Quality Audio Software Pipeline

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Agenda

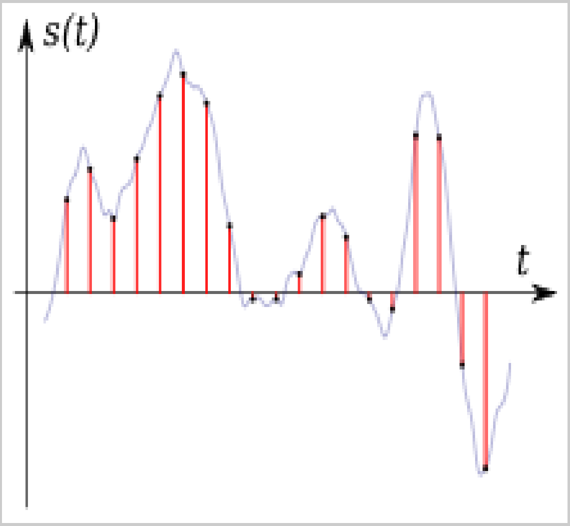
- Scope is limited to Audio quality considerations in software audio pipeline
- Journey of Audio frame in a Multimedia system
- List of issues observed during audio processing
- Configuration and Usage
- Best practices

What is an audio frame ?

- An audio frame is one audio sample
- For a stereo data one left channel and one right channel sample forms one frame
- Example – One 16 bit left data + One 16 bit right sample = One 32 bit audio frame
- For multi-channel one sample from channels combined forms one audio frame

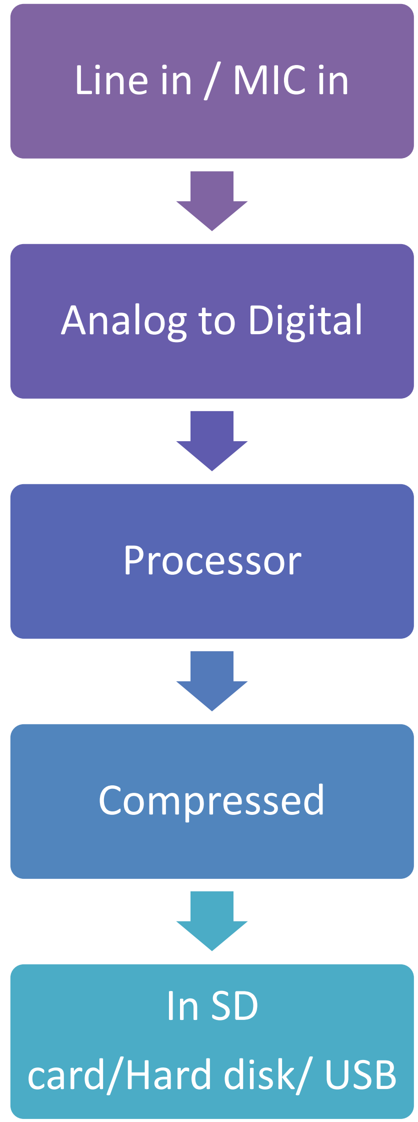
What is sampling rate?

- Rate at which audio data is sampled
- Number of samples per unit of time
- Audio data is generally sampled at 44.1KHz or 48KHz
- Voice data are sampled at 8KHz or 16KHz



Analog signal (light blue) with a sampled signal (red) with a fixed spacing or *sampling rate*.

