What's all this Digital stuff about?

An introduction to digital communication

Early days

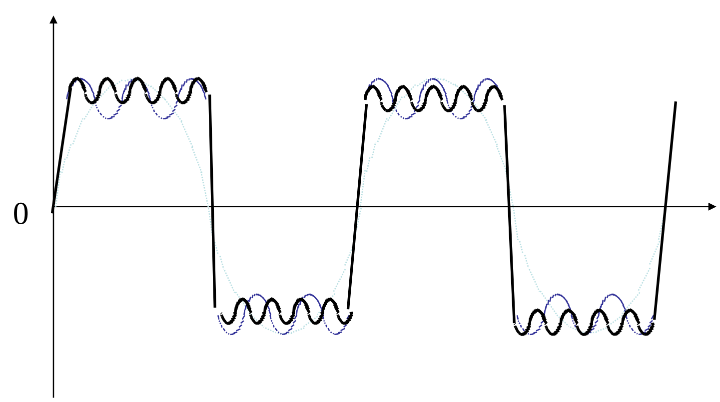
- We live in an "info-communication" world, where telecommunications, computing and media technologies are converging into one multi-media technology
- The mathematical foundations of the theory on which the technology is built goes back onto the 1840s, while the coding theory has its origins even earlier.

Pre-history

- Babylonians coded discrete astronomical observations
- Egyptians developed trigonometry for astronomy and architecture
- Newton found continuous mathematical functions to “interpolate” discrete observations
- Napoleon’s scientists demonstrated that ciphers could be broken
- Fourier showed that continuous functions could be represented by a series of weighted harmonics

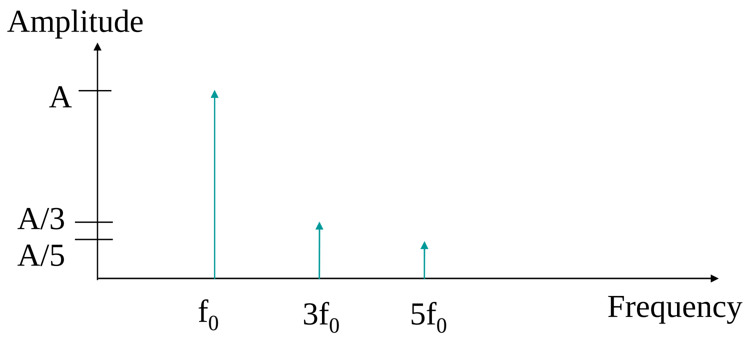
Square Wave – Fourier series

$$f(t) = (4/\pi)[\sin\omega_0 t + (1/3)\sin 3\omega_0 t + (1/5)\sin 5\omega_0 t]$$



Square Wave Spectrum

- Another way of describing the square wave, is by plotting the amplitude of each of the frequency components.



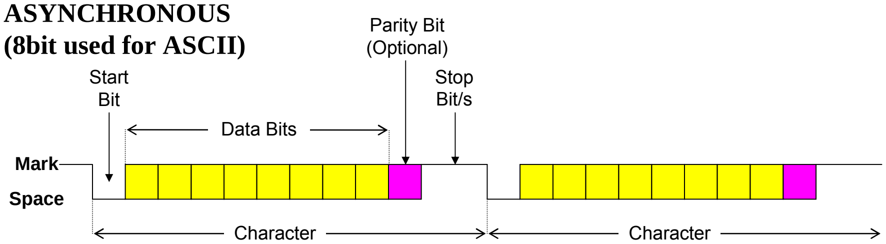
Telegraphy

- 2000 years ago Greeks and Chinese used signal fires
- 1791 Chappe brothers semaphore (fifteen symbols/minute, symbols encode messages)
- 1809, Soemmering's telegraph used 35 wires and sent one letter per minute
- 1830 Henry's electric telegraph: 1835 Morse code – 20-30 words per minute, used by railroads for scheduling and Western Union for telegrams
- 1837 Cooke & Wheatstone demonstrated an electromagnetic code wheel telegraph
- First used Time Division Multiplexing in 1873

Machine telegraphy

- Baudot code used 5 bits to represent letters
- Murray refined the code in 1901 to add control codes and optimise code to minimise tape punch wear
- 1903 Krum patented a “codebar” page printer to print from telegraph codes
- Machine telegraphy widely used in Second World War, particularly with enciphered text
- Computers first used to decrypt the machine telegraph text

ASYNCHRONOUS
(8bit used for ASCII)



Teletype Model 33

Telephony and switched networks

- Early telephone networks connected to a "central office" where operators "patched" calls
- Calls between central offices were "trunk" calls, often handled at a regional level where regional offices had no local access
- Early telephone technology built on technology also used for telegraph line switching
- In 1891, Strowger invented the electromechanical step-by-step switch that allowed customers to automatically call their number without operator intervention. Signalling moved from switch to switch until call connected
- TELEX network is a switched DATA network that used similar switching technology to the POTS to connect data circuits. They used Asynchronous data protocols
- Crossbar switches allowed data control via a separate integrated data network to control switch operation
- Signalling system allowed the path to be selected independently of the hardware to connect the call. Signalling system used asynchronous data packets to carry the signalling over a nearly-synchronous (plesiochronous) data carrier network (Common Channel Signalling – ITU CSS7)

POTS Signalling

- Associated Channel Signalling: that is, DC signalling passed along the same channel as the voice (or data in the switched data network)
- In trunk networks, signalling changed to DTMF (audio) signals – led to "phone phreaking"
- From 1977, ITU-T introduced Common Channel Signalling (CCS7) which separated the signalling data from the voice channels
- By 1980, the signalling system evolved into a comprehensive packet switching network that used packets in a synchronous (or nearly synchronous) data channel

