Simulation of Access for 3G IP WCDMA to fixed packet switched networks using OPNET^{TM}

A Heath and R A Carrasco

School of Engineering and Advanced Technology, Staffordshire University
PO BOX 333, Beaconside, Stafford ST18 0DF, Staffordshire, U.K.
Emails: A.L.Heath@staffs.ac.uk, R.A.Carrasco@staffs.ac.uk

An efficient code holding technique for a Wide-band Code Division Multiple Access (WCDMA) Internet Protocol (IP) access network is proposed, which reduces the number of codes required and allows BandWidth to be supplied On Demand (BWOD). Performance is evaluated in terms of end-to-end delays for different multimedia applications.

***Introduction*:** In Third Generation (3G) systems it is anticipated that wireless, wired and satellite communications will all be integrated as one system, in which multimedia data will be transferred[1]. In IMT-2000, or Universal Mobile Telecommunications System (UMTS) mobile users will access the fixed networks for different traffic types, WCDMA is the most promising proposed air interface[2]. Within UMTS there are separate circuit and packet switched channels; in this paper an entirely packet switched system is proposed, in which all services are treated in the same way. This paper addresses the needs of different data types in terms of end-to-end delay and the amount of traffic transmitted, which have been modelled and simulated in OPNET^{TM} simulation environment. Similar issues have previously been studied in an ATM access network using a service adaptable virtual connection tree algorithm in [3, 4].

There are 512 downlink-scrambling codes [2], so this paper proposes a way of allowing many users to share those codes.

System Model and Simulation: The simulation model is constructed using two user-defined nodes, the Base Station (BS) and the Mobile Station (MS). The radio link's characteristics between the BS and MS are modified by one or more of the fourteen customised design stages of the radio transceiver.

The state machine for a MS is shown in Fig. 1, the MS is initialised and has an IPv4 address assigned to it, the Wait state is then enterred. The transmission and receive processes are modelled dynamically and can occur simultaneoulsy from the Wait state (parallel processes).

When the MS has an uplink IP packet to send a code request message is sent to the BS on the control channel. The MS then awaits for a response from the BS or a timeout period, whilst queuing uplink packets. If a code is not allocated within the timeout period then another request message is transmitted; in this model this is repeated until a free code is allocated, increasing end-to-end delays of packets, this process could be limited to 'x' attempts before deciding to block the application. When a code is allocated, the MS de-queues, encapsulates and transmits the uplink packets across the air interface in 10ms packets to the BS. When there are no packets left in the queue then the code is held on to for the channel-hold time. If the channel-hold time elapses and no more uplink packets are generated then a code de-allocate message is transmitted to the BS and the code is released and available for use by other users. For the receiving process, the MS waits for a downlink code allocation message on the control channel from the BS. When a downlink code has been allocated packets are received with the allocated scrambling code until a code de-allocation message is received.

Fig. 2 shows the state machine of the BS, there are four events that could occur simultaneously, namely: 1. Receive an uplink code request from a MS, 2. Receive an uplink IP packet, 3. Receive an uplink code de-allocate request or 4. Receive a downlink IP packet for a MS currently in the BS's cell.

When an uplink code request is received from a MS, a free code is searched for, if one is found, it is sent to the MS on the control channel. If there are no free channels then a code not allocated message is sent to the MS. If an uplink IP packet is received from a MS it is de-capsulated and then re-assembled in a buffer before being forwarded to its final destination across the IP. When the BS receives an uplink code de-allocation request, the code is freed and is available for other users.

If a downlink IP packet is received that must be forwarded to a MS located within the cell, the packet is placed in a queue (a separate queue is generated for every MS) and if the MS already has a downlink code then the packet is encapsulated and transmitted to the MS. If there is no code already allocated to the MS then one is searched for, if there are no free codes the packets are queued until one becomes available. When a new code is found, a message is transmitted to the MS on the control channel, and the packets are encapsulated and transmitted to the MS. The BS transmits all the packets in the queue to the MS, until the queue is empty. If no more packets arrive within the channel hold time then the code is de-allocated and a last message is sent to the MS.

