

Network MObility (NEMO) support in Interworking Heterogeneous Mobile Networks

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Abstract—With the expansion of the Next Generation Mobile Network (NGMN), ubiquitous service access is increasingly becoming a reality. As a result, an increasing number of users are accessing applications and services while travelling, especially on public transportation systems, thus giving rise to group mobility scenarios. Therefore, it is essential for the NGMN to be equipped with group mobility management support for achieving seamless session handoffs for a group of users or devices moving together. For a group of members moving between two networks, location transition management was initially addressed in the Internet Engineering Task Force's (IETF) Network MObility (NEMO) basic support protocol. However, the two networks in this case are homogeneous in nature. This paper proposes the integration of NEMO support for an NGMN architecture for enabling group mobility management between multiple heterogeneous networks. Results and analysis illustrate that by integrating NEMO support to an NGMN reduces handoff latency, transient packet loss, jitter, and signaling cost for both end users and service providers.

Keywords—NEMO; Network Mobility; MIP-NEMO; SIP-NEMO; IMS; UMTS; CDMA2000; WiMAX; SIP; Mobile IP

I. INTRODUCTION

The expansion of the Next Generation Mobile Network (NGMN) has broadened the horizons of ubiquitous service access. As a result, there is a gradually increasing trend for applications and services for being accessed by commuters travelling on public transportation systems such as busses, trains, ships, and aircraft, leading to group mobility scenarios. Therefore, a group of users moving from A to B on some sort of a transportation carriage can be defined as network mobility where the moving network is called a mobile network [1]. All devices on board the vehicle, irrespective of their capabilities, will be able to achieve global connectivity via a special gateway router installed in the vehicle. As the vehicle travels passing various heterogeneous mobile networks and domains, the mobile router will be responsible for provisioning uninterrupted connectivity for the devices attached to the mobile network.

Therefore, it is essential for efficient group session handoff mechanisms to be in place for collectively handling seamless service access for the nodes of a mobile network travelling through these heterogeneous networks of an NGMN. For a

group of members moving between two similar networks, location transition management was initially addressed in the Internet Engineering Task Force's (IETF) Network MObility (NEMO) basic support protocol by defining an extension to Mobile IP v6 (MIPv6) [2]. The main drawback of MIPv6-NEMO is that all traffic, to and from the mobile network must pass through a bi-directional tunnel, which results in sub-optimal routing [3]. Therefore, in light of alleviating the problems of SIP-NEMO, a Session Initiation Protocol (SIP) based version of NEMO is proposed [4]. The main difference between the MIP and SIP versions of NEMO is that the latter shifts network mobility management to the Application Layer.

In order to integrate network mobility support for interworked heterogeneous networks, SIP-NEMO protocol must be incorporated into the NGMN. To the best of our knowledge, the only noteworthy contribution published in this regards is [5]. Although, the proposed loose and tight coupling architectures in [5] are capable of vertical session handoffs, they fail to guarantee seamless session continuity [6]; hence the motivation for this work. Therefore, this article proposes a novel network mobility management architecture for collectively handling seamless session handoffs for the nodes of a mobile network travelling through an NGMN of interworked heterogeneous wireless and cellular networks by using the SIP-NEMO protocol. The basic NGMN architecture used for the implementation of SIP-NEMO is based on an authors' previous contribution [7]. The inclusion of SIP-NEMO for facilitating group based session mobility support introduces a significant reduction in signaling overhead, hence substantially increases the end user quality-of-service (QoS).

The remainder of this paper is organized as follows. Firstly the concepts of SIP-NEMO assisted group mobility are explained. Next the interworking platform and the integration of SIP-NEMO to the NGMN is presented. Thereafter the simulation results and validation are presented prior to the concluding remarks.

