

VoIP Mobility in IP/Cellular Network Internetworking

Wanjiun Liao, National Taiwan University

Jen-Chi Liu, Industrial Technology Research Institute

ABSTRACT

This article explores VoIP mobility in the context of IP and cellular networks interworking. ITU-T Rec. H.323 gateways provide the interconnection between IP networks and switched circuit networks. They allow a call originating from an SCN phone to be transmitted over an IP network to an H.323 terminal, or bridged to another SCN phone. While H.323 provides interoperability with other SCN terminals, the major efforts have been focused on IP/wired SCN (PSTN, ISDN, etc.) interworking. In this article we discuss the challenges associated with the interworking between IP networks and cellular networks through H.323 gateways, and propose an innovative approach using the existing call transfer supplementary service to provide VoIP mobility in the H.323 IP telephony networks. The proposed approach uses existing components in the H.323 standard, thereby allowing VoIP mobility service in hybrid IP/cellular networks to be a value-added feature in the existing H.323-compliant Internet telephony systems.

INTRODUCTION

Internet telephony, also known as voice over IP (VoIP), promises to deliver real-time, two-way, synchronous voice traffic over the Internet or corporate intranets. The dominant standard of Internet telephony is International Telecommunication Union — Telecommunication Standardization Sector (ITU-T) Recommendation H.323 [1, 2]. H.323 specifies technical requirements for multimedia communications over packet-switched networks, including system components, control messages and functions for component communications, and services. Call setup and other call control signaling messages are carried out-of-band, sent through different paths from those for the payload traffic.

VoIP mobility refers to mobility in the scope of IP telephony, where mobility may include terminal mobility, user mobility, and service mobility. Terminal mobility denotes the ability of a terminal to change physical location while the connection is still maintained. User mobility is defined as the ability to communicate regardless of the terminal type in use. Service mobility is the ability of a user to obtain a particular service

independent of user and terminal mobility. The existing activities in the international standards bodies toward VoIP mobility include European Telecommunications Standards Institute (ETSI) Telephone and Internet Harmonization over Networks (TIPHON) Working Group (WG) 7 and ITU-T Study Group (SG) 16, H.323 Mobile Annex. TIPHON mobility addresses user and service mobility for VoIP services, and supports the concept of *global multimedia mobility*, which assumes that future terminals may connect to several types of access networks, and the choice of access should be made dynamically according to individual needs [3]. H.323 Mobile Annex addresses mobility issues in layer 2, timers, header compression, the H.323 system architecture, and all three aspects of mobility: terminal, user, and service.

Cellular networks offer seamless mobility support for voice services with high-quality connection at the expense of being built on complex and costly connection-oriented networking infrastructure that lacks the inherent flexibility, robustness, and scalability found in IP networks. H.323 allows interoperability between IP networks and the switched circuit network (SCN) through H.323 gateways, and hence may support a call across different cellular network types, say Global System for Mobile Communications (GSM) to H.323 to Advanced Mobile Phone System (AMPS). The current version of H.323, however, does not address interworking for cellular networks (i.e., the wireless SCN part) and IP networks. This missing component is indicated in Fig. 1 by the dotted lines. While H.324 gateways do support low-bit-rate transmissions — for example, public switched telephone network (PSTN), cellular phone — to H.323, the signals have to be transmitted through the PSTN to the H.323 system, making transmission very inefficient. Meanwhile, even if a new gateway type is introduced to perform signaling conversion between IP and cellular networks as shown in the dotted lines in Fig. 1, the call signaling defined in H.323 does not support seamless mobility in hybrid IP/cellular networks. We will examine the impairment of using the original H.323 signaling for VoIP mobility in a later section.

In this article a lightweight approach using the existing call transfer supplementary service (SS-CT) for VoIP mobility is proposed. Unlike

