Audio

System Design

ECE 542 Spring 2005
Dr. T. Tran

Rice University
Agenda

- Introduction: The Wide World of Audio
- Analog and Digital Audio
- Algorithms for Audio Processing
- OMAP Audio Algorithms
- DSP Architectures for the Future of Audio
- OMAP Audio Design
- Class D Audio Amplifier
Agenda: Introduction

- Introduction: The Wide World of Audio
  - Analog and Digital Audio
  - Algorithms for Audio Processing
  - OMAP Audio Algorithms
  - DSP Architectures for the Future of Audio
  - OMAP Audio Design
  - Class D Audio Amplifier
The audio life cycle

- Audio takes a journey from lips to ears

Audio in the 21\textsuperscript{st} Century

- More and more technologies are converging onto consumer entertainment electronics

Different phases, different faces

- Not just a ‘one size fits all’ approach

What we have to do

- Become a reliable partner with our customers

How TI can meet the customers’ needs

- Delivering outstanding audio technologies and audio system solutions
Audio in the 21st Century

In the “Good Old Days”...

- A new music delivery format once every ~20 years
- Portability was only for those with very strong arms

Some signs of change in the last five years:

- Significant new physical formats - MD, SACD, DVD
- New non physical formats – Streaming audio over networks
- New broadcast sources - DAB, Digital TV, HDTV
- More and more people are making their own music

Now every month brings something new

- The musical WWW reaches critical mass
- With some audio standards even set by a software company...
- Soon, every tune in the world can reach every ear in the world (provided they have IP addresses...)
- GOOD AUDIO EXPERIENCE MAKES BETTER VIDEO EXPERIENCE, NOT THE OTHER WAY AROUND!
Agenda: Analog & Digital Audio

- Introduction: The Wide World of Audio
- **Analog and Digital Audio**
- Algorithms for Audio Processing
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- DSP Architectures for the Future of Audio
- OMAP Audio Design
- Class D Audio Amplifier
The whole world is analog and digital only plays a bit part
Analog Audio: Inputs

- **Microphone input**
  - Electret (moving capacitive plates) or dynamic (moving coil) condenser microphone: Electret requires DC bias to function
  - Very low level input (~100mV): pre-amp of about 20dB required
  - Most critical input as far as noise sensitivity: requires special care!
  - Mic performance is very important for good speech recognition, video phone, karaoke, and full-duplex speakerphone
Microphone Front-End Issues

You need a high performance solution with low component count, small board area and component bulk, simple integration into your digital control environment

- PGA2500 programmable gain 10dB to 65dB in 1dB step.
- Extremely low noise and low distortion, 1nV/√Hz and 0.00002% THD
- Simple interface to A-to-D converters: careful matching of supply voltages and signal levels can create an elegant input stage design
Analog Audio: Inputs

- **Line level inputs**
  - Line level amplitude varies greatly depending on the audio source, for example portable CD to DVD player.
  - Typically, **Line In = 200mV_{rms} to 2V_{rms}, where 1V_{rms} = 2.828V_{p-p}**
  - **Issues related to line level:**
    - Portable devices can’t drive audio system to the maximum power.
    - High-end DVD players can overdrive the audio system inputs and cause severe distortion.
  - **Audio Engineering Society (AES) has published many standards to define the audio levels for different types of equipment but only a few manufactures follow this recommendation.**
Analog Audio: Inputs

- Example of audio line level input problem

Input level is clipped at 3.3V, overdriving the CODEC

2Vrms/5.656Vp-p

ADC

3.3V

L-Audio
R-Audio

ground
One solution to the clipping problem

Add a voltage divider network to limit the AC input signals to the CODEC supply voltage. For example,

$$3.3V = 5.656V \frac{R2}{(R1 + R2)}$$

R2 = 1.4 R1
Let R1 = 20K -> R2 = 28K,
Keep input impedance greater than 20k
Analog to Digital Conversion Issues

You need enough dynamic range and performance to cope with almost any situation at almost any sample rate, but without burning lots of power, taking lots of board space or spending lots of money

- A clear case of where the best product is in fact the best compromise between competing requirements; in real-world products, you don’t want the extra cost, board area and heat that a few extra dB’s brings - and you’ll lose a lot of versatility and performance if you try to save any more money