II. SIP-NEMO ASSISTED GROUP MOBILITY

Two main solutions have been proposed at different layers for managing the mobility of a moving network and for assuring seamless connectivity for the members of the entire group [8]. The first of its kind, as defined by the IETF, is MIPv6-NEMO, which operates at the Network Layer [2]. Although the NEMO protocol is capable of reducing the signaling cost related to

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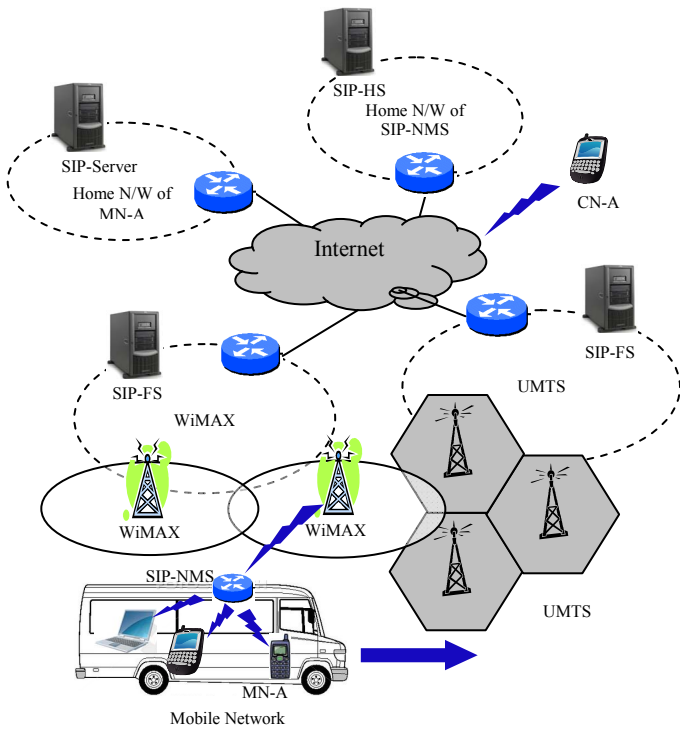


Fig. 1. SIP-NEMO Assisted Mobility Management Architecture.

location updates and the complexity of handoffs; the use of MIPv6 also has some limitations such as sub-optimal routing, increased path lengths, and packet header overheads due to bi-directional tunneling [3]. In order to overcome these drawbacks of MIPv6 based NEMO, a SIP based network mobility management scheme called the SIP-NEMO is proposed at the Application Layer [4].

As illustrated in Fig. 1, since the SIP-NEMO protocol is an extension of the basic SIP protocol, SIP clients can easily roam and operate within a SIP-NEMO environment. The main element of the SIP-NEMO architecture is a SIP Network Mobility Management Server (SIP-NMS), which roams with the mobile network. Each SIP-NMS has one corresponding SIP Home Server (SIP-HS). As the mobile network roams from one network (say, 3G cellular) to another (say, WiMAX), the on board SIP-NMS negotiates a new point of attachment via a SIP Foreign Server (SIP-FS).

Next, the SIP-NMS recovers all ongoing sessions and informs the SIP-HS about its new location via the new point of attachment. Therefore, the SIP-NMS is also considered as the gateway of a mobile network. It is designed to keep all attached nodes globally reachable and to maintain existing sessions as the mobile network changes its point of attachment. Therefore, the attached nodes are not aware of the mobile networks changing of its point of attachment or moving from one subnet to another.

Each SIP-NMS has a corresponding SIP-HS, which takes care of the following tasks; accepting the registration from the SIP-NMS, recording the current location of the registered SIP-NMS, and forwarding requests to corresponding SIP-NMSs.

The SIP-FS is informed of its new location (point of attachment) as and when a SIP-NMS attaches to a new network. Therefore, when a user attached to a particular roaming mobile network needs to be contacted; the corresponding SIP-HS is initially queried, which subsequently forwards this request to the relevant SIP-NMS of the targeted mobile network.

The SIP-FS is fundamentally a SIP Back-to-Back User Agent (B2BUA) on a foreign network. However, the SIP-FS plays an important role by providing a Uniform Resource Identifier (URI)-list service, which reduces signaling cost and handoff delay as a result of optimized routing [9]. As the mobile network roams, its on-board SIP-NMS combines the source and destination addresses of all its ongoing SIP sessions into a URI list. This URI list is embedded into the SIP ReINVITE request and forwarded to the SIP-FS of the newly attached network. When the SIP-FS receives the SIP ReINVITE request with a URI list, it individually generates SIP ReINVITE requests for all sessions that need to be handed-off to the new network.

III. INTEGRATING SIP-NEMO INTO THE NGMN

Prior to presenting details of the proposed integration of SIP-NEMO to the NGMN, an overview of the underlying NGMN platform for interworking heterogeneous wireless and cellular networks is presented.