Audio Capture Path



Audio Playback Path

From SD card/Hard disk/USB

↓

Un-compressed

↓

Processor

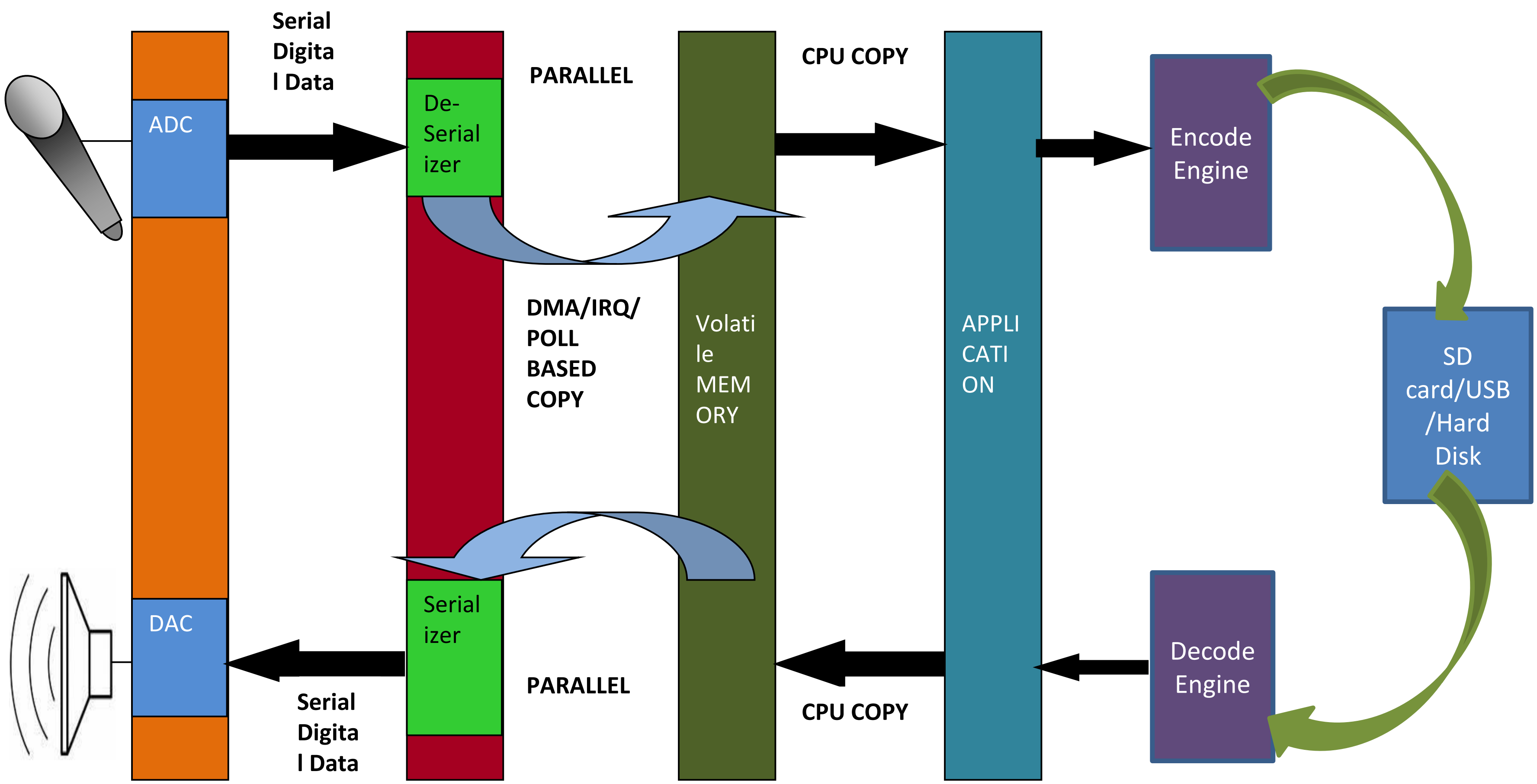
↓

Digital to analog