Computers

- Computers were based on Turing's Computable Numbers paper, which hypothesised a "universal machine"
- Turing helped build Colossus, the first stored programme computer, in Bletchly Park as part of the WW2 code breaking effort.
- Eckert and Mauchly, with von Neumann as the mathematician consultant, built Eniac for the US military
- Large computers were expensive, so access via telegraph lines allowed multiple users. Code standardised to 8-bit ASCII code in 1963
- External access was via teleprinters using telegraph lines, either fixed or via a switched telegraphy network
- As switched telephony networks came to dominate in the late 1970s, modems were used to convert machine telegraphy signals into audio for carriage over audio telephony switched circuits
- High speed local storage used large multi-track data tape recorders, and later, hard disks serially recording "blocks" of data. Data transport protocols introduced, but unique to each machine
- Data blocks on disk were "interleaved" to facilitate seek on a high speed rotating disk

Mini-computer system 1976

Data Networks

- Used to provide wider access to mainframe computers
- IBM first used audio tones to carry data over a switched telephony network
- IBM's SABRE network for airline bookings in 65 cities and over 2000 terminals
- Late 1960s, ARPANET developed; layers of coding differentiated, and protocols established
- Asynchronous data Packets (frames) built around HDLC frame operating in a synchronous channel

HDLC Frame Format

8 Bits	8 Bits	8 Bits	0 to N Bits	16 Bits	8 Bits
FLAG 01111110	**ADDRESS**	**CONTROL**	**DATA**	**FCS** CRC-16	**FLAG** 01111110

- **Data** - an arbitrary number of bits, can be of any length
  - only present in *Information* and *Unnumbered frames*
- **Control** - defines the type of frame
  - *Information*, *Supervisory*, *Unnumbered*
- **Address** - always identifies the address of the secondary station. There are three types: -
  - Individual - the secondary station's own address
  - Group - common to a number (group) of stations, but not all
  - Broadcast (FF_H or 11111111_2) - common to all stations
- **FCS** - Frame Check Sequence, for error correction (CRC-16)
- **Flag** - bit oriented protocols delineate frames with a special bit sequence or character (in HDLC - **01111110**)

Ethernet (Local Area Network)

- Ethernet arose from Xerox PARC, and supported by Digital Equipment and Intel
- Standardised as IEEE802 in 1983 which defined the logical link protocol which is frame based and based on the High-level Data Link Control (HDLC) frame
- Carries asynchronous packets over a synchronous (or nearly synchronous) data channel
- Differentiates physical logical layers to facilitate interoperability of many data protocols, and takes an object-oriented approach to specifying interfaces between levels leaving application designers to code the various interfaces as they wish

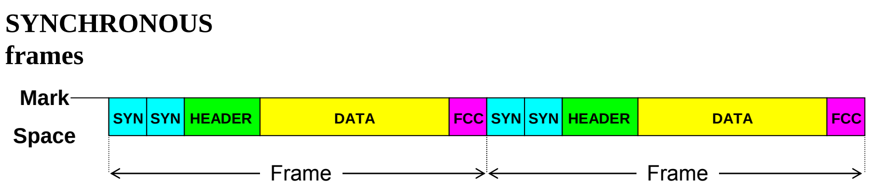
OSI Network layers

OSI Layers	OSI	ITU-T	Internet	IEEE 802 Ethernet
7. Application	File transfer and access management	X.500 Directory Services X.400 Message handling		
6. Presentation	Presentation			
5. Session	Session			
4. Transport	Transport		Transport Control Protocol (TCP)	
3. Network	Internet protocol (IP)	I series (ISDN) X.25 (Packet Switching) X.21 (Circuit switching)	Internet Protocol (IP)	
2. Data link	8802.2			802.2 Logical Link Control
1. Physical	8802.3/4.5			802.3/4/5

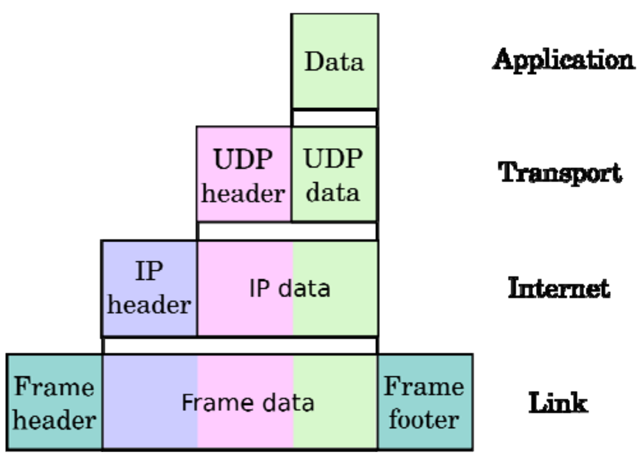
Figure 1: Comparison of OSI layers with other standards **(Duck, Bishop and Read (1996) *Data Communication for Engineers*. Addison-Wesley, Harlow. P. 16.**

Synchronous frames

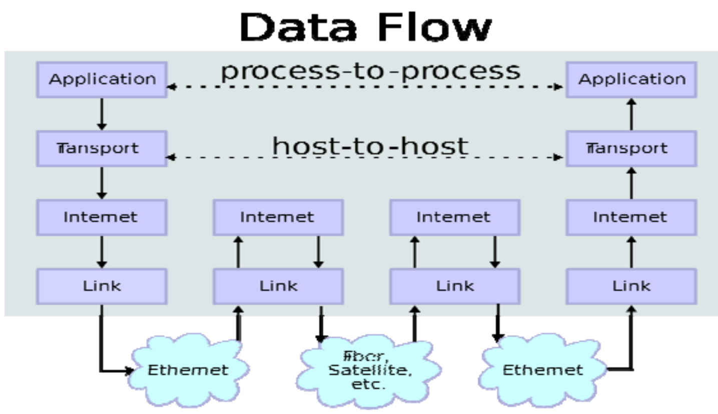
- Synchronous frames live in a continuous series of bytes that are filled with synchronising "flags" if there is no data present.
- Packets (frames) of data are inserted into the synchronous (or near synchronous) data stream as long as there is at least one "flag" between each packet



