SOFT, VERTICAL HANDOVER OF STREAMED VIDEO

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Abstract

The delay and jitter of an RTP stream transported by TCP over WLAN are examined as candidates for assisting handover between networks. Jitter was found to be a good indicator of when a mobile is close to the edge of a WLAN cell and was used to spawn a vertical handover mechanism. The implemented handover scheme used an averaged delay difference to complete the handover of the video stream.

1 Introduction

It is expected that the demand for video streaming that is evident in the fixed Internet will be mirrored in the wireless Internet. The high cost of video streaming over GPRS makes it quite likely that viewing relatively long programmes will be more prevalent with WLAN enabled devices. Most of the possible scenarios for users accessing video streaming services suggest that the users will have low mobility during such sessions. Nonetheless, there may be short periods of mobility as the user moves from one WLAN access area to another and so the problem of vertical handover needs to be addressed.

2 Related Work

In order to comment on and relate work to that described here, it is useful to introduce the concepts of *soft handover* and *hard handover*. While these terms have been used elsewhere in the literature in some senses, some specific meaning is required here. In the context of this work, soft handover is a handover in which the same data is delivered to the mobile device via two access networks simultaneously. Clearly, this can be resource intensive, but it does make the probability of data loss during the handover very small. In contrast, a hard handover is one in which data is streamed via one network at any time: thus at some specific time, a decision is made to receive the data through one network rather than the other. While hard handover is more parsimonious with network resources, it can result in data loss at the user terminal, depending on how quickly the handover is effected and how much data the terminal is transmitting/receiving.

The IEEE 802.11b standard for WLANs allows for handover between overlapping WLAN cells at the link layer [6]. Since this only permits connection to one WLAN at a time, it falls into the category of hard handovers. In [1] it was shown that the handover latency can be significant and subject to large variations. During this latency period the station is unable to send or receive traffic and packets queued at the old WLAN will be lost, making it unsuitable for the handover of multimedia traffic.

An approach which has received most interest within the research community is that of Mobile IP [3], which enables a Mobile Node (MN) to receive IP packets through a packet forwarding procedure. This approach is well suited to solving the problem of locating a mobile that may be attached to one of a number of networks. However, handovers in Mobile IP are slow and packets can be lost during the handover procedure [2], making it unsuitable for the handover of video traffic.

Extensions to the mobility support provided in the Session Initiation Protocol (SIP) [10] have also been proposed for delay-sensitive communication [4,5]. 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 [4] it was estimated that packet loss can occur for up to 50ms during the handover period. This estimate did not take into account network initialisation procedures such as AAA or address assignment, and this can be a major part of the overall handover delay. SIP therefore is not good at providing seamless handover of streamed video between networks.

Clearly, the above solutions are inadequate for handover of streamed video, primarily because they break the old connection before they make the connection to the new cell. As loss of video data can result in video artefacts, soft handover is the preferred option and forms the basis of the approach proposed here.

In [9] a modified transport layer protocol used path delay to initiate a soft handover with a voice application. The work presented here considers the use of jitter and delay in the soft handover of video in the application layer. Providing handover functionality in the application layer means changes do not have to be made to operating systems and protocol stacks.

An approach to realizing handover based on a new socket abstraction was proposed in [7]. While this is an interesting contribution it did not address *when* to handover satisfactorily.

3 Jitter and Delay in Handover

In order to evaluate the use of jitter and delay as the basis of a handover mechanism for a roaming MN in an uncongested WLAN, experiments were performed to investigate the relationship between distance from the *Access Point* (AP) and jitter and delay in an uncongested WLAN. Both line-of-sight and non-line-of-sight scenarios were examined.

A video server and MN were connected to an 802.11b WLAN as shown in Fig. 1. The server streamed RTP packets over TCP to the MN via the AP. This RTP stream was used to model CBR video traffic. The packets were transmitted at 30 packets per second and contained 1448 bytes of RTP data each, corresponding to a video bitrate of 347,520 bits/s. The MN was initially beside the AP, and was then moved at a steady walking pace away from the AP until the range of the AP was exceeded.