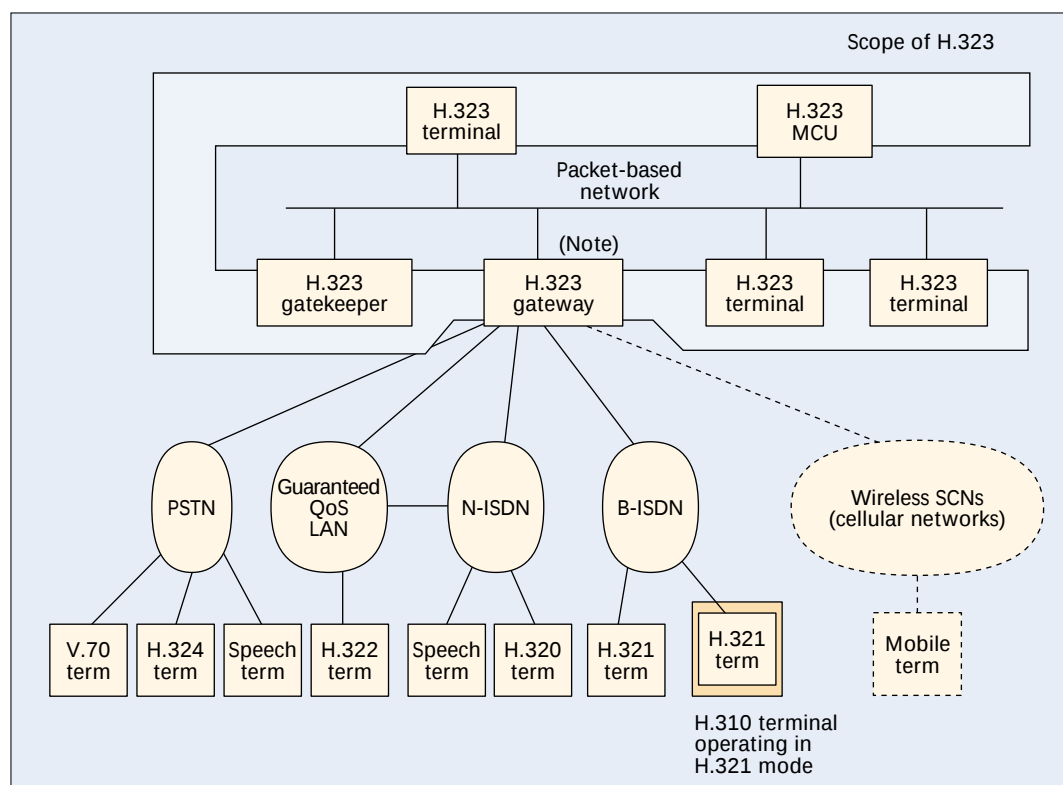


Figure 1. Interoperability of H.323 terminals. Note: A gateway may support one or more of the PSTN, PCS, N-ISDN, and/or B-ISDN connections.

H.323 allows the interoperability between IP networks and switched circuit network through H.323 gateways, and hence may support a call across different cellular network types. The current version of H.323, however, does not address the interworking for cellular networks (i.e., wireless SCN part) and IP networks.

previous work [4] that considered VoIP mobility in H.323 terminals, using dynamic join and departure of an ad hoc multipoint conference to handle connection handoff, we would like to study IP/cellular network interworking in this article. The proposed approach makes use of the existing components in the H.323 standard, thereby allowing VoIP mobility in hybrid IP/cellular networks to be a value-added feature in existing H.323-compliant Internet telephony systems.

The rest of the article is organized as follows. First, we present a brief summary of the H.323 architecture and its supplementary services. Next, we identify the problems associated with interworking between IP networks and wireless SCNs through H.323 gateways, and describe the proposed approach as well as the associated signaling messages for VoIP mobility. Finally, our concluding remarks are included.

H.323: AN OVERVIEW

This section begins with a brief summary of the H.323 entities and protocols involved, followed by a description of the services using these entities and protocol to make a call and the associated supplementary services.

