Description of ResE Audio Applications and Requirements

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Outline

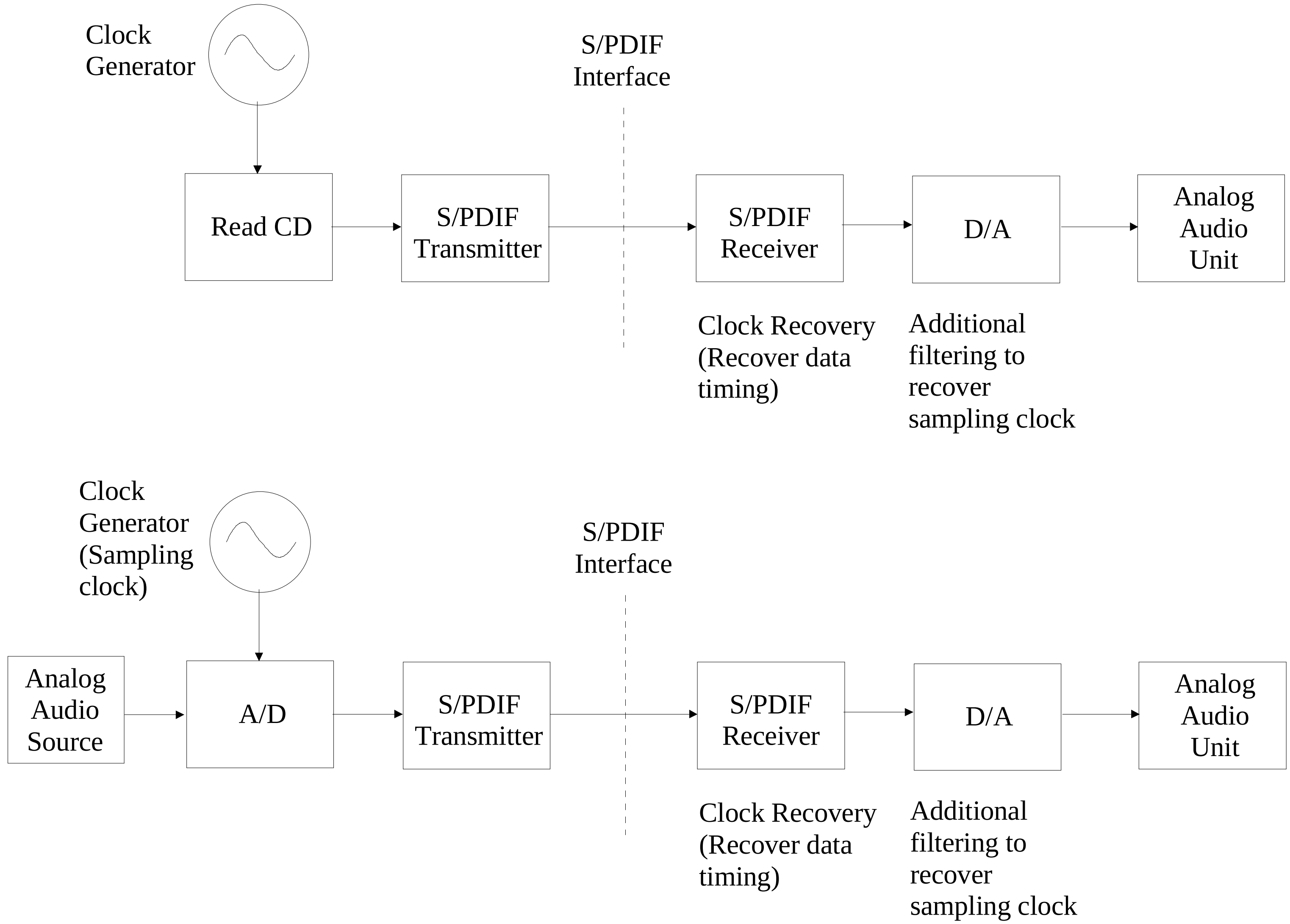
- Introduction
- Digital audio – background
- Digital audio interface standards
- Properties of digital audio signals at IEC 60958 Interface
- Backup
  - More detailed version of presentation
  - References

Introduction

- This is the second of three related VG presentations
  1) Description of ResE Video Applications and Requirements
  2) Description of ResE Audio Applications and Requirements
  3) Jitter and Wander Requirements for ResE Applications

- The backup slides contain a more detailed version of the presentation
- For convenience, each presentation contains the complete (i.e., combined) reference list for all three presentations, at the end of the backup slides

Digital Audio – Background



Digital Audio – Background (Cont.)

- Digital audio source
  - Read CD, at rate controlled by source clock generator
  - Read encoded audio from local server, at a rate controlled by source clock generator
  - Sample analog audio source, at rate controlled by clock generator (sampling clock)
- Transmit digital audio to receiver
  - In consumer applications, this interface is the standardized S/PDIF (Sony/Philips Digital Interface)
  - Professional interfaces are also standardized (interface details described shortly)
- Perform clock recovery at receiver and detect the incoming bits
- Perform further filtering of recovered clock in or prior to D/A converter to produce sampling clock
  - Jitter/wander (especially jitter) requirements for sampling clock are much tighter than those needed for clock/data recovery
  - The additional filtering in or prior to the D/A converter essentially “cleans up” the recovered data clock
  - More detail on digital audio jitter and wander described shortly

