TI Developer Conference
March 7-9, 2007 • Dallas, TX

Minds in Motion

MPEG Surround Audio Coding on TI Floating-Point DSPs

Fraunhofer Institut Integrierte Schaltungen

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Agenda

- The Fraunhofer Institute for Integrated Circuits
- The MPEG Surround Coding Algorithm
- Audio Coding for TI Floating-Point Processors
- Conclusions

The “Fraunhofer Gesellschaft“ - FhG

- Largest private research organization in Europe
- Non-profit organization, founded in 1949
- 56 Institutes in 40 locations in Germany
- Offices in Europe, USA and Asia
- Permanent staff 12,500, mainly scientists and engineers

Minds in Motion

Fraunhofer IIS
Institute for Integrated Circuits

"Home of mp3"

- Founded in 1985,
  staff ~450 in 4 locations
- Audio and Multimedia Cluster:
  - More than 100 engineers in Erlangen
  - MPEG Audio encoders and decoders
  - MPEG-4 Advanced Audio Coding (AAC, HE-AAC v2)
  - MPEG Layer-3 (mp3)
  - MPEG-4 Video Coding, AV Streaming, Virtual Acoustics
  - Digital Rights Management

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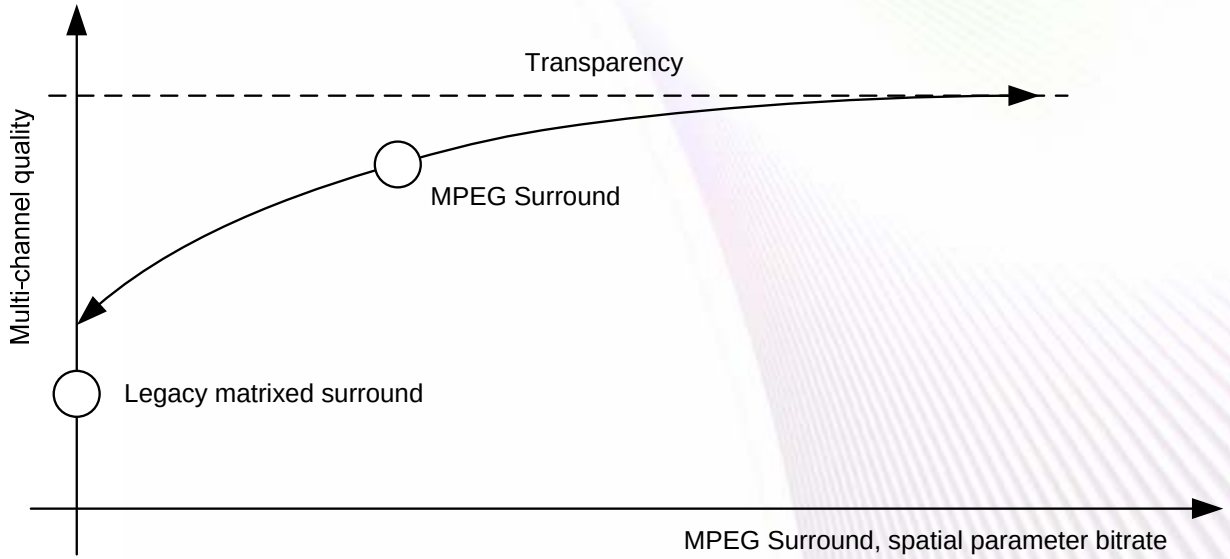
The MPEG Surround Coding Algorithm

- Overview
- MPEG Standardization
- Applications
- Technical Description
- Features and Modes
- Profiles and Levels
- Summary

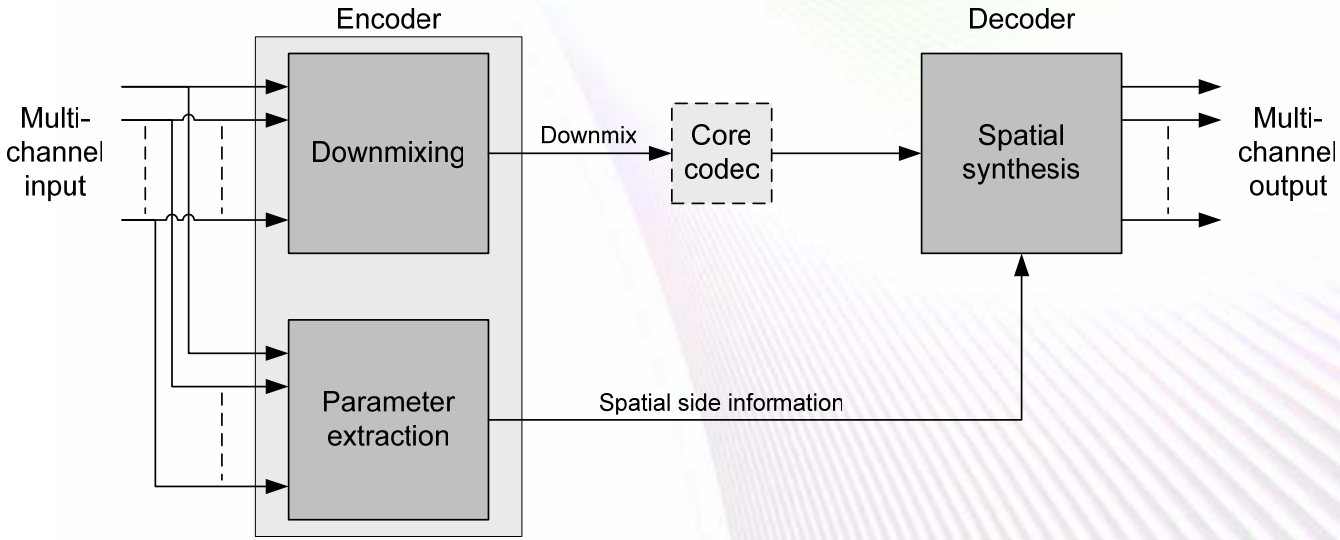
MPEG Surround Overview

- Efficient high-quality multichannel audio coding
- Backwards compatible with existing mono/stereo systems
- Significantly better quality than matrixed surround systems
- Highly scalable from lowest bitrates to highest quality