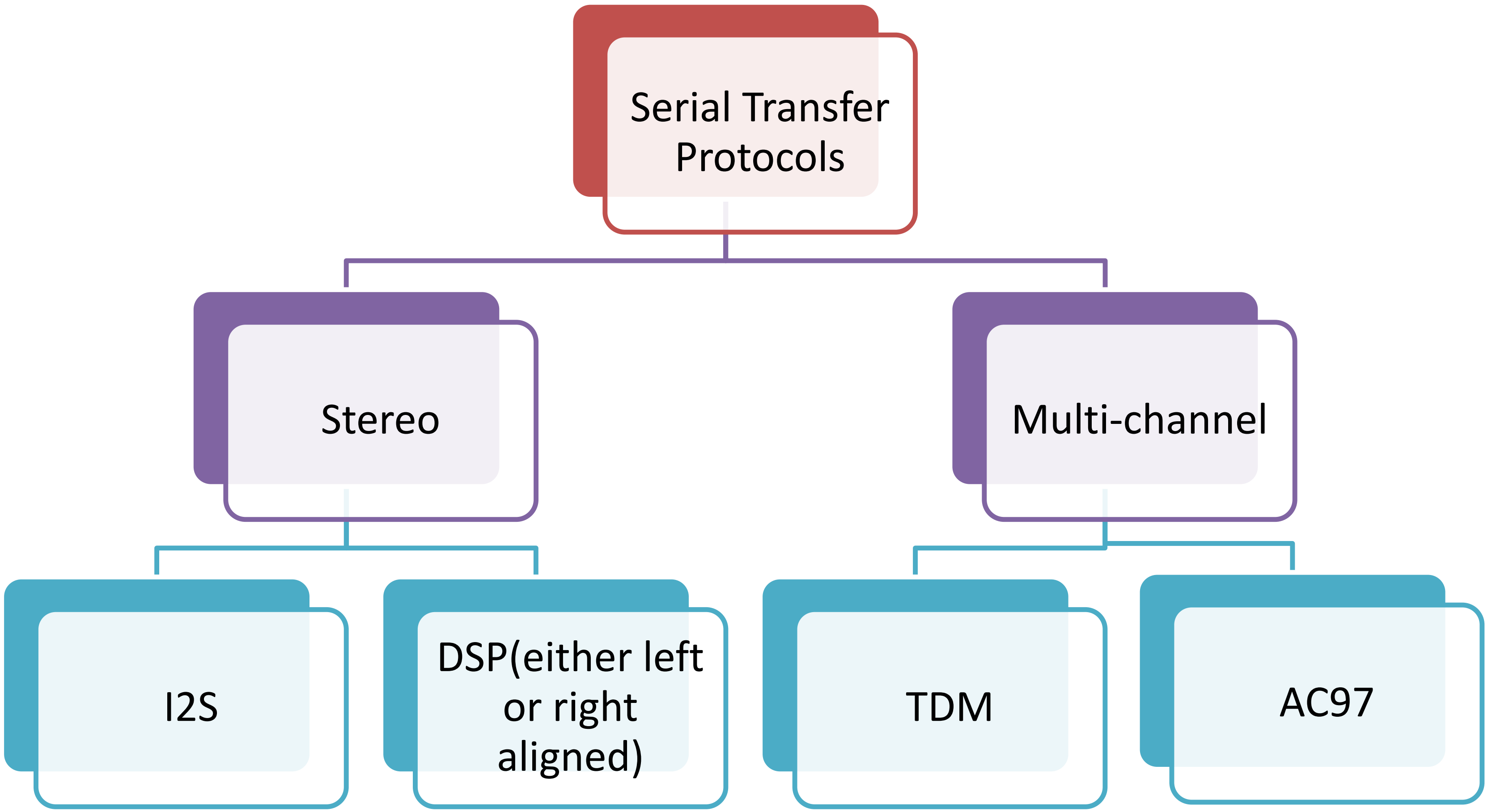
↓

Speaker/Line-out

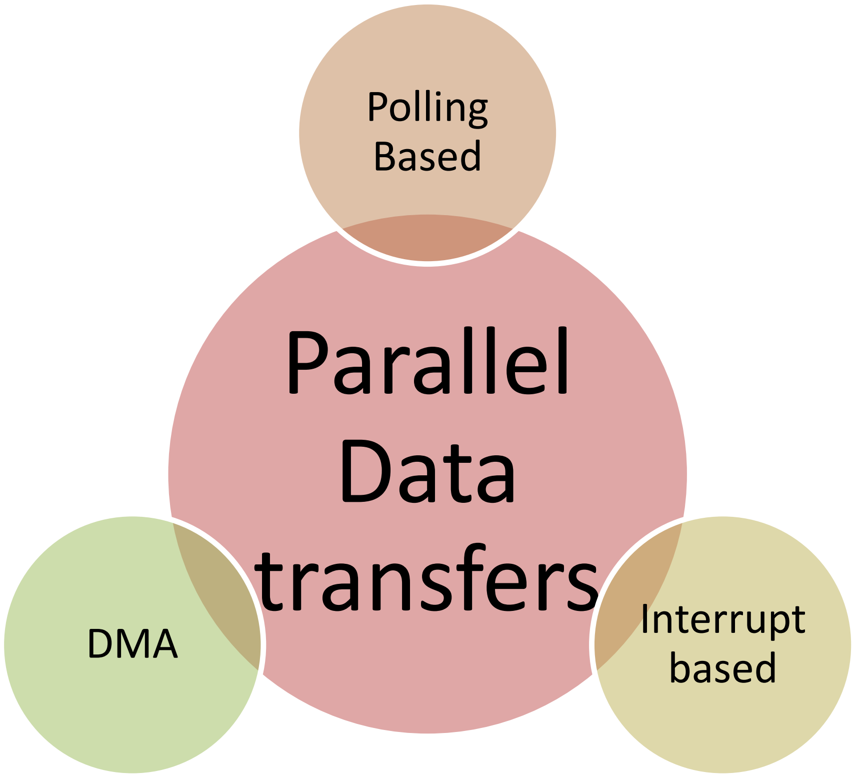
Journey of an Audio frame in a multi-media system