Digital Audio Interface Standards

- Two classes of interface, for two applications, are defined
  - Consumer applications
  - Professional applications
- Both interfaces (including jitter specifications) are standardized in IEC 60958
  - Part 1: General (IEC 60958-1 [18])
  - Part 3: Consumer applications (IEC 60958-3 [19])
  - Part 4: Professional applications (IEC 60958-4 [20])
  - The consumer interface is equivalent to the S/PDIF (Sony/Philips Digital Interface)
  - The professional interface is equivalent to the AES3 specification [22]
  - The professional interface is also equivalent, with some differences, to the EBU 3250-E specification
  - Jitter specifications in IEC 60958-4, AES3, and EBU 3250-E are the same
  - Jitter specifications in IEC 60958-3 and IEC 60958-4 (i.e., consumer and professional interfaces) have significant differences
- Specification of nominal rates in AES5 [23]
- Specification of wander/synchronization in AES11 [24]

Properties of Digital Audio Signal at IEC 60958 Interface

- Digital audio signal uses a bi-phase line coding
  - Each data bit occupies 2 UI (unit intervals)
  - Always have a transition at at data bit boundary
  - Additional transition in the middle of a 1 bit
  - No transition in the middle of a zero bit
- Data is carried in frames
  - Each frame is 64 bits, or 128 UI
  - Each frame is composed of 2 subframes of 32 bits (64 UI) each
  - The 2 subframes can be used to carry 2 channels of data
  - Each subframe carries data representing one audio sample
  - Therefore, each channel carries data at a sample rate equal to the frame rate
  - Subframe structure
    - Audio sample word – up to 24 bits (need not use all 24, but pad if fewer used)
    - Preamble, parity bit, and other overhead (see backup slides and references for more details)

Properties of Digital Audio Signal at IEC 60958 Interface

❑Nominal frame rates

- Basic rates defined in [23] are 44.1 kHz (consumer applications) and 48 kHz (professional applications) Corresponds to 5.6448 and 6.144 Mbit/s, respectively
- Also define double, quadruple, half, and quarter rates in [23]
  - Consumer applications – 11.025, 22.05, 88.2, 176.4 kHz
    - Corresponds to 1.4112, 2.8224, 11.2896, and 22.5792 Mbit/s
      - »Bits are the UIs described above, and not the 2-UI bits
  - Professional applications – 12, 24, 96, 192 kHz
    - Corresponds to 1.536, 3.072, 12.288, and 24.576 Mbit/s

Properties of Digital Audio Signal at IEC 60958 Interface

❑ Frequency accuracy requirements

- This is the amount the source (sampling) clock is allowed to deviate from nominal, in the long-run
- Note that the same long-term frequency accuracy requirement applies at all interfaces over sufficiently long time intervals
    - This is because bits are not created or destroyed by the network
        - the long term average rate at which bits cross an interface is the same when the averaging time is sufficiently long
    - Note that frequencies averaged over shorter intervals at various interfaces may have deviations that are larger than the long-term frequency accuracy requirements
        - Deviations over shorter time intervals are specified in the form of jitter and Maximum Time Interval Error (MTIE) requirements (see the third VG presentation referred to in the Introduction)
- The specified long-term frequency accuracy is also the minimum long-term frequency offset the receiver must tolerate
    - A ResE network inserted between the transmitter and receiver must also tolerate this frequency offset

Properties of Digital Audio Signal at IEC 60958 Interface

❑Frequency accuracy requirements – Consumer applications

- IEC 60958-3 defines 3 levels of accuracy for the sampling clock
- Level I (high accuracy mode): $\pm 50 \times 10^{-6}$ ($\pm$ 50 ppm)
- Level II (normal accuracy mode): $\pm 1000 \times 10^{-6}$ ($\pm$ 1000 ppm)
- Level III (variable pitch shifted clock mode): The standard indicates that signal in this mode can be received by specially designed receivers.
  - A note indicates that the frequency range is under consideration, but that a range of 12.5% (125000 ppm) is envisaged
- IEC 60958-3 indicates that receivers should be able to lock to signals with Level II accuracy
  - I.e., $\pm$ 1000 ppm pull-in range
  - Indicates that if a receiver's pull-in range is less, it should exceed the Level I tolerance ($\pm$ 50 ppm) and shall be specified as a Level I receiver

Properties of Digital Audio Signal at IEC 60958 Interface

❑Frequency accuracy requirements – professional applications

- AES11 specifies 2 levels of frequency tolerance
  - Grade 1
    - frequency tolerance of ± 1 ppm
    - Pull-in range of ± 2 ppm
  - Grade 2
    - frequency tolerance of ± 10 ppm (note that the ± 10 ppm tolerance is indicated in AES5 also)
    - Pull-in range of ± 50 ppm
  - Equipment designed to provide a Grade 1 signal shall only be required to lock to other Grade 1 signals
- AES11 defines the Digital Audio Reference Signal (DARS) for studio applications
  - May be used to time all the equipment in a studio
    - May also time equipment by incoming audio or video signal
  - DARS is classified as Grade 1 or Grade 2
  - DARS may be referenced to GPS

Properties of Digital Audio Signal at IEC 60958 Interface

❑Frequency accuracy requirements – professional applications (Cont.)

