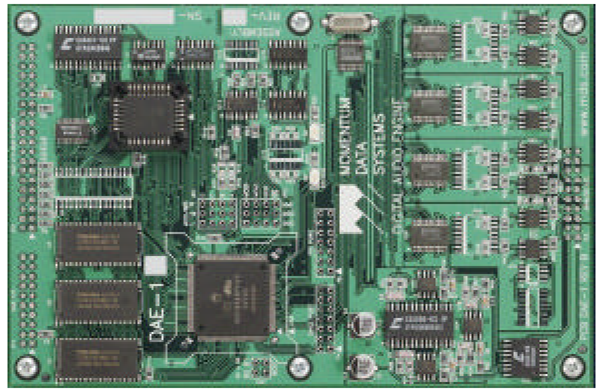
MOMENTUM
DATA SYSTEMS

17330 Brookhurst St., Suite 140, Fountain Valley, CA 92708
Phone: 714-378-5805 / Fax: 714-378-5985
email: dsp@mds.com / web URL: http://www.mds.com



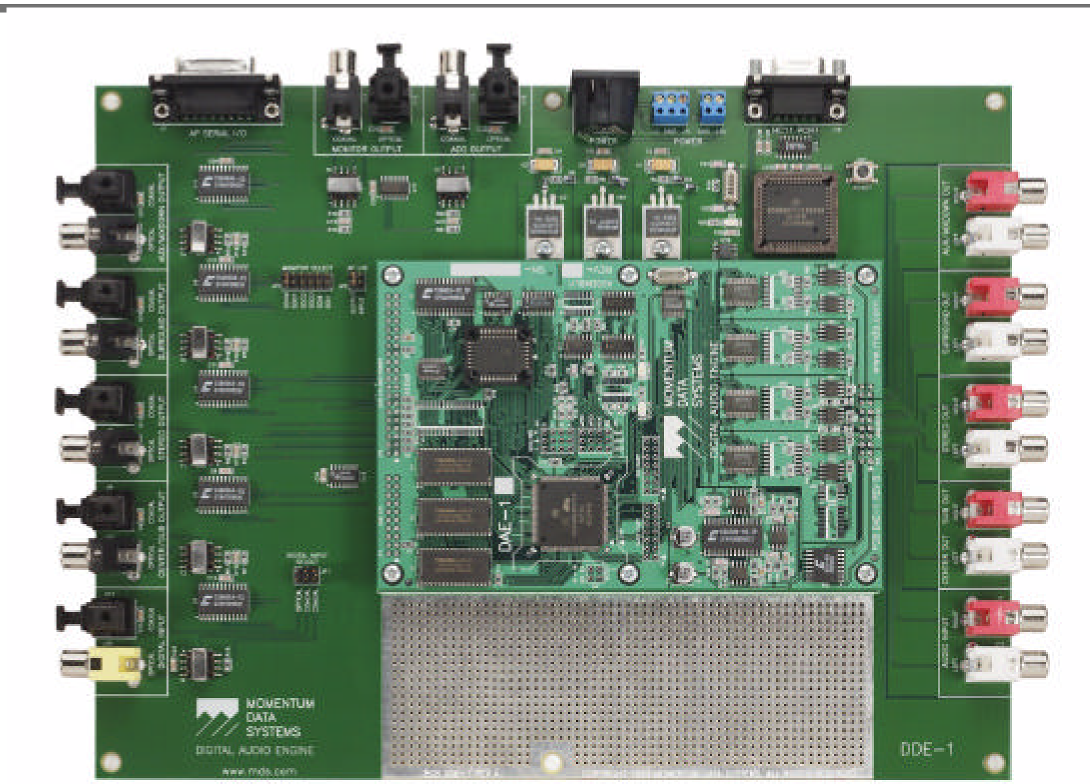
DAE overview

Digital Audio Engines are **'off the shelf' modules** providing all the functions for the heart of multichannel audio decoding. They can be used in surround sound audio or home theatre systems.

