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The Impact of IMS Mobility Management with Google Mobile Phone on Quality of Experience

***Nadia Awad*[1]*, *Musa H. Wali*[2]**

[1] *Medical Devices Techniques Department, Babylon Technical Institute-Al-Furat Al-Awsat, Technical University, Iraq*

[2] *Electronic and Communication Department, Faculty of Engineering, University of Al-Qadisiyah, Iraq*

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The widespread of a Wireless Local Area Network (WLAN) access network in residential areas provi low-cost and high-speed access to the Third-Generation Partnership Project (3GPP) services for end us This has virtually made WLAN to be a complementary access network to Universal Mo Telecommunications System (UMTS). The interoperability of WLAN and UMTS is made possible v the use of a mobility management mechanism in IP Multimedia Subsystems (IMS) architecture. Cur problems in mobility management mechanisms are found in device mobility such as mobile phones. S problems affect several layers and disrupt services. In this paper, IMS mobility management mechani with G1 mobile phone as an IMS client testbed is implemented, and its impact on the Quality of Experie (QoE) during end-to-end VoIP call session and its conformity to the 3GPP requirements of seamless ser continuity is evaluated. The QoE driven mobility management mechanism is then proposed and evalua to improve the QoE and to meet the 3GPP requirements of seamless service continuity during the hando between UMTS and WLAN access networks. Preliminary results show that there is an improvemen QoE and the requirements of 3GPP service continuity were met. The proposed Quality of Service (Q driven mobility management mechanism will help in the development of dual-mode mobile handsets v QoE adaptability.

1. Introduction

Many radio access technologies in mobile devices have supplied end users with multiple choices of access technology to use in a particular scenario. The choice of access technology is based on the low cost and its QoS. The two main factors that motivate the handover mechanism in a scenario whereby an ongoing VoIP session under UMTS access network discovers a WLAN access point are the cheap rate and the high-speed connection which can be obtained by using WLAN access networks [1]. In spite of the small area coverage of WLAN compared to UMTS, the handover from video streaming applications. Minimizing signaling and controlling message overheads, improving low handover latency, having virtually zero packet loss rate and minimal disruption of services during the handover period are the most challenging research areas to be considered at designing and implementing phases of handover management mechanism between UMTS and WLAN access networks. However,

* *Corresponding author.*
E-mail address: nadia.al-khalidi.iba@atu.edu.iq (Nadia Awad)

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meeting 3GPP requirements of seamless service continuity during the handover process [3] [4] and hence improving the QoE is the overall target. The QoE in VoIP, also known as Perceived Quality of Service (PQoS), is a subjective measure that takes into account what the end user is experiencing with the provided VoIP services. Mean opinion score is used to measure QoE, which is a correlation between what the end user expects to hear/see and what that user actually hears/sees. The QoE degradations are obstacles to VoIP service providers, as they increase technical support budget, customer churn rate, and lower revenue streams.

Most of the ongoing research on mobility management in IMS is mainly based on simulation studies focusing on interworking between UMTS and WLAN, improving SIP performance and protocol development. It was demonstrated in [5] that the flow of SIP messages during the changeover process in the IMS core network corresponds with the well-known central server queuing network model. This was done through performance modeling and evaluation of the IMS mobility management mechanism. Simulation studies in [6] have quantified the maximum number of UMTS subscribers that can be admitted to WLAN subject to maintaining the level UMTS (QoS) and respecting WLAN policies. In [7], IMS support in inter-worked UMTS and WLAN was discussed. Also, mobile IP based solution by using cross-layer signaling was described. Performance results showed that the handover delay was varied greatly. An architecture for end-to-end QoS control in a wireless and wired environment with effective QoS translation was discussed [2]. DiffServ and RSVP architectures together with UMTS/WLAN interworking and IEEE 802.11e were used. The results showed that the combination of IEEE 802.11e and RSVP kept delays smaller than DiffServ core. In contrast, this paper goes beyond the current research trend by implementing a real-world testbed and proposing QoE driven mobility management mechanism in order to improve the QoE and to meet the 3GPP requirements of seamless service continuity during the handover between UMTS and WLAN access networks.

As an IMS client, the G1 mobile phone was built with the Android Software Development Kit (SDK), which has a bright future for UMTS access networks. The Android SDK kit was offered by the alliance (open handset alliance) for the new application's purpose.

