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EFFICIENT APPLICATIONS AND ARCHITECTURE OF MODERN DIGITAL SIGNAL PROCESSORS

UČINKOVITE APLIKACIJE IN ARHITEKTURE MODERNIH DIGITALNIH SIGNALNIH PROCESORJEV

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Abstract

Digital signal processors have found their roles in various fields of science and technology. With the appearance of problems related to the processing of large quantities of data in real time, it was necessary to develop a system that would execute procedures very rapidly and at low cost. The most common application in real time is the digitization and mathematical processing of audio, video, temperature, and voltage data, etc., resolved using parallel operations. Various producers of digital signal processors have developed processors and evaluation models that enable developers to quickly and efficiently create unique applications in communications and visual systems, biomedicine, meteorology, etc. In this article, the basic performance and architecture of the modern digital signal processor are described in detail with emphasis on the most common applications. A practical example of the use of a digital signal processor for numerical integration is presented. A comparison with commonly used processors is performed to confirm its efficiency..

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Povzetek

Digitalni signalni procesorji se pojavljajo v različnih panogah znanosti in tehnologije. S pojavom problemov, ki zahtevajo procesiranje velikih količin podatkov v realnem času, je bilo potrebno razviti sistem, ki je sposoben izvajati operacije z večjo hitrostjo in nižjimi stroški. Najpogostejše aplikacije v realnem času so digitalizacija, matematično procesiranje avdio in video signala, temperature, napetosti ipd., ki se izvajajo z vzporednimi operacijami. Različni proizvajalci digitalnih signalnih procesorjev so razvili procesorje in ocenjevalne postopke, ki omogočajo razvijalcem hitro in učinkovito ustvarjanje edinstvenih aplikacij na področju telekomunikacij, vizualnih sistemov, biomedicine, meteorologije ipd. V članku je podrobno opisano osnovno delovanje in arhitektura modernih digitalnih signalnih procesorjev s poudarkom na najpogosteje uporabljenih aplikacijah. Predstavljen je praktični primer aplikacije digitalnega signalnega procesorja za numerično integracijo. Za potrditev učinkovitosti je podana primerjava z drugimi pogosto uporabljenimi tipi procesorjev.

1 INTRODUCTION

Digital signal processors (DSP) are used for collecting large amounts of data, which are the subject of mathematical transformations that give very good results in real time systems. Due to their basic characteristics, DSP application vary from practical everyday devices (cell phone, camera, etc.) to medical, military, scientific research and evolutionary models.

The first appearance of the DSP was in the 1970s, and it was first dominant in telecommunications, high-speed modems, military applications and medicine, because these fields could financially support the development of the expensive technology at that time. A group of engineers from Texas Instruments (TI) presented the first commercial DSP whose architecture is the closest to today's DSPs at International Solid-State Circuits Conference (ISSCC) in February 1982. Their first device was the TMS32010 with 5 million instructions per second and with 55,000 transistors, [1]. To enter the consumer market, they created a talking and listening doll named Julie, and the TMS320C17 was used for voice recognition. They also wanted to attract more customers and expand into more areas, so they started from the basic knowledge of digital signal processing and observed huge losses of energy; their aim was to reduce it, [1]. Nowadays, most of the devices that process graphics and sounds cannot be imagined without a specialized DSP processor.

In this paper, the authors analyse the basic features and architecture of DSPs. The paper is structured as follows: Section 2 presents basic performance and the architecture of DSPs; Section 4 presents the most commonly used applications and algorithms using DSPs; practical implementation is presented on a Texas Instrument evaluation model in Section 4.

2 BASIC PERFORMANCE AND ARCHITECTURE OF THE DSP

A DSP is a microprocessor that has high data flow and can process fast streaming, e.g. multimedia data processing. The execution time of the program using a DSP can be predicted and thus desirable results are guaranteed. It is possible to obtain different behaviour from the system through the reprogramming of the DSP with relevant software, i.e. with decoding algorithm execution, [2]. Programs written for regular processors are written in high-level