Encapsulation of Application Data



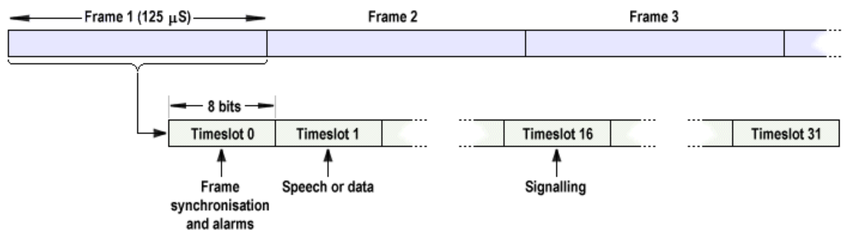
Data flow in an inter-network



Digital Telephony (ISDN)

- ITU-T settled on a voice channel for telephony to be 300 Hz to 3.5 kHz, which with guard bands rounds to 4 kHz channel
- By applying Nyquist rules, the ITU-T settled on an 8 kHz sampling rate, and 8 bit samples, leading to a data stream of 64 k bits per second
- European version of ISDN used TDM to fit 30 voice channels and two data channels (for synchronisation and signalling) into a 2 Mb/s synchronous (or nearly synchronous) frame – called “primary-rate ISDN”

E1 Digital Carrier format



- The E1 frame format (timeslots 1-15 and 17-31 are used for data)

Digitising Audio

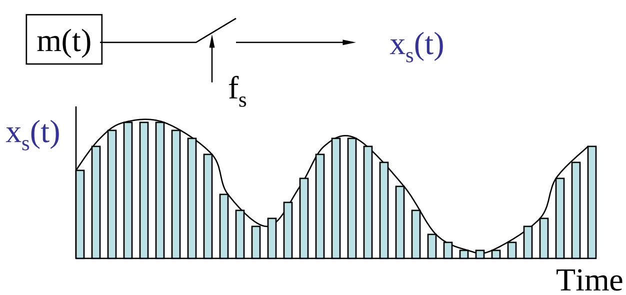
- In 1947, Goodall describe PCM experiments in which he explored the effect of bit depth
- He found that if the volume of the analogue voice signal was regulated (compressed), three bits produced intelligible speech, six bits was not quite enough, but some more bits would be enough
- In telephony, the unregulated volume is addressed by using non-linear sampling (digital compression)
- In the 1980s, musical instrument manufacturers used digital compression because of the high cost of memory

Digitising process

- Band limit audio to ensure that NO frequencies above half the sample rate exist
- Sample and hold the audio throughout the sample period
- Convert the audio into PCM in an A/D converter – there are many different approaches, but contemporary encoders use either a “flash” (parallel) approach, or 1 bit oversampled delta-sigma encoder
- Package audio for transmission, storage or processing
- Convert digital to analogue in a D/A converter, which could take one of many forms
- Filter the analogue output to remove any image frequencies and to smooth the output

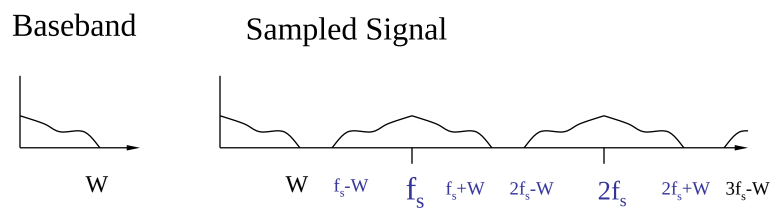
Sampling Theorem

- *Paraphrasing:* ***A sampled waveforms contains ALL the information without any distortions, when the sampling rate exceeds twice the highest frequency contained by the sampled waveform.***



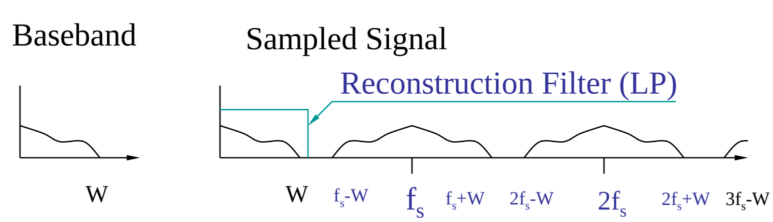
Spectrum

- For a sampled message signal the spectrum might be:



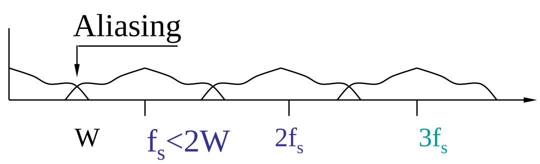
Reconstruction

- The original message signal can be reconstructed from the sampled signal by simply low pass filtering the sampled signal.
- The reconstruction filter removes the image frequencies
- The recovered energy is low unless the samples are held for the full sample period



