Elektronika
Telekomunikacije
Računarstvo
Automatika
Nuklearna tehnika

ZBORNIK APSTRAKTA I PROGRAM

52. KONFERENCIJE ETRAN-a

Palić
8 – 12. juna 2008. godine

Beograd, jun 2008.

SEKCIJA ZA ELEKTRIČNA KOLA I SISTEME I PROCESIRANJE SIGNALA – EK

SEDNICA EK 1 Digitalni filtri i transformacije
Predsedava: Vidosav Stojanović
Ponedeljak, 9. 6. 2008, 9:00, sala 5

EK1.1
DIREKTNA SINTEZA IIR FILTARA POGODNA ZA KOMPLEMENTARNU DEKOMPOZICIJU

Dragana Živaljević, Vidosav Stojanović, Elektronski fakultet u Nišu

U radu je opisana direktna sinteza rekurzivnih digitalnih filtara čija karakteristična funkcija pripada klasi prelaznih IIR filtara. Karakteristika slabljenja u propusnom opsegu varira između maksimalno ravne i mini-max karakteristike. Sve nule prenosa nalaze se na jediničnom krugu tako da je ispunjen uslov za komplementarnu dekompoziciju. Izvršena je dekompozicija filtara neparnog reda na paralelnu vezu dva fazna korektora a zatim je sabiranjem formiran komplementarni niskofrekventni a oduzimanjem visokofrekventni filtar. Ovi filtri generalno imaju osobine da su mnogo manje osetljivi na promenu dužine digitalne reči nego standardni filtri.

EK1.2
REALIZACIJA DIGITALNIH FILTARA SA ALGEBARSKOM PETLJOM

Jelena Ćertić, Elektrotehnički fakultet u Beogradu, Miroslav Lutovac, Univerzitet u Novom Pazaru, Ljiljana Milić, Elektrotehnički fakultet u Beogradu

U radu je pokazano da je moguće realizovati digitalne filtre sa algebarskom petljom iako takvi filtri ne mogu da se realizuju standardnim postupcima. Na primeru filtra devetog reda je pokazano da su filtri sa algebarskom petljom robusniji na kvantizacije koeficijenata posle primene frekvencijskih transformacija.

EK1.3
REALIZACIJA DIGITALNIH FILTARA ZA VELIKE BRZINE OBRADE PRIMENOM PROTOČNE OBRADE SIGNALA

Miroslav Lutovac, Univerzitet u Novom Pazaru, Milenko Ćirić, Tehničko Remontni Zavod Čačak

U radu je predložen originalan pristup razvoju algoritama i projektovanja digitalnih filtara korišćenjem protočne obrade signala. Testiranje rada digitalnog filtra je urađeno korišćenjem simboličkog procesiranja.

EK1.4
DESIGN OF EFFICIENT POLYNOMIAL-BASED FILTERS USING OPIMIZATION METHOD FOR DISCRETE-TIME MODIFIED COMB FILTERS

Đorđe Babić, Računarski Fakultet, Knez Mihailova 6/VI, Beograd

This paper presents several novel structures for conversion between arbitrary sampling rates. The method allows arbitrary number of zeros at multiples of both input and output sampling rates, thus it has both good anti-imaging and good antialiasing properties. The novel structures are derived from continuous-time equivalents of CIC and modified comb filters. The implementation can be based on either the modified Farrow or transposed modified Farrow structure. By choosing one of these two alternatives appropriately, it is possible to shift most of the operations to lower sampling rate. Furthermore, it is possible to improve frequency domain performance by choosing continuous-time equivalent of the modified comb filter. In this way, we obtain very efficient filters for arbitrary sampling rate conversion, which are optimized directly in digital domain.

EK1.5
MODIFIKACIJA ICI ALGORITMA ZA IF ESTIMACIJU ZA SIGNALE SA MALIM SNR ODNOSOM

Semir Tuzović, MTEL, Podgorica, Igor Đurović, Elektrotehnički fakultet, Podgorica