- This function is the ‘gateway to the soul’ of your equipment, and you want the reassurance of the long experience of the Burr-Brown division of TI
- Oversample by N pushes the quantization noise way above the audio spectrum. Sampling at Nyquist rate (2 times the highest frequency component) requires higher order anti-aliasing filter.
- Digital LPF removes the noise generated by the ADC.
- Sample Rate (SR) Decimation reduces the data rate for audio processing
Oversampling ADC

- Noise Density
- Digital LPF Passband
- Noise shaping characteristic (sigma-delta A/D)

Nf$_s$ Sample Rate

f$_s$/2

Nf$_s$/2

Frequency
Oversampling ADC

- Watch for these potential problems
  - Analog input filter is required to limit the audio bandwidth. Without this, out-of-band noise folds back into the audio spectrum and degrades the CODEC performance.
  - Isolate the analog power supply by using a high performance linear voltage regulator.
  - The reference voltage Vref needs to be decoupled properly. This voltage is used for sampling threshold comparison so it needs to be very clean.
  - Routing all the input signals carefully and away from high speed and high current switching signals.
D-to-A Conversion Issues

The required state-of-the-art in replay keeps rising; better dynamic range, less distortion, lower solution cost, greater filtering flexibility for both DSD and PCM operation over a wide Fs range.

- You try to make your clocks as good as possible but they are never perfect; you need good tolerance to clock jitter.
- Not made of money: you need wide dynamic range and THD in real applications using affordable external components, not made out of Unobtanium.
- Good control of out-of-band noise is critical in broadcast applications, and in general any application where the D-to-A output feeds any kind of converter or modulator.
- Format flexibility (PCM, DSD) and filter transient response choice to deliver low delay and good sound quality.
Interpolation filter increases the sample rate by N for the oversample DAC to process the data.

Analog lowpass filter is required to remove the sampling noise. This filter is included in the TLV320AIC23 device.
Oversampling DAC

- Watch for these potential problems
  - Isolate the analog power supply by using a high performance linear voltage regulator.
  - Isolate the analog filter supply from the DAC supply as recommended by the manufacture.
  - The reference voltage Vref needs to be decoupled properly. This voltage is used for sampling threshold comparison so it needs to be very clean.
TLV320AIC23 Stereo CODEC

Analog input filters

DSP

AIC23

AVDD (14)
AGND (15)
HPVDD (8)
HPGND (11)
LOUT (12)
ROUT (13)
LHPOUT (9)
RHPOUT (10)
CLKOUT (2)
VMID (16)

BMW (1)
DGND (28)
DVDD (27)
LVDD (20)
RLINEIN (19)
MICBIAS (17)
MICIN (18)
LRCIN (5)
DIN (4)
DOUT (6)
LRCOUT (7)
BCLK (3)
CSB (21)
SDIN (23)
SCLK (24)
MODE (22)
XTI/MCLK (25)
XTQ (26)
S/PDIF Digital Interfaces

- S/PDIF, Sony/Philips Digital Interface, has been adapted as an I/O interface for digital audio and other consumer electronics equipment.

- S/PDIF connector comes in two types, optical and/or RCA. Optical provides higher performance due to noise isolation but it costs more because of the additional optical transceiver.

- Low jitter is a must for this interface, embedded clock and data on a single conductor.

- New TI SPACT technology completely eliminates performance degradation through plastic optical connections (long cables, low supply voltage...) even at high sample rate
Effect of Interface Jitter

THD+N [%]

Audio Signal Frequency [Hz]
Effect of Interface Jitter

THD+N [%]

SPACT DIR
‘Analog’ DIR

Audio Signal Frequency [Hz]

THD+N [\%]
Algorithms for Audio Processing

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### Lossless Coding
- Coding that is bit exact, i.e., the original signal is recovered bit for bit. Examples of lossless coding include numerical coders:
  - Arithmetic coding,
  - Ziv-Lempel (LZW) coding / WinZip
  - Huffman coding / Fax
  - Run Length Coding
  - Meridian Lossless Packing (MLP) / DVD Audio

### Lossy Coding
- Coders that create an approximate reproduction of the input signal, i.e., “sounds like” the original speech or audio or “looks like” the original image. Examples of lossy coding includes source coders:
  - Sub-band coding / MP3
  - Transform coding / JPEG2000
  - Vector quantization / MPEG-4
Lossless Coding

