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Retrieving Packets from Losing during Service Disruption Time, During Vertical Handover among UMTS and WLAN

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ABSTRACT

In voice activity based communication Choi et al show that using of mutual silence period to do the vertical handover (VHO) is a perfect choice to minimize the packet loss. However according to their proposal, after starting the VHO and before completing it, if one party breaks the silence pushing VoIP packets in channel, those will be lost though those packets may contains significant even very worthy information. In this paper we have shown that by using a small buffer, monitoring the VoIP packets at service disruption time, managing those in the buffer and then sending packets from it after completing VHO until the buffer becomes empty. In our experiment it was found that the proposed method improves the Choi et al's proposal and prevents the packet losses surprisingly.

Keywords – **VHO, voice activity, VoIP, SIP, service disruption, MSP.**

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1. Introduction

In voice activity based communication, losses of packets is a critical issue during the vertical handover. Usually there a large probability of losing packets during VHO even if enhanced signaling mechanism [6] is used. Those losses of packets, generally occurred in service disruption time [1], may decrease the QoS. So for an uninterrupted and quality assured communication VHO should be performed when channel is idle and no party is sending VoIP packets.

To have the exact flavor of QoS, we should emphasis on decreasing the packet losses during VHO. A good choice of making VHO is at mutual silence period [6] to minimize the losses of packets as then both parties are not sending packets. From an observation it is seen that 60% of the VoIP call duration [2] either party does not send packets. That is why triggering VHO at mutual silence period (MSP) will be nice option. The length of mutual silence period always may not be found as our requirement like for service disruption time [4]. So pre and post registration should be applied. In searching for a mutual silence period a margin [5] time is usually set. After that margin time VHO is performed even if MSP is not found. This is done to force the system to do VHO with in a desired period.

However, doing VHO at MSP does not give us assurance of zero packet loss as either party may break the silence just after the start of VHO process by sending VoIP packets. According to the technique of Choi at el in [6], if such packets arrive within service disruption, those will go to hell as there is no mechanism to buffer and to send them next. However those packets may contain very useful information, i.e. password, token, which is necessary for the other party. In this paper we have proposed technique to overcome that problem of Choi [6]. Choi et al where they have minimized the six-state model of Brady [3] to two-state model to analyze the two-way conversation. But it suffers from QoS as it cannot protect the losses of packets which were arrived at service disruption time. This critical issue is solved in our proposal.

This paper is organized in very few sections where section-2 describes Choi et al's method and its problem where as section-3 will explain the proposal of that paper. Section-4 is used to demonstrate the result and to show the improvement of our method over Choi et al's one. Section-5 draws conclusion.

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2. CHOI ET AL'S TECHNIQUE TO DO VHO

Choi et al in [6] have used mutual silence period to make vertical handover (VHO) to minimize the rate of packet loss. So they in their algorithm have searched for a silence insertion descriptor (SID) packet in both the packets of uplink and downlink. As soon as they have found a mutual SID, they have started detachment and attachment procedure to make the VHO.

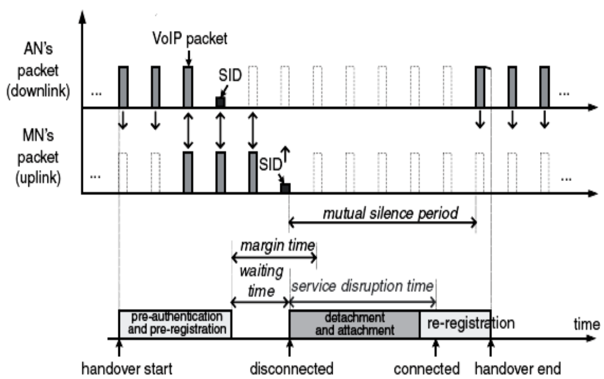

Figure 1: CHoi et al's model of mutual silence.

2.1 Problems of Choi et al's proposal: However after starting the detachment and before completing the re-registration if any party breaks the silence, VoIP packets arrive, those packets will be lost. In the following example access network (AN)'s two packets in the middle will be lost which may contain important message and will degrade the QoS.

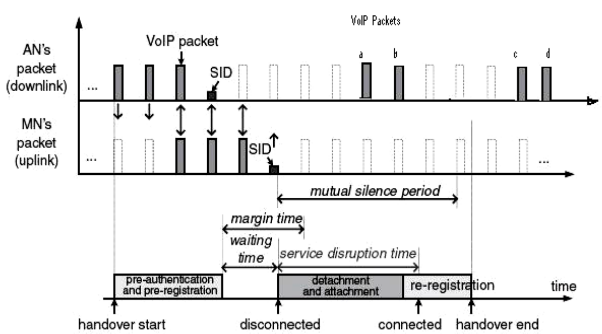

Figure 2: Breaking of silence before completing VHO

3. PROPOSED TECHNIQUE

This proposal is provided based on Choi et al's algorithm to overcome the losses of packets which were arrived during service disruption time. As the disruption time is with in 0.1 second to 0.4 second [4], the number of packets introduced within that period will be very few in number. But the contradiction is shown in [8] where the authors had argued that the VHO latency time is in 4 to 5 second. Another experiment in [7] shows that VHO latency can be of 7 seconds. Using analytic model and testbed experiments, authors in [8] shows that VHO latency can easily be as high as 3 second. So we can argue that VHO latency can be enough high within which a significant message can arrive. But any packets, even if any significant message, arrived at service disruption time will be destroyed [6] as they in [6] have not taken any care about such packets. However we are much careful about the QoS to the customer.