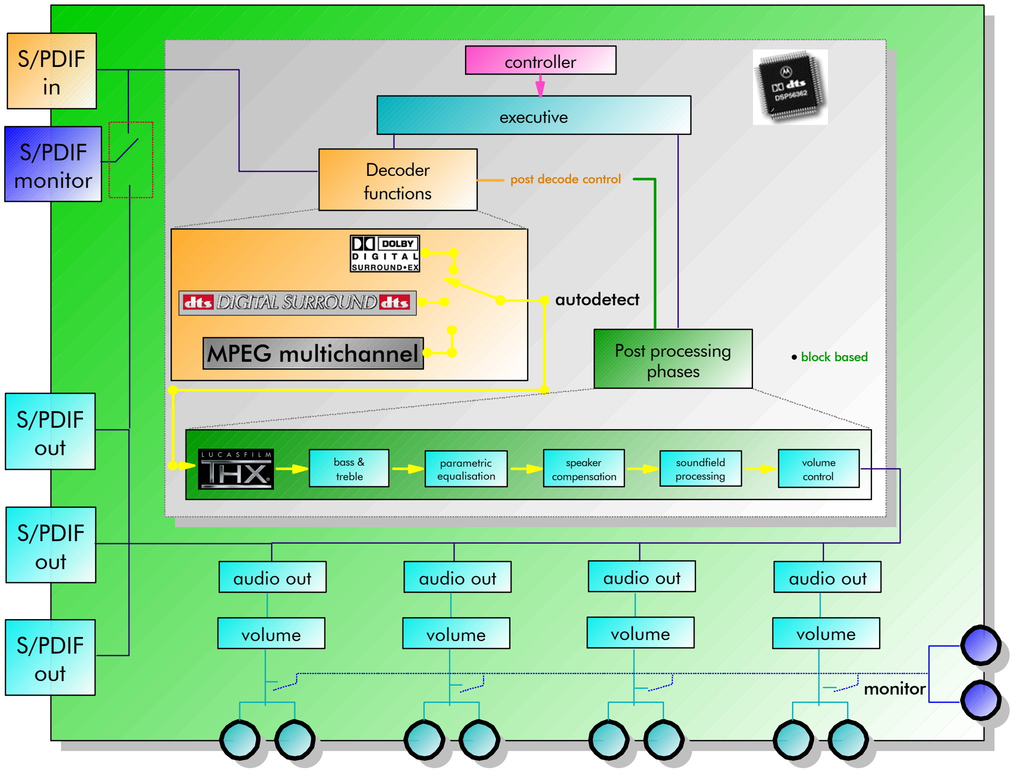
Digital Audio Engines include digital processing for all the main multichannel audio decoding formats, plus Post Processing Phases such as equalisation and sound fields, with the facility for manufacturers to add their own unique post processing features. They also provide **high quality analog output**, on the same compact module. 'Mixed signal' design allows a cost effective, compact final system but demands very careful attention to detail in the design to preserve the analog signal quality: Digital Audio Engines are specially designed to preserve the analog audio quality.

During development, manufacturers can use a Digital audio Development Environment (DDE) which includes a 'motherboard' with all the necessary test connectors; and the Momentum OEM Interface (MOEMI) which provides control over all aspects of the digital decoding and post processing phases so that it is easy to set up and evaluate a test system. The MOEMI software has a 'logging' function which allows the developer to capture the commands sent to the Digital Audio Engine and use these in their own software product.

The Digital audio Development Environment (DDE) comprises all the components - software and hardware - needed to evaluate the Digital Audio Engines and to develop, test and debug systems using them.



Digital Audio Engines



The [Digital Audio Engine](#) itself is a compact, 'off the shelf' module which includes all the digital processing needed fo rmultichannel audio decdoing, as well as the high quality analog audio reconstruction.

All Digital Audio Engines use as their processing core the Motorola DSP5636x processors. These provide the three main surround sound decoding formats - Dolby Digital, DTS and MPEG - as well as the post processing phases such as THX, equalisation and sound field processing. The Digital Audio Engine modules provide a quick and simple way to use the processing power of the DSP5636x in a ready made circuit board with all the necessary support logic.

Each member of the Digital Audio Engine family is aimed at different market needs, depending on the analog quality required, and the audio decoding methods supported.

Digital Audio Engine 1 (DAE-1) is the standard module. It has eight channels of analog audio output with 24 bit resolution using converters capable of 92 kHz sample rate. The THD+N of 96 dB is matched by the analog volume controls.

Digital Audio Engine 2 (DAE-2) is aimed at low cost systems. It has six channels of analog audio output with 24 bit resolution at 48 kHz sample rate. THD+N is 92 dB, and the volume control is performed digitally to keep component costs down. DAE-2 also uses the new Motorola DSP56366 processor which implements MP3 audio decoding.

Digital Audio Engine 3 (DAE-3) is aimed at high end systems. It has eight channels of analog audio output with 24 bit resolution using converters capable of 192 kHz sample rate. THD+N is 100 dB and is matched by the analog volume controls. DAE-3 also has eight channels of analog audio input, needed for DVD audio systems to preserve full signal quality in DSP 'bypass' mode.

Digital Audio Engine 4 (DAE-4) adds a second, user programmable, DSP56362 to the DAE-3 configuration.

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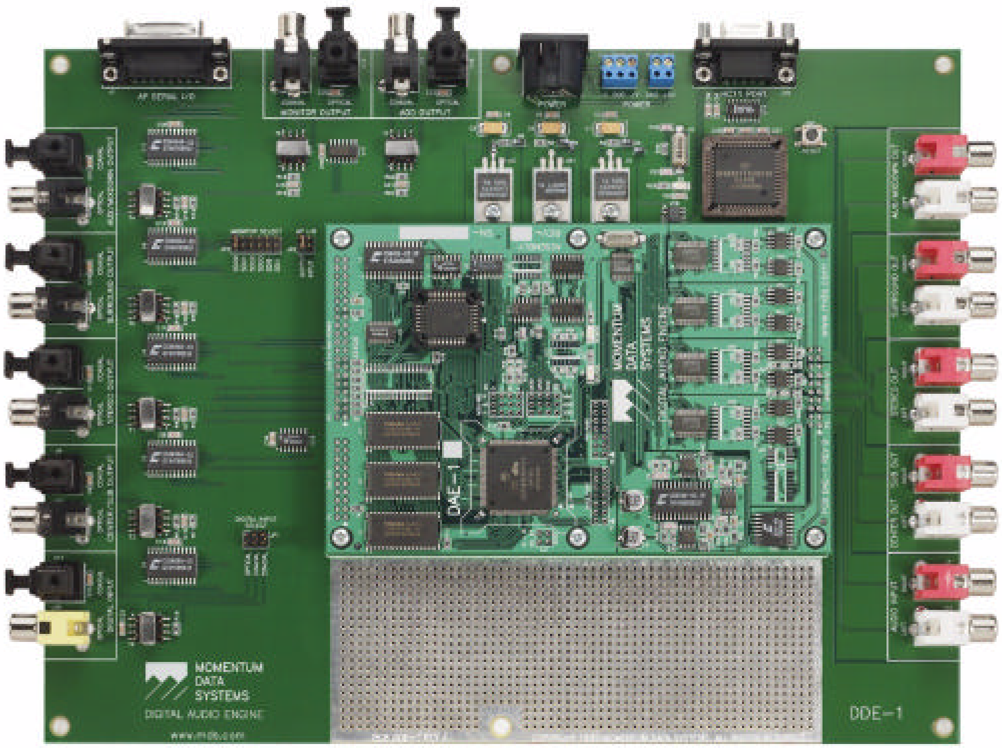
Digital audio Development Environment



