests prove that its performance is satisfying. The disadvantage is that the result won't be same as the original signal.

3. METHODOLOGY

The below figure show block diagram of the sub band coding system. The first frequency subdivision splits the input speech signal into two equal width segments, a low pass signal and a high pass signal. The second subdivision splits the low pass signal from the first stage into two equal bands, a low pass signal and high pass signal. Finally, third frequency subdivision splits the low pass signal from the second stage into two equal bandwidth signals. Thus the signal is subdivided into four frequency bands. Decimation by a factor of '2' is performed after the frequency subdivision. By allocating different number of bits per sample to the signal in the four sub bands, reduction in bit rate of the digitized speech signal can be achieved.

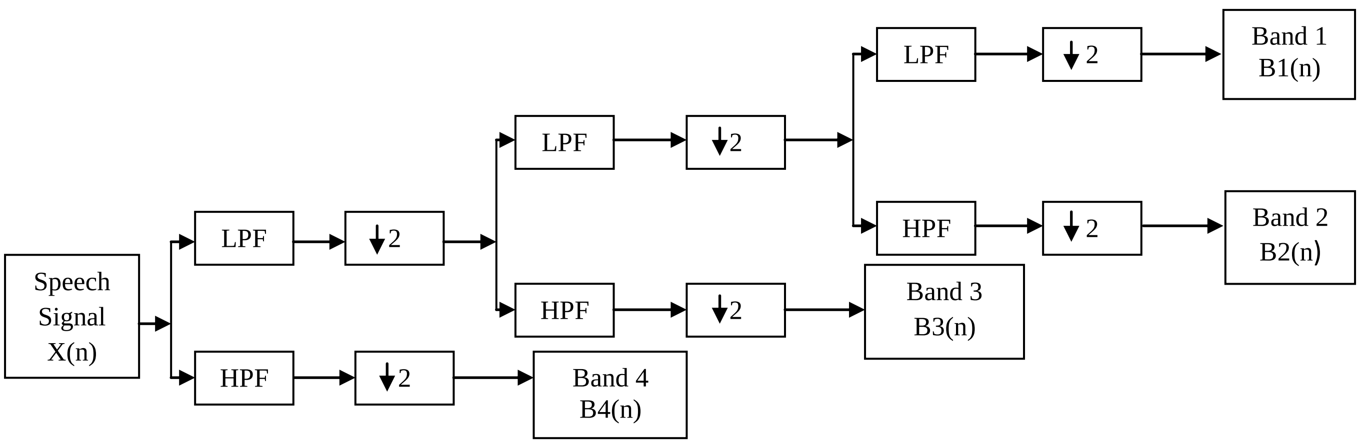

Fig - 1: Sub-band coding of speech signal – Frequency Subdivision

3.1 FIR Filter

Finite Impulse Response (FIR) filter is a filter whose response to any finite length input is of finite duration because it settles to zero in finite time. Finite Impulse Response digital filter has exact linear phase, highly stable relatively easy to design, computationally intensive, arbitrary amplitude-frequency characteristic, less sensitive to finite word-length effects and real-time stable signal processing requirements etc. Thus it is widely used in different digital signal processing applications. FIR filter is described by differential equation. The output signal is a convolution of an input signal and the impulse response of the filter. The output is given by

$$y(n) = \sum_{k=0}^{N-1} h(k)\ x(n-k) \qquad \text{(Eqn.1)}$$

Where,

$x(n)$ is the input signal.

$h(n)$ is the impulse response of FIR filter.