In this paper, IMS-based testbed is implemented in order to evaluate the impact of its mobility management mechanism on QoE during end-to-end VoIP call sessions. The G1 mobile phone is used as an IMS client mobility device. The conformity to the 3GPP requirements of seamless service continuity is also evaluated. QoE driven mobility management mechanism is proposed and evaluated in order to improve the QoE and to meet the 3GPP requirements of seamless service continuity during the handover between UMTS and WLAN access networks. The rest of the paper is organized as follows, Section 2 describes the testbed developed for this evaluation, this will be followed by handover scenario in Section 3. QoE evaluation and experimental results are detailed in Section 4. QoE driven mobility management mechanism in order to improve the QoE and to meet the 3GPP requirements of seamless service continuity is presented in Section 5, Conclusion and future work are described in Section 6.

2. Experimental Testbed

The testbed has an Open IMS Core [8] for RTP session establishment, handover and termination.Fig.1 shows the overall testbed that developed to find the impact of IMS mobility management mechanism with G1 mobile phone on QoE and its conformity to 3GPP seamless service continuity. The G1 mobile phone [9] was used as an IMS client for UMTS and WLAN access networks. On the other end of the LAN connection, an open-source IMS-Communicator [10] is used as an IMS client. IMS-Communicator is implemented in JAVA and its SIP [11] stack is built on top of the JAIN-SIP [12]. The testbed uses Hutchison 3G UK Limited for UMTS access. NETGEAR wireless-G Router WGR614 v9 provides the WLAN access point. Mobility Management Application Serve (MM-AS) was developed on top of the JAIN-SIP and deployed for service continuity and acted as Back-to-Back User Agent (B2BUA).

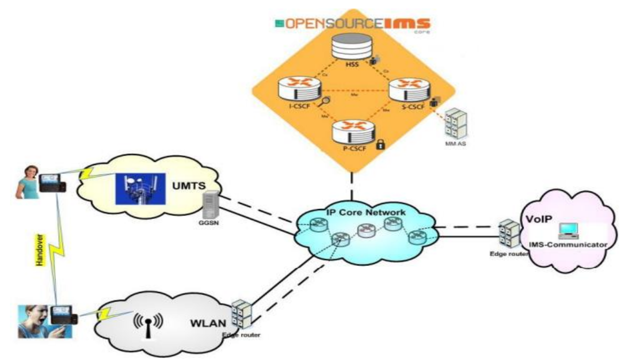

Figure 1. Overview of the IMS testbed components.

2.1. Open IMS Core

The testbed was based on the Next Generation Network (NGN) IMS architecture. The goal of IMS is to bring together mobile, wireless, and fixed broadband data networks into a single network architecture where all forms of data transmission are hosted in all IP environments utilizing the SIP protocol (SIP). Since Open IMS core [1] is an open-source IMS implementation, it is used. It contains the fundamental components of any IMS architecture described by 3GPP, 3GPP2, ETSI TISPAN, and PacketCable. The testbed can manage the handover mechanism through IMS architecture where the handover control messages can be exchanged by using SIP and Session Description Protocol (SDP) [13].

2.2 G1 Mobile Phone.

G1 mobile phone was used in the experimental setup because it is built with Android SDK [14]. The Android platform is an open software stack that includes an operating system, middleware, and important apps for mobile devices. The application framework APIs used by the core applications are completely accessible to developers. The application architecture is created to make component reuse simple. Each program can share its capabilities, which can then be used by other applications (subject to security constraints enforced by the framework). The user can change components thanks to the same technique. Developers can create and own Android applications without requesting Google's permission, which is another benefit of the Android platform. The G1 mobile phone may also join WLAN, EDGE, UMTS, and Bluetooth access networks. For both UMTS and WLAN access networks in this testbed, G1 mobile is utilized.

Android is ported using SIP and RTP stacks in order to be compatible with IMS architecture. A light weight MjSip [16] for mobile apps is the foundation of the SIP and RTP stacks for Android that HSC [15] has made available. The UMTS and WLAN signal strength monitoring functions for the SIP and RTP stack have been added in order to activate the handover mechanism. These functions were developed as JAVA methods (WLAN-Probe and UMTS-Probe) under a JAVA class named Handover Module (HM). WLAN and UMTS probes methods interact with the native

WifiManager class which provides the primary API for managing all aspects of WLAN connectivity and TelephonyManager class which provides access to information about the telephony services such as UMTS connectivity on the device. Fig. 2 shows the ported RTP and SIP stacks. The principal entry point for all SIP methods, including INVITE, ACK, CANCEL, BYE, MESSAGE, PRACK, and UPDATE, is the SIP Manager. For the purpose of sending and receiving handover alarms using the SIP reInvite technique, the SIP Manager communicates with HM, IMS CSCFs, and MM-AS.

