Hardware and Software Solution Signal Compression Algorithms Based on the Chebyshev Polynomial

M. M. Musaev, U. A. Berdanov, and K. E. Shukurov

Abstract—**In this paper, signal compression algorithm of Chebyshev polynomial is implemented. Upper case and lover case paralleled algorithm of compression signals processing suggested. However, audio and speech signal compressing accuracy in ADSPBF-561 dual core processor are analyzed.**

Index Terms—**Discrete cosine transform, Chebyshev polynomial, signal processors, parallel programming, library automation programming.**

I. Introduction

Today spectral methods of digital signal processing includes several transformations. Among them, Fourier transform, cosine and many similar transformations.

These transformations differ each other with basis functions and the calculation processes. Using these transformations, first we calculate spectral values and based on them spectral values signal can be restored. After that using spectral value we can research internal properties of the signal. In these values are reflected analysis of signal, filtration process, signal compression processes. Processors and digital signal processors provides the process of calculation in digital signal processing. Digital signal processing is mainly implemented in real time systems. In the real time systems signals are received, processed and transmitted simultaneously. During signal processing in real-time systems, a signal incoming from the analog-to-digital converter using spectral methods obtains spectral values of signal [1]. Using these spectral values we can organize of the signal processing.

The one of the main direction of digital signal processing is the signal compression. Today, there are many methods of digital signal compression based on different mathematical methods. By using these digital compression algorithms in real time systems, we can solve kind of problems like saving memory space, time delay, signal processing time, performance of digital signal processors etc. Consequently, working on digital signal compressing and developing algorithms of digital signal compressing of the real time systems becomes important.

II. Algorithm and Numerical Method

Till today in many signal processing devices used algorithms are based on classical methods. The classic methods works with signal coefficients and using these coefficients compresses signal. These algorithms refers Newton, Lagrange, method of least squares and others. Using classic algorithms to digital signal compressing gives a good effect but it can’t give normal results when you use in real time systems for digital signal processing. The main reason is increasing amount of signal values leads to increase number of solvable equation. Consequently, the solving the equations requires time, this leads to a delay which is not desirable for real time systems. Therefore, today many real time systems use digital signal processors. General structure of the process of direct and inverse transformation in a specialized signal processor is given in the illustration below (Fig. 1).

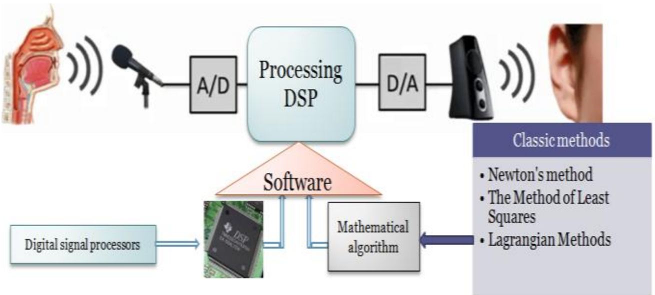

Fig. 1. Structure of signal compression.

This represented method contains process of analyzing the signal using a polynomial [2]. In this case, using the transformation method creates the following structure (Fig. 2). This structure can show that the process recovering of signal is performed by solving equation and the degree of the polynomial.

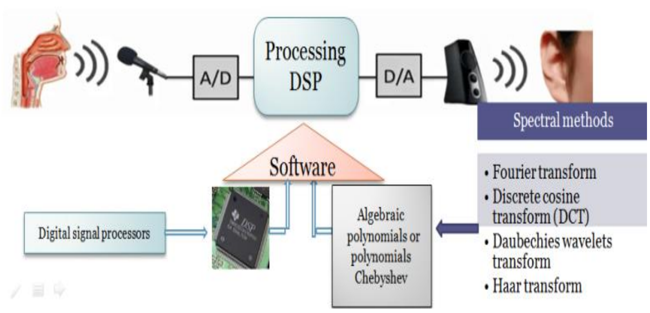

Fig. 2. Structure expression of signal to polynomial.

