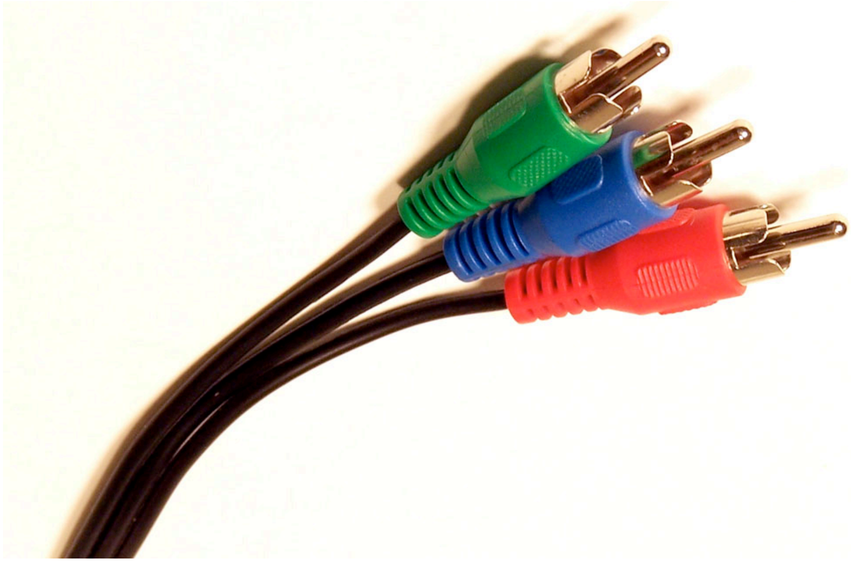
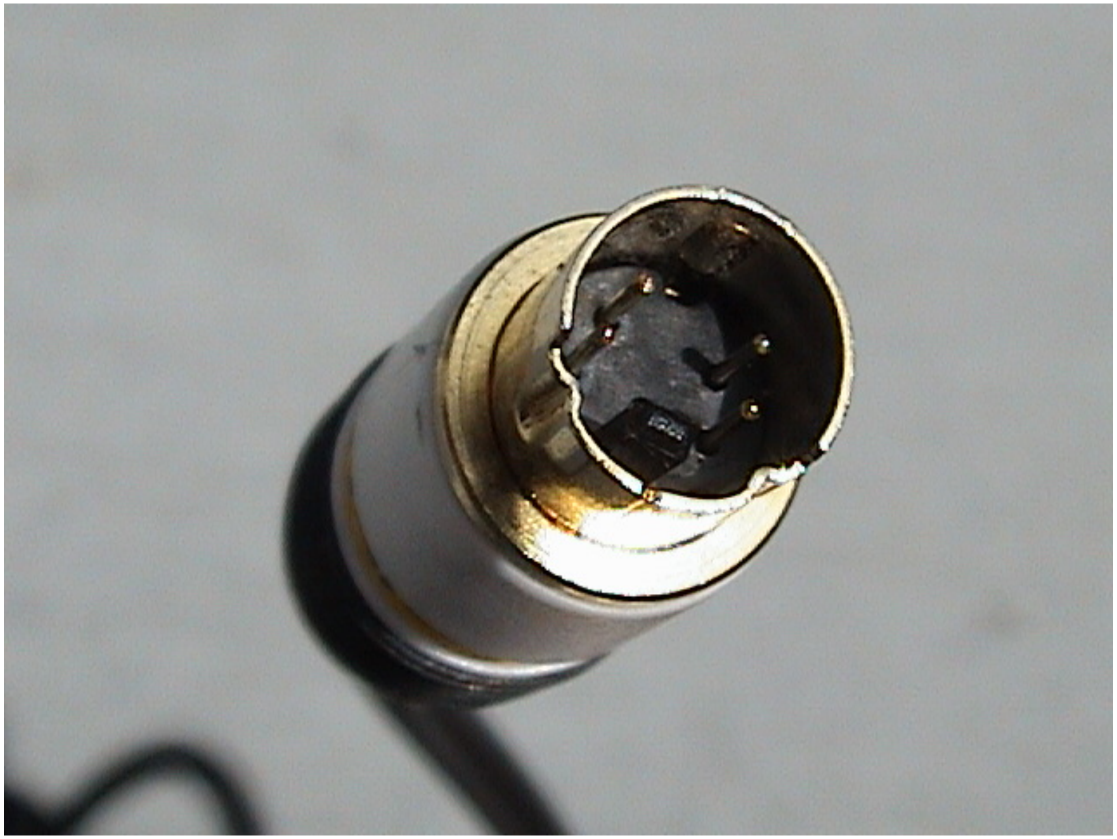
Chapter 5: Video