The user profile consists of voice, e-mail, File Transfer Protocol (FTP), web browsing and database access. Voice has a Type of Service (ToS) setting as interactive voice(6) whilst the others are set to best effort(0) [RFC 791]. Voice connections employ the user datagram protocol (UDP) whilst the other applications use the transmission control protocol (TCP) on the transport layer [RFC 768 and RFC 793]. The MS and BS transmitters and receivers have a standard isotropic antenna pattern and use QPSK modulation. All data types are transmitted at a data rate of 1,536,000 bits/sec with a transmission power of 0.398W.

In the initial simulation, there is a BS and 75 MSs, which contains 31 laptop computers, 23 handheld terminals and 21 mobile telephones which are arbitrary. The BS acts as a gateway node to the fixed IP enabled network. The same simulation was repeated for 10, 90 and 105 MSs to find the capacity of the network.

Simulation Results: A mobile telephone conducting a packet switched voice application is allocated code 17 for 1.5s and code 1 for 0.5s. This is because the code hold time is set to 0.1s and when a user is speaking, packets are sent continuously, so there are distinct talk and pause periods. Due to the nature of the applications used by the laptop computer and the handheld terminal, the code hold time is set to zero, as one packet is randomly transmitted. Therefore, codes have to be allocated each time a packet is sent, unless packets are held in the queue, this same argument can be used for the downlink codes.

The mean end-to-end delay of packets and the traffic transmitted from 75 MSs for different applications is shown in Fig. 3. From this graph it can be seen that voice applications have the lowest end-to-end packet delays with a mean value of 0.104s (which for voice delays above 0.25s can become noticeable), email 5.7s, FTP 7.52s, HTTP 23.24s and Database 82.8s. This graph also shows that the most of the network traffic consists of voice 4700bytes/s next HTTP, then email then database and lastly FTP. Most of the voice packets have a delay in the order of 25ms which is acceptable but some have higher delays as much as 1s; these packets could be dropped without any detriment to the Quality of Service (QoS). The hold time procedure along with a high value of ToS gives priority to the voice traffic, in that its end-to-end delay is lower than for other applications by a factor of 54 compared to email. These values are sufficient for voice but values approaching 900s is too long to wait for a database query, this problem must be investigated further by using either ATM, IPv6 or MPLS. In the simulated system no packets are lost or dropped, which is why there are such high delays for some applications.

Fig. 4 shows the effects of changing the number of MSs in the system. Mean end-to-end delay and mean traffic sent across the network is plotted for 10, 75, 90 and 105 MSs within one cell. As the number of users increase the amount of database traffic is reduced, but their mean end-to-end delay being highest at 75 mobile users. The end-to-end delay of voice packets is highest when there are 90 MSs but this is also when the most voice traffic is being sent. Voice end-to-end delays remain much lower than the delays for other applications, even when most of the network traffic consists of voice.

Conclusion: A WCDMA air interface has been successfully implemented in the OPNET™ simulation environment. A channel holding time has showed that for voice a code was held for 1.5s; this made the end-to-end delay for voice packets 54 times faster than for email. This method is efficient, as a separate code is not required for every single user, reducing the number of codes required thus increasing capacity. Codes are allocated independently for uplink and downlink channels as some applications are asymmetric, such as web browsing and downloading applications.