programming languages, but programs for the DSP are more commonly written in an assembly language because of the standard DSP architecture (multiple memory spaces, buses, irregular sets of instructions and highly specialized hardware), [3]. A DSP is a microprocessor designed for fast problem solving in digital signal processing, in particular for the rapid execution of arithmetic and logical operations and has the capability of executing one or more parallel multiply-accumulate (MAC) operations in one instruction cycle. The time of the MAC operations execution is not a primary feature of the DSP, but faster MAC operations provide better bandwidth. Due to the latter, two or more MAC units are embedded in modern DSPs. MAC operations are common in DSP applications, and they are used for vector multiplication, digital filters, correlations and Fourier transformations, [4], [5]. DSPs are commonly used for real-time processes, and they receive real time signals for audio, video, temperature, pressure or location that have to be digitized and mathematically processed in real time. They are designed for fast execution of the finite impulse response filters (FIR), which are used in digital signal processing. A FIR filter is implemented in real-time and uses circular buffering carried out through the steps listed below. The 14 steps are running parallel on a DSP, unlike on a traditional microprocessor where they are serially executed [4], [5]. Because the algorithm has to be executed quickly, internal DSPs architecture allows the execution in one cycle operations of the loop which contains steps 6-12 and they are repeated circularly, [5].

Selecting an adequate digital signal processor is an important but not easy task due to the great number of available processors. It is necessary to consider the following, [6]:

- architectural features when selecting a DSP, it is important to pay attention to onchip memory, input/output options, RAM etc. because DSPs are not multifunctional
- execution speed even though there are two basic measurement units of the CPU clock speed (MHz) and the number of instructions processed per second that a computer can process (MIPS), due to the various numbers of multiple operations of different DSPs, an alternative measure is based on a speed performance benchmark algorithm.
- type of the arithmetic although most of the PDSPs use fixed-point arithmetic, floating point arithmetic is more efficient, more precise and needs less execution time but, because of optimized DSP arithmetic, the speed is approximately equal. For temporarily storing the results of DSP with fixed-point arithmetic, the additional accumulator registers are joined.
- word length DSP with fixed-point designed for telecommunications uses 16-bit word length and processors intended for high-quality 24-bit word length audio applications. DSP with floating point arithmetic uses the 32-bit word length.

In standard microprocessors, based on Von Neumann architecture, operations are executed sequentially, which commonly results in data flow congestion, as shown in Fig. 1. When aspiring for a faster processor and faster execution of the mathematical instructions in digital signal processing, it is necessary to separate the buses, i.e. use dual bus architecture (separate memory for data and memory for program instructions). This concept of processor is called Harvard architecture, and it is used in most of the modern DSPs; it is presented in Fig. 1, [5]–[7]. The use of two separate memory buses assures simultaneous data and instruction flow and provides the ability for fetching more options in every instruction cycle, [8].

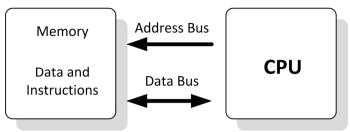


Figure. 1. Von Neumann processor architecture

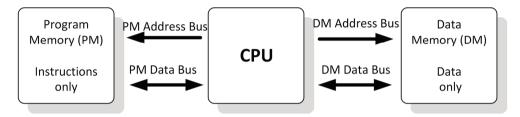


Figure 1: Harvard processor architecture

The DSP processor consumes most of the loop execution time in the algorithms, so it has a built-in CPU instruction cache that can store the 32 most commonly used programming instructions. This processor concept is called Super Harvard Architecture (SCHARC) (presented in Fig. 2) designed by engineers of the Analog Devices company, which unified the enhanced DSP under the name SHARC®DSP. To accelerate the information flow, they have connected it to the data memory I/O controller, which provides high-speed parallel and serial communications ports, [5].

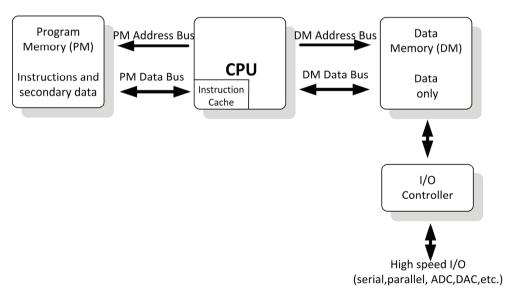


Figure 2: Super Harvard processor architecture

A specific feature of the Harvard architecture is the instruction overlap, i.e. instruction pipelining which allows the CPU to execute all execution steps (fetch, decode, execute) in parallel, [6]. The ability for instruction pipelining (presented in Fig. 3) is a significant element for achieving high processor performances in digital signal processing, [2].

