FTF | FREESCALE TECHNOLOGY FORUM
POWERING INNOVATION

Introduction to MCU Audio and USB Audio Streaming

FTF-CSD-F0110

Rudan Bettelheim

June 2012

Abstract

Find out why audio accessories are migrating from DSP to MCU control, and discover the mysteries of USB audio streaming. If you cannot tell your USB isochronous synchronous from your USB isochronous asynchronous stream, this session is for you. You will also learn about the full range of Freescale MCU audio solutions.

Agenda

- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demo
- Freescale MCU audio solutions

Agenda

- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demo
- Freescale MCU audio solutions

History of Audio: *From Edison to the iPod*


1857
Phonatograph


1876
Telephone


1877
Phonograph

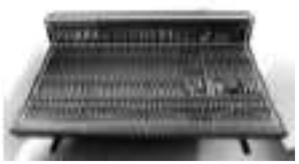

1887
Gramophone


1898
Telegraphone


1909
Triode Vacuum Tube


1935
Magnetophon


1955
Multitrack Recorder


1958
Mixing Console


1962
Compact Cassette


1966
Dolby A-Type Noise Reduction


1971
Digital Audio Processor (Delay)


1982
Compact Disc


1991
Multitrack Digital Recording/Editing


2001
Portable Media Player

Agenda

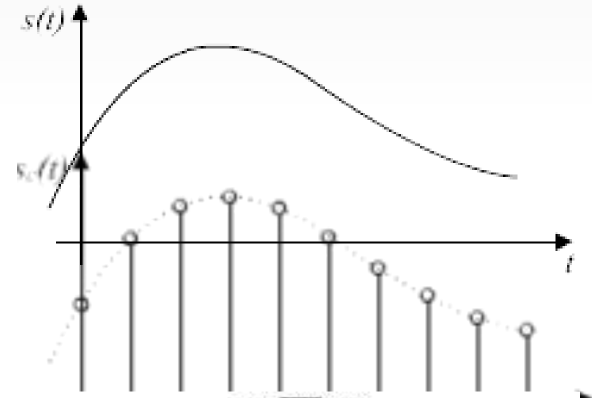
- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demo
- Freescale MCU audio solutions

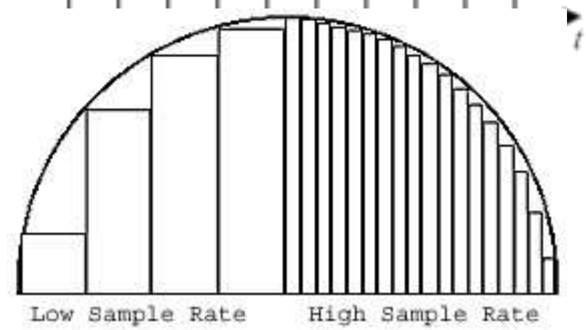
Making it Digital

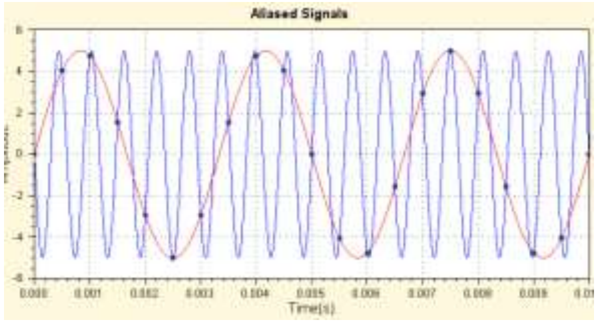
Digital Sampling

- Sample rate defines the number of samples per unit of time.
- For time-domain signals, the unit is Hertz.

- The Nyquist–Shannon sampling theorem states that perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled.







Quantization/Dithering – Noise to the Rescue

- Analog-to-digital conversion maps a continuous range of analog values to *n* discrete digital codes
- **Quantization** errors are unavoidable. These errors are the difference between the input analog value and the quantized digital value
- **Dithering** is the process to mathematically remove unwanted quantization noise from the audio signal. Replacing them with a constant, fixed noise level
- **Dither removes the correlation between the quantization errors and the audio signal, making them inaudible**

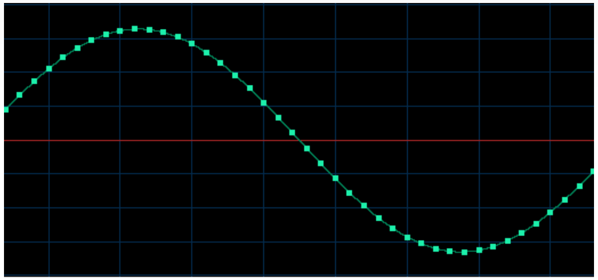


Figure 1 - 980Hz sine wave, -60dB, 24 bits, 44.1kHz

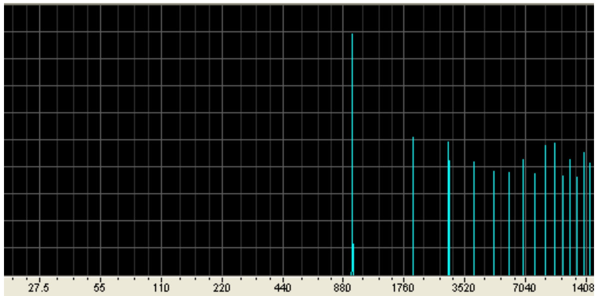


Figure 3 - frequency spectrum for 980Hz sine wave, -60dB, 16 bits, 44.1kHz

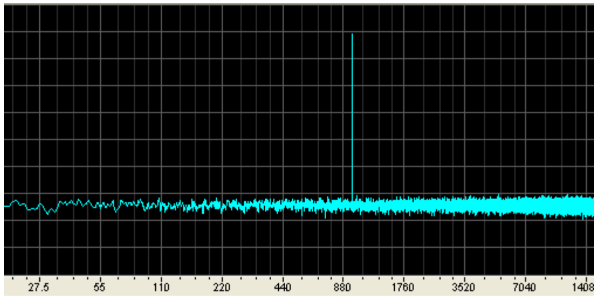


Figure 5 - frequency spectrum for 980Hz sine wave, -60dB, 16 bits, 44.1kHz, with dithering

Sample Rate, Bit Rate & Bit Depth

Sample Rate:

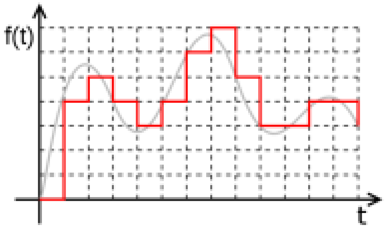
Number of samples per unit of time. Determines the highest frequency that can be reproduced.

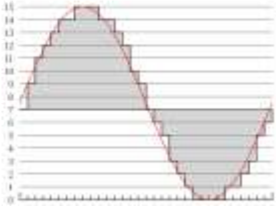
Bit Rate (or Data rate):

The amount of digital information that is recorded per unit of time.

Bit Depth (or Resolution):

The number of bits of information recorded for each sample. Determines dynamic range and the noise floor.

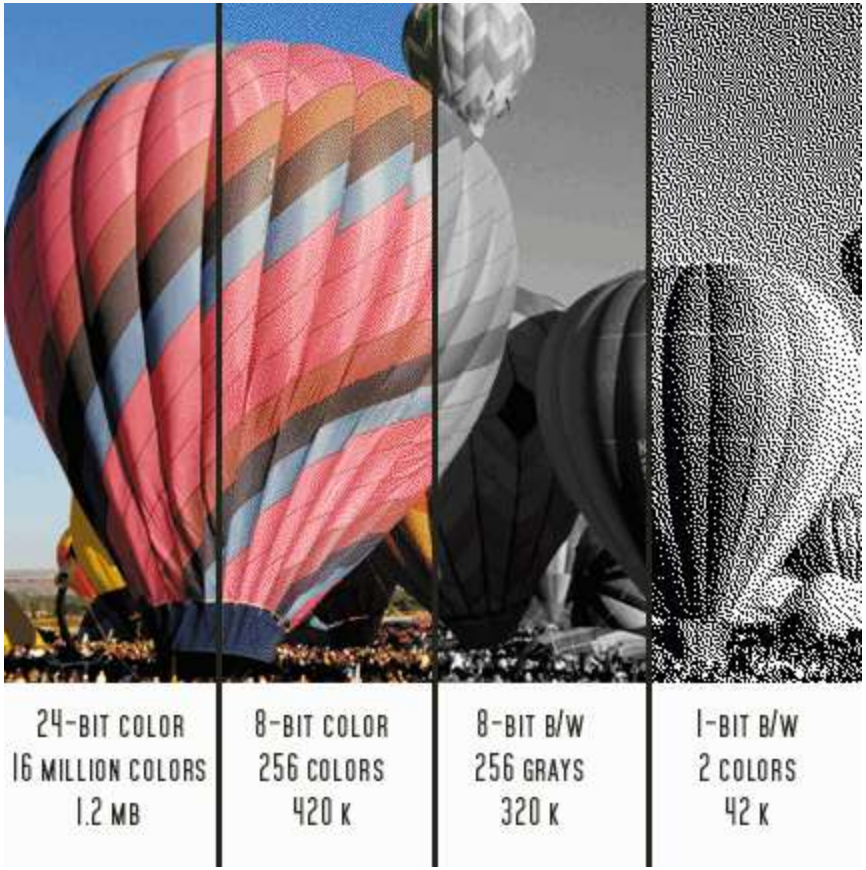




Bit Rate - Resolution

- Bit rate represents the amount of information (bits) that is stored per unit of time in a recording.
- Expressed in bits per second (bit/s or bps).
- The higher the bit rate, the greater the amount of information being transmitted. This means higher quality.
- In Digital Multimedia bit rate is the number of bits used per unit of playback time to represent audio or video after source coding (data compression).
- The encoding bit rate of a multimedia file is the size of a multimedia file in bytes divided by the playback time of the recording (in seconds), multiplied by eight.

A picture is worth a thousand words …



Common Audio Sampling Rates

Sampling Rate	Use
8,000 Hz	Telephone and encrypted walkie-talkie, wireless intercom and wireless microphone. Adequate for human speech but without sibilance.
16,000 Hz	Wideband frequency extension over standard telephone narrowband 8,000 Hz. Used in most modern VoIP and VVoIP communication products.
22,050 Hz	AM Radio quality. Speech recording.
44,100 Hz	Audio CD, MPEG-1 audio (VCD, SVCD, MP3), Pro audio gear.
48,000 Hz	The standard audio sampling rate used by professional digital video equipment. Also used for sound with consumer video formats like DV, digital TV, DVD, and films. Professional audio gear.
96,000 Hz	DVD-Audio, BD-ROM (Blu-ray Disc) audio tracks, HD DVD (High-Definition DVD) audio tracks. Pro audio gear and audio on professional video equipment.





How many Bits to use?


32-bit


24-bit


16-bit




8-Bit

freescale™

What Bit Rate should I use?

Mono - Stereo

128-320kbps

64kbps

32-48kbps

24kbps

Common Audio Bit Rates

Bit Rate	Use
800 bit/s	Minimum necessary for recognizable speech
8 kbit/s	Telephone quality
32 kbit/s	AM quality MP3
128 – 192 kbit/s	Standard MP3 bitrate quality used.
224 – 320 kbit/s	VBR to highest MP3 quality
1411.2 kbit/s	Compact Disc Digital Audio (Stereo)

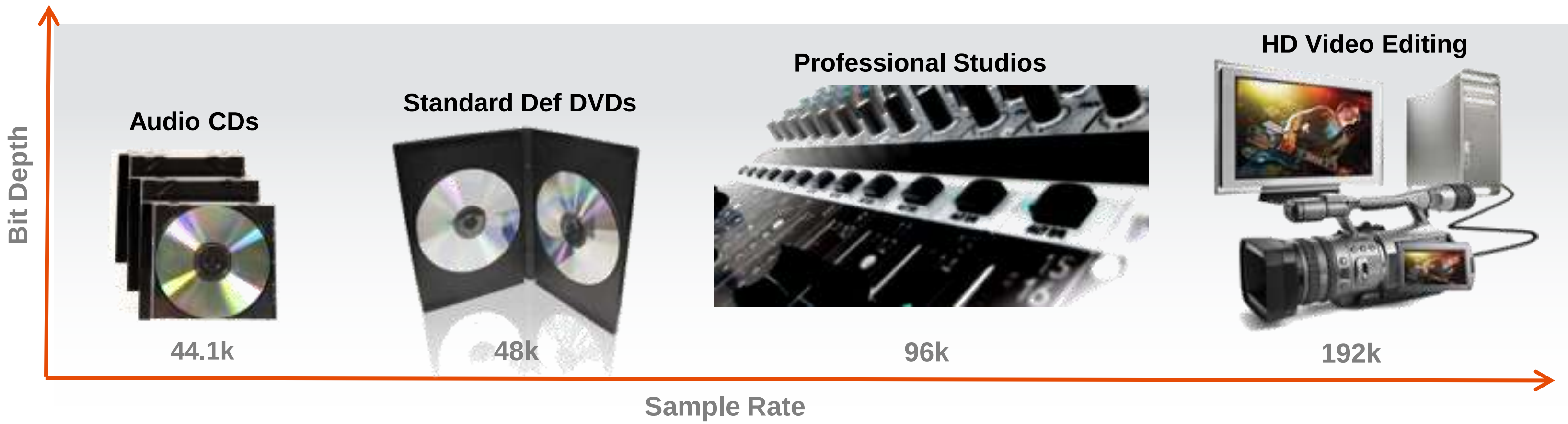
- To fully define a sound file's digital audio bit rates: the format of the data, the sampling rate, bit depth, and the number of channels (e.g. mono, stereo, four-track), must be known.