Vremensko-frekvencijske (TF) distribucije često se koriste za estimaciju trenutne frekvencije (IF) signala koji brzo mijenjaju spektralni sadržaj. Estimacija IF-a može se obaviti određivanjem pozicije maksimuma TF distribucije u nekom trenutku. Dobra estimacija IF-a na osnovu pozicije maksimuma TF distribucije može se postići primjenom ICI algoritma (algoritam o preklapanju intervala povjerenja) kada varijansa šuma nije velika. U radu je prikazana primjena modifikovanog ICI algoritma za estimaciju IF-a, kao jedan od metoda koji se može primijeniti u uslovima visokog šuma. Polazeći od osnovnog ICI algoritma kod modifikovanog se uvode Wienerov i median filtar čime se smanjuje uticaj šuma, kao i maksimalna razlika dvije susjedne vrijednosti IF-a, čime se u velikoj mjeri eliminišu značajna odstupanja IF-a od njene stvarne vrijednosti. Na kraju su prikazani dobijeni rezultati u Matlabu za 50 iteracija na primjeru IF-a oblika |t| za različite vrijednosti varijanse šuma.

EK1.6
LOW COMPLEXITY SOURCE CODING BASED ON HADAMARD TRANSFORM

Dragutin Šević, Institut za fiziku, Visoka škola ERSS, Beograd,

Bratislav Marinković, Institut za fiziku, Visoka škola ERSS, Beograd,

Slavica Marinković, Visoka škola ERSS, Beograd

This paper presents a simple solution for low complexity source coding, based on Hadamard transform. Simulation results are presented and discussed.

LOW COMPLEXITY SOURCE CODING BASED ON HADAMARD TRANSFORM

Dragutin Šević, *Institut za fiziku, Visoka škola ERSS, Beograd, sevic@phy.bg.ac.yu*
Bratislav Marinković, *Institut za fiziku, Visoka škola ERSS, Beograd, marinkov@phy.bg.ac.yu*
Slavica Marinković, *Visoka škola ERSS, Beograd, smarinkovic@vet.edu.yu*

Abstract – *This paper presents a simple solution for low complexity source coding, based on Hadamard transform. Simulation results are presented and discussed.*

1. INTRODUCTION

The research in the communications part of our "Networked control systems" project is concerned with designing simple algorithms, aware of limited energy, processing and storage resources, that will achieve fast, efficient and reliable communication in a networked control system. Low-complexity source coding (compression) and channel coding (error protection) techniques are among the main areas of interest. This paper presents a simple solution for low complexity source coding, based on Hadamard transform. Section 2 shortly reviews the Hadamard transform. Implementation of Hadamard transform based on short kernel 2-tap filters is described in Section 3. Simulation experiments were conducted and their results are presented in section 4.

Experimental results presented in this paper are simply obtained by efficient use of short kernel 2-tap filter bank. However, seeing the equivalence between the 2-tap short kernel filters used in our work and the Hadamard transform provides better framework for future research.

2. HADAMARD TRANSFORM

Unlike the sine transforms, the elements of the basis vectors of the Hadamard transform take only the binary values ±1, and are, therefore, well suited for low complexity applications [1]. The Hadamard transform matrices, $\mathbf{H}_n$, are N x N matrices, where $N=2^n$. These can be easily generated by the core matrix:

$$\mathbf{H}_1 = \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} / \sqrt{2} \qquad (1)$$

and the Kronecker product recursion:

$$\mathbf{H}_n = \mathbf{H}_{n-1} \otimes \mathbf{H}_1 \qquad (2)$$

In series form the Hadamard transform pair is given by:

$$v[k] = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} u[m](-1)^{b(k,m)}, \;\; 0 \le k \le N-1 \qquad (3)$$

$$u[m] = \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} v[k](-1)^{b(k,m)}, \;\; 0 \le m \le N-1 \qquad (4)$$

where

$$b(k,m) = \sum_{i=0}^{n-1} k_i \, m_i \; , \quad k_i \, , \, m_i \; = 0,1 \qquad (3)$$

and $\{k_i\},\{m_i\}$ are binary representations of k and m, respectively, that is,

$$\begin{aligned} k &= k_0 + 2k_1 + ... + 2^{n-1} k_{n-1} \\ m &= m_0 + 2m_1 + ... + 2^{n-1} m_{n-1} \end{aligned} \qquad (4)$$

3. IMPLEMENTATION OF HADAMARD TRANSFORM USING SHORT KERNEL 2-TAP FILTERS

Hadamard transform could be implemented by a well known 2-tap filtering process. Low band signal $L[n]$ and high band signal $H[n]$ are derived from a pair of succesive signal points $X[n]$ and $X[n+1]$ by following equation:

$$L[n] = \big(X[n]+X[n+1]\big) \cdot 0.5 \qquad (5)$$

$$H[n] = \big(X[n] - X[n+1]\big) \cdot 0.5 \qquad (6)$$

Multiplication by 0.5 instead of $1/\sqrt{2}$ keeps output in the same range as input.

