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Non-linear Down-sampling and Signal Reconstruction, Without Folding

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***Abstract*—This paper presents a theory of 1.5 factor non-linear down-sampling, reconstruction and noise elimination. For linear down sampling of two or three factor, one sample is taken and next one or two samples are not taken/discarded. Here in non-linear down sampling two or three samples are taken and the next one is not taken. The purpose of this non-linear down sampling is to send less data samples in voice communication. Though one sample is discarded after taking two samples value of this sample can be reconstructed from values of other samples. Here, two samples are at original sampling period, T_s interval and next two samples are at $2T_s$ interval. High-frequency sharp changes were extracted when sampled at T_s interval. From received signal, discarded sample can be reconstructed from nearby four samples (Previous two and next two). When original signal contains higher frequency some error signal is introduced, after reconstruction. This error signal depends on original signal. Error signal is eliminated using original signal. Down-sampling is performed after sampling and signal reconstruction is performed just before hearing the sound.**

***Index Terms*—PCM, LPC, Down-sampling, Interpolation, Frequency folding and Noise Elimination.**

I. Introduction

In cellular communication PCM or DPCM is used. Before sampling an analog low-pass filter is used to avoid folded signals of higher frequencies [1]. DPCM is better technique [2] because here less data is needed to be transferred. In speech compression Linear Predictive Coding (LPC) is used. LPC is based on AR signal modeling. LPC is the basis of speech compression for cell phones, digital answering machines. LPC reduces the transmitted data by factor of more than twelve [3]. LPC is a lossy compression scheme. LPC is specifically tailored for speech. It does not work well for audio in general. However if anyone wants to use LPC compression technique he can use he can use it in this compressed signal; LPC signal reconstruction is also needed before this signal reconstruction. Anyone can also use DPCM technique in this compressed signal. In adaptive rate sampling technique using LEVEL CROSSING SAMPLING SCHEME (LCSS) [4] bit-rate varies with input and time. If we want to use LCSS technique in mobile communication network will face difficulties to manage variable bit-rate and the technique is not always compressing data. Non-uniform sampling methods can be efficient for image compression [5], not for voice communication. In voice communication a real-time system should be considered. Here it is not possible to send double data in one second and to disconnect system for next one second as network is busy, like data transfer through internet. In silence detection algorithm for speech detection [6] speech and silence are separated by efficient coding. In this algorithm a frame is declared as silence or speech. Here compression ratio is not always constant but loss of data is small.

Using the proposed compression technique 33.3% bandwidth can be saved or number of mobile phone user can be increased by 50%. The proposed technique can also be used for internet voice communication, because LPC technique [7] can be applied on it.

Sampling rate can be reduced to reduce bit-rate of data transfer, but speech quality will degrade in reducing sampling rate. In mobile communication voice signal is sampled at rate of 8000Hz per second. For this frequencies up to 4000Hz can be manipulated using this sampling scheme. If sampling rate is reduced to 5334Hz (2/3 of 8K Hz), frequencies up to 2667Hz can be manipulated. In the proposed compression method signals up to 3700Hz can be manipulated, when frame contains no frequencies carrying major power and smaller than 1000Hz. The proposed technique can manipulate signals up to 3000Hz when signal contains no frequencies, carrying major power and smaller than 300Hz. When any frame contains both signals of high and low frequencies, a hi-pass filter should be used before using this compression system.

Processors can be created for the proposed operation so that samples can be retrieved very quickly. Value gotten from interpolation is mainly depends on values of previous two samples and past two samples. When samples are 8 bit PCM, 32 bit input are available for 8 bit output in reconstructing discarded samples. Fast digital circuit can be created for this input and outputs using K-map and sub-circuit minimization [8].

The proposed paper also gives a theory of 1.5 factor non-linear down-sampling which can't be proved by frequency domain analysis. To prove this theorem trigonometry, formulas of sum and difference in angle [9] and values of sine and cosine are used.