In basis of this structure lies the Parseval equality. In Parseval's Theorem, the sum of squared coefficient of the signal and the sum of squared of the coefficients of the spectrum determines the energy [3]. Based on the above-mentioned we can say that the signal energy is equal to the sum of its spectral coefficients. That’s:

$$P = \frac{1}{T}\int_{-t_a}^{t_a} s^2(t)dt = \sum_{k=0}^{n} c_k^2 \qquad (1)$$

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Based on Chebyshev's polynomial proposed for signal processing in the spectral form, we can write the following

$$f(x) = \sum_{k=0}^{m} A_k(u) \tag{2}$$

In this structure the process of digital signal processing solved by discrete cosine transform [4]. In cosine transformation, direct conversion is determined by formula (3) and inverse by formula (4)

$$C_k = \sum_{k=0}^{N-1} f(i)\, D(k,i), \quad k = 0 \ldots N-1; \tag{3}$$

$$f(i) = \sum_{k=0}^{N-1} C_k D(k,i), \quad i = 0 \ldots N-1 \tag{4}$$

f(i) – signal values, C_k– spectral values, D(k,i) - transformation matrix.

In this method, it is sufficient to obtain the spectral coefficients of the signal. The process of signal compression is implemented by these spectral coefficients [5]. In this method amount of spectral coefficients varies according to $N=2^n$. We select the polynomial corresponding to this. Spectral coefficients for a signal $N=8$, we take Chebyshev's polynomial's with a degree $k=2$ and $k=3$ as initial conditions.

$$N=8, \quad h=\frac{1}{N-1}, \quad x=[0:h:1], \quad q=\frac{x_0+x_N}{2}, \quad u=\frac{x-q}{h} \tag{5}$$

$$A_0(u)=1, \quad A_1(u)=u, \quad A_2(u)=u^2-5\frac{1}{4}, \quad A_3(u)=u^3-9\frac{1}{4}u; \tag{6}$$

$$f(x) = A_0(u)+A_1(u)+A_2(u) \quad or \quad f(x) = -4\frac{1}{4}+u+u^2 \tag{7}$$

This is Perseval's method that in the processing of analyzing of signals, some parts of divided signal might be expressed. Then, following formula is came (9).

$$\sum_{k=0}^{N-1} f(x_k) D(k,i) = \sum_{j=1}^{N-1}\left(\sum_{k=0}^{m} A_k u\right) D(k,i); \tag{9}$$

After distributing of above expression, following equation systems are appeared.

$$\begin{cases} \frac{3363}{1189}A_0 = C_0 \\ -\frac{8098}{1257}A_1 = C_1 \\ \frac{1022}{81}A_2 = C_2 \\ -\frac{926}{1375}A_1 = C_3 \\ \frac{3363}{1189}A_2 = C_4 \\ -\frac{89}{443}A_1 = C_5 \\ \frac{703}{784}A_2 = C_6 \\ -\frac{148}{2919}A_1 = C_7 \end{cases} \tag{10}$$

$$\begin{cases} \frac{3363}{1189}A_0 = C_0 \\ -\frac{8098}{1257}A_1 + \frac{959}{362}A_3 = C_1 \\ \frac{1022}{81}A_2 = C_2 \\ -\frac{926}{1375}A_1 - \frac{2851}{124}A_3 = C_3 \\ \frac{3363}{1189}A_2 = C_4 \\ -\frac{89}{443}A_1 - \frac{2168}{293}A_3 = C_5 \\ \frac{703}{784}A_2 = C_6 \\ -\frac{148}{2919}A_1 - \frac{1363}{718}A_3 = C_7 \end{cases} \tag{11}$$

Let's make a simpler of these equation systems. And then, coefficient of A_0, A_1, A_2, A_3 is found, and below result is obtained.

TABLE I: THE EXPRESSION OF THE SPECTER OF CHEBYSHEV'S POLYNOMIAL

	$\kappa=2$	$\kappa=3$
A_3		$A_3=\frac{127}{945}\left(C_5-\frac{527}{133}C_7\right)$
A_2	$A_2=-\frac{184}{3007}(C_2+C_4+C_6)$	$A_2=\frac{184}{3007}(C_2+C_4+C_6)$
A_1	$A_1=-\frac{558}{4111}(C_1+C_3+C_3+C_7)$	$A_1=-\frac{881}{6269}\left(C_1+C_3+\frac{5447}{440}A3\right)$
A_0	$A_0=\frac{1189}{3363}C_0$	$A_0=\frac{1189}{3363}(C_0)$

According to the compression of signals, in the sending signal to the long distance and by the calculating of spectral coefficients such as A_0, A_1, A_2, A_3 values which are recorded in the memory where small part of memory is required and compressed [6].