The transfer function of a causal FIR filter is obtained by taking the z-transform of impulse response of FIR filter $h(n)$.

$$H(z) = \sum_{k=0}^{N-1} h(k)\ z^{-k} \qquad \text{(Eqn.2)}$$

3.1.1 Low Pass Filter

Low pass filter allows signal below the cut off frequency and attenuates signal above the cut off frequency. The impulse response of LPF is given by

$$h_d(n) = \begin{cases} \frac{\sin w_c n}{n\pi} & ; \quad n \neq 0 \\ \frac{w_c}{\pi} & ; \quad n = 0 \end{cases} \qquad \text{(Eqn.3)}$$

3.1.2 High Pass Filter

High pass filter attenuates signal below the cut off frequency and allows signal above the cut off frequency. The impulse response of HPF is given by the equation

$$h_d(n) = \begin{cases} \frac{-\sin w_c n}{n\pi} & ; \quad n \neq 0 \\ 1 - \frac{w_c}{\pi} & ; \quad n = 0 \end{cases} \qquad \text{(Eqn.4)}$$

3.2 Windowing Techniques

Windowing methods are used for the design of Finite Impulse Response (FIR) filter, in particular to convert an ideal impulse response of infinite duration to a finite impulse response (FIR) filter design. In digital signal processing a window function also known as an apodization function or tapering function is a mathematical function that is zero-valued outside of some chosen interval. When another function or waveform/data-sequence is multiplied by a window function, the product is also zero-valued outside the interval. The main aim of window function is to provide accurate type of the responses with reduced side lobes and comparatively less pass-band and stop-band ripples. The proposed method concerns with the design and implementation of FIR filter using different window method. The following are the types of windows used in the proposed method.

3.2.1 Rectangular Window

A function which is constant inside the interval and zero elsewhere is called rectangular window which describes the shape of its graphical representation. The rectangular window is defined as

$$w(n) = \begin{cases} 1 & ; \quad 0 \leq n \leq N-1 \\ 0 & ; \quad otherwise \end{cases} \qquad \text{(Eqn.5)}$$

Where,

N is the order of the filter.

3.2.2 Hanning Window

The Hanning window is a raised cosine window and can be used to reduce the side lobes while preserving a good frequency resolution compared to the rectangular window. The Hanning window is defined as

$$w(n) = \begin{cases} 0.5 - 0.5\cos\left(\frac{2\pi n}{N-1}\right); & 0 \leq n \leq N-1 \\ 0 \qquad ; & otherwise \end{cases} \qquad \text{(Eqn.6)}$$

3.2.3 Hamming Window

The hamming window is like the Hanning window and also a raised cosine window. The hamming window exhibits similar characteristics to the Hanning window but further suppress the first side lobe. The hamming window is defined as

$$w(n) = \begin{cases} 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right) ; & 0 \leq n \leq N-1 \\ 0 \qquad ; & othertwise \end{cases} \qquad \text{(Eqn.7)}$$

3.2.4 Blackman Window

The Blackman window is similar to Hanning and Hamming windows. An advantage with Blackman window over other windows is that it has better stop band attenuation and with less pass band ripple. The Blackman window is defined as

$$w(n) = \begin{cases} 0.42 - 0.5\cos\left(\frac{2n\pi}{N-1}\right) + 0.08\cos\left(\frac{4n\pi}{N-1}\right); 0 \le n \le N-1 \\ 0 \quad ;otherwise \end{cases} \quad \text{(Eqn.8)}$$

3.2.5 Kaiser Window

The Kaiser window with parameter is defined as

$$w(n) = \begin{cases} \dfrac{I_0\left[\alpha\sqrt{1-\left(\frac{2n}{N-1}-1\right)^2}\right]}{I_0(\alpha)} \quad ; \quad 0 \le n \le N-1 \\ 0 \quad ; \quad otherwise \end{cases} \quad \text{(Eqn.9)}$$

Where,

α is the adjustable parameter which determines the shape of the window and thus controls the trade-off between main-lobe width and side-lobe amplitude.

$I_0(\alpha)$ is the modified zeroth-order Bessel function of first kind.

