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FEATURING: DSP decodes multi-channel audio

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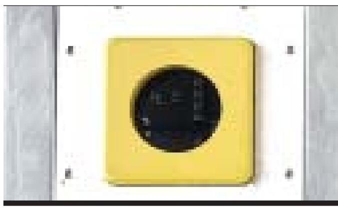


FEATURE

Advanced DSP supports home theater and surround-sound audio

by Paul Wheeler, *Engineering Manager, Analog Devices K.K. (Japan)*
and Sharon St. Ours, *Program Manager, Analog Devices (USA)*

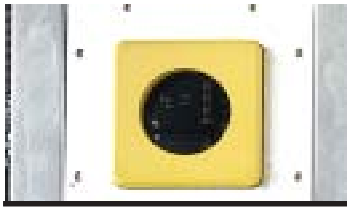
All home theater systems, from the most exotic to the most inexpensive, depend on some form of digital signal processing to increase fidelity, expand the number of audio channels, and enhance the "you-are-there" experience of DVD movies and audio. Some of these processing tasks involve front-end detection; others involve post-processing after the channels have been separated. Analog Devices' engineers explain some of the decoding algorithms the processor must handle.

"With such audio sources as PCM, SACD, DVD Audio, DTS, AC3 and post processing algorithms such as DPLII, Neo6, Surround EX, ES Matrix, Bass, and delay, the DSP must support a myriad of combinations."

Home theater systems range in features and in price. The basic system is the "Home Theater in a Box" which is available from most electronics distributors and resellers. The system will generally consist of a home theater receiver, surround speakers (usually four), a center channel speaker and a powered subwoofer. Often a DVD player is also included as well as a remote control. The DVD player will often accommodate newer formats like DVD-Audio or SACD (Super Audio Compact Disc), and will output a variety of video formats such as component video, S-Video or progressive scan for high definition television.

For the audiophile or theater buff, a higher-end system is in order. The television will be upgraded to a plasma screen and the audio components will be selected for their enhanced features. Certainly a high-end audio home system will include seven speakers and a subwoofer to take advantage of the full 7.1 surround formats available today. The DVD player will be able to play multiple formats automatically such as DVD-Audio, DVD-Video, DVD-R, Video CD, CD-R/RW media and MP3 discs.

The receiver, or AVR, is the heart of the high-end home theater system. This component will handle the audio decode algorithms like Dolby Digital and DTS. In addition the receiver will reproduce the sound effects of a dubbing stage for the "true" theater experience in your home. The audio decoders include virtual surround, which is a multi-channel effect from just two speakers, stereo to multi-channel decode, multi-channel surround using all six or eight speakers, or discrete surround decoding. The AVR component will also handle multiple zones and an array of inputs and



outputs including coaxial and optical.

With all the various combinations of decode formats, virtualization and post processing, this system has to be highly flexible. A high performance digital signal processor (DSP) is ideal for running the decoders and has the programmability and flexibility to perform the additional functions like automatic room equalization. Whether the signal processing is performed in the DVD component or the AVR system, the digital signal processor is the central processing engine of the entire system. With such audio sources as PCM, SACD, DVD Audio, DTS, AC3 and post processing algorithms such as DPLII, Neo6, Surround EX, ES Matrix, Bass, and delay, the processor must support a myriad of combinations while dynamically detecting changes in the input streams to invoke the correct decoding software. .

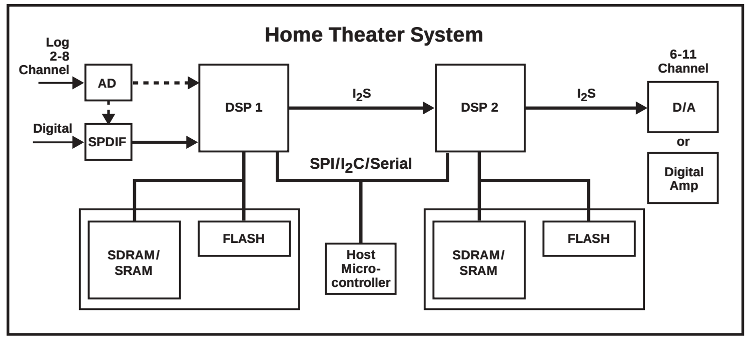


Figure 1

Figure 1: Components for a Mid- to High-end AVR system.