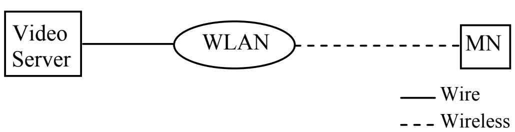

Figure 1. WLAN Setup.

Fig. 2(a) shows a graph of the delay experienced by the RTP packets when the AP and MN were located outdoors with line-of-sight between the AP and MN throughout the test. Fig. 2(b) shows a graph of the corresponding jitter experienced by the RTP packets. The X-axis roughly corresponds to distance between the MN and the AP, where the first packet on the X-axis corresponds to when the MN was beside the AP, and the last packet corresponds to when the MN was about to go outside the range of the AP. Fig. 2(a) shows that the packets experienced no delay until the MN was approaching the edge of the WLAN cell, and that the delay increased exponentially as the MN reached the edge of the cell. Fig. 2(b) shows that the instances when the RTP packets experienced jitter corresponded to when the packets were being delayed.

Fig. 3(a) shows a graph of the delay experienced by the RTP packets when the AP and MN were located within a building and there was no line of sight between the AP and the MN. Fig. 3(b) shows a graph of the corresponding jitter experienced by the RTP packets. Again, the X-axis roughly corresponds to distance between the MN and the AP. Table 1 shows the number of retransmissions in each of the delay spikes in Fig. 3(a) It shows that packets experienced significant delay when the number of packets retransmitted was high.

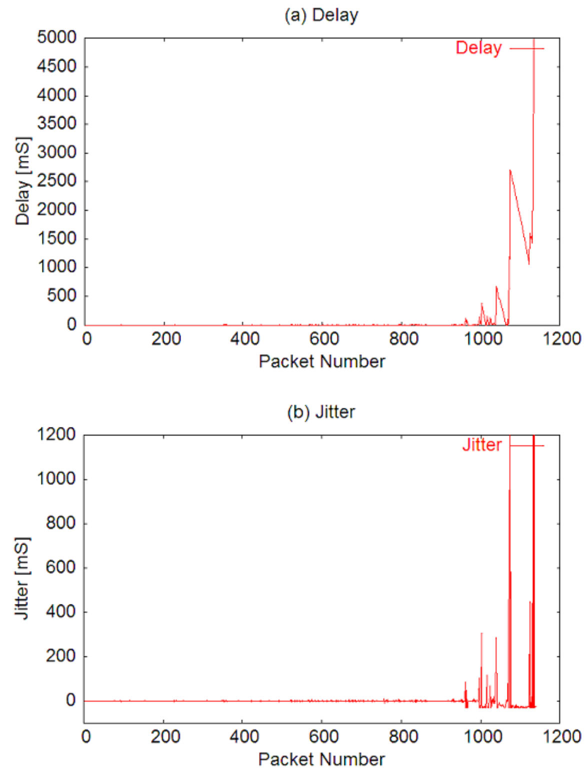

Figure 2. Delay and Jitter in line-of-sight test.

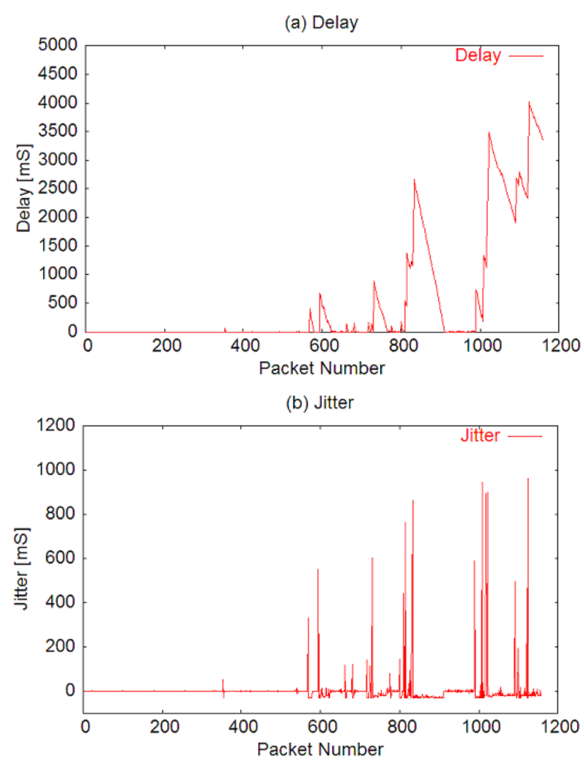

Figure 3. Delay and Jitter in non-line-of-sight test.

