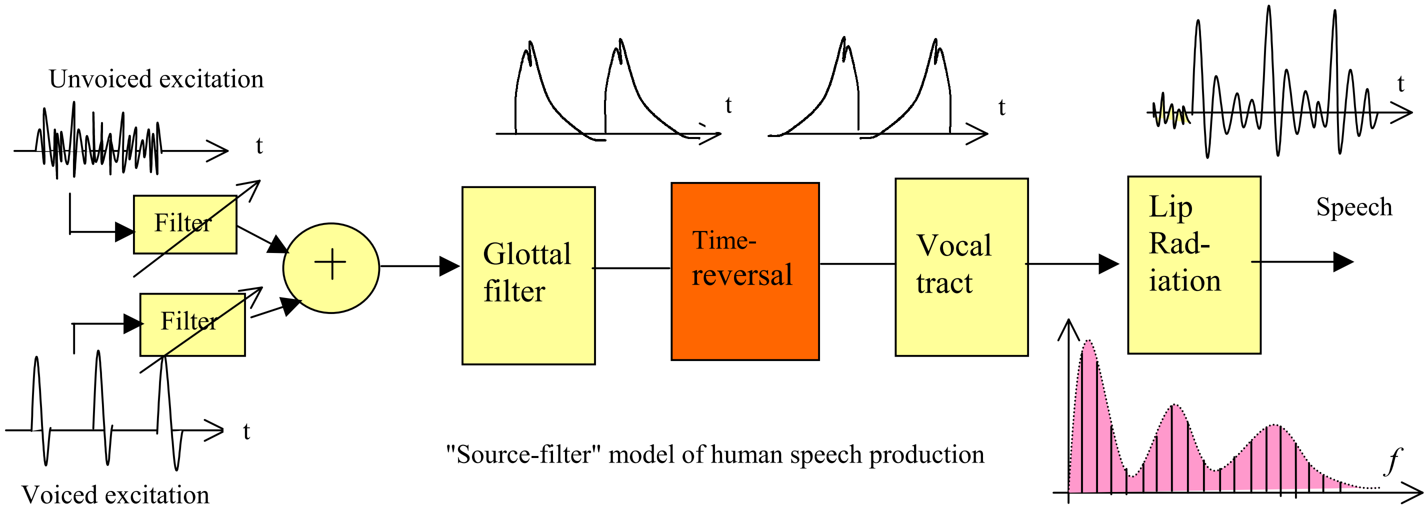
Digital Signal Processing (DSP) for Mobile Communications

Speech coding for mobile & VoIP

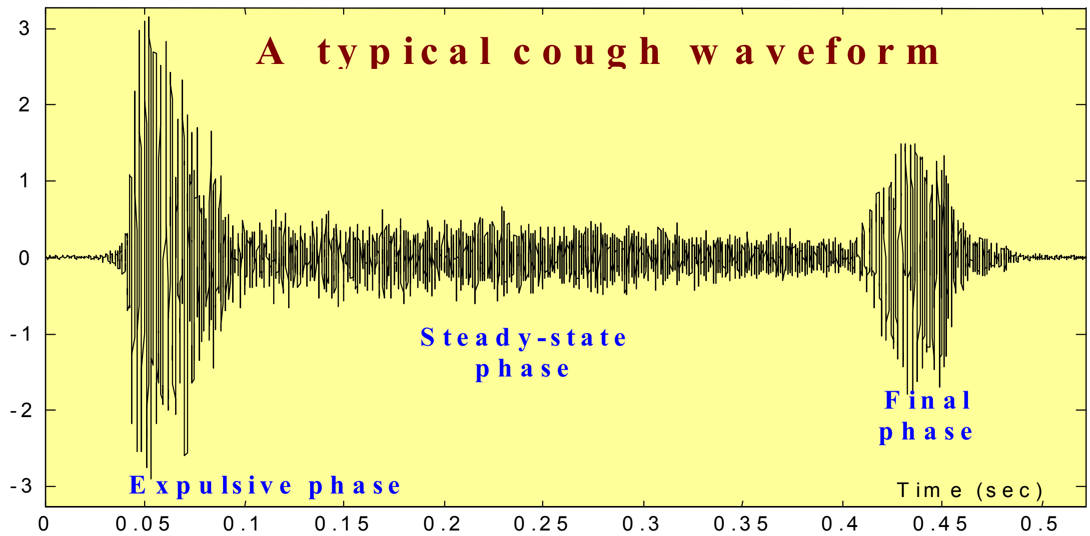
To reduce the bit-rate required for transmitting telephone quality speech, a new approach to speech compression is needed. Current mobile phones compress to about 13,000 b/s (bit per second) and we aim to achieve 4,000 b/s and lower using a method called 'DAP-STC'. Compression is generally achieved by spectrally analysing speech segments to derive a model of the human speech process. DAP-STC is a new and potentially more accurate way of doing this with a modified speech production model. A variable bit-rate version achieves an average 2,400 b/s and is suitable for packetised speech as may be transmitted over computer networks.



