Subband Coding of Speech Signal Using Scilab

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Abstract

Subband coding decompose the input signal into different frequency bands .After the input is decomposed to its constituents, we can use the coding technique best suited to each constituent to improve the compression performance. Decompose a signal into components by applying frequency-selective filtering. Then select the best coding technique that best suits each component (subjectively and objectively). Subband coding applications are Speech coding ITU-T G.722 Encode high-quality speech at 64/56/48 kbps. Interest in signal processing long predates computers. As long as people have tried to send or receive information through electronic media, such as telegraphs, telephones, television, radar, etc., there has been the realization that these signals may be affected by the system used to acquire, transmit, or process them. Sometimes these systems are imperfect and introduce noise, distortion, or other artifacts. Understanding the effects these systems have and finding ways to correct them is the foundation of signal processing . This paper depicts analysis side of the system implemented using freeware language SCILAB. The synthesis side includes speech production The digital filters have been implemented for extraction feature .The signal processing application of SCILAB has become boon.

Keywords: Subband Coding, Speech ,Scilab ,Opensource,ITU-T G.722,LPF,HPF

1. Introduction

Sub-band coding (SBC) is a kind of transform coding [3, 4]. A signal is divided into a number of different frequency bands and encodes each one independently. It enables a data compression by discarding information about frequencies which are masked. The result differs from the original signal, but if the discarded information is chosen carefully, the difference will not be noticeable, or more importantly, The basic concept of ''Frequency Domain Coding of Speech'' methods is to divide the speech into frequency components by a filter bank (sub-band coding), or by a suitable transform (transform coding), and then encode them using adaptive PCM. Recent developments and examples of the ''Vocoder-driven'' adaptive transform coder for low bit-rate applications is also discussed [9].In ''Subband Coding of Speech Signals Using Decimation and Interpolation''—a structure of a two-channel quadrature mirror filter with low pass filter, high pass filter, decimators and interpolators, is proposed to perform subband coding of speech signals in the digital domain. The results show that the proposed structure significantly reduces the error and achieves considerable performance improvement compared to delta-modulation encoding systems [10]. Subband coder reduces and controls quantization noise.

Resampling is done to interface two systems with different sampling rates.
1. Decimation: It is a process to decrease the sampling rate.
2. Interpolation: It is a process to increase the sampling rate.
"Down sampling" is a process of removing some samples, without the low pass filtering. A signal is down sampled only when it is "oversampled"(i.e. sampling rate > Nyquist rate). This combined operation of filtering and down sampling is called "Decimation". To down sample by a factor of M, we must keep every Mth sample as it is and remove the (M-1) samples in between.
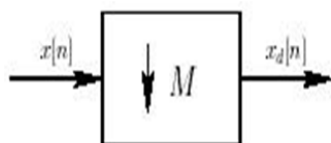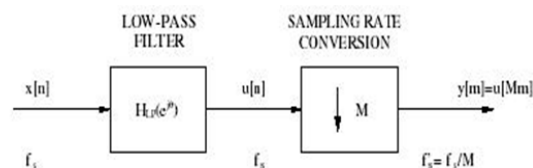

Fig 1.1: Symbol

Fig 1.2: Block diagram of a decimator

"Up sampling" is the process of inserting zero- valued samples between original samples to increase the sampling rate. (This is called "zero-stuffing"). Given a sequence x[n] , we can define

IPASJ International Journal of Electronics & Communication (IIJEC)

Web Site: http://www.ipasj.org/IIJEC/IIJEC.htm

A Publisher for Research Motivatin........

Volume 2, Issue 5, May 2014

Email: editoriijec@ipasj.org

ISSN 2321-5984

$$x_u[n] = \begin{cases} x[n/L], & n = 0, \pm L, \pm 2L, \ldots \\ 0, & \text{otherwise} \end{cases}$$

Where xu[n] is the sequence up-sampled from x[n] by a factor of L.This means that xu[n] is generated by padding (L-1) zeros between every sample of x[n].
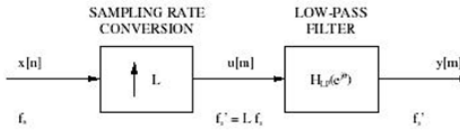