- Types of video signals
  - Component video
    - Three separate cables carry the RGB or YCbCr signals (Analog)
    - Best form of analog video



Pictures from Wikipedia

- S-Video
  - One wire for luminance
  - One wire for both chroma component



- Composite video
  - Single RCA cable carries luminance and chroma component
  - Signals interfere
- For even cheaper connections, VCRs have a connector that broadcasts signals in Channel 3/4. Signals are modulated and demodulated, losing fidelity

Digital connections

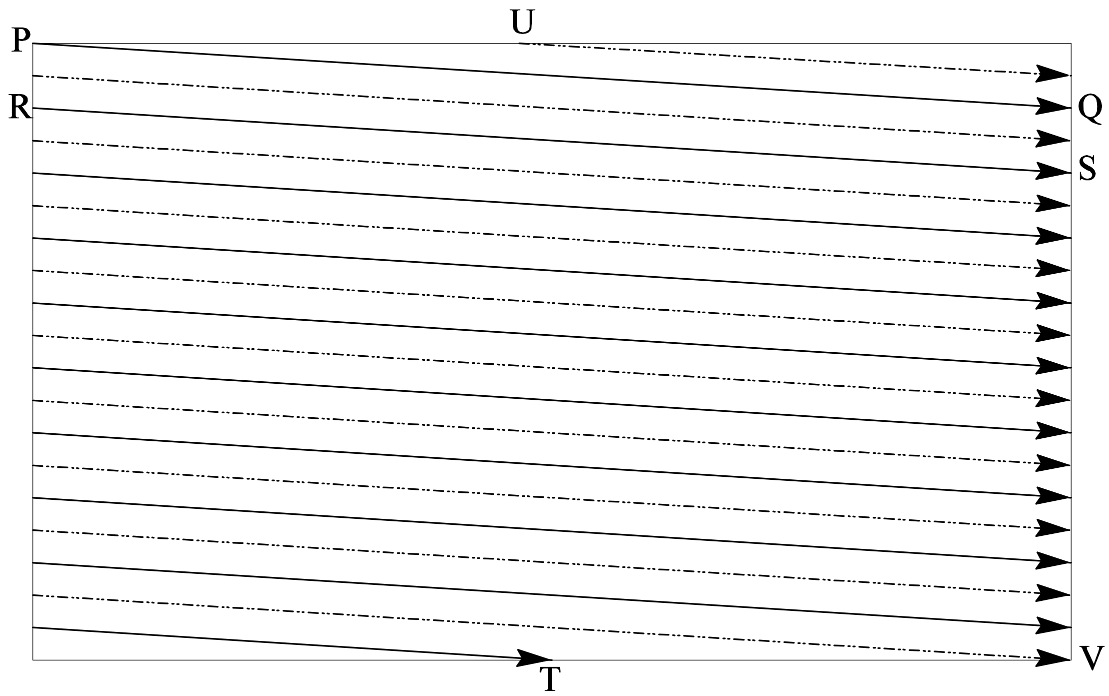
- DVI
  - Example display modes (single link):
    - HDTV (1920 × 1080) @ 60 Hz
    - UXGA (1600 × 1200) @ 60 Hz
    - WUXGA (1920 × 1200) @ 60 Hz
    - SXGA (1280 × 1024) @ 85 Hz
  - Example display modes (dual link):
    - QXGA (2048 × 1536) @ 75 Hz
    - HDTV (1920 × 1080) @ 85 Hz
    - WQXGA (2560 × 1600) pixels (30" LCD)
    - WQUXGA (3840 × 2400) @ 41 Hz