The Digital audio Development Environment (DDE) provides a **ready made base** so that the Digital Audio Engine modules can be used immediately for **development and testing**.

The Digital audio Development Environment is a motherboard which accepts any of the Digital Audio Engine modules, and provides a **complete set of analog and digital audio connectors**, plus an RS232 connector for linking to a PC serial port, and a connection point for a power supply.

The DDE has eight analog output connectors, and two analog input connectors.

Although not normally used in a consumer system, the Digital Audio Engine does provide **four stereo digital audio output** channels - the DDE provides digital audio (S/PDIF) drivers and connectors so that the **digital audio output can be monitored** during development, bypassing the analog conversion.

A single stereo digital audio (S/PDIF) input driver and connector provides for the Digital Audio Engine to take S/PDIF input.

A pair of digital audio monitor connectors allow the incoming digital audio, or any one of the outgoing digital audio, channels to be monitored: the outgoing monitor channel is selected through a 'header' on the motherboard.

An RS232 interface allows a PC serial port to be connected to the Digital audio Development Environment so that the Digital Audio Engine module can be controlled from a PC - using the Momentum OEM Interface (MOEMI) or the developer's own program.

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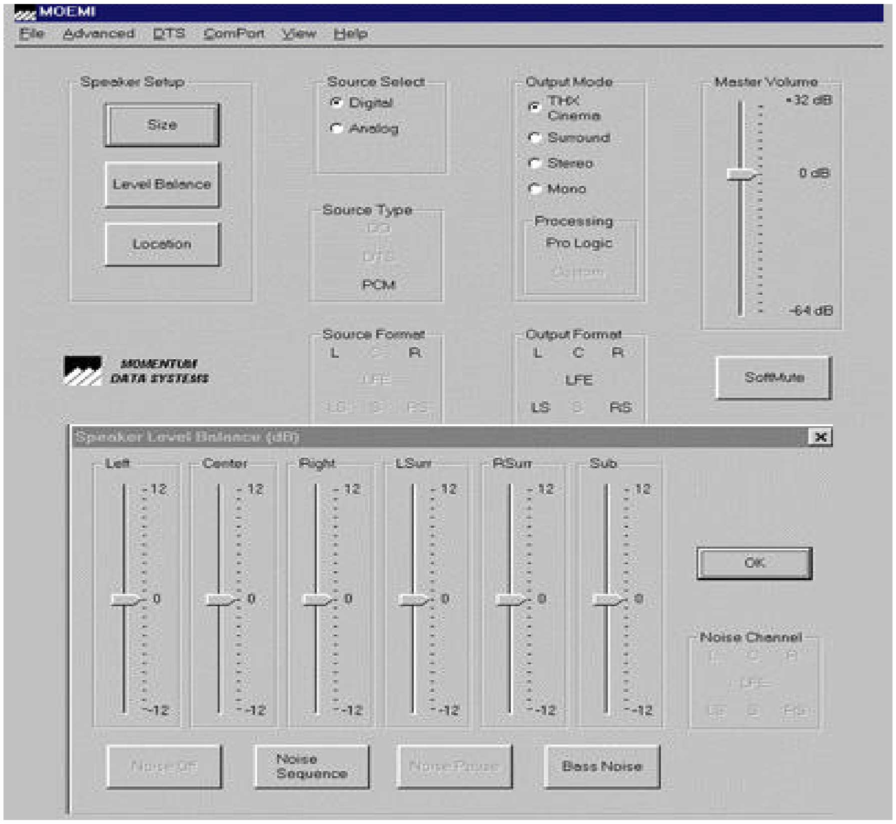
Momentum OEM Interface



The Momentum OEM Interface (MOEMI) provides a **ready made software interface** to control the Digital Audio Engine (via a Digital audio Development Environment (DDE) from a PC using the RS232 serial port.

MOEMI provides **complete control** over the audio decoding and the Post Processing Phases. It also shows the audio decoding status.

During development and testing, the combination of DDE and MOEMI can be used to build a prototype system quickly. The MOEMI software then allows the developer to experiment with settings for the audio decoding and the Post Processing Phases, finding the best settings and learning how to control them. A ‘command logging’ feature lets the commands that are sent to the DAE be captured and analysed, so that these can be built into the developer’s own system.

