A Novel Scheme to Support WiMax/WiFi Vertical Handoff using SIP

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Abstract-In recent days vertical handoff has become one of the most challenging issues in wireless and mobile networks, when users roam between different access networks. Managing vertical handoff have been proposed and realized on different layers. In this paper a novel scheme named Modified SIP (MOD-SIP) is proposed for WiMax/WiFi vertical handoff in application layer. Our proposed scheme aims to provide (i) better charging procedures during handoff to support network service providers, (ii) minimizes the workload on CH to perform handoff (iii) a better way to eliminate the stale address situation. We have developed a prototype model in java and tested it in different real-time user scenarios. We have measured throughput and packet loss and the performance characteristics of MOD-SIP are studied. The results reveal that both handoffs i.e. from WiMax to WiFi network and from WiFi to WiMax network work well and fulfill our aims, but have some limitations in packet loss and handoff latency. Hence further works are carried out to reduce packet loss and achieve low latencies.

Keywords-MOD-SIP, WiFi, WiMax, Vertical Handoff, SDP(Session Description protocol), Handoff Phases.

I. INTRODUCTION

Future generation wireless networks is striving to integrate different wireless access networks such as IEEE 802.11, IEEE 802.16, 3G, GPRS, UMTS to achieve a ubiquitous computing environment. Hence heterogeneous wireless networks have to cooperate in providing users with better quality of service (QoS) and seamless mobility. One of the important wireless technologies today is WiFi. Standard personal devices like Laptop, mobile devices uses WiFi technology to connect to the Internet. But its range is very limited. The new emerging wireless technology is IEEE 802.16 WiMax(Worldwide Interoperability for Microwave access). WiMAX is a relatively new but very promising standard for wireless communication because it provides the speed of WiFi and the coverage of UMTS (Universal Mobile Telecommunications System). Wireless LAN’s limited coverage range makes it difficult to support a ubiquitous computing environment. 3G can offer universal network access its access rate is very limited. WiMax can provide high speed internet access in wide area. Hence it is natural to combine WiMax and WiFi and create a better wireless solution to provide high speed Internet for mobile users. In general in heterogeneous networks seamless handoff is achieved with the help of handoff management protocols. Handoff management have been proposed in different OSI layers. Mobile IP[1] a standardized protocol by IETF for IP Mobility functions in the network layer. SCTP[3] (Stream Control Transmission protocol), functions in the transport layer. When these protocols are used modifications have to be done in the mobile devices at network and transport layers, respectively. Session Initiation Protocol (SIP) [2], which functions in the application layer, is transparent to the underlying networks. Hence, modifications are not required at underlying layers. We have done a survey of seamless vertical handoff schemes for WiFi/WiMax heterogeneous wireless networks[12]. Based on that, we have chosen Session Initiation Protocol in the application-layer and studied the handoff characteristics of the mobile node.

Several works have been carried out in vertical handoff, but most of them is related to WWAN and WLAN [6]. Authors in [7] have considered real-time applications but they have not given importance to network service provider environment. In [11] authors have proposed a scheme in which a CH bicasts in the handoff region. This adds additional load to the CH and becomes critical in real user environment. Vertical handoff between WiFi and WiMax is presented in [12] but they have used mSCTP which does not support network service provider environment. In our paper we have presented MOD-SIP by modifying the Mid-call mobility in SIP in order to fulfill the proposed criteria in WiMax/WiFi heterogeneous networks. Overview of IEEE802.11 and IEEE802.16 is presented in section II. SIP based terminal mobility with MOD-SIP is explained in section III. The fourth section deals with the vertical handoff between WiMax and WiFi networks. Simulation and evaluation using MJ SIP is presented in section V followed by conclusion.

II. PROTOCOL OVERVIEW

A. Overview of WiFi

The IEEE 802.11 standard [15] provides low cost and effective wireless LAN service. The deployment of high speed network (11Mbps in 802.11b and 54Mbps in 802.11a/g) can be easily established by the free and unlicensed spectrum (2.4GHz in 802.11b/g and 5GHz in 802.11a). The IEEE 802.11b standard is one of the most commonly used standards for the WLAN. There are 11 available channels in this standard and 3 of them are non-
overlapping channels. On the PHY layer, it employs the Direct Sequence Spread Spectrum (DHSS) technique with Complementary Code Keying (CCK) modulation scheme. This standard operates in two modes: One is the Ad Hoc Mode and the other is the Infrastructure Mode. The Ad-hoc mode of operation allows the computing devices within range of each other to discover and communicate in peer-to-peer fashion without involving central access points. In an Infrastructure type of a WLAN, an Access Point is used to connect computing devices to wired nodes. In this paper we have considered the later type of a WLAN setup.

