SigmaDSP processors for audio signal processing

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Abstract. In our everyday live, we constantly use devices that use digital signal processors for audio signals processing. The digital signal processing of audio signals have achieved dominance in many audio systems like ‘Hi-Fi’ music systems, radio devices, portable music players and mobile phones. Signal conversion to digital domain opens new possibilities of signal processing and algorithms that were impossible or very difficult to implement in analog domain. One representative of digital signal processors for audio applications is SigmaDSP processor (Analog Devices). The paper deals with the analysis of the SigmaDSP audio processors, their properties and outlines the possible applications.

Keywords: SigmaDSP, SigmaStudio, digital signal processing

1 Introduction

It was in the 1960s when a new discipline of signal processing began to form – digital signal processing. From this time and especially years between 1980 and 1990, a mass replacement of analog signal processing to digital signal processing occurred. For example, music was usually recorded and transmitted in analog form until the 1980s when the digital recording became a common way how to reproduce the music via the CD player.

Currently, the most common way how to handle the audio and speech is to use digital signal processing methods. However, before the digital signal processing methods are used, the signal must be converted into digital form by analog to digital converters. There are two very important advantages to digital signals. First, digital signals can be reproduced exactly – there is no noise caused by a physical media. Second, digital signals can be manipulated easily. Since the converted signal is just a sequence of zeros and ones, the DPS (Digital Signal Processor) and algorithms of the digital signal processing can be applied to audio and speech signals. In audio digital signal processing chain, the analog to digital conversion is done via audio codecs, which consists of the precise, high resolution ADC (analogue to digital converter) and DAC (digital to analogue converter). The audio codec can form an internal part of the DSP or can be implemented as independent part of the digital signal processing chain [4, 5]. The second option is most suitable for most of the applications, which require executing demanding signal processing algorithms in the real-time. However, there
are also applications like automotive and portable audio applications, where it is required to conduct less demanding digital signal processing algorithms like filtration, equalization, mixing, dynamic processing, multi band compression as well as basic low-level DSP functions. SigmaDSP digital audio processors are suitable to perform these algorithms and in the same time to meet the requirements for the low power, high resolution audio digital signal processing applications.

2 Digital signal processing of audio signals

The DSP serves for manipulation of signal converted to digital form. There are two types of the DPS applications – non real-time and real-time. Non real-time signal processing involves manipulating of signals that have already been stored and digitized. However, many audio applications require execution of the algorithms in real-time. For this reason, the hardware, software and instruction set of DSPs is optimized for high-speed numeric processing what is basis for real-time processing of digital data representing analog signals. There are many applications where the DSPs can be used and some of them are listed below:

- Audio signal processing (compression and decompression like MP3, speech recognition and generation, filtration and effects application – echo, reverb, etc.),
- Video signal processing (digital cameras, digital television, etc.),
- Automotive (security and safety systems),
- Medicine (tomography, magnetic resonance imaging, ultrasound, etc.),
- Military (radars, sonars, navigation systems, etc.),
- Telecommunications (multiplexing, compression, echo control, etc.).

DSP processes data in real time, therefore is ideal for applications where there is a null tolerance for delay. DSP manipulates with signals, which were digitized by external ADC or embedded ADC in advance, by using mathematical functions. Among the basic operations, which can be applied to audio signal processing are so called low level operations (absolute value, logical operators, delay functions, etc.), filtering (IIR, FIR), mixing and dynamic processing. Examples of dynamic processing operations are:

- Midnight mode – loud sections of an audio signal are attenuated and very quiet sections are amplified into audibility level, while dialog remains unchanged and intelligible. This permits higher overall loudness with better clarity for the intended active listeners, especially in television applications, with much less chance of disturbing nearby non-listeners,
- Dynamic Bass boost block provides input level-dependent bass boost. Lower level signals are boosted more than higher-amplitude signals. Using a variable-Q filter, this algorithm dynamically adjusts the amount of boost,
- Dolby Pro Logic – Transforms ordinary stereo content into rich, full-range surround sound and ensures that you hear all the subtleties and detail of the original content.
2.1 Analog to digital conversion