The example screen shot above shows the MOEMI screen for adapting audio to loudspeaker sizes and positions. The user can adjust for size of each pair of speakers separately, balance the levels, set the decoding for different numbers and combinations of speakers, and adjust delays to suit different speaker locations. MOEMI also provides a standard ‘noise’ output which can be applied to each speaker in turn to assist in setting the levels uniformly.

The example screen also shows the source material’s format and the audio decoding method that is being used, as well as the currently selected output format.

MOEMI allows complete control of all the Post Processing Phases in a similar way. It should not be thought of as a suggested ‘user interface’ but rather as a very useful ready made prototype which assists in development and testing, and serves to show all the functions which can be supported. A programmer’s library supports developers in building their own systems to make use of the MOEMI command set.

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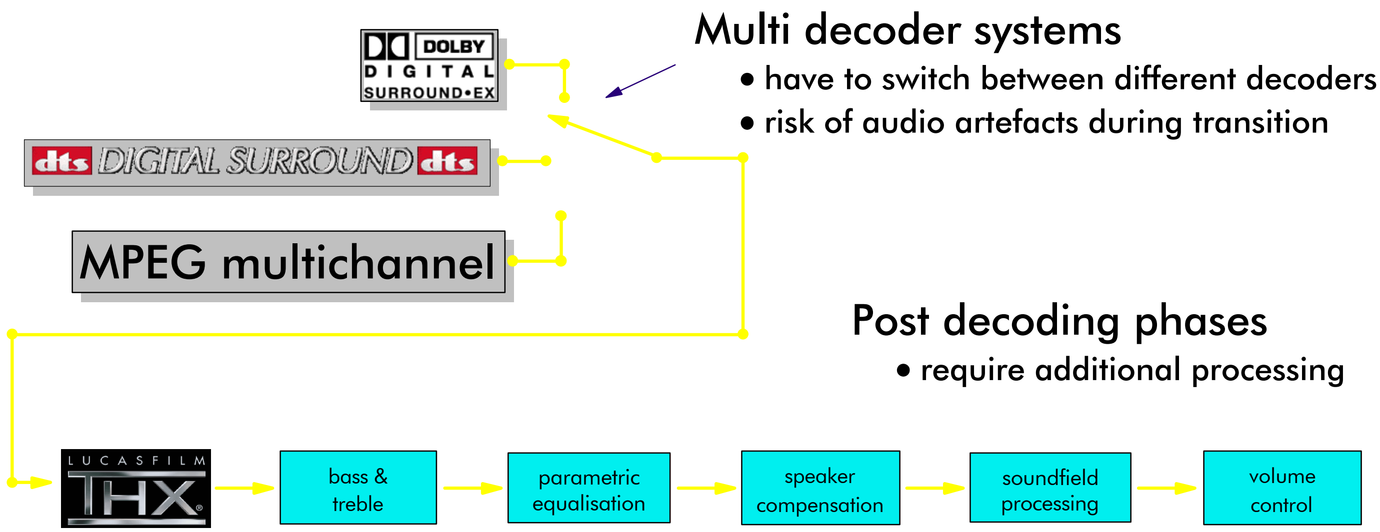
An adaptable digital audio module



Multichannel audio standards and products are evolving quickly: consumers and manufacturers alike want to products that support all the primary standards while at the same time allowing for new or custom affects. Multi decoder systems have to switch between the different decoders: when they do so there is a risk of audio switching artefacts so the switching has to be handled smoothly - this requires some processing power.

Post processing phases such as sub woofer management also need additional processing power - and these phases need to be switched in and out according to the user's requirements.

The Motorola DSP56362 is designed to handle all the major multichannel audio decoding standards (Dolby Digital, DTS and MPEG2) in a single device, while still having enough processing power left to deal with the post processing such as sub woofer management, sound field effects, 3D virtual surrounds, and Lucasfilm THXTM Cinema processing.

The device is programmable, with main functions in ROM, which gives a useful mix between fixed and programmable functions. Its processing power is sufficient to handle all stages of a digital audio system up to and including volume control. This makes it possible to design a low cost digital audio system on a single chip with the minimum of analog or external components.

Digital volume control keeps an audio system cheap, but has limitations. To allow for gain the average signal has to be kept low compared with the number of bits available to represent it: otherwise the amplified signals would be clipped when they reached the largest number that could be represented with the word length. This means the signals cannot use the full word length of the device, which is wasteful and increases quantization noise. A 24 bit device like the DSP56362 has a long enough word length to allow "headroom" without losing too much precision, but higher quality systems use the analog volume control.