A. Interworking Architecture for Heterogeneous Networks

The interworking platform used in this analysis is illustrated in Fig. 2. Interested readers may refer to [7] for more specific and detailed information on its design. One of the primary design considerations of this architecture is that all networks are loosely coupled for data routing and tightly coupled via the IP Multimedia Subsystem (IMS) for session control signaling [10]. Therefore, session mobility management is facilitated via the IMS at the Application Layer. In order to

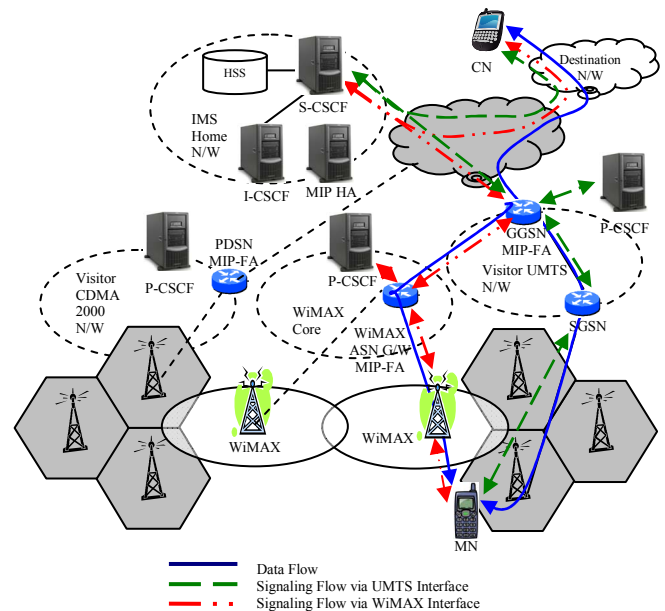


Fig. 2. Interworking Architecture for Heterogeneous Networks.

guarantee terminal mobility, MIPv4 is implemented at the Network Layer.

When a Mobile Node (MN) roams between a WiMAX and a UMTS network, inter-network roaming takes place. The message flow relating to inter-network roaming (i.e., for a vertical handoff) from WiMAX to UMTS is illustrated in Fig. 3. Firstly the standard UMTS attach and PDP context activation procedures are performed by the MN. Once the system acquisition is done, the next step is to setup the data pipeline. The MN's IP address acquisition is initiated by sending MIP registration request to its MIP Home Agent (MIP-HA) via the MIP-Foreign Agent (MIP-FA), which is the Gateway GPRS Support Node (GGSN). This mechanism is based on the specifications given under [11]. Followed by this is the exchanging of the MIP Binding Update message between the MN and the Corresponding Node (CN) for avoiding triangular routing [12].

The next stage is where the IMS-SIP session handoff procedures take place. This requires sending a SIP Re-INVITE (with the Call-ID and other identifiers corresponding to the ongoing session) to the destination. Followed by this comes the resource/precondition reservation for the UMTS interface. This is important because some clients require certain preconditions (that is, QoS levels) to be met before establishing a session. Next, the CN sends a 183 Session Progress response. A PRecondition ACKnowledgment (PRACK) request follows afterwards as an acknowledgement for the 183 Session Progress response. Then a 200 OK response is generated by the CN to the MN. This is followed by an UPDATE confirmation sent to the source, in which the MN confirms that the resources are reserved at its local segment. Once the destination receives an UPDATE request, it generates a 200 OK response. Now the MN is ready for proceeding with the session flow via the new interface. It is important to note that until such time the new

data flow is initiated via the UMTS interface, the old flow via the WiMAX interface remains active. Finally, the resources of the WiMAX session is liberated through a SIP BYE and 200 OK message exchange. Thus the model follows the make-before-break handoff mechanism. Inter-system roaming from UMTS-to-WiMAX can also take place in a similar manner.

B. SIP-NEMO Integrated Interworking Architecture

As previously mentioned, one of the primary design considerations of our interworking architecture is that, both the coupling mediator and the session mobility manager is the IMS. Further, SIP is the main signaling protocol used by the IMS for session mobility management. Therefore, with the above NGMN architecture, session mobility management for a group of users (or a mobile network) can successfully be achieved by introducing SIP-NEMO into the IMS. Preliminary works on the possibilities of integrating SIP-NEMO into the IMS are outlined in [5]. Further, the authors of [5] propose two architectures for IMS assisted group mobility. Nevertheless, neither of them is capable of collectively handling seamless service access for the nodes of a mobile network travelling through heterogeneous networks in an NGMN. Therefore, this section explains how SIP-NEMO is integrated into the NGMN architecture for successfully achieving network mobility.