Huffman Coding

- Assign the most frequent events to the shortest code words.
- Non-duplicating prefix, for example 110 and 11011 can’t be code words

Huffman coding example:

<table>
<thead>
<tr>
<th>Bus Status</th>
<th>Probability</th>
<th>Code word</th>
</tr>
</thead>
<tbody>
<tr>
<td>On time</td>
<td>0.5</td>
<td>0</td>
</tr>
<tr>
<td>Late</td>
<td>0.35</td>
<td>10</td>
</tr>
<tr>
<td>Early</td>
<td>0.125</td>
<td>110</td>
</tr>
<tr>
<td>Wrecked</td>
<td>0.025</td>
<td>111</td>
</tr>
</tbody>
</table>

Run Length Coding

- Special start and stop codes for repeated data.
- Only send the value and how many times the value is repeated; for example 5555 5555 is coded as 85.
Human Audio Perception

- Properties of auditory perception can be used to exploit irrelevancy in audio that hides noise where you cannot perceive it.
- Strategy: Identify signal components for which the ear is less sensitive to distortion.
- Effects are typically signal dependent.
- Effects are broadly categorized into:

<table>
<thead>
<tr>
<th>Time domain</th>
<th>Frequency domain</th>
</tr>
</thead>
<tbody>
<tr>
<td>Prevalent when signal contains large transient components</td>
<td>Prevalent when signal is approximately periodic</td>
</tr>
<tr>
<td>Due to mechanical processes and early neural processing</td>
<td>ExpRESSED using non-uniform frequency scale derived from physiology of cochlea</td>
</tr>
<tr>
<td>Masking components occurring just before and after a transient</td>
<td>Subband coding (AC-3, DTS, MP3, etc.)</td>
</tr>
</tbody>
</table>
Exploiting Perceptual Effects

Typically encoders/decoders perform these steps:

1. Analyzes input in frequency domain (critical subbands).
2. The ear masks (does not hear) neighborhood low amplitude signals of a certain frequency in the presence of a nearby higher-amplitude signal. Thus the psychoacoustic model helps ignore these low amplitude signals well (frequency irrelevancy).
3. The psychoacoustic model also helps gauge which subbands the ear does not hear as well. Scales the number of bits in relation to hearing threshold and masking signals.
Block Diagram of an MP3 Encoder
(by layer)

Digital Audio Input → 32 Sub-band Filterbank → Modified Discrete Cosine Transform → Psychoacoustic Model + 1024 Points FFT → External Control → Side Information and optional Ancillary Data

Layer 1
Layer 2 enhancements
Layer 3

Bit Allocation, Quantization, and Coding → Huffman Coding → Bitstream Formatting → Encoded Audio Bitstream
Widely Used Audio Standards

Widely Used ISO: MPEG Audio Coders:
- MPEG-1 Audio Layers 1, 2 and 3 (MP3 is the popular name for MPEG-1 Audio Layer 3)
- MPEG-2 Audio Layers 1, 2 and 3: 32-256 kb/s/ch European DVD and HDTV
- MPEG-2 Advanced Audio Coder (AAC): 8-160 kb/s/ch Japanese HDTV
- MPEG-4 Speech and Audio Coders - MPEG-2 AAC (with extensions)

Other Audio Coders:
- Sony ATRAC: Primarily 2-ch coders used in Mini-Disk players at 144 kb/s/ch
- Microsoft WMA: Designed for 1- and 2-ch streaming, 5 kb/s/ch to 128 kb/s/ch, 8 to 48 kHz sampling
- Real Audio: Designed for 1- and 2-ch streaming, 5 to 96 kb/s/ch, 8 to 48 kHz sampling
- Dolby Digital AC-3: Theater/Home multichannel standard. 5.1 channels, US DVD (5.1 ch/320 kb/s), US HDTV
- DTS (Digital Theater System): Home multichannel competitor. 5.1-ch system w/48 kHz sampling.
- MLP (Meridian Lossless packing): Lossless coder for DVD-Audio. Multichannel system up to 24bit/192kHz

Other Standardized Audio Processes:
- Dolby Prologic
- DTS NEO:6
- THX
Audio Effects Classification

Amplitude based effects
- Volume, Panning, Tremolo, Noise gating, Compression/Expansion, ...