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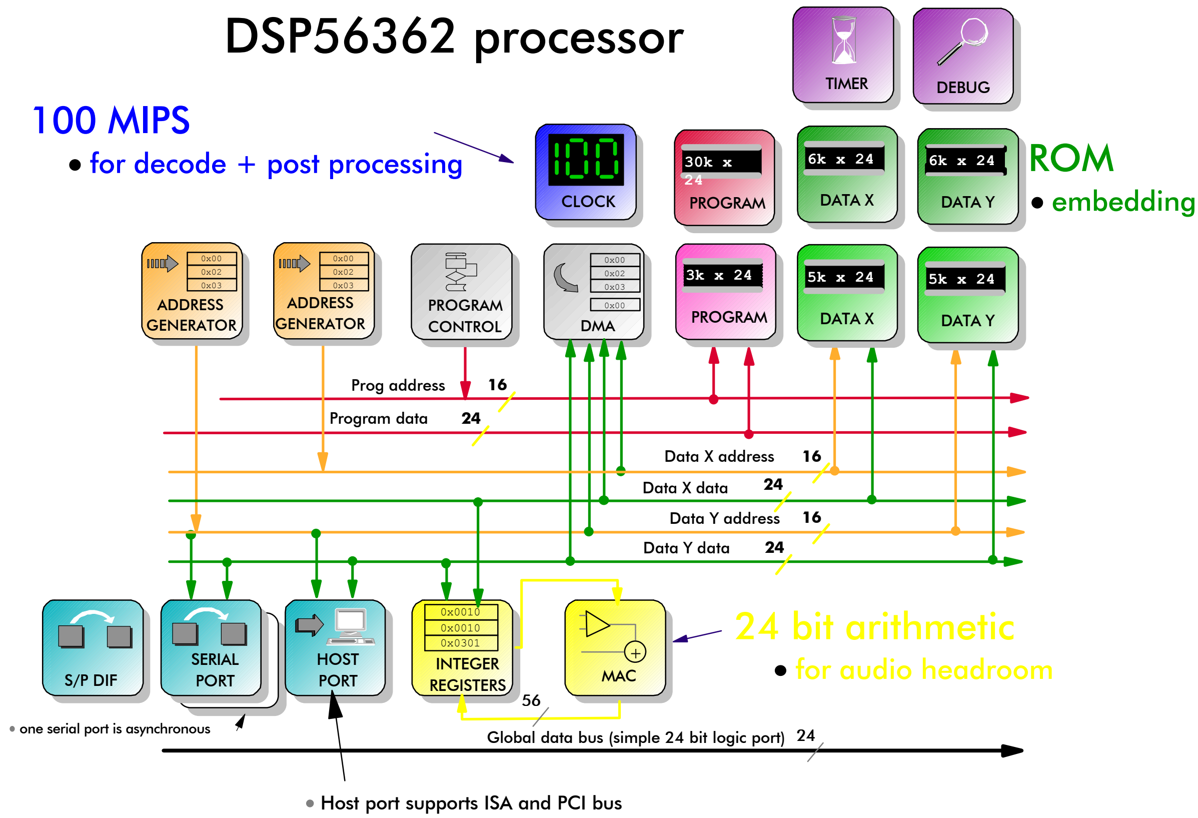

The DSP56362 uses Motorola's standard DSP5630x core, with audio I/O peripherals and a mix of ROM and RAM memory. The three main multichannel audio decoding standards (Dolby Digital, DTS and MPEG) require 24 bit precision and use about 50 million instructions per second (MIPS). The DSP5630x core is a 24 bit Digital Signal Processor which runs at 100 MIPS, so about 50 MIPS are available for post processing phases. Internal arithmetic uses 56 bit accumulators which preserve precision during calculations. A 56 bit barrel shifter allows the audio data to be scaled without overhead: this is an important feature for unpacking compressed bitstreams for multichannel audio decoding.

The external memory bus is 24 bits wide, so that external memory can be accessed at high speed: this is necessary, for example, when processing sound fields where recreating the target venue's ambience requires access to models in external memory.

The DSP56362 has I/O peripherals which are aimed at multichannel audio. The Enhanced Serial Audio Interface (ESAI) provides up to four receivers and six transmitters for 'glueless' interface to ADCs, DACs, and S/PDIF receivers and transmitters. The Digital Audio transmitter (DAX) supports many digital audio formats including IEC 6-1937 and AES/EBU formats required by applications such as DVD and HDTV. Having the digital audio transmitter on chip also helps to reduce overall system cost.

The Serial Host Interface (SHI) is used for external DSP control and system communications. There are additional control interfaces for boot operation, direct analog attenuation control, and input 'data throttling' for 'on demand' execution.

14 kword on chip RAM and 42 kword on chip ROM provide storage for most consumer audio applications. For applications such as sanctified processing, which require large amounts of memory, there is a glueless 24 bit interface to SRAM and DRAM, up to 16 Mword data spaces and one 16 Mword program space.