Bit Rate = Sample Rate x Bit Depth x No. of Channels

File Size (bytes) = Bit Rate x Seconds / 8

How does this impact file sizes?

Bit Depth	Sample Rate	Bit Rate	File Size for 1 minute stereo mix	File Size for 3 minute stereo mix
16	44,100	1.35 Mbit/sec	10.1 Mb	30.3 Mb
16	48,000	1.46 Mbit/sec	11.0 Mb	33 Mb
24	96,000	4.39 Mbit/sec	33.0 Mb	99 Mb
MP3 File	128 k/bit rate	0.13 Mbit/Sec	0.94 Mb	2.82 Mb



Analog or Digital?

- Since the introduction of the Compact Disc in the 1980’s, Digital Technology has become the standard for recording and storage of Audio.
- But… Why?
  - Digital signals are robust.
  - Digital signals can be transmitted/copied without distortion.
  - Digital signals can be played back without carrier degradation.
  - Digital signals can be easily manipulated.
  - Digital signals can be obtained in different options/formats.
  - Generally speaking, “*Digital*” has become synonymous with “*High Sound Quality*”

Audio Fidelity at Lower Cost

- What are the attributes for good audio?
  - Traditional metrics can be deceiving…






  - Richness, detail and clarity…

Key Audio Parameters

Level:

Different audio devices respond to different input levels, this by producing a given output level, certain output distortion, etc.

Frequency Response:

Output levels of an audio device when stimulated with different frequencies of known level.

Total Harmonic Distortion plus Noise (THD+N):

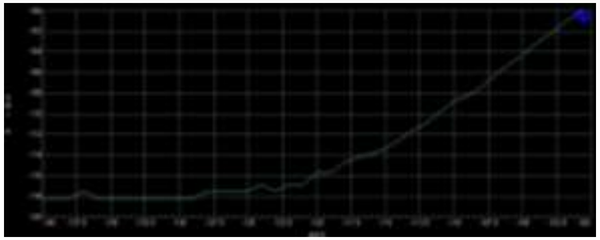
Unwanted addition of new tones and noise to the audio signal.

Phase:

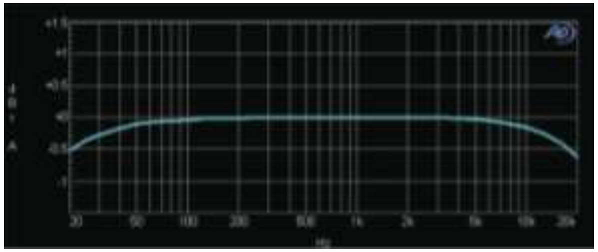
Positive or negative time offset in a cycle of a periodic waveform.

Crosstalk:

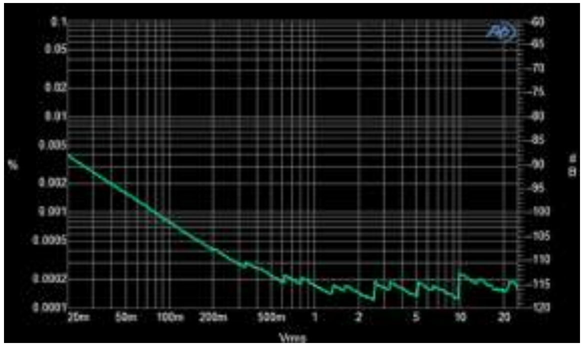
Signal of one channel appearing at a reduced level in another. It is largely the result of capacitive coupling between channel conductors in the device..



Output vs. Input Level



Frequency Response



THD+N vs. Level

Headroom, Dynamic Range and Noise Floor

Signal to Noise Ratio (SNR):

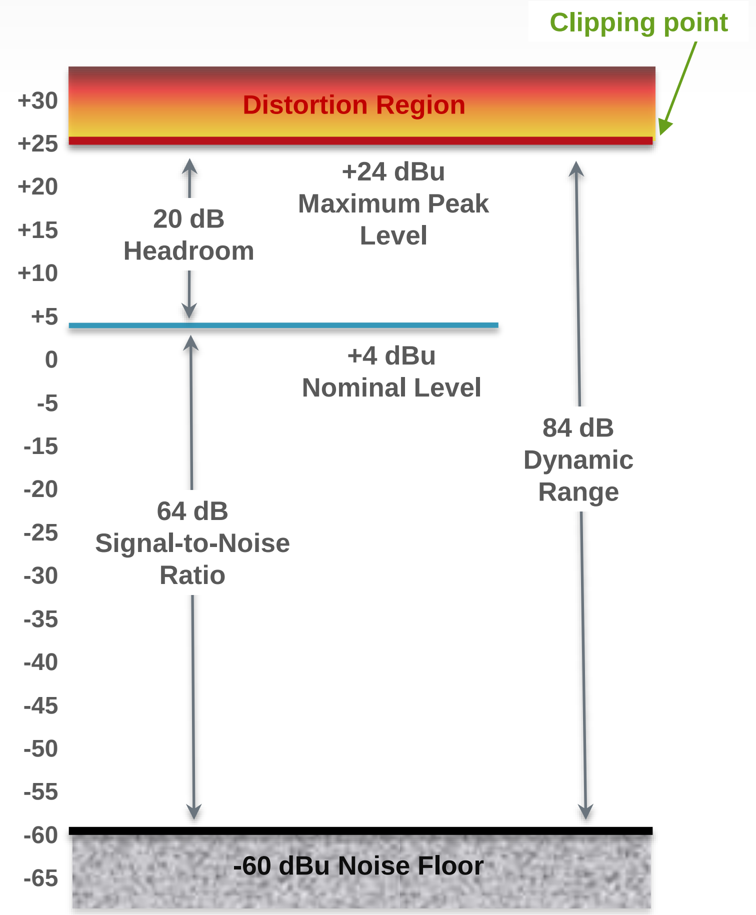
Level of the signal with respect to noise.

Dynamic Range:

Ratio between the strongest un-distorted signal and the minimum discernable signal.

Headroom:

Difference between SNR and Peak Maximum Signal Level (above which signals begins clipping).



Dynamic Range Comparisons

Audio Device/Application	Dynamic Range
AM Radio	48 dB
Analog Broadcast TV	60 dB
FM Radio	70 dB
Analog Cassette Player	73 dB
Video Camcorder	75 dB
ADI SoundPort Codecs	80 dB
16-bit Audio Converters	90 to 95 dB
Digital Broadcast TV	85 dB
Mini-Disk Player	90 dB
CD Player	92 to 96 dB
18-bit Audio Converters	104 dB
Digital Audio Tape (DAT)	110 dB
20-bit Audio Converters	110 dB
24-bit Audio Converters	110 to 120 dB
Analog Microphone	120 dB

How to achieve High-Fidelity?

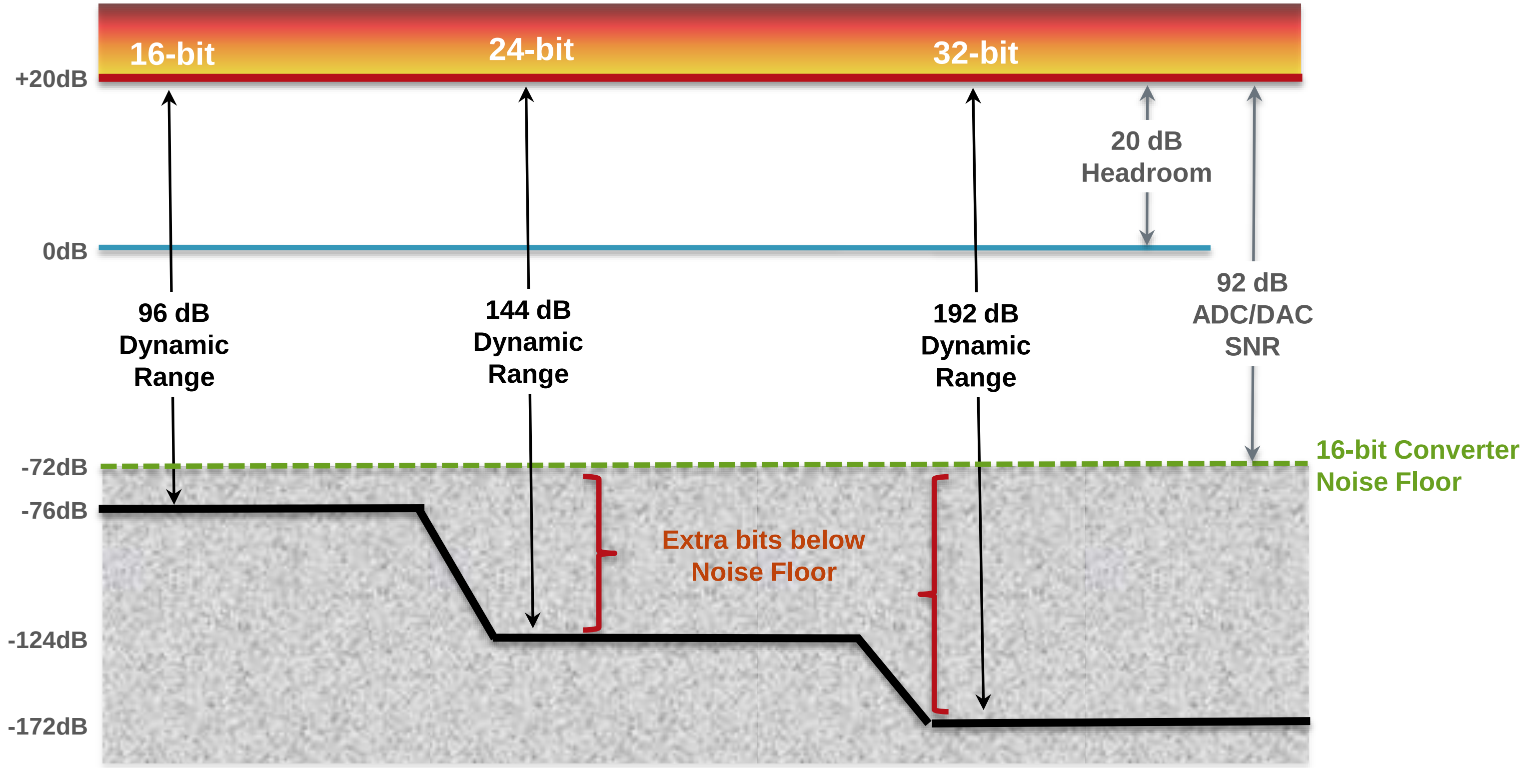
- High-fidelity audio processing can be categorized into:
  - "Consumer CD-Quality":16-bit Codec, with Dynamic Range around 85-93 dB.
  - "Professional-Quality": 20 to 24-bit conversion with Dynamic Range between 110-112 dB.

- Identify the "weakest link" in the chain:
  - Analog Input Signal, ADC or DAC word size, MCU/DSP processing, Post-Processing and Output Analog Circuitry.

- Digital Filter/Algorithm's SNR should be greater than the rest of the components in the chain.
  - With either Lower bit MCU/DSP using Double-Precision Math or Higher bit MCU/DSP using Single-Precision Math.
  - This decision should be based on application and throughput needs.

