Introduction to DSP

A signal is any variable that carries information. Examples of the types of signals of interest are

- Speech (telephony, radio, everyday communication)
- Biomedical signals (EEG brain signals)
- Sound and music
- Video and image
- Radar signals (range and bearing).

Digital signal processing (DSP) is concerned with the digital representation of signals and the use of digital processors to analyse, modify, or extract information from signals.

Many signals in DSP are derived from analogue signals which have been sampled at regular intervals and converted into digital form. The key advantages of DSP over analogue processing are

- Guaranteed accuracy (determined by the number of bits used)
- Perfect reproducibility
- No drift in performance due to temperature or age
- Takes advantage of advances in semiconductor technology
- Greater flexibility (can be reprogrammed without modifying hardware)
- Superior performance (linear phase response possible, and filtering algorithms can be made adaptive)
- Sometimes information may already be in digital form.

There are however (still) some disadvantages

- Speed and cost (DSP design and hardware may be expensive, especially with high bandwidth signals)

- Finite wordlength problems (limited number of bits may cause degradation).

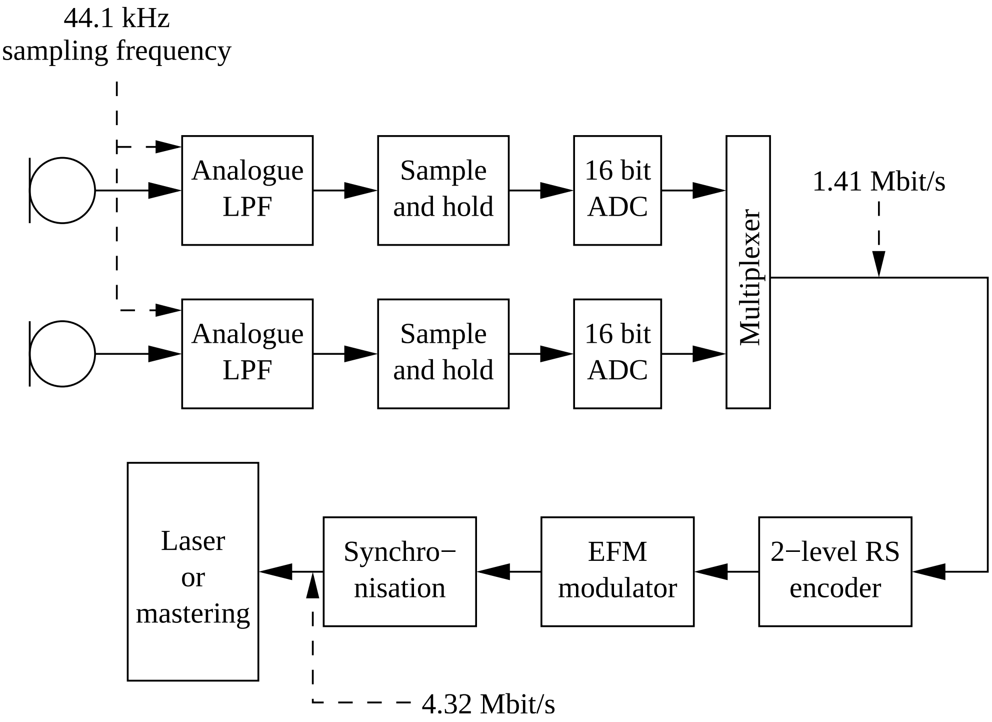
Application areas of DSP are considerable:

- Image processing (pattern recognition, robotic vision, image enhancement, facsimile, satellite weather map, animation)
- Instrumentation and control (spectrum analysis, position and rate control, noise reduction, data compression)
- Speech and audio (speech recognition, speech synthesis, text to speech, digital audio, equalisation)
- Military (secure communication, radar processing, sonar processing, missile guidance)
- Telecommunications (echo cancellation, adaptive equalisation, spread spectrum, video conferencing, data communication)
- Biomedical (patient monitoring, scanners, EEG brain mappers, ECG analysis, X-ray storage and enhancement).