Generally, the digitizing of analog signal is accomplished in three steps - sampling, quantization and coding. During this process several factors can influence the quality of the representation of the audio signal in digital domain. There is a distortion that cannot be fully avoided, because it is fundamental part of the digitizing process – quantization noise and aliasing. In the signal transfer path, there are electronic components, that have not the same ability to transfer all frequencies equally, therefore other types of distortion arise – linear, non-linear and modulation distortion. Some parameters of analog-digital converters influence digitizing process more significantly and they change character of signal. Among these parameters, the Signal-to-Noise Ratio (SNR), the ratio of the amplitude of the desired signal to the amplitude of noise signals at a given point in time, is one of the most important parameters. In case of ideal ADC, only quantization distortion arises and SNR is given by the formula [5]

\[
\text{SNR}_{\text{dB(MAX)}} = 6.02 \cdot \text{N} + 1.76\text{dB}
\]

where N is number of bits of ADC. In a real ADC, other sources of noise influence the conversion process and subsequently, the effective resolution will be less than N-bits. Therefore, actual resolution of ADC can also be expressed by another parameter known as Effective Number of Bits (ENOB). It specifies the number of bits in the digitized signal above the noise floor. The lower SNR is, the lower number of bits determines signal value. The formula for calculation is as follow [5]

\[
\text{ENOB} = \frac{(\text{SNR} – 1.76\text{dB})}{6.02}
\]

Another important parameter is Full Scale Range (FSR) - the difference between the maximum and the minimum signal strength that ADC can process. For applications which demand high quality audio systems, it is necessity to use 24bit audio codecs and sampling frequency up to 96 kHz.

2.2 Digital signal processors specialized for audio signal processing

In case of configuration of system which consists from digital signal processor and audio codec, it is necessary to realize individual algorithms of digital signal processing in DSP. The typical example is the configuration of system with digital signal processor ADSP BF-533 Blackfin together with audio codec AD1836 [6]. Modern multimedia applications, audio systems in mobile devices but also applications like security monitoring systems require to realize not only basic algorithms of digital signal processing like filtering, FFT, compression and so on, but also algorithms of adaptive filtration. Digital signal processors with their internal architecture are adapted for realization of these algorithms (chained performing of instructions, FFT and video accelerators).

Along with standard DSP processors also DSP processors exist with specially modified architecture for maximizing of audio signal processing performance. Their task is digitizing of input signal, its transformation by using sequence of loaded
algorithms and its further modification back to analog form or forwarding modified signal for further processing.

In technical practice, there are some applications where main factor is compactness and low power consumption while keeping option to realize basic digital processing algorithms, echo and noise cancelation and distortion elimination. SigmaDSP audio processors are especially suited for this kind of applications. SigmaDSP processors are fully programmable, single-chip audio DSPs that are easily configurable through the SigmaStudio Graphical Development Tool, and they are ideal for automotive and portable audio products. SigmaDSP chips are available with integrated Sample Rate Converters, ADCs, DACs, output amplifiers and another internal circuitry.

Advantages of specialized DSP are:

- Low power consumption (starting from 7 mW), what makes them ideal for using in portable devices and automobiles,
- Number of build-in algorithms,
- Usage of special circuits – for example Safeload registers, which serve for changing parameters in real time without disturbing noises in output – without clipping and popping or Target/Slew RAM - a hardware-optimized function that allows volume or other parameter level changes to ramp to subsequent levels without audible clicks/pops,
- Possibility to reduce main processor utilization from performing calculations needed for audio signal processing.

2.3 SigmaDSP architecture

The basic SigmaDSP architecture comprises these components (Fig. 1.):

- Program RAM stores functionality and execution information, meaning it dictates the signal flow and chain of operations (+, -, x, /, etc.) that are entered for execution,
- Parameter RAM stores parameter information (manually entered or the result of a core operation),
- The data memory stores serial information, usually audio samples or the result of core operations. Each block of a signal flow consumes some small part of the data memory,
- Safeload registers help load parameters smoothly. The software safeload mechanism enables the SigmaDSP core to load new parameters into RAM while guaranteeing that the parameters are not in use. This prevents an undesirable condition where an instruction could execute with a mix of old and new parameters,
- Target/Slew RAM is a hardware-optimized function that allows volume or other parameter level changes to ramp to subsequent levels without audible clicks/pops,
- Depending on the processor there are also additional registers: data capture; DSP core control; and serial input and output control.
Analog Devices has currently 3 generations of SigmaDSP processors:
- AD1940/AD1941
- ADAU1701/ADAU1702/ADAU1401/ADAU1401A
- ADAU1761/ADAU1781/ADAU1442/ADAU1445/ADAU1446