Serial Data Protocols



Parallel Audio Transfer



- DMA transfers are preferred other modes are CPU intensive and may impact real time performance.

Audio software pipeline - perspective

- In any multi-media system audio quality is one of the very first things that help the product stand out.
- Considerable precautions are taken in designing the audio hardware so that it does not introduce any noise or anomaly in the audio. So it becomes utmost important for the software developer to take care while he processes the audio.

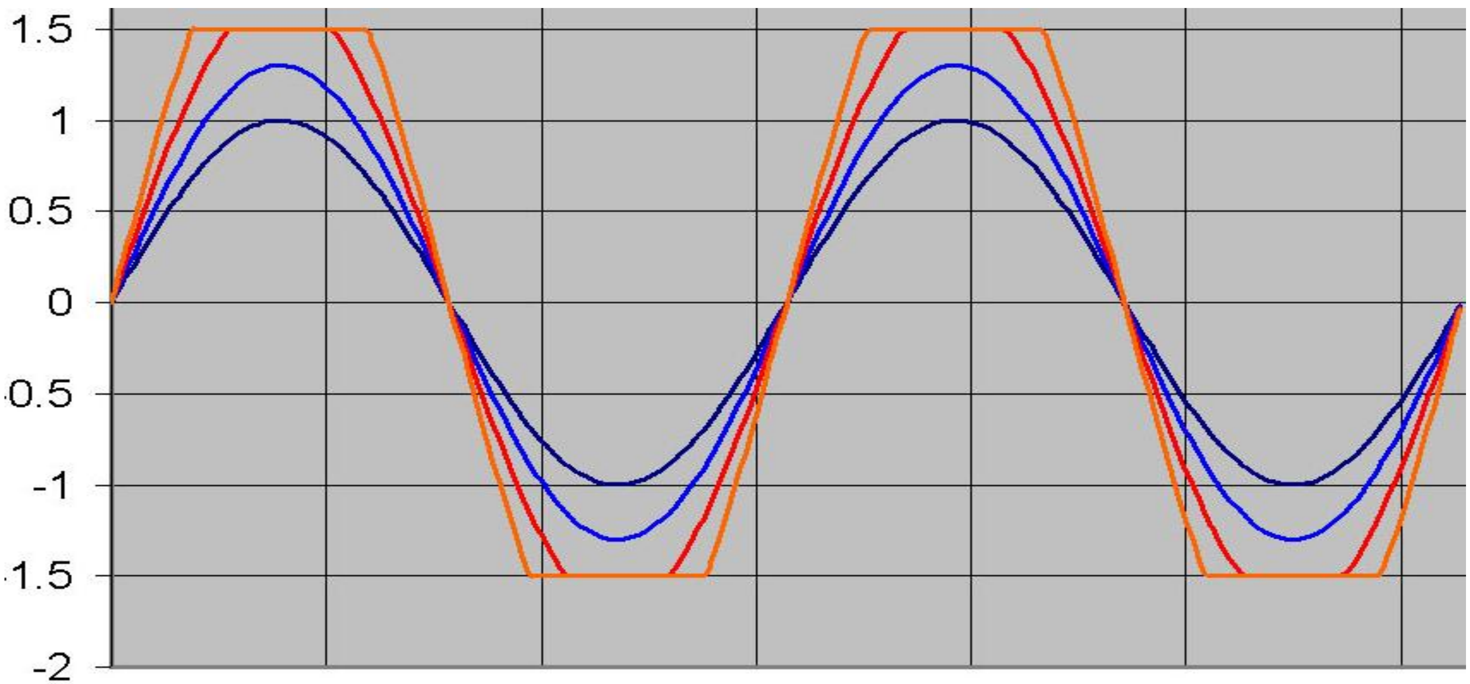
Common issues in audio pipeline

The following list brings out some of the very common audio quality concerns

- Audio Overrun
  - More of concern in the audio capture path.
  - Occurs if the hardware audio data capture rate is faster than the rate at which it is consumed by the software.
- Audio Under-run
  - More of concern in the audio play-out path.
  - Occurs if the audio data play-out rate is faster than the rate at which data is fed to the hardware by the software.