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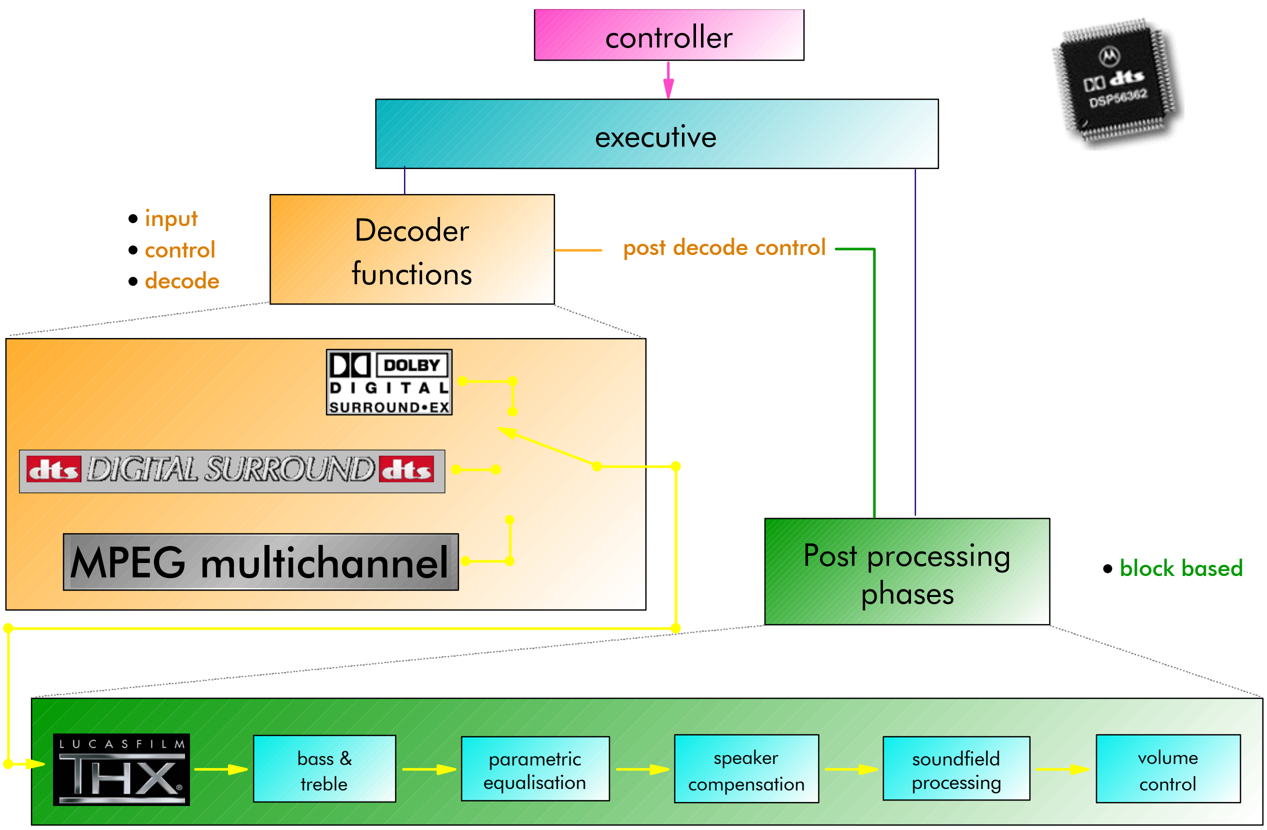
Software architecture



In addition to the hardware, the DSP56362 has a defined software architecture which aims to make it easier for programmers to control standard functions and to add new post processing phases.

A software 'executive' manages the sequence of operation of different software components, provides communication between components, and monitors processing which depends on time measurements. The executive coordinates four main types of components:

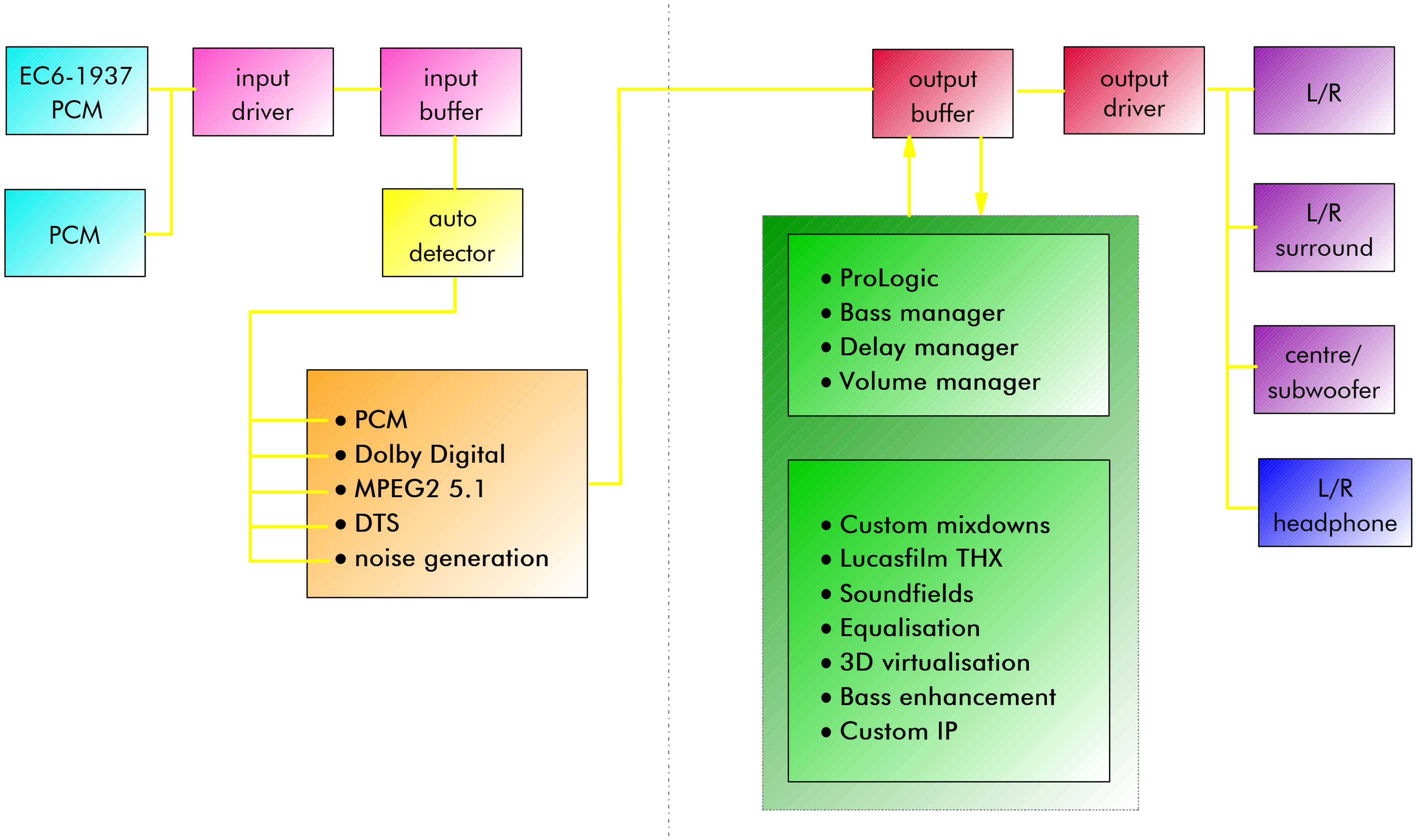
- multichannel decoders
- input and output drivers
- post processing phases (PPPs)
- external controllers

