UPGRADE
The European Journal for the Informatics Professional
http://www.upgrade-cepis.org
Vol. V, No. 1, February 2004

UPGRADE is the European Journal for the Informatics Professional, published bimonthly at <http://www.upgrade-cepis.org/>

Publisher
UPGRADE is published on behalf of CEPIS (Council of European Professional Informatics Societies, <http://www.cepis.org/>) by NOVÁTICA <http://www.ati.es/novatica/>, journal of the Spanish CEPIS society ATI (Asociación de Técnicos de Informática <http://www.ati.es/>).
UPGRADE is also published in Spanish (full issue printed, some articles online) by NOVÁTICA, and in Italian (abstracts and some articles online) by the Italian CEPIS society ALSI <http://www.alsi.it> and the Italian IT portal Tecnoteca <http://www.tecnoteca.it/>.
UPGRADE was created in October 2000 by CEPIS and was first published by NOVÁTICA and INFORMATIK/INFORMATIQUE, bimonthly journal of SVI/FSI (Swiss Federation of Professional Informatics Societies, <http://www.svifsi.ch/>).

E-mail addresses for editorial correspondence: <rfcalvo@ati.es>, <nicolet@acm.org> or <rcarniel@dgt.uniud.it>

E-mail address for advertising correspondence: <novatica@ati.es>

Upgrade Newslist available at <http://www.upgrade-cepis.org/pages/editinfo.html#newslist>

ISSN 1684-5285

Next issue (April 2004):
"Unified Modeling Language (UML)"

Wireless Networks - Telecommunications' New Age

Guest Editors: Mehmet Ufuk Çaglayan, Vicente Casares-Giner, and Jordi Domingo-Pascual

Joint issue with NOVÁTICA*

Mosaic

* This monograph will be also published in Spanish (full issue printed; summary, abstracts and some articles online) by NOVÁTICA, journal of the Spanish CEPIS society ATI (Asociación de Técnicos de Informática) at <http://www.ati.es/novatica/>, and in Italian (online edition only, containing summary abstracts and some articles) by the Italian CEPIS society ALSI and the Italian IT portal Tecnoteca at <http://www.tecnoteca.it>.

VoIP Services for Mobile Networks

Ai-Chun Pang and Yi-Bing Lin

This paper describes the Universal Mobile Telecommunications System all-IP approach for wireless Voice over Internet Protocol (VoIP). In this approach, the IP Multimedia Core Network Subsystem (IMS) provides real-time multimedia services for mobile subscribers, and utilizes the common IP technology to support the services controlled by the Session Initiation Protocol (SIP). We elaborate on the functionalities of IMS network nodes. Then we describe the application level registration and call origination procedures to illustrate the inter-operation between the IMS network nodes for the SIP-based VoIP applications.

Keywords: IP Multimedia Core Network Subsystem (IMS), Session Initiation Protocol (SIP), Universal Mobile Telecommunications System (UMTS), Voice over Internet Protocol (VoIP).

1 Introduction

Next generation telecommunications networks will provide global information access for the users with mobility, which is achieved through integration of the Internet and the Third Generation (3G) wireless communications techniques. As consumers become increasingly mobile, they will demand wireless access to services available from the Internet. Specifically, mobility, privacy and immediacy offered by wireless access introduce new opportunities for Internet business.

One of the most important applications for integration of the Internet and the 3G wireless technologies is Voice over Internet Protocol (VoIP). In this paper, we use the Universal Mobile Telecommunications System (UMTS) all-IP approach as an example to illustrate how a 3G mobile network can utilize the common IP technology to support multimedia and voice services controlled by the Session Initiation Protocol (SIP). We first introduce SIP. Then we describe the UMTS all-IP architecture. Finally, we show how SIP is utilized in the UMTS IP Multimedia Core Network Subsystem (IMS) to support VoIP services.

2 Session Initiation Protocol

SIP [2] is an application-layer signalling protocol over IP networks, which is designed for creating, modifying and terminating multimedia sessions or calls. Two major network elements are defined in SIP: user agent and network server. The user agent resides at SIP endpoints (or phones), which contains both a User Agent Client (UAC) and a User Agent Server (UAS). The UAC (or calling user agent) is responsible for issuing SIP requests, and the UAS (or called user agent) receives the SIP request and responds to the request. There are three types of SIP network servers: proxy server, redirect server and registrar. A proxy server forwards the SIP requests from a UAC to the destination UAS. A redirect server receives the SIP requests from a UAC and responds with the destination UAS address. To support user mobility, the user agent informs the network of its current location by explicitly registering with a registrar. The registrar is typically co-located with a proxy or redirect server.

