

MSc projects 2009

Barry Cheetham

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BC09MSc1: 'Just intonation' midi synthesiser

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This project requires an understanding of musical notation, some aspects of harmony and the willingness to learn to program in MATLAB.

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Description of Project

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The 'equal tempered scale' is a tuning scheme for keyboard musical instruments, such as pianos, which allows them to be played in any key. It is used by the 'musical instrument digital interface' (midi) standard, though midi has been extended to accommodate different tuning schemes. The 'equal temperament' scale gives only an approximation to the harmonic relationships underlying Western classical music. For example the 'major fifth' from C to G or D to A occurs, in principle, because the frequency of the upper note is $3/2$ times that of the lower note. With 'equal temperament' tuning the ratio becomes '2 to the power $7/12$ ' which is 1.49831. The difference is small, but significant, and greater discrepancies occur for other intervals. The harmonious 'common chord' (e.g. C E G) basic to much classical music is significantly affected by equal temperament tuning as may be demonstrated easily. For example, see the reference: <http://www.phy.mtu.edu/~suits/scales.html>

A 'just tuned' musical instrument would produce ideal intervals, with the associated pleasant sounding coincidence of harmonics, for all intervals. Starting on the note C, a 'fifth' (C-G), 'fourth' (C-F) and 'major third' (C-E) interval would produce notes of frequency $3/2$, $4/3$, and $5/4$ times the fundamental frequency of C. Notice the ratio of small integers. Two other examples are a 'minor third' (C-E flat): $6/5$ and a minor sixth: $8/5$. A keyboard could easily be tuned to produce these intervals exactly for intervals starting on C, and it could be played in 'C major' and sound very nice. Unfortunately, intervals starting on notes other than C would then not always be correct, and some may be very far from correct. Playing a keyboard 'just tuned' for 'C major' in some other key such as F sharp major would produce poor harmonic sound.

The problem with 'just tuning' was realised by composers even earlier than J.S. Bach (1685 to 1750) and various tuning methods pre-dating 'equal temperament' were used. Bach wrote two sets of 24 keyboard pieces referred to as 'The well tempered Clavier'. Each set has one study in each of the 24 possible major and minor keys. It is

sometimes assumed that Bach had discovered the concept of 'equal temperament' but there is some debate about whether 'well tempered' meant 'equal temperament'. It is possible that Bach retuned his instrument from piece to piece during performance.

The purpose of this project is to modify an existing midi synthesiser written in MATLAB such that the effect of tuning systems other than 'equal temperament' can be heard. The difference between 'equal temperament' and 'just tuning' may thus be demonstrated for music in a particular key. The tuning can then be made to change according to which key a performer is playing in. Ultimately, the challenge is to make the synthesiser change the tuning automatically according to its analysis of the music. Hence the synthesiser would produce perfect 'just intonation' intervals regardless of what key the music is played in. The project requires an understanding of musical notation, some aspects of harmony and the willingness to learn to program in MATLAB.

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BC09MSc.2: Damaged packet delivery and PLC for VoWLAN

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This project requires an understanding of wireless LANs as presented in the module 'Mobile Computing' (COMP60242), and the willingness to learn to program in MATLAB.

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Description of Project

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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 as with TCP. 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 DLL 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.

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BC09MSc.3 : Viterbi decoder for forward error correction in mobile networks.

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Description of Project

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There are two possible projects concerned with correcting bit-errors in mobile LAN equipment. Both involve a study of the Viterbi decoding algorithm for convolutional coding.

The aim of the first project is to investigate a power saving strategy whereby the decoder is de-activated when bit-errors are not occurring. A new way of doing this has been devised but remains to be fully investigated.

The aim of the second project (probably more challenging) is to study 'forward error correction' (FEC) as used in IEEE802.11 wireless LAN technology and elsewhere. The IEEE802.11 standard employs 'convolutional coding', a cyclic redundancy check and retransmissions to accommodate bit-errors that will usually occur on wireless links. The convolutional decoding is achieved by a 'soft decision input' Viterbi decoder, and any packets that fail the cyclic redundancy check after the best efforts of error correction are normally discarded. The aim of this project is to understand the principles of a 'Viterbi' convolutional decoder which has 'soft decision' outputs as well as soft decision inputs, and to apply this to improve the FEC performance of a wireless LAN. A well known algorithm is known as 'SOVA', but it is intended that we develop our own version, perhaps in MATLAB. This project is related to BC07MSc2, but will concentrate more on the algorithm than the VoIP application.

