Wireless Networks 9, 157–164, 2003

vGPRS: A Mechanism for Voice over GPRS

MING-FENG CHANG, YI-BING LIN and AI-CHUN PANG *
Department of Computer Science and Information Engineering, National Chiao Tung University, Hsinchu, Taiwan, ROC

Abstract. This paper proposes vGPRS, a voice over IP (VoIP) mechanism for general packet radio service (GPRS) network. In this approach, a new network element called VoIP mobile switching center (VMSC) is introduced to replace standard GSM MSC. Both standard GSM and GPRS mobile stations can be used to receive real-time VoIP service, which need not be equipped with the VoIP (i.e., H.323) terminal capabilities. The vGPRS approach is implemented using standard H.323, GPRS, and GSM protocols. Thus, existing GPRS and H.323 network elements are not modified. Furthermore, the message flows for vGPRS registration, call origination, call release and call termination procedures are described to show the feasibility of our vGPRS system.

Keywords: gatekeeper, GGSN, GPRS, SGSN, VoIP

1. Introduction

In early 2000, only a small portion of GSM subscribers used data services. This low data penetration rate is due to the fact that the existing GSM systems do not support easy access, high data rate and attractive prices. GSM operators must offer better services to stimulate the demand. The solution is *General Packet Radio Service* (GPRS). GPRS reuses the existing GSM infrastructure to provide end-to-end packet-switched services. The GPRS core network has also been developed for IS-136 TDMA systems, and is anticipated to evolve as the core network for the third-generation mobile systems.

Recently, supporting telephony services over IP network or the so called voice over IP (VoIP) is considered as a promising trend in telecommunication business. Particularly, integrating mobile phone services with VoIP becomes an important issue, which has been intensively studied [3,9,14]. Thus, it is natural to support voice service over GPRS. In 3G TR 21.978 [1], VoIP service for GPRS is defined based on the H.323 protocol [11], which nicely integrates GPRS with H.323-based VoIP network. In this approach, the mobile phones or mobile stations (MSs) must be equipped with vocoder and H.323 terminal capabilities. In this paper, we propose vGPRS, a VoIP service for GPRS [13]. Our approach supports standard GSM and GPRS MSs to enjoy real-time VoIP service without modifying the existing GSM/GPRS network nodes as well as H.323 family protocols. Other approaches such as 3G TR21.978 cannot accommodate standard GSM and GPRS MSs to receive VoIP service. Instead, the MSs must be equipped with the H.323 terminal capabilities. Furthermore, this previous approach may not offer real-time VoIP based on the existing GPRS architecture.

This paper is organized as follows. Section 2 introduces the vGPRS network architecture and shows how GPRS can be modified to support vGPRS. Section 3 describes the registration procedure that enables a GSM or a GPRS MS to enjoy VoIP service. Section 4 describes how an MS originates a call through vGPRS. Section 5 describes how an incoming call is delivered to an MS. Section 6 compares vGPRS with the 3G TR21.978 approach.

* Corresponding author.

2. vGPRS network architecture

Figure 1 illustrates the GPRS architecture. In this figure, base transceiver station (BTS), base station controller (BSC), mobile switching center (MSC), home location register (HLR), and visitor location register (VLR) are GSM network nodes. GPRS introduces two new network nodes; namely, *serving GPRS support node* (SGSN) and *gateway GPRS support node* (GGSN), to the GSM architecture. An SGSN receives and transmits packets between the MSs and their counterparts in the public switched data network (PSDN). To connect to an SGSN, a packet control unit (PCU) is implemented in the BSC. The BSC forwards circuit-switched calls to the MSC, and packet-switched data (through the PCU) to the SGSN. A BSC can only connect to one SGSN. The GGSN interworks with the PSDN using connectionless network protocols, such as Internet protocol and the OSI connectionless network protocol, or connection oriented protocols such as X.25. Both SGSN and GGSN interact with the GSM location databases, including the HLR and the VLRs, to track the location of the

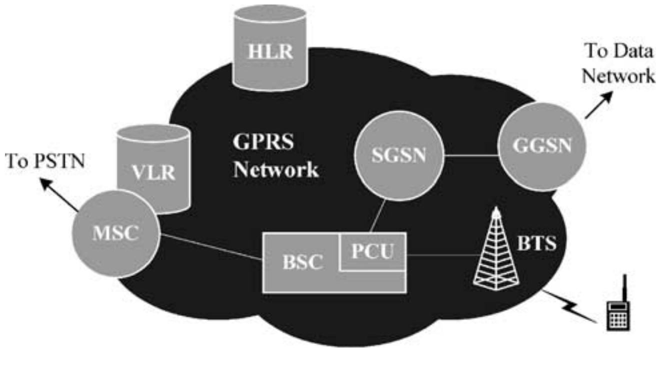

Figure 1. The GPRS network.