- HDMI
  - High definition Multimedia Interface
    - uncompressed, all-digital audio/video interface
    - High-Bandwidth Digital Content Protection (HDCP) DRM
    - Without HDCP HD-DVD & Bluray can restrict quality to DVD
    - Supports 30-bit, 36-bit, and 48-bit (RGB or YCbCr)
    - Supports output of Dolby TrueHD and DTS-HD Master Audio streams for external decoding by AV receivers

Analog video

- Interlaced Raster Scan
  - Way to increase refresh frequencies by alternating odd and even scan lines in separate refresh
  - NTSC has a notion of blacker than black signal that triggers a beginning of line
  - 525 scan lines at 29.97 frames per second
  - VHS: 240 samples per line, S-VHS: 400-425, Hi-8: 425, miniDV: 480x720)
  - PAL and SECAM: 625 scan lines, 25 frames per second
    - NTSC: 6 MHz, PAL&SECAM: 8 MHz

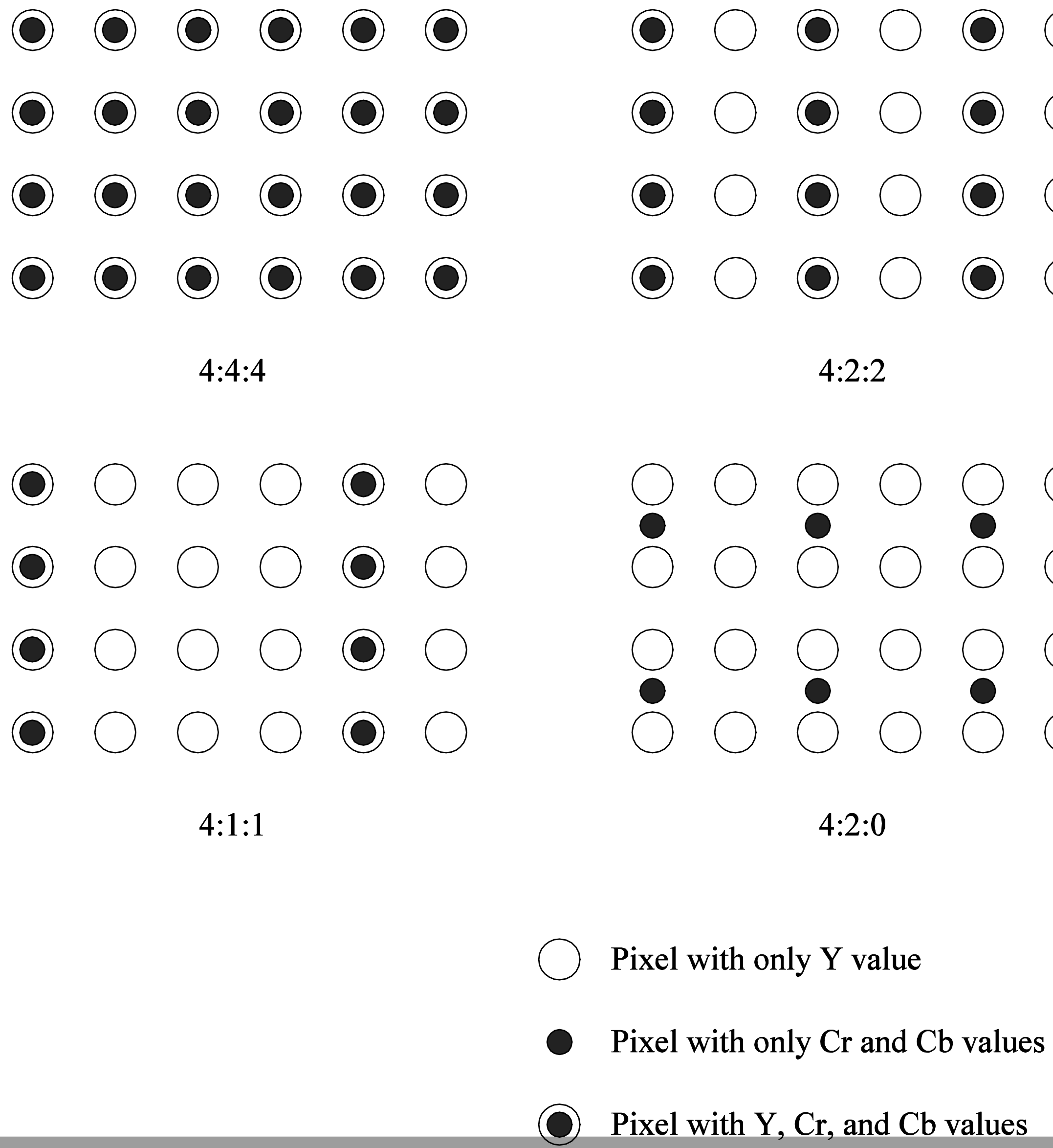
Interlacing



Digital video - Chroma subsampling

- 4:4:4, 4 pixels of Y, Cb and Cr each
- 4:2:2 : Cb and Cr are half
  - NTSC uses this subsampling