H.323 ENTITIES AND SIGNALING PROTOCOLS

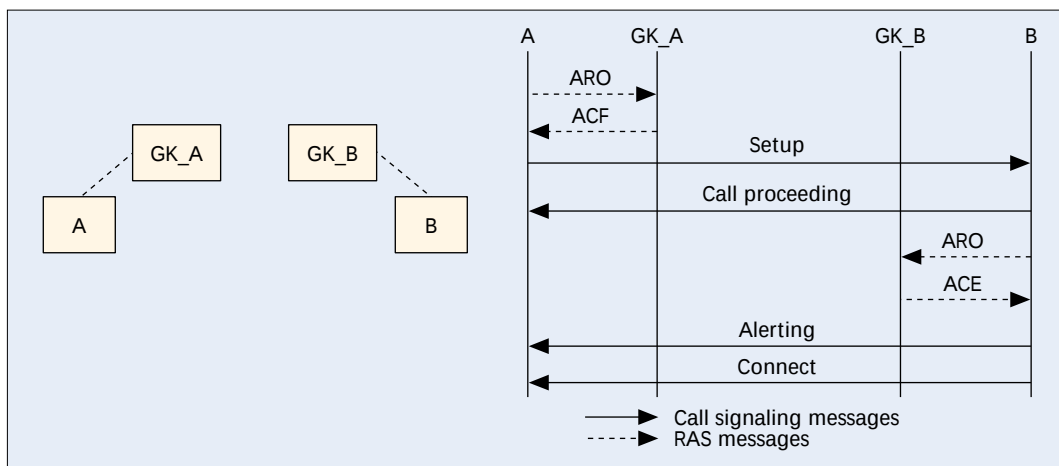
ITU-T Recommendation H.323 is an umbrella standard (a series of recommendations) to describe system components, control messages, and procedures among components, and services for multimedia communications over IP-based networks. The major system components include the H.323 terminal, gateway (GW), gatekeeper

(GK), and multipoint control unit (MCU). The H.323 terminal is an IP host with the capability to terminate H.323 signaling and payload channels at the end user. The GW which bridges the circuit-switched phone networks and the packet-switched data networks performs signaling translation and possible media transcoding between the H.323 terminal and other SCN terminal types. The gatekeeper provides control services to the system, including address translation, access control, zone management, and authentication, authorization, and accounting (AAA) services. A zone is defined as a collection of H.323 endpoints¹ managed under a single GK. The MCU serves a multipoint conference call, namely, a call taking place among three or more terminals.

In the call model of H.323, the signaling for basic call setup and teardown between two endpoints is defined in H.225.0 [5], and that for end-to-end control operations as well as capability exchange in H.245 [6]. H.225.0 RAS specifies signaling between an endpoint and the GK for service registration, admission, and operation status. Unlike both H.225.0 call signaling channels and H.245 control channels, which are transported through reliable Transmission Control Protocol (TCP) connections, RAS channels are delivered through unreliable User Datagram Protocol (UDP) transmissions. H.235 includes signaling for user authentication and data encryption. H.341 defines the management information base (MIB) for the control and management of an H.323 network. The H.450.x [7] series specify the supplementary services of IP telephony, such as call forwarding and call waiting between endpoints.

¹ An endpoint may be a terminal, GW, or MCU, which terminates signaling and payload channels.

H.225.0 defines signaling for basic call setup and release. The signaling messages may be sent directly between the endpoints (direct endpoint call signaling) or routed through GK (GK-routed call signaling).



■ Figure 2. Call setup by direct endpoint call signaling.

BASIC CALL AND SUPPLEMENTARY SERVICES IN H.323

H.225.0 defines signaling for basic call setup and release.² The signaling messages may be sent directly between the endpoints (direct endpoint call signaling) or routed through GKs (GK-routed call signaling). In both approaches, an endpoint should request admission from the GK before connection establishment and request disengagement after connection release, if the endpoint is registered to the GK. As a result, gatekeepers can keep track of all the registered system entities and manage the system resources properly.

Figure 2 shows an example of direct endpoint call signaling, where endpoints A and B are registered to GKs GK_A and GK_B, respectively. Endpoint A first initiates an admission request (ARQ) to gatekeeper GK_A. If the call is allowed, GK_A sends in reply the resolved address of endpoint B in an admission confirmation (ACF); otherwise, it rejects the request using admission reject (ARJ). Endpoint A then issues the Setup message to B using the resolved address. Endpoint B returns the Call Proceeding message to indicate that the setup message is in process. If B would like to accept the call, an ARQ/ACF exchange takes place between B and GK_B, followed by the Alerting message sent to calling endpoint A to indicate that the called party has been notified of the incoming call. Finally, endpoint B responds with the Connect message to A, indicating that the H.225.0 call signaling channel has been established.