It is easy to see that reconstruction is achieved by:

$$Xr[n]=L[n]+H[n] \qquad (7)$$

$$Xr[n+1]=L[n]-H[n] \qquad (8)$$

which could be also derived from series form of Hadamard length-2 transform. Using equations (5) and (6) the two band decomposition of signal is achieved, see left part of Figure 1.

Four band decomposition is obtained by repeating the two band decomposition, as shown sketched in Figure 1.

Synthesis, shown sketched in Figure 2, is based on equations (7) and (8). Note that, for the sake of simplicity, index operations are omitted from Figures 1 and 2.

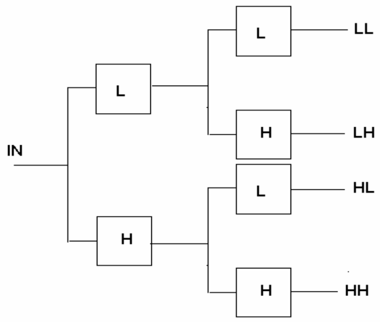

Figure 1. Analysis filter bank

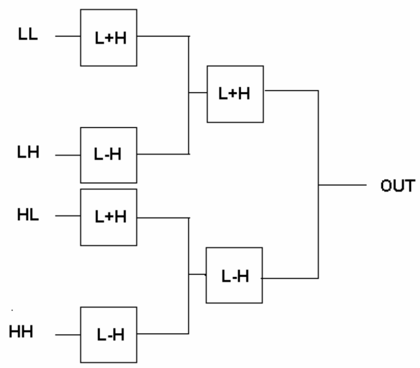

Figure 2. Synthesis filter bank

One of advantages of used algorithm is possibility of using integer arithmetic. However, errors are introduced in integer arithmetic when odd numbers are divided by two. Perfect reconstruction could be obtained by increasing the number of subband signal levels, or by any other way to keep a record of divisions of odd numbers. Anyway, this will lead to increase of entropy of signal. Sophisticated algorithm for reduction of number of subband signal levels was proposed in [2]. At this stage of our investigations, we don't see necesity to use it.

Calculations from first and second level of analysis and synthesis could be combined, similarly to as proposed in [3]. Moreover, combining the analysis calculations partially reduces the erorr introduced by integer divisions.

4. SIMULATION RESULTS

To give good insight into behaviour of proposed compression technique, three types of test signals were used: "distorted" sine sequence, "smooth" sine sequence, and pseudo random sequence. "Distorted" means that quantization was very coarse, "smooth" means oposite as far as posible, pseudo random was generated by random number generator. Test signals are shown on Figures 3, 4 and 5.