Delay Spike	Number of Packets Retransmitted by TCP
Packet Number 568-586	2
Packet Number 594-627	9
Packet Number 729-766	3
Packet Number 810-912	15
Packet Number 990-1159	37

Table 1: Retransmitted Packets in non-line-of-sight test.

In Fig. 2(b) and 3(b), the level of jitter is significant when the RTP packets are experiencing significant delay in the WLAN. This suggests the level of jitter in a stream of RTP packets could be used to initiate a soft handover. (When a soft handover is initiated a second duplicate stream is requested from the video server.) It also suggests that delay can be used to initiate the handover. However it is less straightforward to measure delay in a single stream. Fig. 3(b) shows that there is not a direct linear relationship between jitter and delay.

To avoid handovers being initiated when the delay is relatively small, for example the delay at packet number 600 in Fig. 3, a moving average is used to improve the correlation between the level of the jitter and the delay. For the jitter shown in Fig. 3(b), tests performed showed that a moving average of five elements gave the best correlation. Fig. 4, shows a graph of the resulting moving average.

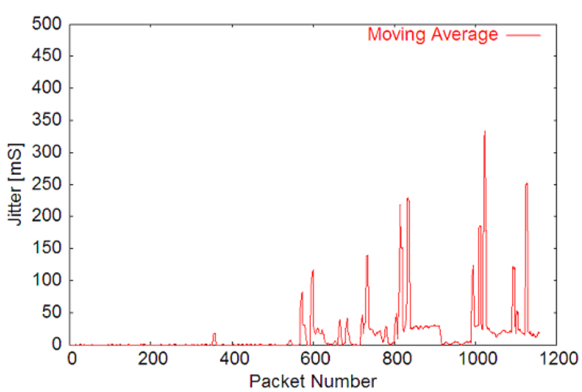

Figure 4. Moving Average of Jitter in non-line-of-sight test.

As a refinement to a scheme that uses jitter to initiate a soft handover, a timer in the MN can be used to check periodically to see how much a delayed packet is delayed by. In this way the moving average can be updated even when delayed packets have not yet arrived. This scheme can enable another stream to be started when the current TCP link is lost, for example in an area of poor signal strength.

During a soft handover, the MN receives two streams and must decide which one to use and then drop the other. It is best that the client chooses the stream that is experiencing the least delay, as delay is high when the MN is close to the edge of the cell and when the MN is in an area in the cell where the received signal strength is poor and a high amount of retransmissions are occurring. Delay is proposed here instead of jitter in deciding which stream to choose, as jitter is less effective at determining the level of delay in a WLAN and also because the MN can easily compare the delays in two WLANs. The MN does this by comparing the arrival time of the received packets in both streams and determining which stream is experiencing the least delay. The difference between the delays being experienced by the two streams is referred to here as *delay difference*. Having decided which stream to use, the MN then drops the other stream.

4 Outline of Handover Scheme

For the purposes of this investigation it is assumed that the video server is in the same metropolitan area as the MN and that the server is not overloaded. In this case the end- to-end delay from the server to the MN via the paths used by the MN can be assumed to be approximately equal under uncongested conditions. Further, it is assumed that for a MN that is streaming near the edge of a WLAN, the delays between the server and the APs is of a lower order than the delays experienced by packets in getting through the WLAN.

Fig. 5 shows content being streamed through two access networks during a soft handover in the above scenario. The MN has two WLAN interfaces, so that it can handover from one WLAN to another. The MN uses soft handover in the application layer, enabling the video decoder in the MN to present an uninterrupted video stream to the user. Microsoft researchers in conjunction with some academics have recently proposed a mechanism by which a single WLAN interface can be used to simultaneously access multiple WLANs [12]. Hence, it is reasonable to assume that future devices will have such functionality.

