Digital processing of audio and video signals

Finalità

This course aims at exploring the design and application aspects of Digital Signal Processing, whose most common exploitation area is in the processing of audio and video signals.

The organization of both the contents and the teaching method has two distinctive features: the contents are enriched, year after year, with advanced application topics, also thanks to the fundamental contribution of the students; as for the teaching method, half of the time is spent in the laboratory, applying the techniques discussed in class to digital signals acquired on a PC, through the numerical software Matlab.

Assessment is based on an oral exam and on the production of a small essay – either an application project or a research on an advanced topic – carried out by students organized in small groups (two to four).

Programma

DESIGN OF DIGITAL FILTERS

Relationship between difference equations, block diagrams and flow graphs. Structures for Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) filters: direct forms, cascade and parallel forms; transposed forms; structures for linear phase FIR filters.

Design methods for digital filters: specifications, choice of the response type, coefficients calculation and realization structure.

Synthesis of linear phase FIR filters: the windowing method; the Kaiser window method; the optimal (equiripple) method: the alternation theorem and the Parks-McClellan algorithm; the frequency sampling method.

Synthesis of IIR filters from analog filters: review of Butterworth and Chebychev analog filters; pole-zero placing; the impulse invariance method; the matched Z transform; the bilinear transformation method.

EFFECTS OF FINITE PRECISION ARITHMETIC

Discrete time random signals: properties, statistical and time averages.

Correlation, covariance and spectral representation. Use of the Z transform in computing average power.

Effects of coefficients quantization; effects of roundoff errors in fixed point arithmetic; effects of zero input limit cycles.

SPECTRAL ANALYSIS AND ESTIMATION

Spectral estimation of stationary signals: the periodogram. Spectral estimation of nonstationary signals: analysis of speech signals and the spectrogram. Estimation of the autocorrelation.

SOURCE CODING ON A PERCEPTUAL BASIS

The Discrete Cosine Transform (DCT) for image compression. The speech signal: acoustic phonetics and the acoustic theory of speech production; digital models for speech signals. Analysis and synthesis of speech signals: Linear Predictive Coding (LPC).

Propedeuticità

Elaborazione Numerica dei Segnali A (Digital Signal Processing A)

Testi consigliati

A. V. Oppenheim, R. W. Schafer, J. R. Buck, “Discrete-Time Signal Processing, 2nd Ed.”, Prentice-Hall, 1999.