Performance Estimation of a SIP based Push-to-Talk Service for 3G Networks

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Abstract

Push-To-Talk (PTT) is a two-way, on-demand voice and data communication service, which is gaining in popularity due to its resource and cost efficiency. In this paper we present and evaluate a PTT service for an all-IP packet-switched 3G type network using the Session Initiation Protocol (SIP) as the call management protocol. In order to evaluate the service we demonstrate a possible implementation within the 3GPP UMTS Rel.5/6 specification based on the All-IP Multimedia Subsystem (IMS). Initial performance results indicate the viability of such a service using SIP and RTP/IP protocols.

1. Introduction

The concept of Push-To-Talk (PTT) in cellular mobile phones is one that has been around for several years now, however over the last year the concept has been taken up by many of the major network operators as well as telecommunications manufacturers. Traditional PTT is based on the concept of two-way radio communications as implemented with walkie-talkies. The introduction of PTT into digital cellular mobile networks effectively allows mobile phones to become walkie-talkies with unlimited range.

Until this year the only wireless service provider in North America to provide a PTT service was Nextel. Nextel uses a proprietary push-to-talk network developed alongside its voice network and cannot currently offer roaming between competing voice networks. The service being available through the 'Direct Connect' feature with PTT enabled phones produced mainly by Motorola under the iDEN brand. The service is used for communications by messengers - particularly in the New York City area, construction workers, factory workers, delivery workers and other businesses that require the use of dispatch-oriented communication services. A number of U.S. based carriers have recently announced the release of PTT services that will be layered onto their data networks. Another method of providing PTT services is through a downloadable application available to the latest Symbian OS based devices. Motorola, Nokia, Ericsson and Siemens are currently leading the development of a PTT service over 3G networks known as 'Push To Talk Over Cellular' (PoC). PoC is based on half-duplex Voice over IP (VoIP) technology. The use of IP technology means that the service uses cellular access and radio resources more efficiently than the traditional circuit-switched cellular services as network resources are reserved only for the duration of talk spurts instead of for an entire call session. The Session Initiation Protocol (SIP) [8] is to be used for call setup and teardown. The Real Time Transmission Protocol (RTP) [7] is to be used for the transmission of voice packets within a call. The RTP Control Protocol (RTCP) is used for Talker Arbitration – this is the process of determining which call member is allowed to talk and which call member listens at any given time during a call. As with the existing Nextel PTT service the PoC PTT service allows for Private Calls, between two individuals and Group Calls, between more than two individuals. A jointly developed specification [10, 11, 12, 13, 14, 15] has been produced that has also gained support from the likes of AT&T, Sony Ericsson, Sonim and Cingular. The specification is based on the All-IP Multimedia Subsystem (IMS) [5] as defined by the 3G standards organisation 3GPP. The specification uses existing 3GPP, OMA and IETF. The specification has been submitted to the OMA as a baseline to provide an access independent and globally interoperable standard for PoC. The PoC specification is intended to reduce market fragmentation and provide end users with an easy to use PTT service that enables roaming between different voice networks.

This paper presents and evaluates a PTT service for a UMTS network [1] based on the IMS with SIP as the call management protocol that is similar to the proposed PoC service and is inspired by the existing PTT service offered by Nextel. Voice packets are transmitted using the RTP. Talker Arbitration signalling is also sent using the RTP by utilising the RTP header extension field. As this architecture conforms to 3GPP standards it is essentially the same as the architecture that has been submitted for PoC to the OMA. The performance of both, the Private Call and Group Call features of the proposed PTT service in UMTS [1] are evaluated through a computer simulation using a C++ based UMTS network simulator.

2. PTT Service and Architecture for UMTS Rel. 5/6

2.1 SIP and UMTS Network Architecture Rel. 5/6

SIP is an application layer control protocol similar to HTTP that can establish, modify and terminate multimedia sessions or calls. SIP has been designed as part of the IETF suite of protocols to provide for IP based multimedia services, including protocols for reserving network resources, for handling real-time data and quality-of-service. However, the operation of SIP is independent of any of these protocols and it can be used in conjunction with other call setup and signalling protocols. SIP supports session control, user location, user capabilities, user availability, call setup, and call handling.