Figure 4 illustrates the proposed SIP-NEMO integrated NGMN architecture. The integration of SIP-NEMO to the IMS framework is planned in such a way that it complies with the existing 3GPP-IMS and 3GPP2-IMS standards. Therefore, the B2BUAs in each network, which are also Proxy-Call Session Control Functions (P-CSCF) to the IMS, are modified for handling the URI list service. With this modification, the P-CSCF can now function as a SIP-FS to a SIP-NMS that may

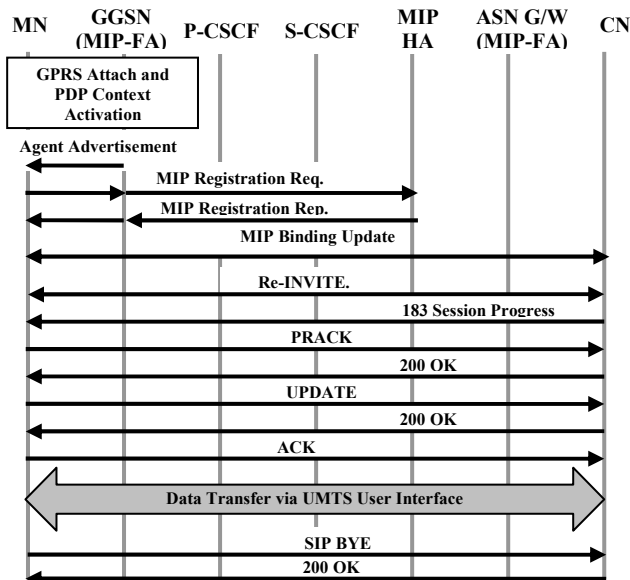


Fig. 3. WiMAX to UMTS Single User Session Handoff.

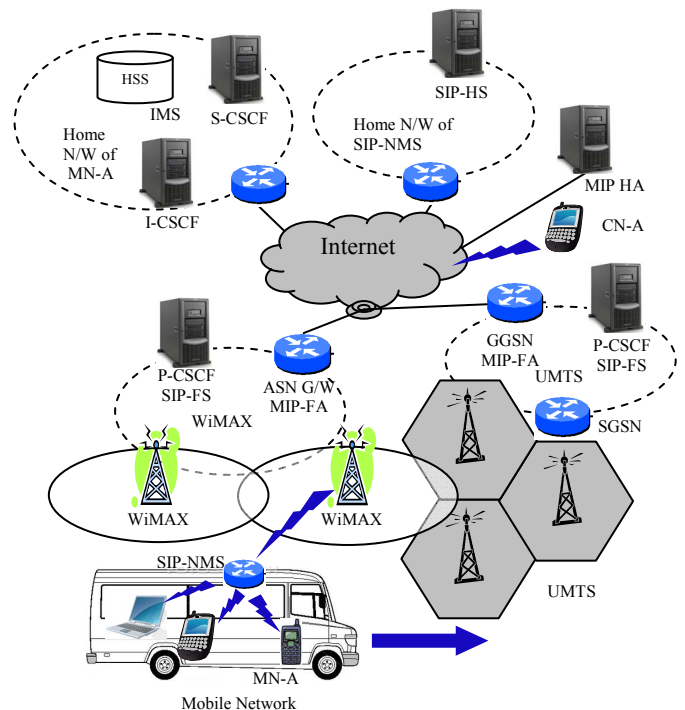


Fig. 4. Proposed SIP-NEMO Integrated Interworking Architecture.

