

COMP30291 Digital Media Processing

Level: 3 **Credit rating:** 10 **Degree(s)** all CS
Prerequisites: None **Co-requisites:** None
Duration 12 weeks **Lectures:** 22 in total at 2 lectures per week
Example classes: additional **Laboratories:** None
Assessment 2 hour exam
Lecturer: Dr. Barry M.G. Cheetham barry@man.ac.uk

Aims:

This course is concerned with the application of digital signal processing (DSP) to speech, music and video in the fields of computing and telecommunications. Students will gain an appreciation of the technology, the design tools currently available and some of the most important techniques for media processing in general purpose and embedded systems.

Learning Outcomes:-

- 1) Understand the significance of digital signal processing in multi-media technology, storage and communications.
- 2) Familiarity with fundamental concepts such as 'linearity', 'time-invariance', 'impulse response', 'convolution', 'frequency response', 'z-transforms' and the 'discrete time Fourier transform'. as applied to signal processing systems.
- 3) Knowledge of digital filters and their application to digitised sound and images.
- 4) Understand how FIR and IIR type digital filters: may be designed and implemented in software.
- 5) Use the "MATLAB" language and "signal processing toolboxes" for designing, implementing and simulating digital signal processing (DSP) operations as applied to speech, music and images..
- 6) Specify the "real time" implementation of DSP operations using special purpose fixed point 'DSP microprocessors'.
- 7) Understand analogue/digital conversion as required for the digital processing of analogue signals.
- 8) Understand the discrete Fourier transform (DFT), its applications and its implementation by FFT techniques. Gain some knowledge of the 2-D FFT and its application to image processing and compression.
- 9) Knowledge of how speech is digitised and compressed by A-law, ADPCM and CELP. Also knowledge of how music is digitised and compressed by MP3 taking advantage of the FFT and perceptual masking.

Assessment of Learning Outcomes: 2 hour written examination

Contribution to programme learning outcomes: A2, A3, A5, B1, B2, B3, C4

Reading List and Supporting Material:

- (1) S.W. Smith, "Scientist and Engineer's Guide to Digital Signal Processing " California Tech. Publishing, 2nd ed., 1999, available complete at: <http://www.dspguide.com/>
- (2) Dr. Cheetham's lecture notes, supporting material and solutions to examinations and other problems are available at: <http://www.cs.man.ac.uk/~barry/mydocs/COMP30291>

Syllabus:**Section 1: Introduction (1 lecture):**

Definition of continuous time (analogue), discrete time and digital signals.
Sampling and quantisation in general terms.
Introduction to signal processing as applied to speech, music and multimedia.

Section 2: Analogue and digital filters (1 lecture):

Transfer function and frequency-responses of analogue and digital filters.
Low-pass, band-pass and band-stop analogue and digital filters.
Butterworth low-pass gain response approximation.

Section 3: Digital signal processing (5 lectures)

Recursive and non-recursive difference equations. Signal flow-graphs and their implementation by simple computer programs.
Linearity, time invariance and impulse-response for discrete time systems.
Definition of finite impulse response (FIR) and infinite impulse response (IIR) type digital filters. Stability and causality. Time-domain convolution.
Frequency response as discrete time Fourier transform (DTFT) of impulse-response.
Gain and phase responses. Linear phase and group delay. Inverse DTFT.
Use of MATLAB for analysing digital filters.

Section 4: FIR type digital filters (2 lectures)

Design of FIR digital filters by the Fourier series approximation method.
Implementation on personal computers and in real time on dsp chips.
Effect of increasing order and use of non-rectangular windows. Alternative methods.

Section 5: IIR type digital filters and the z-transform (4 lectures)

System function, $H(z)$, as z-transform of impulse response.
Relationship between system function, difference equation, signal flow-graph and software implementation of FIR and IIR type digital filters.
Poles and zeros. Distance rule for estimating the gain response of a digital filter from an Argand diagram (z-plane) of poles and zeros. Design of a digital IIR "notch" filter and a resonator by pole/zero placement. Application to these filters to sound recordings.
Design of IIR digital filters using MATLAB.

Section 6: Digital processing of multimedia signals (3 lectures)

Sampling theory, aliasing and the effect of quantisation and 'sample and hold' reconstruction. Sampling rate conversion and oversampling to simplify analogue filters.
Overall design of digital systems for processing speech, music and multimedia.

Section 7: The discrete Fourier transform (DFT) & its applications (3 lectures)

Derivation of DFT from DTFT. Inverse DFT. Effects of windowing and frequency-domain sampling. Non-rectangular windows. The 'fast Fourier transform' algorithm (FFT). Use of the FFT for spectral estimation and media processing. 2-D FFT and its application to an image. Case studies in MATLAB.

Section 8 Processing speech, music & video (3 lectures)

Digitising speech with bit-rate compression (A-law, ADPCM, CELP). Digitising music (CDs, MPEG & MP3)

Discussion of exercises and problems