In order to demonstrate the SIP based PTT service in a 3G network, we have chosen UMTS. UMTS Rel.5/6 specifies the provision of traditional circuit and packet switched services over a single converged IP based packet-switched (PS) network. Control signalling is facilitated by an all-IP multimedia core network subsystem (IMS) with full IP packet support including full UMTS call control capabilities. The IMS works in conjunction with the PS Core Network (CN). Since SIP has been chosen by 3GPP as the signalling protocol between user equipment (UE) and the IMS as well as between components within the IMS, UMTS Rel.5/6 it is the ideal platform for the implementation of the proposed service.

Figure 1 depicts the main components of the UMTS Rel.6 architecture, that is the UMTS Radio Access Network (UTRAN), the PS CN and the IMS CN. The Serving and Gateway GPRS Support Nodes (SGSN and GGSN) constitute the PS core network. Each SGSN is connected to a number of Radio Network Controller's (RNCs) thereby acting as serving node for all mobiles that are under coverage of these RNCs. The GGSN is the point of packet data network (PDN) interconnection between external networks and the UMTS Public Land Mobile Network (PLMN). The IMS consists of the Call State Control Function (CSCF), the Media Gateway Control Function (MGCF) and Media GateWay (MGW). The CSCF can perform a number of functions depending on whether it is operating as a Proxy CSCF (P-CSCF), Serving CSCF (S-CSCF) or Interrogating CSCF (I-CSCF). The P-CSCF is the first contact point for the UE IMS; the S-CSCF actually handles the session states in the network; the I-CSCF is mainly the contact point within an operator's network for all IMS connections destined to a subscriber of that network operator, or a roaming subscriber currently located within that network operator's service area. The MGCF controls the MGW used in connections to external networks.

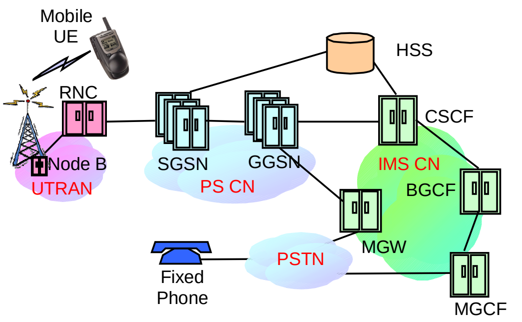

Figure 1: UMTS Rel. 5/6Model with PS CN & IMS Domain

2.2 Proposed PTT System Architecture

Figure 2. depicts the generic PTT system architecture that we propose to include into the UMTS Rel.5/6 core network. In the following the system architecture is described in general terms and then a mapping onto the UMTS Rel.5/6 IMS network elements is proposed.

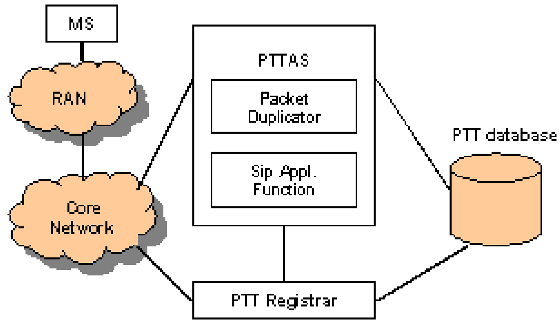

Figure 2. PTT System Architecture

2.2.1 Description of PTT System Architecture entities

PTTAS (Push to Talk Application Server): The PTTAS provides call-processing functions for dispatch services (Private Call, Group Call) to be implemented and it includes the packet duplicator function that is required for Group Calls – when the PTTAS receives the RTP voice packets from the current talker within the group call it makes copies of those packets for every other call member and then sends these copied packets to them.

PTT Registrar: The PTT Registrar provides SIP registration and authentication functions for dispatch services in PTT. The PTT registrar is responsible for generating the mobile stations contact address, and placing this address in the PTT database during the registration process. For this architecture it has been decided to create the PTT registrar and specify it as a separate logical entity. Physically though, the PTT registrar may be included on the same platform as the PTTAS or PTT database.

PTT database: The PTT database provides the system repository functions for subscribers, including SIP contact addresses (updated during SIP Registration), subscriber profiles and restrictions, passwords, subscriber address-book, buddy lists, and group lists. The PTT database will be operator provisioning point for the dispatch services, as well as a user self-provisioning point for users to build and modify their address-books, build buddy lists, and generate user defined groups.