Spatial Audio Coding

- Efficient and backwards compatible coding of multichannel audio
- Transmission of downmix and compact side information



MPEG Standardization

- ISO/MPEG Call for Proposals on ‘Spatial Audio Coding’ (March 2004)
- Proposals of **Fraunhofer/Agere** and **Coding Technologies/Philips** showed best and complementary performance
- Reference Model Zero: Combination of both systems (verification of merge April 2005)
- International Standard January 2007

Applications

Attractive for upgrading existing mono/stereo distribution systems to surround

- Digital Audio Broadcasting
  - DAB Surround, HD Radio
- Digital Video Broadcasting
  - ISDB, DVB-T, DVB-H, DMB
- Internet Streaming/Download
  - IPTV / Video-on-Demand

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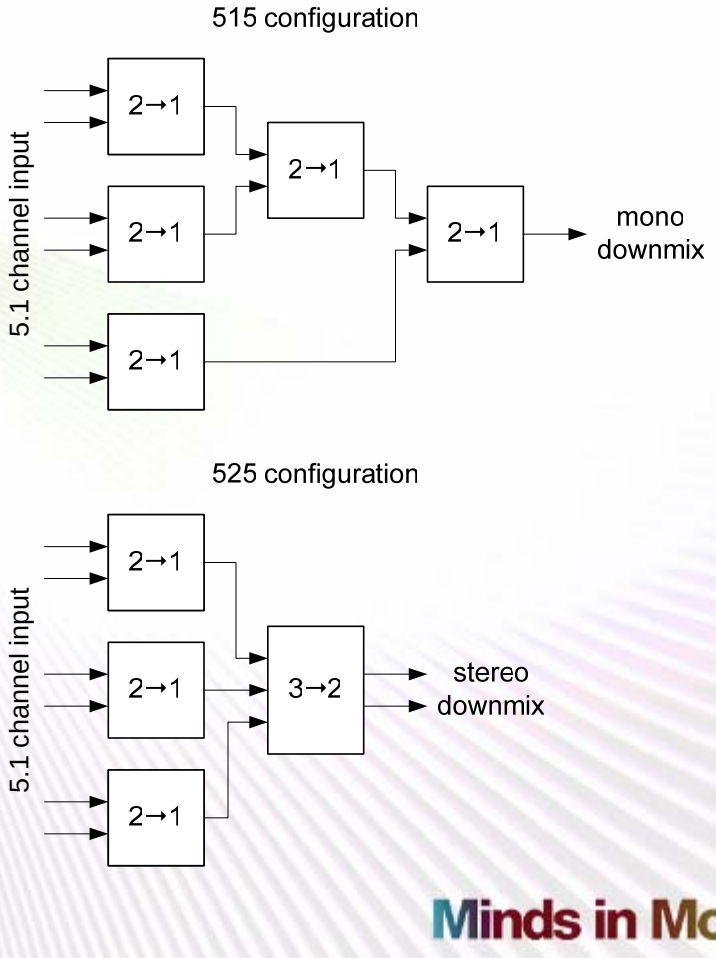
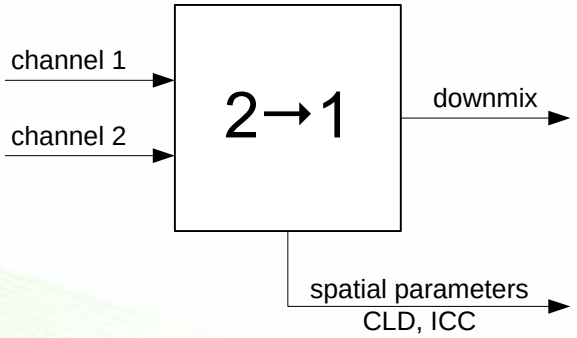
Encoder Structure

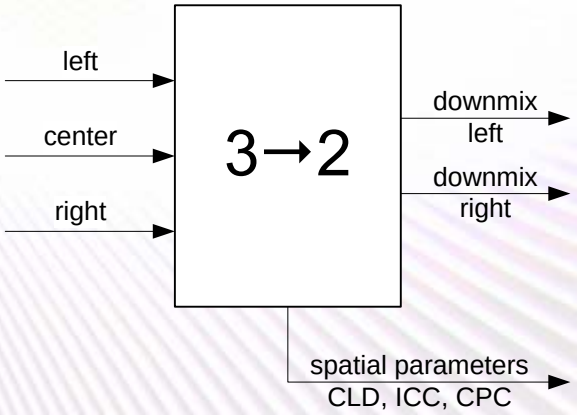
- Processing in subband domain
- Hierarchical tree structure
- Two-to-one (TTO) and three-to-two (TTT) boxes
  - Downmixing
  - Parameter extraction
- Quantization and coding



Spatial Parameters

- Channel Level Difference (CLD)
- Inter-Channel Correlation/ Coherence (ICC)
- Channel Prediction Coefficients (CPC)
- Quantization of parameters according to psychoacoustic criteria





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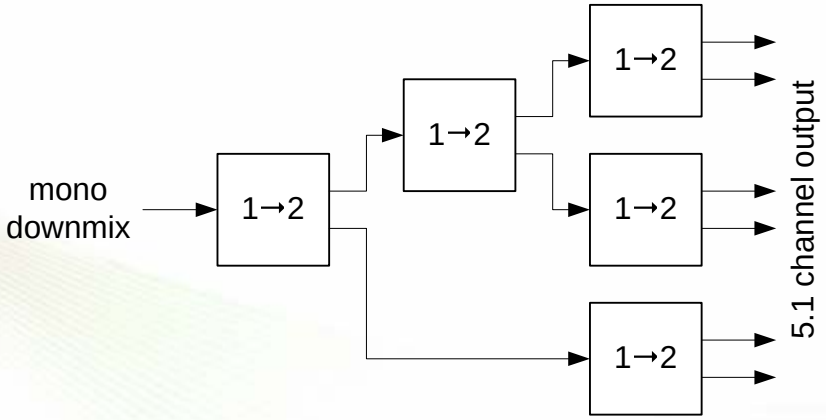
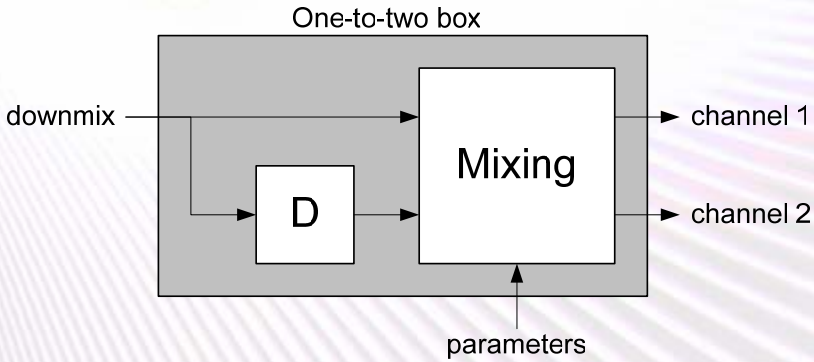
Decoding Principle