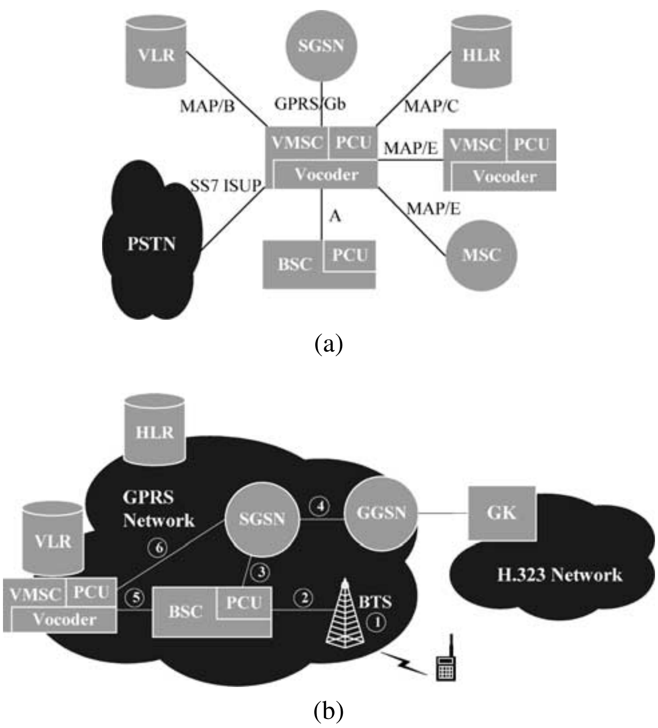

Figure 2. (a) the VMSC interfaces and (b) the vGPRS network architecture.

MSs. The reader is referred to [12] for details of GSM and GPRS.

The vGPRS approach provides VoIP service to standard GSM and GPRS MSs. In vGPRS a new network node VoIP MSC (VMSC) is introduced to replace MSC. The VMSC is a router-based softswitch and its cost is anticipated to be cheaper than an MSC. Figure 2(a) shows the interfaces between the VMSC and other GSM/GPRS network nodes. The GSM signaling interfaces of the VMSC are exactly the same as that of an MSC. That is, it communicates with the BSC, the VLR, the HLR, and the MSC (either GSM MSC or VMSC) through A, B, C and E interfaces, respectively. The VMSC also interfaces with the public switched telephone network (PSTN) via the SS7 ISUP protocol. Unlike an MSC, the VMSC communicates with SGSN through GPRS Gb interface. In other words, while an MSC delivers voice traffic through circuit-switched trunks, the VMSC delivers VoIP packets through GPRS network. In figure 2(b), the data path of a GPRS MS is (1) ↔ (2) ↔ (3) ↔ (4). The voice path is (1) ↔ (2) ↔ (5) ↔ (6) ↔ (4). In this voice path, (1) ↔ (2) ↔ (5) is circuit switched as in GSM. Thus, both standard GSM MSs and GPRS MSs can set up calls as if they are in the standard GSM/GPRS network. At the VMSC, the voice information is translated into GPRS packets through vocoder and packet control unit (PCU). Then the packets are delivered to the GPRS network through the SGSN (see path (6) ↔ (4)). An IP address is associated with every MS attached to the VMSC. In GPRS, the IP address can be created statically or dynamically for a GPRS MS. On the other hand, a standard GSM MS cannot be assigned an IP address through GPRS. Thus, in vGPRS the creation of the IP address is performed by the VMSC through the standard GPRS Packet Data

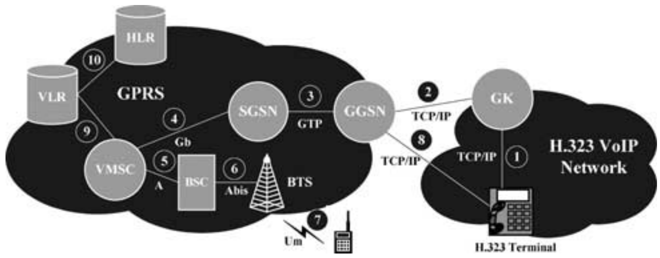

Figure 3. Connection between an H.323 terminal and a GSM MS.

Protocol (PDP) context activation procedures (to be elaborated in section 3). The VMSC maintains an MS table. The table stores the MS mobility management (MM) and PDP contexts such as TMSI (Temporary Mobile Subscriber Identity), IMSI (International Mobile Subscriber Identity), and the QoS profile requested. These contexts are the same as that stored in a GPRS MS (see section 13.4, GSM 03.60 [7]). In vGPRS, H.323 protocol [11,14] is used to support voice applications. The VMSC executes the H.323 protocol just like an H.323 terminal.

A gatekeeper *GK* in the H.323 network (see figure 2(b)) performs standard H.323 gatekeeper functions (such as address translation). Figure 3 illustrates the connection between an H.323 terminal and a GSM MS. In this figure, the TCP/IP protocols are exercised in links (1), (2), and (8). The GPRS tunneling protocol [8] is exercised in link (3), and GPRS Gb protocol [6] is exercised in link (4). The standard GSM protocols are exercised in links (5)–(7), (9), and (10). The H.323 protocol is implemented on top of TCP/IP, and is exercised between the VMSC and the nodes in the H.323 network. The H.323 packets are encapsulated and delivered in the GPRS network through the GPRS tunneling protocol.

3. Registration

Several types of messages are described in this and subsequent sections. A message with the "MAP_". prefix represents a GSM MAP message. The "Um_", "A_" and "Abis_" prefixes denote air interface, A interface, and Abis interface, respectively. The *Registration, Admission and Status* (RAS) and Q.931 messages are defined in H.323 and H.225 protocols [10,11]. In GSM, a registration procedure is performed when an MS is turned on or when the MS moves to a new location area. Without loss of generality, this section describes the registration procedure by assuming that registration is performed when the MS is turned on. The registration procedure for MS movement is similar and will be briefly elaborated at the end of this section. The GSM registration messages (steps 1.1, 1.2, and 1.3) are delivered through the path (7) ↔ (6) ↔ (5) ↔ (9) ↔ (10) in figure 3. The H.323 messages (steps 1.4, 1.5, and 1.6) are delivered through the path (7) ↔ (6) ↔ (5) ↔ (4) ↔ (3) ↔ (2). The message flow is illustrated in figure 4 and are described in the following steps.

