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Soft, Vertical Handover of Streamed Multimedia in a 4G Network

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Abstract

In this paper the soft, vertical handover of streamed multimedia in a 4G network is considered. We propose a soft handover solution in which the mobile client controls the handover. This solution requires no modifications to existing wireless networks. The second stream required for the soft handover is duplicated just above the transport layer, rather than requiring the server to play out a second stream that needs to be synchronised with the existing stream. Such a scheme is outlined, and the results are presented that show how the scheme functioned in an emulated environment.

I. INTRODUCTION

Fourth-generation (4G) wireless communication systems [1] will be made up of different radio-networks providing access to an IPv6 [2] based network layer. In densely populated areas, for example, 3G will augment ubiquitous 2.5G networks by providing higher bit-rate access. In hotspots and in corporate campuses, Wireless LANs will complement these systems further by providing even higher bit-rates. Other wireless access networks envisaged for 4G include satellite networks, fixed wireless access (e.g. IEEE 802.16) and PANs (Bluetooth, 802.15, UWB).

Multimedia is expected to be a main application of 4G networks. However, multimedia streams can be sensitive to packet loss, which in turn can result in video artefacts. Such packet loss can often occur when there is an interruption to a connection when a user is moving between networks that are autonomous.

In cellular networks such as GSM, a call is seamlessly handed over from one cell to another using hard handover without loss of voice data. This is managed by network based handover control mechanisms that detect when a user is in a handover zone between cells and redirect the voice data at the appropriate moment to the *mobile node* (MN) via the cell that the MN has just entered.

In 4G networks a handover between different networks is required. A handover between different networks is usually referred to as a *vertical handover*. As 4G networks are comprised of independent networks, there may be no network based handover control mechanism. Therefore, a hard handover may not be possible for 4G as multimedia data may be lost. Instead, soft handover can be used. This ensures that data is not lost by allowing the MN to attach to two networks simultaneously during the handover period, and the same data is sent to the MN via the two networks during this period. The price paid for soft handover is the use of resources in two networks, rather than one, but only during the handover phase.

This paper outlines a scalable soft handover system that enables a MN to control the handover of a multimedia stream in a 4G network. The rest of the paper is laid out as follows. The next section examines related work, while the following section describes the proposed scheme in detail. Section IV presents results from an emulated environment. The final section presents the conclusion.

II. RELATED WORK

Mobile IP [3] has been proposed as a solution for mobility support in IP networks. Mobile IP uses hard handover. Handovers are slow and packets can be lost during the handover procedure [4]. It is therefore unsuitable for the handover of streamed multimedia.

The mobility support provided in the Session Initiation Protocol (SIP) [5] has also been proposed for real-time communication [6,7]. SIP is an application layer protocol for establishing and tearing down multimedia sessions. It can help provide personal mobility, terminal mobility, and session mobility. SIP uses hard handover and “in-flight” packets can be lost during the handover period. In [6] it was estimated that packet loss can happen for up to 50ms during the handover period. So SIP does not provide seamless handover of streamed video in networks.

The scheme proposed below uses soft handover and so is appropriate to the needs of streamed multimedia.

III. PROPOSED SYSTEM

The architecture of the proposed system is shown in Figure 1. The system consists of a multimedia server that is connected to the Internet and a MN that is streaming a multimedia session from the multimedia server while roaming from wireless access network to wireless access network. The server has a handover agent that handles the soft handover. The MN also has a handover agent. The server’s handover agent is located between the transport layer and the normal play out function of the server, while the MN’s handover agent is located between the transport layer and the normal decoding function. The MN has 2 radio interfaces, each with its own IP address.

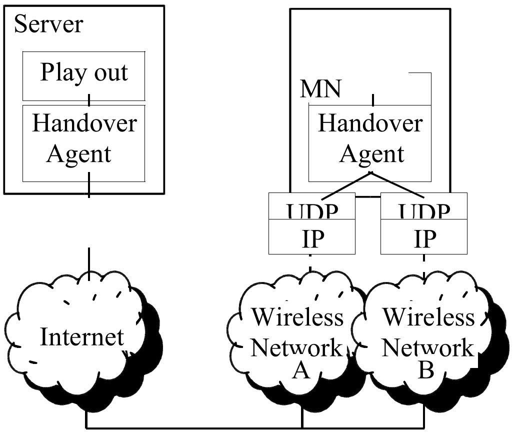

Figure 1. Proposed handover scheme.

Initially the MN is receiving the multimedia content from the server via one wireless network, and the other radio interface is looking for another wireless network. When it discovers another wireless network, the MN sends a START_HANDOVER command to the server's handover agent. On receiving this command, the server's handover agent duplicates each multimedia packet it receives for this multimedia session from the play out part of the server, until it receives the END_HANDOVER command from the MN (Figure 2). The START_HANDOVER command supplies a number of parameters to the server's handover agent: a session id, an IP address and port number. The session id pertains to the current multimedia streaming session in the MN, and is used by the server's handover agent to decide which packets to duplicate. The IP address and port number refer to the IP address and port number that will be used by the MN's radio interface in the wireless network just discovered. The server's handover agent uses these values with the duplicated packets when it inserts them into the UDP transport layer. Therefore the MN receives two streams from the server during the handover period, one through each wireless network, enabling the MN's handover agent perform a soft handover. The MN's handover agent decides which packets to pass on to the decoder, the original packets or the duplicate packets.