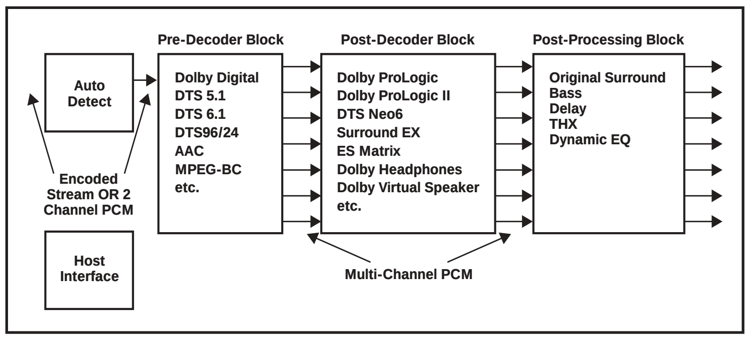


Figure 2

Figure 2: Audio DSP functional blocks include a source detector and a variety of decoders.

A Mid- to High-end AVR

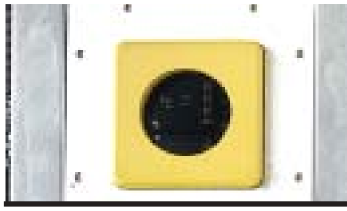
Figure 1 shows a basic block diagram for a mid to high-end home theater system that uses two 32-bit floating point digital signal processors. One signal processor may be sufficient depending on the individual system's processing requirements. For lower cost systems, the signal processor may run the various processing algorithms in internal ROM and RAM excluding the need for external memory.

To describe the types of processing that an audio signal processor must perform, we will first explore the audio sources that are available on the market today.

Automatic input detection

The audio DSP must be capable of recognizing the various formats in real time and invoking the suitable decoder codec as soon as possible. Any error in detection could result in noise at the output. This auto-detection process must run frame-by-frame in parallel with the decoding process. (The Analog Devices' SHARC signal processor, for example, makes use of a modified Harvard architecture to input samples in the background while decoding the previous frame.) The auto-detection process is based on the IEC61937 International standard for non-linear PCM encoded bit streams. This standard defines how encoded streams are sent to the DSP via the Sony-Philips Digital Interface Format (S/PDIF) receiver. The audio processor must determine if the input stream is an encoded stream (such as AC3, DTS) or is a PCM stream. The auto-detect module uses the IEC61937 standard as a base for this recognition in tandem with the stream characteristics of each format, which will be detailed in the next section.

One example is Dolby Digital (Dolby AC3), a multi-channel audio encoding technology developed by Dolby Labs, and publicly announced in mid 1991. This encoding mechanism is a "lossy" algorithm, which means that it takes advantage of the masking effects of the human ear to reduce the amount of information needed to convey a particular sound. This algorithm uses a "dynamic bit allocation" methodology. Frequencies that the



human ear are more sensitive to are allocated more bits to accurately reconstruct the original sound. The algorithm is capable of encoding from one (mono) to 5.1 audio signals, with the main signals supporting 20 Hz to 20KHz, and the ".1" channel supporting 3 to 120 Hz (LFE channel). The algorithm can support sampling frequencies of 32, 44.1 and 48kHz, and data rates from 32kbps to 640 kbps. (The basic Dolby Digital decoding method is shown in Figure 3.)

Figure 3: The Basic Dolby Digital (AC3) decoding process.