- Processing in subband domain
- One-to-two (OTT) and two-to-three (TTT) boxes
  - Decorrelation
  - Mixing according to parameters





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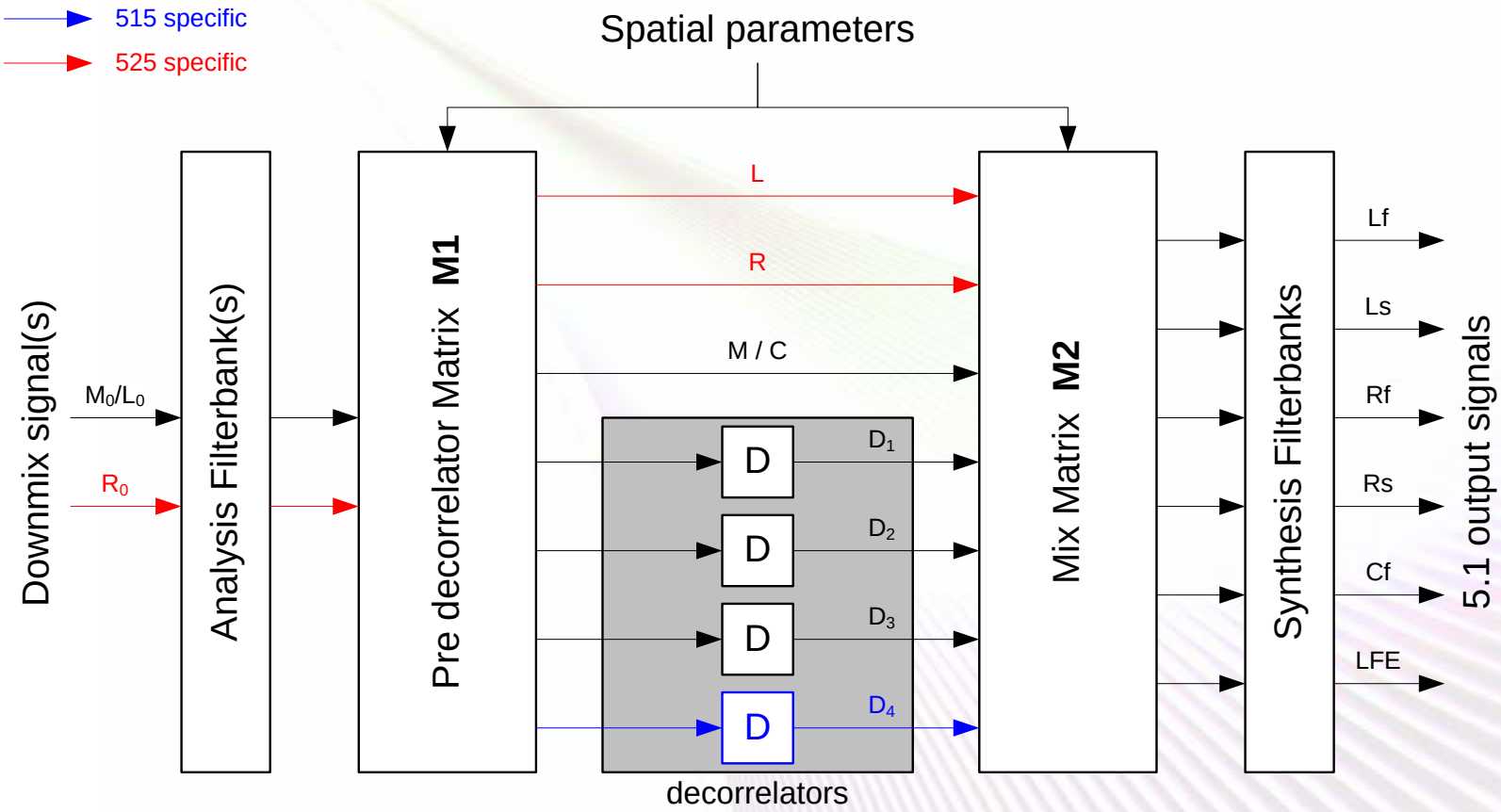
Decoder Structure



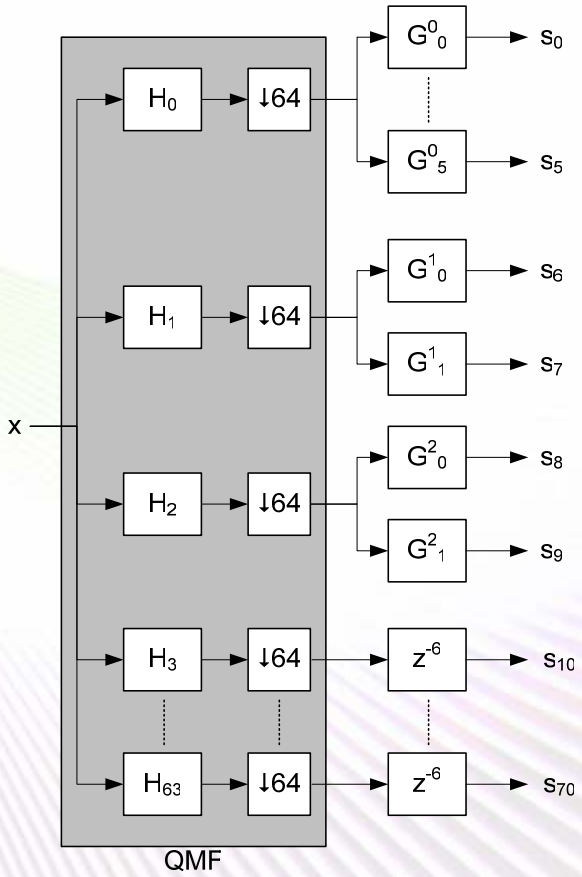
Filterbank

Hybrid filterbank:

- Complex-modulated 64 band QMF bank
- Nyquist filtering of lowest 3 QMF bands

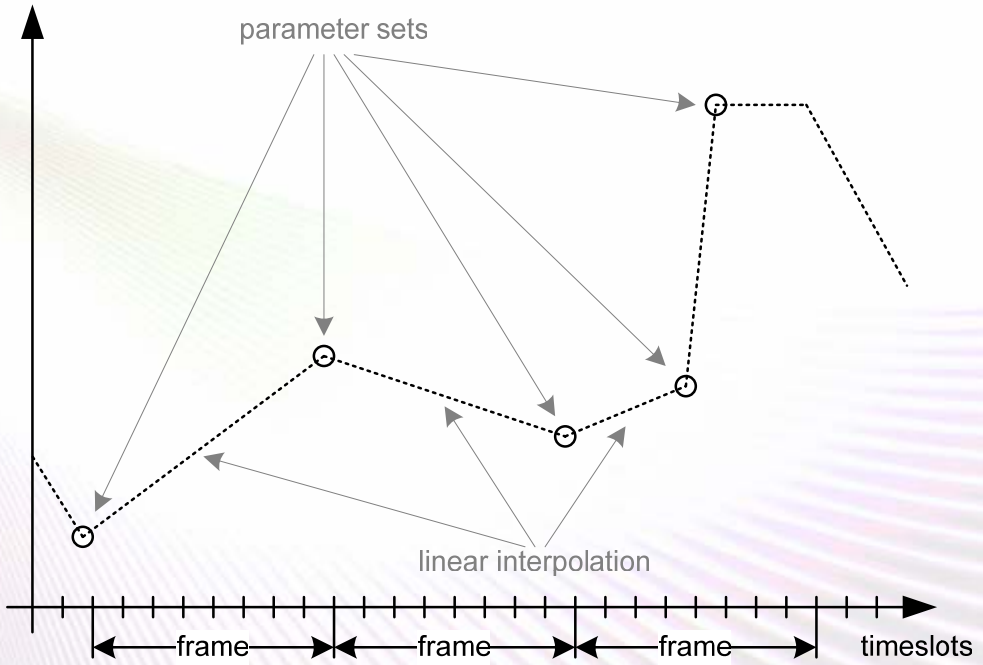
→71 hybrid subbands

1 timeslot = 64 samples



Time/Frequency Resolution

- Customizable frame length
- Parameter sets
  - One or more per frame
  - Linear interpolation
- Grouping of hybrid subbands to parameter bands according to perceptual criteria



Decorrelation

- Recreates diffuse sound fields
- Improves spaciousness

Decorrelators:

- Create output signals with correlation close to zero
- Reverberation filters in subband domain
  - All-pass filters
  - Spectral and temporal envelope shaping

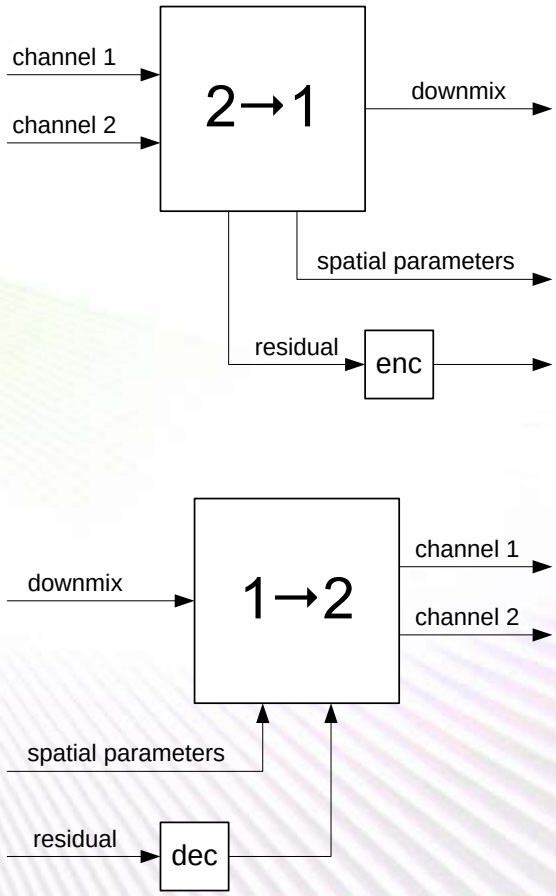
Temporal Shaping Tools

Improve temporal structure of output signals

- Subband Temporal Processing (STP)
  - Temporal shaping of decorrelated signals in the subband domain
- Guided Envelope Shaping (GES)
  - Temporal shaping of direct part of output channel signals according to transmitted envelope data

Residual Coding

- Extraction and AAC encoding of residual signals on encoder side
- Residuals replace decorrelated signals on decoder side
- Improved quality for high-bitrate scenarios (up to transparency)
- Scalable residual bandwidth