SIP Manager creates a SIP listener interface that manages incoming and outgoing SIP messages and offers wrapping of the SIP stack's functions. After acknowledging the INVITE methods, the SIP Manager works with the Media Manager to start audio RTP sessions. The AV Rev/Trans bloc, which combines RTCP, RTSP, and RTP, is used by the Media Manager to receive and transmit RTP sessions. Android SDK 1.5 release 2 under the Eclipse IDE plugin with Android Development Tools (ADT) were used to perform the development under Android SDK platform.

At the moment G1 mobile phone only provides access to encode and decode PCM audio format for RTP session (see AudioRecord class). Due to this limitation only Pulse Code Modulation (PCM) audio format is used throughout our testbed to demonstrate the handover mechanism.

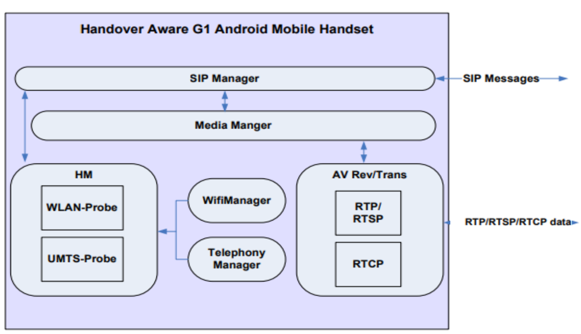

Figure 2. G1 Android Platform Architecture with HM, RTP and SIP Stacks

2.3 Mobility Management Application Server.

A SIP based MM-AS was developed as B2BUA and deployed into Cipango [17] application server. Cipango is an open-source JAVA platform for developing, deploying and integrating any SIP application. MM-AS is responsible for communicating SIP messages amongst the IMS clients in order to carry out handover operations. The developed MMAS supports IP Multimedia Service Control (ISC) interface according to 3GPP. MM-AS ignores any SIP headers that does not understand according to RFC 3262 [18]. When acting as B2BUA, MM-AS copies the remaining Route headers unchanged from received SIP Invite request to the new SIP Invite request as defined in 3GPP standard [19] for a B2BUA. MM-AS must know each and every ongoing RTP session and registration for future handover mechanism.

3. Handover Scenario

The G1 Mobile phone is subjected to disruption of service continuity when moving from one access network to another during an active communication session. During the handover, as the mobile switches its point of attachment in the network, the mobile terminal may end up interacting utilizing its second interface in the new network. Based on the type of movement and access network, the handover is primarily categorized as Inter-subnet and Intra-subnet. Due to space limitation, this paper demonstrates the inter-subnet handover which is further classified as inter-technology handover, i.e., the handover between the UMTS and WLAN and vice versa. The goal of this scenario is to evaluate and investigate the impact of mobility on QoE under IMS mobility management with Android G1 mobile phone. Application service provided under this scenario is the end-to-end voice communication. The mobile has IMS subscription with its user profile and service information in the Home Subscriber Server (HSS) of the IMS network and always establish and terminate session via the IMS.

Fig. 3 illustrates the handover scenario from WLAN to UMTS for an ongoing voice communication. Then the mobile performs an origination handover from WLAN to UMTS. The result is the voice communication over UMTS.

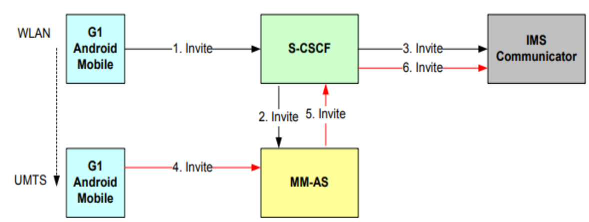

Figure 3. Originating handover from WLAN to UMTS

Before the handover occurs, a voice session is established over WLAN access network. This is illustrated in the steps 1-3 in Fig. 3. The handover is initiated by the mobile (step 4 in Fig.3). This may be based on measurement of the WLAN signal strength in the mobile that triggers the handover procedure when the signal strength becomes weaker.

When the MM-AS receives the call setup request in step 4 in Fig. 3, it knows that a handover is required. This is because the MM-AS already has an ongoing session from the mobile, and knows therefore that the mobile wants to move from WLAN to UMTS. To perform the handover, the MM-AS sends a SIP reInvite message to the IMS-Communicator. The reInvite traverses along the SIP signalling path of the existing voice session. It carries the session description protocol with the IP address and port information used for this voice session. When the SIP reInvite arrives at the IMS-Communicator, it will redirect the RTP traffic to the newly provided IP address and port information. Finally, the MM-AS closes the WLAN based call leg by sending a SIP Bye message via the S-CSCF to the mobile. This completes the handover procedure.