Time delay
- Modulated delay: Chorus, Flanger, Phaser, Vibrato, ...
- Fixed delay: Echo/Delay, Reverb, ...

Waveform shaping effects
- Vacuum tube simulator, Non-linear effects

Frequency filters
- Noise Reduction, Pitch-shifter, Vocoding, Equalizer, ...

Compression/Expansion

Impulse Response

Reverb
Acoustic Effects Library Example

**Equalizer** Adjust the level of signal frequencies components*

*From Parametric Equalization on the TMS320C6000 DSP (SPRA867) app note and code for C6711 DSK
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Audio Algorithms

Audio Algorithms for C54xx/C55xx DSPs

System Software (Play, Stop, FF, Rew, Repeat)
EQ (5-band)
Volume (Stereo)
SRC(Stereo)
AC3 Decode
MP3 decode
MP3 encode
AAC decode
AAC encode
WAV Playback
WAV Record
ADPCM Playback
ADPCM Record
ATRAC-3 decode
ACELP.NET decode

TwinVQ
Dolby Digital 2 channel  Class - C
Dolby Digital 5.1 channel  Class -C
Real G2 decode
WMA decode
WMA w/ DRM decode
WMA encode
Lucent ePAC
Liquid Audio SP3 (ver. 1.0)
MPEG-1 Layer I Encoder
MPEG-1 Layer I Decoder
MPEG-1 Layer II Encoder
MPEG-1 Layer II Decoder
AC-3 2 ch Encoder (DDCE) Class A
Intertrust
723.1
AAC Decoder

Features

- MPEG-4 AAC Decoder with simple and Low complexity Profile.
- Decoder outputs pcm file format from 16-bit AAC Encoded samples
- all sample rates (8 kHz to 48 kHz)
- all bit rates (8 kbit/s to 288 kbit/s)
- mono, stereo, dual-channel, joint stereo
- The given decoder supports both; ADIF (Audio Data Interchange Format) and ADTS (Audio Data Transport Stream) input formats.
- automatic bitstream synchronization and error recovery
- audible fast-forward/rewind mode
- TI eXpressDSP compliant
- C5500 large memory model object code library

Performance

Low memory library

- 25.5 MHz (peak) and 23.5 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 13.5 Kwords of SARAM (const and program) and 11 Kwords of DARAM including real-time I/O buffers and stack

Normal memory library (with more memory)

- 20.5 MHz (peak) and 18.5 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 21 kwords of SARAM (const and program) and 11 Kwords of DARAM including real-time I/O buffers and stack
AAC Encoder

Features

- The given Encoder implements AAC Low complexity (LC) profile of the standard up to 2 channels (stereo).
- The given encoder supports both; ADIF (Audio Data Interchange Format) and ADTS (Audio Data Transport Stream) input formats.
- It supports sampling frequency from 8 kHz to 96 kHz as specified by the standard.
- Encoder outputs AAC file format from 16- bit raw PCM samples.
- It supports the bit-rate up to maximum depending upon sampling frequency as specified by the standard (see 8.2.2 of ISO/IEC 13818-7).
- It supports all necessary tools and features, so that the given algorithm is standard compliant.

Performance

- 73 MHz (peak) and 64 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 27 kwords data RAM including real-time I/O buffers
- 14 kwords data ROM
- 23 kwords program ROM
MP3 Decoder

Features
- layer-3 only
- 16-bit PCM output
- all sample rates (8 kHz to 48 kHz)
- all bit rates (8 kbit/s to 320 kbit/s)
- variable bit rate (VBR) mode
- mono, stereo, dual-channel, joint stereo
- CRC error protection
- automatic bitstream synchronization and error recovery
- audible fast-forward/rewind mode
- TI eXpressDSP compliant
- C5500 large memory model object code library

Performance
- 20.97 MHz (peak) and 15.38 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 11 kwords data RAM including real-time I/O buffers and stack
- 8 kwords data ROM
- 10 kwords program ROM
- Tested against MPEG-1 and -2 compliance suites
MP3pro Decoder 2/2

Features
- Spectral Band Replication Technology using ancillary data field (designed by Coding Technologies.)
- Supports .mp3 files
- Supports a maximum of up to two channels
- All mode (stereo/joint stereo/dual channel/single channel)
- Ancillary data extraction
- Variable bit-rate decoding