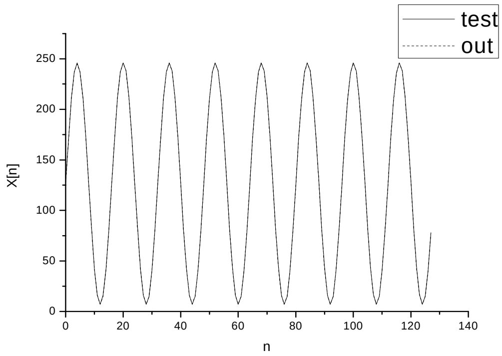

Figure 3. Distorted sine sequence

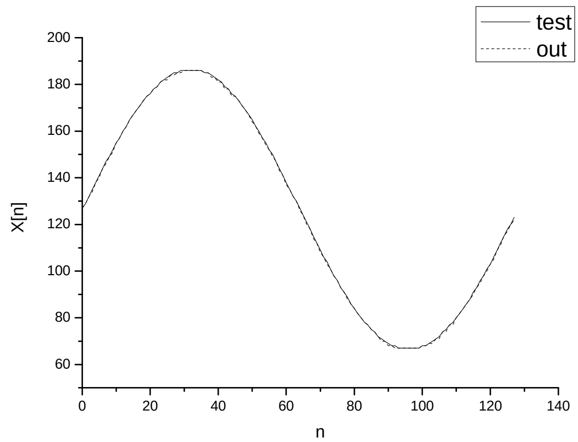

Figure 4. Smooth sine sequence

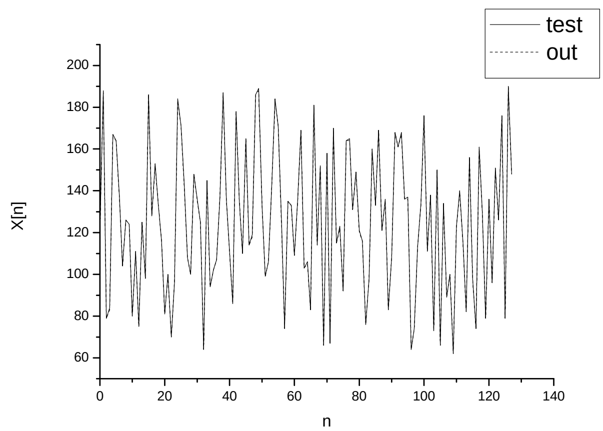

Figure 5. Pseudo random sequence

The signal to noise ratio, SNR, peak signal to noise ratio, PSNR, and first order entropy are calculated in usual maner. Calculated entropy coresponds to required bit rate. As a indicator of overall entropy reduction, averaged entropy of subbands is calculated and presented in Tables 1, 2 and 3, denoted as "avg entropy".

Reconstructed filter out signals for two-band and four-band analisys are also shown on figures 3, 4 and 5. However, reconstructed signals are visibly indiscernable from test signals.

Comparing data shown in Tables 1, 2 and 3 we see that better entropy reduction is obtained by four band signal decomposition, which increases compression losses, as expected.

Table 1. *Simulation results for distorted sine signal*

Distorted sine sequence, signal entropy=4.984148		
	2 band	**4 band**
L band entropy	**4.055540**	
H band entropy	**3.067946**	
LL band entropy		**3.203627**
LH band entropy		**2.715826**
HL band entropy		**2.477217**
HH band entropy		**2.238609**
avg entropy	**3.561743**	**2.658819**
SNR [db]	**42.853**	**37.553**
PSNR [db]	**77.722**	**72.422**

Table 2. *Simulation results for smooth sine signal*

Smooth sine sequence, signal entropy=5.833571		
	2 band	**4 band**
L band entropy	**5.738205**	
H band entropy	**1.584612**	
LL band entropy		**4.812500**
LH band entropy		**2.539717**
HL band entropy		**1.530994**
HH band entropy		**0.000000**
avg entropy	**3.661408**	**2.220803**
SNR [db]	**44.980**	**43.849**
PSNR [db]	**74.593**	**73.463**

Table 2. *Simulation results for pseudo random signal*

Pseudo random sequence, signal entropy= 6.147979		
	2 band	**4 band**
L band entropy	**5.402115**	
H band entropy	**5.476410**	
LL band entropy		**4.375000**
LH band entropy		**4.687500**
HL band entropy		**4.538910**
HH band entropy		**4.625000**
avg entropy		**4.556602**
SNR [db]	**45.996**	**44.800**
PSNR [db]	**74.147**	**72.951**

4. DISCUSION AND CONCLUSION

Presented simulations show that nearly lossless compresion is achieved. Perfect reconstruction could be obtained by increasing the number of subband signal levels. This will lead to increase of entropy of signal. Sophisticated algorithm for reduction of number of subband signal levels was proposed in [2]. For our aplications we don't see necessity to use it.

Better entropy reduction is obtained by four band signal decomposition, which increases compression losses.

It should be pointed out that to obtain practical results presented in this paper it is not necessary to have any knowledge about Hadamard transform. However, seeing the equivalence between the 2-tap short kernel filters used in our work and the Hadamard transform provides better framework for future research.

Acknowledgment

This work was supported by bilateral project "Networked control systems - Pavle Savic" and project 141011 of MS RS.

REFERENCES

[1] Anil K. Jain, "Fundamentals of Digital Image Processing," Prentice Hall, 1989.

[2] Kazunari Irie and Ryozo Kishimoto, "A Study on Perfect Reconstructive Subband Coding," in *IEEE trans. on Circuits and Systems for Video Technology,* vol 1. no 1, march 1991, pp. 42-49.

[3] Dragutin Sevic, "Improvements in Subband Coding of Images," PhD Thesis (in serbian), Faculty of Electrical Engineering, Belgrade, 1995.