Six basic types of SIP requests are defined, which are described as follows.

- INVITE is used to initiate a multimedia session, which includes the routing information of the calling and called parties, and the type of media to be exchanged between the two parties.
- ACK is sent from a UAC to a UAS to confirm that the final response to an INVITE request has been received.
- OPTIONS is used to query the user agent's capabilities such as the supported media type.
- BYE is used to release a multimedia session or call.
- CANCEL is used to cancel a pending request (i.e., an uncompleted request).
- REGISTER is sent from a user agent to the registrar to register the address where the subscriber is located.

After receiving a request message, the recipient takes appropriate actions and acknowledges with a SIP response message. The response message carries a return code indicating the

Ai-Chun Pang received the BS., MS. and PhD. degrees in Computer Science and Information Engineering from *National Chiao Tung University* (NCTU), Taiwan, in 1996, 1998 and 2002, respectively. She joined the Department of Computer Science and Information Engineering, *National Taiwan University* (NTU), Taipei, Taiwan, as an Assistant Professor in 2002. Her research interests include design and analysis of personal communications services networks, mobile computing, voice over IP and performance modelling. <acpang@csie.ntu.edu.tw>

Yi-Bing Lin received his BSEE degree from *National Cheng Kung University*, Taiwan, in 1983, and his PhD. degree in Computer Science from the University of Washington, USA, in 1990. He is Chair Professor at Providence University and a Professor in the Department of Computer Science and Information Engineering, *National Chiao Tung University*, Taiwan. His current research interests include design and analysis of mobile telecommunications networks. Dr. Lin is an IEEE Fellow and an ACM Fellow. <liny@csie.nctu.edu.tw>

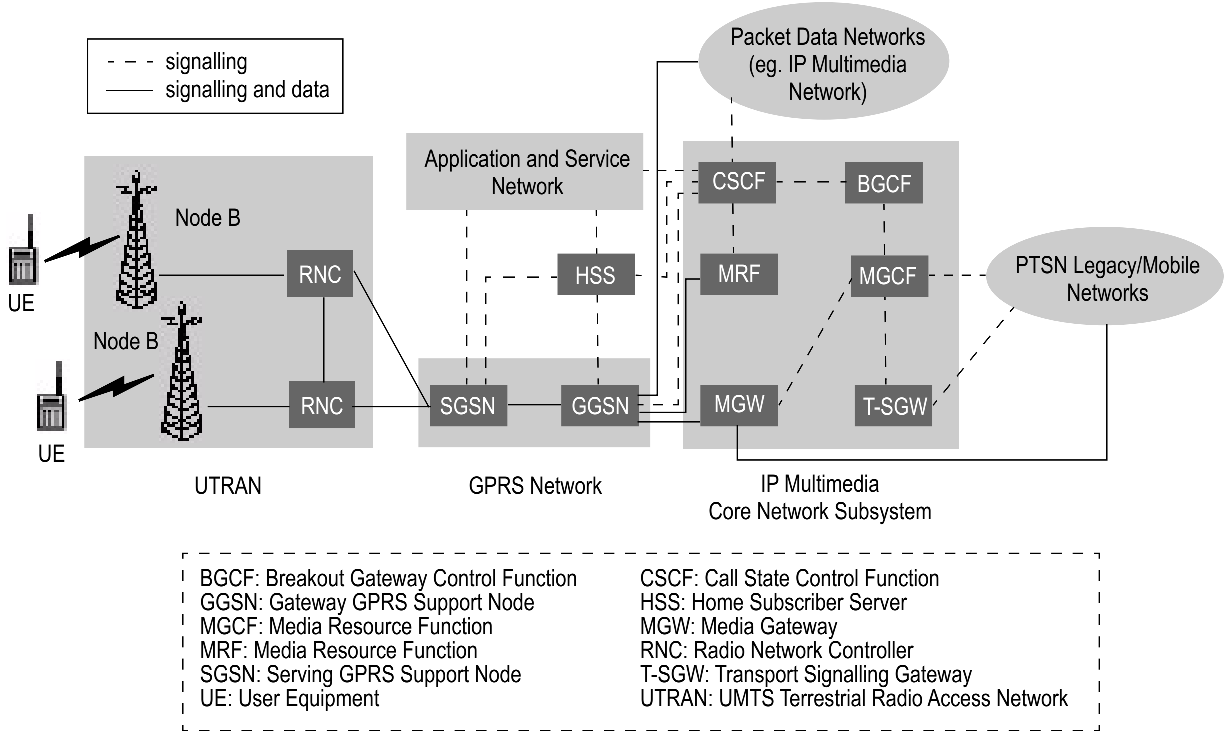

Figure 1: UMTS All-IP Network Architecture.

execution result for the request. Examples of the code are 100 (the command is currently being executed), 200 (the command was executed normally) and 510 (the command could not be executed because a protocol error was detected). In the following section, we elaborate on how SIP can be supported for VoIP services over UMTS.