3.3 Performance Measures

3.3.1 Signal- to- Noise Ratio

Signal-to-noise ratio (SNR) is a measure used for comparing the level of a desired signal to the level of background noise. It is defined as the ratio of signal power to the noise power, often expressed in decibels.

$$SNR = 10\log\frac{\sum_{n=0}^{M-1} S_i^2(n)}{\sum_{n=0}^{M-1} N_i^2(n)} \quad \text{(Eqn.10)}$$

Where,

$S_i(n)$ is the input signal.

$N_i(n)$ is the noise signal.

4. RESULTS AND DISCUSSION

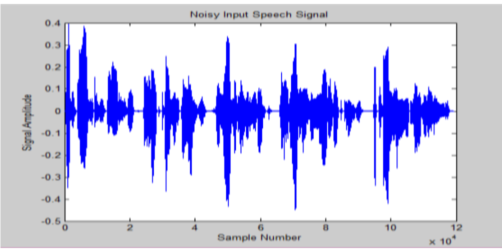

Fig – 2: Noisy Input Speech Signal

The voice sample of the male is taken for analysis of the speech signal from the TIMIT (Texas Instruments and Massachusetts Institute of Technology) data base. TIMIT data base was compiled using 630 speakers and each speaker spoke ten utterances giving a total of 6300 sentences.

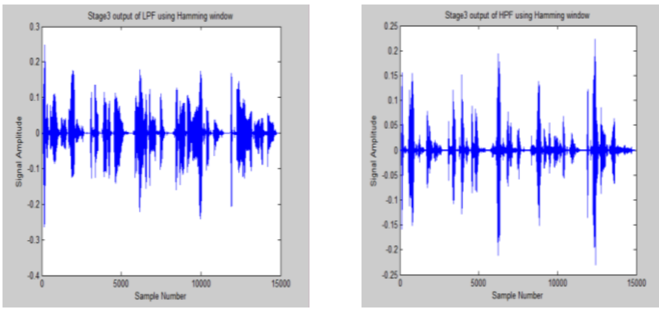

Fig – 3: Band 1 and Band 2 output of Hamming Window

The above figures show the band1 output signal contains most of the information when compared to band2 signal. Since most of the speech signals are present in the lower frequency bands, band2 contains less information than band1 signal.

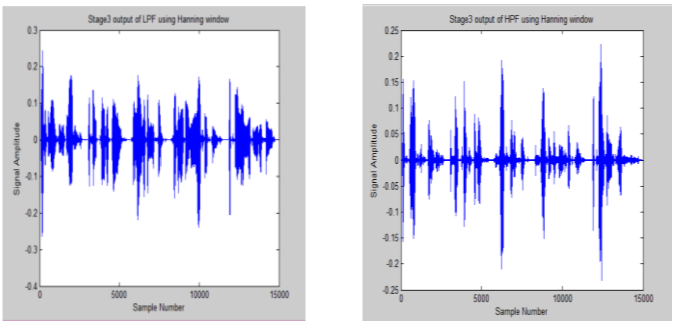

Fig – 4: Band 1 and Band 2 output of Hanning Window

The above figures show that the most of the speech signals are present in the lower frequency bands when compared to higher frequency bands.

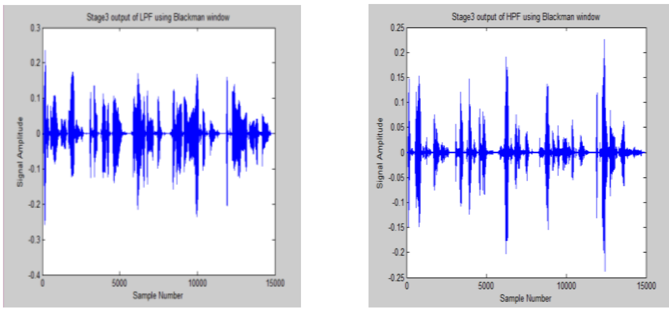

Fig – 5: Band 1 and Band 2 output of Blackman Window

The above figures show that the most of the speech signals are present in the lower frequency bands when compared to higher frequency bands.