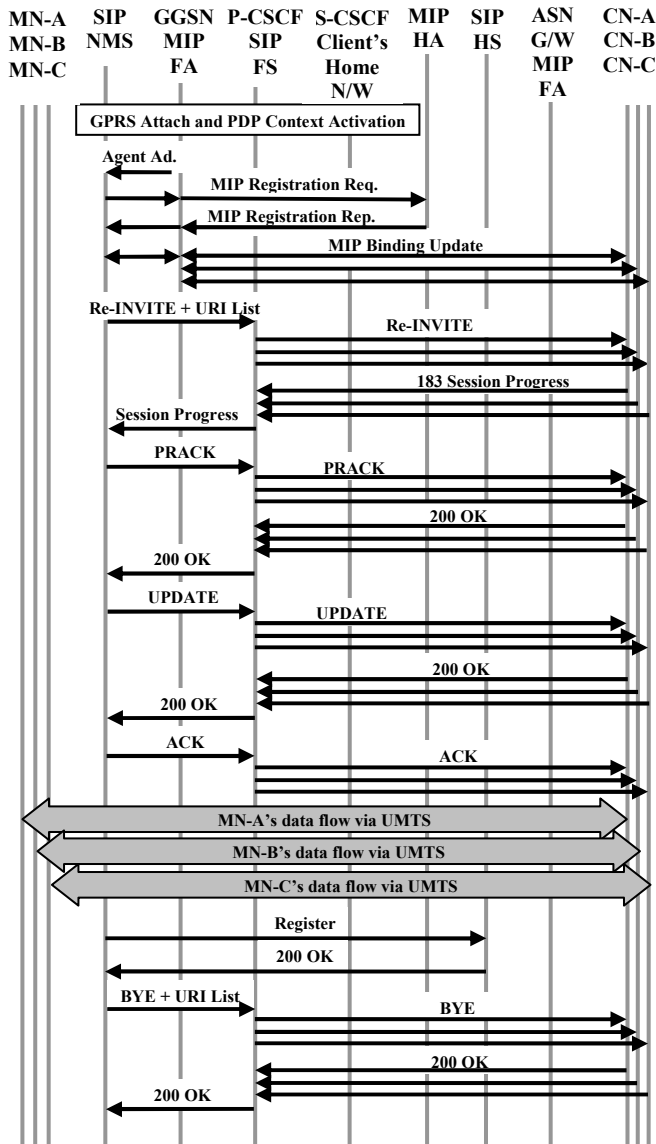


Fig. 5. WiMAX to UMTS Group Session Handoff.

roam into its network, thus easily be able to handle network mobility. Therefore, the SIP-NMS acts as a Network Address Translator (NAT) for the mobile network's clients trying to connect to the IMS. Then again, the SIP-NMS also appears as a roaming SIP user from the SIP-FS's point of view. Each SIP-NMS has its corresponding SIP-HS.

When a new SIP client (say, MN-A) joins the mobile network, the following registration sequence takes place. MN-A first sends a REGISTER request to the SIP-NMS. Next, the SIP-NMS changes the CONTACT field in the SIP header from the MN-A's address to the SIP-NMS's URI address as a part of its NAT mechanism. Followed by this, the SIP-NMS forwards the REGISTER request to the P-CSCF, which is also the SIP-FS in this case. The P-CSCF examines the SIP header to identify the entry point to the MN-A's home domain. Next, the SIP-FS forwards the REGISTER request to the MN-A's (IMS)

home network. The registration process in the IMS involves the Serving-CSCF (S-CSCF) and the Home Subscriber Server (HSS). A copy of the SIP-NMS's address is stored at the HSS as MN-A's contact address. Finally the S-CSCF of the IMS replies with a 200 OK message to the MN-A via the SIP-NMS.

As the mobile network, which consists of onboard clients roams from WiMAX to UMTS, the SIP-NMS must first recover its global reachability. Therefore, the SIP-NMS must obtain an IP address for its UMTS interface and ReINVITE all

ongoing sessions via the UMTS network. Fig. 5 illustrates session handoff. Similar to the previous explanation, firstly the standard UMTS attach and PDP context activation procedures are performed by the SIP-NMS. The actual IP address allocation for the SIP-NMS is initiated by sending the MIP registration request to its MIP-HA via its MIP-FA, which is the GGSN in this case. Followed by this is the exchanging of a MIP Binding Update message between the SIP-NMS and its existing SIP destination clients for avoiding triangular routing [12].

