Analytical Modeling of IMS based Interworking in Heterogeneous Mobile Data Networks

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***Abstract*-Analytical evaluation of performance measures for vertical handoff management in an interworked Next Generation Mobile Network (NGMN) is presented in this paper. This analysis is based on a novel architecture for interworking previously proposed by authors. It enables the Universal Mobile Telecommunications System (UMTS) cellular technology to interwork with Wireless Local Area Networks (WLAN) under a common platform. The proposed framework exploits the IP Multimedia Subsystem (IMS) as a universal coupling mediator for real-time session negotiation and management. The analysis includes vertical handoff performance measures such as delay, transient packet loss, and signaling overhead/cost. The latter part of this paper includes an OPNET based simulation platform for verification of the results obtained by the analytical model.**

I. Introduction

Ubiquitous data services and relatively high data rates across heterogeneous networks could be achieved by interworking 3G cellular networks with WLANs. This will enable a user to access 3G cellular services via a WLAN, while roaming within the range of a hotspot. Thus WLANs can be considered as a complementary technology for 3G cellular data networks as well as a compulsory element of the future NGMN [1]. A variety of internetworking architectures for 3G Cellular and WLANs have been proposed [2]. By and large, these internetworking architectures may be categorized as tight coupling, loose coupling, and peer-to-peer networking (also referred as no-coupling) [3], [4]. However, these approaches seem to provide limited internetworking capability as neither of these designs has successfully addressed the issue of seamless service continuation.

Having identified the importance of this need, we have recently proposed a solution for achieving session continuation during a vertical handoff between WLAN and UMTS networks [5]. The significance of our proposed architecture is that it uses a novel approach, that is, the use of the 3GPP's IMS [6] for supporting real-time session negotiation and management with additional controls as inspired by [7], [8]. The IMS, as introduced in UMTS Release 5 within its core network, comprises of the required characteristics for control of real-time multimedia sessions and plays an essential role in the provision of IP multimedia services in a UMTS network. This paper primarily focuses on using an analytically modeled approach for further analyzing the performance of vertical handoff management over a packet switching service domain. Based on this, an analysis on vertical handoff performance measures such as delay, transient packet loss, and signaling overhead/cost is presented. This further investigates the validity of this architecture by using the OPNET simulation platform developed for [5].

The reminder of this paper is organized as follows. The next section presents an overview on the proposed architecture. Followed by which comes the sections on analytical modeling and performance analysis. Lastly, an OPNET based model is included for verification prior to the concluding remarks.

II. IMS based WLAN-UMTS Interworking

This section provides a brief overview of our proposed interworking mechanism with specific attention towards its vertical handoff mechanism. Interested readers may refer to [5] for more specific and detailed information on the architectural design.

As per the illustration in Fig. 1, the flow of data originates from the source Mobile Host (MH), through the Serving Gateway Support Node (SGSN) and the Gateway GPRS Support Node (GGSN), and reaches the destination network. This model uses the Visitor-GGSN approach to avoid the inter-PLMN (Public Land Mobile Network) backbone and to make data routing simpler for the network operator [9]. In whichever approach, the data flow bypasses the IMS network. Thus the IMS is said to follow the philosophy of using different paths for user data and signaling through the network. The SIP [10] signaling messages originate from the MH via the UMTS Terrestrial Radio Access Network (UTRAN), through the SGSN and GGSN, out to the Call Session Control Functions and finally to the destination network. It is important to note that when the MH requires establishing a session, this request is always sent to the (Home) Serving – Call Session Control Function via the (Visiting) Proxy – CSCF of the IMS. During the exchange of IMS-SIP signaling, both the SGSN and GGSN act as routers by merely forwarding IMS-SIP messages.

The data originating from the WLAN is routed via a SGSN emulator to the UMTS GGSN. It essentially emulates the WLAN as another SGSN belonging to the same UMTS network. Thus mobility can be managed by the UMTS network. Some of the functionalities of the BSS are bypassed in this approach and the load on the UMTS network, created by the high volumes of WLAN data traffic, may also be