Example: audio signal reconstruction in CDs

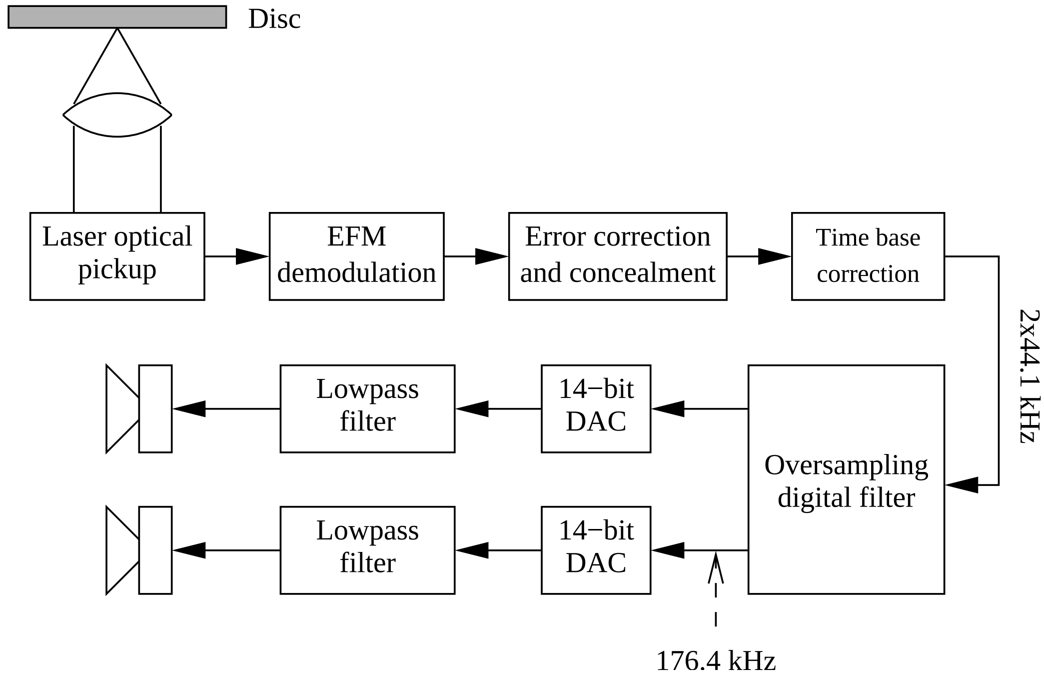
Information on a compact disk is recorded on a spiral track as a succession of pits (10^6 bits/mm^2).

The recording or mastering process is depicted below:



- Analogue signal in each stereo channel is sampled at 44.1 kHz and digitised to 16 bits (90 dB dynamic range), resulting in 32 bits per sampling instant.
- Encoded using a two-level Reed-Solomon code to enable errors to be corrected or concealed during reproduction.
- An EFM (eight-to-fourteen) modulation scheme translates each byte in the stream to a 14 bit code, which is more suitable for disc storage (eliminates adjacent 1's, etc.)
- The resulting bit stream is used to control a laser beam, which records information on the disc.

The audio signal reconstruction process is demonstrated below:



- Track optically scanned at 1.2 m/s
- Signal is demodulated, errors detected and (if possible) corrected. If correction is not possible, errors are concealed by interpolation or muting
- This results in a series of 16 bit words, each representing a single audio sample. These samples could be applied directly to a DAC and analogue lowpass filtered
- However, this would require high specification lowpass filters (20 kHz frequencies must be reduced by 50 dB), and the filter should have linear phase. To avoid this, signals are upsampled by a factor of 4. This makes the output of the DAC smoother, simplifying the analogue filtering requirements.
- The use of a digital filter also allows a linear phase response, reduces chances of intermodulation, and yields a filter that varies with clock rate.