- 4:1:1 : Cb and Cr are factor of four
  - DV uses this subsampling
- 4:2:0 : Cb and Cr are subsampled, effectively 4:1:1
  - Used in JPEG, MPEG and HDV

Chroma sub-sampling



Digital video standards

- **CCIR Standards for Digital Video**
- CIF stands for Common Intermediate Format specified by the CCITT.
  (a) The idea of CIF is to specify a format for lower bitrate.
  - (b) CIF is about the same as VHS quality. It uses a progressive (non-interlaced) scan.
  - (c) QCIF stands for "Quarter-CIF". All the CIF/QCIF resolutions are evenly divisible by 8, and all except 88 are divisible by 16; this provides convenience for block-based video coding in H.261 and H.263

Digital video specifications

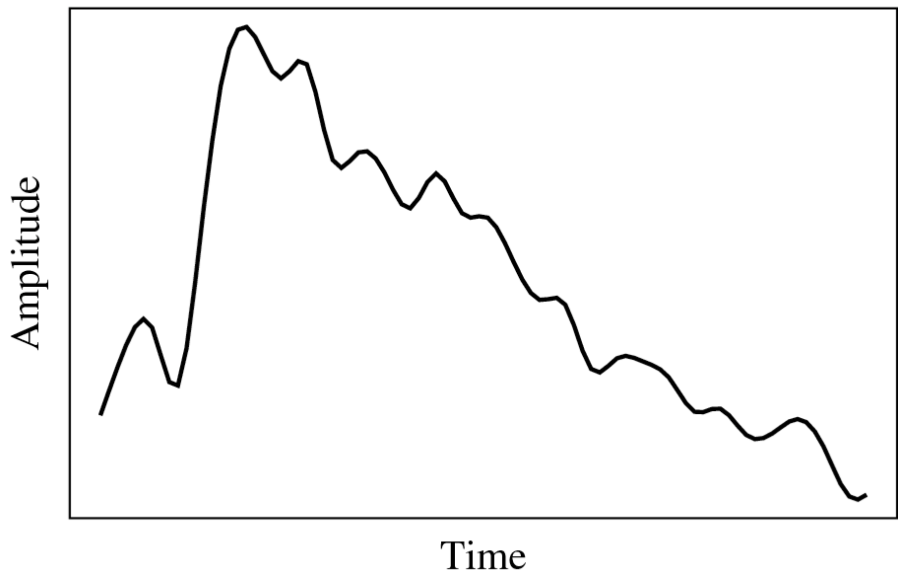
	CCIR 601 525/60 NTSC	CCIR 601 625/50 PAL/ SECAM	CIF	QCIF
Luminance resolution	720 x 480	720 x 576	352 x 288	176 x 144
Chrominance resolution	360 x 480	360 x 576	176 x 144	88 x 72
Colour Subsampling	4:2:2	4:2:2	4:2:0	4:2:0
Fields/sec	60	50	30	30
Interlaced	Yes	Yes	No	No

