Sub-band Coding of Speech Signals using Multirate Signal Processing

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1. THEORY

A variety of techniques have been developed to efficiently represent speech signals in digital form for either transmission or storage. Since most of the speech energy is contained in the lower frequencies, we would like to encode the lower-frequency band in more bits than the high-frequency band. Sub-band coding is a method where the speech signal is subdivided into several frequency bands and each band is digitally encoded separately.

An example of a frequency subdivision is shown in the Figure 1. Let us assume that the speech signal is sampled at a rate Fs samples per second. The first frequency subdivision splits the signal spectrum into two equal width segments, a lowpass signal ($0 < F < Fs/4$) and a highpass signal ($Fs/4 < F < Fs/2$). The second frequency subdivision splits the lowpass signal from the first stage into two equal bands, a lowpass signal ($0 < F < Fs/8$) and a highpass signal ($Fs/8 < F < Fs/4$). Finally, the third frequency subdivision splits the lowpass signal from the second stage into two equal bandwidth signals. Thus, the signal is subdivided into 4 frequency bands, covering 3 octaves, as shown in the figure.

Decimation by a factor of 2 is performed after frequency subdivision. By allocating a different number bits per samples to the signals in the 4 sub-band, we can achieve a reduction in the bitrate of the digitalized speech signal.

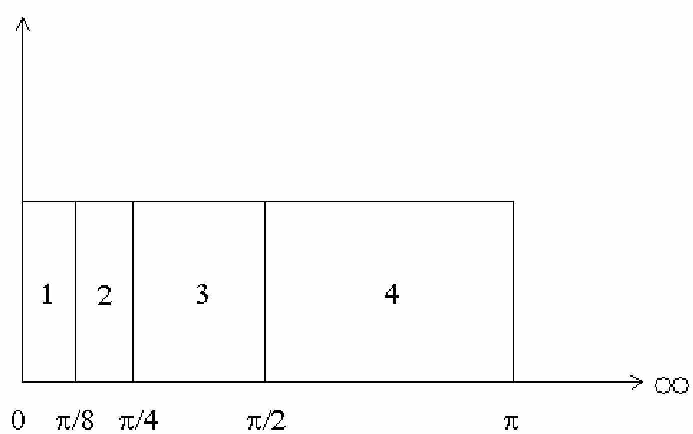

Fig. 1: Frequency Subdivision.