Next the SIP-NMS sends a ReINVITE request with a URI list to the P-CSCF, which is also the SIP-FS, in order to recall the ongoing SIP sessions. Then the P-CSCF reproduces individual INVITE messages for all CNs. After each SIP CN receives the ReINVITE request according to the URI list, resource/precondition reservations for the UMTS interface will initiate for each of these sessions. Individual precondition negotiation is vital since some clients/applications require certain preconditions (that is, QoS levels) to be met before handing-off a session. As illustrated in Fig. 5, all precondition reservation signaling takes place via the P-CSCF/SIP-FS. The point to note here is that, the SIP-FS combines all responses received by destination clients in the URI list to a single message and forwards it to the SIP-NMS. Hence the session handoff signaling overhead is considerably reduced. After the precondition reservations, each CN will resume the ongoing sessions via the new contact address of the SIP-NMS.

The point to note in this case is that, our proposed solution is based on a make-before-break handoff, which is therefore capable of assuring seamless session handoffs for the SIP-NMS clients, unlike the solution given in [5]. Next, a REGISTER request is sent to the SIP-HS by the SIP-NMS. The reason being that, as the SIP-NMS attached mobile network roams into the UMTS network, the new location information (i.e., the IP address) must be updated with its SIP-HS. Note that this does not involve the IMS since it is handled by the NEMO architecture. Once the SIP-HS responds with a 200 OK message, the SIP-NMS attached mobile network will be globally reachable (via the UMTS network). Finally, the breaking of the session via the WiMAX network takes place by using the URI list service.

IV. RESULTS AND VALIDATION

A. Simulation Platform

In order to investigate the performance of the presented architecture, simulations are performed using the OPNET Modeler 14.0 platform. A fully functional SIP-IMS model for OPNET is constructed and integrated to the OPNET's existing

UMTS Special Module, which is currently available under the contributed models library of the OPNET University Program [13]. Modifications are made for SIP Proxy Servers (UASs) to function as different CSCFs, UAC processes to communicate with modified UASs, IMS-SIP based messaging to flow between CSCFs, introduce roaming facility between multiple domains, and introduce process delay controls (i.e. for messages sent between CSCFs and the HSS queries). As a result, a UMTS network that is fully capable of using IMS based SIP signaling for session management is developed. Furthermore, below the IMS architecture, a MIP v4 framework is also constructed for providing IP mobility.

The next task is creating a heterogeneous network with MIP and SIP signaling as illustrated in Fig. 4. Since IMS and MIP protocols are implemented at the core network, their behavior can be considered to be independent to the underlying Physical and Link Layers. Taking the facts and limitations of OPNET simulator into consideration, an all-IP heterogeneous test bed is created by interworking a UMTS network with a WiMAX network. The reader is referred to [14] for specific details of this simulation platform. Next, a mobile network consisting of a group of SIP UACs are created. This group is connected via a mobile SIP B2BUA, which acts as the SIP-NMS. The SIP-NMS and the P-CSCF/SP-FS are capable of handling SIP URI list services. Firstly, the on-board SIP-NMS combines the source and destination addresses of all its ongoing SIP sessions into a URI list. Next, this URI list is embedded into the SIP ReINVITE request and sent to the SIP-FS of the newly attached network. When the SIP-FS receives the SIP ReINVITE request with a URI list, it individually generates SIP ReINVITE requests for all the ongoing sessions to be handed-off to the new network. This platform is used for simulating and studying SIP-NEMO assisted vertical handoff delay and transient packet loss for SIP sessions being handed-off from WiMAX-to-UMTS. The results are compared against a non SIP-NEMO scenario.

B. Simulation Results

The average vertical handoff delays for SIP clients roaming from WiMAX-to-UMTS and UMTS-to-WiMAX are illustrated by Fig. 6 and Fig. 7 respectively. Regardless of being onboard a SIP-NEMO supported mobile network or otherwise, the average vertical handoff delay for a single session from WiMAX-to-UMTS is 192 ms and the same from UMTS-to-WiMAX is 174 ms. The reason for both the SIM-NEMO and the normal handoff method to incur equal delays in this case is due to the SIP URI list containing a single entry where the SIP-NMS merely works as a SIP proxy. Also, the reason for WiMAX-to-UMTS to show a relatively higher handoff delay is due to the relatively low bandwidth and the complicated structure of the UMTS Terrestrial Radio Access Network (UTRAN).