Performance
- 66 MHz (peak) and 59 MHz (average) for stereo 44.1 kHz at 96 kbit/s
- 25.5 Kwords of Program memory
- mp3 part of mp3PRO decoder = 10Kwords,
- PRO part of mp3PRO decoder = 15.5Kwords
- 35 Kwords of Data memory
- mp3 part of mp3PRO decoder = 17Kwords,
- PRO part of mp3PRO decoder = 18Kwords
MP3 Encoder

Performance

- 93 MHz (peak) and 61 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 24.2 kwords data RAM including real-time I/O buffers
- 8.3 kwords data ROM
- 30 kwords program ROM
ADPCM Decoder

Description
- The ADPCM decoder supports all the bit rates and sampling frequencies, which are specified in ITU G726 Standard. I.e. Sample Rates 8, 11.025, 16000, 22.050, 32 and 44.1 KHz and 2, 3, 4 and 5 bit encoding/decoding of Mono streams.

Performance
- 3.78 MIPS (peak) and 3.76 MIPS (average) for mono 8 kHz
- 20.72 MIPS (peak) and 20.5 MIPS (average) for mono 44.1 kHz
- 977 words data RAM
- 1069 words program ROM
ADPCM Encoder

Description

- The ADPCM encoder supports all the bit rates and sampling frequencies, which are specified in ITU G726 Standard i.e. Sample Rates 8, 11.025, 16000, 22.050, 32 and 44.1 KHz and at 2, 3, 4 and 5 bit encoding/decoding. Mono encoding and decoding is supported.

Performance

- 4.71 MIPS (peak) and 4.68 MIPS (average) for mono 8 kHz
- 25.137 MIPS (peak) and 24.86 MIPS (average) for mono 44.1 kHz
- 990 words data RAM
- 1750 words program ROM
WMA Decoder

Description
- The WMA C55x all bit rate decoder supports all the bit rates, which are specified in WMA decoder Standard. It supports version 2, version 7 as well as version 8 encoded bit streams.

Performance
- 44.4 MHz (peak) and 17.5 MHz (average) for stereo 44.1 kHz at 128 kbit/s
- 27 kwords data RAM
- 11.5 kwords data ROM
- 16.5 kwords program ROM
5-band Equalizer

Features
- N bands
- Gains adjustable from –15db to +15 db in 1db steps
- Distortion free flat spectrum at 0 dB gain
- No unpleasant audible artifacts during gain changes
- 16-bit mono PCM input and output audio data
- Center frequencies and bandwidths are configurable off-line
- Implemented by 2nd order IIR sections in cascade
- TI eXpressDSP xDAIS compliant

Performance
- 7.64 MIPS (peak) and 7.58 MIPS (average) for 5 bands
- 4.24 MIPS (peak) and 4.24 MIPS (average) for 3 bands
- 208 words data RAM including real-time I/O buffers
- 266 words data ROM
- 1427 words program ROM
Speech Recognition

Key Features

VoCon is optimized for small vocabularies and offers the following advanced features:

- **Continuous speech input**
  VoCon can recognize natural speech, so there is no need to leave gaps between words.

- **Speaker Dependent Recognition**
  User creates own vocabulary via user word training.

- **Speaker Independent Recognition**
  Vocabulary consists of optimized acoustic models corresponding to pre-defined grammar and word list.

- **Whole-word and Phoneme-based Recognition**
  Ensures high recognition accuracy and the design of specific words and acronyms.

- **Keyword Activation**
  Filters and recognizes keywords out of natural spoken sentences.

- **Out-of-Vocabulary Rejection**
  Rejects words that are not part of the vocabulary.

- **Speaker Adaptation**
  Adapts speaker-specific language characteristics to speaker-independent systems.

- **Dialogue Capabilities**
  Support the design of the user interface via grammars and integration of word probabilities.