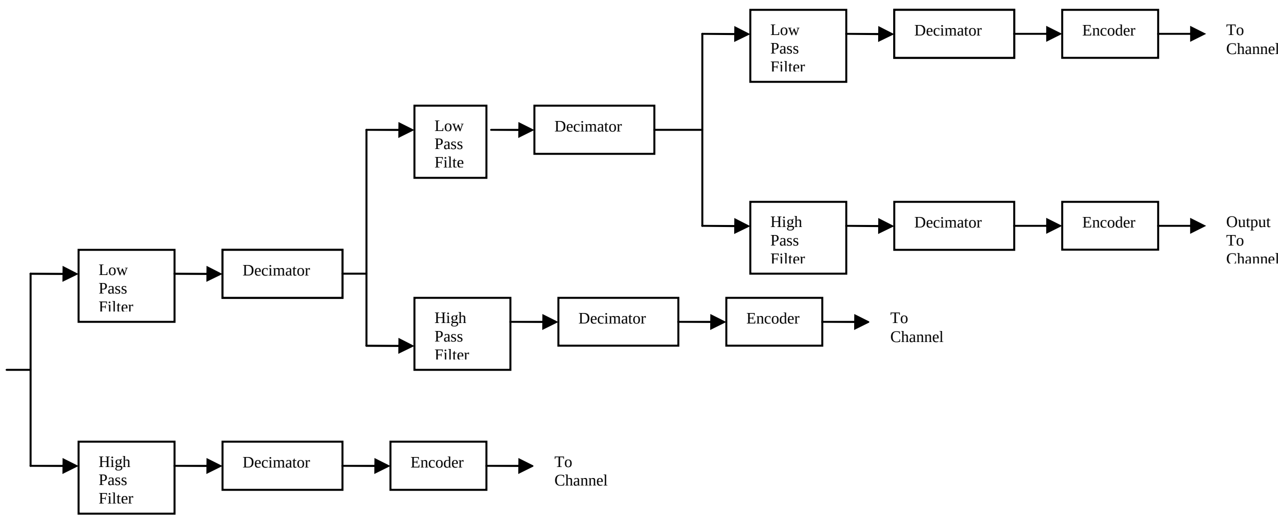

Fig 2a. Block Diagram: Subband Speech Coder

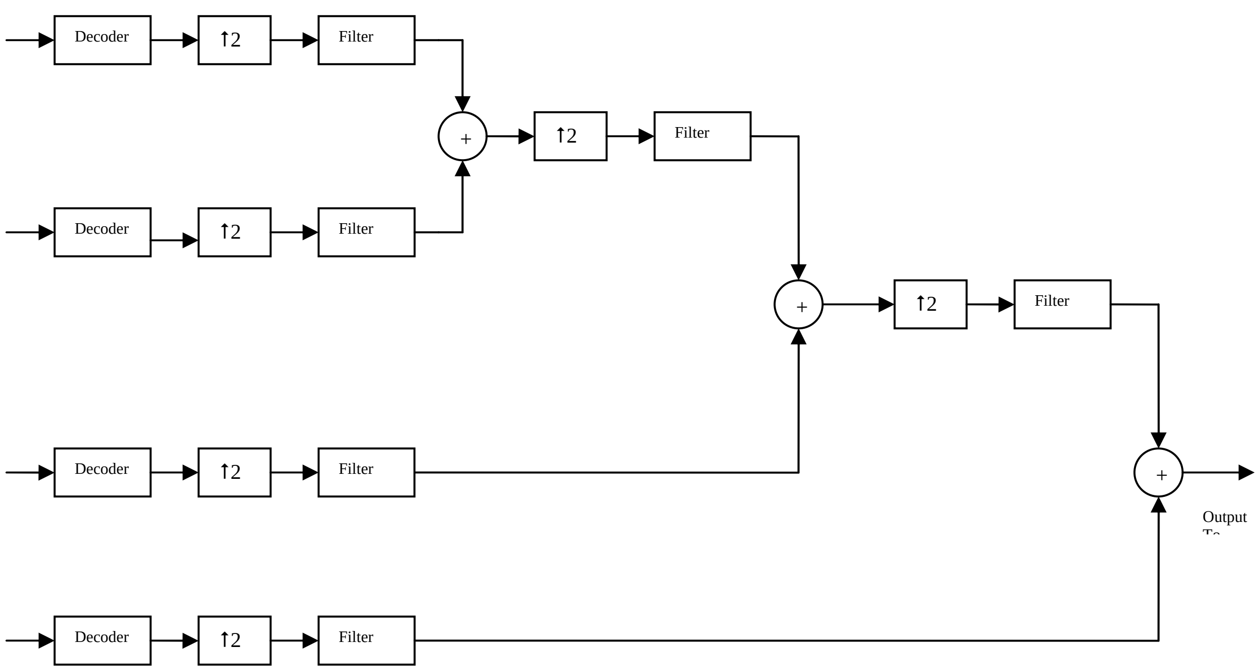

Fig. 2b: Synthesis of Subband Encoded Signal

2. **SIMULATION & RESULTS**

Using the MATLAB Siumlink model firstly a spoeeh signal was made as shown in the following Fig. 3.

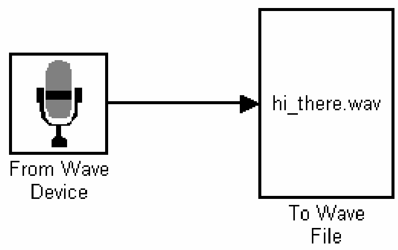

Fig. 3: Generation of Speech Signal.