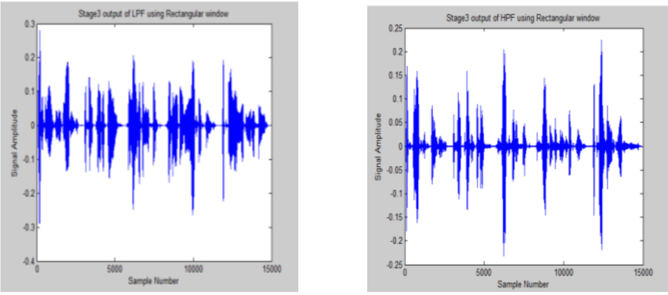

Fig – 6: Band 1 and Band 2 output of Rectangular Window

The above figures show that the most of the speech signals are present in the lower frequency bands when compared to higher frequency bands.

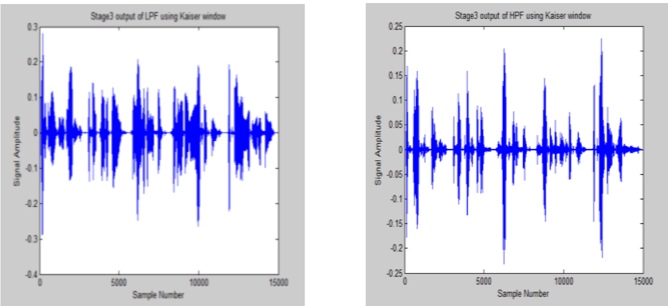

Fig – 7: Band 1 and Band 2 output of Kaiser Window

The above figures show that the most of the speech signals are present in the lower frequency bands when compared to higher frequency bands.

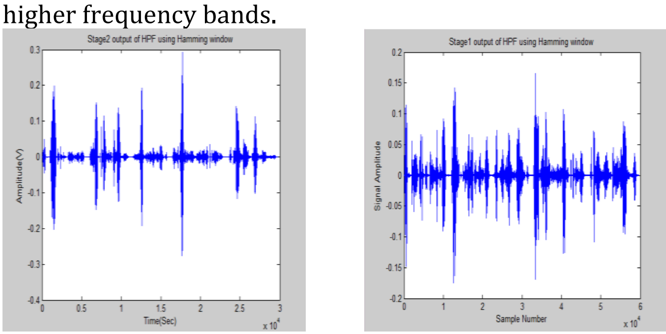

Fig – 8: Band 3 and Band 4 output of Hamming Window

The above figures show the band 3 and band 4 contains less information when compared to band 1. So most of the speech energy is contained in the lower frequency band than the higher frequency band.

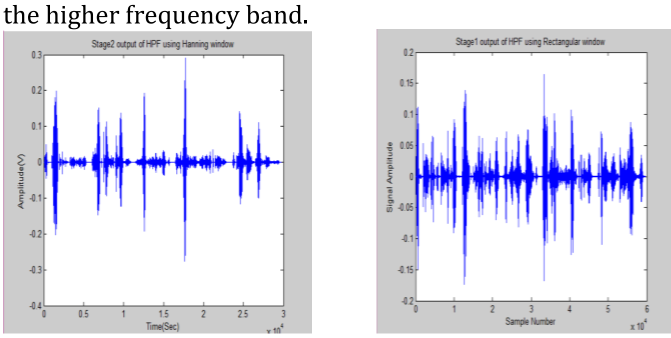

Fig – 9: Band 3 and Band 4 output of Hanning Window

The above figures show the band 3 and band 4 contains less information when compared to band 1. So most of the speech energy is contained in the lower frequency band than the higher frequency band.