Fig 1.3: Symbol for up-sampler   Fig 1.4: Block diagram of an interpolator

"Interpolation" is the process of upsampling followed by filtering (to remove the undesired spectral images.) The result is a signal sampled at a higher rate. The interpolation factor (L) is the ratio of the output rate to the input rate
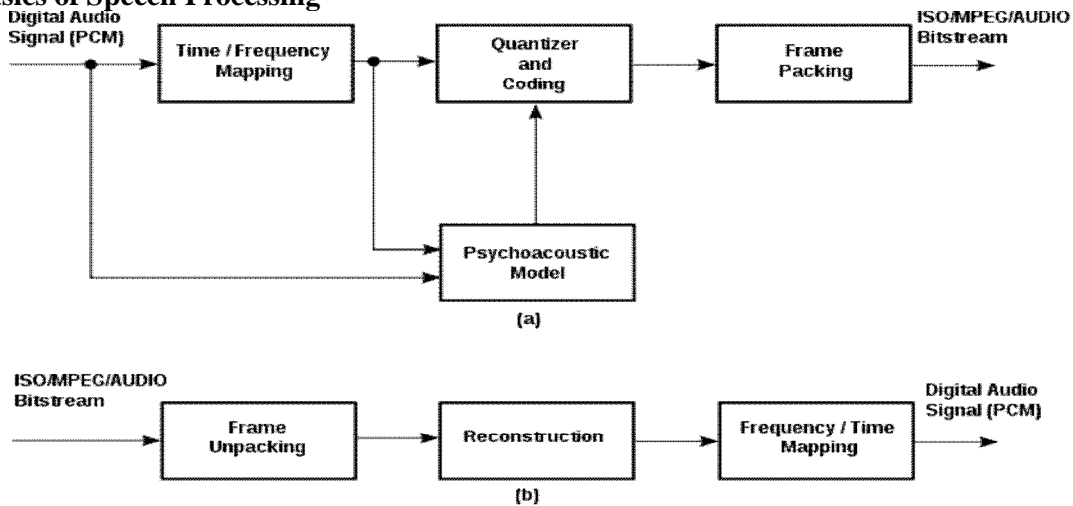
2. Basics of Speech Processing

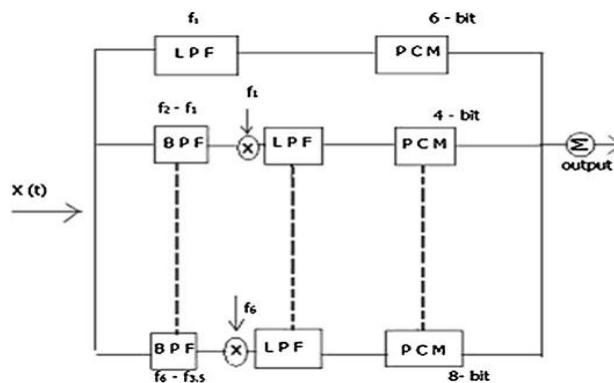Fig 1.4: Block diagram of Subband coding

Fig 1.5: Block diagram of Subband coding transmitter

Transmitter consists of one LPF and six BPFs. All BPFs outputs are multiplied by the lowest frequency component of those bands at the multiplier block. Then outputs are PCM and then added by summer. Finally the summed output is put into channel.At the receiver signals are decoded by seven decoders. Then each signal is passed through LPF of cut-off frequency f1, f2–f1, f3–f2 etc. From second to seventh signal outputs are multiplied by their respective lowest frequency components and then passed through BPFs of f2–f1, f3–f2 etc. Then the outputs are summed up to get the replica of the original signal. Data rate from the above signal is reduced from 64 to 19.5 Kbps [1].
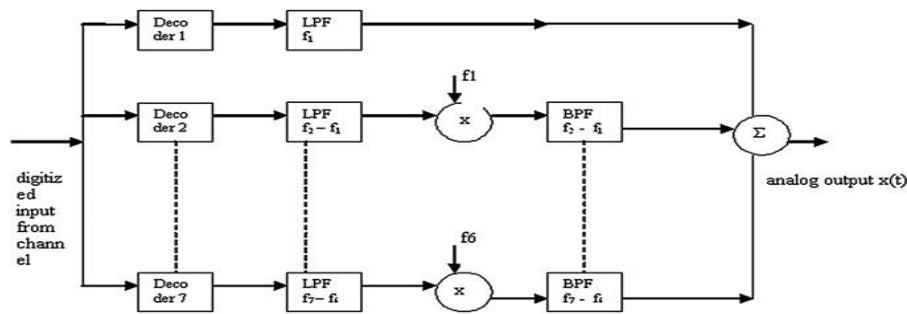
Fig 1.6: Block diagram of Subband coding Receiver

3.Implementation of Subband Coding Using Scilab Open Source Software:

Scilab is stated as scientific laboratory which basically software package for numerical computation for engineers and scientists. Scilab is endowed with powerful tools and easy syntax. Matrix being the basic fundamental object for calculation matrix manipulation can be easily handled. It is basically an interpreted language and is multiplatform so available on different OS such as Linux, Windows & MacOSX. The associated facilities in conjunction with open environment are helpful for signal processing, optimization & control It is easily compatible with Matlab so m-files can be easily used as it has Matlab to Scilab translator tool.

Source output ⬚⬚⬚⬚⬚⬚⬚⊗ analysis filter bank ⬚⬚⬚⬚⬚⬚⬚⊗sub-sampled ⬚⬚⬚⬚⬚⊗encoded.