Step 1.1. An MS is turned on. To join in the network, the MS performs registration following the standard GSM location update procedure [5]. The MS sends a Um_Location_

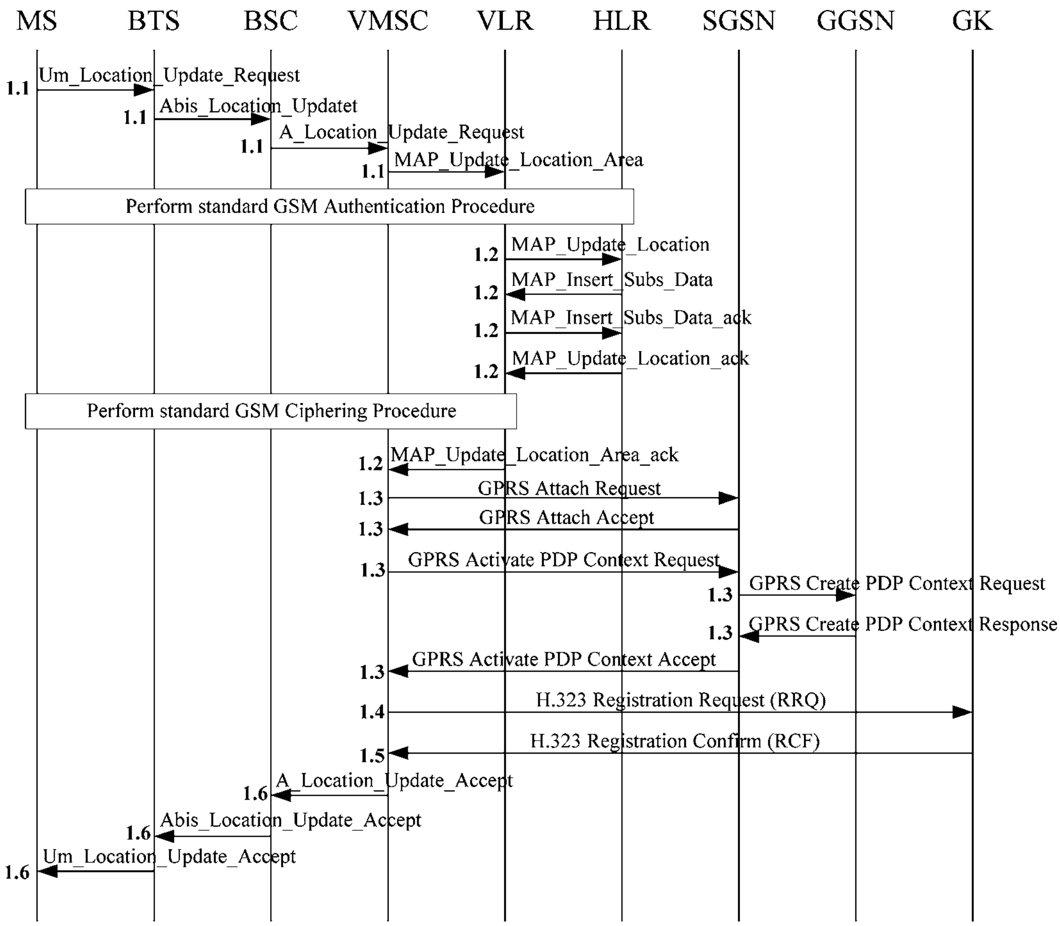

Figure 4. Message flow for vGPRS registration.

Update_Request message to the BTS. The BTS forwards this request to the BSC using the GSM Abis message Abis_Location_Update. The BSC then sends A_Location_Update message to VMSC through GSM A interface. The VMSC issues GSM MAP message MAP_Update_Location_Area to the VLR. We assume that the MS provides the IMSI (international mobile subscriber identity) for registration. Our message flow can be easily extended for the case when the TMSI (temporary mobile subscriber identity) is used [12] (which is likely to occur for location update due to MS movement). The standard GSM authentication procedure is exercised between the MS and the HLR to authenticate the MS. The details are omitted.

Step 1.2. The VLR sends message MAP_Update_Location to the HLR, and obtains the subscription profile of the MS (the profile indicates, e.g., if the MS is allowed to make international calls) from the HLR through the MAP_Insert_Subs_Data messages exchanged. The VLR then sets up the standard GSM ciphering with the MS. Then it sends the MAP_Update_Location_Area_ack message to VMSC to indicate that the registration is successful.

Step 1.3. The VMSC performs GPRS attach to the SGSN by exchanging the GPRS Attach Request and Accept message pair. Following the standard GPRS PDP context activation procedure, the VMSC activates a new PDP context just like a GPRS MS does. In the activation procedure, the IMSI of the MS is used by the GGSN to retrieve the HLR record to obtain information such as IP address (we assume that the IP address is allocated dynamically). The PDP context record for the MS is created in the GGSN, which includes fields such as IMSI, IP address, QoS profile negotiated, SGSN address, and so on. Since the activated PDP context is dedicated for transmitting the H.323 signaling, the QoS profile can be set to low priority and network resource would not be wasted. At this point, an IP session is established so that the VMSC can communicate with the gatekeeper *GK* in the external H.323 network.

Step 1.4. By using the H.323 signaling PDP context, the VMSC initiates the end-point registration to inform the *GK* of its transport address and alias address (i.e., MSISDN) through the RAS Registration Request (RRQ) message.

Step 1.5. The *GK* creates an entry for the MS in the address translation table, which stores the (IP address, MSISDN) pair. Then it confirms the registration by sending the RAS Registration Confirm (RCF) message to the VMSC. The VMSC then creates the MS MM and PDP contexts for the MS and stores these contexts in its MS table.