Common issues in audio pipeline contd.

- Clipping
  - A form of wave form distortion
  - Occurs when the ADC and DAC gains are set to a very high value



Issues observed in audio path contd.

- Incorrect sampling rate configuration
  - Sampling rate configuration of the system does not match the desired audio data sampling rate.
  - Faster or slower audio depending on whether the sample-rate configured is more or less than the required.
- Buffer width mismatch
  - Significant mismatch in the driver and application buffer sizes increases the probability of audio under-runs and over-runs.
- Channel swap
  - One channel data is swapped with the other channel data

Issues observed in audio path contd.

- Noise, glitches, breaks
  - Causes:
    - Software processing rate does not match with the hardware data rate
    - Improper DMA configuration
    - Noise included by the audio hardware
  - Impacts:
    - Audio quality degradation

Considerations for quality Audio Software Pipeline

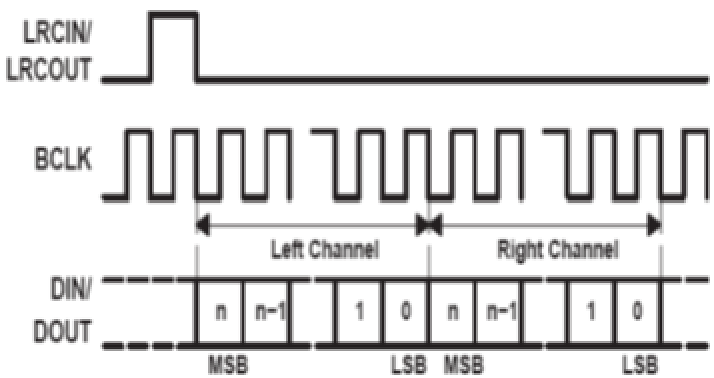
- Serial Interface Modes: DSP Mode and I2S mode
- Shift register level overruns
- TDM and low latency configurations
- DMA configurations
- Sample Rate and Buffer Sizes
- Software resampling

Serial Interface Modes: DSP Mode and I2S mode

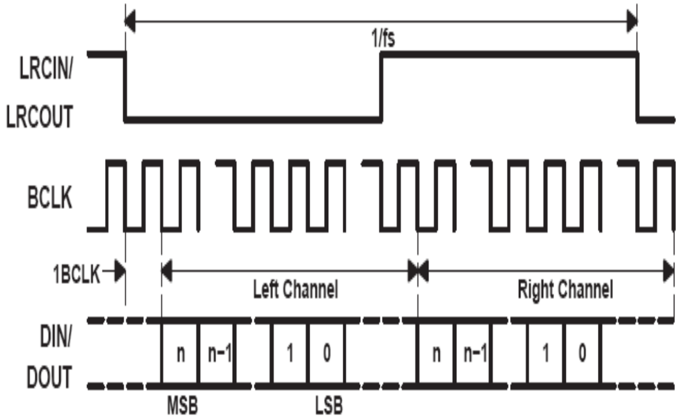
- DSP mode and I2S mode are the two standard interfaces between Audio Codec and the Processor. These modes are used to transfer the digitized Audio data from the decoder to the processor while recording and vice versa while playback.