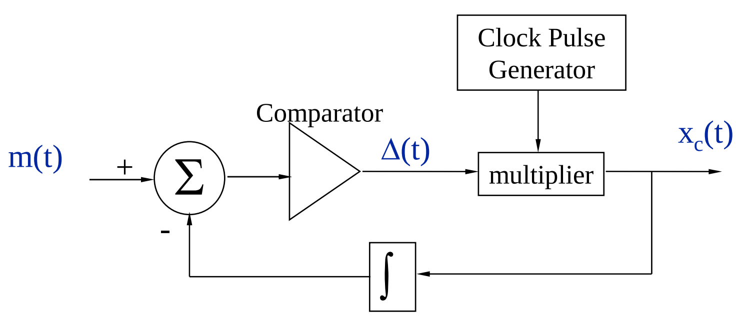
Aliasing

- When f_s is less than $2W$, overlapping of the spectra occur, resulting in spurious frequency components appearing in both the baseband and in the spectrum around the sampling frequency.
- These frequencies are mixed with wanted signals in the baseband and become audible, resulting in distortion in the recovered signal



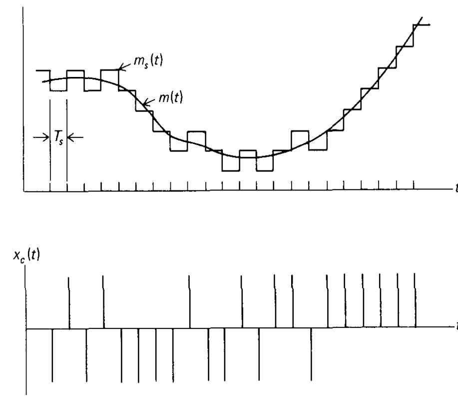
Delta Modulation

- Delta modulation (also called single bit A/D conversion) encodes the message signal as a sequence of binary pulses.



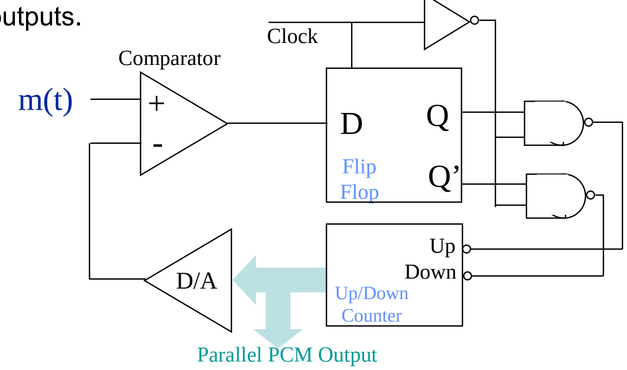
Delta modulation Waveforms

- In this example the pulse output is the output of the comparator.
- The pulse train produces the square wave representation of the message after it has been integrated.
- A low pass filter recovers the message from the squared waveform.
- The sampling frequency is much higher than the PCM sampling frequency: 2Mb/s for 24 bit, 192 kHz sampling rate

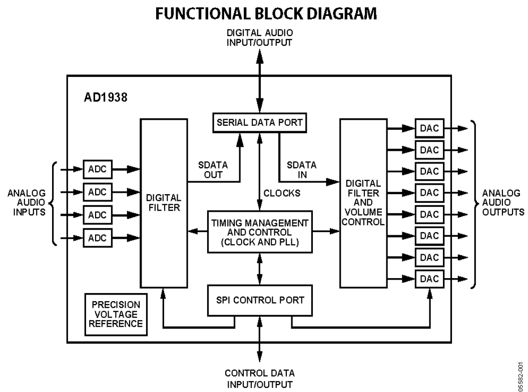


A Delta-sigma PCM Encoder

- A variant of the delta modulator can be configured with an up/down counter and D/A converter as a digital integrator.
- The raw code can then be taken directly from the counter outputs.



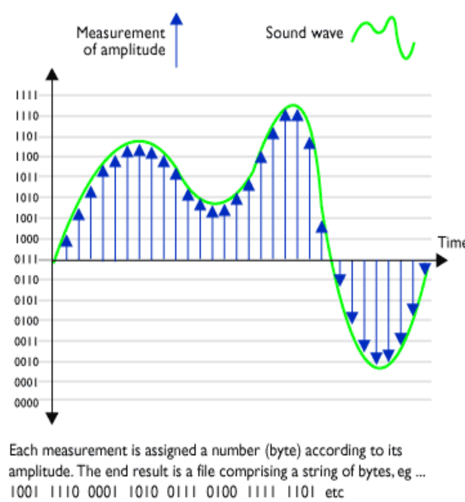
E.g. A/D, D/A converter AD1938



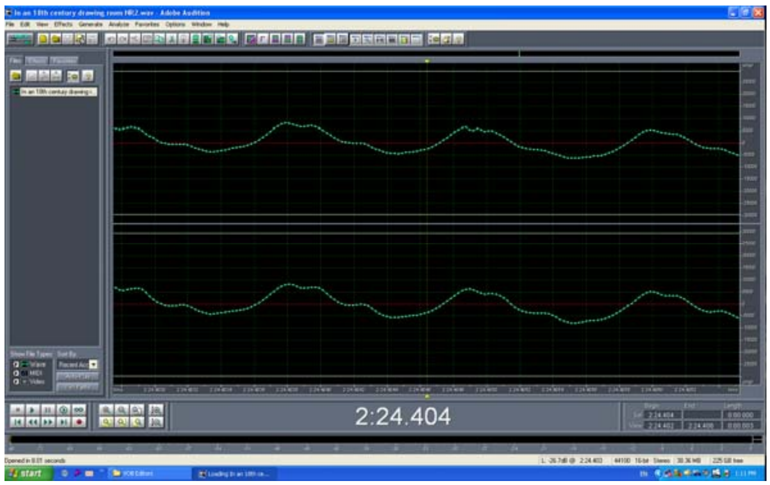
- The AD1938 uses 1 bit conversion in the ADC to create 24 bit PCM, 8 kHz to 192 kHz sample rates