B. Overview of WiMAX

WiMAX, is a new emerging Wireless technology based on the IEEE 802.16 standard[14]. Its main objective is to provide broadband wireless access over long distances. WiMax base stations can offer greater wireless coverage of about 5 miles, with LOS (line of Sight) transmission within bandwidth of upto 70mbps. The most popular WiMax standards are IEEE 802.16d and IEEE 802.16e. The IEEE 802.16d was proposed in 2004 called fixed WiMAX created by WiMAX Forum [2], but it does not support for mobility. The second standard is IEEE 802.16e proposed in 2005. It is introduced to support for mobility and is called mobile WiMAX. WiMAX specifies the MAC layer and the PHY (physical) layer. These specifications describe the air interface between a Base Station (BS) and a Mobile Station (MS). WiMAX operates in two modes. One is point-to-multipoint (PMP) mode and the other is mesh mode. In PMP mode every Mobile Station (MS) makes its own connection to the Base Station (BS), whereas in mesh-mode every MS gets connected to BS through the other MS. WiMAX is based on the RF technology called Scalable Orthogonal Frequency Division Multiple Access (SOFDMA). This can be described as a division of the frequency band into several sub-carriers. And also the inclusion of (MIMO) Multiple Input Multiple Output, means that both transmitter and receiver use multiple antennas along with flexible sub-channelization schemes enable the Mobile WiMAX technology to provide high data rates and larger coverage and better performance.

III. SIP MOBILITY SUPPORT

SIP (Session Initiation Protocol) as defined by the IETF in RFC 3261[2] is an application layer control signaling protocol used to establish, modify and terminate sessions in an IP based network. Such sessions could be among two or more users and could include Internet Telephone calls, multimedia distribution and multimedia conferences. SIP runs on top of several different transport protocols. SIP can support terminal, session, personal and service mobility [4]. Terminal mobility is one in which a MN moves between different access networks and continues any ongoing session without any break during its movement. Two types of terminal mobility is supported by sip. One is the Pre-Call mobility, and the other is the Mid-Call mobility. In Pre-call mobility a connection is established during the beginning of a new call when the MN has moved already and to a foreign network, whereas in mid-call mobility during the middle of an ongoing session.

A. Existing Mid-call mobility

In most of the existing Mid-Call mobility support, SIP re-INVITE method is used as a solution. Here the MN’s movement to a new network is directly informed to the CH using the SIP request. According to [2] a SIP User agent is capable of initiating a request and modify an existing session. This is done by sending a new INVITE message using the same Call-ID.

![Fig. 1. Mid-Call Mobility](image)

(i)In the re-INVITE method[2], there is a direct end to end communication between the MN and the CH. In such case, the CH is responsible for performing the handoff by establishing a new media session using the IP address of the MN in the new network. Hence the CH must be capable enough to handle the handoff situation completely which becomes difficult in a real user scenario. (ii)When the CH due to some reason has lost the address of the mobile host, it must have a fall-back mechanism to overcome the situation. For example when we have two mobile hosts having a conversation and when driving through a tunnel both loses its connection for a while and when the connection is regained, both would have got a new IP address totally different from that of the old one[4]. Such situations can be avoided by sending retransmissions of invitations also to the SIP server located in the MN’s home network. The CH can relocate a MN by contacting the SIP server in the MN’s
Home network. (iii) In network service provider environment information about location change has to send to the SIP server before the new SIP session starts between the MN and CH which is not possible in the existing mid-call mobility support. This is important to enable proper charging because prices in two different networks can be different. But charging procedures using AAA server is not considered and it is beyond the scope of the paper.

C. Modified mid-call mobility (MOD-SIP)

Fig. 2. Modified Mid-call Mobility

In this paper we propose a novel method which provides solutions for all the problems in the existing mid-call mobility support. But in our MOD-SIP the handoff is totally performed by the server and the CH is totally unaware of the handoff taking place. This totally reduces the load on the CH to perform handoff. And also the change in access technology is handled by the server. This scheme also supports the stale address situation because every move of MN is registered in the server and known to the server.

Handoff for decision is made in the second phase. Here the decision is made based on some threshold levels of performance metrics. In handoff process this phase is more critical because if decision is not made properly there will be high degradation in the QoS. According to the predefined Threshold level handoff takes place from one network to another network.

The third phase is responsible for switching the traffic to a new network. It means that the connection with an old network is terminated and all the traffic traverse the new network. It means that the connection with an old network is terminated and all the traffic traverse the new network.

V. Simulation And Evaluation

We have developed a prototype model by modifying MjSip open source Java SIP stack[16] according to our proposed solution. We have written our own coding in java for WiMax and Wi-Fi networks. Here the SBC(session border controller) of the MjSip is modified to support handoff and used as the SIP server. Fig. 3 shows our simulation network model. We have defined MN which is a dual mode terminal,
which can be connected to both WIFI and to WIMAX network. Static IP addresses were assigned for mobile nodes, SIP server and Access Points. A new SIP message called INVITE_HANDOFF is defined which is similar to the SIP INVITE message. This message is sent to the SIP server from the MN whenever handoff occurs. The movement pattern of MN in the mid of a session from WiMax to WiFi network and vice versa is studied. The handoff process is initiated by the decision for handoff which is a simple decision function written inside MjSip stack to initiate handoff.