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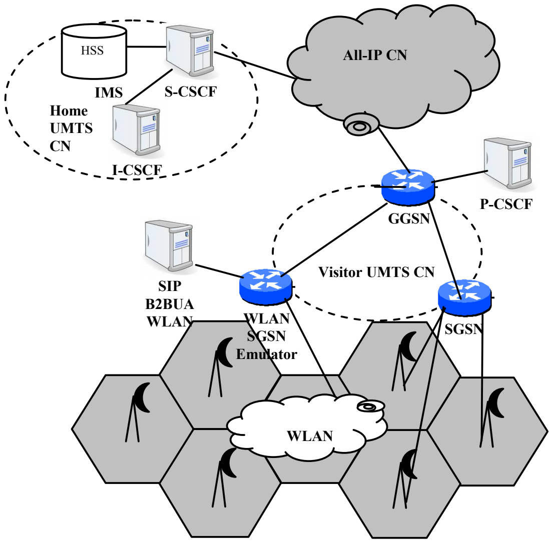

Fig. 1. The proposed interworking architecture.

sufficiently reduced. Furthermore, a MH does not require any change of IP addressing between the WLAN and UMTS network as long as the two networks are connected to the same GGSN.

Prior to establishing a SIP session, the MH requires performing a service registration function to let the IMS know its location. The MH acts as a SIP client and sends a SIP registration message to its home system through the P-CSCF. The basic steps for a SIP service registration can be summarized as follows. Firstly, the IMS Home Subscriber Server (HSS) for the MH is notified of its current location for the HSS to update the subscriber profile accordingly. Next the HSS checks if the MH is allowed to register in the network based on the subscriber profile and operator limitations, and grants authorization. Once authorized, a suitable S-CSCF for the MH is assigned and its subscriber profile is sent to the designated S-CSCF. After the activation of the PDP context and the service registration, the MH is ready to establish a media/data/call session. As illustrated in Fig. 2, the sequence of the SIP session origination procedure can be described as follows.

The mobile origination procedure is initiated by a SIP INVITE message sent from the UMTS interface of the source MH. This initial message is forwarded from the P-CSCF to the S-CSCF of the originating network, via the CSCFs of the terminating network, and finally to the destination. Next, the destination responds with a 183 Session Progress containing a SDP answer (if the INVITE contains a request to follow the precondition call flow model) with information of media streams and codecs that the destination is able to accept for this session. The acknowledgement for the reception of this provisional response (PRACK) follows afterwards. If the destination does not receive a PRACK response within a determined time, it will transmit the provisional response. When the PRACK request successfully reaches the destination a 200 OK response is generated by the destination with an SDP answer. Next an UPDATE request is sent by the source containing another SDP offer, in which the source indicates that the resources are reserved at his local segment. Once the destination receives the UPDATE request, it generates a 200 OK response. Once this is done, the MH can start the media/data flow and the session will be in progress (via the UMTS interface). As the WLAN interface becomes active, the need for a mechanism for a pure SIP (or Application Layer) based session handoff arises and the SIP REFER method is chosen for explicitly transferring the session to the new interface [11]. Under realistic conditions, vertical handoff decision must ideally be triggered by a network selection mechanism. Since network selection criteria are beyond the scope of this report, a manual triggering for handoff is considered.

As illustrated in Fig. 2, the basic steps for IMS-SIP based session handoff is as follows. The UMTS interface notifies the WLAN interface with a SIP REFER request (step 8). The REFER request contains a "Refer-To" header line containing the destination SIP URI and a "Replaces" header line identifying the existing session to be replaced by the new session. Next the WLAN interface sends the CN an INVITE message with the "Replaces" header received from the previously received REFER request (step 10). Also the new IP address and port numbers are also included in the SDP body of this INVITE message. The receipt of the "Replaces"

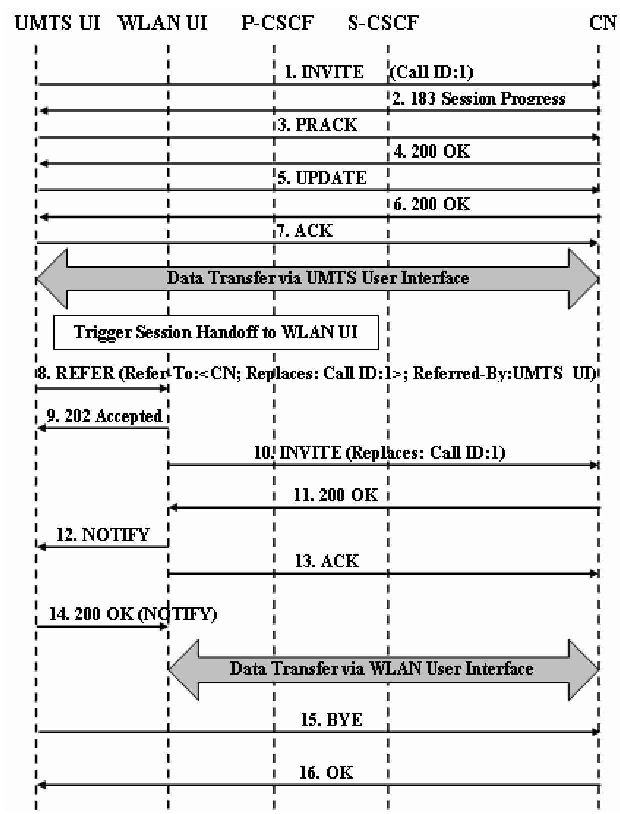

Fig. 2. IMS-SIP based session handoff.