High Definition TV

- US style:
  - MPEG 2 video, Dolby AC-3 audio
  - 1920x1080i - NBC, CBS ..
  - 1280x720p - ABC, ESPN
  - 1920x1080p - Xbox 360, PSP3
    - 1920x1080p24 – cinematic
- HDV uses rectangular pixels: 1440x1080
- For video, MPEG-2 is chosen as the compression standard. For audio, AC-3 is the standard. It supports the so-called 5.1 channel Dolby surround sound, i.e., five surround channels plus a subwoofer channel.

What is Sound?

- Sound is a wave phenomenon like light, but is macroscopic and involves molecules of air being compressed and expanded under the action of some physical device
  - Since sound is a pressure wave, it takes on continuous values, as opposed to digitized ones



Digitization

- **Digitization** means conversion to a stream of numbers, and preferably these numbers should be integers for efficiency
  - Sampling means measuring the quantity we are interested in, usually at evenly-spaced intervals
  - Measurements at evenly spaced time intervals is called sampling. The rate at which it is performed is called the *sampling frequency*. For audio, typical sampling rates are from 8 kHz (8,000 samples per second) to 48 kHz. This range is determined by the Nyquist theorem
  - Sampling in the amplitude or voltage dimension is called **quantization**

Quality	Sample Rate (Khz)	Bits per Sample	Mono / Stereo	Data Rate (uncompressed) (kB/sec)	Frequency Band (KHz)
Telephone	8	8	Mono	8	0.200-3.4
AM Radio	11.025	8	Mono	11.0	0.1-5.5
FM Radio	22.05	16	Stereo	88.2	0.02-11
CD	44.1	16	Stereo	176.4	0.005-20
DAT	48	16	Stereo	192.0	0.005-20
DVD Audio	192 (max)	24(max)	6 channels	1,200 (max)	0-96 (max)

Nyquist theorem

- The Nyquist theorem states how frequently we must sample in time to be able to recover the original sound. For correct sampling we must use a sampling rate equal to at least twice the maximum frequency content in the signal. This rate is called the **Nyquist rate.**
- **Nyquist Theorem**: If a signal is **band-limited**, i.e., there is a lower limit f_1 and an upper limit f_2 of frequency components in the signal, then the sampling rate should be at least $2(f_2 - f_1)$

Signal to Noise Ratio (SNR)

- The ratio of the power of the correct signal and the noise is called the *signal to noise ratio* (**SNR**)
  - a measure of the quality of the signal.

- The SNR is usually measured in decibels (**dB**), where 1 dB is a tenth of a **bel**. The SNR value, in units of dB, is defined in terms of base-10 logarithms of squared voltages, as follows:

$$SNR = 10\log_{10}\frac{V_{signal}^2}{V_{noise}^2} = 20\log_{10}\frac{V_{signal}}{V_{noise}}$$

Common sounds

Threshold of hearing	0
Rustle of leaves	10
Very quiet room	20
Average room	40
Conversation	60
Busy street	70
Loud radio	80
Train through station	90
Riveter	100
Threshold of discomfort	120
Threshold of pain	140
Damage to ear drum	160