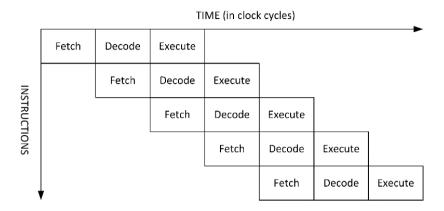


Figure 3: Instruction pipelining

The number of levels of parallel instruction execution differs from processor to processor: as the number of levels is higher, the performances of the processor are better, i.e. studying the parallel instruction execution leads to reduced average execution time of the instructions. Aiming to enhance the memory and speed memory access in one instruction cycle, various producers have modified the Harvard processor architecture in different ways, [2].

For DSP performance improvement, two approaches of parallel processing were developed: VLIW (Very Long Instruction Word) and SIMD (Single-Input Multiple-Data). The VLIW processor architecture is suitable for numerically demanding algorithms due to embedded multiple units for the parallel execution of instructions in one cycle. More details about parallel processing can be found in [2], [9], [10]. The SIMD processor architecture is used in operations of big data groups, e.g. matrix operations, image processing, graphics, simulations, numerical analysis, etc., [2].

3 MOST COMMONLY USED APPLICATIONS AND ALGORITHMS

It is important to consider the requirements of the applications that would be executed on the desired DSP. Dominant producers in sales and the development of the DSPs are presented in Table 1 with a list of applications and algorithms from literature:

Table 1: An overview of DSP applications and algorithms

Producers	DSPs	APPLICATIONS AND ALGORITHMS		
Analog Devices	ADSP-21xx 16bit, fixed point; 32bit, floating and fixed point	wideband sinusoidal (WS) speech, [11], Dual Tone Multi-Frequency (DTMF) signals, [12]; image processing and resilient propagation algorithms, [13]; intravascular ultrasound, [14], active power filter, [15]; image reconstruction algorithms, [16]		
	Blackfin	Optimization of MP3 decoder, [17]; audio equalizer, [18], driver fatigue detection system, [19], [20]; fuzzy logic controller, [21], guitar effectors [22], H.264/AVC encoder, [23], graphic equalizer, [24]		
Lucent Technologies and AT&T	DSP16xx 16bit, fixed point; DSP32xx 32bit, floating point	multineuron recordings, [25] multi-channel dual-tone multiple frequency detection, [26]; digital lock in amplifier, [27]; matrix-pencil approach, [28]; noise cancellation, [29]; control of brushless DC (BLDC) drives, [30]		
Motorola	DSP561xx 16 bit, fixed point; DSP560xx 24 bit, fixed point; DSP653xx 24 bit, fixed point; DSP96002 32 bit, floating point	Extracting signal components, [31]; real-time speech compression, [32]		
	StarCore	Radix-4 FFT, [33]; least mean square adaptive filter algorithm, [34]; convolutional face finder algorithm (for teleconferencing, security access control, etc.), [35]		
Texas Instruments	TMS320Cxx 16 bit, fixed point; TMS320Cxx 32 bit, floating point	rapid prototyping, [36]; acoustic OFDM transmitter, [37]; voltage frequency control of induction motor drive, [38]; LISA models, [39]; active noise control, [40]; noise reduction in speech signals, [41]		
	TMS320LF	temperature humidity detection, [42]		

DSP is present in all areas where the information is processed in digital form or controlled using digital processors, some of which are shown in Table 2, [5], [43].

AREA	DSP algorithm	APPLICATION	
Communication	Speech coding/decoding; speech	Digital mobile telephony, [44];	
	encryption/decryption; speech	multimedia computers, secure	
	recognition; speech synthesis;	communications; satellite phones;	
	speaker identification; echo	robotics; automotive applications;	
	cancellation; data compression;	multimedia workstations;	
		speakerphones; modems;	
	Modem algorithms	Digital mobile telephony; digital audio	
		broadcast; digital television	
Consumer	Noise cancellation; audio	Consumer/professional audio; music;	
	equalization; ambient acoustics	multimedia computers, [45]; advanced	
	emulation; audio mixing and	user interfaces	
	editing; sound synthesis		
	Vision; image	Robotics; security; multimedia	
	compression/decompression;	computers; navigation; digital video	
	image compositing	[46]; digital photography; consumer	
		video; advanced user interfaces;	
Industrial,	Image processing, beamforming	Magnetic resonance imaging (MRI)[47];	
medicine and		ultrasound, [48]; CT; ECG, [49]; process	
military		monitoring and control, [50], [51];	
		vision systems, [52]; navigation;	
		radar/sonar. [53]: digital radio:	