Further detail: The contents of WLAN packets are convolutionally coded to allow a degree of bit-error correction if errors occur. The convolutional decoding at the receiver is efficiently done by a Viterbi decoder which these days is usually a 'soft decision' input Viterbi decoder. 'Soft decision' input means that the receiver will assign a level of confidence to each bit it receives: '111' could represent 'definitely 1'; '000' could represent definitely '0', with '110', '101' and '100' representing 'probably 1', 'perhaps 1' and 'guess 1' respectively. Similarly for '001' etc. A receiver expecting say 1 volt or 0 volt pulses would make these soft decisions according to how close the noise affected received voltage is to 1 volt or zero volts. A conventional Viterbi convolutional decoder will output a 'hard decision' bit stream

which is hopefully correct. The bit-stream will likely include a 'cyclic redundancy check' (CRC) allowing the receiver to have a good idea that the data is wrong when it is wrong. A conventional WLAN receiver will just discard Viterbi decoded data when it fails its CRC check. If we are to consider making some use of the data in packets which fail the CRC, i.e. damaged packets, rather than just discarding them the use of a Viterbi decoder whose output as well as the input consists of 'soft decision' bits, may be beneficial. The aim of this project is to study convolutional coding, soft decision decoding and the many available decoding algorithms. Ultimately the requirement is a demonstration of a working soft decision output Viterbi decoder and an assessment of its effectiveness for the purpose describe above. This project is strongly related to a previous one but is focussed specifically on the soft output decoder.

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BC09MSc4: Effect of bit-errors on compressed speech and music.

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Description of Project

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Speech, music and multi-media are generally compressed for transmission over wired and wireless networks to reduce the required bit-rate. The effect of bit-errors tends to become more serious the more the compression. The aim of this project is to investigate the effect of bit errors as may occur in Mobile Communication over a WLAN, on the quality of the sound when a number of different compression techniques are used. The G711 (64kb/s), G726 (32kb/s) and G729 (8 kb/s) standard compression techniques for 'narrow band' speech should be investigated along with MP3 for music. The project could be extended to pictures. The aim is to produce an application for demonstrating the effect of bit errors and the effect of commonly used forward error correcting (FEC) coding. The use of software to automatically evaluate 'mean opinion scores' (MOS) for narrowband speech will be part of this project. The MOS software and the speech and music coding techniques will be available, but something of their underlying principles must be understood.

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BC09MSc5: An application for teaching Westerners to recognize and reproduce tones in Chinese

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This project is for a Computer Scientist with knowledge of Chinese, preferably Mandarin, and an interest in developing interactive educational software which includes sound analysis and reproduction. This will require a study of some basic principles of DSP (digital signal processing). Studying the mechanism of human speech and its perception will be required, and learning the high level programming language 'MATLAB' will probably be an advantage.

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Description of Project

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For many Westerners studying Chinese, the recognition and reproduction of the four basic tones in Mandarin, and the many more in Cantonese, are a source of great difficulty. Some 'wai-guo-ren' simply cannot discern the differences as they are so attuned to other aspects of intonation. Reproducing the tones can be an even more difficult problem. Imagine the frustration of a poor old Englishman who has learnt his vocabulary only to find that no Chinese person has a clue what he is saying because the tones are all wrong.

This project is for a Computer Scientist with knowledge of Chinese, preferably Mandarin, and an interest in developing interactive educational software which includes sound analysis and reproduction. An understanding of the basic mechanisms of human speech and its perception and some principles of Digital Signal Processing will be needed. The use of MATLAB is recommended for this project.

The main aim of the project is to develop an application that demonstrates the use of Mandarin tones to a Westerner, analyses his/her attempts to reproduce Chinese words and sentences correctly, and provides feedback and suggestions for corrective action. Specific objectives are as follows:

- (i) Consult professional publications on the teaching of Chinese to foreigners and, if possible, to contact language teaching staff in the University or elsewhere to gain from their experience. The teaching methods adopted by Chinese teachers of foreigners and the computer software packages currently available should be considered and references on automatic speech recognition as applied to tonal languages should be researched as background.
- (ii) Study the physical mechanisms responsible for voice production.

- (iii) Develop tools for segmenting recorded sections of spoken Mandarin to isolate single words. Then devise the means of measuring the fundamental frequency of the voice at suitable intervals to determine how the pitch of the voice varies within each word.
- (iv) Given a 'pin-yin' representation of the segment of spoken Chinese, the application should then be enabled to impose the standard symbols over the vowels to indicate the tones (right or wrong) as detected by the application. These can be compared with the correct tones and imperfections in either the software or the speaker can be identified.
- (v) Analyse the tones of a clear native speaker of Chinese to verify the software before moving on to a trainee.
- (vi) Build the tools into an application which produces written Chinese (probably in pin-yin), invites the user to speak the words or phrase, and then analyses the result. The correct pronunciation could then be given as sound output.
- (vii) Introduce the option of producing Chinese words and inviting the user to identify the tones via a keyboard input.
- (viii) Develop a suitable 'graphical user interface' with convenient feedback from the analysis.
- (ix) Subjectively test the software on a user and, by seeking the advice of a professional teacher, give an assessment of its potential for further development.

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BC09MSc.6: Estimating network QoS parameters & simulating audio/video conferencing links.

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This project requires an understanding of wireless LANs as presented in the module 'Mobile Computing' (COMP60242).

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Description of Project

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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.

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DO NOT USE THIS ONE IN 2009:-

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BC09MSc.7 : Investigating voice over wireless LAN protocols

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Description of Project

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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".

NB This project will probably be restricted to VoIP and the use of standard IEEE802.11 protocols in the normal 'contention mode'.

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