- **Context-switching capabilities**
  Ensures shortest turnaround time when switching vocabulary.
Speech Recognition

Solutions from Scansoft: VoCon ASR

Performance (implementation on C5402)
- about 80 MIPS
- 100 kwords program ROM
- 112 kwords data RAM

Scansoft ASR software VoCon, ASR3200
Speech Recognition

Solutions from Advanced Recognition Technologies: smART Speak XG

Key Features

- Name dialing - speaker independent
- Incorporates core modules (smARTspeak NG):
  - Name dialing - speaker dependent
  - Continuous digit dialing - speaker independent/ dependent
  - Menu navigation and device control – speaker independent / dependent
  - Unique adaptation to user’s voice for enhanced performance with accents, while maintaining an excellent out of the box experience

Performance

- runs on ARM or DSP
- about 8 MIPS
- 46.7 kwords program ROM
- 167 kwords constant data (continuous digit dialing, 20 fixed commands, 100 names)
- 30.5 kwords data RAM
DSP Architectures

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A/V Receiver System Diagram

ROOM #1 – 7.1 ANALOG AMPLIFICATION

OMAP5910
Encoder/Decoder
Dolby Digital (AC-3)
WMA
Multichannel AAC
MP3
Effects
Bass Management
Treble Control
Dolby Headphone
Virtual 3-D
Downmixing
Hall Effects

S/P DIF Transceiver
DIX1700

16-20 Bit 48 Khz Conversion

2-6 channels

ROOM #2 – 7.1 DIGITAL AMPLIFICATION

POWER STAGE

DIGITAL PWM

REMOTE SPEAKER

DSP & Analog AV Solutions

HIGH
MID
LOW
I added 'one big arrow from DA610/C6714 to Room #1 and Room#2

marilyn connell, 4/30/2003
A/V Receiver System Diagram

150W x 5.1 Channels
(0.09% THD @ 8 ohms BTL)
I added 'one big arrow from DA610/C6714 to Room #1 and Room#2

marilyn connell, 4/30/2003
Typical requirement is 32-48 bit dynamic range with a 48-76 bit precision for accumulator.

Audio Source

Audio Encode/Decode
- Dolby Digital (AC-3)
- DTS
- Multichannel AAC
- Etc...

Audio Processing
- Dolby Pro Logic
- DTS NEO:6
- Graphic Equalizer

A/V Receiver Applications: The Whole Gamut

Audio HiFi Playback

Interface format to typical DAC’s are fixed point.

No Floating-point DAC!!

Typical A/D
16~24-bit @ 11.25kHz~48kHz

Typical requirement is 16-24 bit dynamic range with 32-48 bit precision for accumulator.

Typical requirement is 24-30 bit dynamic range with 48-64 bit precision for accumulator.

Typical DAC 24-bit @96kHz
Audio Application MP3 Jukebox: Reduced Gamut

HiFi Audio
Home Theater
Super Audio CD
DVD Audio
HDD server
AV receiver
DVD Receiver

Hi Fidelity
Automotive Audio
Car Audio and Entertainment

Outdoor, Mobile

Home

General Audio
Mini Stereo system
Boombox

Casual

Portable Audio
Portable CD, MP3
PDA Audio
Wireless Phone Audio
Portable Digital Radio

CD Quality

- 16-bit words, 44.1 kHz is the recreation goal of the MP3 decoder
- Tradeoff bit rate vs. storage space
- Typically in a noisier environment than the A/V Receiver (i.e. what SNR do you need when you are jogging?)

And...

- Often other concerns like cost, power consumption, portability etc. outweigh performance for a « leaner » system
Digital Audio Devices

2002

Floating Point C67xx Core

- DA610/C6713
- C6711
- C6712

Fixed Point C54xx & C55xx Core

- C5416
- C5410
- C5510
- C5509
- 5910

Fixed Function Digital Audio Processors

- TAS3001
- TAS3002/3004
- TAS3103

- Code Composer Studio
- DSP/BIOS
- xDAIS Algorithm
- Standard
- Third Parties
- Fully Programmable

- Floating Point C67xx Core
- Fixed Function Digital Audio Processors
- Functionality

- Dolby™
  - Dolby Digital™ (AC3)
  - ProLogic™
  - ProLogic™ II
  - Dolby Headphone™
- Fraunhoffer™
  - MPEG AAC (LC)
- DTS™
  - Consumer 5.1™
  - ES 6.1™
  - Neo 6.1™
  - DTS 96/24™
- MP3 (encode/decode)
- AAC (encode/decode)
- Windows Media Audio™ (decode)
- ATRAC3™ (decode)
- Security features
- Speaker Equalization
- Effects
- Fixed Function devices
- Low pin count
- GUI tools
Rita I added C5502 but it needs positioning

marilyn connell, 4/30/2003
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OMAP Audio Design