When signal coefficients increased to $N=16$, high compressing level might be obtained. In processors Blackfin SHARC and TigerSHARC of the company Analog devices algebraic functions, audio signals and speech signals using the above specified method analyzed and obtained results. Results of analysis are shown on Table I.

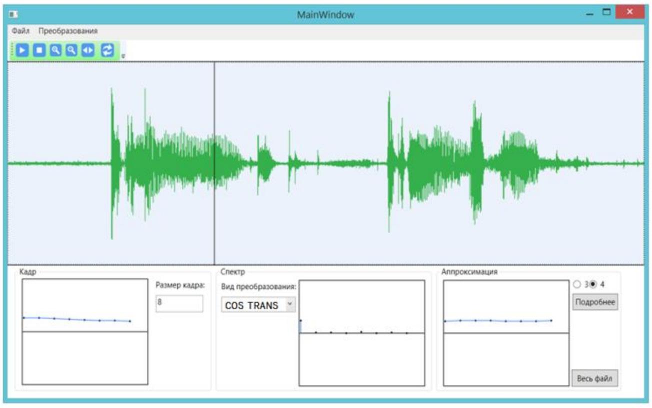
TABLE II: SPEECH AND AUDIO SIGNALS FOR THE COMPRESSING VALUES OF THE CHEBYSHEV POLYNOMIAL

	Signal coefficients $N=8$			
	Algebraic functions& trigonometric function		Audio signals &Speech signals	
	coefficients	signal after compressing	to signal compression	signal after compressing
$\kappa=2$	8	3	8 mb (.wav)	3 mb (.asp)
$\kappa=3$	8	4	8 mb (.wav)	4 mb (.asp)
	Signal coefficients $N=16$			
	Algebraic functions & trigonometric function		Audio signals &Speech signals	
	coefficients	signal after compressing	to signal compression/fi rst signal	signal after compressing
$\kappa=2$	16	3	16 mb (.wav)	3 mb (.asp)
$\kappa=3$	16	4	16 mb (.wav)	4 mb (.asp)

III. Medium Software Implementation

When this algorithm is implemented in real systems, the most requirement calculating process is to take a specter of a signal. That why, paralleling is required for this process [7].

For paralleling, library of programming tools, technologies, programming tools of parallel processing has created in multi-core processors if uppercase programming language used. For example, Open MP technology, TBB, IPP library, Visual Parallel Studio and so on can be good example. These libraries and programming packages are used for paralleling methods (Fig. 3).



1.	Speech signal coefficients:	-0,258	-0,2547	-0,2443	-0,2324	-0,222	-0,2129	-0,2052	-0,1968
2.	Spectral coefficients:	-0,228	-0,0041	-0,0085	0,0010	-0,019	0,0002	-0,0002	0,0012
3.	Recovered speech signal coefficents:	-0,227	-0,2346	-0,2343	-0,2299	-0,224	-0,2209	-0,2127	-0,2028

Fig. 3. Programming with C#.

If we use low level programming language for programming of the algorithm, we cannot use paralleling libraries [8]. The parallel algorithms of creating spectral values of signals are calculated by multiplying the matrix to the vector. And this expression is shown (3) below.

Flows can be divided as a result of paragraphing by multiplying the matrix to vector in parallel case. m=1…j is contained by number of these threads. If multiplying the matrix to vector divided $m=2$ threads, formula (12) is appeared.

$$C_k(L_k : K_k) = \begin{cases} L_k = \sum_{i=0}^{N/2-1} f(i)D(k,i) \quad if \quad i = 0...N/2-1; \quad k = 0....N-1; \\ K_k = \sum_{N/2-1}^{N-1} f(i)D(k,i) \quad if \quad i = N/2-1.......N-1; \quad k = 0....N-1; \end{cases} \tag{12}$$

Here, Lk and Kk are threads multiplying the matrix to the vector.