Table 2: DSP fields of use and applications

Communication systems and audio application

O Adaptive echo and noise cancellation

Application for adaptive filtering, i.e. attenuation of undesired echo in a telecommunication network, provided by modelling the echo path using an adaptive filter and subtracting the echo path output approximation, [54].

o Digital mobile telephony

Digital signal processors embedded in mobile phones are used for signal and data processing (e.g. for speech coding, measuring consolidation of signals, voice mail, modulation and demodulation, etc.). Modern DSP chips are optimized for wireless communication, and they provide affordable and high-quality products, [55], [56].

o Digital television

Interactivity, internet access, shopping, recording shows for watching later, etc. are just some examples of what digital television provides to consumers. DSP plays a key role in the processing, coding/decoding and modulation/demodulation of video and audio signals. For example, compressed video and audio before transfer and perfect image and voice are impossible without DSP, [55].

o Digital audio adjustment of the voice

The major example of DSP application is the improvement of audio quality and its functionality. Audio adjustment of different voices is used in film, television and radio engineering to develop the sound background, [55].

o Creating artificial speech

With the development of semiconductor technology and digital signal processors, artificial voices have almost assumed the voice quality of real human speech (e.g. Speak and Spell, TI, 1982.), [55].

o Speech recognition

The speech recognition system is based on a training system for the recognition, digitization, and storage of every spoken word. The recognition step is based on the search for matching words for every spoken word which is digitized and saved in the base. The problem occurs when the system cannot recognize speech, e.g. due to the insufficient breaks between words, fast speaking, unclear word pronunciation or presence of background noises. To resolve these problems, DSP has two major operations: parameter insulation (in order to create a sample, a clean pattern is chosen from spoken word) and pattern matching (pattern is compared with patterns in memory), [55].

Biomedical applications

Most modern medical applications, such as electrocardiography (ECG), digital stethoscopes, pulse oximeters, etc., require DSP processing. One of the DSP processors appropriate for that application is Texas Instruments TMS320C5515, based on fixed-point arithmetic. Texas Instruments has developed an MDK (Medical Development Kit) based on the C5515 DSP processor that supports all developing medical applications, [57].

o Electrocardiography monitoring

Electrocardiography (ECG) is a procedure for data collection about the electrophysiology of the human heart. DSP is needed to read digital signals from an analogue-to-digital converter (ADC) over a serial peripheral interface (SPI), for noise reduction and for decoupling the key features of the ECG, [55].

Anaesthesia control

An automated closed control system with embedded DSP processor for separating signals which come from brain serves to control the anaesthetic in the patient's body and to monitor the patient's condition. DSP plays a key role in the separation of auditory evoked response (AER) from background EEG signal. AER is part of the EEG signal: a few times weaker, but a significant signal. AER is an electrical reaction of the brain to external sounds, so it is essential for a transition assessment from consciousness to unconsciousness when the patient is anaesthetized, [55]

Meteorology

DSP is used for temperature control of the sensor wire at constant temperature used in wind speed measuring instruments. DSP executes extra operations such as linearizing the output voltage of anemometers and controlling the user interface directly or using a control program on a master computer, [58].

4 PRACTICAL IMPLEMENTATION OF THE EVALUATION MODEL

The TMDXEVM8148 evaluation model is based on Texas Instruments processor DM814x/AM387x for developing applications sensitive to power supply, consumer, and medical video applications which require less video streaming, [59]. The digital media processor (DM8148) provides fast and high-quality creation of unique applications such as video security, video conferencing, navigation, advanced portable consumer electronic devices with high end gaming support, digital signage, smart home controller applications, etc. The evaluation model has two processors: master ARM Cortex-A8 processor, which goes up to 1 GHz, and slave processor TI C674x VLIW DSP which goes up to 750 MHz, [59], [60].

EVM works with GStreamer, which helps in creating programs for parallel execution and creates different multimedia applications: streaming, video editing, etc. The C6Accel API allows the memory share between DSP and ARM, i.e. parallel working. Generally, it is used for easy intercommunication between the ARM and DSP. The C674x processor architecture contains a bi-level internal core, the cache memory with the support of external memory. On the first level, the memory is divided into L1P (software cache) and the L1D (data cache). If the requested information is not contained in the cache memory, it is then retrieved from the next lower program levels: L2 or external memory, [10]. The architecture of the cache processor C674x is shown in Fig. 4. L1PChronic and L1D are built into the SRAM cache to 32 KB. All memory and data paths are controlled by the cache memory controller, [61].