The duration of the speech signal was about 2 second and the sampling frequency was 9600 Hz. The spectrum (0 to 4800 Hz) and the plot of the signal are as follows:

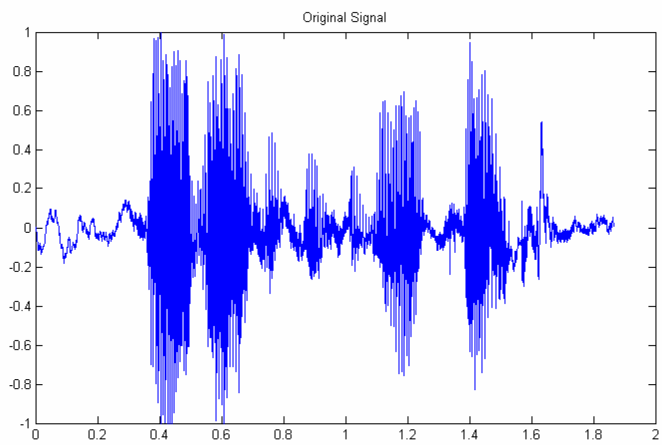

Fig 4a: Original Signal Plot.

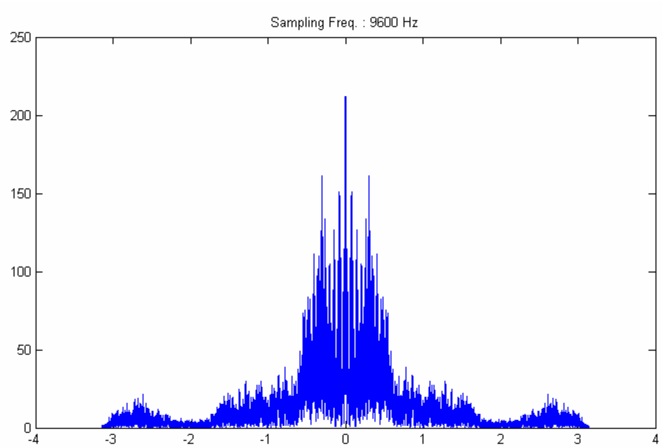

Fig. 4b: Original Signal Spectrum

Then a third order sub-band coder is used to get three outputs. The lowest octave (containing the information for the frequency ranges 0 to 1200 Hz) is coded using 16 bits, the second octave (containing the information for the frequency ranges 1200 to 2400 Hz) is encoded by using 8 bits and the final octave containig the least amount of data is encoded using 4 bits if data. The subband spech encoder thus made is as follows in Fig. 5a. The corresponding decoder is made and the speech signal is recovered.