Step 1.6. VMSC informs the MS that the location update request is accepted by the HLR. At this point, the registration procedure is completed.

4. The MS call origination

This section describes MS call origination where the MS is the calling party. The called party can be another MS in the same GPRS network or an H.323 terminal in the H.323 VoIP network. The called party can also be a traditional telephone set in the PSTN, which is connected indirectly to the GPRS network through the H.323 network. Without loss of generality, we assume that the called party is an H.323 terminal user. The signaling and voice paths are illustrated in figure 3. The message flow (figure 5) is described in the following steps:

Step 2.1. To originate a call, the standard GSM traffic channel assignment, authentication, and ciphering setup procedures [4,5] are performed to set up the radio link between the MS and the BTS. Then the digits dialed by the MS are sent to the BTS in a Um_Setup message. The BTS forwards the message to the VMSC through the GSM Abis and A interfaces.

Step 2.2. The VMSC sends the MAP_SEND_INFO_FOR_OUTGOING_CALL message to the VLR, which asks the VLR to check if the service requested by the calling party is legal, e.g., if the calling party is allowed to make this call. After authorization is successfully performed, the VMSC checks the PDP context record of the MS and identifies the routing path to the GGSN based on the GPRS tunnel ID and the GGSN number (see section 9.3, GSM 03.60 [7]).

Step 2.3. Based on the dialed digits, the VMSC starts the voice path setup procedure as follows. Through the RAS Admission Request (ARQ) and Admission Confirm (ACF) message pair exchange, the gatekeeper provides the VMSC the destination's call signaling channel transport address. The message path is (4) ↔ (3) ↔ (2) in figure 3.

Step 2.4. The VMSC sends the Q.931 Setup message to the destination through the GGSN. The Q.931 Call Proceeding message is sent back to the VMSC to indicate that enough routing information has been received and it does not expect to receive more routing information from the VMSC. The message path is (4) ↔ (3) ↔ (8) in figure 3.

Step 2.5. The H.323 terminal exchanges the RAS ARQ and ACF message pair with the gatekeeper. It is possible that an RAS Admission Reject (ARJ) message is received by

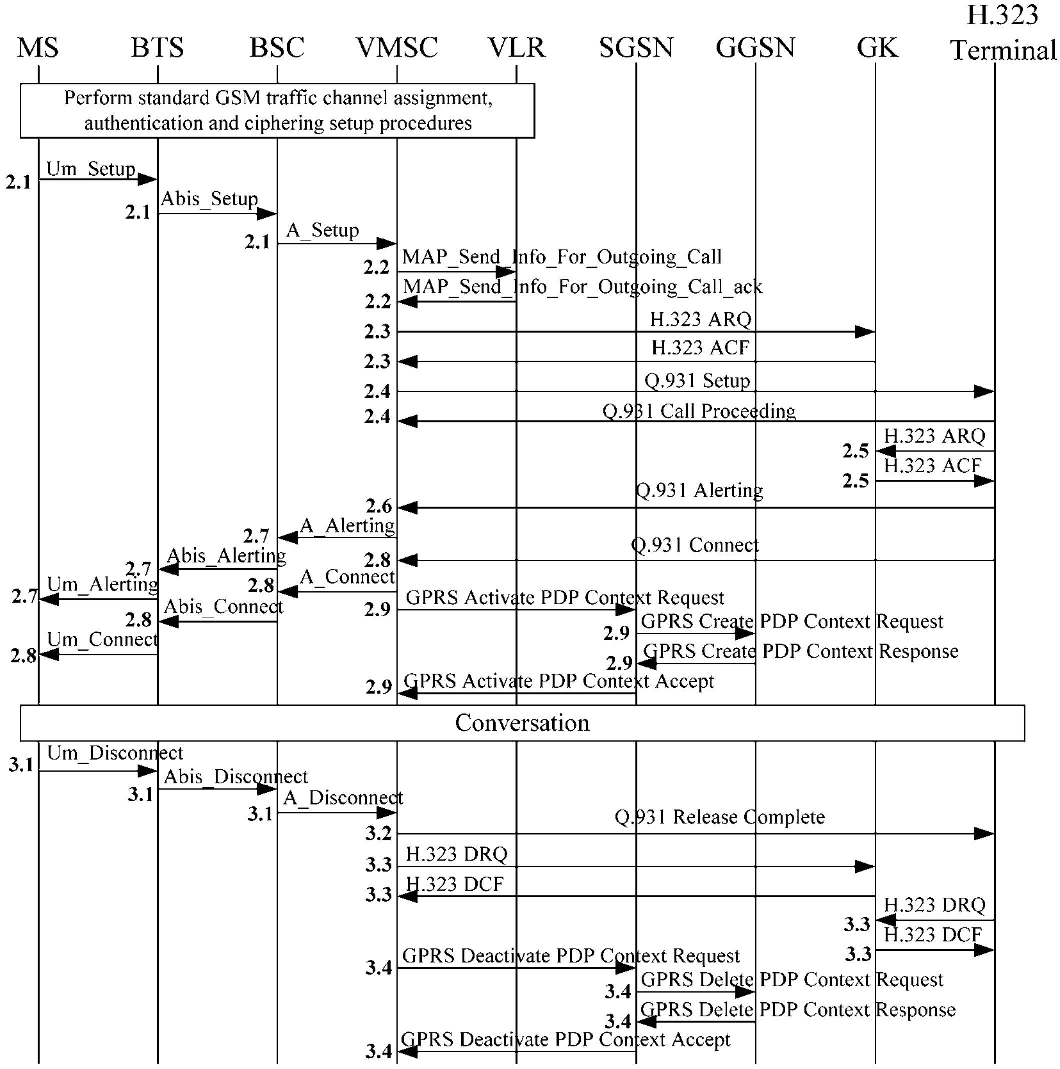

Figure 5. Message flows for MS call origination and call release.

the terminal and the call is released. The message path is (1) in figure 3. If the call admission is accepted, the following steps are executed.