header is what indicates that the initial session is to be replaced by the incoming INVITE request and hence be terminated. Now the WLAN interface has successfully established a direct signaling relationship with the CN. Once the WLAN interface has successfully established a session with the CN, it sends a NOTIFY request to the UMTS interface updating the final status of the REFER transaction (step 12). This NOTIFY message contains the session information of the newly established session allowing the UMTS interface to subsequently retrieve the session (if so desired). Once the data flow is established between the WLAN and CN, the UMTS interface tears down its session with the CN (steps 15-16). Also note that in the event that the provided information in the replaced header does not match any existing session the triggered INVITE does not replace the initial session and will be processed normally. Thus any failed session handoff attempt can not destroy the initial session.

III. Analytical Modeling

An analytical model is derived for evaluating the proposed scheme for analyzing the QoS metrics and measures involved in session and mobility management. More precisely, QoS metrics such as handoff delay, total packet loss, and signaling overhead are analyzed. The derivations of the numerical results have been omitted in the following sections for brevity.

A. Handoff Delay

A standard vertical handoff delay during mid-session mobility consists of the following sub-procedures (or delays); D_1 = Link layer HO delay, D_2 = movement detection delay, D_3 = address allocation delay, D_4 = session re-configuration delay, and D_5 = packet re-transmission delay. The vertical handoff delay/s at the network layer (and above) are calculated independent of the link layer delay D_1 and mainly consist of D_3 and D_4. According to our proposed architecture IMS based vertical handoff, there is no DHCP related address allocation; hence it can be argued that the main contributor for network layer based vertical handoff delay is D_4. The session re-configuration delay (D_4) mainly consists of IMS based session handoff delay (D_{IMS}).

Let $D(S, H_{a-b})$ denote the end-to-end transmission delay of a message of size S sent from a (an MN away) to b via wireless and wired links. Thus, $D(S, H_{a-b})$ can be expressed as follows [12]:

$$D(S, H_{a-b}) = \left[\frac{S}{B_{wl}} + L_{wl}\right] + \left[\frac{S}{B_w} + L_w\right] \times H_{a-b} + [H_{a-b} + 1] \times L_{proc} \quad (1)$$

where, S is the average size of a signaling message, H_{a-b} is the average number of hops between a and b, B_{wl} is the bandwidth of the wireless link, B_w is the bandwidth of the wired link, L_{wl} is the latency of the wireless link, L_w is the latency of the wired link, and L_{proc} is the processing delay at node. Therefore, by using (2), D_{IMS} for the architecture given by Fig. 1 and Fig. 2 can be expressed as in equation (3).

$$\begin{aligned} D_{IMS} = {} & D(S_{Refer}, H_{UMTS\text{-}WLAN}) + D(S_{202Accept}, H_{UMTS\text{-}WLAN}) \\ & + D(S_{INVITE}, H_{WLAN\text{-}CN}) + D(S_{183\text{-}SP}, H_{WLAN\text{-}CN}) \\ & + D(S_{Notify}, H_{UMTS\text{-}WLAN}) + D(S_{PRACK}, H_{WLAN\text{-}CN}) \\ & + D(S_{OK}, H_{WLAN\text{-}CN}) + D(S_{UPDATE}, H_{WLAN\text{-}CN}) \\ & + D(S_{OK}, H_{WLAN\text{-}CN}) + D(S_{ACK}, H_{WLAN\text{-}CN}) \\ & + D(S_{OKNotify}, H_{UMTS\text{-}WLAN}) + \Delta \end{aligned} \quad (2)$$

where, Δ is additional IMS (application layer) latency due to HSS lookup process. It is worth reminding that make-before-break handoff is applied in the proposed handoff scenarios, which helps compensate for large handoff delays. However, for purpose of a complete analysis of the vertical handoff delay, the standard straight forward case of break-before-make handoff scenario is used.