The source output is passed through a bank of filters.This filter bank covers the range of frequencies that make up the source output.The passband of each filter specifies each set of frequencies that can pass through The outputs of the filters are subsampled thus reducing the number of samples The justification for the subsampling is the Nyquist rule and its extension justifies this downsampling

Scilab Simulation Used for Analysis, Scilab codes for Subband coding is given Below

```
[x,Fs,bits]=wavread("/home/aiktc/Documents/beep-01a.wav");
n = length(x)
//Low Pass
of Length = 19, Wc = 0.5, Hamming Window
[wft_LPF,wfm_LPF,fr_LPF]=wfir('lp',17,[0.25,0],'hm',[0,0])
//High Pass Filter of Length = 19, Wc = 0.5, Hamming Window
[wft_HPF,wfm_HPF,fr_HPF]=wfir('hp',17,[0.25,0],'hm',[0,0])
//LPF output
Y_lpf = convol(x,wft_LPF)
//HPF output
Y_hpf = convol(x,wft_HPF)
//Downsampling by a factor of 2
Downsampling_XLPF = Y_lpf(1:2:length(Y_lpf));
Downsampling_XHPF = Y_hpf(1:2:length(Y_hpf));
figure(1)
subplot(3,1,1)
plot([1:n],x)
xtitle("ORIGINAL Speech SIGNAL");
subplot(3,1,2)
plot([1:length(Y_lpf)],Y_lpf)
xtitle("Low Pass Filtered Speech SIGNAL");
```

```
subplot(3,1,3)
plot([1:length(Y_hpf)],Y_hpf)
xtitle("High Pass Filtered Speech SIGNAL");
figure(2)
subplot(2,1,1)
plot([1:length(Y_lpf)],Y_lpf)
xtitle("Low Pass Filtered Speech SIGNAL");
subplot(2,1,2)
plot([1:length(Downsampling_XLPF)],Downsampling_XLPF,'r')
xtitle("LPF Output Downsampled by 2");
figure(3)
subplot(2,1,1)
plot([1:length(Y_hpf)],Y_hpf)
xtitle("High Pass Filtered Speech SIGNAL");
subplot(2,1,2)
plot([1:length(Downsampling_XHPF)],Downsampling_XHPF,'r')
xtitle("HPF Output Downsampled by 2");
```

The intention of this work is to design and implement a SUBBAND CODING system. We have successfully designed an optimum low pass filter for minimize the amplitude distortion. From this Low pass filter we have designed a High pass using these filters we have successfully simulated a Subband of Speech Signal. The result shows that the output is a perfect reconstruction of the input speech signal.

4. Result & Conclusions

We have successfully implemented the sub-band coding system by designing . The frequency response characteristics of LPF and HPF used. The speech signal on which sub-band coding is to be performed is given as an input For this we recorded a speech signal using the tool sound recorder i.e available in the windows with the following specifications
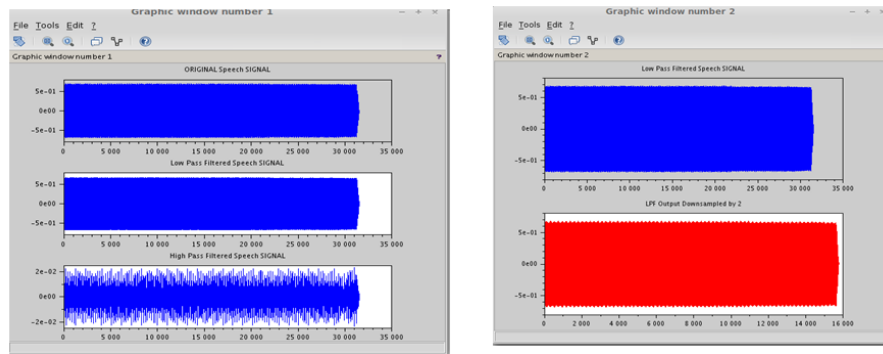

Fig 1.7 DownSampled By 2 For Low Pass Signal   Fig1.6:Speech Signal (Original,Lowpass, and High Pass Signal)
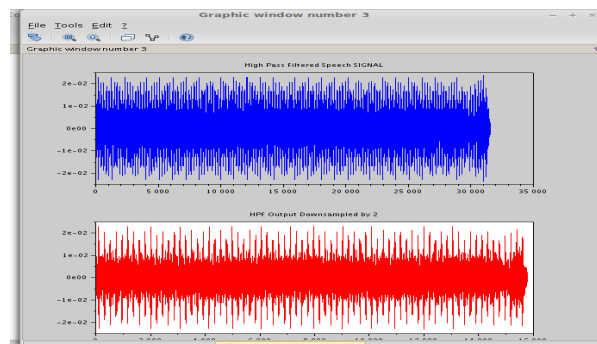

Fig 1.8 DownSampled By 2 For High Pass Signal

To enable higher quality compression, one may use subband coding. First, a digital filter bank divides the input signal spectrum into some number (e.g., 32) of subbands. The psychoacoustic model looks at the energy in each of these subbands, as well as in the original signal, and computes masking thresholds using psychoacoustic information. Each of the subband samples is quantized and encoded so as to keep the quantization noise below the dynamically computed masking threshold. The final step is to format all these quantized samples into groups of data called frames, to facilitate eventual playback by a decoder.Decoding is much easier than encoding, since no psychoacoustic model is involved. The frames are unpacked, subband samples are decoded, and a frequency-time mapping reconstructs an output audio signal. subband decomposition of a signal offer a time-frequency representation of the signal Because both representations are rearrangements of each other,they are equivalent in terms of their information content or representation of the signal.

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