Fixed-Point Dynamic Range

- In theory, every 1-bit of resolution equals to 6 dB of Dynamic Range.
- The higher number of bits used to process an audio signal will result in a reduction in quantization error (noise).



Audio Signal Chain

- **Device Hardware** (Networking components, Processor, Memory, ADC/DAC, etc).
- **Digital Audio Interfaces:** The fundamental building block of digital audio is its *transport*.
  - Inter IC Sound™ (I2S)
  - Sony/Philips Digital Interface (S/PDIF)
  - AES/EBU (AES3)
  - Universal Serial Bus (USB Audio)
  - Firewire (IEEE1394)
  - HDMI (High Definition Multimedia Interface)
- **Networking** technology/connection method (e.g. ENET, WiFi)

Signal Chain Hardware

Codec:

- Software codec is used to encode/decode digital formats (i.e. MP3). In the hardware world, it is an interface to and from the analog domain. There are three audio converter types.
  - Analog-to-digital converter (ADC)
  - Digital-to-analog converter (DAC)
  - ADCs and DACs combined on the same device (CODEC).

Control Interface:

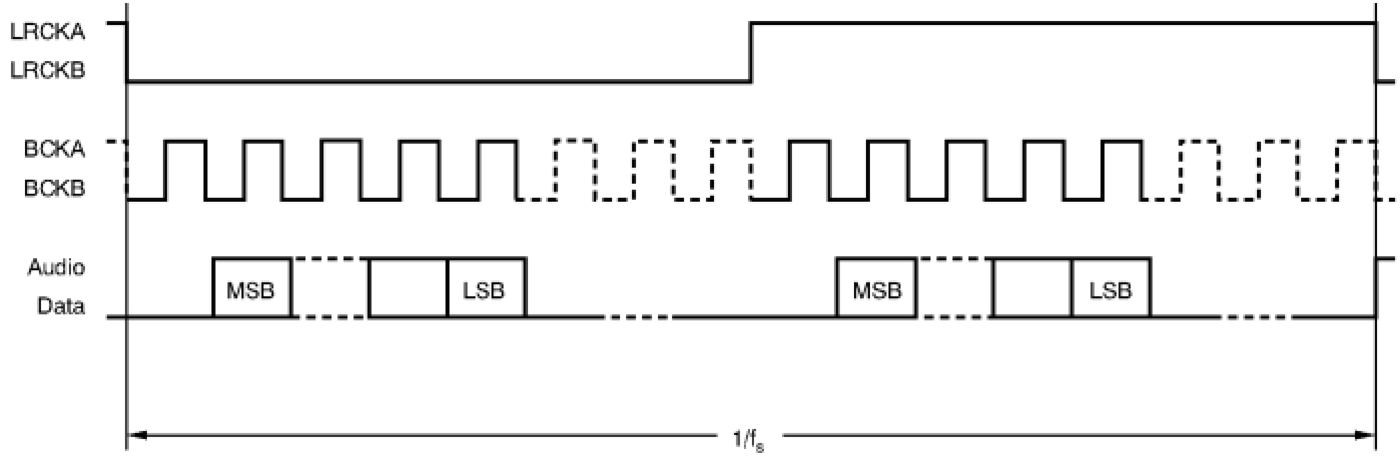
- There are a variety of control interfaces. Simple converters typically come with hardware control interfaces. Control pins are usually tied to V_{DD}, GND or GPIO processor pins.
- Software-controlled interfaces are typically driven by either I^2C or SPI serial ports found on MCUs and DSPs.
- Devices driven in software mode usually offer more flexibility than their hardware-controlled cousins. Software-controlled converters usually have registers inside that can be written from an external source

Inter IC Sound (I²S)

- I²S is the most commonly used format for transporting audio between ICs: three signals, data, bit clock and wordclock. I²S typically transports 24-bit PCM audio in stereo.

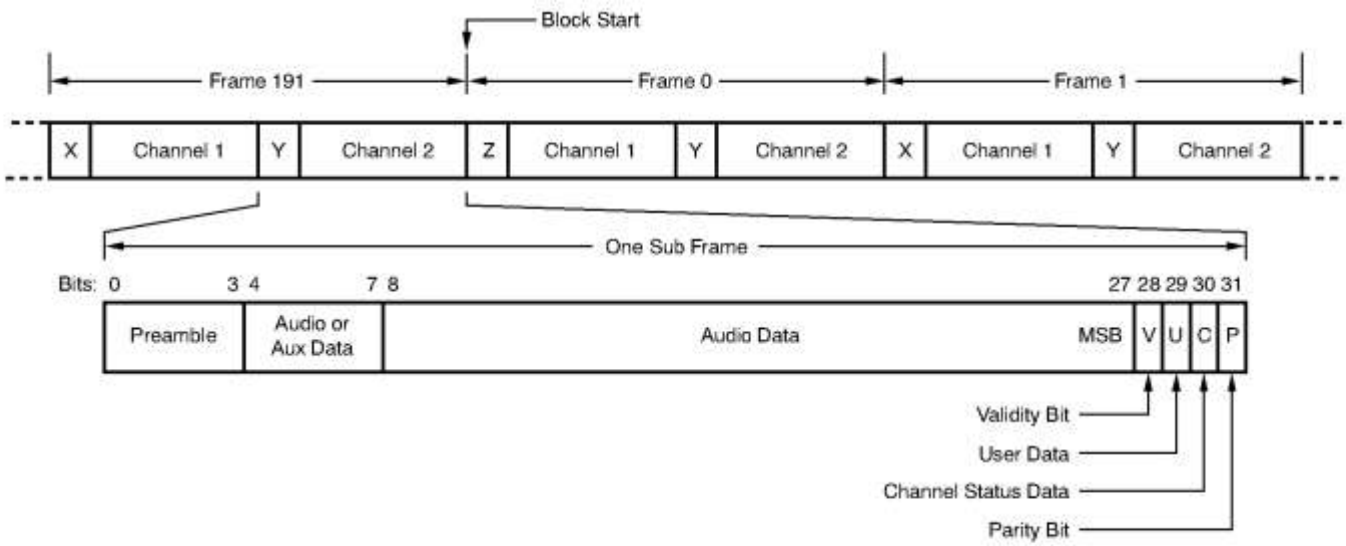
- I²S is a very simple structure to understand. Typically transmitted in a block with two subframes, each subframe is usually 32 bits long where each frame represents the left or right channel in a stereo pair.

- The bit clock is used by the receiver to differentiate between a string of 1s or 0s. The data itself is a serialized 16- or 24-bit value of the audio sample. The left/right clock (LRCK or wordclock) tells the receiver whether it's dealing with the left channel of data, or the right. Most serial ports on DSPs can handle I²S easily.



Sony/Philips Digital Interface (S/PDIF)

- S/PDIF is a single-ended signal, either optical or coaxial. It is found on virtually every DVD, set top box, gaming consoles, and anything else that connects to a TV or home theater system.
- In its most vanilla format, S/PDIF can transport up to 24-bit PCM audio in stereo. Encoded data streams from the DVD can be transmitted instead of regular PCM data. An example is Dolby™ AC-3 data.
- Additional data is also sent in the data stream such as recorded sample rate, source data, copy validity and parity bits. The data is sent in blocks of 192 frames. Each frame has two sub-frames (left and right channels)
- Each Channel Status bit is mapped with a different meaning. In consumer S/PDIF, this can signify things like "copy protected" or "source data rate."



S/PDIF Interface Applications

- The S/PDIF interface has two primary purposes:
  - Transfer compressed digital audio and carry the signal between the output of a computer or DVD player to a home theater system designed for Dolby Digital or DTS surround sound.
  - The S/PDIF interface is extensively used to inter-connect commercial/professional audio equipment.
- The main advantage of SPDIF Digital transference vs. original analog transmissions is noise immunity.

AES/EBU (AES3)

- AES/EBU is the professional audio version of S/PDIF, and is electrically based on the RS422 standard.
- It is able to carry two channels of PCM audio over several different transmission
mediums including balanced and unbalanced lines and optical fiber. Normally it uses XLR connectors.
- Many consider it to be the "balanced version of S/PDIF." The logical format of AES/EBU is identical to consumer S/PDIF. However, the 192 bits generated in the channel status word are mapped differently.
- An S/PDIF transmitter or receiver can be used with S/PDIF or AES/EBU. However, many devices require conversion to CMOS voltage levels to work with these ICs.
- For AES/EBU, an isolation transformer can be used to help with issues such as common-mode noise and grounding issues between units.

USB Audio

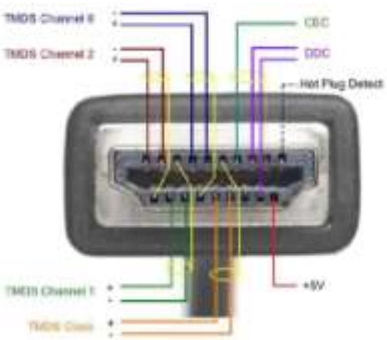
- At a high level, upon connection to the host, the USB device shares its descriptor data (i.e., what it can do, what it needs, etc.) in addition to its identification (VID/PID). The host loads special drivers for matching VID/PID, if required. This process is called enumeration.
- USB standard supports three different transfer speeds:
  - 1.5 Mb/s – Low speed
  - 12 Mb/s – Full Speed
  - 480 Mb/s – High speed
- Regarding channel bandwidth to USB products, the basic channel requirements (without any overhead) are:
  - Number of channels × bit depth × sampling frequency (e.g. CD audio = 1.4 Mb/s)
- However, USB overheads can be rather intense, taking up to 25 percent additional bandwidth.
  - Half-duplex, six-channel, 16 bits, 48 kHz (home theater) is possible with full speed devices (4.6 Mb/s).
  - Full-duplex, stereo, 24 bits, 96 kHz is virtually impossible at full speed (9.2Mb/S), but easily done at high speed.
- Windows™, Mac™ and Linux currently have prewritten drivers for basic USB audio codecs.

Firewire

- Running at a basic rate of 400 Mbps, Firewire appears to have plenty of bandwidth for streaming multiple audio channels. In fact, within the pro audio market, it has been the interface of choice for many years.
- However, the rewards of Firewire don't come for free. Today, there are no "single chip, drop-in" IC's on the market that support Firewire for streaming audio. The solution requires significant programming skills, as Firewire relies heavily on its interface ICs to manage the low-level protocols. This is different to USB, where the host CPU takes care of the low-level interface (and in many cases by the operating system manufacturer).
- Alternatively, the benefit of the external IC's low-level protocol is that it places less work on your system's CPU. It, frees up your CPU to do more audio processing and run with fewer interrupts. In fact, sustained transfer tests of a USB 2.0 high speed (480Mb/S) versus FireWire 400 interface, show that USB 2.0 high-speed rarely exceeded sustained transfers of over 280 Mbps. However, for the number of channels being transferred, this may not be a major concern.
  An advantage of 1394 is it can run over CAT5 cabling (with the right companion cable EQ IC), giving it a significant advantage over USB (USB's limit is nearer to five meters). Companies such as TC Applied Technologies and their Dice II family of devices have simplified streaming I^2S audio in and out of a PC significantly.