B. Packet Loss

The total packet loss (*Pkt_loss*) during a session can be defined as the sum of all lost packets during the vertical handoff while the MN is receiving the downlink data packets. It is assumed that the packet loss begins when the L2 handoff is detected and all in-flight packets are lost during the vertical handoff time. Thus, it can be expressed as follows [12]:

$$Pkt\_loss = \left[\frac{1}{2T_{ad}} + D_4\right] \times \lambda_d \times N_m \quad (3)$$

where, T_{ad} is the time interval between MIP agent advertisements, λ_d is the downlink packet transmission rate, and Nm is the average number of handoffs during a session. Nm is known as t_s/t_r, where t_r is average network resident time and t_s is average call (session) connection time.

C. Signaling Overhead/Cost

The resultant signaling overhead/cost of mobility management during vertical handoff can be analyzed as follows. The signaling overhead is the accumulative traffic load on exchanging signaling messages during the MN's communication session. The overhead incurred by a message can be defined as [13]:

$$O_{message} = S_{message} \times H_{a-b} \quad (4)$$

Thus the total signaling overhead incurred by vertical handoffs during a given data session can be expressed as follows:

$$O_{IMS} = \lambda_m \sum_{i=1}^{n_1} \left(S_{IMS-Invite-i} \times H_{(a-b)-i}\right) + \lambda_s \sum_{i-1}^{n_2} \left(S_{IMS-ReInvite-i} \times H_{(a-b)-i}\right) \quad (5)$$

$$O_{IMS} = \left[\lambda_m \sum_{i=1}^{n_2} \left(S_{IMS-ReInvite-i} \times H_{(a-b)-i}\right)\right] \frac{\lambda_s}{\lambda_m} + \lambda_m \sum_{i=1}^{n_1} \left(S_{IMS-Invite-i} \times H_{(a-b)-i}\right) \quad (6)$$

where n_1 and n_2 represent the number of messages involved in each handoff/message sequence. If λ_m is the average network mobility rate of a MH and λ_s is the average call (session) arrival rate, λ_s / λ_m can be called as the call-to-mobility rate (CMR). If the average call (session) duration/holding time is μ^{-1}, the average call (session) completion rate will be μ. Hence λ_s / μ can be called as Utilization. Thus the total signaling overhead incurred by vertical handoffs during a given data session can also be expressed as follows:

$$O_{IMS} = \left[\mu \sum_{i=1}^{n_2} \left(S_{IMS-\mathrm{Re}\,Invite-i} \times H_{(a-b)-i} \right) \right] \frac{\lambda_s}{\mu} + \lambda_m \sum_{i=1}^{n_1} \left(S_{IMS-Invite-i} \times H_{(a-b)-i} \right) \tag{7}$$

IV. Performance Analysis

The following numerical results are generated using 3GPP-SIP messages. Table 1 shows the typical SIP message sizes and other related parameters. IMS-SIP values are based on [14]. Other related parameters have been partly obtained from [12] and [13] to maintain consistency. The relative distances in hops are illustrated in Fig. 3. Based on these assumptions, the following analytical results are derived for equations (2), (3), (6) and (7) for the scenario of a pure SIP based vertical handoff.

TABLE I
IMS-SIP Message Sizes and Parameter Values

Message	**Size (Bytes)**	**Parameter**	**Value/s**
INVITE	736	B_{wl}	16 Kbps – 54 Mbps
Re-INVITE	731	B_w	100 Mbps
183 Ses. Pro.	847	L_{wl}	2 ms
PRACK	571	L_w	0.5 ms
200 OK	558	L_{proc}	0.001 sec
UPDATE	546	λ_m	2-10 $h^{-1}MH^{-1}$
ACK	314	λ_s	2-10 $h^{-1}MH^{-1}$
REFER	750	μ^{-1}	2-6 min
200Accepted	550	CMR	0.1-10
NOTIFY	550	Δ	100 ms
OK NOTIFY	550	T_{ad}	1 sec
		λ_d	64 Kbps – 1Mbps

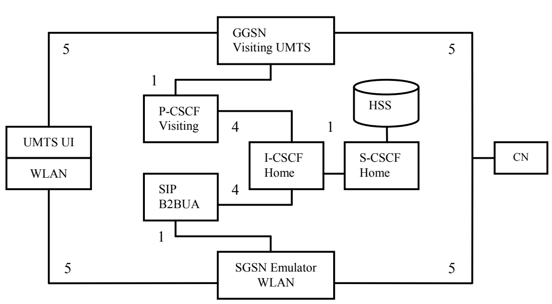

Fig. 3. Relative distances in hops.