The MPEG Surround Coding Algorithm

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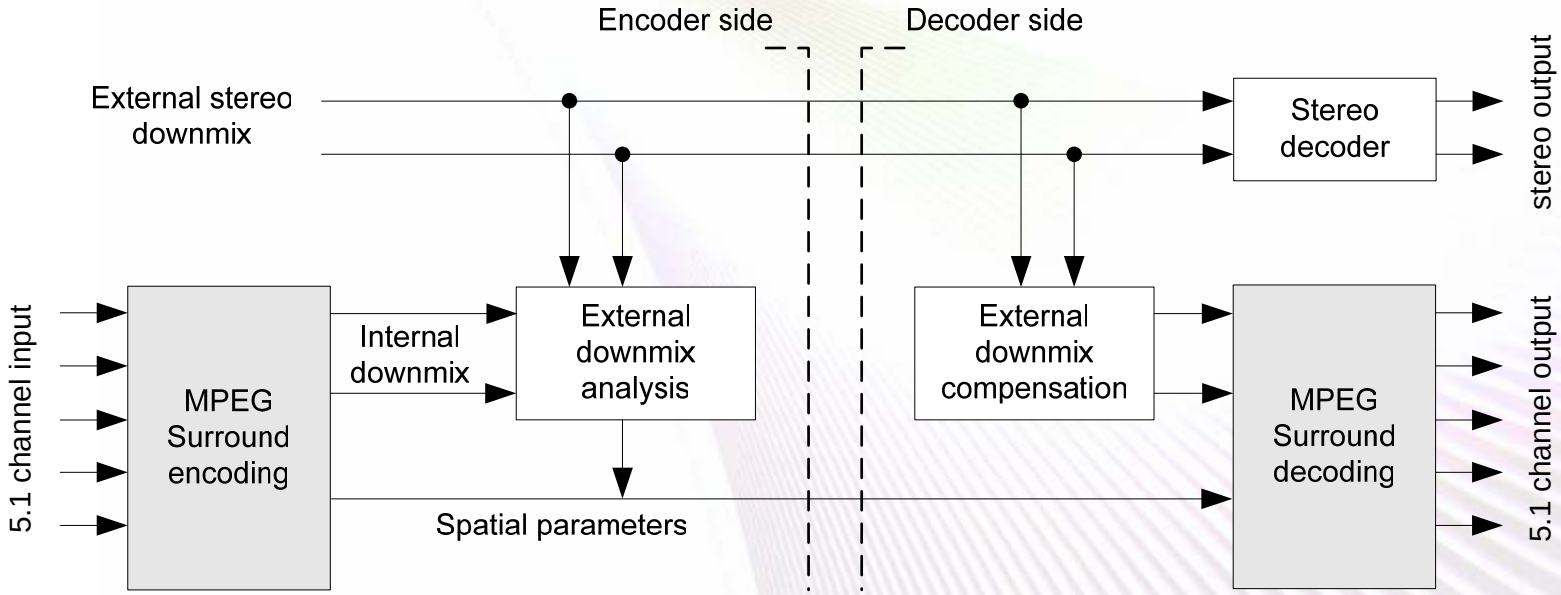
External Downmix

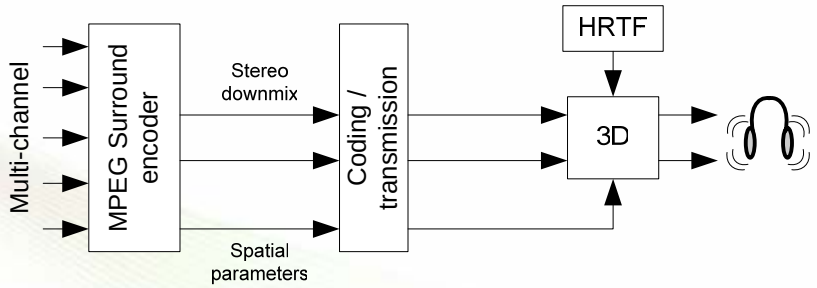
- Transmission of external ‘artistic’ downmix
- Downmix compensation on decoder side

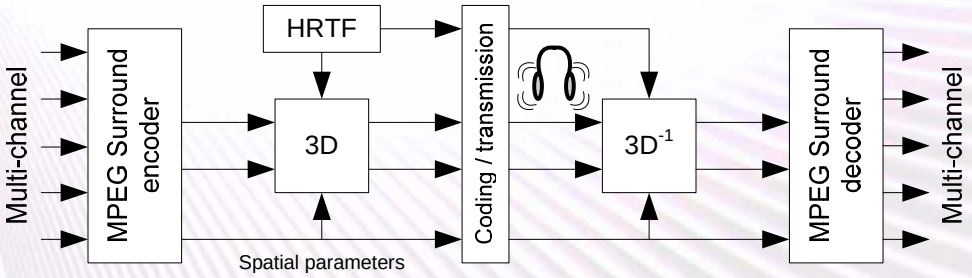


Binaural Processing

→ Virtual headphone surround

- Binaural decoding
  - Binaural processing on decoder side
- 3D stereo mode
  - Binaural processing on encoder side for direct headphone listening
  - Inversion for MPEG surround decoding