The originating handover from UMTS to WLAN for ongoing voice session over UMTS takes the same path as illustrated in Figure 3. The result is the voice session over WLAN. Similar handover procedures are followed as described above, but this time the handover is from UMTS to WLAN.

4. Results and Discussion

In order to evaluate the impact of the IMS mobility management with G1 Android phone on QoE, the mobility scenario discussed in Section 3 is used.

4.1 QoE Evaluation and Experimental Results

Extensive tests were carried out at the same location using the same WLAN router and G1 mobile phone, measuring MOS values and WLAN's Received Signal Strength Indicator (RSSI). Access to IMS application services and control functions passed through the Local Area Network (LAN). There were several other WLAN access points in range but the G1 mobile phone was only authorized and authenticated to only one WLAN router connected to the testbed. The tests were conducted during work days in the afternoons. The averaged values of relevant measurements were taken in a sample of 30 runs.

The handover mechanism will introduce extra latency Dh in which voice session will be disrupted due to switching of access networks. Packet losses Ph will also be experienced due to access network switching. The latency Dh is comprised of Dd, Da, Dn, and Ds, where Dd is the delay contributed by the process of detecting the handover when one of the access networks' signal strength degrades. Da is the delay caused by the authentication and authorization process when roaming between different administrative domains. Dn is the delay caused by the network layer which includes access network activation and IP address allocation, IP V4 is used in both access networks. Ds is the delay caused by the application layer, this is mainly due to signaling traffic, SIP REGISTER and INVITE requests are issued during the handover process in order to re-establish the media session. The latency Dh is therefore expressed as:

$$D_h = D_d + D_a + D_n + D_s \qquad (1)$$

In the context of the scenario detailed in Section 3, Da = 0, as there is no authentication and authorization process involved because the handover mechanism takes place in the same administrative domain. Furthermore, Ds is only contributed by SIP INVITE requests, this is because the IMS client would have already been registered in the current administrative domain.

The G1 mobile phone does not simultaneously support multiple access networks. The 3G is only activated once WLAN is out of range, and the 3G is only switched off when WLAN is in range. From the experimental tests conducted in the testbed, it takes an average of 5 seconds to switch from one access network to another and to automatically obtain an IP address. This process harms the QoE and service continuity, in 5 seconds the loss is almost 250 RTP packets for PCM audio format. The statistical data were recorded from the time the WLAN interface was off to the time when the UMTS interface was on. Figure 4 depicts QoE degradation MOS values are calculated by Equations (2-5) defined in [20] and [21].

$$R = R_0 - I_d - I_{e-eff} \qquad (2)$$

where R is the rating factor, R0 = 93.2, is the maximum score that can be achieved by codecs, Id is the impairment factor caused by end-to-end delay, Ie−eff is the effective equipment factor which depends on the equipment, packet loss, and packet loss robustness, given by

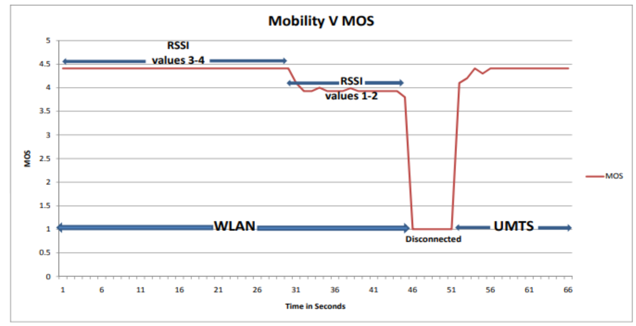

Figure 4. QoE degradation during handover process

$$I_{e-eff} = I_e + (95 - I_e)\frac{P_{pl}}{B_{pl}} \qquad (3)$$

where, Ie is the equipment impairment (i.e., codec quality), Bpl is the packet loss robustness and Ppl is the packet loss rate in percentage. In the case of the packet loss business:

$$I_{e-eff} = I_e + (95 - I_e)\frac{P_{pl}}{fracP_{pl}BurstR + B_{pl}} \qquad (4)$$

where BurstR is the burst ratio defining the ratio of the average length of observed bursts in an arrival sequence to an average length of bursts expected under random loss. Equation 5 calculates MOS values:

$$MOS = \begin{cases} 1, & for\ R \le 0 \\ 1 + 0.035R + R(R-60) * (100-R)7 * 10^{-6} & for\ 0 < R < 100 \\ 4.5, & for\ R \ge 100 \end{cases} \qquad (5)$$

The packet loss rate was monitored through RTCP reports scheduled at 1-second intervals. The severe QoE degradation is clearly seen in Figure 4 from 46 to 50 Seconds where the MOS value is 1. This is due to inactive WLAN and 3G access network interfaces during the handover process from WLAN to UMTS. The tests also showed that there were noticeable QoE degradations even before the WLAN was out of range, this is because the signal strength of WLAN gets weaker with distance from the access point. The RSSI from Android WIFI Manager Class was used, and it was found that there were RTP packet losses at RSSI values between 1 and 2, this is clearly seen in Figure 4 from 30 to 40 seconds where the MOS value is around 3.9. The RSSI value takes integer values from 1 to 4, 1 being the worst and 4 the best.