- AES11 does not discuss any distribution of timing references between studios (i.e., it does not discuss a synchronization network)
  - AES11 indicates that when an incoming signal to a studio differs in phase and/or frequency from the DARS of that studio
    - Frame alignment is necessary if only the phases differ
    - Sample rate conversion is necessary if the frequencies differ
      - »Presumably, this means interpolation in going to higher frequencies and discarding a small amount of information in going to lower frequencies

❑Maximum phase offset requirements (peak-to-peak wander) – professional applications

- Maximum phase offset between input and output of digital audio equipment (wander generation)
  - ± 5% of a frame period (± 6.4 UI)
- Input wander tolerance of digital audio equipment
  - ± 25% of a frame period (± 32 UI)
- Wander accumulation within a studio (e.g., traversing a chain of digital audio devices [26], [27])
  - ± 25% of a frame period (± 32 UI)
- Between studios may have larger wander and/or frequency differences; in latter case sample rate conversion is necessary

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter specifications

- As indicated earlier, jitter requirements for sampling clock are much tighter than jitter requirements at digital interface (receiver input)
  - Sampling clock jitter requirement is driven by level of jitter that causes audible effects
    - –Depending on the particular audio source and jitter frequency, this can range from less than 1 ns rms to more than 100 ns rms [29], [30]
    - –Effect of jitter tends to be greater at higher jitter frequencies and higher audio source frequencies
  - Digital interface jitter requirement is driven by need to perform clock and data recovery with acceptable bit error ratio (BER)
    - –Assumed that receiver and DAC can cope with any jitter within the interface requirement
    - –Receiver and DAC will contain the necessary filtering to perform clock and data recovery, and to bring the sampling clock jitter to within limits
      - »Some implementations may use two-stage filtering process: wide band clock recovery circuit, followed by narrow band jitter cleanup filter

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter specifications (Cont.)

- Interface jitter (referred to as *Network Limit*) specification
  - Related to jitter tolerance; essentially, receiving equipment must tolerate the jitter that is allowed to accumulate in the network
    - Here, the network is whatever equipment the digital audio traverses in getting from the source to the receiver
    - Network can include both digital audio equipment and intermediate transport (e.g., ResE) network(s)
      - »Reference model – worst-case network connection expected in practice
    - Any jitter accumulation over the reference model must be within digital interface jitter requirement
    - Assumed the audio remains in the digital domain as it traverses the reference model
      - »A/D and D/A occur at endpoints
  - Often specify jitter tolerance to sinusoidal input jitter
    - Sinusoidal jitter tolerance mask expresses minimum peak-to-peak sinusoidal input jitter that must be tolerated as a function of frequency
- Jitter generation specification
  - Referred to in IEC 60958 and AES-3 as *intrinsic jitter*
  - Amount of jitter a piece of digital equipment is allowed to produce when the input digital signal is jitter-free
- Jitter transfer
  - Maximum allowable output jitter, excluding generated jitter, for a specified level input jitter
    - Often specified in the form of a frequency response to sinusoidal input jitter

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter generation specifications

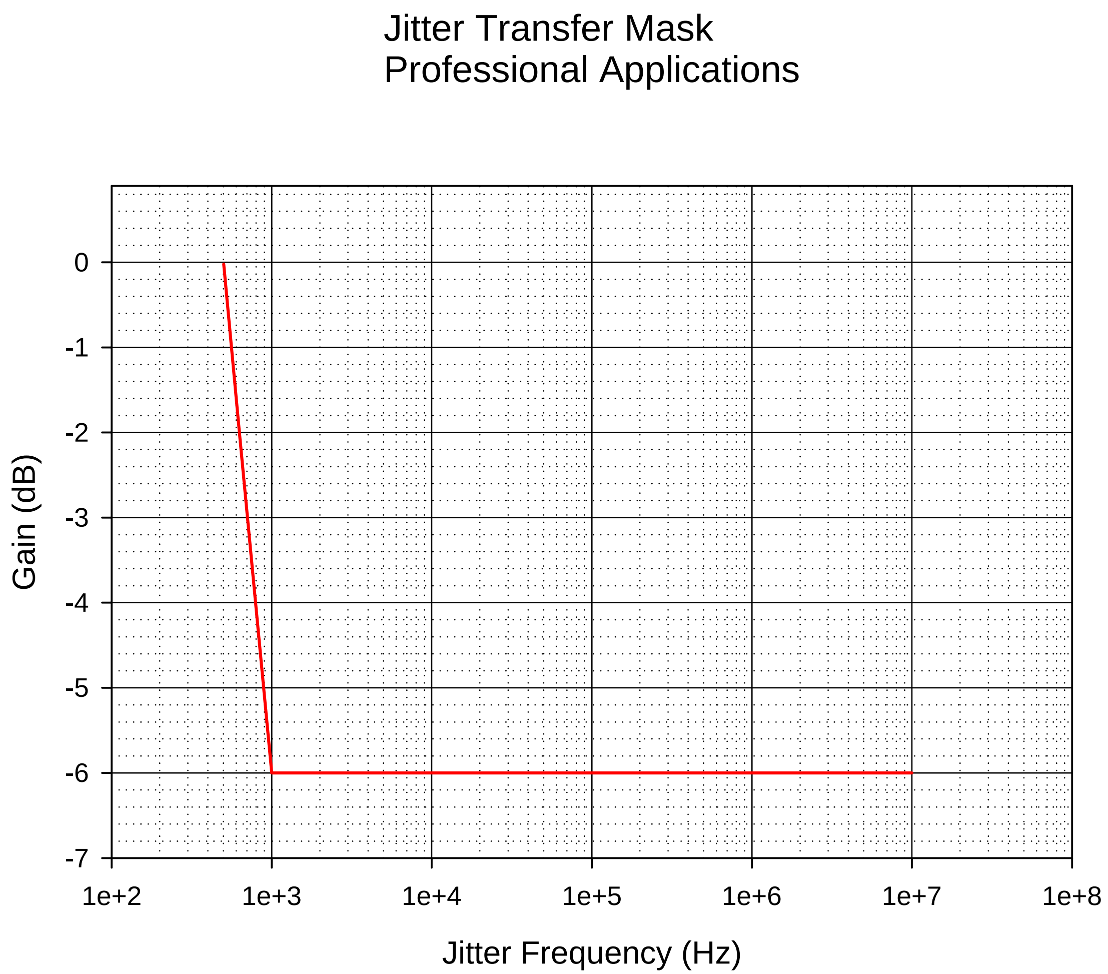
- Consumer applications: peak jitter ≤ 0.05 UI
  - Peak-to-peak jitter ≤ 0.1 UI
- Professional applications: peak jitter ≤ 0.025 UI
  - Peak-to-peak jitter ≤ 0.05 UI
- Jitter measurement filter
  - Same for both consumer and professional applications
  - 700 Hz (3 dB point) first-order (minimum phase) high-pass filter
  - Pass-band gain of unity
  - Roll-off to 70 Hz