Each of the two WLAN interfaces in the MN has a unique IP address, so that the two streams are independently routed through the networks. Most of the time the MN has one connection to the server streaming the desired video stream, while the second wireless interface looks for another AP. When it discovers another WLAN, it registers with the network and performs any necessary initialisation procedures. When the jitter in the current WLAN exceeds a certain threshold the MN makes another connection to the server and the server initiates a second, identical video stream to the MN via the new WLAN.

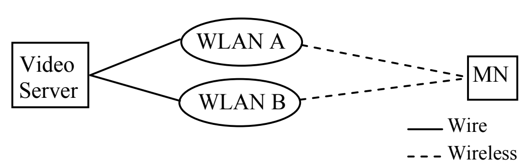

Figure 5. A Soft Handover.

As Fig. 2(a) and 3(a) showed, the packets of a stream passing through an uncongested WLAN where the MN is close to the edge of the cell or where the MN is in an area in the cell where the received signal strength is poor, will arrive at the

MN later than the packets of a stream in an uncongested WLAN where the MN is not close to the edge of the cell and the received signal strength is good. In the proposed handover scheme the MN records the arrival time of each packet using the MN's internal clock.

When the MN is receiving a single stream of RTP packets it continuously monitors the jitter. To calculate the jitter, the MN compares the arrival time of each packet with that of the previous packet in the stream.

When the MN receives two streams of RTP packets it calculates the delay difference between the two streams by comparing the arrival time of each packet with the arrival time of the corresponding packet in the other stream.

An application layer agent was created to implement this scheme and was used for gathering the results presented below.

5 WLAN Handover Results

Tests were performed to demonstrate a soft handover of streamed video when a roaming MN is about to go outside the range of an uncongested AP, as shown in Fig. 6, in the metropolitan scenario described above.

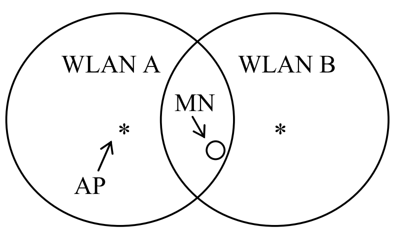

Figure 6. Handover at edge of WLAN cell

A video server was fitted with two LAN cards and was connected to an 802.11b AP and Nistnet [11] as shown in Fig. 7. The MN was multihomed and had a WLAN and a LAN card. Nistnet is a network emulation tool that is frequently used to emulate network conditions at the IP layer. It was used here to take the place of a second WLAN, and it emulated packet delay in a second WLAN cell. The cable between Nistnet and the MN was sufficiently long to allow the MN to roam outside the range of the AP.