2.2.2 Mapping the PTT System onto UMTS

In the UMTS registration process the I-CSCF assigns a S-CSCF to a user performing SIP registration. In the proposed PTT service, the S-CSCF will perform the functions of the PTT Registrar, this is also the case with the PoC system [11], as the S-CSCF behaves as a registrar, i.e. it accepts registration requests and makes its information available through the location server (e.g. Home Subscriber Server (HSS)). The HSS was chosen to perform the functions of the PTT database, as within the UMTS domain this is the master database for a given user. It is the entity containing the subscriber-related info to support the network activities actually handling calls/sessions. The PTT database provides similar functionality to the Group and List Management Server (GLMS) [11] as specified in the

PoC architecture specification. The PTTAS will be a separate entity; effectively this is a SIP application server (AS) that provides the services described in section 2.2.1. The PTTAS conforms to 3GPP standards, as the AS is an IMS CN Subsystem entity that offers value added IM services and resides either in the user's home network or in a third party location, it provides similar functionality to the PoC Server [11]. The PTT architecture for UMTS described here conforms to the requirements outlined by the 3GPP for the conferencing service within the IMS based on SIP [3, 4, 6]. The conferencing service provides the means for a user to create, manage, terminate, join and leave conferences, which are handled by a server within a home network of the conference creator. It also provides the network with the ability to give information about these conferences to the involved parties. Participants to conferences may be internal or external to the home network. Conferencing as defined by the 3GPP applies to any kind of media stream by which users may want to communicate, this includes e.g. audio and video media streams as well as instant message based conferences or gaming. Figure 3 shows the mapping of the PTT architecture onto UMTS Rel. 5/6 described.

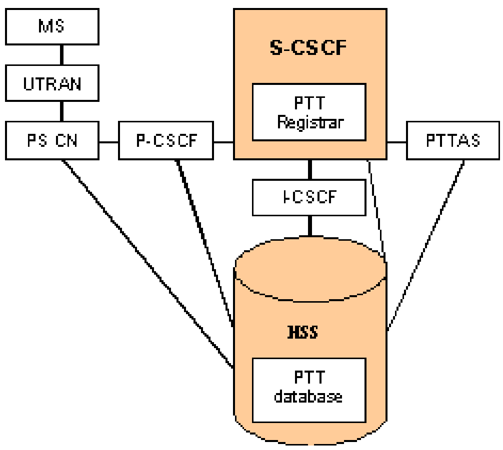

Figure 3. PTT architecture for UMTS Rel. 5/6

3. PTT Signalling Diagrams

In line with the UMTS Rel.5/6 specification, the proposed PTT service is based on SIP signalling. In the following three key signalling scenarios of the PTT service are described, that is PTT Registration, PTT Private Call Setup and Teardown and PTT Private Call Talker Arbitration.

3.1 PTT Registration

The procedure used for PTT Registration is essentially the procedure used for normal registration in the IMS and is depicted in Figure 4. Prior to registration the following items must be provisioned into various network elements. The mobile station (MS) is provisioned with the users unique PTT Address and associated password. The HSS is provisioned with subscriber info. This info is accessed via the users PTT Address. Within the registration flow the SIP Register message is used to inform the S-CSCF of the users current location. The P-CSCF receives the SIP Register message and performs a DNS query to determine the address of the I-CSCF and sends it the message. The I-CSCF then queries the HSS to validate that the user is unregistered and is allowed to register. If allowed, then a response is sent to the I-CSCF indicating the S-CSCF selection. The I-CSCF then sends the SIP Register to the S-CSCF. The S-CSCF then informs the HSS that the user has registered and the HSS updates its database with the S-CSCF for that user. The S-CSCF requests a user authentication from the HSS. The HSS selects an authentication vector and sends it to the S-CSCF. The S-CSCF sends a ‘SIP 401 Unauthorised’ message to the I-CSCF to indicate that the registration needs to be authenticated by the MS. The message is then forwarded to the MS where the MS sends an authentication response in a SIP Register message to the S-CSCF. The S-CSCF sends a ‘Cx Pull’ to the HSS that allows the S-CSCF to download subscriber data and service related info. The S-CSCF indicates to the MS that authentication was successfully completed by sending a SIP 200 OK message.