Fig. 4 illustrates the behaviour of the vertical handoff delay against the wireless link bandwidth. The behavioural trend of the graph indicates that the proposed method of handling IP mobility with the help of the IMS-SIP Refer method gives rise to relatively high vertical handoff delay. One of the main causes of this high vertical handoff delay is the number of application layer/IMS based processing latencies. Another interesting observation is that when the bandwidth of the wireless link increases from zero to 2 Mbps the handoff delay decreases exponentially. It also indicates that for relatively wider bandwidths (say, beyond 512 Mbps) the handoff delay decrease becomes relatively slower. Therefore, this indicates that vertical handoff delays cannot be simply reduced by purely increasing the wireless link bandwidth.

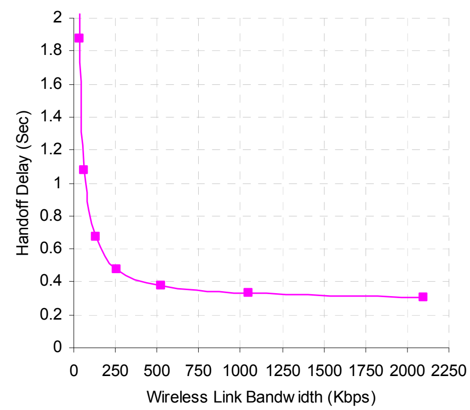

Fig. 4. Vertical handoff delay vs. wireless link bandwidth.

From a rather different perspective, Fig. 5 illustrates how vertical handoff delay behaves against increasing session handoffs. Confirming with the trends exemplified in Fig. 4, graphs in Fig.5 also indicate relatively high transient handoff delays for relatively low data rates (i.e., 256 Kbps and 512 Kbps). The graphs also indicate that handoff delays for relatively high data rates (i.e., 11 Mbps and 54 Mbps) are much closer to each other. Furthermore, according to the presented analytical model, an important conclusion to be

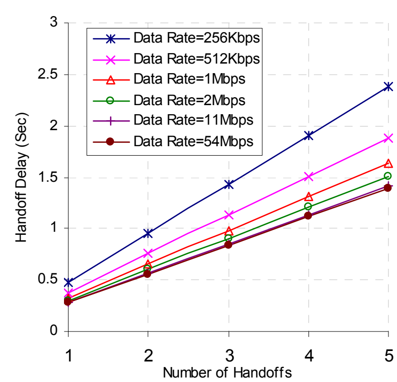

Fig. 5. Vertical handoff delay vs. number of handoffs.

derived is that, the transient packet loss for a single vertical handoff lies between 477 ms (for a 256 Kbps data link) and 278 ms (for a 54 Mbps data link).

Fig. 6 demonstrates the total transient packet loss during the vertical handoff for variable downlink packet transmission rates as the number of handoffs increase (in the case of a break-before-make handoff scenario). According to equation (3), the packet loss during a vertical handoff is relatively proportional to the vertical handoff delay. Therefore, since the proposed approach has a relatively higher handoff delay, the resultant packet loss also follows a similar trend. Another interesting observation of the plots in Fig. 6 is that the graphs diverge from one another as the packet transmission rates and number of handoffs per session increases. The reason for the graphs to show such behaviour is the application layer based additional IMS related latencies. These additional latencies substantially contribute towards increasing handoff delays as the number of handoffs per session increases and eventually the packet loss. However, it is important to note that the proposed model uses a make-before-break handoff technique to avoid such transient packet loss.

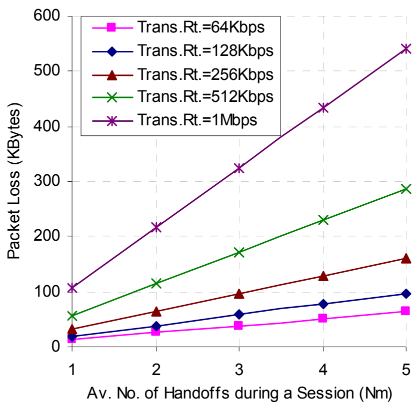

Fig. 6. Total packet loss vs. average number of handoffs during a session (Nm).

Fig. 7 illustrates the behavior of the total mobility management related signaling overhead/cost against the CMR when λ_s is constant. That is, for the graphs in Fig. 6, the call (session) arrival rate (λ_s) is fixed at λ_s = 2 hr^{-1}MH^{-1}, λ_s = 4 hr^{-1}MH^{-1}, λ_s = 6 hr^{-1}MH^{-1}, λ_s = 8 hr^{-1}MH^{-1}, and λ_s = 10 hr^{-1}MH^{-1}. According to these graphs in Fig. 7, the signaling cost reduces exponentially with the increase of CMR since the mobility rate *(λ_m)* declines when the session arrival rate *(λ_s)* is constant.