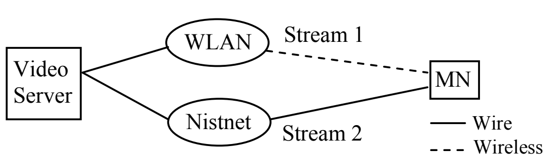

Figure 7. Soft Handover test setup

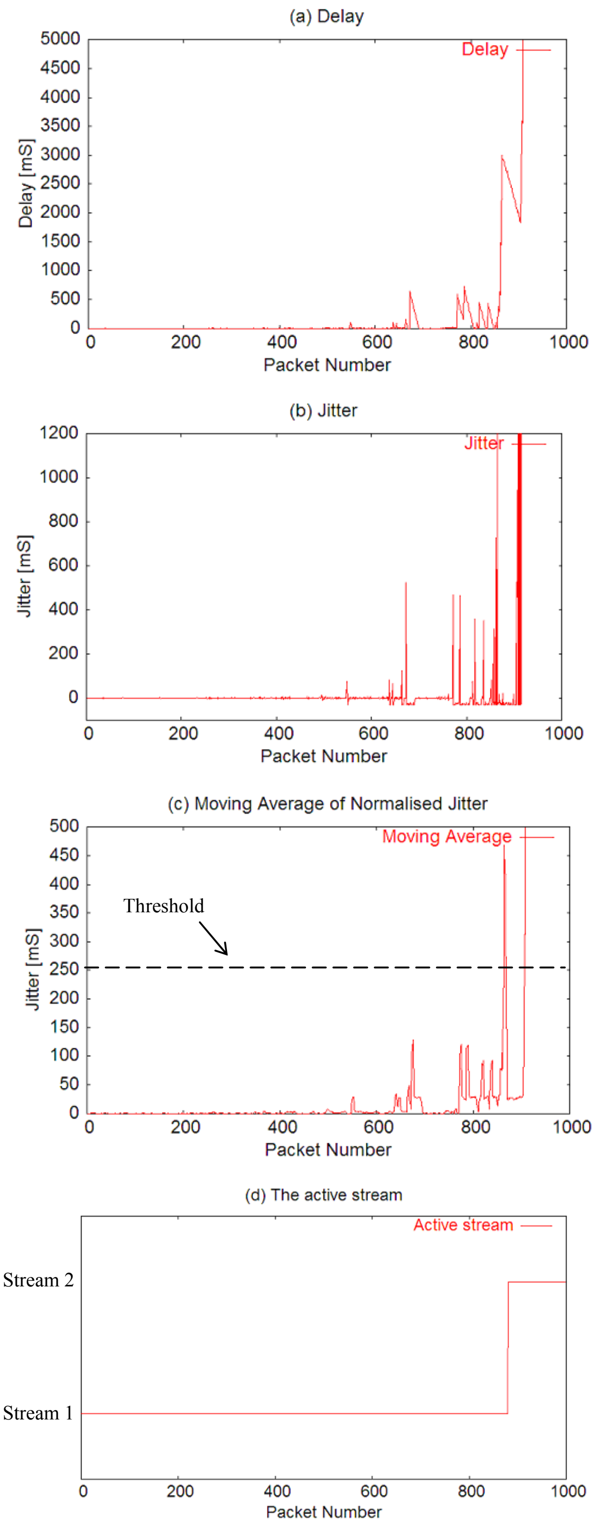

Figure 8. Delay, Jitter and Handover decision

The server streamed a single stream of RTP packets over TCP to the MN via the AP. This RTP stream was used to model CBR video traffic. The packets were transmitted at 30 packets per second and contained 1448 bytes of RTP data each, corresponding to a video bitrate of 347,520 bits/s. There was a line of sight between the AP and MN throughout the test. The MN was initially beside the AP, and was then moved at a steady walking pace away from the AP until the range of the AP was exceeded.

The traces shown in Fig. 8(a) and Fig. 8(b) show the delay and jitter respectively, experienced by the RTP packets of the initial stream during the test. Fig. 8(c) shows the moving average of the jitter shown in Fig. 8(b). When the moving average of the jitter exceeded the threshold shown in Fig. 8(c) a second stream was requested by the MN from the server. Nistnet applied a delay of 10ms to all packets arriving from the second stream. When the MN received the two streams it calculated the delay difference, and dropped the initial stream as it had the higher delay. Fig. 8(d) shows the point when the MN performed a handover to the second stream.

It is desirable to limit the frequency of handovers as the operation of handing over consumes significant network and device resources. Using a moving average of the jitter and choosing a high threshold in Fig. 8(c) limits the frequency of handovers.

6 Conclusion

In this paper we showed that the jitter experienced by RTP packets passing through a WLAN can be used by a roaming MN in an uncongested WLAN to initiate a soft handover. We showed that delay difference between two steams in a soft handover can be used by the MN to decide which stream to use and then drop the other stream. We described a scheme that can be implemented in the MN to achieve this. Finally we presented results showing a successful handover from a WLAN to an emulated WLAN.

Further investigation is planned to consider longer end-to-end delays and to determine the most appropriate threshold level for initiating a second stream. In particular, this will focus on the problems associated with handover between WLAN and a cellular system with high latency. Also, the behaviour of the scheme in a congested WLAN environment is planned.

Acknowledgements

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