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter transfer specifications

- Consumer applications
  - Maximum gain peaking: 3 dB
  - No specification for jitter attenuation
- Professional applications
  - Maximum gain peaking: 2 dB
  - Jitter attenuation is not required, but if it is provided, it should be within the mask on the following slide
    - –No additional specification below 500 Hz (beyond the 2 dB gain peaking limit)
    - –20 dB/decade roll-off between 500 Hz and 1 kHz, from 0 dB to –6 dB
    - –Constant attenuation of –6 dB from 1 kHz to 10 MHz
    - –See mask on following slide

Properties of Digital Audio Signal at IEC 60958 Interface



Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter tolerance/network limit specifications

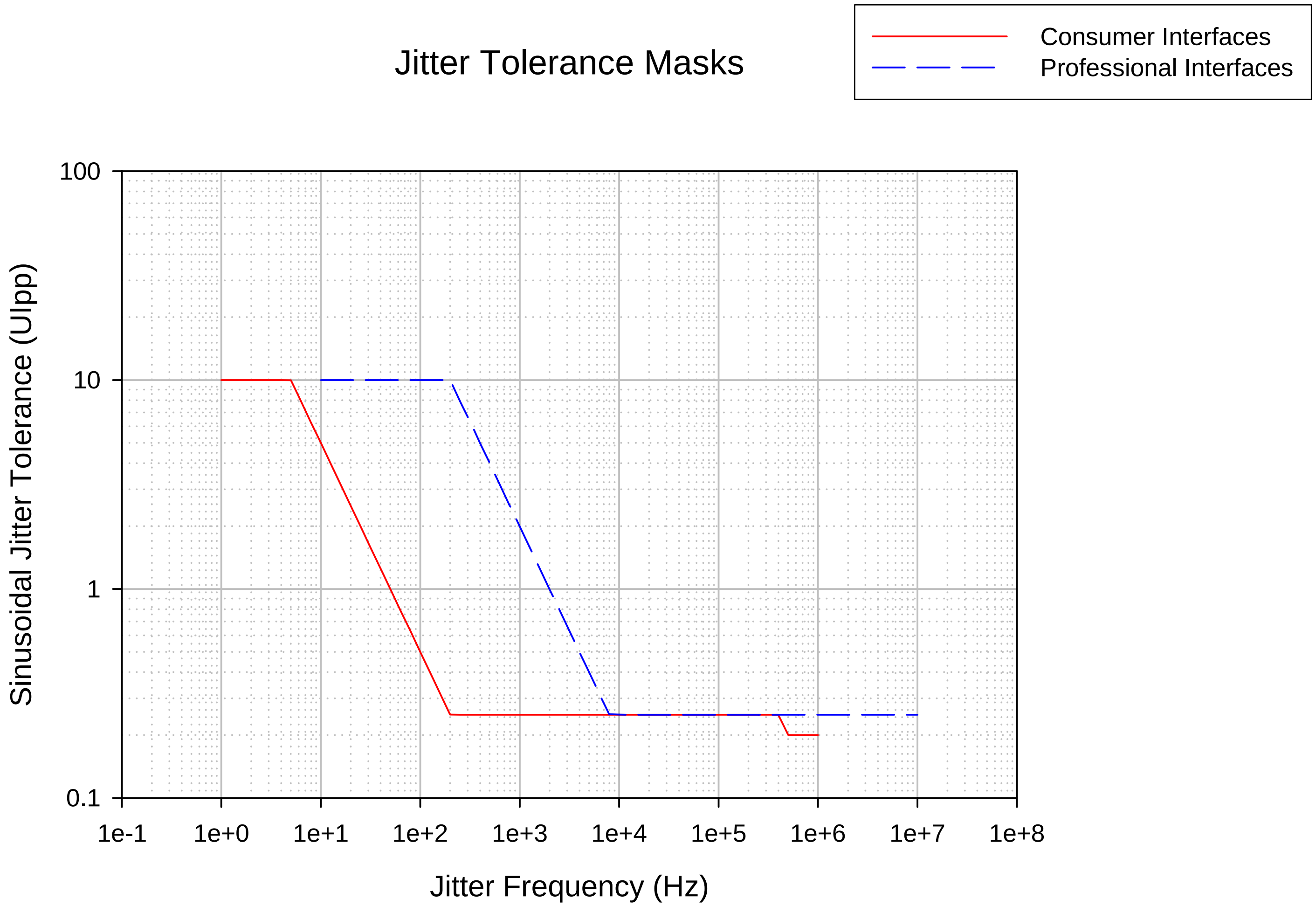
- Represents the amount of jitter the DAC must cope with and still produce an acceptable sampling clock
  - If accumulated jitter exceeds the mask, sampling clock may have excessive jitter, resulting in audible effects
- Professional equipment is required to tolerate higher level of jitter than consumer equipment
  - Appears that this tends to allow consumer equipment to use a single filter for clock recovery and jitter cleanup
    - narrow band jitter cleanup filter would not tolerate higher frequency jitter levels of professional interface
    - Professional equipment would tend to use 2 filters – wide-band clock recovery followed by narrower-band jitter reduction