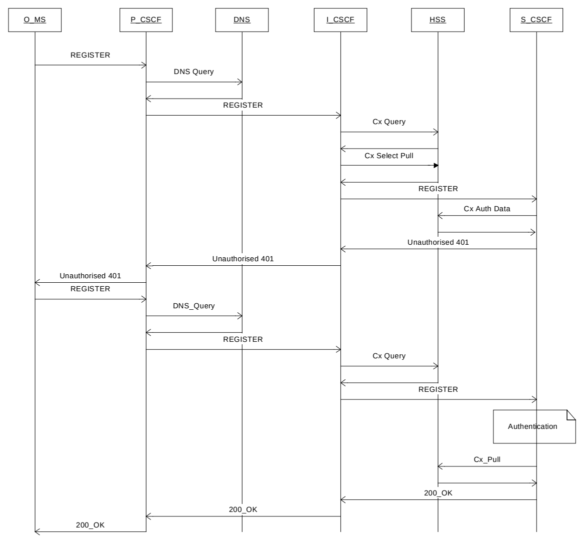

Figure 4. PTT Registration Signalling Flow

3.2 PTT Private Call Setup and Teardown

A PTT call may be made after PTT Registration has taken place. The call setup sequence is shown in Figure 5. The MS sends a SIP INVITE, an example contents is shown in Table 1, and directs it to the PTTAS whose address was received during the PTT Registration process. The INVITE specifies the PTT Address of the calling and called members in the ‘From’ and ‘To’ addresses as well as the Contact address of the MS. The SDP body specifies amongst other details the type of call, in this case a Private Call, the IP address and port in the MS for the bearer as well as the vocoder type used. The PTTAS then obtains the subscriber profiles from the HSS of both the originating and terminating MS. The PTTAS then sends a SIP INVITE to the terminating MS. The terminating MS then alerts the user to the call and responds to the PTTAS with a SIP 180 Ringing. When the user accepts a call the terminating MS sends a SIP 200 OK that contains the IP address and port of the MS where the bearer traffic should be sent to and the selected vocoder. The PTTAS responds to SIP 200 OK from the terminating MS by sending a SIP ACK to the terminating MS and sending a SIP 200 OK to the originating MS containing the terminating MS's bearer port and vocoder information. The originating MS receives the SIP 200 OK from the PTTAS and responds with a SIP ACK, then indicates to the user that they may talk by emitting a tone and sending the voice packets to the terminating MS.

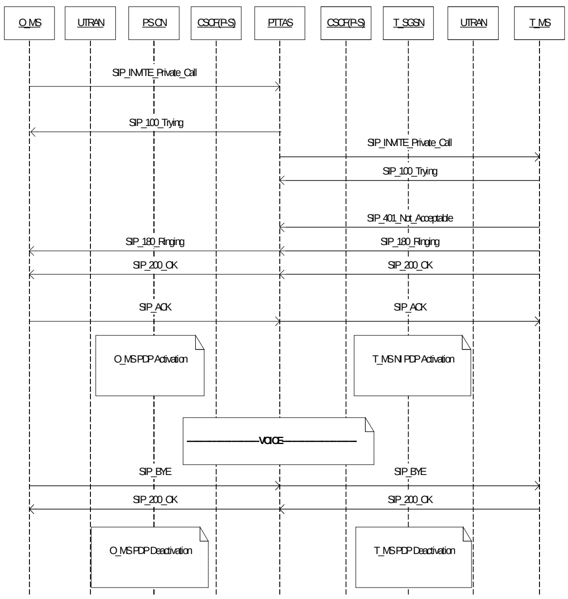

Figure 5. PTT Private Call Setup and Teardown

```
INVITE sip:username(T_MS)@carriername.net SIP/2.0
Via: SIP/2.0/UDP O_MS_IP_Address
From: <sip:username(O_MS)@carriername.net>
To: <sip:username(T_MS)@carriername.net>
Contact: <sip:username(O_MS)@O_MS_IPAddress >
Content-type: application/sdp
Call-ID: 1
User-agent: ms/1.0/1.0
Cseq: 1 INVITE
Content-Length: xxx
v=0
o=
t=0 0
c=IN IP4 O_MS_IP_Address
a=Call_Type:Private_Call
m=audio 5002 RTP/AVP 100
a=rtp_map:100 EVRC+
```