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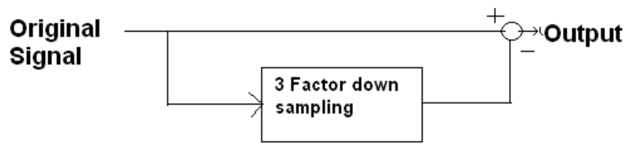

Fig. 1. A similar operation of our compression

II. Operation

Fig. 1 shows a similar operation of the proposed compression. In diagram (Fig. 1), 3 factor down-sampled [10] data is subtracted from original signal. In our compression 3 factor down-sampled data is discarded from original data. For example when input is-
x(1) x(2) x(3) x(4) x(5) x(6) x(7) x(8).
Output of this system:
x(1) x(2) 0 x(4) x(5) 0 x(7) x(8).
Output of proposed system:
x(1) x(2) x(4) x(5) x(7) x(8).
x(3) x(6) .... will be reconstructed before hearing sound.

III. Mathematical Background

When frequency of the original signal is small compared to F_N original signal can be reconstructed from down-sampled data using interpolation with very small noise.

At high frequency two noise signals are found. One of them comes due to folding [13]. As it is non-linear down sampling original signal is found with less amplitude because of one same sampling period (T_s) and also noise of folding frequency F=(2/3)F_N is found because of one double sampling period ($2T_s$). Assuming a signal of frequency,

$$f = \frac{2}{3}F_N + f1Hz \tag{1}$$

So, Folded signal,

$$S_f = sin(2\pi\frac{(\frac{2}{3}f_N - f1)n}{F_s}) = sin(2\pi\frac{n}{3} - 2\pi\frac{f1n}{F_s}) \tag{2}$$

[As, $Fs = 2 * F_N$]
Original signal,

$$S_m = sin(2\pi\frac{(\frac{2}{3}f_N + f1)n}{F_s}) = sin(2\pi\frac{n}{3} + 2\pi\frac{f1n}{F_s}) \tag{3}$$

From eqn. (2) and eqn. (3), a relation between these two signals is found. That is-
Folded signal - Original signal =$S_f - S_m =$

$$sin(2\pi\frac{n}{3} - 2\pi\frac{f1n}{F_s}) - sin(2\pi\frac{n}{3} + 2\pi\frac{f1n}{F_s}) = \tag{4}$$

-2 $cos(2\pi\frac{n}{3})sin(2\pi\frac{f1n}{F_s})$

Here, $sin(2\pi f1n/F_s)$ is the low frequency noise signal, S_l; it's coefficient [9]-
-2 $cos(2\pi n/3) = 1 for, n\%3 = 1, 2$.
-2 $cos(2\pi n/3) = -2 for, n\%3 = 0$.
So, for two samples n=1, 2:

$$S_f - S_m = S_l \tag{5}$$

The relation expressed in eqn. (5) is not true for third sample which is discarded and this causes noise. It can be written as-

$$S_f + S_l = S_m \tag{6}$$

For original signal of frequency, $f = \frac{2}{3}F_N - f1Hz$
Same equation will be found as eqn. (6) with-
$S_l = -sin(2\pi f1n/F_s)$

According to this relation (eqn. (6)), a part of original signal may create noise. When signals are reconstructed using interpolation, lowest possible frequency signals are reconstructed. For this folding occurs [12]. Folded signal and low frequency noise signal are of same amplitude.

Let, x% of the original signal is creating noise.

According to eqn. (6), following phenomena will occur.

- output = (1 - x%)Original signal + x%(Folded signal- Low frequency noise signal)
- π radian phase-shift occurs between folded signal and low frequency noise signal.
- frequency difference between low frequency noise signal and change in original signal at high frequency $= (2/3)F_N$

To fix this problem, low frequency signal is selected from output through low-pass filter [11]. Then it is shifted by (2/3)F_N Hz multiplying by $sin(2\pi(2/3)F_N t)$ and added with original signal[12]. After that, low frequency signal is eliminated from output using high-pass filter.

Compression ratio is defined by the eqn. (7)

$$CR = \frac{B}{B_o} \tag{7}$$

Where,
B_o= Size of data before compression
B= Size of data after compression

Compression ratio for the proposed compression technique, $CR = \frac{100-33.333}{100} = 0.667$.

IV. Simulation Result

A model matlab code is built. It can manipulate signals up to 3000Hz when signal contains no frequencies, carrying major power and smaller than 300Hz. It works in two ways-

- At low frequency (only interpolation for signal reconstruction)
- At high frequency (noise minimization after signal reconstruction)

It is successful for five male and five female voices, sound of bird (available in matlab sptool), pure sine waves. The important thing is that it is successful over all real human clean voices used for simulation.