The decoder functions are coded into ROM: these are for Dolby Digital, DTS and MPEG2 multichannel audio. Different versions of the DSP56362 can have different combinations of decoders in ROM. The decoder components use about 50 MIPS, which is half the available processor power.

When switching between different decoder functions there is a risk of audio artefacts. The DSP56362 software architecture uses special filtering to smooth the transition between different decoding standards and eliminate these audio artefacts.

After decoding the digital audio data can be passed through sequential Post Processing Phases (PPPs). These can include ROM coded functions and programmable functions: for example bass and treble control, parametric equalization, speaker compensation, sound field processing and volume control are all Post Processing Phases. The Post Processing Phases are all block based: each component receives and processes a block of data which is then passed on to the next phase.

Data flow



The incoming multichannel audio bitstream can be in various formats. DVDs provide Dolby Digital, DTS and MPEG2 sound tracks: multichannel CDs may also use DTS or HDCD, and there may also be conventional PCM audio.

The bitstream enters the DSP56362 through the Enhanced Serial Audio Interface and is packaged into 24 bit words which are sent to the input buffer. Auto detection software recognizes the bitstream type and so determines the appropriate decoder to use. The decoder decompresses the data, decodes it and copies blocks of audio samples into the appropriate channel buffer - left, right, centre, and so on.

The software is capable of dealing with various audio standards including:

- PCM (conventional digital audio)
- Dolby Digital 5.1
- MPEG2 5.1
- DTS
- noise generation

Output from the Post processing Phases is then directed to the appropriate output peripheral left, right, centre, and so on.

The software architecture is provided in ROM, so the DSP simply boots up, configures itself and starts decoding audio bitstreams when it is powered up. DSP programmers can take this 'pass through' mode for granted and concentrate on developing specific Post Processing Phases. The software architecture also allows Post processing Phases to be downloaded in the background while the DSP is decoding audio - and the transitions from 'swapping' in or out a Post Processing Phase are handled elegantly with audio filters that eliminate artefacts. Similarly, the parameters of Post Processing Phases can be changed while running.

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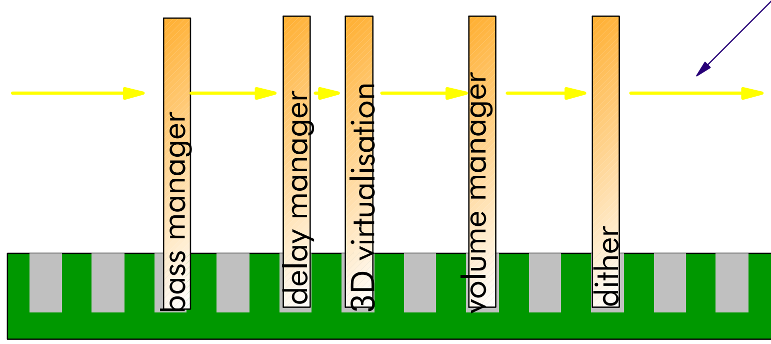
'slot' model of post processing phases (PPP's)

- like a card bus
- 25 slots available
- some reserved for Motorola supplied PPPs
  - ✓bass manager, delay manager, volume manager, dither



block data transfer

- modifying on chip data buffers
- allows re-use of memory

run time insertion

- without audio artefacts
- allows activation and de-activation of PPPs
- memory can be re-used

After decoding the executive invokes the Post Processing Phases. The Post Processing Phases (PPPs) use a 'slot' model. PPPs can be slotted in or out as required. This is a bit like a hardware bus model, except that data passes from one slot to another sequentially.

The software architecture provides 25 slots in which PPPs can be inserted. Some of the slots are occupied by standard Motorola PPPs that are in ROM - for example Dolby Prologic, sub woofer management, volume management, THX processing etc. The executive addresses each slot sequentially to determine whether the slot is active before it is invoked.