Microphone Preamp

- Need to bypass the bias voltage
- Use op amp instead of transistor -> much better noise rejection
- Need to terminate McBSP bus
OMAP Audio Design

Low noise mic
Pre-amp circuit
Audio Clock Design

MCLK, Oversampling Clock = OS x Frame Sync
-> OS = 256, 250, 272 or 384 oversample
Frame Sync = 44.1KHz, 48KHz or 32KHz
BCLK has to be sync’ed with MCLK
Master CODEC drives BCLK & Frame Sync
MULTIPLE CODECs DESIGN

- Master device drives all FS and BCLK inputs of slave devices.
- Only McBSP1 has SRG, generating different BCLK from one CLKS input.
- Only one master is allowed for Sync CODECs
Class D Audio Amplifier

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Class D Audio Amplifier
Power Amplification Issues

All equipment designers are striving for more compact, lighter, cheaper products with less heat generation and uncritical power requirements, while preserving great sound quality.

- Switching amplifiers, both digital-input and analog-input, are the way to go for efficiency improvement and bulk reduction (smaller, or even no, heatsinks). Customers nervous about switching designs; TI has world-class experts to advise.
- Small form factor is sexier - and cheaper to ship! Ultra low profile PCB design enables sub-50mm total product height.
- TI is pioneering Power Supply Compensation for digital-input amplifiers.
- Ease-of-use features of the latest analog-input switching amplifiers make them the solution of choice for sleek products like LCD television.
The “Linear Danger Zone”

Power Dissipation (Sine Wave)

- Vs = 14.4 V
- RL = 4 x 4 Ohm

CLASS AB

TDAATDAA

Danger Zone!
Class D Amplifier

- Legacy Audio Amplifiers (Linear Class AB)
  - Efficiency: 40-60%
    - Bulky and heavy
    - Hot and noisy
    - Shorter battery life
    - Power hogs

- Emerging Audio Amplifiers (TDAA Class D)
  - Efficiency: ~ 90%
    - Light and streamlined
    - Cool and quiet
    - Longer battery life
    - Energy efficient
    - Consumer friendly
    - Cosmetically pleasing
Class D Amplifier

- **Higher Efficiency**
  - >90% Theoretical, / >85% Achievable
    - Lower Current Requirement
      - Longer Battery Life
      - Less Power Input for same Power Output
    - Less Heat Dissipation
      - Smaller Packages
      - Less Heat Sinking
- Smaller Power Supply Requirement
  - Less Weight
- In General – “Greener”
Class D Amplifier

- Superior Distortion Characteristics
- No Slew-rate induced TIM (Transient Inter-Modulation Distortion)
- Very Low Feedback Required
  - Excellent Transient Response
  - Excellent Open-loop Characteristics
- Low Open-loop Output Impedance
  - Less Back-EMF Interference
- Psycho-acoustically Speaking
  - “Tighter Bass Response”
  - “Crisper/Cleaner High End”
TI Class D Amplifier

Digital Source
CD / DVD / DAB / SACD / MP3

24 bit 32-96kHz PCM

Digital Upsampler, Linearity Corrector and Modulator
PCM to PWM Processor

TAS50XX family

Switching Output stage
TAS51XX family

V+

Digital Source

24 bit 32-96kHz PCM

PCM to PWM Processor

TAS50XX family

TAS51XX family

V+
TI Class D Amplifier

Parallel Port Interface

Volume Control

TAS3002
Digital Audio Processor

TAS5100
Digital Audio
PWM Power Output Stage

DIR1703
S/PDIF Receiver

TAS5010
Digital Audio
PWM Processor

PC
Software

Analog Input

Analog Output

I^2C Bus

I^2S Bus

I^2S Bus

32kHz - 96kHz

32kHz - 192kHz

“Thunderbird”
Power Amplifier

I^2S Interface

I^2S Bus

I^2S Bus
TI Class D Amplifier EVM

CD Player

24 bit 44.1 kHz PCM
Toslink Cable

75 Watt Power Supply

Thunderbird Power Amplifier

AC Power Strip

8 ohm Speaker

8 ohm Speaker