Step 2.6. A ringing tone is generated at the H.323 terminal to alert the called party. The Q.931 Alerting message is sent to the VMSC.

Step 2.7. The VMSC sends A_Alerting to the BSC. The BSC sends an Abis_Alerting message to the BTS. The message is then forwarded to the MS through message Um_Alerting. This message triggers the ringback tone at the MS.

Step 2.8. When the called party answers the phone, the H.323 terminal generates the Q.931 Connect message to the VMSC. The VMSC sends the A_Connect message to the BSC. The BSC sends the Abis_Connect message to the BTS. The BTS forwards the message to the MS through the air interface Um.

Step 2.9. The VMSC performs the standard GPRS PDP Context Activation procedure to create another PDP context for real-time VoIP packet transmission.

At this moment, the call is established and the conversation begins. When the conversation is over, the call release procedure is described as follows (see steps 3.1–3.3 in figure 5). We assume that the calling party (the GSM user) hangs up the phone first.

Step 3.1. The MS sends a Um_Disconnect message to the BTS. The BTS then forwards this message to the VMSC through the BSC.

Step 3.2. The VMSC sends Q.931 Release Complete message to the H.323 terminal to release the call.

Step 3.3. Both the VMSC and the H.323 terminal inform the *GK* of call completion by exchanging RAS Disengage Request (DRQ) and Confirm (DCF) message pair. The *GK* records the call statistics for charging.

Step 3.4. At the end of the call, the VMSC performs the standard GPRS PDP context deactivation procedure to deactivate the voice PDP context established in step 2.9.

5. The MS call termination

This section describes the message flow for the MS call termination. We assume that the call is originated from an H.323 terminal, and the MS is the called party. The message flow (figure 6) is described in the following steps:

Step 4.1. The calling party sends the RAS ARQ message to the gatekeeper with the called party's address (i.e., the MSISDN of the MS). From the address translation table, the gatekeeper finds the IP address of the MS and returns it to the calling party through the RAS ACF message.

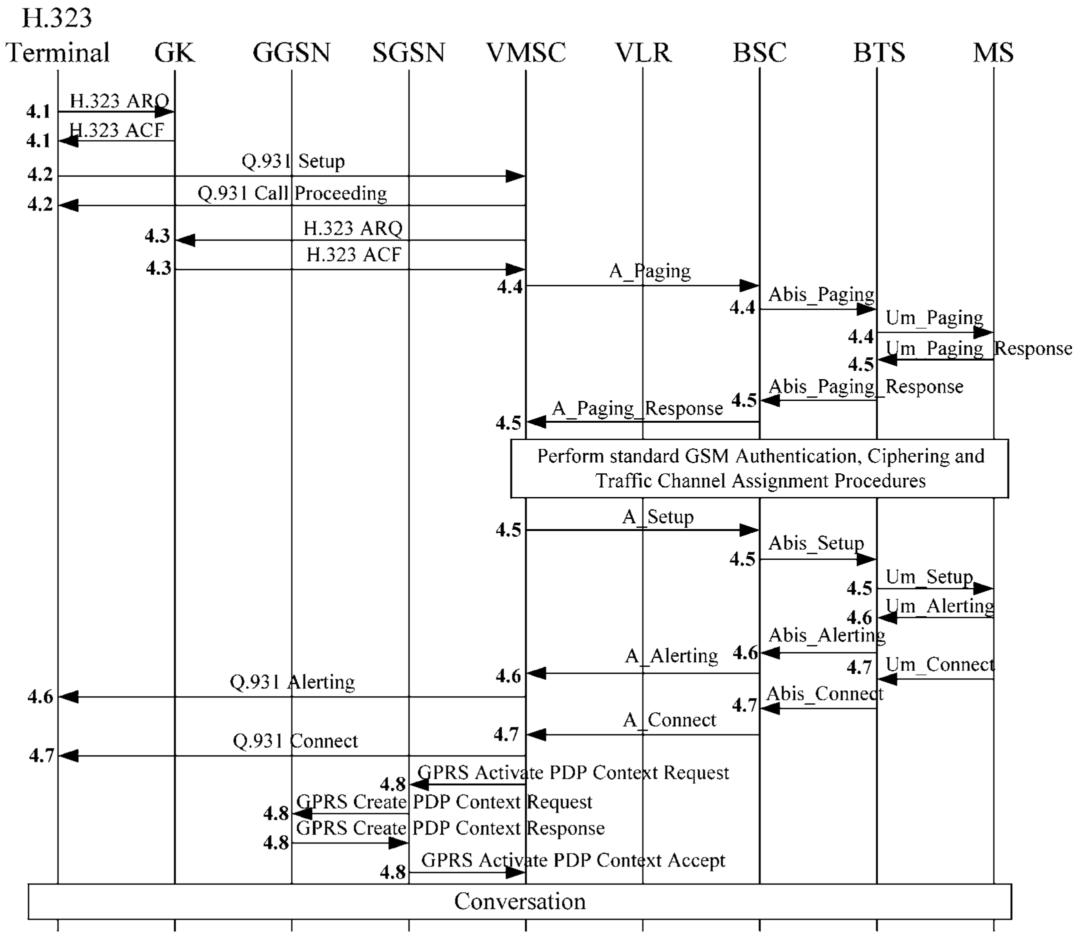

Figure 6. Message flow for MS call termination.