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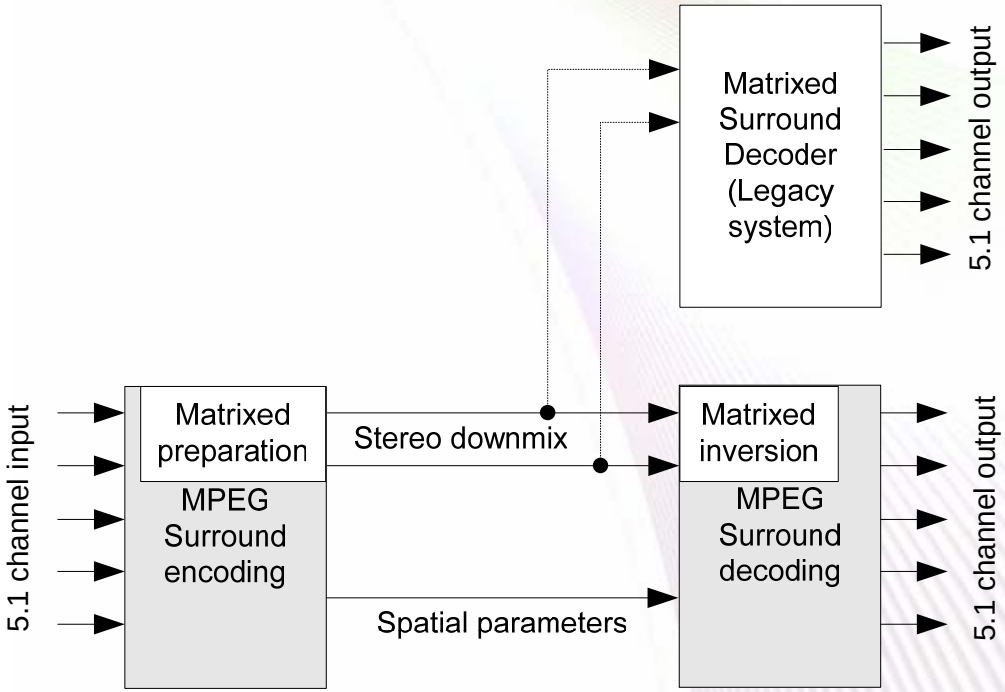
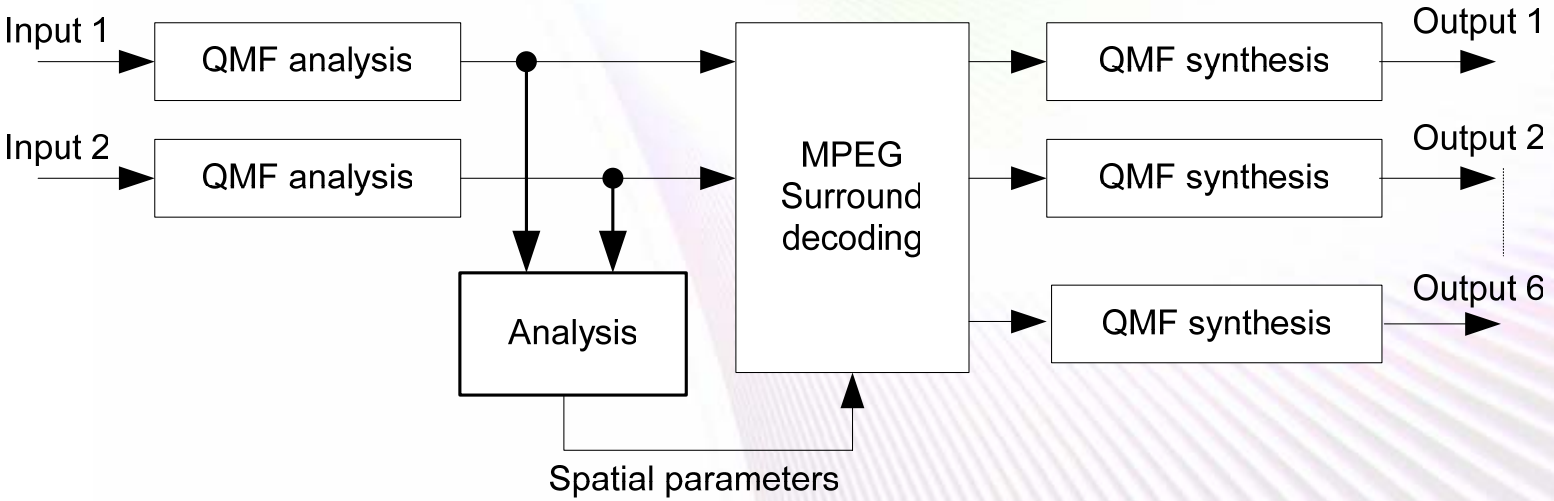
Matrix Compatible Mode

- Encoder creates matrix compatible downmix
- Matrix inversion in MPEG Surround decoder



Enhanced Matrix Mode

- No side information
- Extraction of parameters from matrix compatible downmix
- Significantly better quality than legacy matrixed systems



Low Power Decoder

- Reduced decoding complexity/power consumption at slightly lower quality
- Partially complex processing
  - Real-valued QMF bank
  - Real to complex converters for lower QMF bands
  - Aliasing reduction
  - Low power decorrelators

Profiles and Levels

Level	Configurations	Max. output channels	Max. sampling rate	Residual coding
1	515, 525	2.0	48	-
2	515, 525	5.1	48	-
3	515, 525	5.1	48	X
4	515, 525, 727, 757	7.1	48	X
5	515, 525, 727, 757, arbitrary extension	32	96	X

MPEG Surround Summary

- High scalability from lowest bitrates to transparency
- Backwards compatible
- Support for external (artistic) downmix
- Matrix compatible encoding, enhanced matrix decoding mode
- Binaural processing
- Low power decoder profile

MPEG-Surround-Demo

- Software-Decoder + 5.1 Soundcard
- Demo Items:
  - Money
  - Five voices
  - Effects

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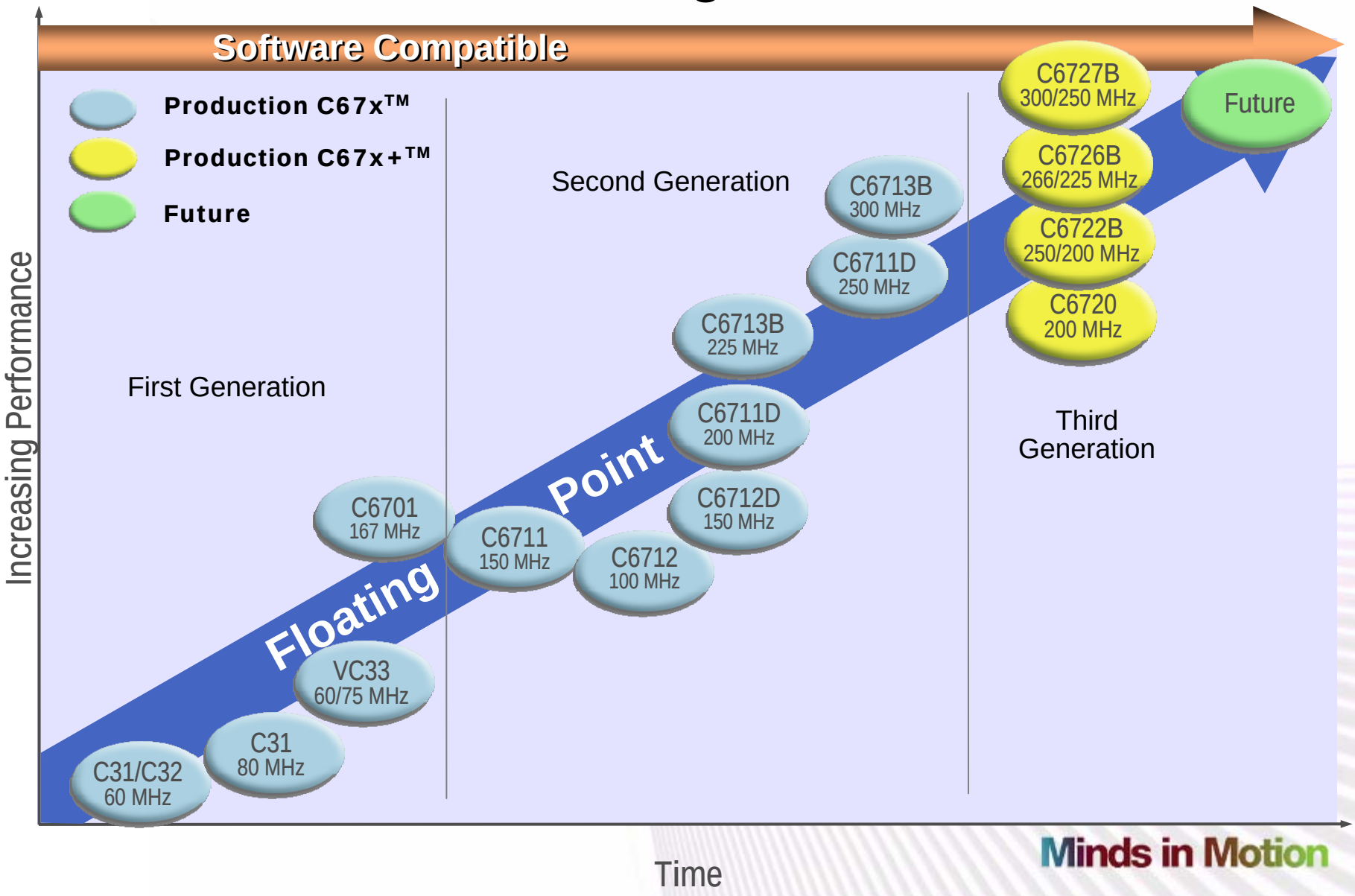
Audio Coding on TI Floating-point Processors