Dn is a predominant factor in Equation (1) because Dd and Ds are in few milliseconds, the average response time of the reinvite SIP message is 6ms and that of Dd is 30ms. The reinvite response time is calculated from the time the request was initiated to the time the 200 Ok response was received.

4.2 QoE Driven Mobility Management Mechanism

By not supporting simultaneous activation of multiple access networks, G1 mobile phone will not meet the 3GPP requirements of seamless service continuity during the handover process. Letting only one access network at a time is of significant importance in preserving power in any portable device [22]. In this context, there should be a tradeoff between power management and seamless service continuity. In order to improve seamless service continuity and hence QoE, make before-break rule is proposed (c.f., Algorithm 1).

Table 1. Algorithm 1

Algorithm 1 QoE Driven Mobility Management Mechanism

```
timer = Timer(30); /* timer (ms) to get RSSI values */
wifi = WifiInfo(); /* get WLAN services */
if wifiInfo != null then
    strength = wifi.getRSSI(); /* WIFI signal strength */
    if strength ≤ 2 then
        Get MOS;
        Activate UMTS interface;
        Wait for IP allocation;
        Send reINVITE with new connection info via MMAS;
        Start new dialog;
        Process BYE message triggered by MMAS ;
        Close the WLAN dialog and all its connections;
    else
        /* strength > 2 */
    end if
else
    /* No WLAN */
end if
```

For the scenario discussed in this paper (c.f., Section 3), instead of activating both access network interfaces at all times, the activation of the 3G access network is only triggered once the RSSI of the WLAN is less than or equal to 2. At this value, the testbed started to experience QoE degradation due to packet losses. The proposed solution will eliminate Dn because both access network interfaces will be active, though WLAN signal strength will be weaker. In this solution, the handover process will be completed before the WLAN is out of range. The proposed solution meets the 3GPP requirements of seamless service continuity because the major contributor of latency is only Ds with which an average of just 3 RTP packets were dropped. This is hardly noticeable to end users.

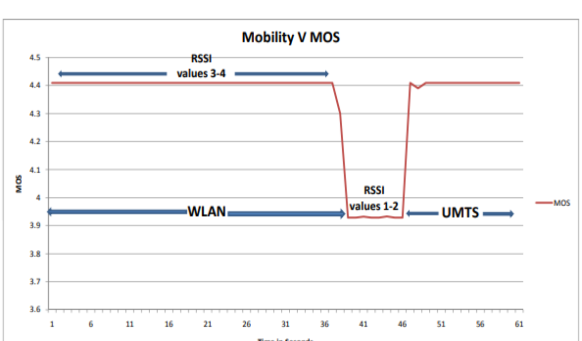

Figure 5. QoE degradation improvement

In the proposed solution, the QoE degradation is mainly caused by the weakening of WLAN signal strength at between 1 and 2 RSSI values. Fig. 5 illustrates the improvement in QoE degradation due to the proposed solution; the lowest MOS value is around 3.9 compared to 1 in Section 4. Note that the proposed solution was emulated by moving the G1 mobile phone to a location where RSSI reads 2, this triggers the handover process by sending a reinvite SIP message and then return to a place where the RSSI values reads more than 2. The reinvite message contains the same RTP connection details

5. Conclusions

This paper has evaluated the impact of IMS mobility management mechanism with Android G1 mobile phones on QoE. The current version of Android does not permit multiple activations of access network interfaces and hence, as shown in this paper, the phone does not conform to the requirements of 3GPP seamless service continuity and therefore suffers QoE degradation during the handover process. A solution to improve QoE and seamless 3GPP service continuity was proposed and implemented, preliminary results showed that there is a significant gain in QoE and the requirements of 3GPP service continuity were met.
Handover of voice or video traffic between access networks with varying QoS policies might result in QoE degradation, future work will include adaptation schemes for appropriate voice or video codecs just after handing over RTP traffic from WLAN to UMTS and vice versa.

Authors' contribution
All authors contributed equally to the preparation of this article.

Declaration of competing interest
The authors declare no conflicts of interest.

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