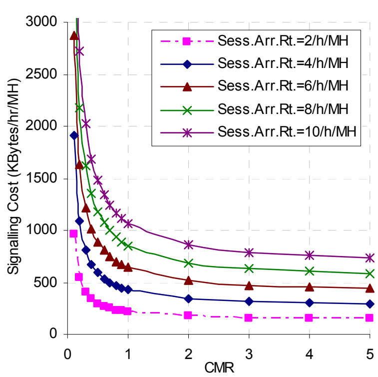

Fig. 7. Signaling cost vs. CMR when λ_s is constant.

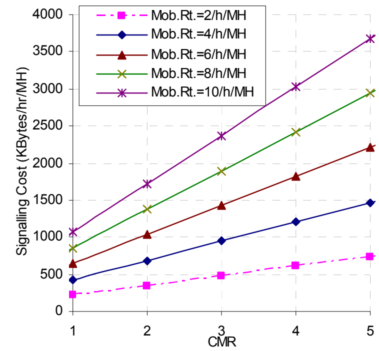

Fig. 8. Signaling cost vs. CMR when λ_m is constant.

Fig. 8 illustrates the behavior of the total mobility management related signaling overhead/cost against the CMR when λ_m is constant. That is, for the graphs in Fig. 8, the network mobility rate *(λ_m)* is fixed is fixed at λ_m = 2 hr^{-1}MH^{-1}, λ_m = 4 hr^{-1}MH^{-1}, λ_m = 6 hr^{-1}MH^{-1}, λ_m = 8 hr^{-1}MH^{-1}, and λ_m = 10 hr^{-1}MH^{-1}. On the contrary to the results of Fig. 6, the graphs in Fig. 7 indicate that the signaling cost increases linearly as the CMR increases. This is because when the CMR increases the session arrival rate *(λ_s)* also increases provided the mobility rate *(λ_m)* is kept constant. This also leads to a very interesting conclusion. That is, more signaling overhead is incurred during the vertical handoff than at the time of the initial session setup.

Next, with the default mobility rate *(λ_m)* fixed to 6 hr^{-1}MH^{-1}, the influence of Utilization on the signaling cost is investigated when the average call (session) completion rate *(μ)* remains constant. That is, for the graphs in Fig. 9, average call (session) completion rate *(μ)* is fixed at μ = 10 hr^{-1}MH^{-1}, μ = 12 hr^{-1}MH^{-1}, μ = 15 hr^{-1}MH^{-1}, μ = 20 hr^{-1}MH^{-1}, and μ = 30

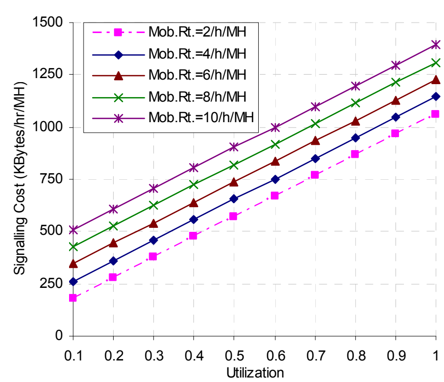

Fig. 9. Signaling cost vs. Utilization when λ_m= 6 hr^{-1}MH^{-1}.

$hr^{-1}MH^{-1}$. With the increase of the Utilization, more handoffs take place and thus signaling costs keep linearly increasing. Furthermore, as μ increases from 10 $hr^{-1}MH^{-1}$ to 30 $hr^{-1}MH^{-1}$, the graphs shows rapidly increasing gradients. This also proves that more signaling overhead is incurred during the vertical handoff process than at the time of the initial session setup.

Lastly, with the average call (session) completion rate *(μ)* is fixed to 15 $hr^{-1}MH^{-1}$, the influence of Utilization on the signaling cost is investigated when the default network mobility rate *(λ_m)* is constant. That is, for the graphs in Fig. 10, average network mobility rate *(λ_m)* is fixed at λ_m = 10 $hr^{-1}MH^{-1}$, λ_m = 12 $hr^{-1}MH^{-1}$, λ_m = 15 $hr^{-1}MH^{-1}$, λ_m = 20 $hr^{-1}MH^{-1}$, and λ_m = 30 $hr^{-1}MH^{-1}$. As the network mobility rate increases, the session setup cost shows a fixed/constant increase (when μ and λ_s are fixed). Hence the graphs show a linear increase with fixed gradients with the y-axis intercept increasing as a result of increasing λ_m. This result re-confirms that the signaling overhead incurred during network mobility is relatively less when compared against the rapidly increasing signaling overhead incurred during session mobility.