Pulse-Code Modulation

- The pulse code modulator converts the message signal into a series of numbers corresponding to the levels into which the message signal is divided.
- PCM for audio is in 2s complement code



Sampled Audio



Quantisation Error

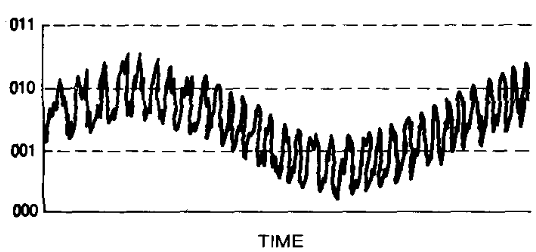
- The quantisation error is the difference at sample time between the actual message signal amplitude and the amplitude represented by the code.
- At best the waveform coincides with the quantisation level; at worst the error is half a quantisation step.
- The quantisation error produces a signal contamination similar to analogue noise when the signal amplitude is large, but when the signal amplitude is small, the errors are *correlated* with the signal so they appear more like harmonic distortion, noise modulation or “musical noise.”

Quantisation Levels

- With a binary number system, the word length (number of bits in the word) determines the number of quantising increments available.
- Thus an n bit word will have 2^n quantising levels. An 8 bit word represents 256 increments; 10 bits, 1024 increments and 16 bits 65536 increments.
- The greater the number of bits in the code word, the better the approximation to the message signal.
- Digital Audio is encoded to PCM with a twos complement code.

Dither

- A small amount of noise added to the modulating signal, with an amplitude about the size of a quantisation level, randomises the effects of quantisation error making it less objectionable.
- The penalty for this quality improvement is much less objectionable slightly increased random noise.
- This added noise signal is called Dither.
- The dither signal effectively adds up to two bits of resolution



Error detection and correction

- In text, the redundancy in language is enough for a competent reader to detect and correct errors: E.g. Tha cut an the hat (the cat on the mat)
- In non-real-time packet-based transport systems it is enough to know that an error has occurred, so a request for retransmission can be issued
- In real-time systems there is no opportunity for errors to cause retransmission, so the must be corrected
- Error correction comes in two kinds:
  – Block codes (e.g. parity, CRC, hamming codes, etc.)
  – Convolution codes: (Viterbi, etc.)
- All error correction involves adding redundancy to the signal

Reed-Solomon Codes

- Forward Error correcting of a single serial data stream can also be performed by adding, prior to transmission, redundant parity bits to the source data.
- These additional bits are generated by a cyclic block code (Reed-Solomon).
- On receipt of the transmitted data, the added parity bits are used to determine which bits, if any, are in error so they can be corrected.

Interleaving

- When an error burst hits a frame of data the whole frame may be lost
- By interleaving sub-blocks of the data before transmission, and restoring the order on reception, the burst of errors is distributed across the sub-frames and is therefore more correctable
- With concatenated error coding, errors can be reduced up to the limit of the channel capacity
- Error coding and processing introduces latency

Viterbi Coding

- In Convolution, or trellis, coding using an algorithm attributed to Viterbi, the output of the encoder depends on the input state and the current state of the encoder.
- The current state represents a memory of the signal that has passed before.
- The fact that convolutional codes contain memory makes them different from block codes that use parity coding within a data block to detect and correct or conceal burst errors.

Decoding Viterbi Sequences

- The Viterbi algorithm provides an effective means of finding the best possible means of finding the best possible sequence given the soft (corrupted by noise) received sequence.
- This is achieved by finding for each sequence of states the most likely path or sequence leading into that state and eliminating the other paths.

Tape Storage

- NHK (Denon) produced a digital tape recorder in the 1960s. They released their first digital recordings in 1971
- In the early to mid-1970s, the BBC began developing a digital recorder. 3M took the BBC technology and produced two digital recorders: a two-track mastering recorder; and a thirty-two track recorder.
- In 1974, Sony announced a PCM recorder on two-inch tape with 58-track stationary head. Success with this machine led to the Sony PCM-1 recorder with external coding feeding a video signal to a Betamax recorder.
- The 3M recorders used 16-bit, 50 kHz sample rate
- In 1977, Soundstream released a four-track recorder based on a proprietary analogue converter feeding an 8-track Honeywell data recorder
- 1982, Sony and Studer introduce families of Digital Audio Stationary Head (DASH) recorders.
- Mitsubishi produced a family of stationary head recorders that had some success but were incompatible with the Sony and Studer machines

Compact disks

- A compact disk is all about robust error correction.
- The raw PCM is organised into six 16-bit words, then reorganised into twenty-four 8-bit words
- Then the new frame is subjected to a RS(24,28) code adding 4-bits capable of correcting two errors
- After interleaving to spread this frame over 109 frames, the new frames are subjected to RS(32,28) code and again interleaved, with a further 8-bit sub-code for timing and start-of-track information.
- Finally, the physical layer is encoded with an 8 into 14 code that has a low number of transitions between 0 and 1. Each frame has 3 more timing bits added, and each frame has a unique 24 bit synchronising word

DAT and ADAT

- In 1987, Sony released a Digital Audio Tape (DAT) as a digital replacement for general consumer use
- The DAT was built on the PCM-1 with a built-in video recorder
- In 1992, Alesis shipped the first ADAT 8-track digital recorders, recoding 16-bit PCM on high-speed helical tracks on super VHS tape