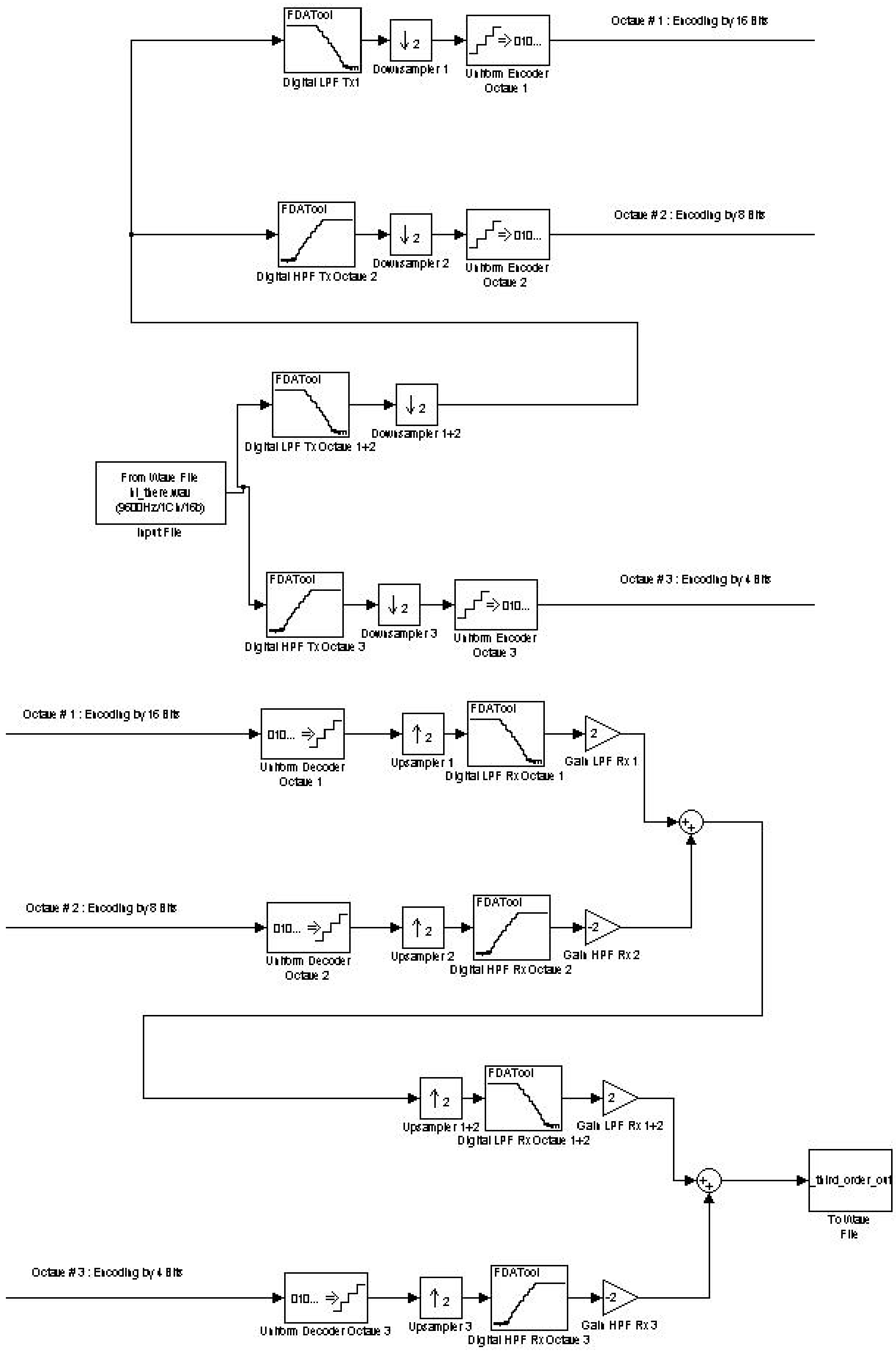

Fig. 5: The Subband Speech Coder Decoder made using MATLAB Simulink Model Editor

The following is the plot of the output speech that is recovered and the corresponding plot. To the ear, apart form the hum induced due to quantization effects and other things, the sound quality doesn't suffer much.

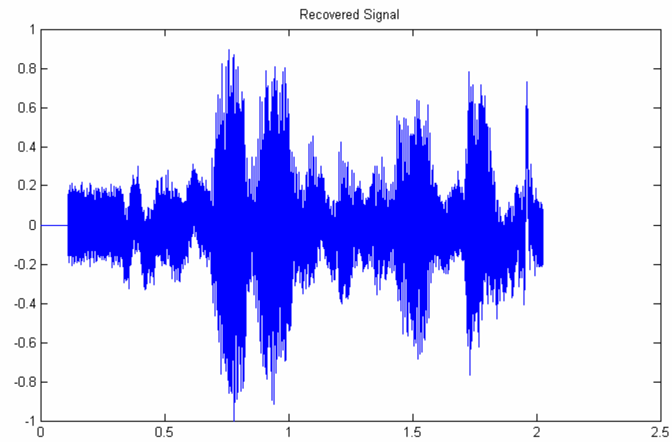

Fig. 6a: Recovered Signal Plot

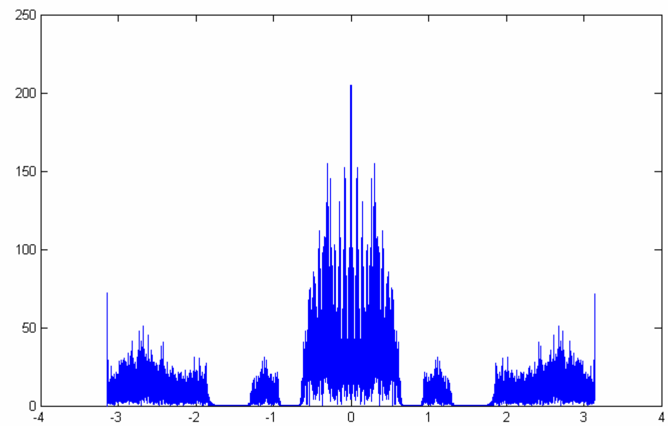

Fig. 6b: Recovered Signal Spectrum

3. **REFERENCES**

[1] John G. Proakis and Dimitris G. Manolakis, "*Digital Signal Processing: Principles, Algorithms and Applications*", Third Edition.