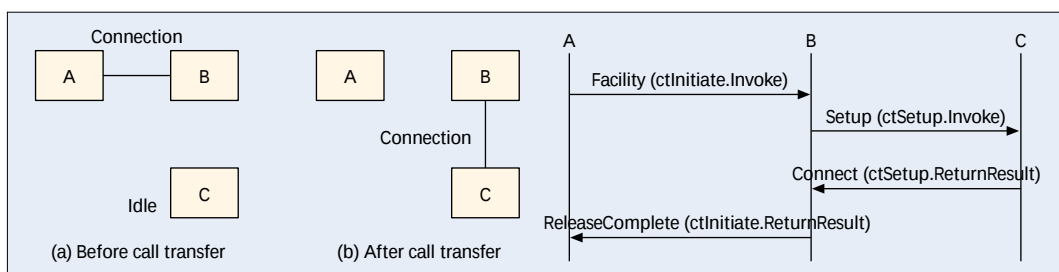
The H.450.x series specify signaling between H.323 entities for supplementary services, including call transfer, call forwarding, call hold, call

waiting, message indication, and N-way conference [8]. Figure 3 shows the operation of SS-CT. SS-CT [9] enables the served endpoint A to transform an existing call between A and B into a new call between endpoint B and an endpoint C selected by A. A call between endpoints A and C is not required to exist prior to the call transfer. On the invocation of call transfer, endpoint A sends a Facility message with call transfer initiate *ctInitiate.Invoke* to endpoint B. The original connection between A and B is retained until a success indication on call transfer is received. Upon receipt of the call transfer request, endpoint B initiates a call establishment toward endpoint C by sending the Setup message with call transfer setup *ctSetup.Invoke* to endpoint C. If C wishes to accept the call, the Connect message with *ctSetup.ReturnResult* is returned. Endpoint B in turn issues the Release Complete message with *ctInitiate.ReturnResult* back to A, indicating that the call transfer has succeeded.

INTERWORKING IN THE HYBRID IP/CELLULAR NETWORKS

Figure 4 shows a typical infrastructure of a hybrid IP/GSM³ network and its major components. The base station (BS) serves as the interface of the mobile host (MH) to the GSM network. It is usually located at the center of a cell and controlled by the base station controller (BSC). The BSC performs handoff, radio channel management, and other activities between the BSs and the MHs. The mobile switching center (MSC) is the heart of GSM, responsible for

² H.245 control signaling follows H.225.0 call signaling, performing operations such as capability exchange, master/slave determination, and logical channel establishment for media transmission. Since the major focus of this article is on H.225.0 call signaling, we will not include H.245 signaling.



■ Figure 3. The call transfer supplementary service.

The H.450.x series specify signaling between H.323 entities for supplementary services, including call transfer, call forwarding, call hold, call waiting, message indication, and N-way conference.