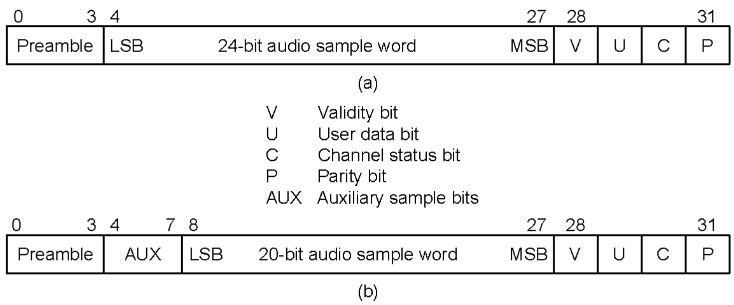
Digital Audio Workstations

- Soundstream built a DAW based on a PDP-11, with custom audio I/O and large Braegen 14 inch disk drives
- 1979, Fairlight CMI. Reborn Fairlight DAW in the 1990s
- 1987, Digidesign releases Sound Tools for the Macintosh computer
- 1991, Digidesign release Protools multi-track recorder. By 1997, Protools had up to 48 tracks and a multitude of sample rates and bit depths
- 1990s, Cool Edit Pro was a powerful DAW for Windows platforms. By version 2, hardware limitations were removed and non-destructive editing was available. Now Adobe Audition
- 1996, Sonic Solutions, grew out of Lucas Film's Edit Droid. Now Sonic Studio
- 2004, Audacity available as a free DAW

AES Standards

- AES3-2003 audio transport standard for two channels
  - Data contained in sub-frames for each channel, combined with space for two voice-grade channels and synchronising data
- AES10-2003 Multichannel audio interface for multi-track audio over coaxial cable or optical fibre
  - Physical link a synchronous stream like ISDN, with audio frames of varied size depending on bit depth and sample rate inserted into the carrier stream with flags separating frames, thus operating like Ethernet
  - 4B5B coded flag inserted at 40-bit channel boundaries
  - Standard does not have control codes, but codes based on HDLC format are listed in the Appendix
- AES47-Multi-channel audio over ATM
  - Includes CCS7 signalling in Virtual channels to carry packets at a minimum guaranteed bit-rate of 100Mb/s
  - Carried over Cat5 cable or PSTN at a 155.5Mbits/s interface

AES-3 Sub-frame format



- Two sub-frames make up a two-channel frame
- 16-bit samples use the 20-bit sample word format with unused bits set to 0

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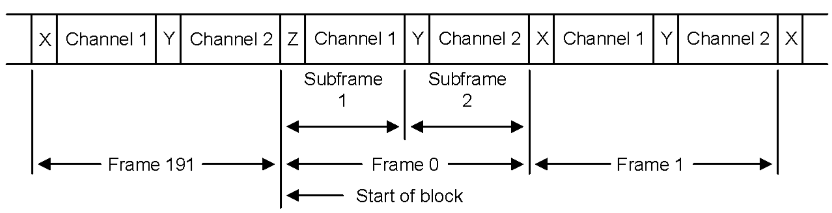
AES 3 frame sequence



- The first subframe normally starts with preamble X. However, the preamble changes to preamble Z once every 192 frames. This defines the block structure used to organize the channel status information. The second subframe always starts with preamble Y.

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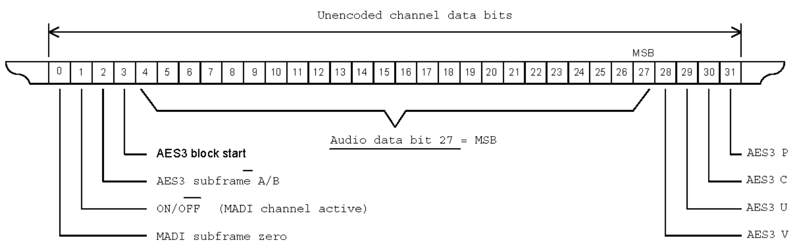
AES 10 Channel data format

- Physical layer establishes a synchronous data stream at 125 Mb/s, with a continuous data capacity of 100 Mb/s
- Peak data rate for 64 channels @ 48 kHz is 98.304 Mb/s
- The rest of the channel is filled with flags and synch



AES10 Frame pattern with one synchronising flag

MADI subframe	0	1	2		26	27	0	1		26	27	0
Audio channel	Ch 0	Ch 1	Ch 2		Ch 26	Ch 27	Ch 0	Ch 1		Ch 26	Ch 27	Ch 0
Sample number	n	n	n		n	n	n + 1	n + 1		n + 1	n + 1	n + 2
AES3 subframe	A	B	A		A	B	A	B		A	B	A

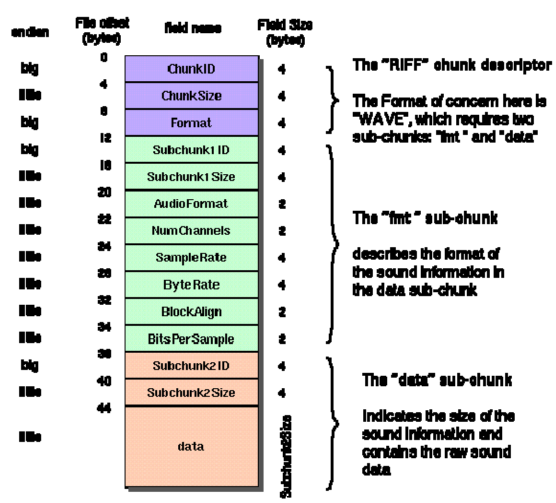
20,8 µs

- **96 kHz with 28 channels working (96 kHz frame pattern, using 10,4 µs framing, one channel-zero flag per frame**