Processing from this point on is block based - each component works on a block of data which is fully processed before being passed on to the next component.

The decoded audio blocks contain information such as the type of decoder, buffer locations, sample rate, number of channels, and so on. This global information is available to all PPPs and to any external controller. Each individual PPP has an associated Post Processing Control block that contains parameters of local interest which convey any audio mode changes to the next PPP. For example data might enter a sound field PPP as 5.1 channel audio, but leave as only two channels for headphones: in this case the sound field PPP would use its local variables to tell the next PPP the current audio mode. This combination of local an global information permits the PPPs to be developed and used independently in a modular way.

Sometimes the order execution of PPPs will have auditory effects. For example, a 3D virtualization PPP may not work properly if it follows down mixing or speaker compensation.

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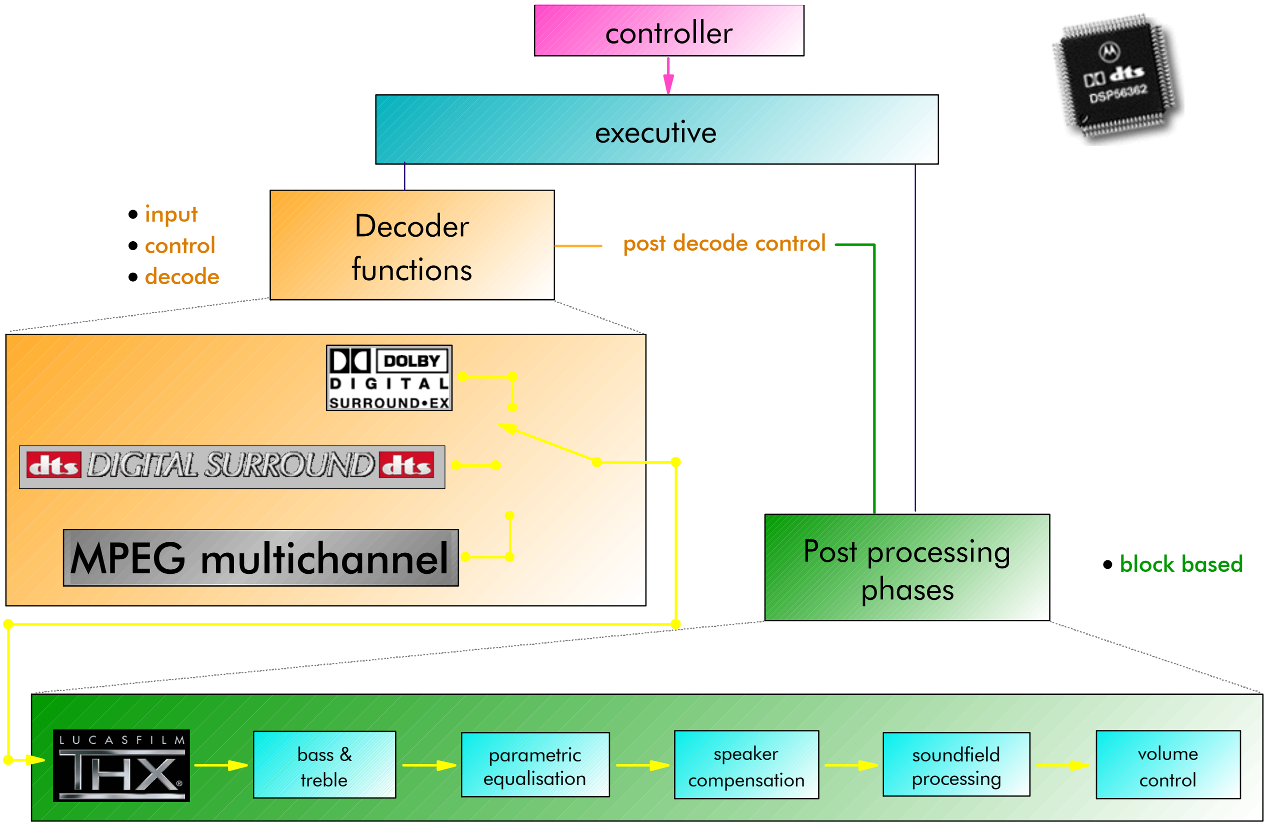
Post processing phases



The executive invokes each active PPP in turn. The PPPs process the data in place (that is, their output overwrites the input data buffer). After all PPPs have been executed, the channels are paired (left/right, centre/sub woofer, left and right surround) and transmitted by the Enhanced Serial Audio Interface to external DACs.

PPPs may be turned on, modified or bypassed in real time without having to reload the entire software chain: filters handle transitions elegantly without artefacts.

PPPs can be written to be position independent by using the 'relative addressing' feature of the DSP5630x core. This allows software to be reused in any configuration or with any number of PPPs.

The software architecture also supports auxiliary channels. For example a second listening room could be handled with a two channel down mix while the primary room continues to receive 5.1 channel surround sound.

An example of a problem that is easily solved because of the software architecture is volume management. When different decoders are used, and different combinations of Post Processing Phases, it can be hard to keep the final operating volume of the system constant. The software architecture provides volume management to solve this problem and allow the user to switch decoders and change post processing without having to keep adjusting the system volume.

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