- TMS320C672x Processor Platform
- Floating-point Audio Coding Basics
- MPEG-Surround Implementation
- Available Audio Codecs
- Prototype Realization

TI Floating-Point DSP Platforms



C67x+™ Floating-Point DSP Platform

	C6722B 200/250 MHz	C6726B 225/266 MHz	C6727B 250/300 MHz
MFLOPS	1200/1500	1500	1500/1800
Memory	128 K Bytes SRAM 32 K Bytes Instr. Cache	256 K Bytes SRAM 32 K Bytes Instr. Cache	256 K Bytes SRAM 32 K Bytes Instr. Cache
McASP	2	3	3
I²C	2	2	2
SPI	2	2	2
HPI	N/A	N/A	Yes
EMIF	16-bit/100MHz	16-bit/133MHz	32-bit/133MHz
dMax	Yes	Yes	Yes
Package	RFP	RFP	GDH/ZDH
Availability	NOW	NOW	NOW

Software Compatible

Typical Applications

Musical Instruments and effects

Sound and musical modification and generation

Pro-audio, synthesizers

Audio broadcast, encoding and studio applications

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TMS320C672x: Features

- 200/250/300 MHz CPU clock
- Enhanced C67x+ core
- Memory Organization
- 32 KByte L1 program cache
- Up to 256 KByte internal RAM
- dMAX DMA engine
- Up to four floating-point additions/cycle

➡ Very cheap and powerful 32-bit floating-point DSP

TMS320C672x: Memory Performance

Cycles for memory transfer (1K word)

DSP, transfer type / memory src→dest	**C6727** (copy loop)	**C6727** (DMA burst transfer)	**C6713** (copy loop, L2 data cache valid/not valid)
int.→int.	1100	(7600)	2100 / 2400
int.→ext.	12300	7700	2300 / 7700
ext.→int.	55500	11000	2600 / 8200
ext.→ext.	68000	13000	2100 / 13700

C672x has no data cache

→ **Avoid direct SDRAM access**

→ **Operate on internal RAM**

→ **Use DMA for data movement**

Performance of DMA: 6 cycles/word

Conclusion:

- Best performance if all data internal
- Performance problems for memory-intensive applications
  →Memory optimizations

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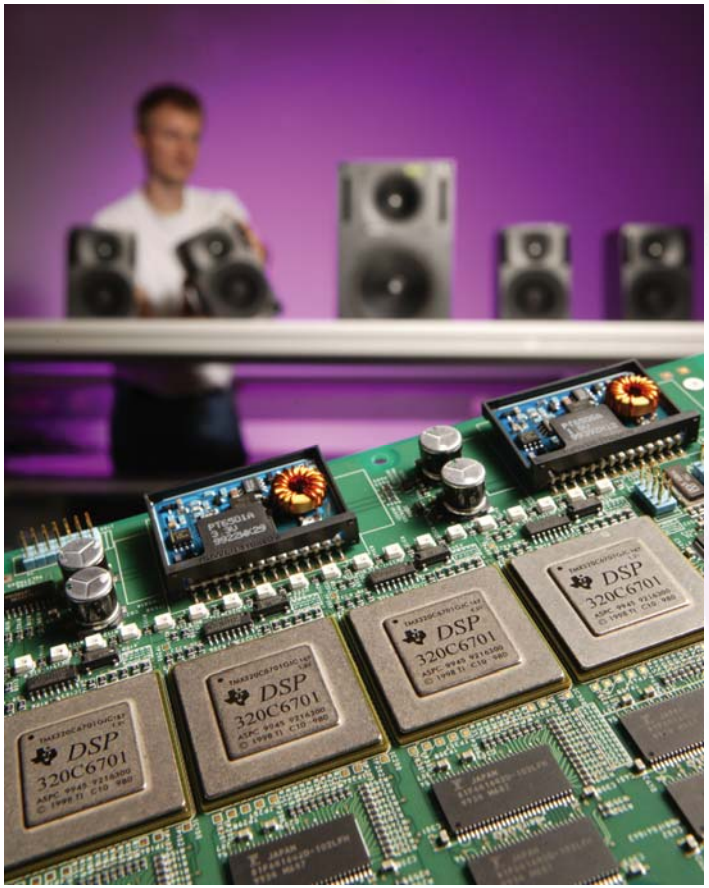
Advantages of Real-Time DSP Systems



- Cost-effective
- Reliable
- Power-efficient
- Powerful interfaces on chip
- Low data delay between input/output