"Source-filter" model of human speech production

Patient Monitoring

DSP is being applied to the monitoring of patients with chronic cough in a collaborative project with Wythenshawe Hospital in Manchester and Fazakerley Hospital in Liverpool . This work emanates from a clinical study of asthmatic, cystic fibrosis and cryptogenic fibrosing alveolitis patients. The requirement is to obtain statistical measurements from long term sound recordings to allow objective assessment of the severity of cough. These measurements are being used for comparing the effectiveness of various treatments and to study the physiological characteristic of pulmonary disease.

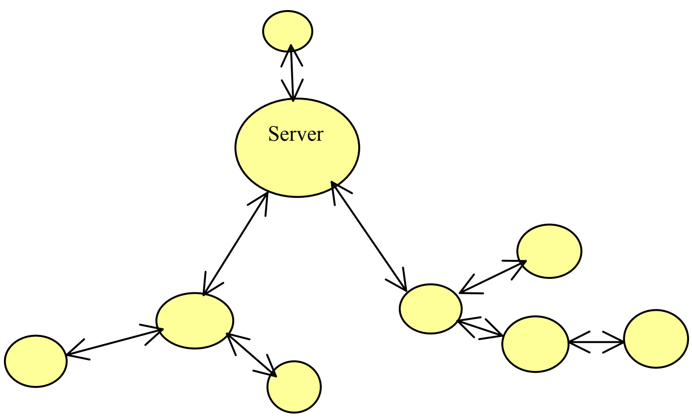


Low power implementation of DSP

The requirement for extended battery life, reduced size and low electromagnetic interference for mobile communication equipment has led to the development of a novel asynchronous DSP device known as CADRE. Compiler support for the new device is needed to avoid manual assembly-level programming. Apart from maximising speed and minimising code size, the compiler is required to produce code which exploits the special features of CADRE to minimise energy consumption. This optimisation at software level builds on the economies achieved at hardware design level and aims to minimise the switching activity when the DSP code is implemented.

Real time multimedia facilities for distance learning over IP networks

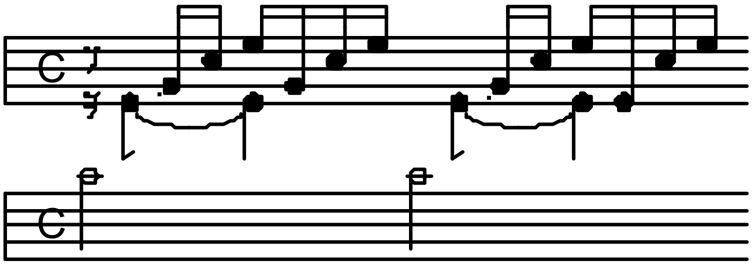
Distance learning can be a major application of fixed and mobile computer networks and the Internet. We are concerned with the introduction of real time multimedia facilities into this application. Delay and network congestion have a more serious effects on real-time traffic than on text and there are interesting ways of adapting face-to face communications, as occur in typical distance learning scenarios, to the available connections and their bandwidths. An adaptive trade-off between speech, data and images is possible and 'ad-hoc' point-to-point routing between participants can reduce the effects of network congestion.

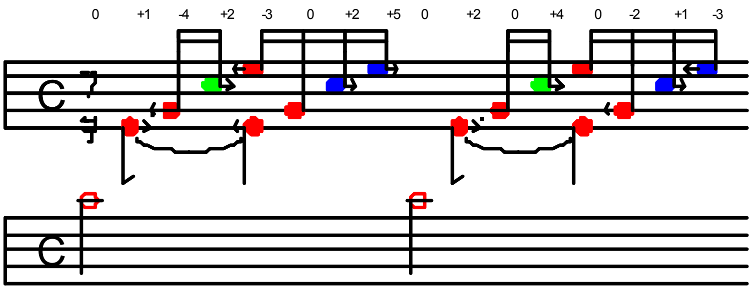


On-line tutorial with point-to-point routing

Extraction of expressive performance parameters from acoustic recordings of music

An effective musical performance involves very fine and subtle variations of timing, dynamics, timbre, articulation, vibrato and other characteristics of the sound. A conventional musical score gives only general directions with respect to these characteristics. Our aim is to be able to automatically measure the expressive characteristics of a live or recorded musical performance and to develop notation, using colour and other markings, for augmenting the score with these measurements in a clear and convenient way. This has many applications, for example in comparing different interpretations of the same piece, providing feedback during rehearsal and diagnosing problems with technique.





Voice over multi-carrier wireless LANs

This work addresses the problem of efficiently integrating wireless telephony and wireless computer networks using a IEEE802.11 standardised 'multi-carrier' physical layer. This allows the sharing of resources, common access to telephone and data services and the interaction between normally separate systems. Traditional "voice over IP" approaches are inefficient in terms of system overheads, and more recent proposals, such as "5-UP" are not compatible with 'ad-hoc' networks. Our technique transmits packetised voice on the multiple carriers individually, one per voice channel, and requires the wireless telephones to conform with standard IEEE802.11 collision avoidance protocols.