Step 4.2. The calling party sends the Q.931 Setup message to the VMSC through the GGSN. When the GGSN receives the Setup packet, it retrieves the PDP context of the MS based on the destination IP address of the packet. The GPRS tunnel ID and the SGSN number of the MS are identified from the PDP context. Then the GGSN routes the packet to the VMSC through the SGSN. The VMSC sends the Q.931 Call Proceeding message back to the H.323 terminal as in step 2.4.

Step 4.3. The VMSC exchanges the RAS ARQ and ACF message pair with the gatekeeper as in step 2.5.

Step 4.4. The VMSC sends message A_Paging to the BSC. The BSC sends an Abis_Paging message to the BTS. The BTS then pages the MS.

Step 4.5. Upon receipt of the paging response from the MS, the network (VMSC and VLR) performs the standard GSM traffic channel assignment, authentication, and ciphering procedures. The VMSC sends A_Setup to the BSC. The BSC sends an Abis_Setup message to the BTS. The BTS forwards this setup instruction to the MS.

Step 4.6. The MS rings and sends a Um_Alerting message to the BTS. The BTS forwards this signal to the VMSC through the BSC. The VMSC sends the Q.931 Alerting message to the H.323 terminal. The H.323 terminal generates the ringback tone.

Step 4.7. The called party answers the call and the MS sends a Um_Connect message to BTS. The BTS forwards this message to VMSC, which results in a Q.931 Connect message delivered to the calling party.

Step 4.8. Finally, the VMSC activates another GPRS PDP context for transmitting VoIP packets as in step 2.9.

At this moment, the call is established and the conversation begins.

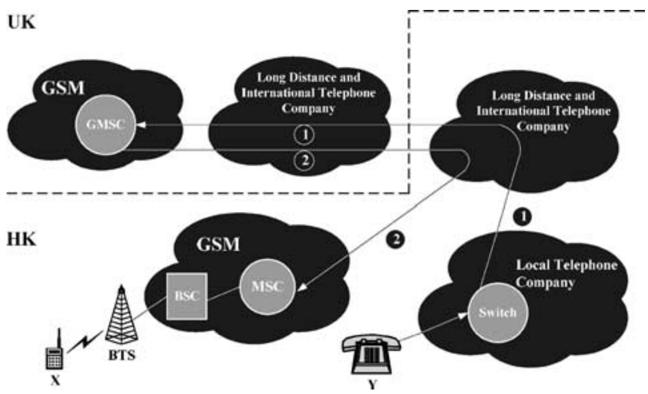

Figure 7. Tromboning phenomenon in GSM.

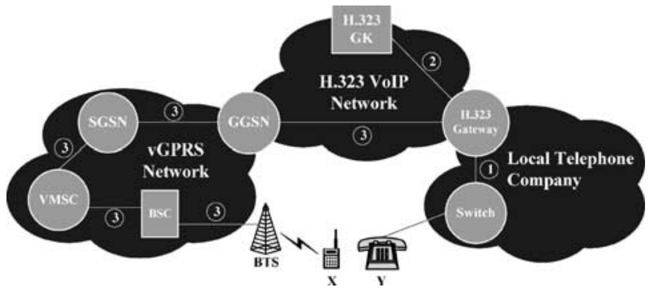

Figure 8. Tromboning elimination in vGPRS.

6. Comparing vGPRS and 3G TR21.978

This section compares vGPRS with the 3G TR21.978 approach. The following issues are discussed.

Real-time communication

vGPRS provides real-time communication that is required for VoIP. This is achieved by using circuit-switched mechanism in the GSM air interface. On the other hand, the 3G TR21.978 approach is affected by the non-real-time packet switching nature in the radio interface. Thus, VoIP with required quality can not be satisfied.

PDP context activation

In vGPRS, after an MS is attached to the GPRS network, a signaling PDP context is activated (see step 1.3, section 3). In GPRS, the PDP context must be activated before a routing path can be established between the GGSN and the MS. In vGPRS, when a call (either incoming or outgoing) to the MS arrives, the call path can be quickly established because the PDP context is already activated (see step 2.2, section 4, and step 4.2, section 5). 3G TR21.978 takes a different approach. In this approach, after the MS has registered to the H.323 gatekeeper, due to the network resource consideration, the PDP context is deactivated (see step 6, figure 7 in [1]). The PDP context must activated again when there are call activities to the MS (see step 1 of figure 8, step 6 of figure 9 in [1]). Thus, the SGSN and the GGSN do not need to maintain the PDP contexts of MSs when they are idle. In this scenario, however, the PDP context must be established and deactivated for every phone call. 3G TR21.978 does not provide details about how the PDP context can be activated for phone calls. We briefly discuss the 3G TR21.978 call path setup as follows. For MS call origination, the PDP context is activated as described in step 1.3, section 3. For MS call termination, a network initiated PDP context activation is required. As pointed out in [7], to perform this task, static PDP address is required (which may not be practical for a large-scaled network). Furthermore, IMSI is used by the GGSN (through GPRS PDP context activation) and the gatekeeper (through MAP_SENDING_ROUTING_INFORMATION and other MAP messages) to obtain MS information from the HLR. It implies that the H.323 gatekeeper should memorize IMSI. Since IMSI is considered confidential to the GPRS network operator, this approach may not work if the GPRS network and the H.323 network are owned by different service providers. When an H.323 terminal makes a call to the MS, the gatekeeper must find the IP address of the GGSN and the IMSI based on the dialed MSISDN. The gatekeeper then requests the GGSN to perform PDP context activation. After the PDP context activation, the routing path between the GGSN and the MS is