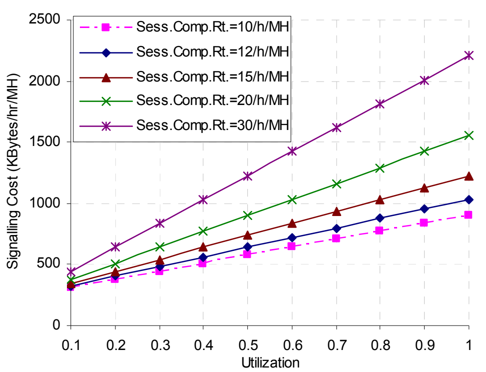

Fig. 10. Signaling cost vs. Utilization when μ = 15 $hr^{-1}MH^{-1}$.

V. Simulation Results and Validation

In order to validate the above numerical results and analysis, a network simulation scenario is modeled using OPNET Modeler 11.5. Since OPNET's standard SIP components does not address the specifications of the 3GPP's IMS substantial modifications are required. Thus a fully functional SIP-IMS model for OPNET is constructed and integrated to OPNET's existing UMTS Special Module. The newly developed SIP-IMS model is an enhanced version of the basic IMS-SIP signaling model, which is currently available under the contributed models library of the OPNET University Program [15]. Fig. 11 illustrates the constructed simulation scenario.

Modifications are made for SIP Proxy Servers (UASs) to function as different CSCFs, User Agent Client (UAC) processes to communicate with modified User Agent Servers (UAS), IMS-SIP based messaging and flow between the CSCFs, roaming facility between multiple domains, and facility for introducing process delay controls (i.e. for

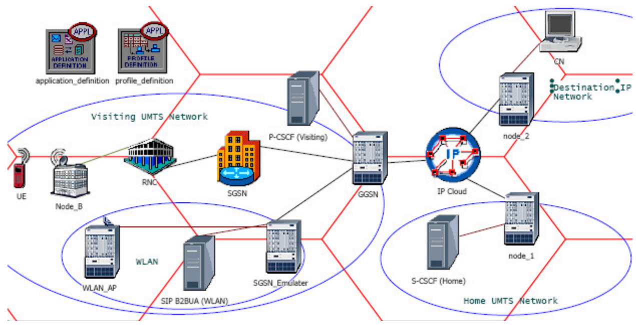

Fig. 11. The OPNET Simulation Model.

messages sent between CSCFs and the Home Subscriber Server queries).

As a result, a UMTS network that is fully capable using IMS based SIP signaling for session management is developed. Next a simple WLAN is connected via an SGSN emulator to the GGSN of the Visiting UMTS Network. The P-CSCF (WLAN) can be seen as a SIP Back-to-Back User Agent (B2BUA), which is capable of interworking with the IMS-SIP and capable of forwarding SIP requests. S-CSCF is the only IMS node implemented in the Home UMTS Network. This is since the I-CSCF is mainly used for SIP Registration process and it is assumed that both UMTS and WLAN interfaces of the UE have already been registered. The Corresponding Node (CN), which is an IMS-SIP UAC, is connected to a destination IP network via a public IP network (IP Cloud). The IMS-SIP message flows basically follow the sequence described in Fig. 2.

Using the newly developed IMS-SIP based platform a series of simulations are performed for evaluating vertical handoff for the previously described scenarios. However, it is worth noting the assumptions made when obtaining these results. Both the UMTS network and WLAN belong to a single IP subnet and IP addressing and routing is statically assigned. Since there are no multiple networks available (except for one UMTS and one WLAN), the need for a network selection algorithm is eliminated. Handoff decisions are individually based on the signal strength of the WLAN, to which the MH either roams into or roams out of.