Table 1. PTT Call Setup INVITE example content

3.3 PTT Private Call Talker Arbitration

Talker Arbitration (TA) is the process by which a member within an ongoing call receives permission to talk by being granted the "*floor*". This is achieved by sending TA signalling between members of a PTT call to assign who is currently talking. For a Private Call the originating MS performs the TA and for a group call the TA is performed by the PTTAS. For the PoC system it is proposed that the PoC Server will perform TA for both Private Calls and Group Calls [13]. TA signalling is sent over the bearer path using the RTP header extension field. For the PoC system the TA signalling is sent using the RTP Control Protocol (RTCP) [13]. RTCP will be incorporated into the simulation at a future date and the relative merits of using RTCP and the RTP header extension field will be evaluated and compared. Also the PoC system uses a total of seven TA messages [13] compared to the four used in this simulation. The TA messages identified for TA are as follows.

- Floor_Open – This indicates to the members within a call that the floor is open as well as showing the active members within the call. This message is sent when the entity performing the TA receives a Floor_Release message.
- Floor_Release – This message is sent by the member currently in control of the floor to release floor control.
- Talk_Request – This is sent by a call member to the entity performing the TA to request floor control.
- Talk_Enabled – This is sent by the entity performing the TA to assign floor control to a call member. This message is sent to all members of the call with the 'id' parameter of the message identifying which member has the floor.

The PTT Private Call Talker Arbitration Signalling Flow displayed in Figure 6 can be described as follows. The originating MS performs TA once the call setup has been completed and the call is in progress. The originating MS has the PTT button pressed and is sending voice packets to the terminating MS. The originating MS then releases the PTT button and a Floor_Open message is sent by the originating MS to indicate that the floor is now open. Upon receipt of the Floor_Open message the user at the terminating MS presses the PTT button to request control of the floor (it should be noted that at any time from when the originating MS sent the Floor_Open message the user at the originating MS could again press the PTT button to request control of the floor again). The terminating MS then sends a Talk_Request message to the originating MS using the 'id' parameter to identify itself. After receiving the Talk_Request message from the terminating MS the originating MS sends Talk_Enabled to the terminating MS with the 'id' set to identify the terminating MS as having control of the floor. After the terminating MS has received the Talk_Enabled it indicates to the user that they may begin to speak and it begins to send RTP voice packets. When the terminating MS then releases the PTT button it sends a Floor_Release to the originating MS to indicate that it has released control of the floor. Upon receipt of the Floor_Release message the originating MS sends a Floor_Open message. The user of the originating MS then presses the PTT button and the originating MS sends a Talk_Enabled message with 'id' parameter set to identify the originating MS as now having control of the floor. The originating MS now indicates to the user that they may begin to speak and begins to send RTP voice packets. When the PTT button is released the originating MS sends out a Floor_Open message.

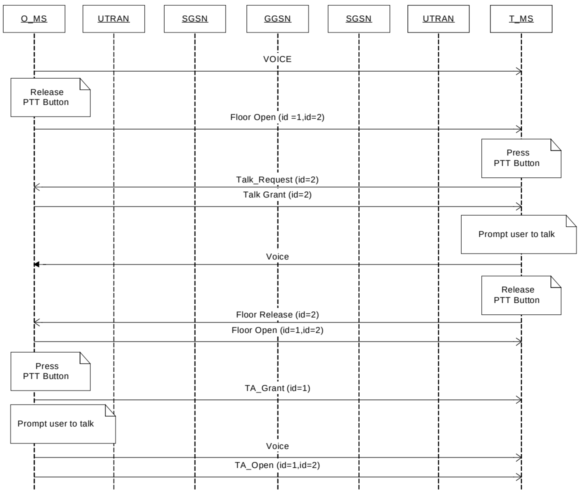

Figure 6 PTT Private Call Talker Arbitration Signalling Flow

Figure 7 gives the packet data structure used for transmitting voice in PTT. The RTP Header conforms to the format of the RTP Fixed Header Fields as specified by the 3GPP[7].