DSP Mode and I2S mode : Waveforms

- DSP Mode of transfer



- I2S Mode of transfer



DSP Mode and I2S mode : notable differences

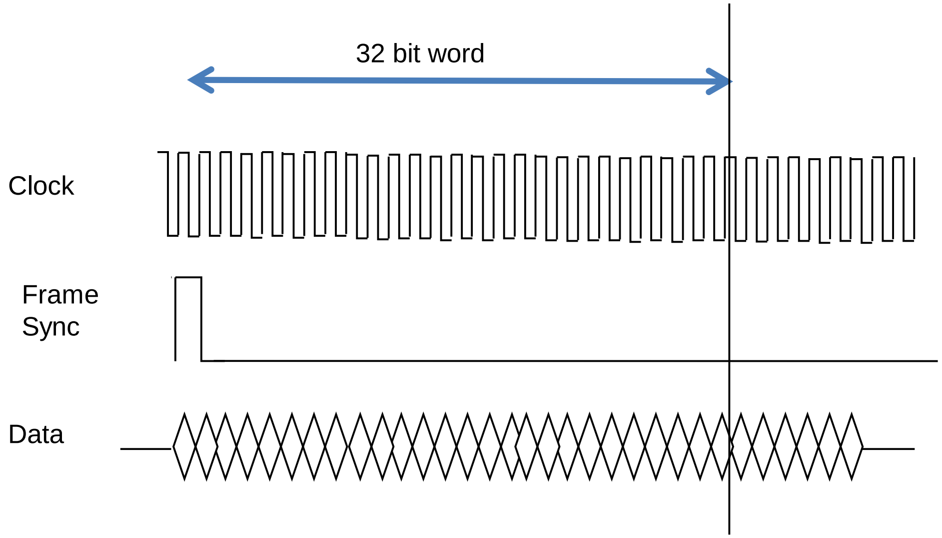
- DSP Mode
  - One single sync pulse to determine the start for frame
  - Bit clock can be much higher than required. The data would be sent as either left justified or right justified
  - Higher bit clock may required faster DMA event generation if the serial port is not configured rightly
- I2S Mode
  - Sync has 50% duty cycle. Its high during left channel data transfer and low during right channel data transfer
  - Bit clock can be lower and may be equal to the no of bits to be transferred during the sync period.
  - Due to slower clock we may get away with un-optimizations on serial port transfer.

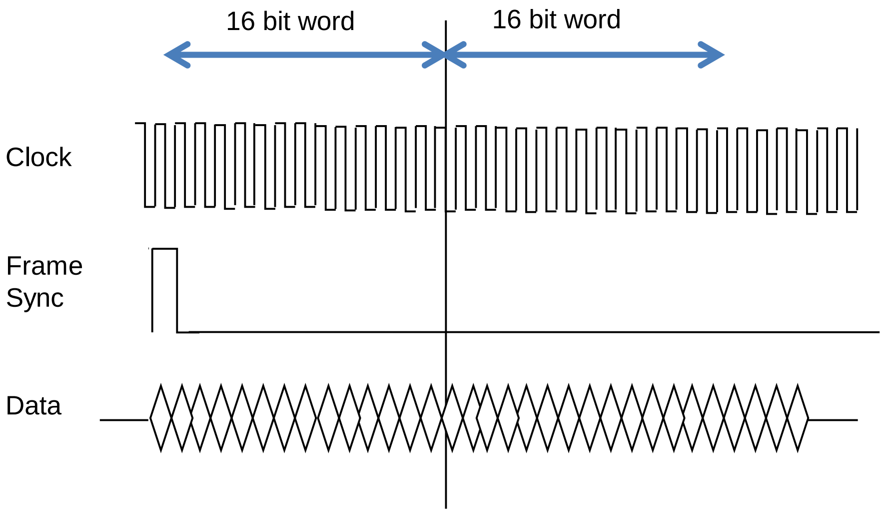
DSP Mode: DMA optimizations

- DMA optimizations are necessary for DSP mode of transfers else the audio could look jittery owing to higher bit clock and the end user may hear click sounds once in a while.
- Transfer configuration that works perfectly fine for I2S mode of transfer should be reviewed if the mode of transfer is DSP.
- Following example illustrates one configuration optimization that is necessary.

DSP Mode: Configuration optimization

- Configuration 1
  - 32 bit Stereo Audio
  - No of frames: 1
  - Word Length: 32
- Configuration 2
  - 32 bit Stereo Audio
  - No of frames: 2
  - Word Length: 16





DSP Mode: Best practices

Minimize the number of frames	• Ideal case would be to have one single frame
One frame per transfer	• This would also avoid DMA transfers within a sample
Trigger at frame boundary	• This gives ample time to schedule DMA transfers
Less loaded DMA	• higher bit clocks (up to 256 fs) result in higher rate of DMA transfers rates which causes higher load on DMA and may result in sample misses.

