Title	**BC04M.1: Estimating network QoS parameters and simulating video conferencing links.**
Description	This project is concerned with interactive real-time audio/video conferencing as may be applied to distance learning over the Internet. The first phase of the project is an investigation of the "quality of service" QoS parameters which quantify the performance of a real Internet link for conferencing, and the accurate measurement of these parameters. A means of measuring QoS parameters such as round-trip delay, jitter and packet loss probability should be devised and used to obtain measurements for a number of different Internet connections, geographical locations, times of day and packet sizes. The second phase of the project will design and implement a simulator that emulates different QoS behaviour based on the measurements obtained during the first phase of the project. This will allow distance learning systems which involve interactive real time communication by Internet to be tested "in house" over typical Internet links before the systems are released to world-wide participants of distance learning courses.
Main Objective	As specified above
Additional Objectives	As specified
Area of Project	Computer networks.
Type of Project	Software development.
Special Expertise Needed	A high-level programming language and an interest in networks.
Equipment Needed	Access to a PC with network access & sound input and output . .
Location	CS labs
Workshop Facilities	None needed.
Technician Involvement	General advice only
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	Involvement in a Ph D project.

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Title	**BC04M.2 Anonymous speech transcription to text**
Description	Anonymous speech files for transcription to text: In the medical and other fields, audio recordings of speech are sent to agents, often abroad, for conversion to text. This clearly creates problems with confidentiality and security of the information. The aim of this project will be to process the audio files firstly to disguise the voice and secondly to identify and remove words which may still allow the persons involved to be identified. The removed words would be stored in a separate encrypted file. Other techniques may be applied to increase the level of confidentiality; e.g. scrambling the order of phrases and sentences according to a key, and identifying and removing identifiable mannerisms.
Main Objective	As specified
Additional Objectives	As specified
Area of Project	Speech technology and Signal Processing.
Type of Project	Software development and interfacing standard peripherals
Special Expertise Needed	A high-level programming language and an interest in speech and DSP.
Equipment Needed	Access to a PC with CDROM, sound input and output facilities microphones and headphones. Compiler, internet access and midi software.
Location	CS labs
Workshop Facilities	None needed.
Technician Involvement	General advice only
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	A possible marketable product.

Title	BC04M.3 : Investigating voice over wireless LAN protocols
Description	The aim is to simulate a cordless voice telephone network which uses a wireless LAN (IEEE802.11) scheme rather than a conventional GSM cellular system. This allows a private telephone network (PABX) as may exist in a small commercial company to operate alongside and interact with data terminals such as computers and PDAs. Various 'voice-over-IP' schemes have been proposed in the research literature and these should be evaluated to discover roughly how many speech users may be accommodated on a single wireless LAN. It should then be possible to propose more efficient 'voice-over wireless LAN' (VoWLAN) schemes without the normal overheads of IP to accommodate yet more speech users with better speech quality. Many voice over wireless LAN techniques propose to use the point co-ordination function (PCF) of IEEE802.11 though Liu and Wu propose a novel distributed co-ordination function (DCF) mode approach which uses the IEEE "beacons" and power saving facility (MAC approach) to achieve a "pseudo-time-division-multiplexing" technique for each speech channel. Many other researchers conclude that PCF would poorly support voice and are now looking at the problem of integrating voice with 'contention mode' wireless networks and exploring the new MAC sub-layer of the proposed IEEE802.11e standard. Popular MAC sub-layer approaches include “distributed fair scheduling” (DFS) and “blackburst”. For a survey and references see: www.cs.man.ac.uk/~barry/mydocs/glasgow.pdf NB It is beyond the scope of this project to implement PCF approaches, "5-up" or the new scheme proposed in the above paper. It will be restricted to VoIP and the use of standard IEEE802.11 protocols in the normal 'contention mode'.
Main Objective	As specified
Additional Objectives	As specified
Area of Project	Wireless computer networks, QoS and 'voice over WLAN'.
Type of Project	Software development and interfacing standard peripherals
Special Expertise Needed	A high-level programming language.
Equipment Needed	Access to a PC with sound input and output facilities microphones and headphones. IEEE802.11 wireless LAN cards.
Location	CS labs
Workshop Facilities	None needed.
Technician Involvement	General advice only
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	Link to a current European research grant.