Fig. 12 illustrates the simulation results for vertical handoff delay obtained for the proposed mechanism. The average vertical handoff delay (from UMTS to WLAN) obtained by the OPNET model for a single VoIP session for an IMS-SIP based mechanism is 340 ms. This vertical handoff delay value (340 ms) obtained by our simulations is approximately close to the value obtained by our analytical modeling (302 ms). The behavioral trend of this graph indicates that the vertical handoff delay is relatively large with IMS-SIP involved in contrast with other vertical handoff mechanisms. This increase is a result of the large sized SIP messages being used for handling IP mobility in the IMS-SIP Refer method and continuous IMS precondition negotiation prior to each session handoff. Furthermore, it also has a relatively higher number of application layer/IMS based processing latencies, which also contributes to a higher vertical handoff delay. As the number of sessions increase the vertical handoff delay shows an exponentially increasing trend. For example, this method shows an (approximate) increase of 30% between the first and second handoffs and approximate increases of 35%, 48% between the next consecutive handoffs. The interesting point about our simulation results is that, to a certain extent, they are inline with results published for a case of similar handoff delays [13],[16]. Furthermore, the most encouraging outcome is that these results suggest efficient ways for the deployment of the IMS as shown in [5].

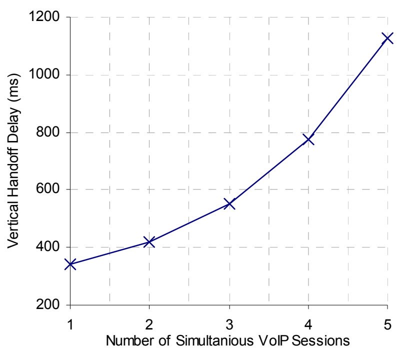

Fig. 12. Mean Vertical Handoff Delay.

Fig. 13 illustrates the simulation results for total packet loss during a vertical handoff for a VoIP call setup over the proposed mechanism. The GSM codec with a 33Kbps data rate is used for this simulation. According to the simulation result, the average packet loss during the vertical handoff for a single VoIP call is 8 Kbps, which is closely inline with the result for packet loss for a 64 Kbps link (12.8 Kbps) given in Fig. 6. As in the case of the explanations given for the increase of packet loss in Fig. 6, we can arguably apply the same for the behavior of the graph in Fig. 13.

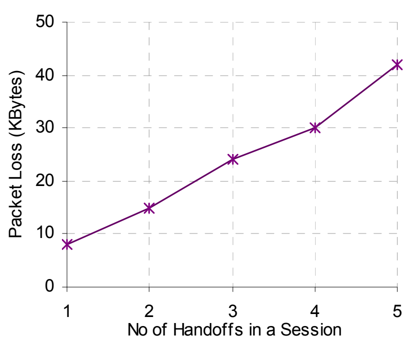

Fig. 13. Mean Packet loss during Vertical Handoff.

Although it has not been analytically modeled, the simulation model is used for measuring the jitter (i.e., the variation in the inter-arrival delay) for VoIP traffic during a vertical handoff. The main factors affecting the jitter during a vertical handoff for the considered VoIP data flow are the voice packet payload length (dependent on the codec used) and the downlink data flow rate. As the graph in Fig. 14 indicates, the jitter is 47 ms for the case of a single VoIP session. This is also a particularly encouraging observation. Since a jitter rate below 50 ms is expected to provide acceptable voice quality in real-time VoIP applications, the above readings indicate the likelihood of maintaining satisfactory (if not seamless) levels of VoIP communications during the vertical handoff process [17]. However, as the numbers of simultaneous handoffs increases the jitter rate tends to increase rapidly and way beyond the 50 ms limit. The reasons for this behaviour are currently under investigation.

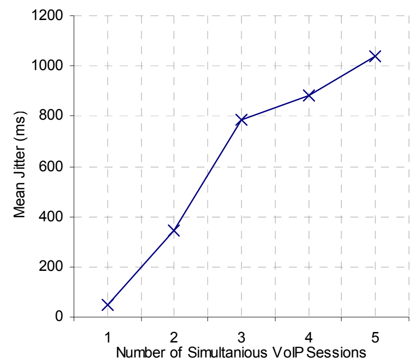

Fig. 14. Mean Jitter during Vertical Handoff.

VI. CONCLUSION

An internetworking model for WLAN and UMTS networks with the 3GPP's IMS acting as an arbitrator was analyzed in this paper. The analytical modeling investigated vertical handoff performance measures such as delay, transient packet loss, and signaling overhead/cost. The derived results indicated that the application layer based additional IMS processing latencies are capable of imposing a substantial impact on the overall performance of the vertical handoff process. Furthermore, it also indicated that mere increase in the wireless link bandwidth does not always help to decrease the vertical handoff delays and reduce transient packet loss. The analysis on the effects of mobility management signaling cost/overhead on CMR indicated that more signaling overhead was incurred during the vertical handoff than at the time of the initial session setup. Lastly an OPNET based network simulation model was used for verification of the analytical model and results.

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