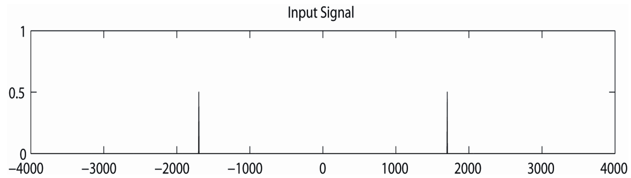

Fig. 2. Original signal, frequency=1700Hz

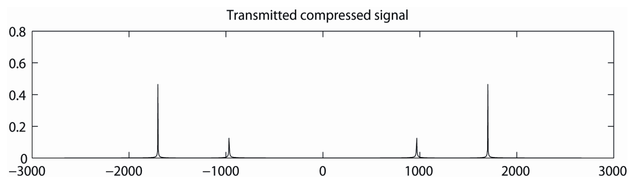

Fig. 3. Signal after down-sampling

A. Normal Case

Normally frequencies, carrying major power of voice signal are smaller than 2200Hz, f $<$ (5/9)F$_N$. Original signal can be reconstructed from transmitted or down-sampled signal in this case easily only using interpolation. Here a pure sine wave of 1700Hz is used as input. Signal spectrums are shown in Fig. 2-4.

Here signal of a female clean voice (matlab, available in matlab sptool) is used as input. Input-output signals and signal-spectrums are given Fig. 5-7.

Here it can be said that input and output signal are approximately same.

B. Exceptional Case

In previous method signals having frequency lower than 2350Hz can be manipulated. To manipulate signals of higher frequency (exception) errors due to down-sampling and interpolation should be considered. When frequency of signal is more than 2000Hz, two significant noise signals produced.

Here a pure sine wave of 2400Hz is used as input. Signal spectrums are given in Fig. 8-10.

Elimination of these noises is vital. When a 2400Hz sinusoidal signal is used, two noise signals have frequencies of (2666.6667 - 2400) = 266.66667Hz and (2666.6667$\times$2 - 2400) = 2933.3333Hz.

They are of same amplitude (approximately) but their amplitude increases in increase of frequency. Their frequency

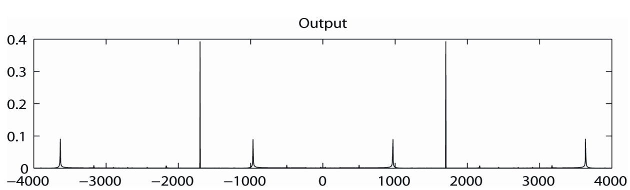

Fig. 4. Reconstructed signal containing noise (Noise is not creating any problem as noise-power is small)

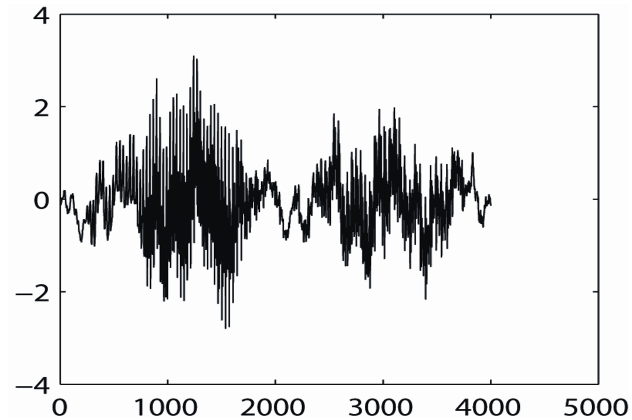

Fig. 5. Input of a female clean voice (matlab, available in sptool) [14]

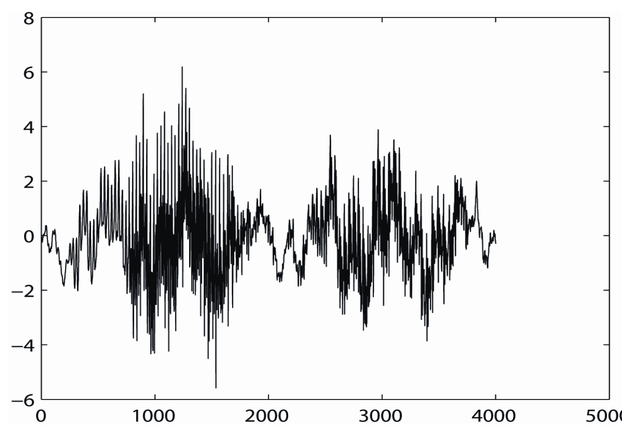

Fig. 6. Output signal

difference is 2666.6667Hz. This frequency difference is applicable for any input signal. So if it is possible to shift low frequency signal by 2666.6667Hz and to subtract it from total signal high-frequency noise will be eliminated. Then low frequency signal will be eliminated using high-pass filter. Noise eliminated signal is shown in Fig. 11.