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter tolerance/network limit specifications (Cont.)

- Consumer applications sinusoidal jitter tolerance mask (all jitter values are peak-to-peak)
  - 10 UIpp between 1 Hz and 5 Hz
  - 20 dB/decade roll-off between 5 Hz and 200 Hz, from 10 UIpp to 0.25 UIpp
  - 0.25 UIpp between 200 Hz and 400 kHz
  - 0.2 UIpp between 400 kHz and 1 MHz
- Professional applications sinusoidal jitter tolerance mask
  - 10 UIpp between 10 Hz and 200 Hz
  - 20 dB/decade roll-off between 200 Hz and 8000 Hz, from 10 UIpp to 0.25 UIpp
  - 0.25 UIpp between 8000 Hz and 10 MHz
- See masks on following slide

Properties of Digital Audio Signal at IEC 60958 Interface



Thank You

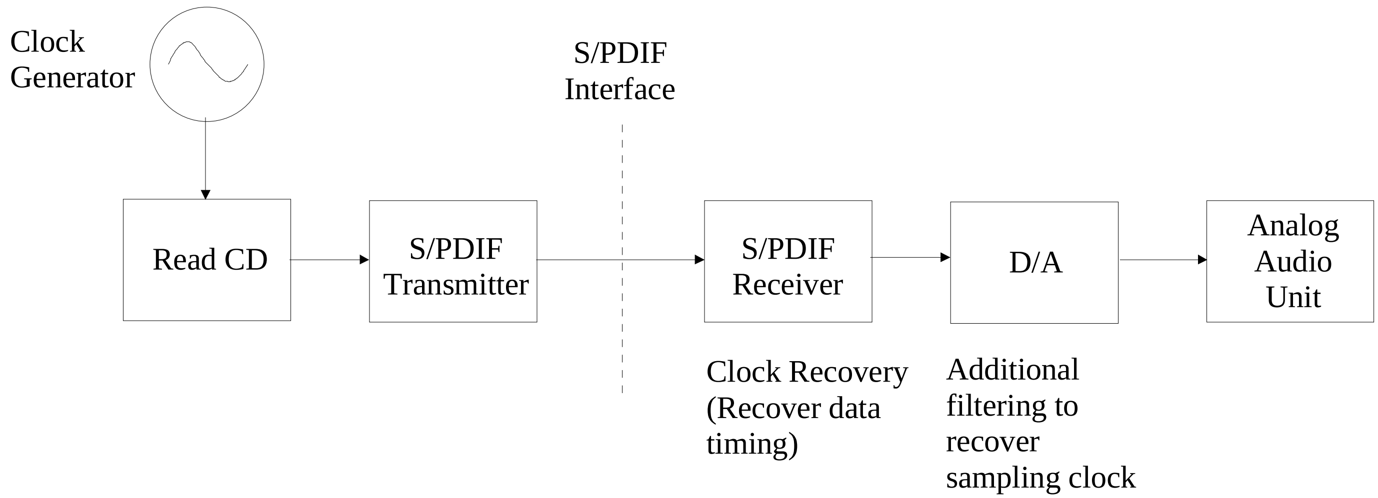
Backup

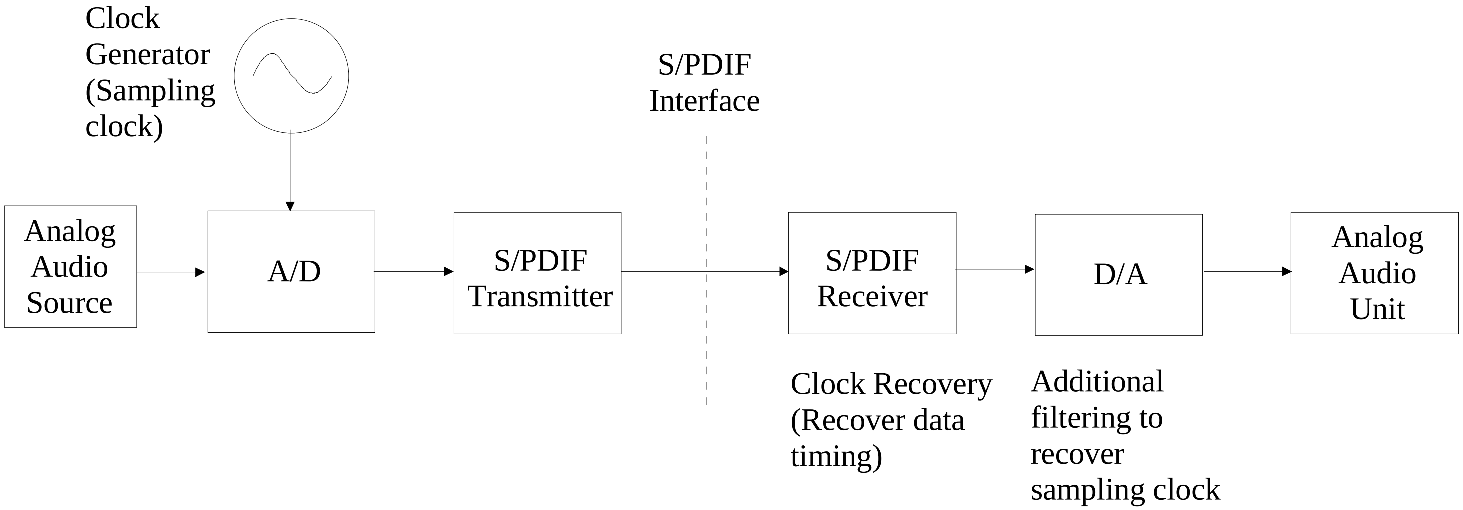
More detailed version of presentation, plus references

Digital Audio - Background

- High-level view of CD player (consumer application)
  - Read CD
    - Requires clock generator
  - Digital Transmitter
  - Digital Receiver
    - Recover data clock
  - D/A converter
    - Recover sampling clock
  - Produce analog audio (speakers, etc.)
  - See schematic on next slide (based on figures in [17])
  - Reference [17] provides a good introduction to digital audio
- Can replace CD by analog audio source and D/A converter
  - Need clock generator for sampling
  - See schematic on next slide

Digital Audio – Background (Cont.)





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Digital Audio Interface Standards (Cont.)

❑Specification of nominal rates

- Professional applications – AES5 [23] and IEC 60988-4 [20]
- Consumer applications – AES5 and IEC 60958-3 [19]
- Note that [23] specifies the actual sampling rates
  - [19] and [20] specify the coding of sampling rates in the frame overhead (not the actual frequency specifications)

❑Specification of wander/synchronization

- Studio applications – AES11 [24]
  - Specifies frequency accuracy, pull-in range, and maximum phase offset (peak-to-peak wander)
  - Specifies Digital Audio Reference Signal (DARS)
- Consumer applications – ICE 60958-3
  - Specifies frequency accuracy and pull-in range

Properties of Digital Audio Signal at IEC 60958 Interface

- See [26] or [27] for a good description of the specifications
- Digital audio signal uses a bi-phase line coding
  - Each data bit occupies 2 UI (unit intervals)
  - Always have a transition at at data bit boundary
  - Additional transition in the middle of a 1 bit
  - No transition in the middle of a zero bit
- Data is carried in frames
  - Each frame is 64 bits, or 128 UI