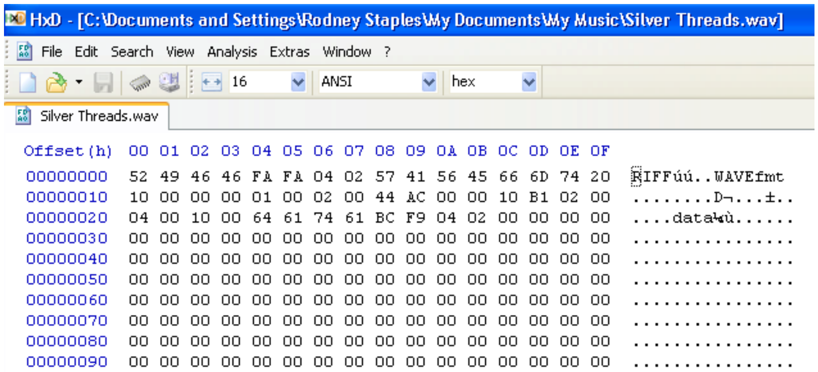
Computer file formats

- Uncompressed files (wav, aiff) contain a header followed by the raw PCM, optional metadata in footer. Raw files just contain the audio samples
- Lossless compression: 22 file types, including those by Apple, Dolby True HD, DTS HD Master Audio, Free Lossless Audio Codec (FLAC) and others
- Lossy compression: 27 file types, including MPEG-1 Layer 3, called mp3
  - Uses polyphase quadratic filter to break file into 32 frequency bands, which allows for identification of frequency bands to encode coarsely or to eliminate because of masking
  - and Modified Discrete Cosine Transform (MDCT) to compress frequency content in each band (Modification overlaps DCT frames)
- Newer codecs like:
  - AAC, an ITU-T designed codec that uses sub-band coding like mp3
  - AC3 introduced by Dolby uses just MDCT
  - Vorbis (Ogg) and Windows Media (wma) use MDCT alone