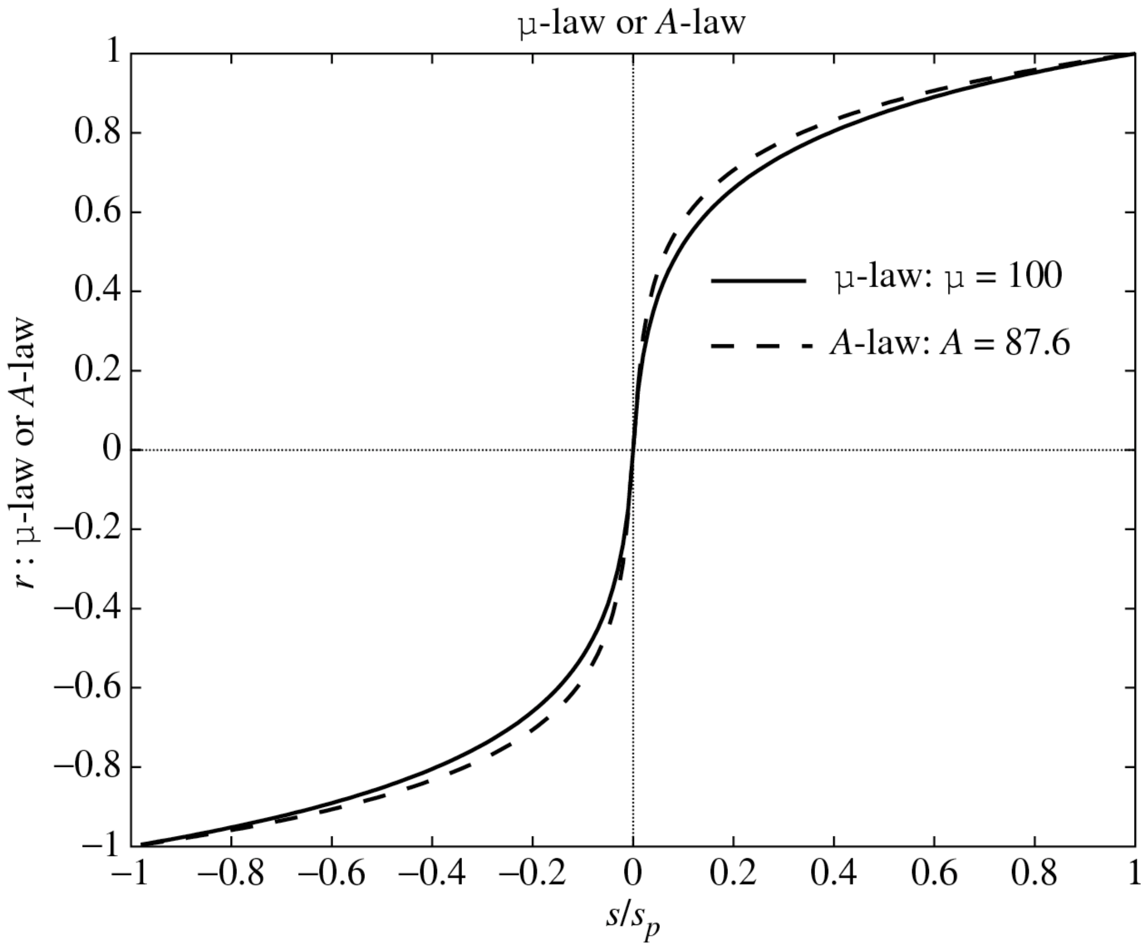
Signal to Quantization Noise Ratio (SQNR)

- If voltages are actually in 0 to 1 but we have only 8 bits in which to store values, then effectively we force all continuous values of voltage into only 256 different values. This introduces a roundoff error. It is not really “noise”. Nevertheless it is called **quantization noise** (or quantization error)
- **Linear and Non-linear Quantization**
  - **Linear format**: samples are typically stored as uniformly quantized values
  - **Non-uniform quantization**: set up more finely-spaced levels where humans hear with the most acuity
    - Weber’s Law stated formally says that equally perceived differences have values proportional to absolute levels:
      - $\Delta\text{Response} \propto \Delta\text{Stimulus}/\text{Stimulus}$

Nonlinear quantization

- Nonlinear quantization works by first transforming an analog signal from the raw s space into the theoretical r space, and then uniformly quantizing the resulting values. Such a law for audio is called μ-**law** encoding, (or **u-law**). A very similar rule, called A-**law**, is used in telephony in Europe

- The μ-law in audio is used to develop a nonuniform quantization rule for sound: uniform quantization of r gives finer resolution in s at the quiet end