As the number of SIP clients onboard the SIP-NEMO supported mobile network starts to increase, benefits of our proposed method become more apparent. For example, as Fig.6 illustrates, when two SIP clients are onboard a mobile network, the SIP-NEMO method reduces the handoff delay by 5 ms in comparison to performing individual handoffs as per [7], which

is actually a reduction by 2.5%. As the number of SIP clients onboard the mobile network increases SIP-NEMO experiences reduced handoff delays. For example, according to Fig. 6, when the number of clients attached to the mobile network becomes 6, the SIP-NEMO framework is capable of reducing the handoff delay up to 16 %. Therefore, due to the reduction in the session handoff message overhead, a diverging pattern is noted between the two corresponding handoff delay graphs for SIP-NEMO and MIP-SIP mechanisms.

Figures 8 and 9 illustrate the normalized transient packet loss against the number of sessions handed-off from WiMAX-to-UMTS and UMTS-to-WiMAX networks respectively (Note: this is in the case of a break-before-make handoff scenario takes place). A relatively higher transient packet loss is observed from the WiMAX-to-UMTS graphs in Fig. 8 in contrast to the UMTS-to-WiMAX graphs in Fig. 9. Since the

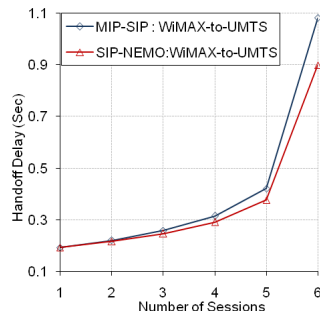


Fig. 6. WiMAX-to-UMTS Session Handoff Delay.

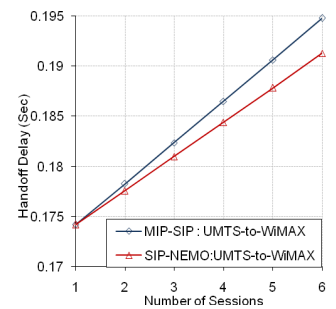


Fig. 7. UMTS-to-WiMAX Session Handoff Delay.

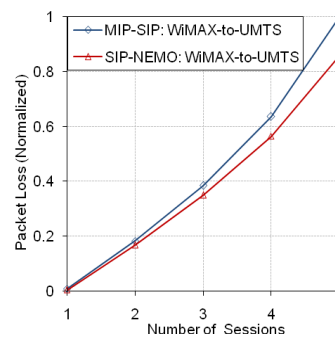


Fig. 8. WiMAX-to-UMTS Packet Loss during Session Handoff.

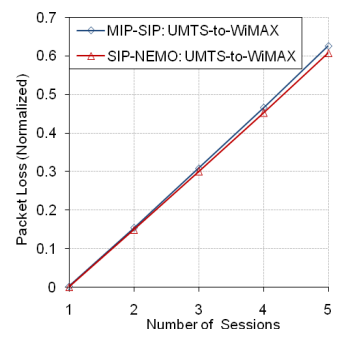


Fig. 9. UMTS-to-WiMAX Packet Loss during Session Handoff.

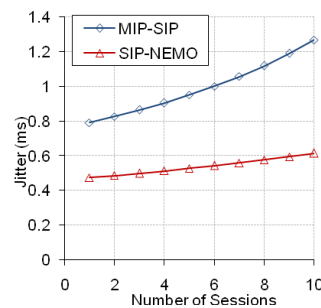


Fig. 10. Jitter Analysis.

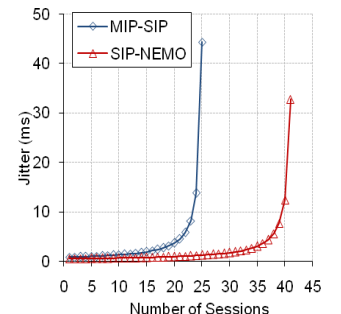


Fig. 11. Jitter Analysis.

transient packet loss during a vertical handoff is directly proportional to the vertical handoff delay, SIP-NEMO assisted session handoffs incur relatively lower packet loss in contrast to the results obtained from [7].