3 VoIP over UMTS

Figure 1 shows the UMTS all-IP network architecture [1][4]. In this architecture, Signalling System No. 7 (SS7) transport is replaced by IP, and the common IP technology supports all services including multimedia and voice services controlled by SIP. The UMTS all-IP network consists of five segments: General Packet Radio Service (GPRS) network, Home Subscriber Server (HSS), UMTS Terrestrial Radio Access Network (UTRAN), IMS, and application/service network. The GPRS network consists of Serving GPRS Support Node (SGSN) and its Gateway GPRS Support Node (GGSN). The SGSN connects to the UTRAN, which provides the mobility management and the Packet Data Protocol (PDP) context activation services to mobile subscribers. The GGSN interacts with the IMS and external packet data networks, and is connected with SGSNs via an IP-based GPRS backbone network. GGSNs and SGSNs communicate with the HSS to obtain the mobility and session management information of subscribers.

The UTRAN adopts the Wideband CDMA} radio technology to provide broadband wireless access. The UTRAN consists of Node Bs (i.e., base stations) and Radio Network Controllers (RNCs) connected by an ATM network. A User Equipment (UE) communicates with one or more Node Bs through the radio interface.

The IMS provides real-time multimedia services for mobile subscribers, which consists of six network nodes:

3.1 Call Session Control Function (CSFC)

CSCF communicates with the HSS for location information exchange, and handles control-layer functions related to application level registration and SIP-based multimedia sessions. The CSCF consists of the following logical components. Incoming Call Gateway (ICGW) communicates with the HSS to perform routing of incoming calls.

Call Control Function (CCF) is responsible for call setup and call-event report for billing and auditing. It receives and processes IMS registration requests, provides service trigger mechanism toward application/service networks, and may invoke location-based services related to the serving network. It also checks whether the requested outgoing communication is allowed given the current subscription. Serving Profile Database (SPD) interacts with the HSS in the home network to receive profile information for the all-IP subscriber. Address Handling (AH) analyses, translates (and may modify) addresses. It supports address portability and alias address mapping (e.g., mapping between E.164 number and transport address).

A CSCF can be interrogating, proxy or serving. The Interrogating CSCF (I-CSCF) determines how to route mobile terminated calls to the destination UEs. That is, the I-CSCF is the contact point for the home network of the destination UE,

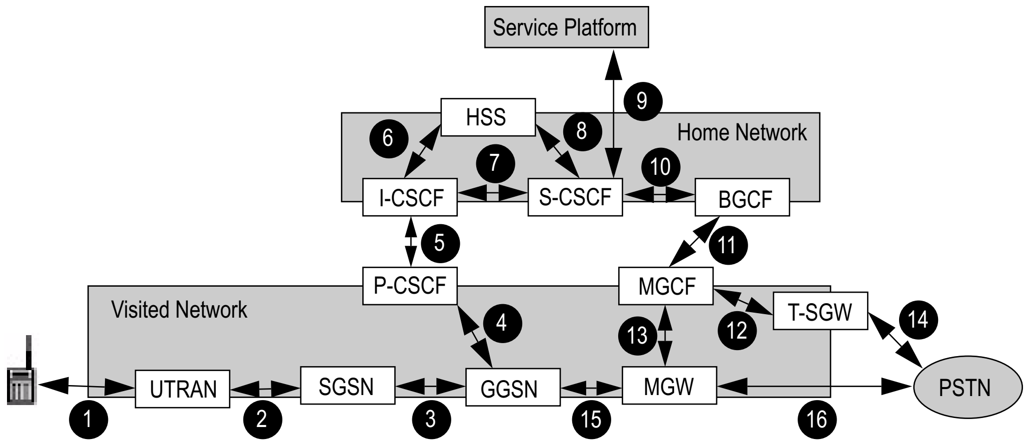

Figure 2: IMS Application Level Registration and Call Origination.

which may be used to hide the configuration, capacity, and topology of the home network from the outside world. When a UE attaches to the network and performs PDP context activation, a Proxy CSCF} (P-CSCF) is assigned to the UE. The P-CSCF contains limited CSCF functions (that is, address translation functions) to forward the request to the I-CSCF at the home network. Authorization for bearer resources in a network is also performed by a P-CSCF within that network. By exercising the application level registration, a Serving CSCF (S-CSCF) is assigned to serve the UE. This S-CSCF supports the signalling interactions with the UE for call setup and supplementary services control (e.g., service request and authentication) through SIP.