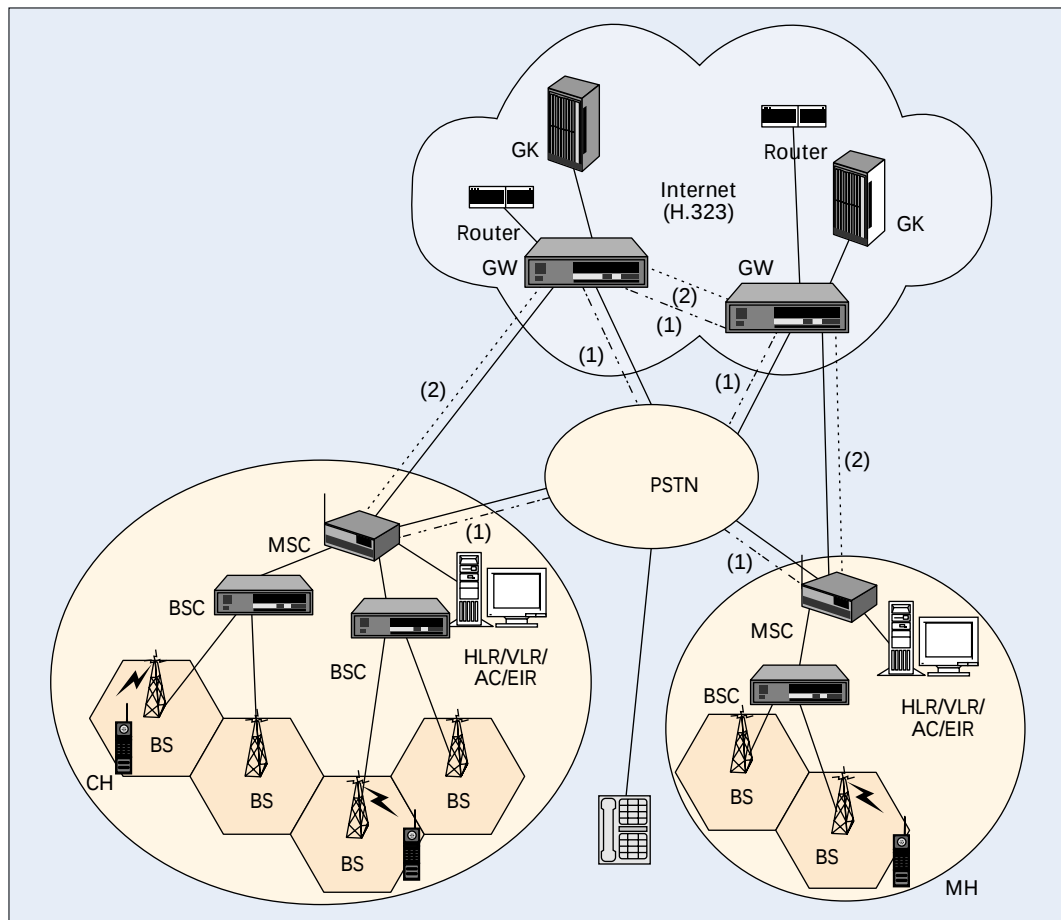


Figure 4. An example of hybrid IP/cellular networks.

connection setup/release/management and call routing to the proper cell. An MSC provides an interface to the public switched telephone network (PSTN), relaying calls between the PSTN and the cellular networks. The roaming capability of GSM is provided through the cooperative support of the home location register (HLR) and visitor location register (VLR).

Assume that the MH in cellular network 2 would like to call the CH (corresponding mobile host) in cellular network 1. Conventionally, the connection between MH and CH is established by the MSCs through the PSTN, as indicated in Path (1) (i.e., MSC–PSTN–MSC) in Fig. 4. The H.323 gateways perform signaling conversion, and hence allow calls to be operated across different cellular networks. Under the current version of H.323, again, the signals are relayed through the PSTN to the H.323 gateways. Path (1) (i.e., MSC–PSTN–GW–GW–PSTN–MSC) in Fig. 4 depicts the interoperability with cellular networks defined in the current H.323 standard, which is inefficient because the path has to go through the PSTN. We would like to go directly to the IP networks through the GWs, as indicated in Path (2) in Fig. 4.

In the following, we will first examine the problem of using current H.323 signaling to support IP/cellular network interworking for VoIP mobility (for both paths (1) and (2), it turns out existing H.323 signaling is inadequate), and then present the proposed approach to overcome the

identified problem and hence to allow interoperability that follows Path (2).

THE PROBLEM OF IP/CELLULAR NETWORKS INTERWORKING THROUGH H.323 GATEWAYS

Figure 5 shows an MH communicating with the CH in the simplified topology of Fig. 4. Assume that the MH is registered with GK_H through GW_H from MSC_H in cellular network 2, and the CH is registered with GK_R through GW_R from MSC_R in cellular network 1. Under the current version of H.323, both the MH and CH are allowed to roam within the location area associated with the corresponding GW while the conversation is taking place. The connection may be broken, however, if a mobile host roams to a location area not connected to the original GW.

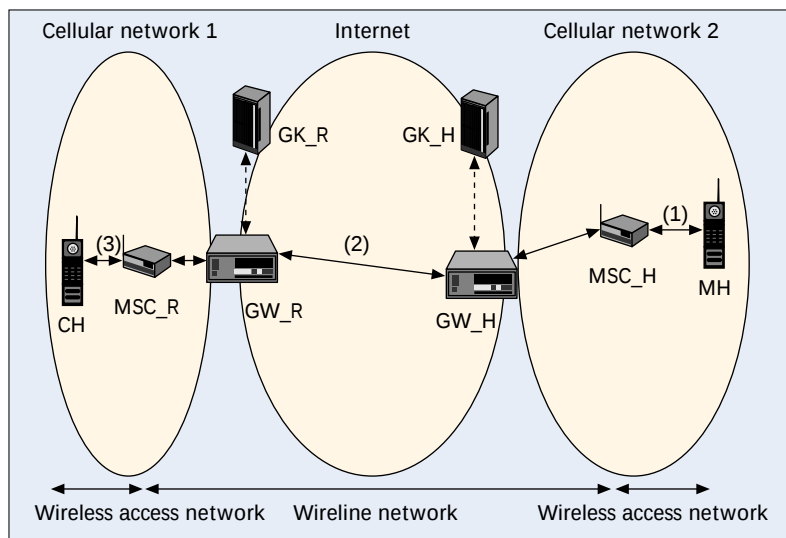
Figure 6 illustrates this phenomenon. The connection between MH and CH comprises three segments:

- MH to MSC_H
- MSC_H to GW_H to GW_R to MSC_R
- MSC_R to CH

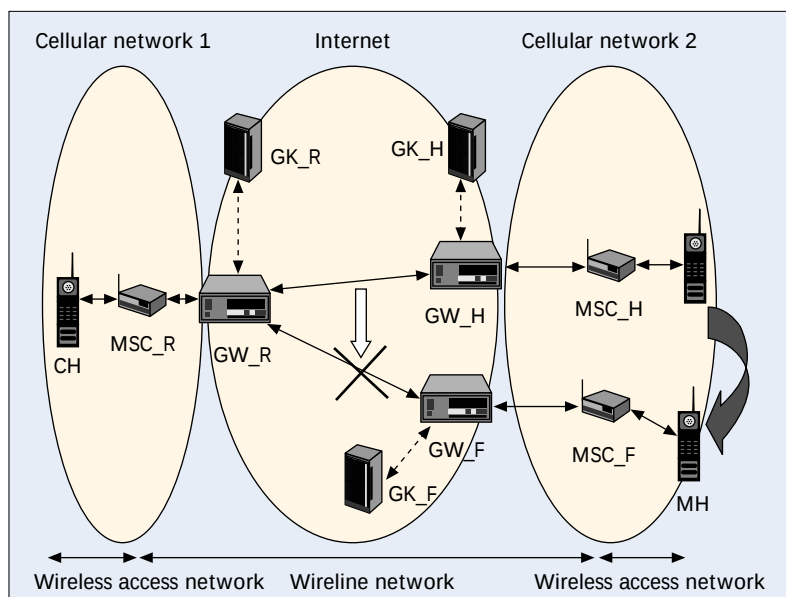
Namely, these are two wireless segments and one wired segment.

Suppose the MH is handed off from MSC_H to MSC_F. Since both cellular networks 1 and 2 support seamless roaming, as mentioned above, connection handoff is taking place transparently in segments (1) and (3) guided by the underlying system in use. Segment (2), however, is not

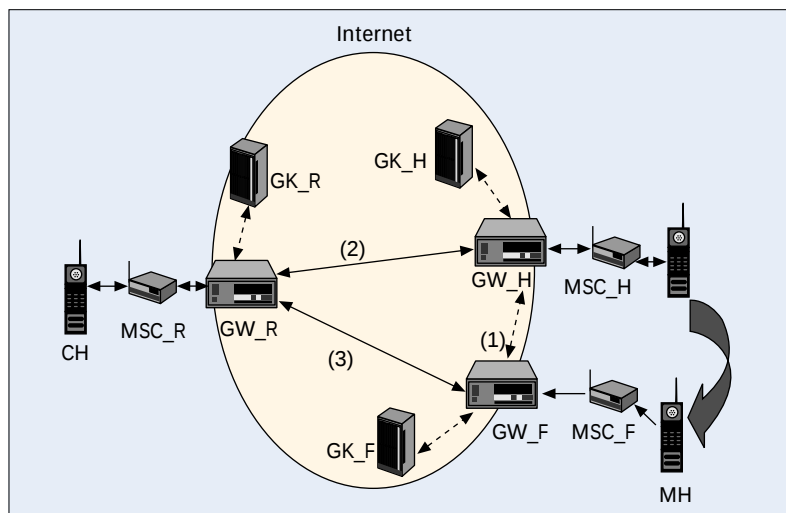
³ In this article we use the term *cellular network* to denote any wireless network with mobility and roaming support, such as GSM, Digital Enhanced Cordless Telephone (DECT), and so on. For ease of explanation, we will use GSM as an example to illustrate how our proposed approach works in the rest of the article. This approach may also be applied to other cellular systems.



■ **Figure 5.** An example of an SCN-H.323-SCN call between an MH and a CH.



■ **Figure 6.** The MH roams from the location area with GW_H to that with GW_F.



