EXPERIMENT DM1

Adaptive signal processing & adaptive systems

Academic in charge: Prof Danilo Mandic

(Room 813, ext. 46271)

Equipment:

Any computing Facility. MATLAB and SIMULINK.

Aims & Outline:

Adaptive digital signal processing is the study of algorithms and techniques which have the capacity to vary in sympathy with changing statistical properties, characteristic of many real signals. Such techniques have been successfully applied in many application areas, for example, channel equalisation in communications, beam forming for seismic prospecting, ECG monitoring in medicine, analysis of multiphase flow and the control of dynamic systems.

The purpose of this experiment is to study algorithms for adaptive signal processing with particular emphasis given on the Least Mean Square (LMS) and Recursive Least Square (RLS) adaptive algorithms. The performance of the above algorithms will be investigated when they are used for echo cancellation, system identification and channel equalisation.

For instance, in "hands-free" mobile telecommunications, the mobile unit which contains the loudspeaker and microphone is placed, for convenience, at some distance from the local speaker. Therefore, when the remote (far-end) speaker is talking, there is acoustic coupling between the loudspeaker and microphone of the mobile unit, which leads to the far-end speaker hearing a disturbing echo of his own voice. Elimination of this echo can be achieved with an adaptive filter, which models the time-varying acoustic coupling.

References:

[1] Chambers J., "Experimental Handout on Adaptive Signal Processing", October 1994.
[2] Haykin, S., "Adaptive Filter Theory", Second Edition, Prentice-Hall, 1992,Much improved on first edition, good sections on numerical issues in adaptive filtering and emerging adaptive techniques.
[3] Widrow, B., and Stearns S.D., "Adaptive Signal Processing", Prentice-Hall, 1985.
Widrow was in the vanguard of adaptive filtering, introductory but strong on applications.
[4] Bellanger, M., "Adaptive Digital Filters and Signal Analysis", Marcel Dekker, 1987.
A first-class book, he has really worked with adaptive filters.