Method for exceptional case can be used when a frame contains only high frequency signals. For an example signal of frequency 3600Hz can be manipulated when frame contains no frequencies, carrying major power and smaller than 1000Hz.

In following example a high frequency practical signal (chirp, available in matlab sptool) is used as input. Input-output signals and signal-spectrums are given in Fig. 12-15.

From figures it can also be said that input and output signal are approximately same after applying noise elimination operation.

V. DISCUSSION

Speech compression algorithms, in mobile satellite systems bring us lossy compressions [15]. Our compression is one of them. It can be used in voice communication. In PCM

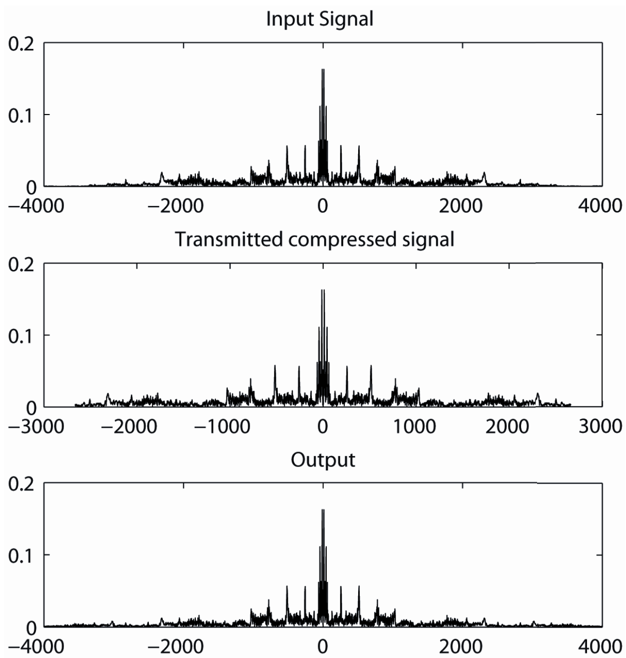

Fig. 7. Input and output signal spectrum of a female clean voice (matlab, available in sptool) [14]

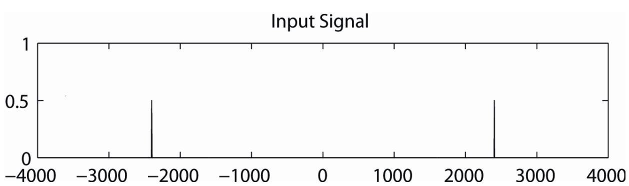

Fig. 8. Original signal of frequency=2400Hz

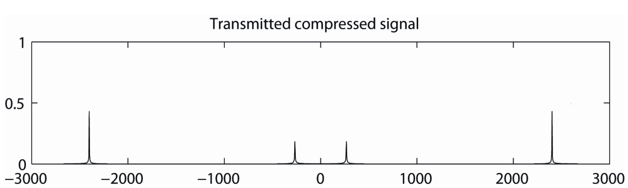

Fig. 9. Signal after down-sampling

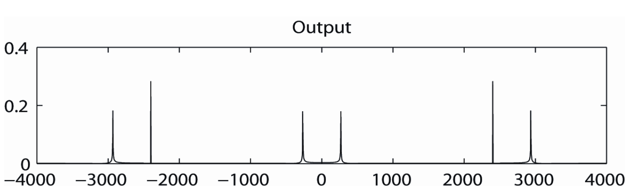

Fig. 10. Reconstructed signal containing noise (noises at 266Hz and 2933Hz are of same amplitude)