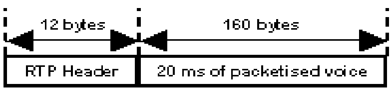

Figure 7 PTT RTP Voice Packet Structure

The Packet Structure used in Figure 8 conforms to the requirements outlined by the 3GPP for RTP Header Extension [7]. This is an extension mechanism that is provided to allow individual implementations to experiment with new payload-format-independent functions that require additional information to be carried in the RTP data packet header. The 'Header Field' is the same as the standard RTP packet header field except that the 'Extension bit' is set to indicate that a variable-length header extension is appended to the RTP header. The 'Profile Type' field allows multiple interoperating implementations to each experiment independently with different header extensions, or to allow a particular implementation to experiment with more than one type of header extension. The 'Extension Length' field counts the number of 32-bit words in the extension, excluding the four-octet extension header. The actual RTP Header Extension consists of the 'TA Message Type' and 'Call Member ID' fields. The 'TA Message Type' field identifies the Talker Arbitration message that has been sent. The 'Call Member ID' field identifies the Call Member that the message is either intended for or has been sent by, depending on which message has been sent.

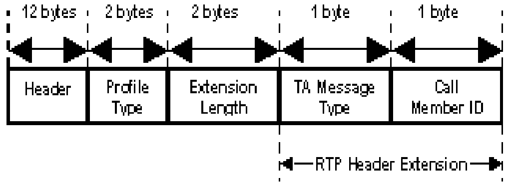

Figure 8. PTT RTP Talker Arbitration Packet Structure

4. Reference and Simulation Models

The simulator used to evaluate the service is known as the Signalling Simulator. This is a stochastic, event-driven simulator [16] implemented in Visual C++. The simulation consists of four main phases, as shown in Figure 9, reading inputs, initialisation, execution and post-processing. Inputs to the simulator are read from specific input files. Before initialisation takes place a check on those files is performed in order to initialisation errors. Then a cell layout and network topology are created and initialised. The mobile terminals are assigned random locations within the cells. Afterwards the mobility and session models drive the simulation by invoking the required UMTS and SIP signalling flows for the activation of a session. Once a session is activated, traffic models are used to determine the periods of bearer activity and inactivity depending on the type of service. The signalling execution persists for the duration of the simulation, which is set at the start of the simulation All collected data is stored in particular data structures, called bins, and are processed at the end of the simulation.

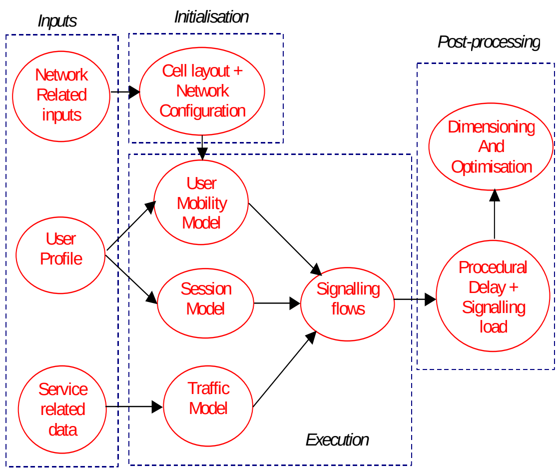

Figure 9 Signalling Simulator Structure

Figure 4 shows the functional model of the PTT architecture within the IMS domain considered in this paper. However, this does not provide the representative physical model that is required to model delays realistically in order to determine PTT SIP based call set up and talker arbitration. In order to model physical delays we propose a more practical realisation of a UMTS IMS as shown in Figure 10. A physical SIP server Network Element (NE) is shown, which hosts the S-CSCF and the P-CSCF. This seems a reasonable concept as every network will require both of these functionalities and interaction between them is likely to be largest for the local network. We assume the computing platform is typically a set of 16 SUN Netra 20 type servers or a similar platform, providing a 25ms SIP to SIP message turnaround duration for up to 800k subscribers per NE. We make similar assumptions for the HSS. Co-locating the I-CSCF and the HLR together provides for fast interrogation behaviour and we estimate that for a similar sized platform to the SIP server, interrogations and HSS Cx data retrieval can take ~55ms per transaction (read, read/write and forward averaged).