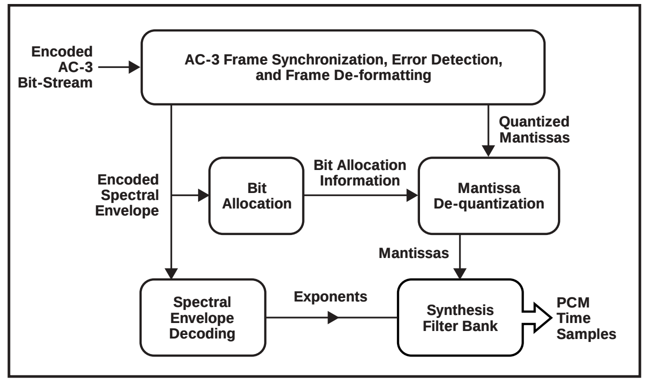


Figure 3

The Dolby Digital (AC3) payload stream is as defined in the ATSC standard for Digital Audio Compression and (is shown in Figure 4).

The frame contains 2 CRC (cyclical redundancy check) words that are used to detect errors in the frame. The frame also contains a BSI field that holds information about the sample rate, the data rate, the number of encoded channels, and the audio services that are available in the stream such as karaoke mode or dynamic range control. An auxiliary field is also reserved for control status. The payload frame contains six audio blocks, which each contain 256 audio samples, resulting in a fixed time of 1,536 audio samples. The actual length of the frame will vary according to sample rate and data rate.

Figure 4: The Payload frame format for a Dolby AC3 encoded stream includes synch and CRC, as well as six audio frames.

Sync	CRC1	Sync Info	BSI	Audio Block1	Audio Block2	Audio Block3	Audio Block4	Audio Block5	Audio Block6	Aux Data	CRC2

Figure 4

Decoding DTS

The DTS method of multi-channel encoding involves two basic processes, sub-band filtering and ADPCM (adaptive pulse code modulation). The encoder operates on various PCM frame sizes, with the longer frame sizes normally kept for low bit rate applications to maintain quality. The current supported frame sizes are 256, 512, 1,024, 2,048 and 4,096 samples. The main difference from the Dolby encoder is the provision for further extensions to the DTS format such as the recent additions to the encoder family of DTS 6.1 and DTS 96/24 (a 96-kHz sampling rate and 24-bit range).

DTS 96/24, the newest addition in the quest for high quality audio, uses the encoding method shown in Figure 5. The "core" signal, which is a 24bit, 48kHz, 5.1 signal is created by filtering the 96kHz signal to a 48kHz bandwidth and encoding using the original DTS encoder. This encoded "core" signal is then decoded to recreate a 5.1, 48kHz PCM signal. This reconstructed signal is then subtracted from the original 96kHz, 5.1 PCM signal to give the difference signal, which includes the coded error information from 0 to 48kHz and the original 48 to 96kHz high frequency informa-

Figure 5: High sampling rate audio encoding uses extension frames to affect a 96-kHz rate.

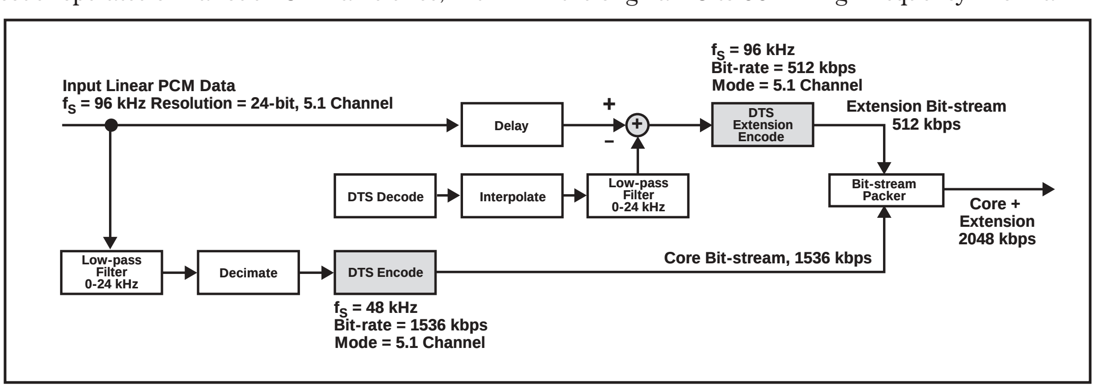
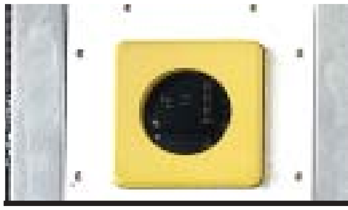