3.2 Breakout Gateway Control Function

(BGCF) is responsible for selecting an appropriate Public Switched Telephone Network (PSTN) breakout point based on the received SIP request from the S-CSCF.

3.3 Media Gateway Control Function (MGCF)

MGCF communicates with the CSCF through SIP to control media channels for connection in an MGW. The MGCF selects the CSCF depending on the routing number for incoming calls from legacy networks.

3.4 Transport Signalling Gateway Function (T-GSW)

T-SGW serves as the PSTN signalling termination point and provides PSTN/legacy mobile networks to IP transport level address mapping, which maps call-related signalling from PSTN on an IP bearer and sends it to the MGCF, and vice versa.

3.5 Media Resource Function (MRF)

MRF performs multiparty call, multimedia conferencing, tones and announcements functionalities. The MRF communicates with the S-CSCF for service validation of multiparty/multimedia sessions.

3.6 Media Gateway Function (MGW)

MGW provides user data transport in the IMS. The MGW terminates bearer channels from PSTN/legacy mobile networks and media streams from a packet network (e.g., Real-time Transport Protocol [5] streams in an IP network).

In the UMTS all-IP network, a UE conducts two types of registration. In bearer level registration, the UE registers with the GPRS network following the standard UMTS routing area update or attach procedures [3]. After bearer level registration, the UE can activate PDP contexts in the GPRS network. Bearer level registration is required to support GPRS-based services. To offer IM services, application level registration must be performed in the IMS. The application level registration is initiated by a UE. As shown in Figure 2 the UE first sends a SIP REGISTER message to the I-CSCF through path (1) → (2) → (3) → (4) → (5). After communicating with the HSS (Figure 2 (6)), the I-CSCF forwards the SIP REGISTER to the selected S-CSCF (Figure 2 (7). Then the S-CSCF interacts with the HSS for obtaining the subscriber's profile and responds SIP 200 OK to the UE through path (7) → (5) → (4) → (3) → (2) → (1)),which indicates that registration is successfully performed.

When a UE makes a call to the PSTN, a SIP INVITE message is issued from the UE to the S-CSCF through path (1) → (2) → (3) → (4) → (5) → (7). The S-CSCF forwards the SIP INVITE to the BGCF. The BGCF selects an MGCF in the visited network and transmits the message to the MGCF (path (10) → (11) in Figure 2).

The MGCF instructs the MGW to allocate the necessary resources for the call (Figure 2 (13)) and delivers the SS7 ISUP Setup message to the called party via the T-SGW (path (12) → (14) in Figure 2). After the call is established, the voice path for this call is (1) ↔ (2) ↔ (3) ↔ (15) ↔ (16).

4 Conclusions

This paper described the Universal Mobile Telecommunications System all-IP approach for wireless Voice over Internet Protocol (VoIP) support. In this approach, the IP Multimedia Core Network Subsystem (IMS) provides real-time multimedia and voice services using SIP. We elaborated on the functionalities of IMS network nodes. Then we described application level registration and call origination procedures to show how a mobile subscriber accesses the wireless VoIP services.

References

[1] 3rd Generation Partnership Project; Technical Specification Group Services and Systems Aspects; IP Multimedia Subsystem Stage 2. Technical Specification 3G TS 23.228 version 5.1.0 (2001–06), 2001.

[2] M. Handley et al. SIP: Session Initiation Protocol. IETF RFC 2543, August 2000.

[3] Y.-B.Lin and I. Chlamtac. Wireless and Mobile Network Architectures. John Wiley & Sons, 2001.

[4] Y.-B.Lin, Y.-R. Huang, A.-C.Pang,and Imrich Chlamtac. All-IP Approach for Third Generation Mobile Networks. IEEE Network, 16(5):8–19, 2002.

[5] H. Schulzrinne et al. RTP: A Transport Protocol for Real-Time Applications. IETF RFC 1889, January 1996.