HDMI - High Definition Multimedia Interface

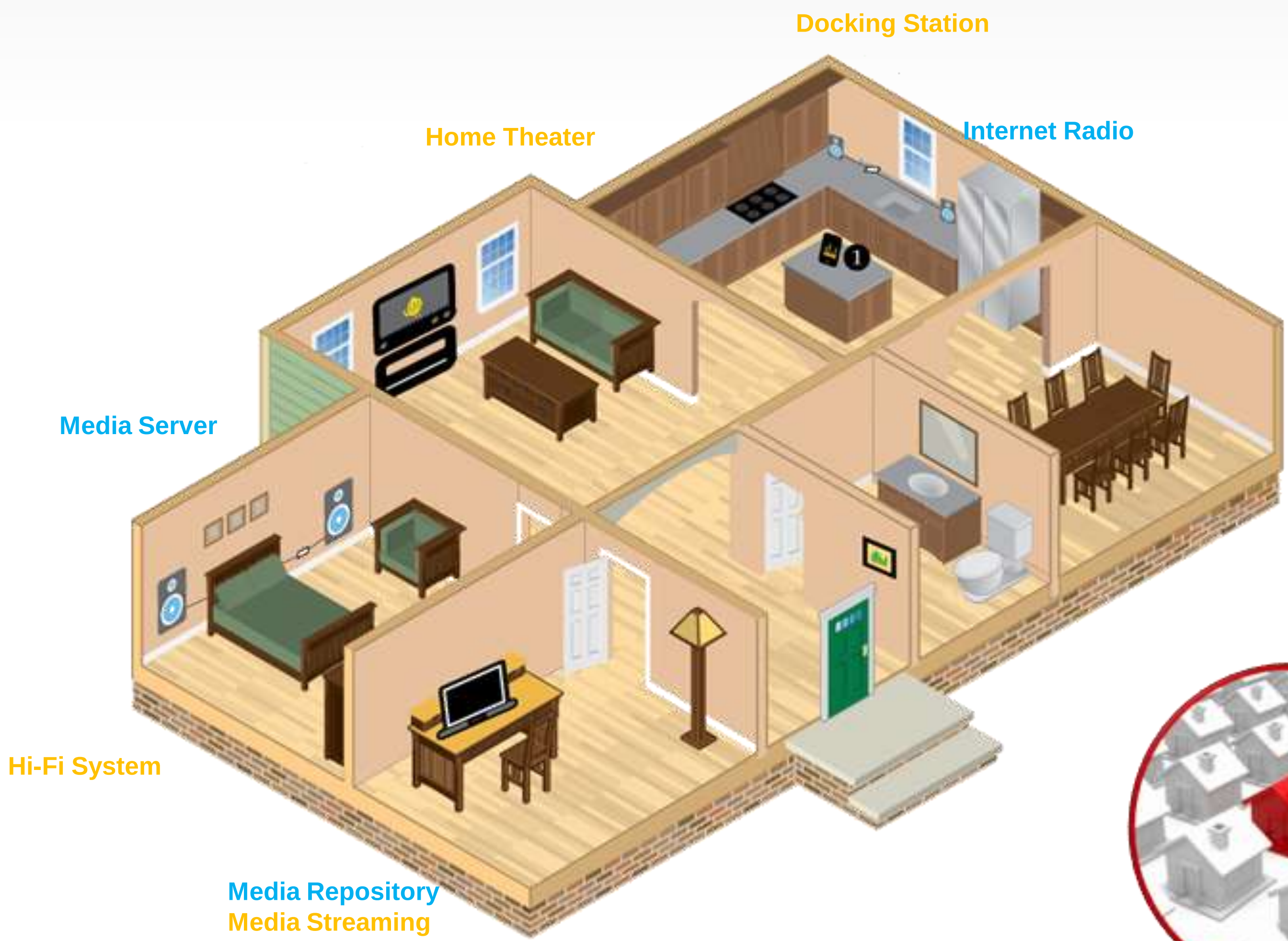
- A 19-pin digital connection that transmits both high-definition uncompressed video and multi-channel audio through a single cable.
  - Pins 1-9 carry the three TMDS data channels (Transition Minimized Differential Signaling – allows to send high-speed digital data). Three pins per channel (+,- and ground/data shield). TMDS data includes both video and audio information.
  - Pins 10-12 carry data for the TMDS clock channel, which helps keep the signals in synchronization. Also uses three separate lines.
  - Pin 13 carries the CEC (Consumer Electronics Control) channel, used for sending command/control data between connected devices.
  - Pin 14 is reserved for future use.
  - Pins 15/16 for DDC (Display Data Channel). Used for communicating EDID (Extended Display Identification Channel) between devices.
  - Pin 17 is a data shield for the CEC and DDC channels.
  - Pin 18 carries a low-voltage (+5V) power supply.
  - Pin 19 is the Hot Plug Detect, for monitoring power up/down and plug/unplug events.
- HDMI is the preferred connection for HD devices.
- HDMI 1.4 supports maximum clock rate of 340 MHz, and a maximum audio throughput of 36.86 Mbits/s.



Audio Networking

- Two or more devices connected (wired or wireless) to allow access to or distribution/sharing of digital audio content
- Examples:
  - LAN-connected network media player playing streamed content from local PC or online server.
  - Multi-channel Audio Distribution among equiment/components in a recording studio.
- Applications: Personal/Home Media, Broadcast, Live Sound, Sound Reinforcement, Studio Recording, Commercial Sound Distribution, Networked Musical Performance.

Networked Audio



Audio Networking Technologies

- Layer 1 (Physical)
  - A-Net (Aviom)
  - MaGIC (Gibson)
- Layer 2 (Data)
  - AES51
  - Ethernet AVB
  - CobraNet
  - EtherSound
- Layer 3 (Network)
  - Dante (Audinate):
  - Q-Sys networking (Q-LAN)
  - UPnP/DLNA
  - DAAP/iTunes

Agenda

- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demo
- Freescale MCU audio solutions

USB Audio Class 1

USB Audio Class 1 standard (1998)

- Supports up to 24 bits / 96 kHz max
- No limitation on sample rate
- Class 1 is tied to USB 1 Full Speed = 12 MHz

- Every millisecond a packet is send
- Maximum packet size is 1024 bytes
- 2 channel x 24 bit x 96000 Hz sample rate= 4608000 bits/s or 576 Byte/ms
- This fits in the 1024 byte limit
- Any higher popular sample rate e.g. 176 kHz needs 1056 bytes so in excess of the maximum packet size

- All operating systems (Win, OSX, and Linux) support USB Audio Class 1 natively
- All support 2 channel audio with 24 bit words and 96 kHz sample rate

USB Audio Class 2

USB Audio Class 2 standard (2009)

- Downwards compatible with class 1
- USB Audio Class 2 additionally supports 24 and 32 bit and all common sample rates
- Class 2 uses High Speed (480 MHz), requiring USB 2 or 3.
- At 480 MHz, it is possible to transfer 60 channels at 24 bits at 96 kHz (132 Mbit/s)
- USB audio class 2 drivers are available in OSX from 10.6.4 and Linux since mid 2010
- Both support sample rates up to 384 kHz.
- Windows Vista and Windows 7 support USB Audio 2
- Windows XP requires a third party USB class 2 driver
  - Thesycon and Centrance have developed a USB Class 2 Audio driver for Windows

Isochronous transfer

- The source device reads the data from local storage (HD or Flash) and stores it in a streaming buffer
- It is then continuously streamed from the buffer to the USB port (Isochronous mode).
- Data is sent out in frames every millisecond.
  - Frames are sent even if the buffer is empty
  - The rate at which the frames go out is determined by a oscillator driving the USB bus
  - This rate may be independent of other source device systems
- This should guarantee a constant flow of the frames
  - System priorities and interrupts may interfere
- Three types of synchronization modes are available for USB isochronous transfers

USB Isochronous Transfers

Synchronous

- The destination device DAC clock is synchronized to the 1 kHz frame rate
  - In practice limited to 48 kHz
  - Tends to be jittery
- Considered the least desirable isochronous mode

Adaptive

- The destination device has a local clock, adjusted according to the average rate of the incoming data stream
  - Less susceptible to jitter, but still not fully stable
  - Synchronization and stabilization often results in large initial latency
    - A variation on Adaptive uses a destination fixed local clock and an Audio Sample Rate Converter (ASRC) to match the rates of the arriving data stream to the local DAC

Asynchronous (Acknowledged as the most reliable and desirable implementation)

- The destination device has a local clock
- The destination device has a receive buffer and signals the source to adjust the rate of data delivery according to the amount of data available in the buffer
  - No jitter, as stable as the local clock
  - Low latency (the receive buffer can be kept small)

USB Audio Interconnect

- USB Interconnect is used for:
  - Audio input devices, such as microphones
    - May also be used to power the input device
  - Headsets for telephony and gaming
  - Audiophile headphone systems
  - PC audio accessories
  - Smartphone and Tablet audio accessories

Agenda

- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demos
- Freescale MCU audio solutions

USB Speaker Demo

- USB Isochronous Asynchronous audio streaming

USB Headset Demo

- USB Isochronous Synchronous audio streaming
- Input and Output

TWR-DOCK Demo

- USB Isochronous Asynchronous audio streaming for output
- USB Isochronous Synchronous audio streaming for input

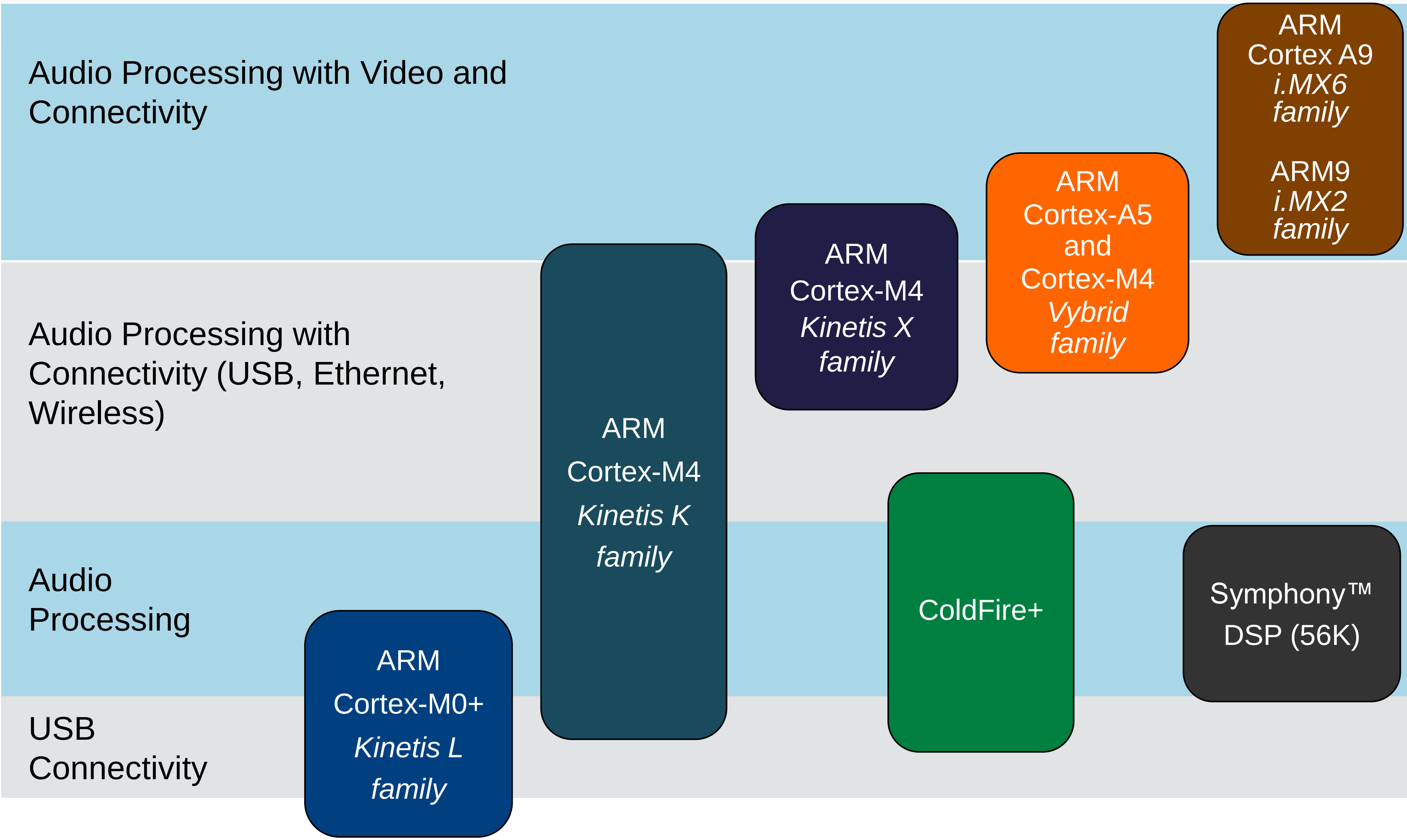
WAV Player

- Graphical User Interface with touch screen
- Media playback from SD memory card

Agenda

- Audio systems evolution
- Audio streaming overview
- Options for USB Audio streaming
- Freescale USB Audio streaming demo
- Freescale MCU audio solutions

Accessories and Digital Audio Processors



freescale™

Accessories and Digital Audio ARM Based Processors

Audio Processing with Video and Connectivity

Audio Processing with Connectivity (USB, Ethernet, Wireless)