3.1 Policy

This few packets may contain very significant message. So we should care of those. We can use the technique, not to destroy, rather to deliver those to the destination. We can use a small buffer to store such packets which will be found within the disruption period. After the end of handover, if we find any idle slot of that party (i.e. AN) we can start sending the packets from buffer until the buffer becomes empty or next VoIP packet from that party is generated. This way we can decrease the losing of packets so as to increase the QoS.

3.2 Implementation Technique

In the following figure, just after completing VHO, we have searched for an idle slot and then the buffer if there is any packet to be transmitted. So we can see that packet **a** is transmitted as an idle slot is found. But packet **b** has been lost due to the introduction of packet c just after a single idle slot after the end of handover. In that case 50% packets are recovered from losing. In fact it will increase the QoS by saving the packets from loss.

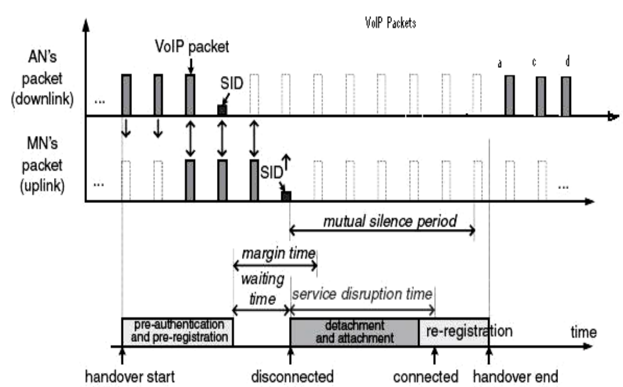

Figure 3: Pading from buffer just after completing VHO if an idle period is found

Another technique can be used to allow all packets to be transmitted by maintaining a queue. As the disruption time

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is very small, very few packets will be stored in buffer. In that case after the end of handover, we can start the sending from buffer. Before sending all packets in buffer, if VoIP packets arrive from that party to be transmitted, those packets can then be stored in buffer as a queue. So then one packet will be stored in queue and at the same time another packet will be sent from queue. If SID packet is found, the queue will then be empty after certain packets have been sent.

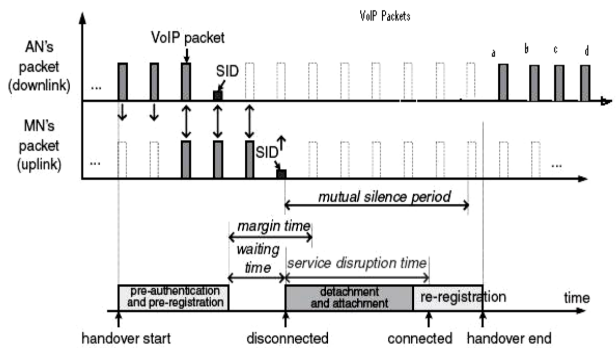

Figure 4: Using a queue in a buffer

So the following procedure can be part of the main vertical handover procedure.

1. Start detachment and attachment procedure
2. If any VoIP packet arrives store in Queue Q
3. Increase the count value by one for each VoIP packets stored in Q
4. Execute re-registration
5. Pop from Q to send packets in to channel and decrease count by one
6. If any VoIP packet arrives store in Q and increase count value by one
7. If Q is not empty, count>0, goto step 5
8. End

4. Result Analysis

We have experimented that proposal using NS-2. It is a nice platform to make programming code and to simulate a network. We have analised our problem in a testbed varying the size of buffer.

4.1 Logical Analysis

Logically we can urgue for the use of a buffer. If we do not use buffer the packets arrive in service disruption time will be lost. If we use a buffer with very small size, it may be overflow to store the VoIP packets. On the contrary if we use a big size of buffer no packet will be lost. However it will be very expensive and inefficient because managing queue in a buffer for a long time will make the VoIP communication system slow in a low bandwith environment. So we should look for the size of a buffer to have a buffer size-managing cost tradeoff.

4.2 Experimental Result

In our experiment, to get the best performance, the size of buffer was used to store 30 VoIP packets. Using buffer size to 10, it was found that a few packets had been lost. Uses of big size of buffer also create a delay in deleivering the packets as the queue contains lot of packets, i.e. 200, to be tranfered though new packets are ready to be transferred. The experiemntal result shows a good improvement over Choi et al method.

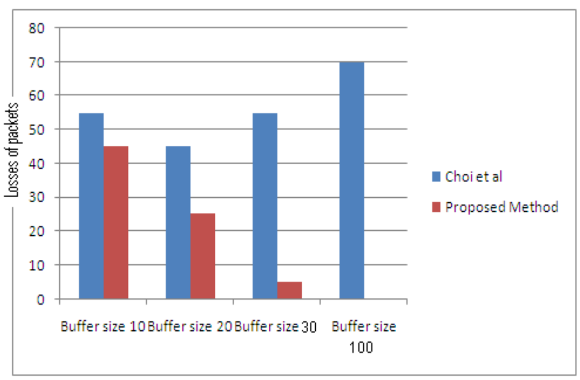

Figure 5: Comparative study of Choi and our proposal agains QoS

5. Conclusion

This proposal is to overcome the problem found in Choi's method. In any communication system, if VHO is necessary our proposed method will make a nice contribution in assuring the QoS to the customer. In transmiting text or audio it will give us a sound solution. We furture wish to use such technique in a moderated way in seamless video transmission.

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