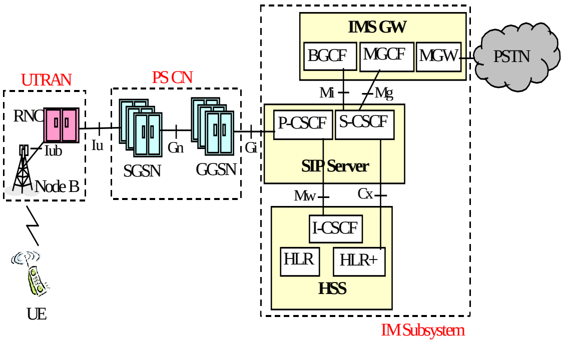

Figure 10 A Practical Realisation of UMTS IMS

Based on the proposed reference model, a typical network configuration is analysed for the dense urban environment, where the UTRAN employs 784 NodeBs and 4 RNCs. The CS domain is comprised of 2 MSCs and 1 GMSC, whereas the PS domain consists of 2 SGSNs and 1 GGSN. With this configuration, all core network-related and mobility signalling including handover are accounted for. In the IMS, we assume that there are 1 P-CSCF, 1 S-CSCF, 1 I-CSCF and 1 MGCF. These elements are co-located with the GGSN. The caller is assumed to be a mobile located in its home network.

5. Simulation Results

Table 2 shows initial performance results for the Private Call function of the PTT service within the proposed UMTS based architecture described in this paper. Table 3 shows initial performance results for the Group Call function. Both tables shows the mean of RAN delay, Core Network and Total delay averaged over all simulated users for mobile originated and mobile terminated call setup and teardown. The TA message results are shown. The TA messages used in the simulation are all of equal size and as a result it was found that the average time for all TA messages (MO and MT) took approximately the same time and are therefore presented in both tables in a single row. Also shown is the delay due to the secondary PDP context activation both at the originating and terminating side.

The results presented in Table 2 for the Private Call Function show that the mean call setup delay including a secondary PDP context activation for a mobile originated call is just under 3sec, which is a very fast call establishment and is much faster than SIP based call control for VoIP as defined by 3GPP [9]. The SIP compression schemes considered in [9] to reduce setup signalling are not required here which simplifies the design.

Mean Delay (Seconds)

	RAN Delay	Core Delay	Total Delay
PTT Private Call Setup MO	0.436085907	0.750333604	1.186419511
PTT Private Call Setup MT	0.506239114	0.700436636	1.20667575
PTT Private Call Teardown MO	0.14939209	0.27513663	0.42452872
PTT Private Call Teardown MT	0.113328509	0.275100312	0.388428821
TA Messages MO & MT	0.038161	0.125018	0.163179
PDP Context Act	1.044350965	0.770174192	1.814525157
NI PDP Act	1.052842351	0.708620626	1.761462977

Table 2. Performance Results for Private Call Setup

Mean Delay (Seconds)

	RAN Delay	Core Delay	Total Delay
PTT Group Call Setup MO	0.73581	1.1505217	1.8863317
PTT Group Call Setup MT	0.69368	0.976176	1.669856
PTT Group Call Teardown MO	0.1246668	0.187891	0.3125578
PTT Group Call Teardown MT	0.192674	0.375172	0.567846
TA Messages MO & MT	0.0041843	0.350027	0.3542113
PDP Context Act	1.54915949	0.77018427	2.31934376
NI PDP Act	2.6284126	1.41724171	4.04565431

Table 3. Performance Results for Group Call Setup

A mobile terminated call setup delay is slightly higher as it includes paging the mobile terminal. The PDP activations both network initiated and otherwise take approximately a half a second longer than both MO (mobile originating) and mobile terminating (MT) call setup. The increase time taken in the PDP activations do not affect the call setup as such because if call setup has been completed before the PDP activation intended for RTP data is completed then the PDP activation for SIP signalling can be used. The time taken for the average TA message to go from the originating MS to the terminating MS was found to be just over 0.3 seconds for a Private Call. The results presented in Table 3 shows an increase in the call setup times in comparison to the Private Call. There is an increase in average time for TA messaging for Group Calls relative to Private Calls - this is caused by the TA messaging having to go to the PTTAS in the IMS for group calls as the PTTAS performs the TA, whereas for Private Calls the TA is performed by the originating mobile station.

6. Conclusions

In this paper we present a novel SIP based Push-To-Talk for 3G networks. We show an implementation of the service for UMTS Release 5/6 that is based on the IMS. The implementation presented is similar to that proposed in the PoC specification. Initial results from a computer simulation based performance analysis indicate that the service will provide satisfactory performance in terms of average call setup delay and talker arbitration delay.

7. References

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