Figure 5



tion. This "difference" signal is then DTS encoded as an "extension" stream, and combined with the "core" encoded stream to give a DTS 96/24 frame.

The encoded frame for DTS is shown in Figure 6. The data frame is made up of five main parts: The synchronization word which defines the frame start; The frame header, which contains information about number of channels and bit rate. There are up to 16 sub-frames, which contain the "core" encoded audio data, an optional user data space and extension data. Extension data may contain the high frequency components (for DTS96/24) or the discrete Surround Back channel (for DTS6.1).

The decode method for this frame is shown in Figure 7. As can be seen, the audio decoder, depending on its processing power, can choose to ignore the extension stream and decode only the "core" audio. This backward compatibility means that older decoders can still effectively decode DVDs with extension stream inclusion without need of additional software. ■

Figure 6: The encoded frame for DTS includes a data frame made up of five main parts: The synchronization word which defines the frame start, the frame header, which contains information about number of channels, and bit rate, sub-frames, a user space, and extension data.

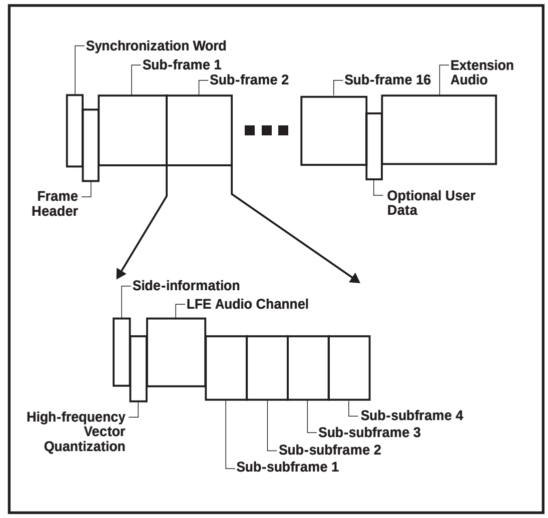


Figure 6

Figure 7: On DTS decode, the audio decoder can choose to ignore the extension stream and decode only the "core" audio.

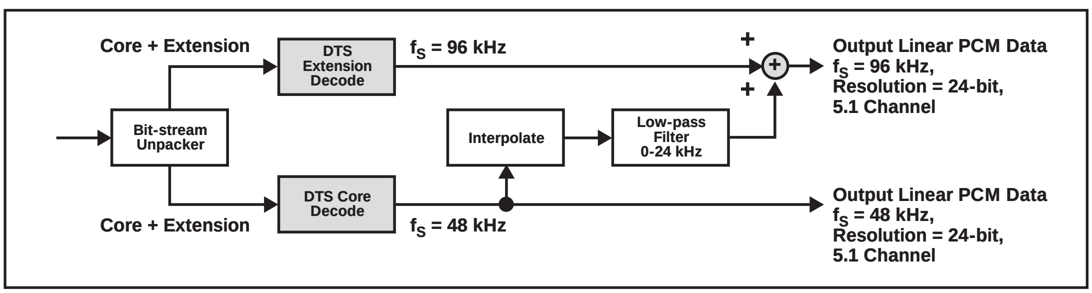


Figure 7

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