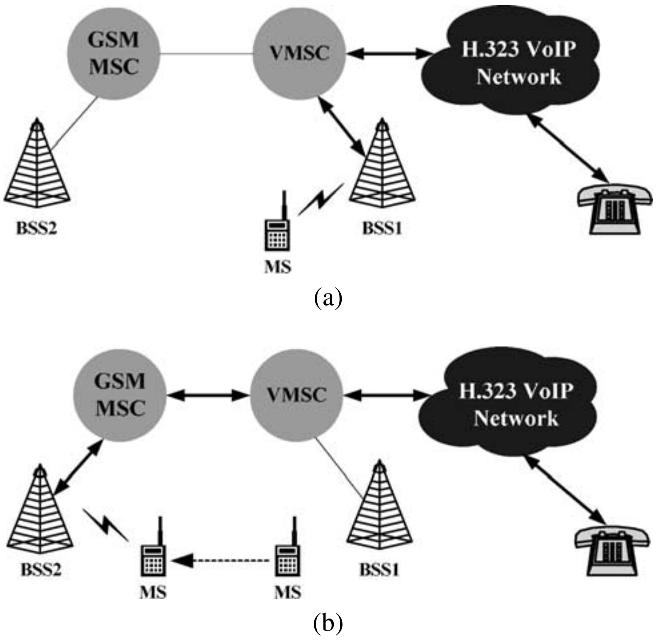

Figure 9. The routing path (a) before and (b) after intersystem handoff.

established, and the network can set up the call to the MS. Clearly, the call setup time is longer in this approach. vGPRS registration and call procedures can be easily modified to deactivate the PDP contexts when the MSs are idle. However, this approach may significantly increase the call setup time and is not considered in the current vGPRS implementation.

Modifications to the existing networks

To receive VoIP service in the 3G TR 21.978 approach, the MS must be an H.323 terminal. Also, the gatekeeper must equipped with GSM MAP because it needs to communicate with the HLR. In vGPRS, on the other hand, standard GSM and GPRS MSs can enjoy VoIP service. Furthermore, the *GK* is a standard H.323 gatekeeper, which only communicate with the GGSN using the standard H.323 protocol. However, vGPRS introduces a new component VMSC that will replace the existing MSC. Since VMSC is a router-based softswitch, this component is anticipated to be much cheaper than the existing MSC.

Tromboning elimination

In GSM, when a GSM user x in one country (say, the UK) roams to another country (say, Hong Kong) and someone y in Hong Kong makes a phone call to x, it will result in two international calls as described in [2]. As illustrated in figure 7, when y in Hong Kong dials x's MSISDN, the call is first routed to x's gateway MSC (GMSC), which is located in the UK (see (1), figure 7). After the GMSC has queried the HLR and the VLR (not shown in this figure), the location of x is identified, and a trunk is set up back to Hong Kong (see (2), figure 7). Thus, the call setup results in two international calls. If the local telephone company in Hong Kong connects to VoIP network (many local telephone companies are evolving into this configuration now), vGPRS can eliminate this tromboning situation; in other words, the call from y to x will be a local phone call, as explained below. When x roams from the UK to Hong Kong, it registers at the *GK* of Hong Kong. As shown in figure 8, when y at Hong Kong makes a phone call, the local telephone company first routes the call to the H.323 gateway through VoIP service (see (1), figure 8). The gateway checks with the *GK* to see if the entry for x can be found in the address translation table (see (2), figure 8). If so, the call setup follows the procedure described in section 5 (see (3), figure 8), which results in a local phone call, and the tromboning situation is avoided. On the other hand, if x is not found in the *GK*, the *GK* will instruct y to connect to the international telephone network as a normal PSTN call. Note that 3G TR21.978 cannot eliminate tromboning. To make a local call setup as illustrated in figure 8, the Hong Kong's gatekeeper needs to communicate with UK's HLR to set up the call (see step 2, figure 9 in [1]). This implies that the Hong Kong's gatekeeper needs to keep x's IMSI that is confidential to UK's HLR, which is not allowed in a real business model.

7. Conclusions

Recently wireless voice over IP (VoIP) has been intensively studied. 3GPP has proposed VoIP service for GPRS in 3G TR 21.978 specification, which nicely integrates GPRS with H.323-based VoIP network. In this approach, the mobile phones or mobile stations (MSs) must be equipped with vocoder and H.323 terminal capabilities. In this paper, we proposed vGPRS, a VoIP service for GPRS, which allows standard GSM and GPRS MSs to receive VoIP service.

In this approach, a new network element called VoIP mobile switching center (VMSC) is introduced to replace standard GSM MSC. The design of VMSC can co-exist with the standard GSM MSCs in our vGPRS system. The issue for coexistence is that the VMSC needs to communicate with the standard GSM MSC for inter-system handoff. This communication is achieved through standard SS7 MAP E protocol. Figure 9 shows the scenario before (see figure 9(a)) and after (see figure 9(b)) inter-system handoff. In this figure, the VMSC is an anchor MSC, which is always in the call path after inter-system handoff. The circuit-switched trunk connection between the VMSC and the GSM MSC is established through the standard GSM inter-system handoff procedure. Note that inter-system handoff between two VMSCs follows the same procedure.

The vGPRS approach is implemented using standard H.323, GPRS, and GSM protocols. Thus, existing GPRS and H.323 network elements are not modified. We described the message flows for vGPRS registration, call origination, call release and call termination procedures. We also showed that for international roaming, vGPRS can effectively eliminate the tromboning situation (two international trunks in call setup) for incoming call to a GSM roamer.