Audio Processing

USB Connectivity

ARM Cortex-M0+ *Kinetis L family*

ARM Cortex-M4 *Kinetis K family*

ARM Cortex-M4 *Kinetis X family*

ARM Cortex-A5 and Cortex-M4 *Vybrid family*

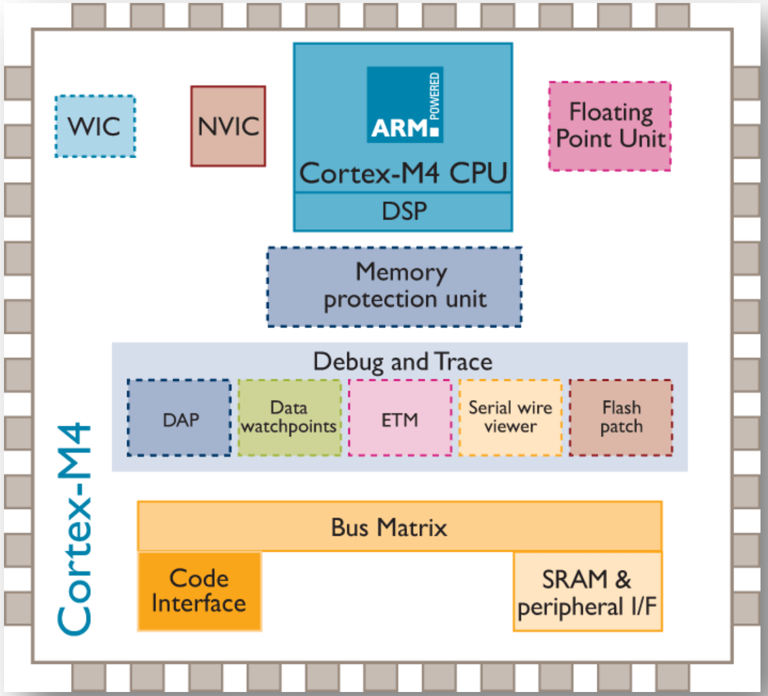
ARM Cortex A9 *i.MX6 family*

ARM9 *i.MX2 family*

freescale™

ARM Cortex-M4 Processor Microarchitecture

- Backwards compatible with ARM Cortex-M3
- New features
  - DSP extensions
  - Single precision floating point unit
- Freescale IP and innovation
  - Available on-chip cache for instructions and data enhanced performance reaching zero wait states
  - Crossbar switch for concurrent multi-master/slave accessing improves system throughput
  - MPU with multi-master protection enhances system safety and security
  - Low-leakage wake-up unit adds flexibility for low-power operation
- Architected for digital signal processing
  - **Motor Control** – advanced algorithms, longer lifespan, power efficiency
  - **Automation** – high calculation and algorithm bandwidth at a low cost
  - **Power Management** – designed for low/battery-powered systems
  - **Audio and Video** – 5x performance improvement over software, helping batteries last longer



freescale™

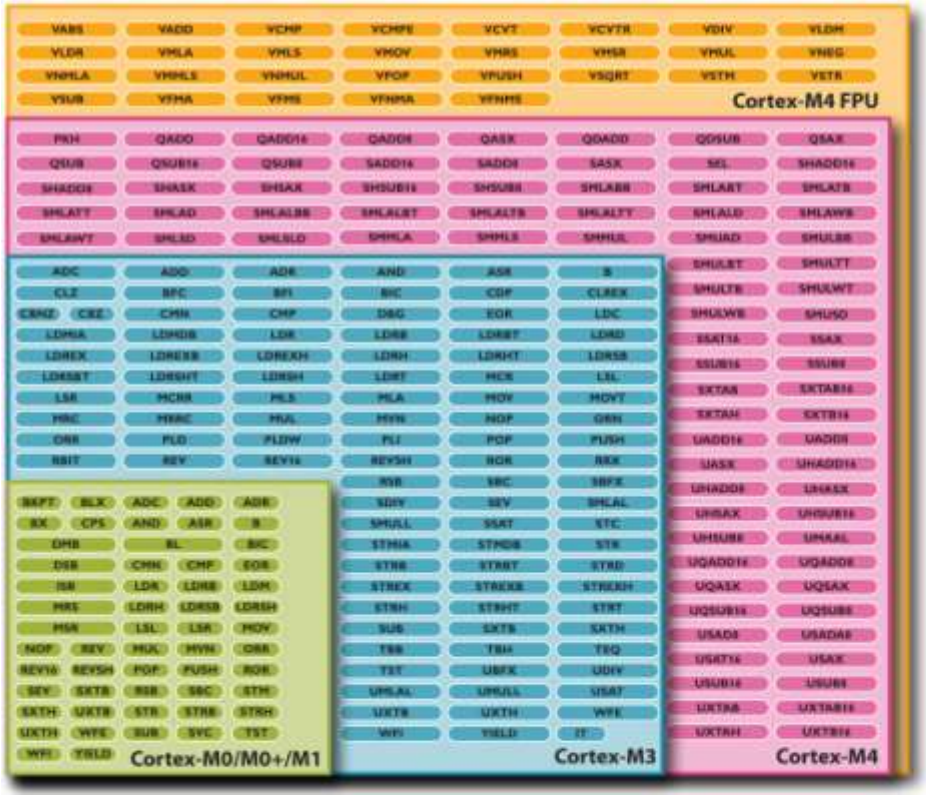
Reduced Instruction Set

Cortex-M0+

Cortex-M0

E A S E O F U S E

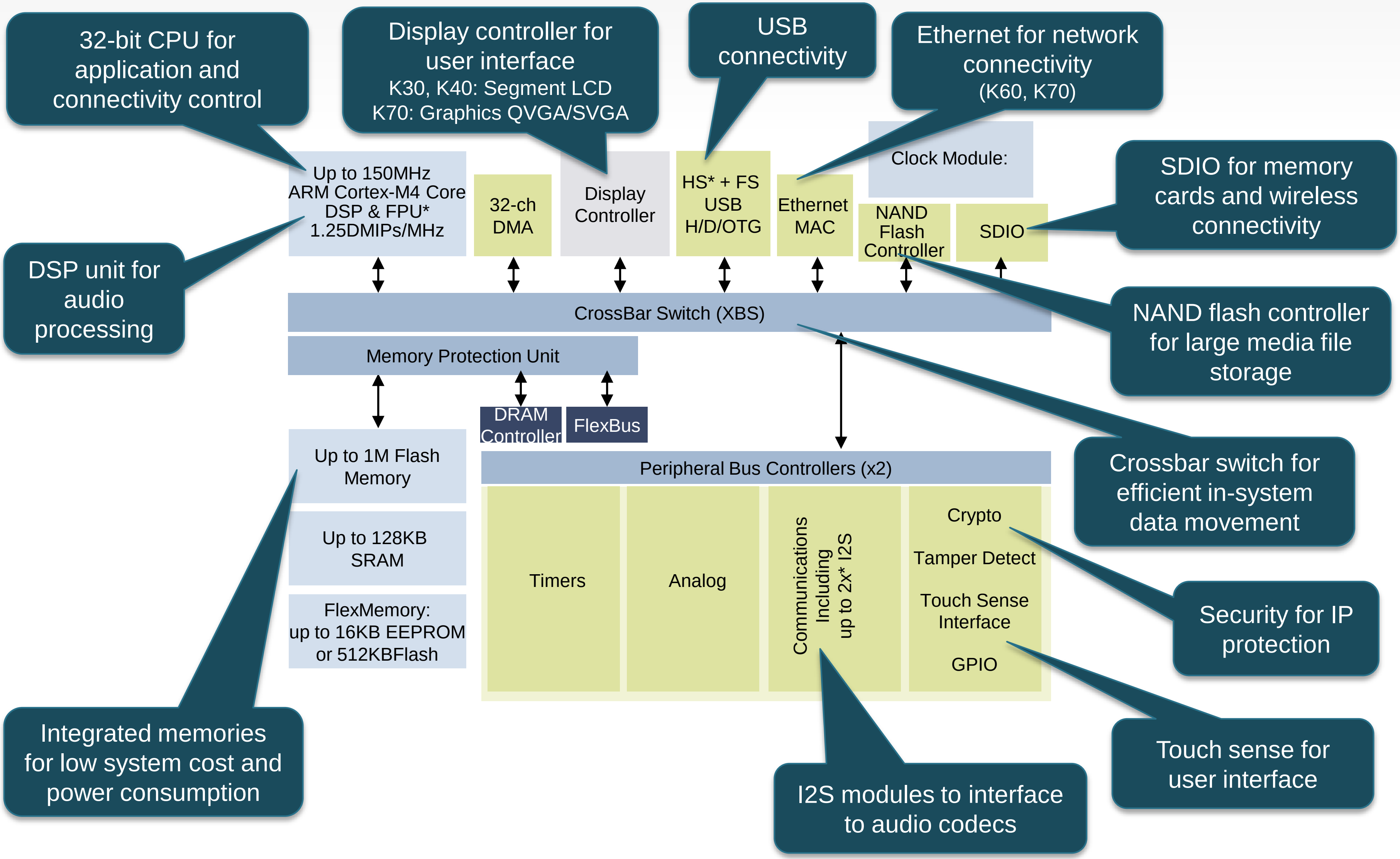
- Only 56 Instructions
- Mostly coded on 16-bit
- Operate on the 32-bit registers
- Option for fast MUL 32x32 bit in 1 cycle



S C A L A B I L I T Y

- **Fully upward compatible to Cortex-M3 and Cortex-M4**

Kinetis MCU



freescale™

Vybrid Controller Solutions

Cortex-A5 + Cortex-M4
Multicore

Cortex-A5 Only
Single Core

Feature Integration

VF7xx - Up to 500 MHz
Dual Heterogeneous Cores, Dual XGA LCD, Camera Interface, OpenVG GPU, Stereo Audio, DDR, Secure Boot, Tamper Detect, Dual USB OTG w/HS PHY, Ethernet

VF6xx - Up to 500 MHz
Dual Heterogeneous Cores, XGA LCD, Camera Interface, OpenVG GPU, DDR, Secure Boot, Tamper Detect, Dual USB OTG w/HS PHY, Dual Ethernet, L2 Switch

VF5xx - Up to 500 MHz
SVGA LCD, Camera Interface, OpenVG, Stereo Audio, DDR, Secure Boot, Tamper Detect, Dual USB OTG w/HS PHY, Dual Ethernet, L2 Switch

VF4xx - Up to 500 MHz
SVGA LCD, Audio, DDR, Secure Boot, Tamper Detect, Dual USB OTG w/HS PHY, Ethernet, NAND Flash Controller

VF3xx - Up to 266 MHZ
WQVGA LCD, Audio, Secure Boot, Tamper Detect, USB OTG w/HS PHY, Dual Ethernet, L2 Switch, Dual Quad-SPI, NAND Flash Controller

Rich Apps in Real Time

Performance

freescale™

F7xx: Consumer, Medical and Factory Automation

Core
Up to 500MHz ARM™ Cortex-A5 with TrustZone
Up to 167MHz ARM™ Cortex-M4

HMI
Dual TFT LCD up to XGA resolution

Memory
32KB I and D L1 Cache for A5, 16KB I and D for M4
512KB L2 Cache and 64KB TCM for M4
On Chip: up to 1.0MB SRAM . ECC support on 512KB
On Chip: LPDDR2/DDR3 DRAM controller
NAND Flash Controller

Analog
2 x 12-bit ADC (16-Ch), 2 x 12-bit DAC

Communication
6 x UART, 2 x CAN, 4 X SPI, 4 X I2C
1 Ethernet MAC with IEEE1588
Dual USB2.0 HOST and OTG with PHY