Ittiam

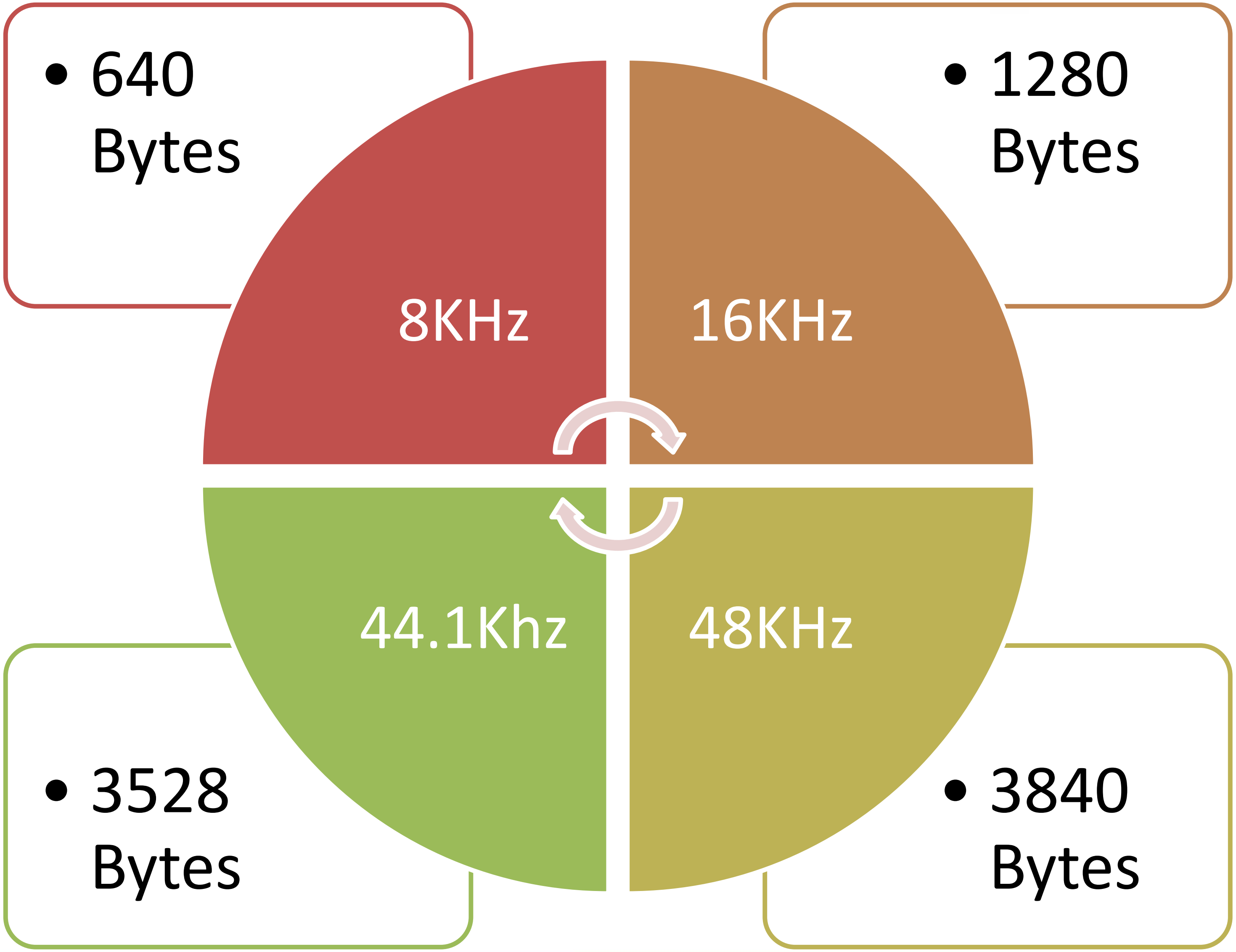
Sample rate - configuration

- Incorrect configuration of sample rate would lead to anomalies at capture and playback.
- Accuracy and Synchronization of clocks in streaming system is utmost necessary to achieve good quality audio. Inaccuracy in Sample rates at capture and playback could lead to
  - Brakes in the audio
  - Overruns and under runs at buffer
  - Loss of Audio Video synchronization

Buffer size - configuration

- Buffer size configuration in the driver as well as application have to be decided based on
  - The application that the system is catering to
  - Sampling rate of the system
  - OS scheduling time
- If one choses higher sampling rates and lower buffers sizes then definitely the system is prone to overruns and under-runs.