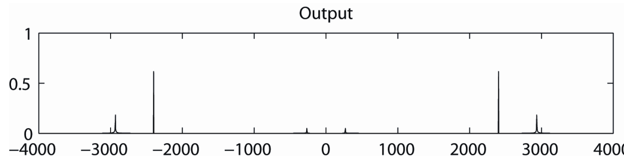

Fig. 11. Signal after elimination of noise

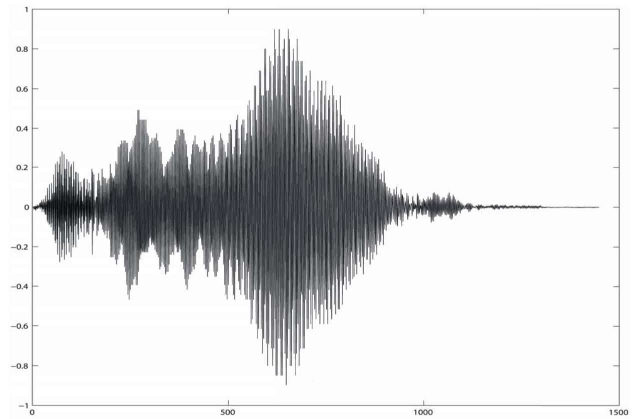

Fig. 12. Input signal of chirp (High frequency signal available in matlab sptool)

8K samples are sent per second. In proposed process 2/3 of these samples will be sent. Using the proposed compression scheme 33.33% bandwidth can be saved or number of user can be increased by 50% in cellular communication. LPC Analysis and Synthesis of Speech are also possible in compressed signal.

The proposed technique will create some noise if signal contains frequencies near to 4K Hz. Voice signals doesn't contain such high frequency terms, moreover a band-pass filter will be used before sampling. So this technique is an efficient one.

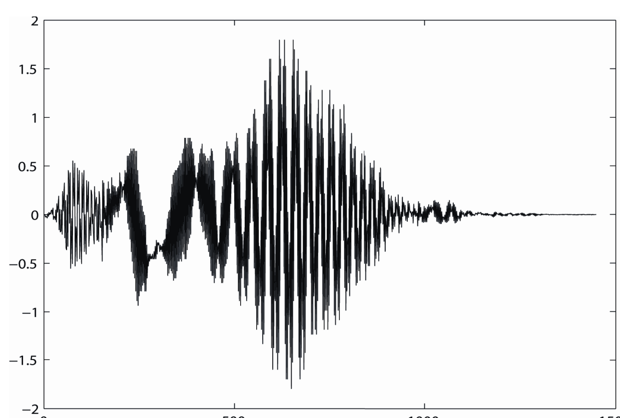

Fig. 13. Signal after interpolation-reconstruction operation (before noise elimination)

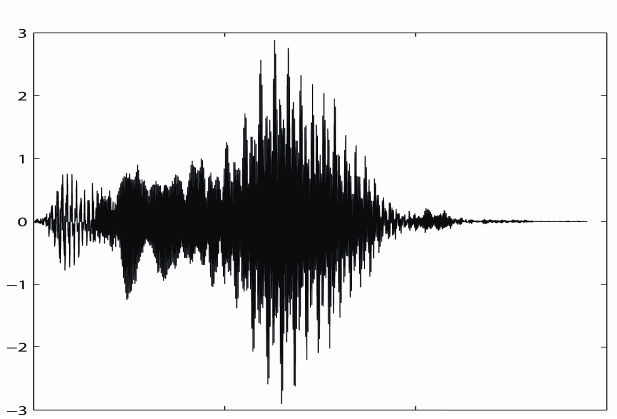

Fig. 14. Output after elimination of noise

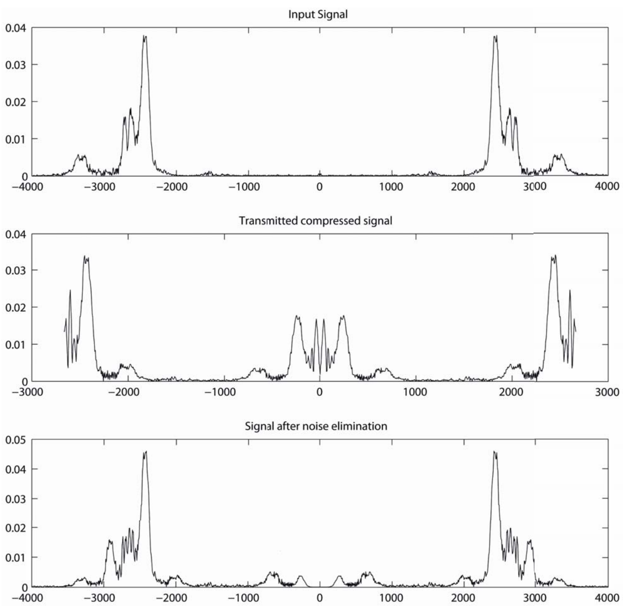

Fig. 15. Input and output signal spectrum of chirp (High frequency signal available in sptool)