Preparation of Audio Reference Code

- Using single precision
  - Standard math functions use double precision by default
  - Single precision → final resolution of 19 bits

  → enough for 16 bit output resolution

- Avoid long integer types
  - Long (40 bit) data types to integer (32 bit)

Optimization of Audio Reference Code

- High Level
  - Powerful C-Compiler → most parts optimized in “C”
  - Algorithmic simplifications
  - Using vector functions for pipeline benefits
  - Sequential data processing
    - Advantage of L1/L2 cache strategy (671x)
  - Memory optimizations (internal memory, DMA)
- Low Level
  - Rewritten in assembler:
    - t/f-mapping (MDCT, FFT)
    - Math functions

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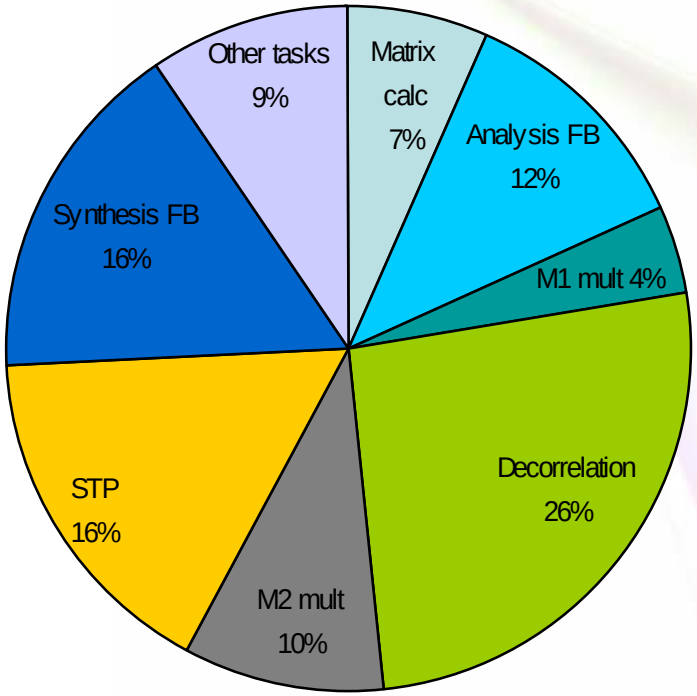
MPEG Surround Optimization Steps

- Reduce memory demand
- ‘Slot based’ processing
- Avoid redundant copying and math operations
- Reorganize nested loops and multidimensional arrays
- Use of vector functions

MPEG-Surround Decoder Workload



High-quality decoder
525 configuration (C6727)

- Filterbanks: 28%
- Decorrelation: 26%
- Matrix operations: 21%
- STP: 16%

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Available Audio Codecs for TI DSPs

All figures in MIPS for 44.1kHz sampling rate	Encoder	Decoder
	TI-C6713	TI-C6713
mp3 (stereo)	70*	35
MPEG-4 AAC-LC (stereo)	60	30
5.1 MPEG-4 HE AAC	300	-
5.1 MPEG-4 HE AAC 96kHz	-	225
mp3 Surround Decoder	-	90
MPEG-Surround	TBD	TBD

* Preliminary results, C-level optimization not completely finished

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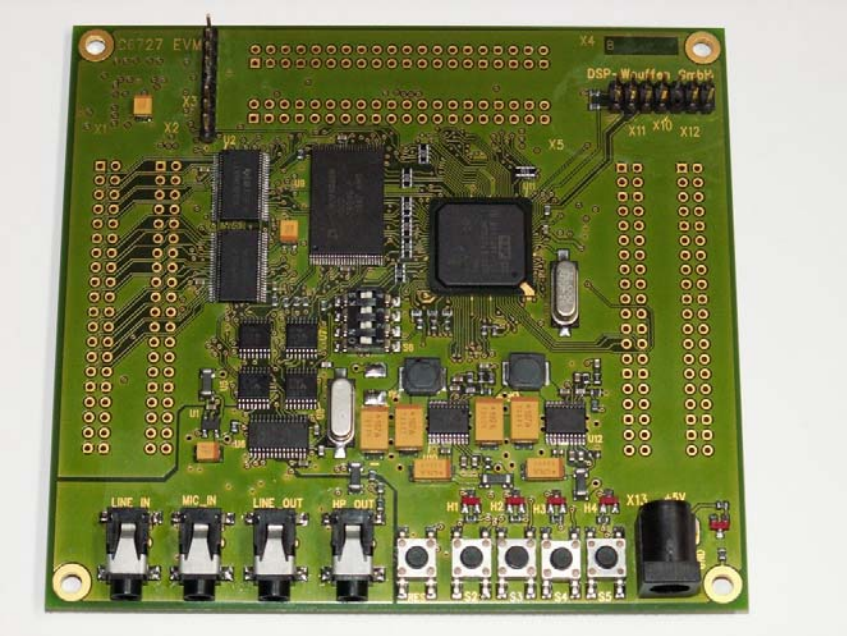
Prototype Realization

- Hardware
  - TMS320C6713 DSK
  - Lyrtech PADK
  - DSP-Weuffen C6727 EVM
- Tools
  - CodeComposer Studio 3.1
  - CodeComposer Simulator C67xx

Lyrtech Professional Audio Development Kit (PADK)

- TMS320C6727 DSP at 300 MHz
- 128 MB SDRAM
- 16 MB Flash
- Coaxial and optical S/PDIF inputs and outputs
- 8-channel analog inputs and outputs
- 24bit 192kHz D/A and A/D converters
- LED and DIP switches
- Hi-speed USB port

DSP-Weuffen C6727 EVM



- TMS320C6727 DSP at 300 MHz
- AIC23 stereo codec
- 128 MB SDRAM
- 512 kB Flash
- 4 pushbuttons, 4 LEDs
- JTAG connector
- Connectors for memory interface expansion
- Connectors for UHPI or expansion for low cost USB-connector-card via UHPI

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Final Conclusion

- MPEG-Surround Audio Coding algorithm is the key for exciting new products

  →Soon available on TMS320C6713/27
- Texas Instruments DSPs are a perfect platform for cost-efficient low resource implementations and rapid prototyping
- Fraunhofer IIS offers implementations for all TI floating-point platforms

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