Buffer size – Selection (20ms Buffer)



TDM and low latency system - Requirements

Very high data-rate	• Because of multi-channel the data rate is very high
Very low latency	• This implies that the system buffer sizes have to be small enough.
No capture over-run	• No capture sample miss • Capture software should be fast enough to match with hardware data rate
No playback under-runs	• No playback sample drop • Playback software rate should be fast enough to match with the hardware rate

TDM and low latency system – Configurations

- Buffer Management
  - Multi level buffering
  - Buffer management strategy needs to handle the condition of capture over-run and playout under-runs.
- DMA configurations
  - Proper configuration of DMA for multiplexing and e-multiplexing of TDM data in the playout and capture path respectively
- OS scheduling
  - OS threads catering to capture and playback should have higher priority so that they get scheduled at the right time intervals

Channel Swap – Introduction

- The channel swap could have the following implications:
  - Left channel data getting swapped with Right channel data in a stereo system
  - channel positions being swapped in a multichannel audio configuration
- Channel swap could occur at the start of recording or playback or it could occur during recording or playback.
- The issue once started gets propagated throughout the entire span of operation.

Channel Swap – Causes (Case I - Start of operation)

- Processor audio serial port is enabled before the serial port DMA initializations
- DMA initializations take a few milliseconds to complete. Audio serial port and DMA are configured for 1 channel data transfer
- Data from the first channel would get over written if DMA transfers do not start in one channel time period
- In case of stereo data left data will get over-written by the right
- In case of multi-channel the first channel may get swapped with the channel number that arrives at a time equal to the time taken for the DMA to start

Channel Swap – Causes (Case II - During operation)

DMA engine is heavily loaded or the audio DMA transfers do not have necessary priority to get scheduled at right intervals

Audio data in the audio serial port shift-register gets missed

The time DMA is scheduled for a data transfer, the audio data in the serial port shift-register might get over-written by the data from the next channel

- This kind of serial port shift register level over-runs are very hard to detect and they need some special techniques like enabling hardware interrupts from the serial port to CPU for over-run conditions.

Software Re-sampling: Considerations

- The audio hardware is hard wired to a fixed sampling rate
- Multiple applications requiring to receive and send data at rates other than the hardware rate
- Implementation of re-sampling in software then becomes necessary so that all the different applications can access the same hardware resource.

Software Re-sampling – system schenario

Capture Rate -> 8KHz

Playback Rate -> 44.1KHz

Hardware does not support different sampling frequencies for capture and playback over a single serial port

Audio driver allows the first process (capture/playback) to configure the hardware sampling frequency. The other process respects the same and does up/down sampling to adjust to the user specified rate

Here we need to configure the hardware to the higher sampling frequency and configure the other for software down-sampling.

Software Re-sampling – Best Practices

- Hardware to be set to the highest of the two samples rates when capture and playback are at 2 different sample rates
- In audio frame works unnecessary software resampling to be avoided.
- Too many software re-samplers result in
  - Increase in system load
  - In turn results in capture and playout thread scheduling issues
  - In turn results in capture overrun and playout underrun and sample drops
  - Hence degrades the over-all system performance

DMA Configuration : Best Practices

- The audio data transfer from audio serial port shift register to memory using DMA has to be scheduled at right intervals. Failing to do so will result in samples misses in capture and sample drops in playback.
- To ensure this DMA configurations should take care of the following
  - Highest priority for the audio DMA transfer queue
  - No other high priority transfer scheduled in the same queue.
  - Optimal DMA transfer configurations to
    - Avoid too many interrupts for a single transfer
    - Minimize the time for each transfer

DMA Configurations for TDM – Best Practices

- DMA to be used for de-multiplexing the captured data and multiplexing the playback data
- CPU based multiplexing and de-multiplexing techniques hog the system.
- Optimal DMA configurations to be done
- Best option is to use a 3 dimensional DMA transfer technique to optimize the number of transfers and the number of interrupts.

Software Audio Pipeline: Key take away

- Glimpse at audio pipeline
- Typical issues observed during recording and playback
- Considerations while designing audio pipeline. Sneak into the real time issues
- Best practices to avoid typical audio issues
- Role of DMA configurations

Thank You

Q&A