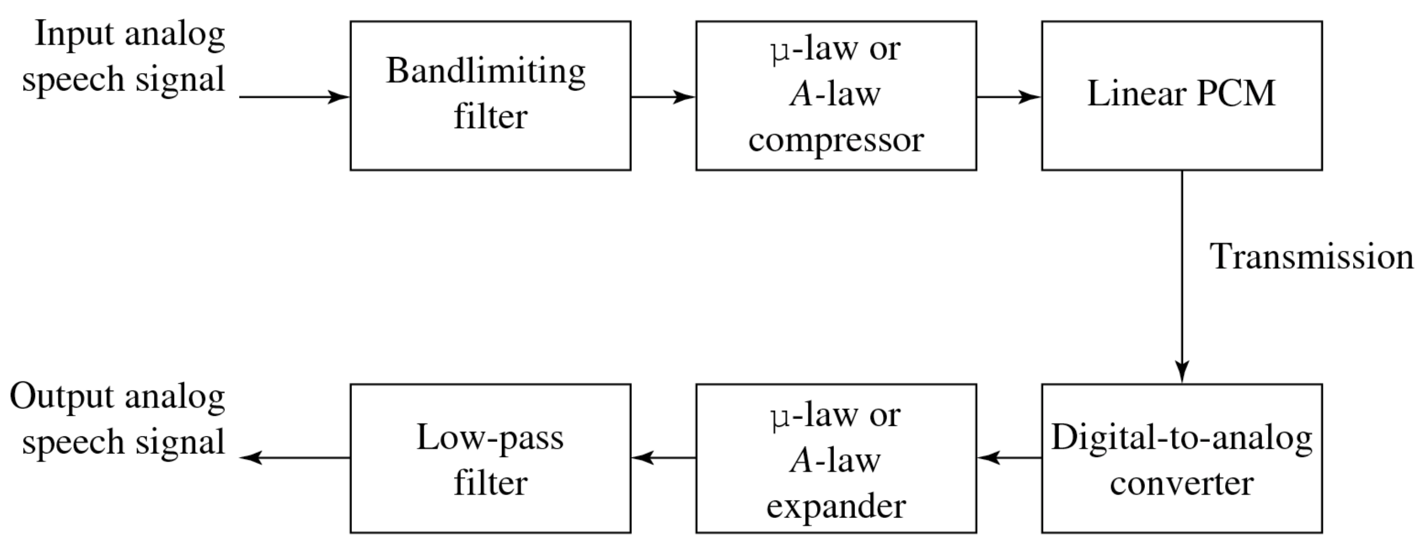


Synthetic sounds

- Frequency modulation (with a magnitude envelope)
- Wav table: the actual digital samples of sounds from real instruments are stored. Since wave tables are stored in memory on the sound card, they can be manipulated by software so that sounds can be combined, edited, and enhanced
- MIDI is a scripting language — it codes “events” that stand for the production of sounds. E.g., a MIDI event might include values for the pitch of a single note, its duration, and its volume.

6.3 Quantization and Transmission of Audio

- producing quantized sampled output for audio is called **PCM** (Pulse Code Modulation). The differences version is called **DPCM** (and a crude but efficient variant is called **DM**). The adaptive version is called **ADPCM**



Differential coding

- If a time-dependent signal has some consistency over time (“temporal redundancy”), the difference signal, subtracting the current sample from the previous one, will have a more peaked histogram, with a maximum around zero

ADPCM

- **ADPCM** (Adaptive DPCM) takes the idea of adapting the coder to suit the input much farther. The two pieces that make up a DPCM coder: the quantizer and the predictor.
  - In Adaptive DM, adapt the quantizer step size to suit the input. In DPCM, we can change the step size as well as decision boundaries, using a non-uniform quantizer.
  - We can carry this out in two ways:

    (a) **Forward adaptive quantization**: use the properties of the input signal.

    (b) **Backward adaptive quantization**: use the properties of the quantized output. If quantized errors become too large, we should change the non-uniform quantizer.

    We can also **adapt the predictor**, again using forward or backward adaptation. Making the predictor coefficients adaptive is called *Adaptive Predictive Coding* (APC)