  - Each frame is composed of 2 subframes of 32 bits (64 UI) each
  - The 2 subframes can be used to carry 2 channels of data
  - Each subframe carries data representing one audio sample
  - Therefore, each channel carries data at a sample rate equal to the frame rate
  - Subframe structure
    - Preamble – 4 bits
    - Audio sample word – up to 24 bits
    - Validity bit – 1 bit
    - User data bit – 1 bit
    - Channel status bit – 1 bit
    - Parity bit – 1 bit

Properties of Digital Audio Signal at IEC 60958 Interface

❑Data frames (cont.)

- Audio data samples need not use the full 24 bits
  - A number of consumer applications use 16 bits
  - If fewer than 24 bits are used for audio data, the unused bits are padded with zeros
    - Therefore, a specified frame rate implies a specified bit rate
  - AES3 defines 4 bits of auxiliary data for the case where fewer than 20 bits of audio information are present ([26] and [27] indicate that this use is rare)
- Detailed description of the validity, user data, channel status, and parity bits, as well as the preamble, are given in [18] – [20], [22], and are summarized in [26] and [27]
  - The details differ for consumer and professional applications
  - The details are not important for the discussion here

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❑Nominal frame rates

- Basic rates defined in [23] are 44.1 kHz (consumer applications) and 48 kHz (professional applications) Corresponds to 5.6448 and 6.144 Mbit/s, respectively
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    - Corresponds to 1.4112, 2.8224, 11.2896, and 22.5792 Mbit/s
      - Bits are the UIs described above, and not the 2-UI bits
  - Professional applications – 12, 24, 96, 192 kHz
    - Corresponds to 1.536, 3.072, 12.288, and 24.576 Mbit/s
- Coding of the rates in frame overhead defined in [19], [20]

Properties of Digital Audio Signal at IEC 60958 Interface

❑Frequency accuracy requirements

- This is the amount the source (sampling) clock is allowed to deviate from nominal, in the long-run
- Note that the same long-term frequency accuracy requirement applies at all interfaces over sufficiently long time intervals
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Properties of Digital Audio Signal at IEC 60958 Interface

❑Frequency accuracy requirements – Consumer applications

- IEC 60958-3 defines 3 levels of accuracy for the sampling clock
- Level I (high accuracy mode): $\pm 50 \times 10^{-6}$ ($\pm$ 50 ppm)
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Properties of Digital Audio Signal at IEC 60958 Interface