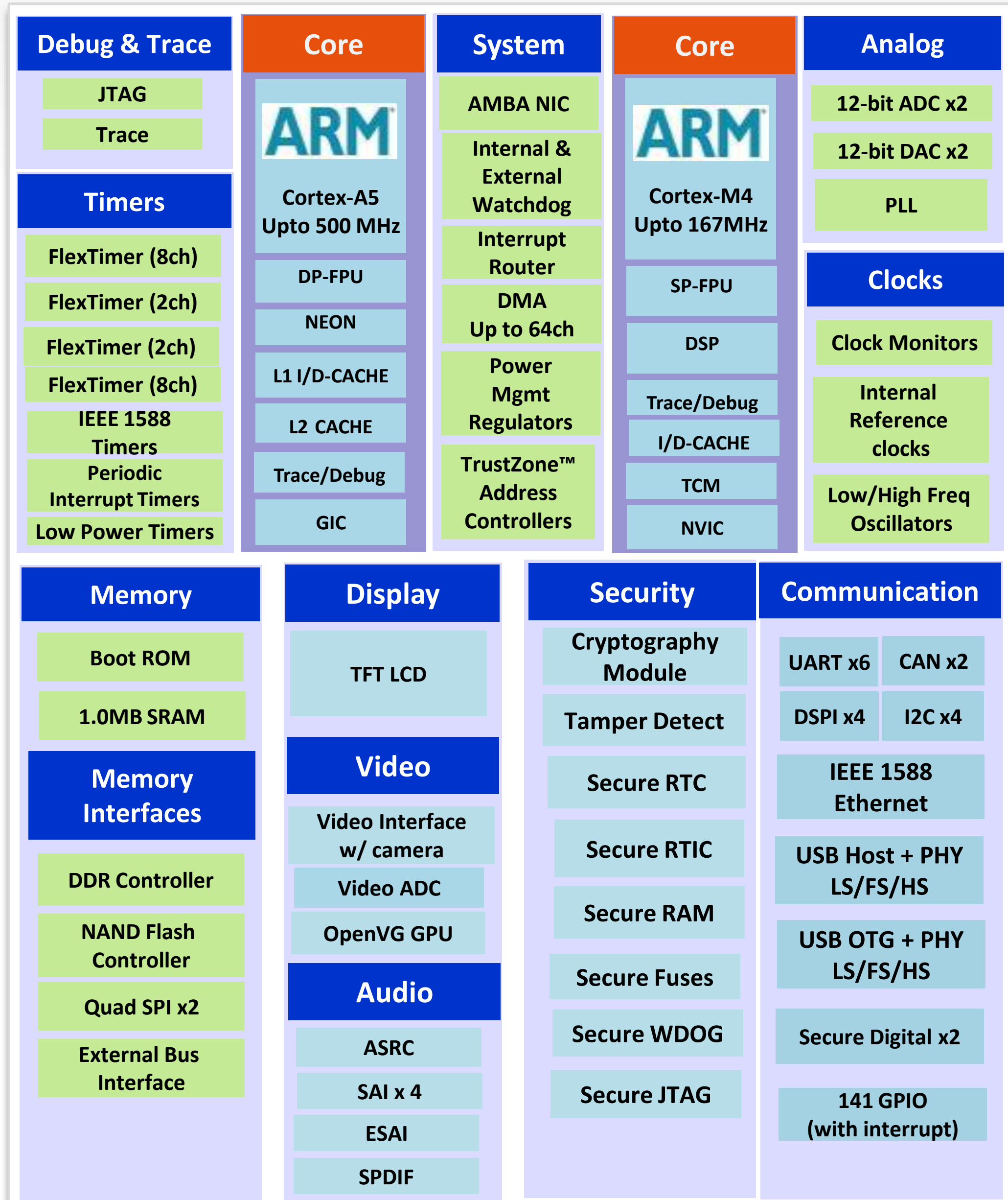
Audio
4 x SAI for full-duplex serial interfaces like I2S, AC97
ESAI – Enhanced Serial Audio Interface
SPDIF

Video
Video Interface unit with parallel camera interface and analog input
OpenVG GPU

Security
Tamper Detect, High Assurance Boot
True RNG

Power Management
Internal regulator (PMIC)

Package
17x17 0.8mm pitch 364-pin MAPBGA
Spec'ed to Freescale Industrial standards (-40 to 85C)

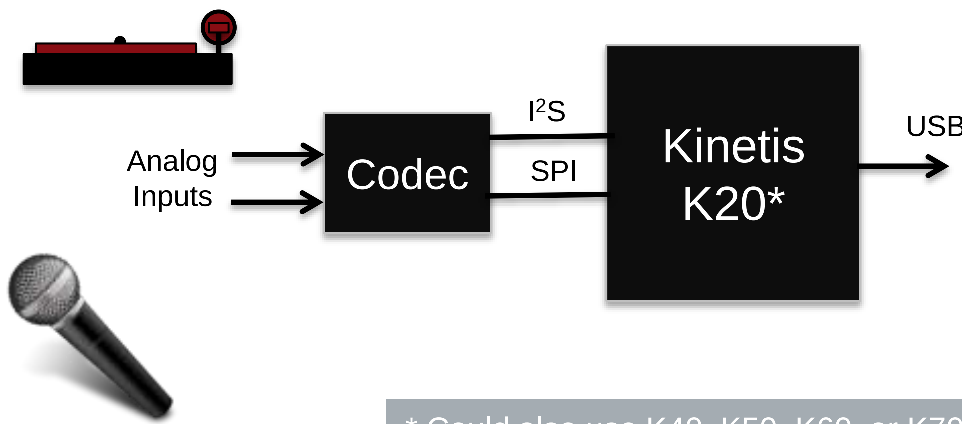


Kinetis USB Audio Input Device Solution

Audio Input Options:
- Microphone
- Line
- Pickup cartridge
- Guitar pickup
- Other analog pickup?

Functions:
- Gain control
- Equalization (e.g. RIAA)
- Sample rate selection

Output:
- Isochronous USB audio stream



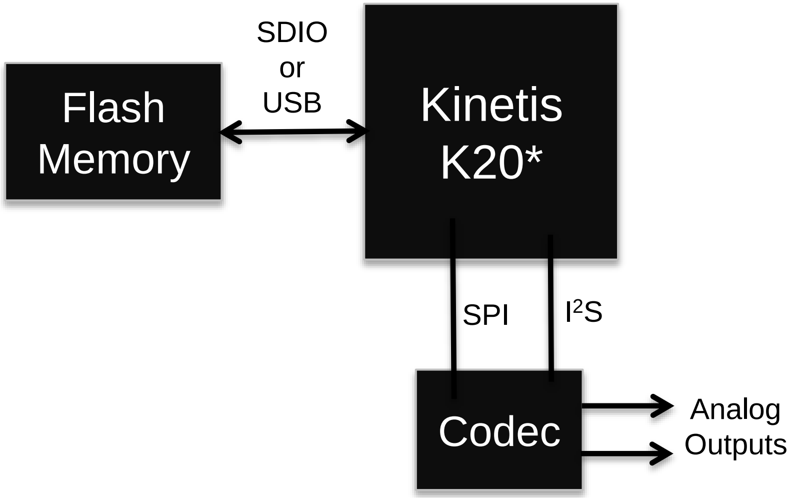
* Could also use K40, K50, K60, or K70

Kinetis Audio Player

Audio Input:
- WAV files
- MP3 files
- AAC files

Functions:
- File decode

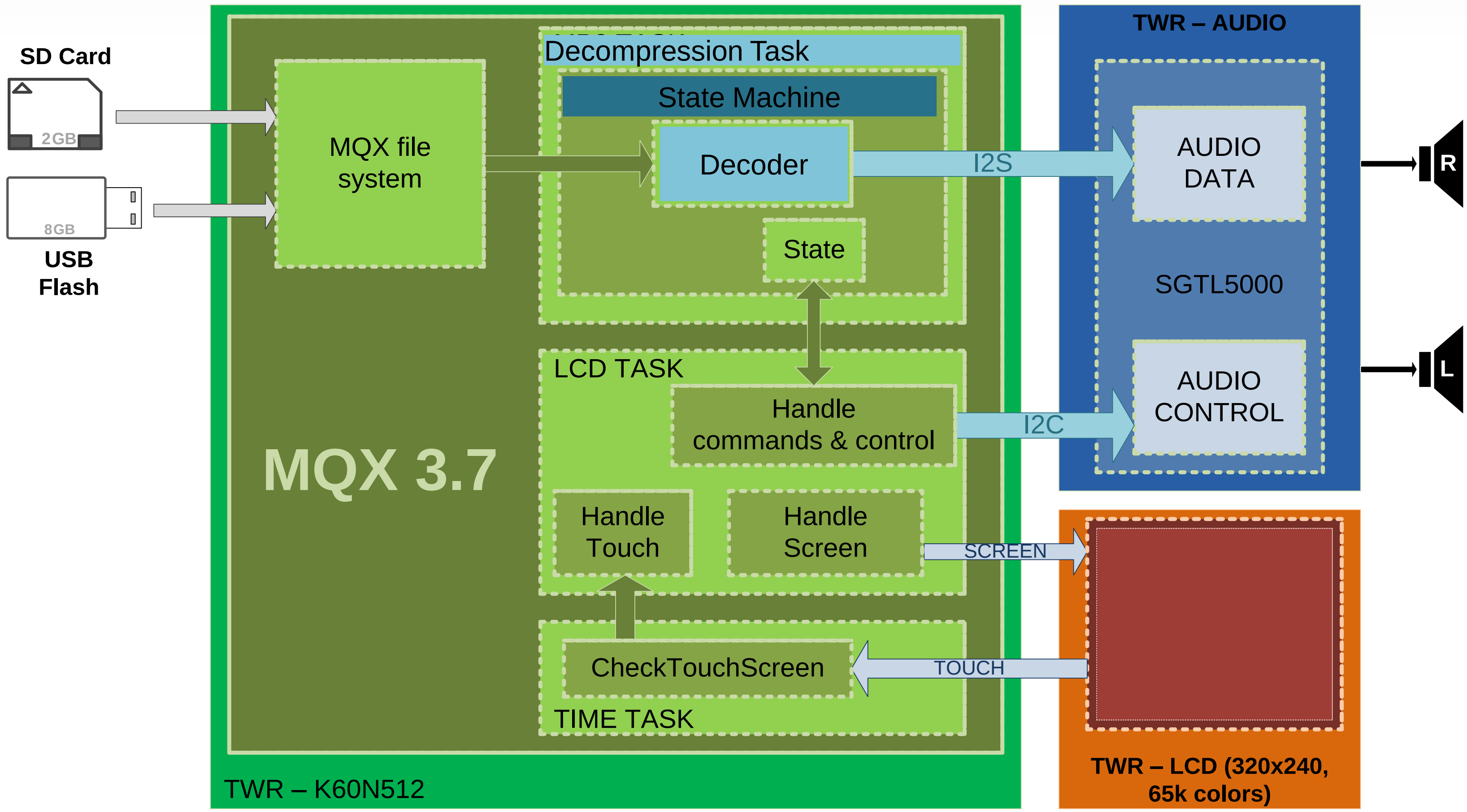
Output:
- Analog line



* Could also use K40, K50, K60, or K70

Kinetis K60 Audio Player Demo: Block Diagram

S/W MP3 decoding and playback under MQX with touch screen LCD using Kinetis K60 MCU



Demo available now; contact Freescale for more information

Kinetis K60 Audio Player Demo: Screen Features of Audio Player demo

Main features:

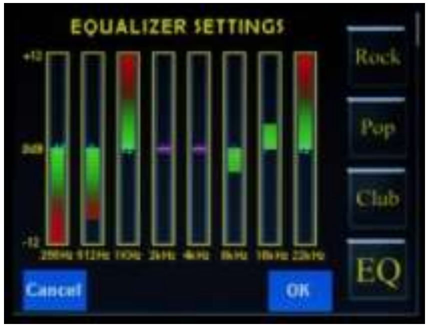
- basic control features (play, stop, pause, play next, play previous)
- basic song information display (title, artist, album, year, name of file)
- plotting of both channels samples in a time domain
- display of actual time and current position of song
- current position change possible using slide bar moving
- volume and balance control

Spectrum Analyser:

- whole frequency spectrum (~40Hz - 20kHz) is divided into16 frequency sub-bands
- display 16 frequency sub-band lines with 10 pixel resolution
- spectrum analyser is based on sub-bands data hidden in every mp3 frame
- each of sub-band line includes average value of both channels in a specific frequency band

Other:

- equalizer setting
- select from playlist



Demo available now; contact Freescale for more information

SoundBar Surround Processor

Audio Input Options:

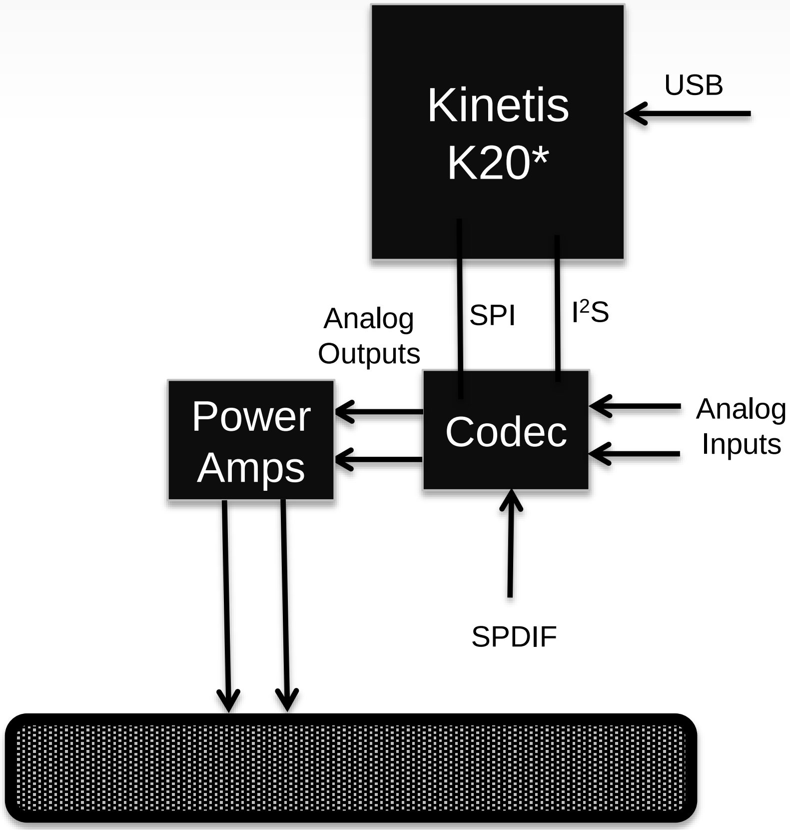
- SPDIF
- Isochronous USB audio stream
- Analog line

Functions:

- Surround processing, selection from
  - Dolby
  - DTS
  - SRS

Output:

- Analog to power amplifier



* Could also use K40, K50, K60, or K70

Demo with SRS or DTS available now, Dolby is in development
Contact Freescale for more information

Headphone Audio Processor

Audio Input Options:

- SPDIF
- Isochronous USB audio stream
- Analog line

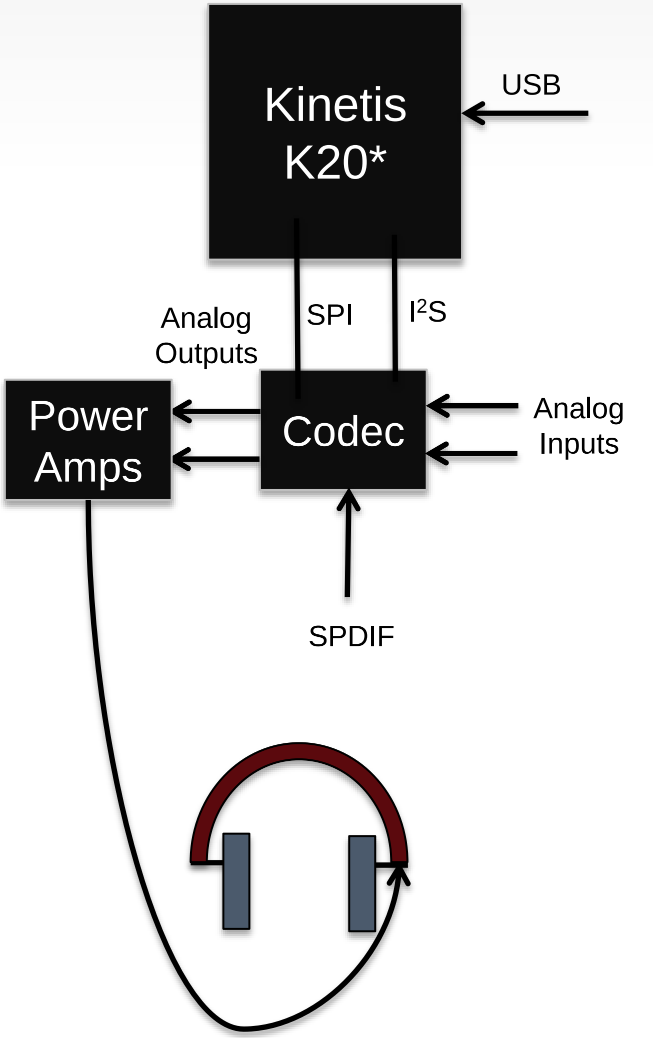
Functions:

- Surround processing, selection from
  - Dolby
  - DTS
  - SRS

Output:

- Analog to power amplifier

Demo with DTS and SRS are available now,
Dolby is in development
Contact Freescale for more information



* Could also use K40, K50, K60, or K70

Example: Kinetis K40 Tower SRS WOW HD SoundBar Decoder

Demo consists of: TWR-K40X256-KIT and TWR-AUDIO-SGTL

- Analog stereo audio input
- Analog stereo audio output
- USB available for digital audio connection
- Low-power capacitive touch sensing

Demo available now; contact Freescale for more information

freescale™

The Freescale Tower System

A modular development platform for 8-/16-/32-bit MCUs & MPUs

- Quickly combine Tower Modules to build a prototype of your application
- Modules sold individually or in kits
- Open Source: Build your own Tower Module to integrate your IP
- Cost-optimized hardware
- Software support from Freescale and Third Parties
- Growing community of Third Party hardware support
- On-line community: www.towergeeks.org

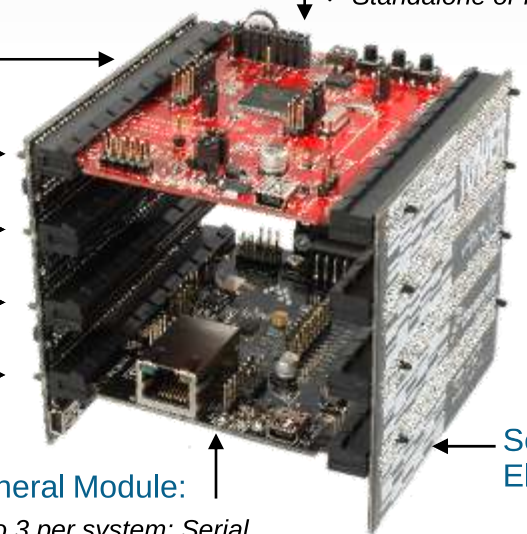
Rapidly build a prototype of your end application

Support for all ColdFire+ and Kinetis MCUs!

MCU/MPU Module:
- *Tower controller board*
- *Standalone or in Tower System*

Primary Elevator

Board Connectors



Secondary Elevator

Peripheral Module:
- *Up to 3 per system: Serial, Memory, LCD,..*
- *Mix & match with different MCU modules*

TWR-MEM

TWR-LCD

TWR-SENSOR-PAK

Available Tower System Modules

www.freescale.com/tower

Processor Modules ($39-$119)

8-bit

TWR-S08LL64
TWR-S08LH64
TWR-S08JE128
TWR-S08MM128
TWR-S08GW64
TWR-S08UNIV

16-bit

TWR-S12GN32
TWR-S12G128

DSC

TWR-56F8257

32-bit - ColdFire

TWR-MCF51JE
TWR-MCF51CN
TWR-MCF51MM
TWR-MCF51QM
TWR-MCF5225X
TWR-MCF5441X

32-bit – Power Arch

TWR-MPC5125

32-bit Kinetis

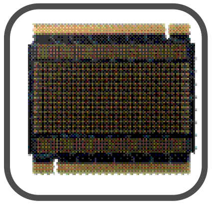
TWR-K60D100M
TWR-K70F120M
TWR-K40X256
TWR-K60N512-IAR
TWR-K60N512-KEIL
TWR-K53N512
KWIKSTIK-K40

Peripheral Modules ($15 – $149)

Serial

TWR-SER
TWR-SER2

Prototyping



TWR-PROTO

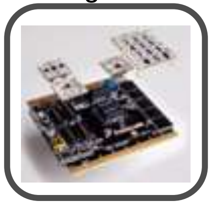
Wi-Fi

TWR-WIFI-RS2101
TWR-WIFI-G1011MI
TWR-WIFI-AR4100

Memory

TWR-MEM

Sensors & Plug-Ins



TWR-SENSOR-PAK
TWR-SENSOR-PAK-AUTO
TWRPI-MMA6900
TWRPI-MPL115A

Displays

TWR-LCD

Medical

MED-EKG

Analog

TWR-ADCDAC-LTC

Audio

TWR-AUDIO-SGTL

Mesh Networking

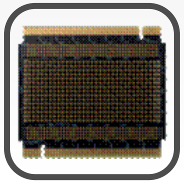
TWR-RF-SNAP

MFi

TWR-DOCK

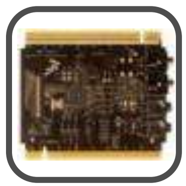
freescale™

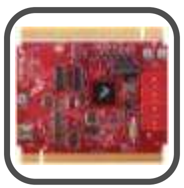
Example Tower System Configurations

 TWR-S08MM128 +  TWR-PROTO +  TWR-SER +  TWR-ELEV = **Medical Prototyping Solution**

 TWR-56F8257 +  TWR-MC-LV3PH +  TWR-SER +  TWR-ELEV = **Motor Control Solution**

 TWR-MCF5225X +  TWR-SENSOR-PAK +  TWR-SER +  TWR-ELEV = **Sensors Solution**

 TWR-K40X256 +  TWR-LCD +  TWR-AUDIO +  TWR-ELEV = **Multimedia Solution**

 TWRK60N512 +  TWR-WIFI +  TWR-MEM +  TWR-ELEV = **Wi-Fi Solution**

Kinetis: Freescale Enablement Bundle

Freescale Tower System

Kinetis MCU modules from $69

- Modular, expandable, open-source h/ware development platform for 8/16/32-bit MCUs/MPUs
- Rapid evaluation and prototyping with maximum h/ware reuse
- Supported by a growing range of peripheral plug-in boards (Wi-Fi, Sensing, Graphics LCD, Audio,...)
- www.freescale.com/tower

Open source, reusable hardware platform

Freescale CodeWarrior IDE

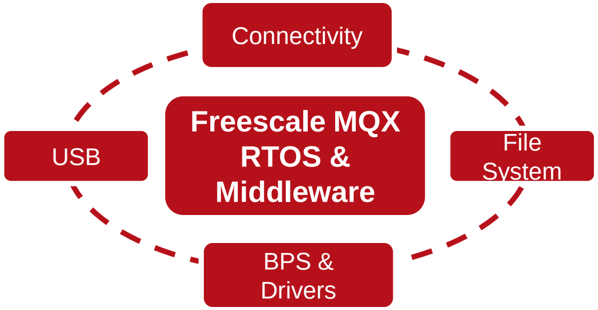
Free of charge up to 128KB

- Eclipse environment
- Includes **Processor Expert code generation wizard**
- Creates MQX-aware drivers
- Build, debug and flash tools
- Software analysis
- Kernel-aware debug
- Special Edition $0 up to 128KB
- www.freescale.com/codewarrior

Powerful IDE with code generation wizard for $0!

Freescale MQX RTOS

Free of charge ($95K est. value)



- Full-featured, scalable, proven RTOS with TCP/IP, USB, Graphics, Security and File Systems plug-ins
- Makes application code more stable, more maintainable and easier to upgrade – reduces time-to-market!
- Compatible with CodeWarrior, IAR, Keil & Green Hills IDEs
- www.freescale.com/mqx

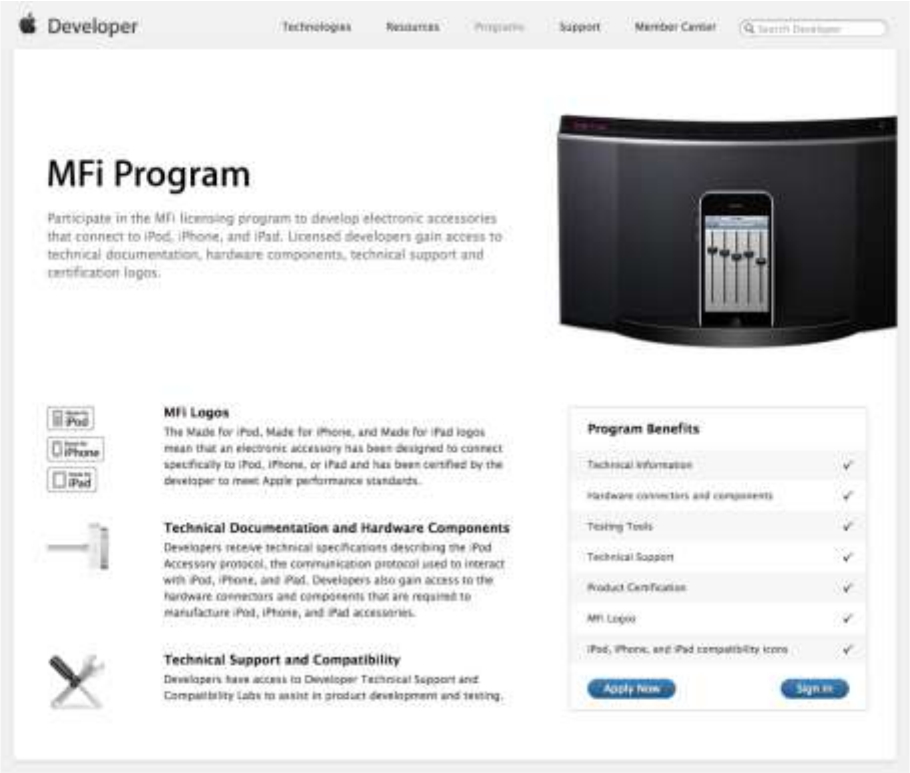
Bundled RTOS for $0!