Today, there are many companies which can produce digital signal processors and multi-core digital signal processors. They are Texas Instruments, Freescale Semiconductor, Analog Devices, Philips Semiconductors, Agere Systems and other companies. Multi-core digital processors are high performance processors. Parallelization of cores gives us the possibility working with many different threads at the same time. Consequently, it will give the opportunity to solve many problems.

ADSP-BF561 is a dual core digital signal processor, it will give us the possibility of parallel processing of signal. The architecture of the ADSP-BF561 digital signal processor is shown in Fig. 4.

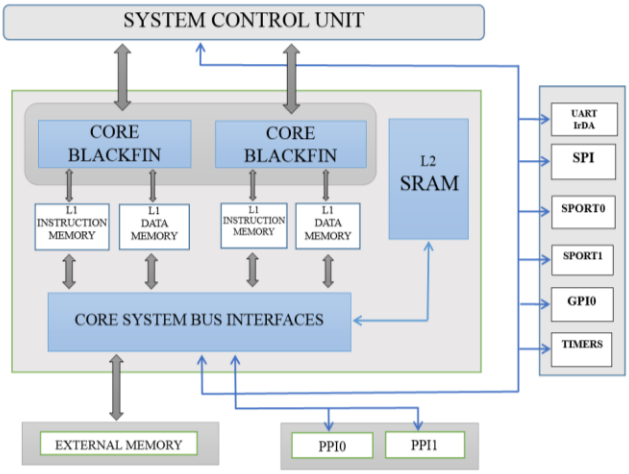


Fig. 4. Functional block diagram.

The Blackfin Processor family expands the performance envelope with the ADSP-BF561. With two high performance Blackfin Processor cores, flexible cache architecture, enhanced DMA subsystem, and Dynamic Power Management (DPM) functionality, the ADSP-BF561 can support complex control and signal processing tasks while maintaining extremely high data throughput [11].

The ADSP-BF561 is a functional extension of the popular Blackfin Processor family and is ideally suited for a broad range of industrial, instrumentation, medical, and consumer appliance applications—allowing for scalability based upon the required data bandwidth and mix of control, plus signal processing needed in the end product.

High-Level of Integration

- 328 Kbytes of on-chip memory configured as:
  1) 32Kbytes of L1 instruction memory SRAM/Cache per core.
  2) 64 Kbytes of L1 data memory SRAM/Cache per core.
  3) 4 Kbytes of L1 scratchpad memory per core.
  4) 128 Kbytes of low-latency shared L2 memory.
- 32-bit Memory Controller providing glue less connection to multiple banks of SDRAM, SRAM, Flash or ROM;
- Two Parallel Peripheral Interfaces Units supporting ITU-R 656 video data formats;
- Two dual-channel, full-duplex, synchronous serial ports supporting eight stereo I2S channels;
- Dual 16 Channel DMA Controllers, supporting one and two-dimension transfers;
- SPI-compatible Port;
- UART with support for IrDA ®;
- 12 timer/counters supporting PWM, pulse width and event count modes;
- 48 Programmable Flags/General Purpose I/O;
- Event Handler;
- Dual Watchdog timers;
- PLL capable of 1x to 63x frequency multiplication;
- 256-ball Mini-BGA and 297-ball Sparse PBGA packages.

Algorithm of dividing to threads is programmed by ADSP-BF561 signal processor of Analog Device Company [9].

TABLE III: THE COMPARISON RESULTS

Paralleling	Performing period (м sec,)		
	n=512	n=1024	n=2048
Sequence	0,082	0,104	0,298
Parallel	0,061	0,076	0,195
Acceleration coefficient	1,34	1,36	1,52

From comparison results, acceleration coefficient based on Amdal theory is evaluated [10]. This acceleration coefficient is found following formula(13).

$$K = \frac{T_s + T_p}{T_s + \frac{T_p}{Q}} \tag{13}$$

T_s-performing period of algorithm in sequence case,
T_p- performing period of algorithm in parallel case,
Q – Number of cores.

IV. CONCLUSION

In conclusion, in audio and speech compression processing only discrete cosine transform was used. And, the basis transform matrix for other type of signal analyzing methods is enough for transforming. In this case, regards to main characteristics of signal the transforming matrix will be chosen. In whole spectral transforming, this type of opportunity is existing. This could be because, current algorithm is useful for all type of signal processing.

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