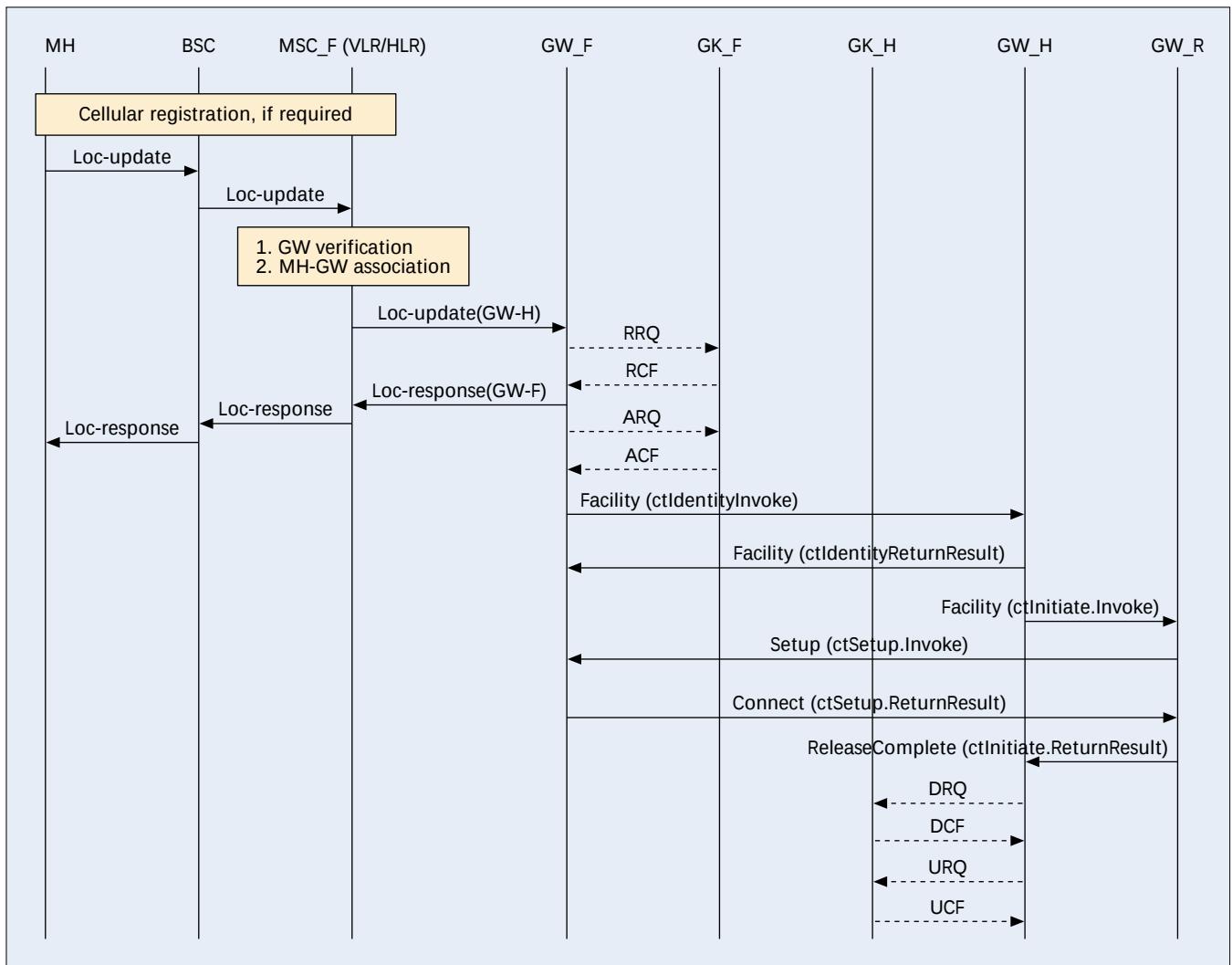
■ **Figure 7.** The SS-CT-based approach to VoIP mobility in hybrid IP/cellular networks.

handed off along with the handoff operation and location update performed in the cellular networks, because the call signaling defined in H.323 does not support mobility in the wireless SCNs. As a result, the MH suffers from a broken connection, and needs to explicitly set up a new connection from scratch to resume the ongoing conversation with the CH.