Each of those products has slightly different DSP core architecture, so there is a minor difference in the amount of processing cycles required to run a given algorithm on each of them. The ADAU1446 can compute 3584 instructions per sample, whereas the ADAU1701 and ADAU1761 can only compute 1024 instructions per sample. The downside of using the ADAU1446 is that it does not have internal audio ADCs or DACs. ADAU1761 has the lowest power consumption among these processors. Choosing the best processor is always based on requirements for the current project. The SigmaDSP processors up to the current generation can realize stream-oriented processing, known as sample-by-sample processing. This means that a sample comes in and comes out the DSP core at rate equal to the sample rate, and these samples are processed sequentially. For this reason, there is no option to compute tasks which require block oriented signal processing – for example FFT and DFT (MP3, WAV/AAC compression/decompression).

2.4 SigmaDSP booting

The ADAU1701, ADAU1702, ADAU1401, ADAU1442, ADAU1445, and ADAU1446 support self-booting from an external EEPROM. Typical architecture for self-boot applications is shown in fig. 2.
At power-up sequence, these ICs can load their program, parameters and register settings from an external EEPROM without a need of microcontroller. All other SigmaDSP ICs require a microcontroller for self-booting.

2.5 SigmaDSP ADAU1761

One member of SigmaDSP products family is ADAU1761. Its core offers 28-bit processing (56-bit double precision), 50 MIPS, it has internal 24-bits Σ-Δ stereo audio converter with sampling frequencies ranging from 8 kHz to 96 kHz, inputs for digital and analog microphones, I2C and SPI interfaces. Functional block diagram of the ADAU1761 is shown in fig. 3.
The record signal path includes very flexible input configurations that can accept differential and single-ended analog microphone inputs as well as a digital microphone input. Each input signal has its own programmable gain amplifier (PGA) for volume adjustment and can be routed directly to the playback path output mixers, bypassing the ADCs. An automatic level control (ALC) can also be implemented to keep the recording volume constant.

The playback path allows input signals and DAC outputs to be mixed into various output configurations. Headphone drivers are available for a stereo headphone output and other output pins are capable of differentially driving an earpiece speaker. Capless headphone outputs are possible with the use of the mono output as a virtual ground connection. The stereo line outputs can be used as either single-ended or differential outputs and as an optional mix-down mono output. Basic specification of the SigmaDSP ADAU1761 is as follows:

- Core Frequency: 50MHz
- Instructions/Sample: 1024
- Program RAM: 1kWORD
- Data RAM: 4kWORD
- Digital I/O Channels: 8
- Analog I/O Channels: 2
- GPIOS: 4

Analog Devices provides the evaluation board EVAL-ADAU1761Z (fig. 4.), that can be connected directly to USB port of the computer via USBi control interface.
board. This allows simple and fast testing and prototyping of designed audio algorithms and also special audio applications like microphone array beamforming.

Fig. 4. The ADAU1761 SigmaDSP evaluation board [1]

2.6 SigmaStudio

The SigmaStudio graphical development tool is the programming, development, and tuning software for the SigmaDSP audio processors and in the latest version, some types of SHARC digital signal processors are also supported. Audio processing blocks can be interconnected in the application layout and in compile phase, the compiler generates DSP-ready code and control layer for setting and tuning parameters. The typical graphic user interface of the SigmaStudio is shown in fig. 5.
For this purpose SigmaStudio includes an extensive library of algorithms to perform audio processing such as filtering, mixing and dynamics processing, as well as basic low-level DSP functions and control blocks. Because there is no C compiler, it is not possible to program own algorithms in C or assembler. If the required algorithm is not present, it is necessary to compose it from already existing algorithms. Depending on architecture and memory capacity not all existing algorithms have to be available and also number of algorithms is limited with hardware capabilities.

3. Conclusion

With increasing number of mobile phones and portable audio devices, digital signal processors with integrated audio codecs obtain more and more important position in audio signal processing. In case that low power consumption, high performance and only stream oriented sample processing is required, SigmaDSP processors are right candidates for these application. SigmaDSP enable anyone to use a fully programmable audio DSP that is easily configurable through the SigmaStudio graphical development tool. Some are designed specifically for the automotive market and specified over an extended temperature range. The SigmaStudio graphical
development tool is the programming, development, and tuning software for the SigmaDSP audio processors. This tool allows engineers with no DSP code writing experience to easily implement a DSP into their design and yet is still powerful enough to satisfy the demands of experienced DSP designers. SigmaStudio links with both Analog Devices evaluation boards and production designs to provide full in-circuit real-time IC control.

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