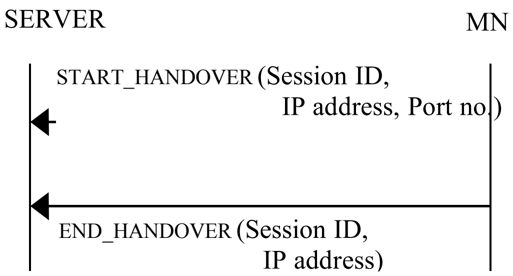

Figure 2. Soft Handover Protocol.

The mobile client controls the handover and no modifications are required to the existing wireless networks that might make up a 4G network. The duplication of the streams in the server's handover agent relieves the server from having to play out a second multimedia session that must be in sync with the current multimedia session being played out. The duplication of the packets adds overhead in the server. However, if it is acceptable to make changes to the network architecture, the server's handover agent can be placed on a proxy, freeing the server fully of the overhead in performing a soft handover.

IV. EMULATION RESULTS

To validate the concept of the proposed system, a prototype system was implemented and tested in the test bed shown in Figure 3.

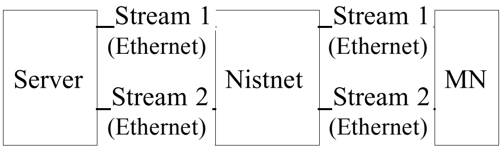

Figure 3. Emulation Test Set-up.

The test bed was based on Nistnet [8], which is a network emulation tool that is frequently used for emulating network conditions at the IP layer. It can emulate network conditions (e.g. packet delay) on a per path basis. The server and MN were connected to the computer containing the Nistnet package using Ethernet links. Two separate paths were used to emulate two wireless access networks, and the MN had two IP addresses.

The server transmitted a stream (Stream 1) of Real Time Protocol (RTP) packets [9] at 100kbps using the User Datagram Protocol (UDP) through the upper of the two paths shown in Figure 3. Nistnet was used to apply delays to the packets it received. These delays were consistent with the delays typically experienced by data packets in a handover zone between cells. After a period of time, the MN sent the START_HANDOVER command to the server (Figure 4). The server's handover agent then turned on Stream 2 through the lower path shown in Figure 3, by duplicating the RTP packets and using the IP address and port number that were received in the START_HANDOVER command. After another period of time, the MN sent the END_HANDOVER command to the server. On receiving this command the server's handover agent then turned off Stream 2 (the duplicate stream) and simultaneously switched Stream 1 to the lower path by applying the IP address and port number that were previously used by Stream 2 to Stream 1's packets.

Figure 4 shows the streams received at the MN, and the points when the MN sent the START_ HANDOVER and END_HANDOVER commands. It shows that the MN's handover agent received Stream 1 and Stream 2 during the handover period, enabling it to perform a soft handover of the RTP stream.

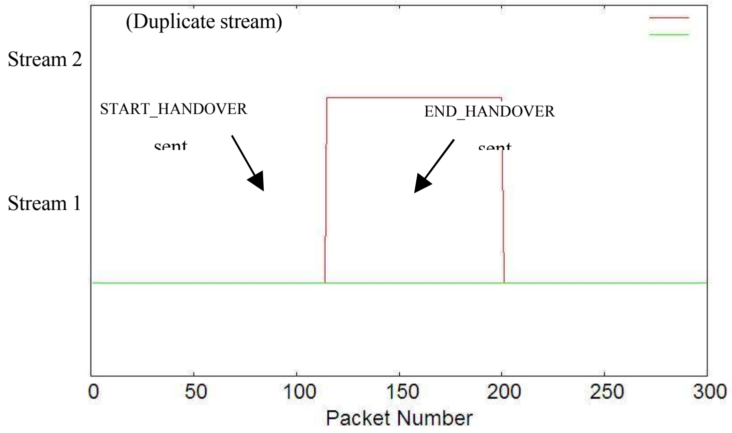

Figure 4. Emulation results.

V. CONCLUSIONS AND FUTURE WORK

In this paper we explained why soft handover is appropriate for the handover of streamed multimedia in 4G networks. We described a scheme to perform the soft handover that enabled the MN to control the handover and required no modifications to the existing stack architectures of wireless networks. Finally, we showed how we demonstrated the scheme using an emulated environment.

Our future work will focus on how the scheme functions in real wireless access networks.

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REFERENCES

[1] Hui & Yeung, "Challenges in the Migration to 4G Mobile Systems", IEEE Comm. Mag., Dec 2003
[2] R. Hinden. "Internet Protocol, Version 6 (IPv6) Specification", Internet-Draft, Oct 1994
[3] C. Perkins, "IP mobility support", RFC (Proposed Standard) 2002, IETF, Oct 1996
[4] Stephane & Aghvami, "Fast Handover Schemes for Future Wireless IP Networks: a Proposal and Analysis" , VTC, May 2001.
[5] M. Handley et al, "SIP: session initiation protocol", RFC (Proposed Standard) 2543, IETF, Mar 1999
[6] Wedlund & Schulzrinne, "Mobility Support using SIP", Proc. ACM WoWMoM'99, USA, Aug 1999
[7] Wedlund & Schulzrinne, "Application Layer Mobility Using SIP", Mobile Computing and Communications Review, Volume 1, Number 2, 2001
[8] Nistnet emulation tool, http://snad.ncsl.nist.gov/itg/nistnet/
[9] Schulzrinne et al, "RTP: A Transport Protocol for Real-Time Applications", IETF RFC 1889, Jan 1996