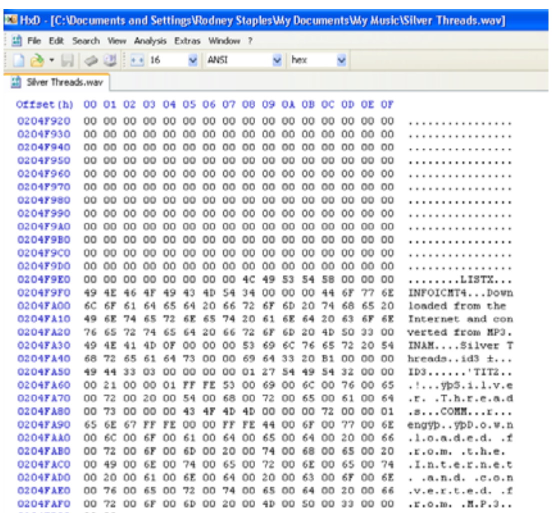
The Canonical WAVE file format



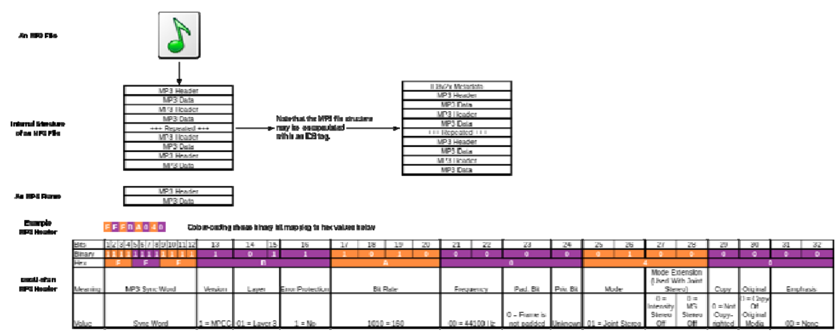
Wav file header



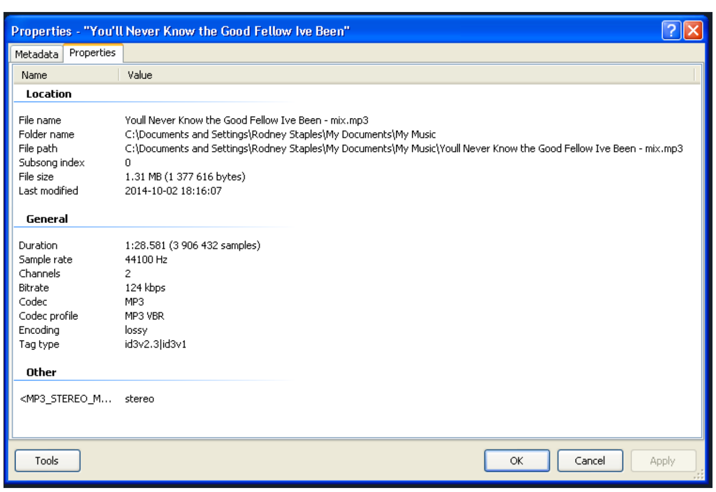
Wav File footer



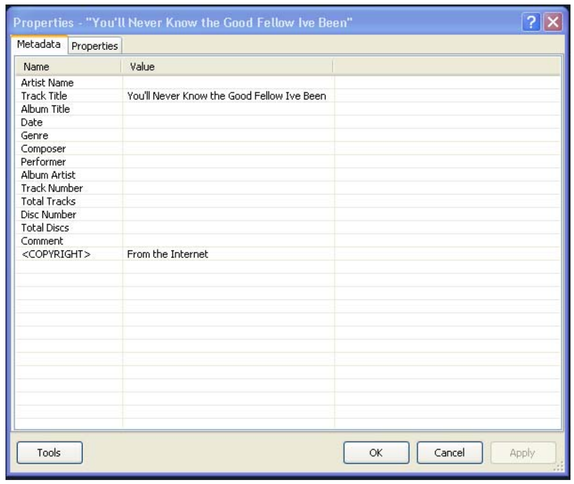
MP3 file structure



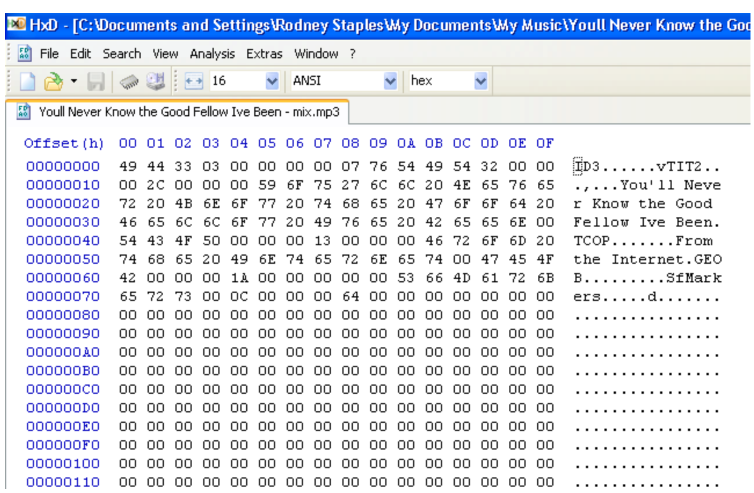
MP3 file data



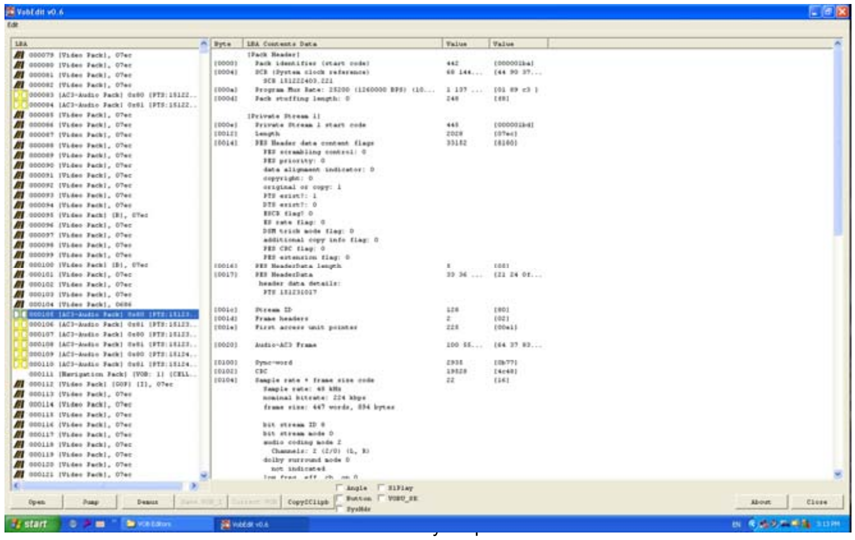
MP3 file metadata



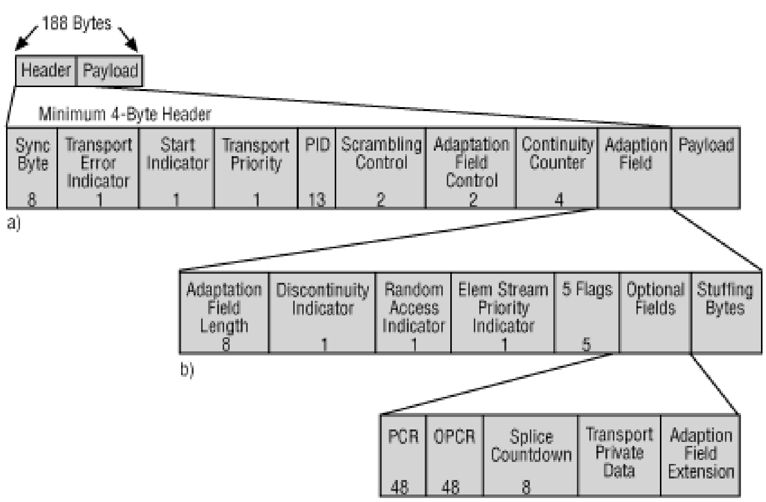
Metadata in an MP3 file header



Audio header in a video transport stream



MPEG 2 video Transport Stream Packets



Audibility

- 1981, Muraoka, Iwahara, and Yamada observe that average listeners cannot *reliably* detect a filter that removed all frequencies above 15 kHz
- Some exceptional listeners could reliably detect 16 kHz filter and a very few could identify 20 kHz filter
- 2000, Shlien found results depend on "learning effect", listening level (set at 70dB SPL), fatigue and masking of surrounding signals
- There is virtually NO energy above 40kHz. Although we can't hear this, it may cause audible inter-modulation in non-linear circuits
- Lavery justifies that sampling rates above 96 kHz are a waste of bandwidth and have no audible effect
- 2000, Blech and Yang show no reliable way of detecting between SA-CD and CD in a double blind study. When level was increased 14 dB the difference in the noise floor facilitated discrimination
- 2004, Nishiguchi, Iwaki and Ando show no significant difference between signal with high frequency components (above 20 kHz) and that without HF components
- Stuart argues for 58 kHz sample rate and 20-bit sample depth with a flat noise floor for transparent recordings

Free Audio Software

- Foobar 2000 player and converter:
  - http://www.foobar2000.org/download
- HxD Hex editor: for reading files directly
  - http://download.cnet.com/HxD-Hex-Editor/3000-2352_4-10891068.html
- VobEdit: for displaying audio in MPEG 2 transport stream:
  - http://www.videohelp.com/tools/VobEdit
- Audacity Digital Audio Workstation:
  - http://audacity.sourceforge.net/