Title	**BC06MSc.1: Estimating network QoS parameters & 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.
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 .
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	Involvement in a Ph D project.

Dr. BMG Cheetham Room: IT403 Tel: 4534 Email: barry@man.ac.uk

Title	**BC06MSc.2 Damaged packet delivery and PLC for VoWLAN**
Description	This project concerns the use of a wireless LAN to provide interactive voice communications, i.e. 'voice over WLAN' (VoWLAN), as well as data. 'Voice over IP' (VoIP) is now widely used for telephony over wired networks and is being increasingly used over wireless networks despite the fact that wired and wireless computer networks were designed primarily for data. Measures which aim to provide or guarantee a certain 'quality of service' (QoS) to telephone traffic are often introduced into 'medium access control' (MAC) mechanisms. The presence of bit-errors in a received packet may be detected by a 'cyclic redundancy check' (CRC) and, in wireless networks, forward error control (FEC) is used to allow the receiver to correct a small number of bit-errors by means of a 'Viterbi' FEC decoder. Despite these measures, voice packets may be irreparably damaged, lost in transmission or excessively delayed so that they are too late to be useful. 'Packet loss concealment' (PLC) is used to replace lost or excessively delayed voice packets by artificially created voice packets in such a way that the distortion, as perceived by a listener, will be as small as possible. The QoS provided by wireless networks will normally be different from that provided by wired networks in that irreparable damage will be more common. Excessive delay will be less common with wireless. With interactive telephony, it is normally not possible to use transport layer acknowledgements and retransmissions when errors occur. Currently, with VoIP and VoWLAN, irreparably damaged packets are discarded and replaced by PLC created packets, even when the number of uncorrected bit-errors is small. The aim of this project is to investigate how packets that are irreparably damaged by a wireless LAN can be made use of if they are not discarded. Packets are considered irreparable when the FEC Viterbi decoder fails to correct all the bit-errors that occurred during transmission. The following questions may be asked: (1) What is the effect of the Viterbi decoder on irreparable packets? (2) If uncorrected bit-errors are simply ignored, under what circumstances is the distortion in the voice sound worse than that created by throwing the whole packet away and using PLC to produce a substitution? (3) Since WLAN cards can perform automatic retransmissions at the MAC layer and there may be several irreparably damaged versions of the same packet available at the receiver, how can these versions be used to produce a packet with fewer or no bit-errors ? (4) If irreparably damaged packets are to be made use of rather than simply discarded, is

	the 'convolutional coding' and Viterbi decoding mechanism used currently by wireless LANs the best option for voice? The investigation may be carried out by means of software simulations using MATLAB or another language.
Area of Project	Mobile computing and signal processing.
Type of Project	Voice over wireless LAN (Mobile Computing)
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.
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	Related to European project : WINDECT

Title	**BC06MSc.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'.
Area of Project	Mobile computing 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.
Safety Issues	Instructions on use of headphones to be read and understood.
Industrial Involvement	Link to a current European research project (WINDECT).