References:

1. Godara L C, Ryan M J and Padovan N, 'Third generation mobile communication systems: Overview and modelling considerations' Annales des Telecomms/Annals of Telecomms v 54, n 1, 1999, p 114-136

2. Holma H. and Toskala A., 'WCDMA for UMTS Radio Access for third Generation Mobile Communications', John Wiley and Sons, (August 2000)

3. Larrinaga F and Carrasco R A, 'Virtual Connection Tree Concept Application over CDMA Based Cellular Systems' IEE Colloquium on ATM Traffic in the Personal Mobile Communications Environment, Savoy Place, London (11 February 1997)

4. Heath, A L and Carrasco R A, 'Virtual Connection Tree over Multiple Access Techniques For 3G Wireless Communication Systems' IEE/IEEE Second International Symposium on Communication Systems, Networks And Digital Signal Processing (CSNDSP2000) Bournemouth University (18-20 July 2000),

Fig. 1 *Mobile Station (MS) state machine*

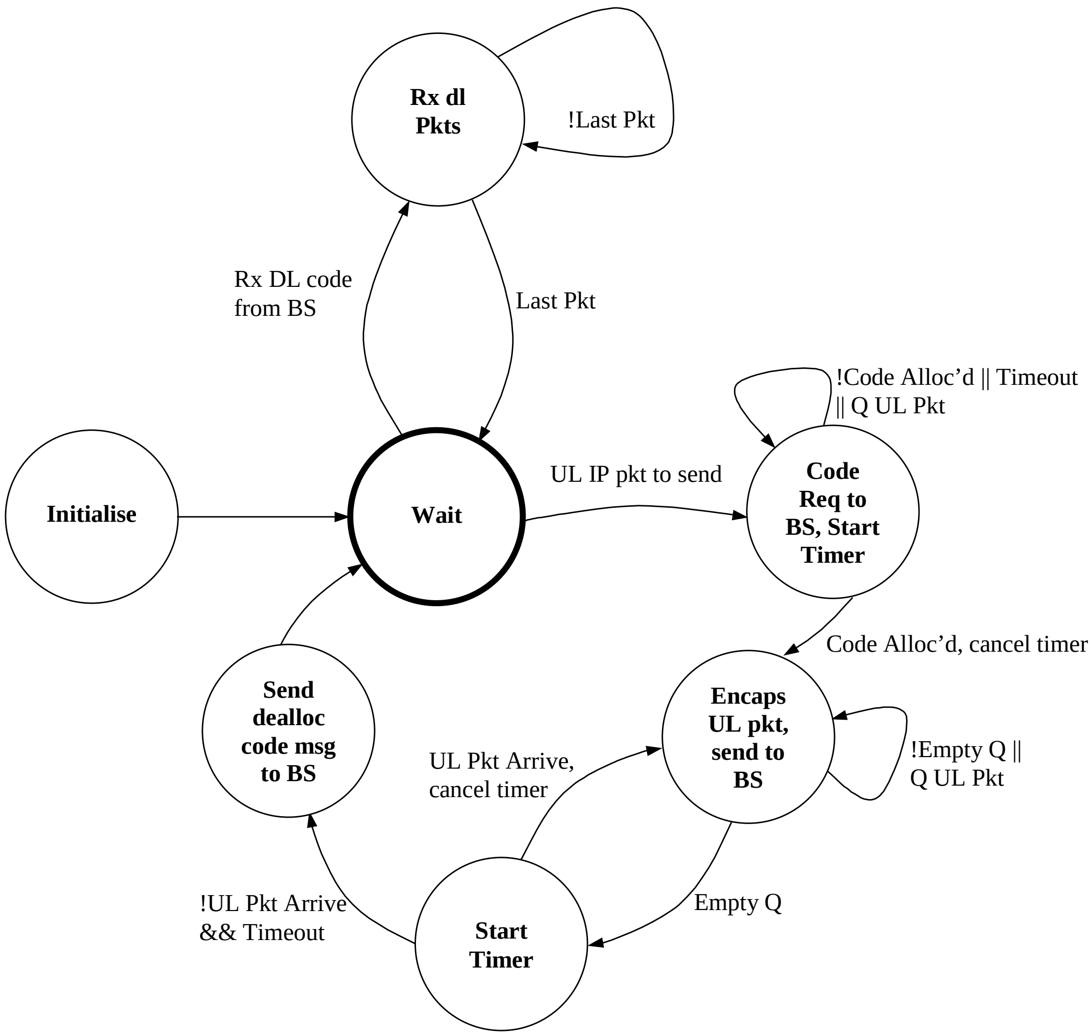

Fig. 2 ***Base Station (BS) state machine***