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- AES11 specifies 2 levels of frequency tolerance
  - Grade 1
    - frequency tolerance of ± 1 ppm
    - Pull-in range of ± 2 ppm
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    - frequency tolerance of ± 10 ppm (note that the ± 10 ppm tolerance is indicated in AES5 also)
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  - May be used to time all the equipment in a studio
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❑Maximum phase offset requirements (peak-to-peak wander) – professional applications

- Maximum phase offset between input and output of digital audio equipment (wander generation)
  - ± 5% of a frame period (± 6.4 UI)
- Input wander tolerance of digital audio equipment
  - ± 25% of a frame period (± 32 UI)
- Wander accumulation within a studio (e.g., traversing a chain of digital audio devices [26], [27])
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Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter specifications

- As indicated earlier, jitter requirements for sampling clock are much tighter than jitter requirements at digital interface (receiver input)
  - Sampling clock jitter requirement is driven by level of jitter that causes audible effects
    - –Depending on the particular audio source and jitter frequency, this can range from less than 1 ns rms to more than 100 ns rms [29], [30]
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  - Digital interface jitter requirement is driven by need to perform clock and data recovery with acceptable bit error ratio (BER)
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Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter specifications (Cont.)

- Interface jitter (referred to as *Network Limit*) specification
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  - Referred to in IEC 60958 and AES-3 as *intrinsic jitter*
  - Amount of jitter a piece of digital equipment is allowed to produce when the input digital signal is jitter-free
- Jitter transfer
  - Maximum allowable output jitter, excluding generated jitter, for a specified level input jitter
    - Often specified in the form of a frequency response to sinusoidal input jitter

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter generation specifications

- Consumer applications: peak jitter ≤ 0.05 UI
  - Peak-to-peak jitter ≤ 0.1 UI
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- Jitter measurement filter
  - Same for both consumer and professional applications
  - 700 Hz (3 dB point) first-order (minimum phase) high-pass filter
  - Pass-band gain of unity
  - Roll-off to 70 Hz

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter transfer specifications

- Consumer applications
  - Maximum gain peaking: 3 dB
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  - Maximum gain peaking: 2 dB
  - Jitter attenuation is not required, but if it is provided, it should be within the mask on the following slide
    - –No additional specification below 500 Hz (beyond the 2 dB gain peaking limit)
    - –20 dB/decade roll-off between 500 Hz and 1 kHz, from 0 dB to –6 dB
    - –Constant attenuation of –6 dB from 1 kHz to 10 MHz
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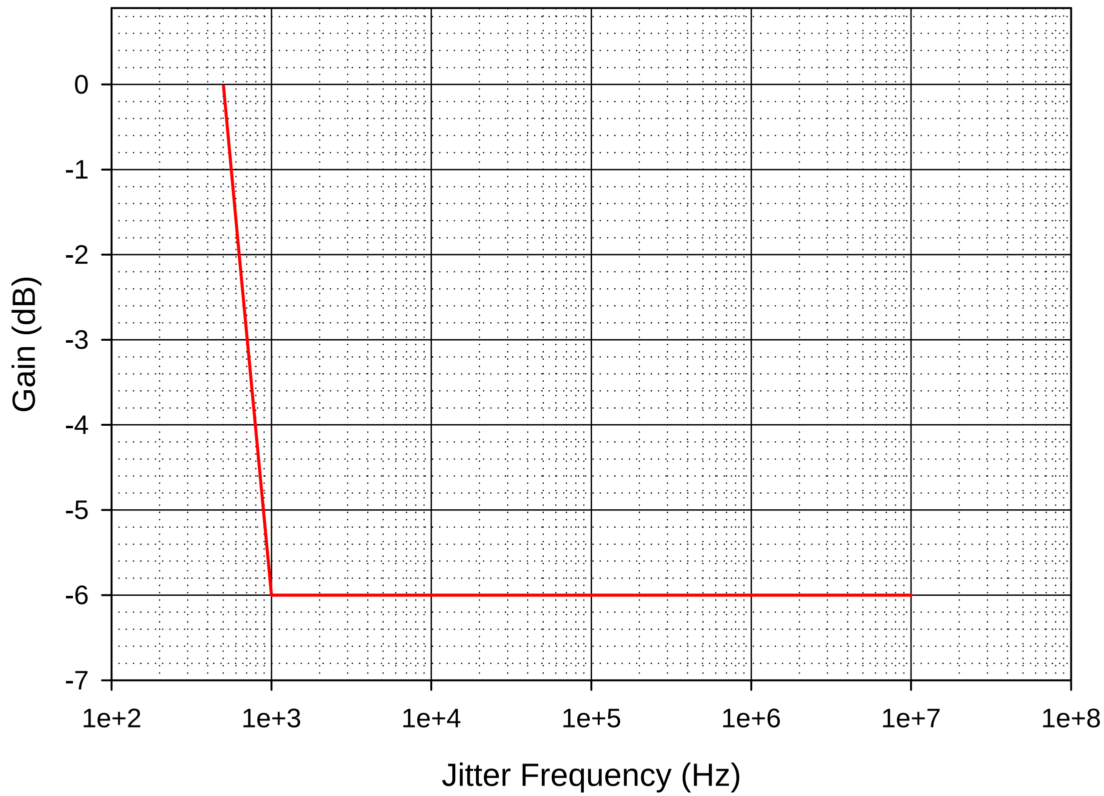
Properties of Digital Audio Signal at IEC 60958 Interface