The simulation period was set to 180 sec. For exchange of RIP messages and for network routing to set automatically it takes 100 sec initially. Between MN and CH the media streaming application with SIP signalization was used. The average media streaming application time between MN and CH was set to 60 sec. During that time MN will move from WiMax to the WIFI network coverage and then return back from the WIFI network. The average end to end latencies across WIMAX network is 120ms and WiFi network is 10 ms for one way.

A. Simulation results

In the simulation the handoff time and packet loss during handoff is measured and its performance is analyzed. Handoff execution consumes maximum time among the overall handoff time. A new SIP session establishment, is defined as a time interval from decision for handoff to the time when new session is established via new network. After exchanging the SIP messages for new session RTP packets between MN and CH is sent via the new network. To calculate the overall handoff delay this time interval needs to be added to new SIP session establishment time. The delay taken for Handoff is calculated according to (1)

$$D_{handoff\_time} = D_{new\_session} + D_{first\_RTP\_packet} (1)$$

The $D_{new\_session}$ is the delay added for the exchange of all SIP messages and $D_{first\_RTP\_packet}$ is the delay added for first RTP packets to traverse to the new network. When new SIP session is established the RTP packets between MN and CH traverse to the new network, but there are still some packets that traverse the old network. Those packets are lost and will be discarded. The proportion of the lost packets represents the packet loss during handoff and is defined as in (2).

$$packet\_loss = 1 - (received\_packets / sent\_packets) (2)$$

The number of packets is measured on the MN (received packets) and CH (sent packets) in time period from SIP INVITE_HANDOFF message to the time when first RTP packets arrive.

B. Handoff From WiMax to WiFi

In Fig. 4 throughput on all interfaces of MN is presented for handoff from WiMax to WiFi. As observed from the graph that S/N ratio of WiFi network felt below THS/N at 145.3 sec of the simulation. This started handoff process. It is also observed that the SIP INVITE_HANDOFF message was sent via WiMAX Tx and Rx of MN, while for the RTP stream still old network has been used (i.e. packet traversing via WIFI Tx and Rx interface of MN). Because of bigger delays in WIMAX network the SIP message exchange was longer than in the previous case. After the SIP 200 OK, MN started to send RTP packets via new network. When this happened throughputs on WIFI Rx and Tx interfaces started to decrease, while throughputs on WIMAX interfaces started to increase.

Table 1 shows the handoff characteristics measurements for WiMax to WiFi. From the results it is observed that handoff time to WIFI network is not critical. This is because of the very fast connection in WIFI network, which enabled quick SIP session setup in 49 ms and first RTP packets transmission of in 14 ms. But the packet loss during handoff is 28 % which can be critical. Such a proportion of discarded packets happened due to bigger delay in WIMAX network and large number of packets in WIMAX network.

C. Handoff From WiFi to WiMax

In Fig. 5 throughput on all interfaces of MN is presented for handoff from WiFi to WiMax. As observed from the graph the S/N ratio of WiFi network felt below THS/N at 124.9 sec of the simulation and the handoff is executed. Immediately the SIP INVITE_HANDOFF message is sent via WiFi TX to the SIP server to handle the handoff situation whereas the RTP packets still traverse the WIMAX TX. After receiving SIP 200 OK from the SIP server the RTP packets start traversing the WIFI TX. Now the throughput on WIMAX receiver and transmitter starts to decrease, while the throughputs on WIFI interfaces starts to increase.
Fig. 5. Throughput on all interfaces of MN (WiFi to WiMax).
Table 2 shows the handoff characteristics measurements for WiFi to WiMax. From the results we can see that total handoff delay is high compared to previous case. This is because of the bigger delay in WIMAX network. The packet loss which is not critical in this case because delay of the WIFI network is very low. Therefore only very small number of packets that traverse WIFI network, and hence the packet loss is only 7%.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>D_new_session</td>
<td>304ms</td>
</tr>
<tr>
<td>D_Ext_RTP_packet</td>
<td>78ms</td>
</tr>
<tr>
<td>D_handoff_time</td>
<td>382ms</td>
</tr>
<tr>
<td>Packet Loss</td>
<td>7%</td>
</tr>
</tbody>
</table>

Table 2. Handoff measurements (WiFi to WiMax)

VI. CONCLUSION

In this paper we have presented a novel scheme MOD-SIP to support network service provider environment by modifying the existing SIP protocol. Our solution also reduces the load on the CH to perform handoff because handling handoff by the CH in real time environment is highly critical. And also when the connection between the MN and the CH is lost for a moment i.e. the stale address situation, is avoided by registering each and every movement of the MN with the SIP server. The handoff latency and packet loss of a Mobile Node in a WiMax/WiFi heterogeneous networks is studied under different conditions. Based on the simulation it is observed that this method enhances our proposed aims but the handoff latency and packet loss in SIP is high to provide seamless handoff. We are further working on it to reduce the packet loss and handoff latency and achieve seamless handoff. This novel scheme MOD-SIP can be extended as such to other heterogeneous wireless networks.

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VIII. REFERENCES