Lastly, Fig. 10 and Fig. 11 illustrate the jitter comparison for WiMAX-to-UMTS vertical handoffs for MIP-SIP and SIP-NEMO mechanisms. The jitter plot is obtained by calculating the variation of the end-to-end delay over UMTS for sessions that are being handed-off. According to the graphs in Fig. 10, the jitter rates are within acceptable limits for VoIP applications. However, the interesting fact is that, these jitter graphs tend to indicate rather exponential curves. This could well be confirmed from Fig. 11, which illustrates jitter values for an extended number of session handoffs. The graphs illustrating the jitter also confirm that the SIP-NEMO based group session management method is capable of successfully supporting a relatively higher number of simultaneous SIP session handoffs, whilst providing acceptable jitter rates.

CONCLUSIONS

This paper proposed a novel network mobility management architecture for collectively handling seamless session handoffs for the nodes of a mobile network travelling through an NGMN of interworked heterogeneous wireless and cellular networks by integrating the SIP-NEMO protocol. The integrating was achieved by introducing SIP-NEMO into the IMS, which was the main coupling mediator of the underlying NGMN architecture. According to the proposal, the P-CSCF of the IMS functioned as the SIP-FS of the SIP-NMS, thus easily being able to handle group mobility. Since both the P-CSCF and the SIP-FS are SIP B2BUAs, this was possible with minimal changes to the existing 3GPP and 3GPP2 standards. The inclusion of SIP-NEMO for facilitating group based session mobility support introduced a significant reduction of signaling overhead, hence substantially increased the end user QoS. Results and analysis illustrated that by integrating NEMO support to an NGMN reduced handoff latency, transient packet loss, jitter and signaling cost for both end users and service providers.

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REFERENCES

- [1] H. Y. Lach, C. Janneteau, and A. Petrescu, "Network mobility in beyond-3G systems," *IEEE Communications Magazine* vol. 41, no.7, pp. 52-57, Jul. 2003.
- [2] V. Devarapalli, et al., "Network Mobility Basic Support Protocol," IETF RFC 3963, Jan. 2005.
- [3] E. Perera, V. Sivaraman, and A. Seneviratne, "A Survey on Network Mobility Support," *Mobile Computing and Communications Review*, vol. 8, 2004.
- [4] C. M. Huang, C. H. Lee, and J. R. Zheng, "A novel SIP based route optimization for network mobility," *IEEE Journal in Selected Areas in Communications*, vol. 24, no. 9, pp. 1682-1691, Sept. 2006.
- [5] W. K. Chiang, A. N. Ren, and Y. C. Chung, "Integrating SIP-based Network Mobility into IP Multimedia Subsystem," in *Proceedings of the IEEE Wireless Communications and Networking*, Budapest, Hungary, Apr. 2009.
- [6] V. K. Varma, et al., "Mobility management in integrated UMTS/WLAN networks," in *Proceedings of the IEEE International Conference on Communications*, Anchorage, Alaska, May 2003.
- [7] K. S. Munasinghe and A. Jamalipour, "A Unified Mobility and Session Management Platform for Next Generation Mobile Networks," in *Proceedings of the IEEE Globecom*, Washington DC, Nov. 2007.
- [8] C. M. Huang, C. H. Lee, and P. H. Tseng, "Multihomed SIP-based Network Mobility Using IEEE 802.21 Media Independent Handover," in *Proceedings of the IEEE Communications Conference*, Dresden, Germany, Jun. 2009.
- [9] G. Camarillo and A. Roach, "Framework and Security Considerations for Session Initiation Protocol (SIP) Uniform Resource Identifier (URI)-List Services," draft-ietf-sipping-uri-services-07, Nov. 2007.
- [10] 3GPP, "IP Multimedia Subsystem (IMS)," 3GPP TS 23.228 Version 6.10.0 Release 6, 2005.
- [11] 3GPP, "Combined GSM and Mobile IP Mobility Handling in UMTS IP CN," 3GPP TR 23.923 v 3.0.0, 2000.
- [12] C. Perkins, "IP Mobility Support for IPv4," IETF RFC 3344, 2002.
- [13] A. H. Enrique Vazquez, Jose Ignacio Fernandez, "SIP-IMS Model for OPNET Modeler," OPNET University Program Contributed Models, 2005.
- [14] K. S. Munasinghe and A. Jamalipour, "A 3GPP-IMS based Approach for Converging Next Generation Mobile Data Networks," in *Proceedings of the IEEE International Conference on Communications*, Glasgow, UK, Jun. 2007.