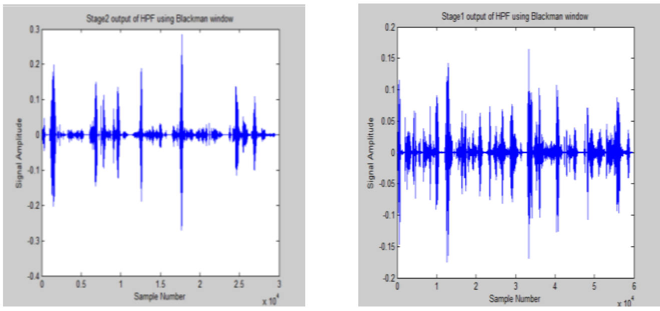

Fig – 10: Band 3 and Band 4 output of Blackman Window

The above figures show the band 3 and band 4 contains less information when compared to band 1. So most of the speech energy is contained in the lower frequency band than the higher frequency band.

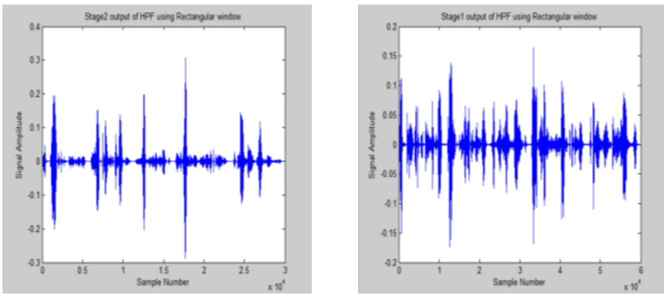

Fig – 11: Band 3 and Band 4 output of Rectangular Window

The above figures show the band 3 and band 4 contains less information when compared to band 1. So most of the speech energy is contained in the lower frequency band than the higher frequency band.

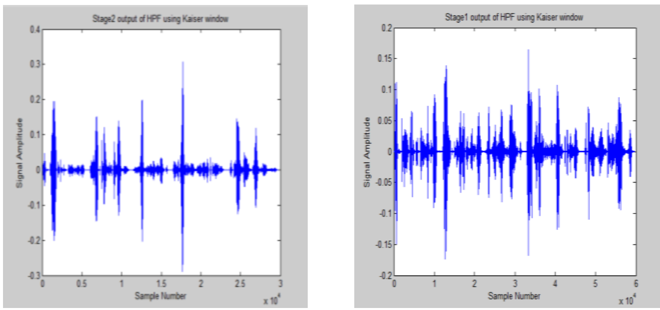

Fig – 12: Band 3 and Band 4 output of Kaiser Window

The above figure show the band 3 and band 4 contains less information when compared to band 1. So most of the speech energy is contained in the lower frequency band than the higher frequency band.

Table-1: Comparison of SNR (dB) for different windowing Methods at band 1 and band 2

Windowing Methods	Band 1	Band 2
Hamming	12.0196	15.4032
Hanning	12.0638	15.4452
Blackman	12.1626	15.6137
Rectangular	11.4783	14.8236
Kaiser	11.5023	14.8469

Table-2: Comparison of SNR (dB) for different windowing Methods at band 3 and band 4

Windowing Methods	Band 3	Band 4
Hamming	14.6716	14.2775
Hanning	14.6662	14.3096
Blackman	14.7829	14.4682
Rectangular	14.7123	13.8619
Kaiser	14.7092	13.8782

From the Table 1 and 2 it is observed that the Blackman window provides maximum Signal to Noise Ratio. So Blackman window is suitable for designing the finite impulse response filters.

5. CONCLUSION

Thus the sub band coding system is designed and the low pass and high pass finite impulse response filters are designed and implemented using different windowing methods such as Hamming, Hanning, Blackman, Rectangular and Kaiser. To evaluate the performance of the different window methods Signal to Noise Ratio values are measured. From the performance measures it is observed that the Blackman window provides maximum SNR values when compared to other windowing methods. So the performance of the Blackman window is better and it is suitable for designing finite impulse response filters.

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