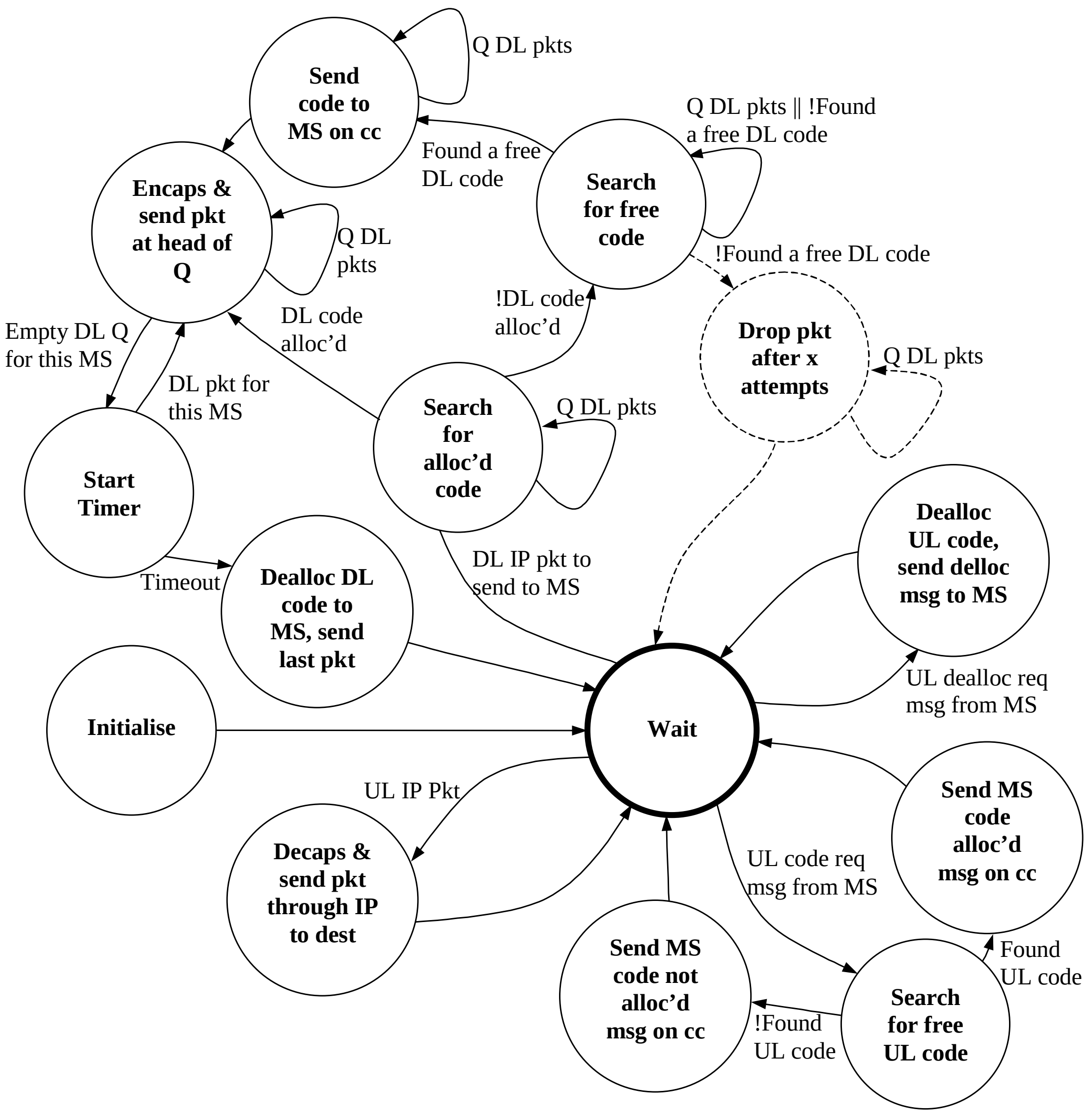

Fig. 3 ***Mean traffic and end-to-end delay, for 75 MSs in a WCDMA IP network***

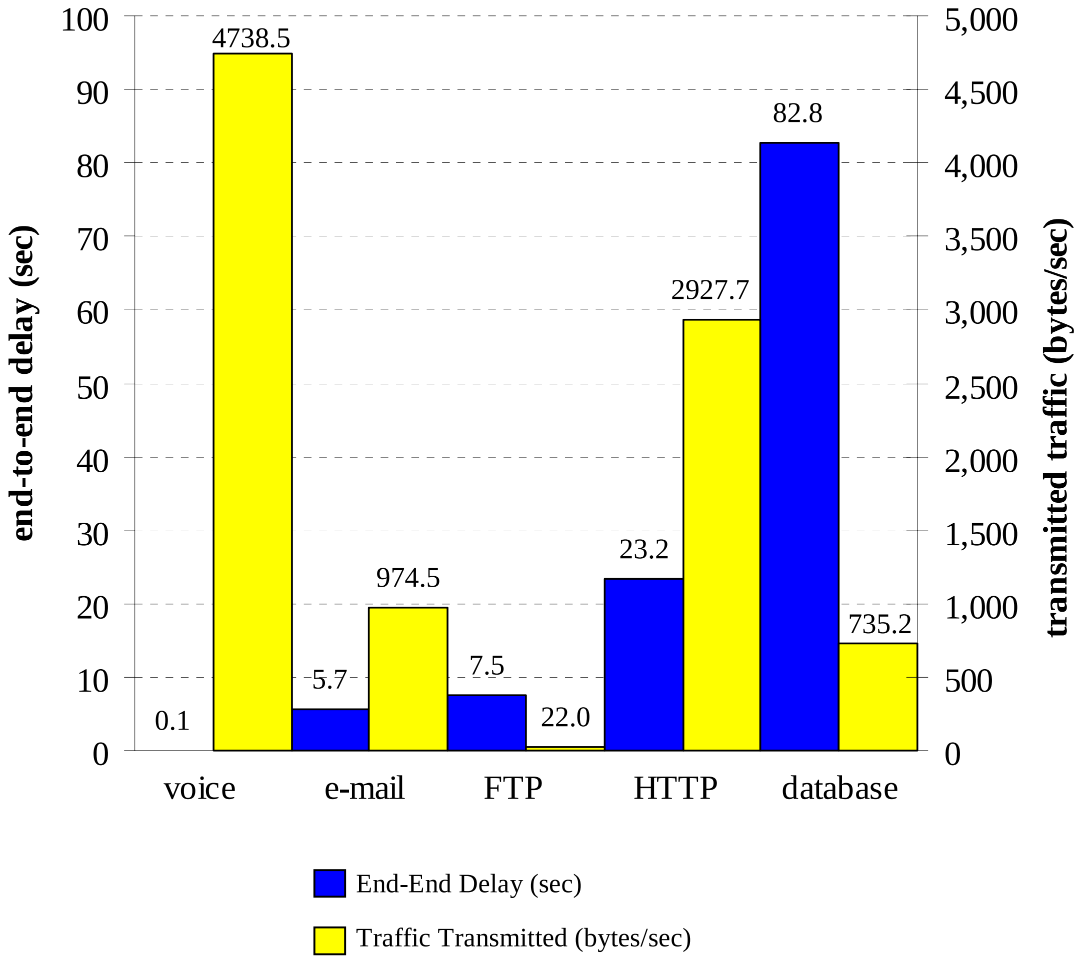

Fig. 4*a) Logarithm of mean packet end-to-end delay,*

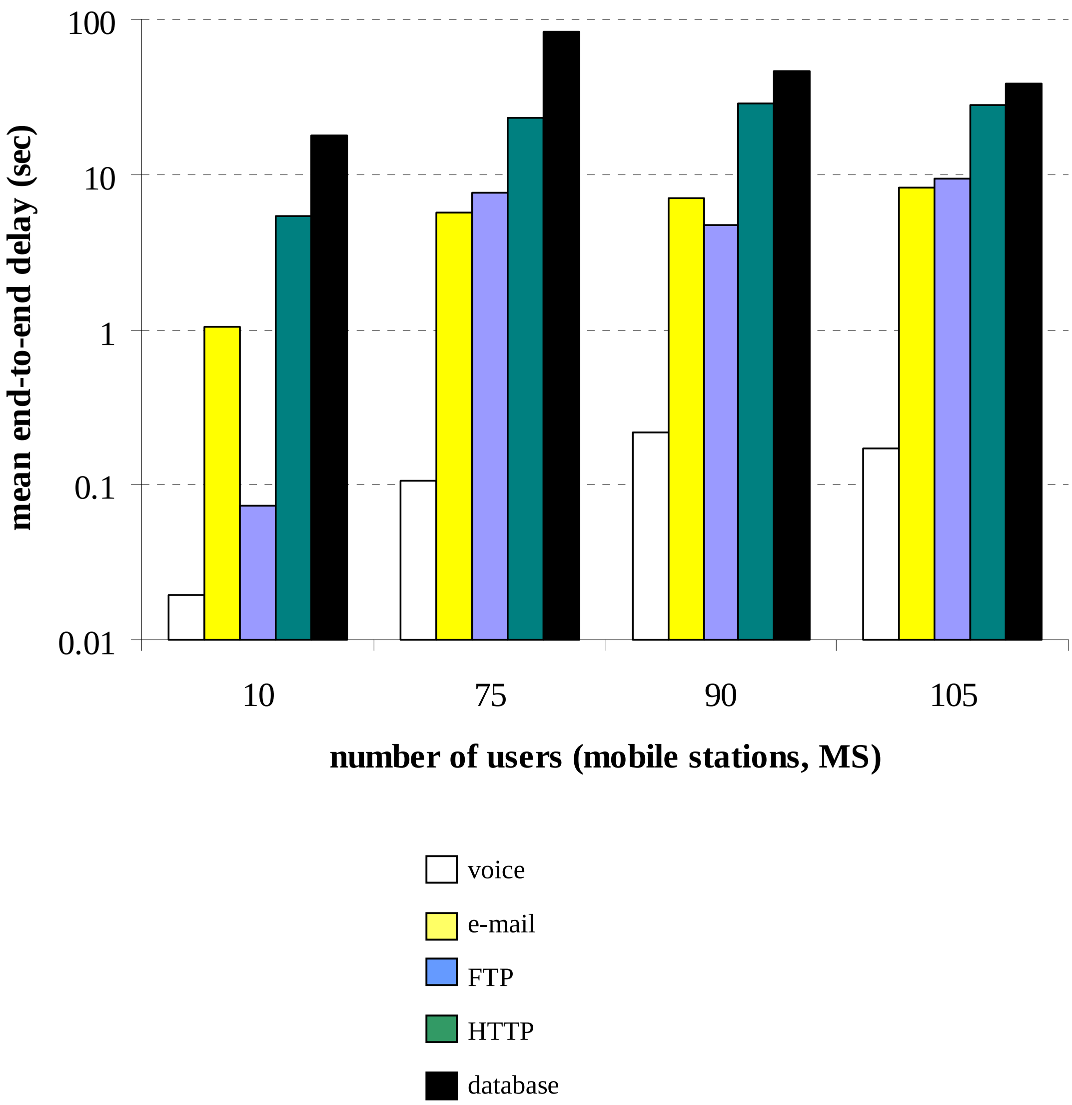

Fig. 4*b) transmitted traffic across a WCDMA IP access network*