The main problem of digital data is that, it takes huge space. Entropy coding is used for compressing data. Huffman coding is an example of entropy coding [16]. These codings are used for zipping files. The main problem of these algorithms is compression is not always possible and compression ratio is not good for speech signals. Modern goal of down sampling is to maintain good quality using smaller memory space [17]. Proposed system will be an efficient way of sending less number of samples. It is better process than reducing sampling rate because it works as a combination of two systems. It can manipulate both high frequency and low frequency signals. In proposed system bit-rate is constant, so network will not face difficulty to transmit signals [18]. So, proposed technique can be used for efficient real-time signal transmission.

VI. Conclusion and Future Work

The proposed compression reduces bandwidth requirement for signal transmission. LPC Analysis and Synthesis of Speech are also possible in compressed signal. We will work for creating a fast processor for this operation. If this operation is performed in normal processors, a large time will be needed but this time is enough small for transmitting and receiving data. For faster implementation of the system FPGA will be an efficient device.

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References

[1] T. Wescott, Wescott Design Services. [Online]. Available in: http://www.wescottdesign.com/articles/Sampling/sampling.html

[2] S. Haykin M. Mohers, *An Introduction to Analog and Digital Communications*, 2nd ed., 1989.

[3] A. Mawla, M. A. Najih, A. R. Ramli, A. Ibrahim, Syed , "Comparing Speech Compression Using Wavelets With Other Speech Compression Schemes," in *IEEE Proceedings*, Putrajaya, Malaysia, 2003.

[4] S. M. Qaisar, L. Fesquet, M. Renaudin, "Computationally Efficient Adaptive Rate Sampling and Filtering" in *EUSIPCO*, Poznan, 2007.

[5] J.W. Mark and T.D. Todd, "A nonuniform sampling approach to data compression," *IEEE Transactions on communication*, vol. COM-29, pp. 24-32, January 1981

[6] C. Rose, R. W. Donaldson, "Real-time Implementation and Evaluation of an Adaptive Silence Detection Algorithm for Speech Compression," in *IEEE Pacific Rim Conference on Communications, Computers and Signal Processing,*, May 9-10, 1991

[7] *Building Your Vector Quantizer Model in Matlab John*, Matlab, 2007.

[8] S. Brown and Z. Vranesic, *Fundamentals of Ddigital Logic with Verilog Design*, Tata McGraw-Hill Edition, 2002.

[9] Trigonometry sum and difference formulas. [Online].Available: http://en.wikibooks.org

[10] S. W. Smith, *The Scientist and Engineer's Guide to Digital Signal Processing*, 2nd ed., 1997.

[11] S. S. soliman, M. D. srinath, *Continuous and Discrete Signals and Systems*, 2nd ed.

[12] J. C. Bellamy, *Digital Telephony*, 3rd ed.

[13] G. Proakis, D. K Manolakis, *Digital Signal Processing Principles, Algorithms and applications*, 4th ed., 30 March, 2009

[14] *Matlab Sptool ,Filter designing and available signals*, Matlab, 2007.

[15] M. Markovic, B. BoSkovic, "Speech Compression Algorithms in Mobile Satellite Systems," in *IEEE Trans.*, pp. 13-15, NiS, Yugoslavia, October 1999.

[16] C. Giria; B. M. Raob; S. Chattopadhyaya, "Split Variable-length Input Huffman Code With Application to Test Data Compression for Embedded Cores in SOCs ," in *International Journal of Electronics*, Vol. 96, Issue 9, pp 935 - 942, September 2009.

[17] L. Fang, O. C. Au, X. Wen, Y. Yang, W. Tang, "An Lmmse-based Merging Approach for Subpixel-based Downsampling," in *17th European Signal Processing Conference (EUSIPCO)*, Glasgow, Scotland, August 24-28, 2009.

[18] A. S. Tanenbum, *Computer Networks*, p-308, 4th ed., 2003.