Acknowledgements

C.-I. Chiu provided useful comments regarding GPRS PDP context procedures. This work was sponsored in part by MOE Program of Excellence Research under contract 89-E-FA04-4, TAHOE Network, Ericsson, InterVideo, FarEastone, National Science Council under contract NSC 89-2213-E-009-203, the Lee and MTI Center for Networking Research, NCTU.

References

[1] 3GPP, 3rd Generation Partnership Project; Technical Specification Group Core Network; Feasibility Technical Report – CAMEL Control of VoIP Services, Technical Report 3G TR 21.978, Version 3.0.0, 3GPP (1999), Available at `http://www.3gpp.org`.

[2] Y.-J. Cho, Y.-B. Lin and C.-H. Rao, Reducing the network cost of call delivery to GSM roamers, IEEE Network Magazine 11(5) (1997) 19–25.

[3] ETSI, Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON): description of technical issues, Technical Report TR101300 V1.1.5, ETSI (1998).

[4] ETSI/TC, Mobile Radio Interface Layer 3 specification, Technical Report Recommendation GSM 04.08, ETSI (1993).

[5] ETSI/TC, Mobile Application Part (MAP) Specification, Version 4.8.0, Technical Report Recommendation GSM 09.02, ETSI (1994).

[6] ETSI/TC, European Digital Cellular Telecommunications System (Phase 2+); GPRS Base Station System (BSS) – Serving GPRS Support Node (SGSN) Interface; Gb Interface Layer 1, Technical Report Recommendation GSM 08.14, Version 6.0.0, ETSI (1997).

[7] ETSI/TC, GPRS Service description Stage 2, Technical Report Recommendation GSM 03.60, Version 7.0.0 (Phase 2+), ETSI (1998).

[8] ETSI/TC, GPRS Tunnelling Protocol (GTP) across the Gn and Gp Interface, Technical Report Recommendation GSM 09.60, Version 7.0.0 (Phase 2+), ETSI (1998).

[9] O. Granberg, GSM on the Net, Ericsson Review (4) (1998) 184–191.

[10] ITU, Call signalling protocols and media stream packetization for packet based multimedia communication systems, Technical Report ITU-T H.225, Version 3, International Telecommunication Union (1998).

[11] ITU, Packet-based multimedia communications systems, Technical Report ITU-T H.323, Version 3, International Telecommunication Union (1999).

[12] Y.-B. Lin and I. Chlamtac, *Mobile and Wireless Network Protocols and Services* (Wiley, New York, 2000).

[13] A.-C. Pang, Y.-B. Lin and Y.-R. Haung, System and method of providing voice communications for radio network, USA Patent Pending.

[14] H.C.-H. Rao, Y.-B. Lin and S.-L. Chou, iGSM: VoIP service for mobile networks, IEEE Communications Magazine 4(38) (2000) 62–69.

Ming-Feng Chang received the Ph.D. degree in computer science from the University of Illinois at Urbana-Champaign in 1991. He is currently an Associate Professor in the Department of Computer Science and Information Engineering, National Chiao-Tung University, Taiwan Republic of China. His research interests include Internet communication, mobile computing and VLSI system design. On-going research projects include "VoIP for Wireless Networks," "Interworking of VoIP Protocols" and "Cache Model for WAP Applications."

Yi-Bing Lin received his BSEE degree from National Cheng Kung University in 1983, and his Ph.D. degree in computer science from the University of Washington in 1990. From 1990 to 1995, he was with the Applied Research Area at Bell Communications Research (Bellcore), Morristown, NJ. In 1995, he was appointed as a professor of Department of Computer Science and Information Engineering (CSIE), National Chiao Tung University (NCTU). In 1996, he was appointed as Deputy Director of Microelectronics and Information Systems Research Center, NCTU. During 1997–1999, he was elected as Chairman of CSIE, NCTU. His current research interests include design and analysis of personal communications services network, mobile computing, distributed simulation, and performance modeling. Dr. Lin is an associate editor of IEEE Network, an editor of IEEE J-SAC: Wireless Series, an editor of IEEE Personal Communications Magazine, an editor of Computer Networks, an area editor of ACM Mobile Computing and Communication Review, a columnist of ACM Simulation Digest, an editor of International Journal of Communications Systems, an editor of ACM/Baltzer Wireless Networks, an Editor of Computer Simulation Modeling and Analysis, an editor of Journal of Information Science and Engineering, Program Chair for the 8th Workshop on Distributed and Parallel Simulation, General Chair for the 9th Workshop on Distributed and Parallel Simulation, Program Chair for the 2nd International Mobile Computing Conference, Guest Editor for the ACM/Baltzer MONET Special Issue on Personal Communications, a Guest Editor for IEEE Transactions on Computers Special Issue on Mobile Computing, and a Guest Editor for IEEE Communications Magazine Special Issue on Active, Programmable, and Mobile Code Networking. Lin is the author of the book Wireless and Mobile Network Architecture (co-author with Imrich Chlamtac; published by John Wiley & Sons). Lin received 1998 and 2000 Outstanding Research Awards from National Science Council, ROC, and 1998 Outstanding Youth Electrical Engineer Award from CIEE, ROC. Lin is an Adjunct Research Fellow of Academia Sinica.
E-mail: liny@csie.nctu.edu.tw

Ai-Chun Pang received her B.S.C.S.I.E. and M.S. C.S.I.E. degrees from National Chiao Tung University in 1996 and 1998, respectively. She is currently a Ph.D. candidate of the Department of Computer Science and Information Engineering, National Chiao Tung University. Her current research interests include design and analysis of personal communications services network, mobile computing and voice over IP.
E-mail: ijpong@csie.nctu.edu.tw