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.

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 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.

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.

SPECTRAL ANALYSIS AND ESTIMATION

Spectral estimation of stationary signals: the periodogram. Frequency resolution and Leakage. Spectral estimation of nonstationary signals: the time-varying DFt and the spectrogram. Applications to the analysis of speech signals.

THE DISCRETE COSINE TANSFORM (DCT)

Definitions and inverse Transforms: the DCT-1 and DCT-2. Energy compaction property. Applications to image compression.

SPEECH SIGNAL MODELS

Review of acoustic phonetics and the acoustic theory of speech production; digital models for speech signals. Analysis and synthesis of speech signals. Pitch freeuqncy and formant frequencies. The VOCODER model and its variations.

LINEAR PREDICTIVE CODING (LPC)

AR and MA filters; predictive filter with impulsive or noise input. Calculation of the Error signal power and its minimization. Yule-Walker equations. Interpretation in the frequency domain: predictive filter as a spectral estimator. Applications to speech signals.

Modalità d'esame

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

Propedeuticità

For this course, it is assumed that the student has already taken a first course in Digital Signal Processing ("Elaborazione Numericadei Segnali A", at Parma University).

Testi consigliati

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