One Stop Shop for Silicon, IDE & RTOS

Developing Accessories for iPod, iPhone and iPad

Made for iPod | Made for iPhone | Made for iPad

To develop Accessories or App-based accessories for iPod®, iPhone® or iPad® devices, you will need an MFi (Made For iPod) license from Apple

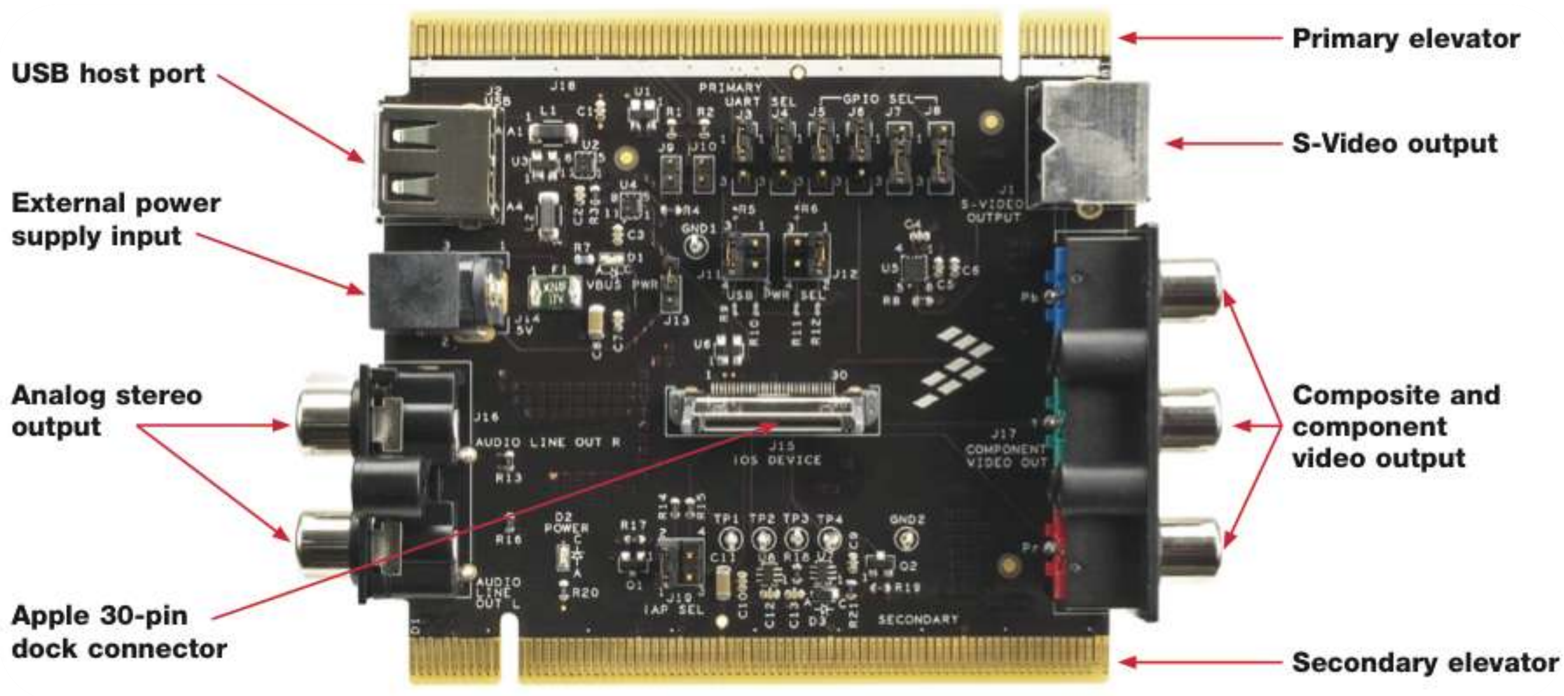


URL: http://developer.apple.com/programs/mfi/

freescale™

TWR-DOCK Module, Available only to MFi Licensees

- Available now!
- May only be purchased through MFi-Portal ($139)



Freescale's MFi Solution – TWR-DOCK

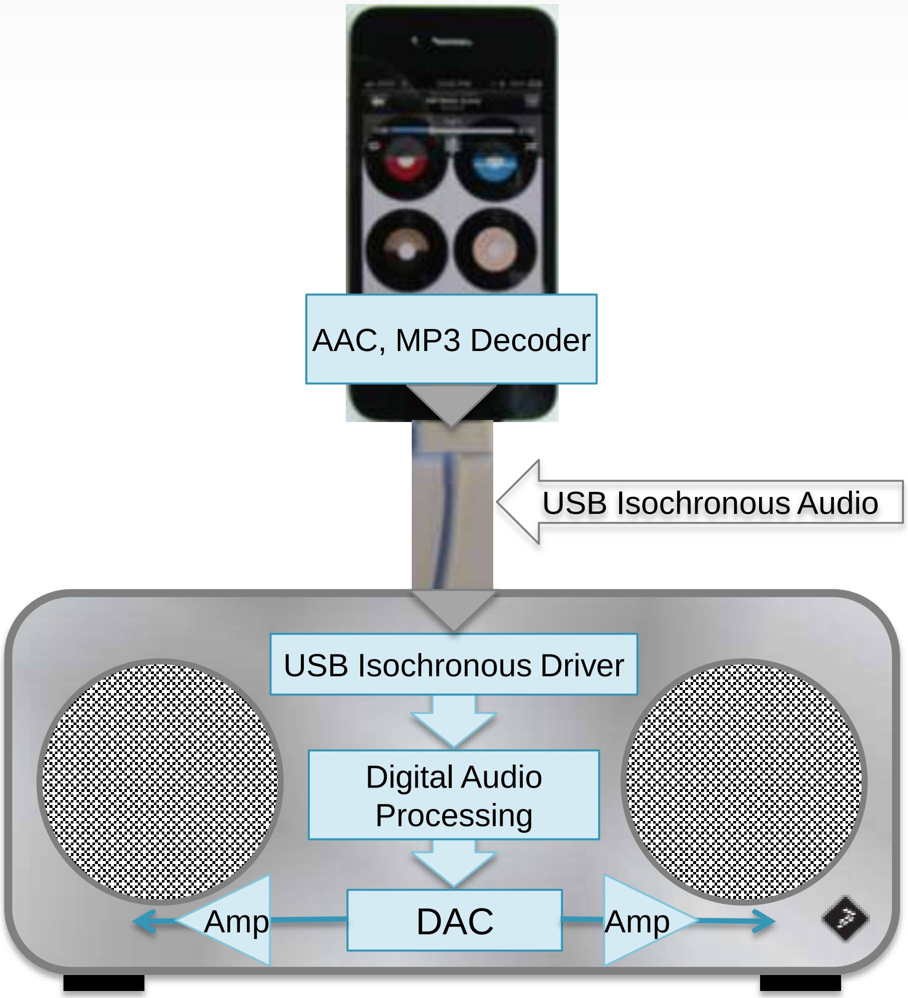
Freescale's MFi solutions are based on the TWR-DOCK peripheral module

- TWR-DOCK supports development and rapid prototyping of electronic accessories for iPod, iPhone and iPad devices.
  - Access to the 30-pin connection
  - Analog audio and video signals with standard RCA and S-Video connectors
  - Digital audio streaming in both directions over USB
  - Control and communication with various devices
- Includes free interface software
- TWR-DOCK concentrates all MFi controlled items on one Tower module, without including any processors or other Freescale products
- TWR-DOCK may be used with a wide range of Tower System MCU/MPU, peripheral, sensor and communication modules
- Kinetis-based demos are available
- *Vybrid support planned for 2012*

freescale™

Fully Digital Speaker Dock

- Best potential audio quality
  - Depends only on quality of source material and speaker dock implementation
- USB Isochronous audio offers two options:
  1. Synchronous – Source and client must have perfectly synchronized clocks
     - Synchronization latency
     - Possible instability
  2. Asynchronous – Source adjust streaming rate according to client control signals
     - Very stable with low latency

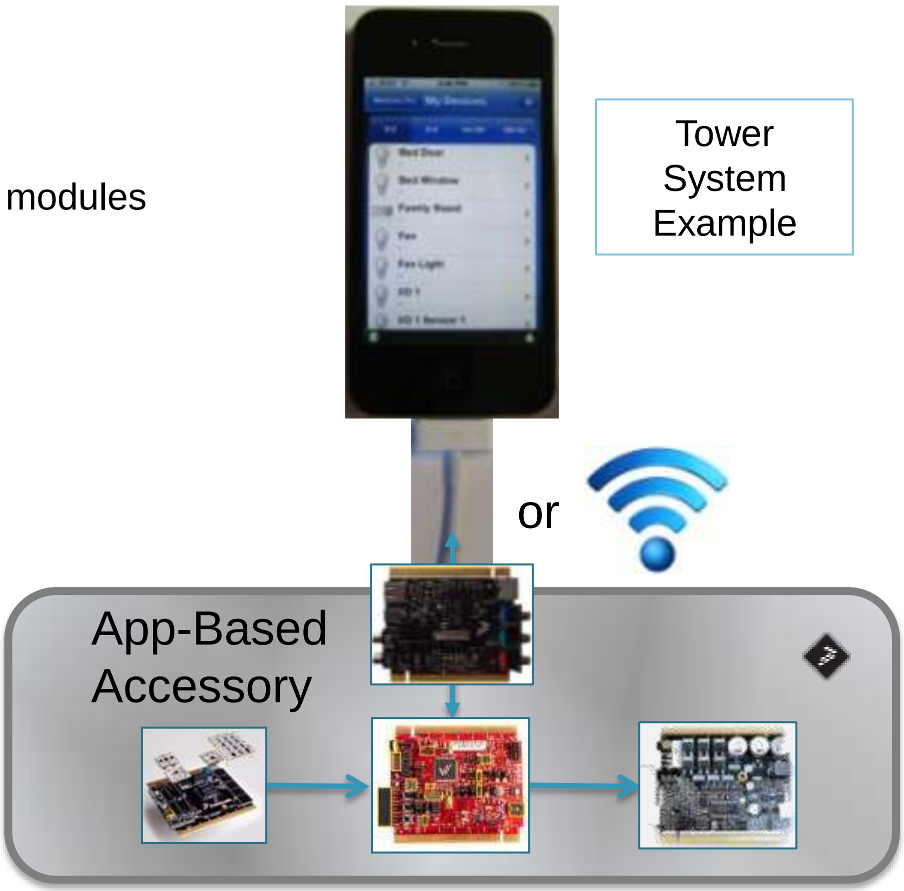


App-Based Speaker Dock

- Uses smartphone based App to control the speaker dock
  - Saves on separate GUI on speaker dock
  - Limited to smartphone or devices that support App-Based Accessories (such as iOS devices)
  - Possible with multiple connection options

App-Based Accessory Example

- The Tower System with TWR-DOCK provides a highly flexible development and rapid prototyping platform for a very wide range of Accessories and App-Based Accessories
- Supports a wide range of:
  - Microcontroller and embedded microprocessor modules
  - Peripheral modules
  - Communication modules
  - Audio modules
  - Display modules
  - Key, touch pad, and keyboard modules
  - Sensor modules
  - Analog modules



Questions

Thank you

Don’t forget to fill out the evaluation form.

Facebook.com/Freescale
Tag yourself in photos
and upload your own!

Tweeting?
Please use hashtag
#FTF2012

Session materials will be posted @ www.freescale.com/FTF
Look for announcements in the FTF Group on LinkedIn or follow Freescale on Twitter

NXP

freescale™