Jitter Transfer Mask
Professional Applications



Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter tolerance/network limit specifications

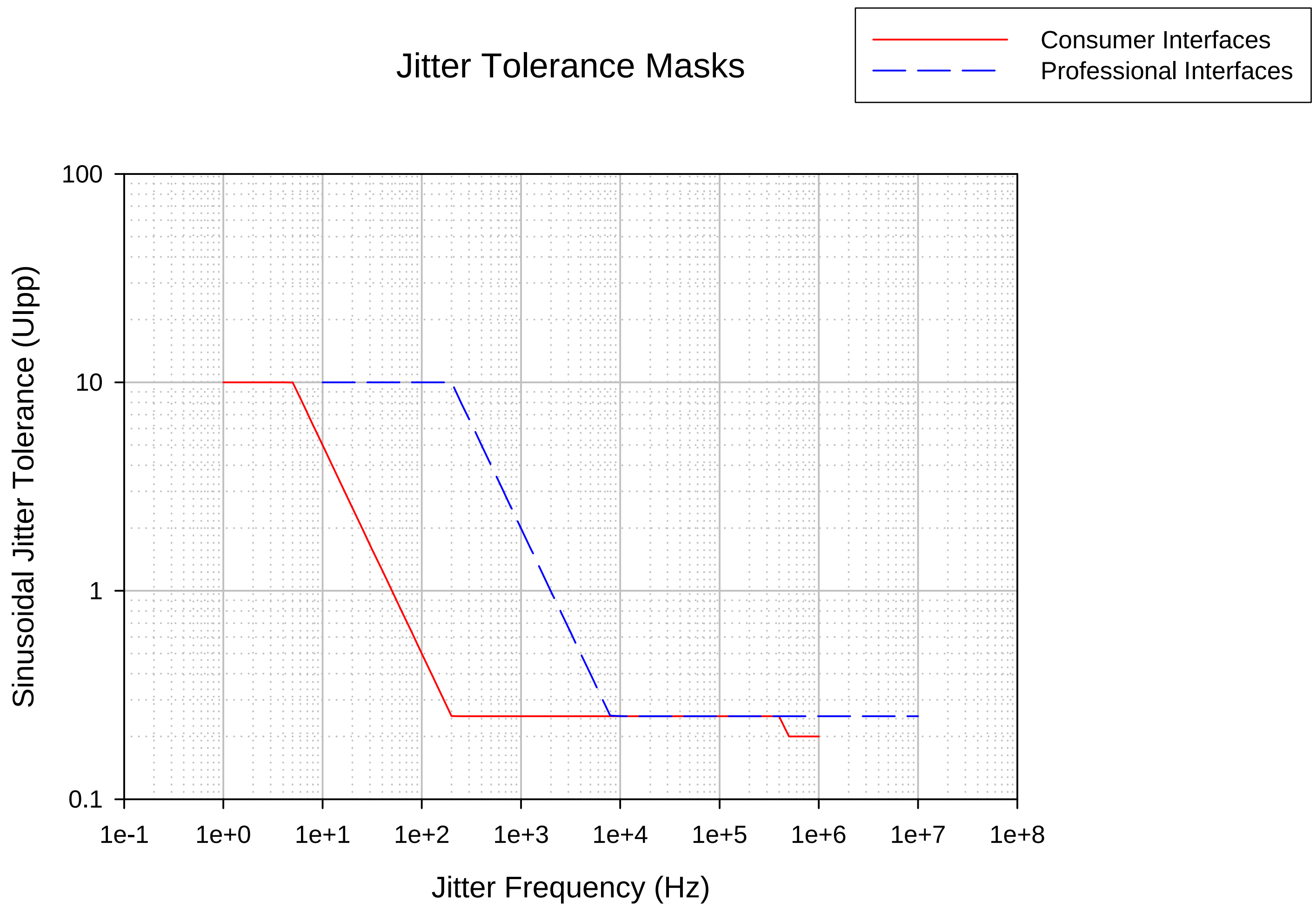
- Represents the amount of jitter the DAC must cope with and still produce an acceptable sampling clock
  - If accumulated jitter exceeds the mask, sampling clock may have excessive jitter, resulting in audible effects
- Professional equipment is required to tolerate higher level of jitter than consumer equipment
  - Appears that this tends to allow consumer equipment to use a single filter for clock recovery and jitter cleanup
    - narrow band jitter cleanup filter would not tolerate higher frequency jitter levels of professional interface
    - Professional equipment would tend to use 2 filters – wide-band clock recovery followed by narrower-band jitter reduction

Properties of Digital Audio Signal at IEC 60958 Interface

❑Jitter tolerance/network limit specifications (Cont.)

- Consumer applications sinusoidal jitter tolerance mask (all jitter values are peak-to-peak)
  - 10 UIpp between 1 Hz and 5 Hz
  - 20 dB/decade roll-off between 5 Hz and 200 Hz, from 10 UIpp to 0.25 UIpp
  - 0.25 UIpp between 200 Hz and 400 kHz
  - 0.2 UIpp between 400 kHz and 1 MHz
- Professional applications sinusoidal jitter tolerance mask
  - 10 UIpp between 10 Hz and 200 Hz
  - 20 dB/decade roll-off between 200 Hz and 8000 Hz, from 10 UIpp to 0.25 UIpp
  - 0.25 UIpp between 8000 Hz and 10 MHz
- See masks on following slide

Properties of Digital Audio Signal at IEC 60958 Interface



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