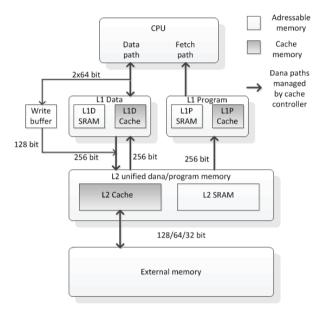


Figure 4: Architecture of the cache processor C674x

The registers ensure the control setting mode and control various processor operations. Interrupt Controller (INTC) is responsible for the control of the interruption in the program and management of the CPU. More details about execution time comparison can be found in, [62].

Let us consider a numerical example. An executing program is given for the Monte Carlo method used in numerical integration functions executed in an integrated development environment (IDE) of the Code Composer Studio (CCSv5) supported by Texas Instruments microcontroller and embedded processors. Execution time of the loop on the digital signal processor without the level of optimization is 3.1373×10^{-306} seconds.

If the optimization level is set, the execution time of the loop of the numerical integration with the Monte Carlo method is $1.112554 \times 10^{-308}$ seconds. For comparison, the execution time of the loop of the numerical integration with the Monte Carlo method on AMD Dual-Core 2.30 GHz processor is 8.78 seconds.

The main problem for an image-and-video processing system is the time of algorithm execution. Different methods for minimization operations and memory access use a different algorithm in every loop iteration, and most of the methods for execution time minimization are based on a pipeline.

Results and execution time comparisons of the image processing from a camera in different stages are presented in Table 3. It can be concluded that the digital signal processor is a better choice for image processing in comparison with other processors regarding the execution time. More examples of execution time comparisons can be found in [63].

Table 3: Execution time for various functions using different processors

Function	Matlab (ms)	ARM (ms)	DSP (ms)
Transform	1536	35.41	36.2
Gaussian filter	252	5.8	3.1
Horizontal interpolation	621.20	6.9	4.7
DX filter	920.3	5.4	0.2

5 DISCUSSION

Due to increasing demand for better performance of processing, there are possibilities for improving performance in clock rate, data and instruction level parallelism, decreasing the switching time of the device, etc., [64]. Owing to the demand for different multiple applications and the possibilities of running multiple tasks, high-performance processors have been developed. Classification of the microprocessors is presented in Figure 5.

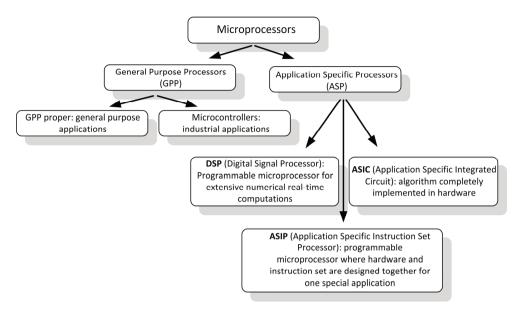


Figure 5: Classification of microprocessors

General purpose processors (GPP) are used in CPUs for PCs and workstations, and have a general purpose. DSPs are microprocessors specialized for signal processing applications and embedded in mobile devices in order to optimise performance and energy consumption, [65]. Nowadays, multi-core processors in PCs use parallel running instructions, and are based on shared or distributed cache memory and can execute up to four instructions per cycle, while high-performance DSP can execute up to eight instructions per cycle, [66]. GPPs generally have Single Instruction Multiple Data (SIMD) architecture to improve their performance in data processing, [67], while DSP has very long instruction word (VLIW) or SIMD operations to improve their performance, as mentioned above.

6 CONCLUSION

Digital signal processors have been undergoing massive development in the last ten years, and they are embedded in different devices (from cell phones to advanced scientific devices).

The particularity of the DSP architecture enables the development of fast and efficient applications in all areas of human activity. Due to the basic architecture of the processor regarding the data collection, data processing and transmission, the DSP achieves its maximum in millions of instructions per second. Although developers of the GPP have increased its performance, the GPP with SIMD has the ability to compute intermediate complex instructions only. Furthermore, GPP includes DSP instructions and implements DSP algorithms but it still often provides only partial solutions, [67], [68].

Practical results, as described in Section 4, show the great advantages of the DSP in comparison with commonly used processors regarding the execution time of the numerical integration.

This paper gives a review of the basic architecture of DSP and the diversity of its application. Digital signal processors may be of great interest to developers who work on application development in these, or similar areas.

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