A CALL-TRANSFER (SS-CT)-BASED APPROACH TO VOIP MOBILITY

In this section we present the proposed approach for VoIP mobility in hybrid IP/cellular networks. Recall that H.450/H.323 defines SS-CT, allowing an existing call to be transferred from a transferring endpoint to the endpoint selected by the transferring endpoint. Taking advantage of the wired characteristics of the connections between MSC and GW, and the seamless roaming support of the cellular networks, the proposed approach uses the modified call transfer signaling procedure plus location update capability supported by the underlying cellular system (say GSM) to realize IP/cellular network interworking. Thus, in addition to recording the regular information about GSM subscribers, the HLR/VLR stores the GW identification associated with a registered mobile terminal. Using the stored GW information associated with the visiting terminal, an MSC can tell if the mobile terminal is roaming from the location area associated with a different GW, and hence launch a handoff if necessary. For example, the HLR/VLR with MSC_H associates GW_H with the MH; similarly, that with MSC_R stores GW_R with the CH.

Figure 7 illustrates the steps of the SS-CT-based approach for VoIP mobility, and Fig. 8 shows the corresponding call signaling messages. When the MH is roaming from the location area controlled by MSC_H to that by MSC_F, a location update is performed. Upon receipt of the location update request, MSC_F (HLR/VLR) compares its directly connected GW (GW_F) with the GW associated with the MH stored in the HLR/VLR (GW_H). If both are identical, it means that the MH still stays in the area associated with the same GW, and no extra operation other than regular GSM location update is required; otherwise, a handoff in the wired part of the connection must be performed. To initiate the handoff, MSC_F sends a location update indication to the previous GW information to the directly associated GW_F. GW_F then initiates RRQ/RCF and ARQ/ACF exchanges with GK_F if the gatekeeper is in use. Using a Facility message to GW_H, the previous GW of the MH, GW_F requests GW_H to invoke a call transfer to the transferred endpoint GW_R, followed by the return Facility message in response to the request. GW_H then starts the regular call transfer procedure to GW_R, asking for a call transfer to GW_F. Finally, GW_H initiates disengage request/disengage confirmation (DRQ/DCF) and unregister request/unregister confirmation (URQ/UCF) exchanges with GK_H, releasing the reserved resources and completing the connection handoff.



■ **Figure 8.** Call signaling of the SS-CT-based approach to IP/cellular network interworking.

CONCLUDING REMARKS

We have proposed the SS-CT-based approach to realize VoIP mobility in hybrid IP/cellular networks. The potential problem of using the original H.323 call signaling to provide IP/cellular networks interworking for VoIP mobility has been identified, and the proposed approach as well as the associated signaling messages have been demonstrated. The SS-CT approach does not introduce any extra component to the current H.323 standard, thereby allowing VoIP mobility service in hybrid IP/cellular networks to be a value-added feature in existing H.323-compliant Internet telephony systems.

REFERENCES

- [1] ITU-T Rec. H.323v2, "Packet Based Multimedia Communications Systems," Mar. 1997.
- [2] G. A. Thom, "H.323: the Multimedia Comm. Standard for Local Area Networks," *IEEE Commun. Mag.*, Dec. 1996, pp. 52–56.
- [3] ETSI TIPHON, "Analysis of Existing Roaming Techniques Applicable to TIPHON Mobility Services," TR 101 338 V1.1.2, May 1999.
- [4] Wanjiun Liao, "Mobile Internet Telephony: Mobile Extensions to H.323," *Proc. IEEE INFOCOM '99*, New York, Mar. 1999; also in ETSI TIPHON TR 101 338 V1.1.2, 1999.

- [5] ITU-T Rec. H.225.0, "Media Stream Packetization and Synchronization for Visual Telephone Systems on Non-Guaranteed Quality of Service LANs."
- [6] ITU-T Rec. H.245, "Control Protocol for Multimedia Communication," Mar. 1996.
- [7] ITU-T Rec. H.450.1, "Generic Functional Protocol for the Support of Supplementary Services in H.323," Sept. 1997.
- [8] M. Korpi and V. Kumar, "Supplementary Services in the H.323 IP Telephony Network," *IEEE Commun. Mag.*, July 1999, pp. 118–25.
- [9] ITU-T Rec. H.450.2, "Call Transfer Supplementary Service for H.323," Sept. 1997.

BIOGRAPHIES

WANJIUN LIAO (wjiao@cc.ee.ntu.edu.tw) received B.S. and M.S. degrees from National Chiao Tung University, Taiwan, in 1990 and 1992, respectively, and a Ph.D. degree in electrical engineering from the University of Southern California (USC), Los Angeles, in 1997. Since August 1997 she has been an assistant professor in the Department of Electrical Engineering, National Taiwan University, Taipei. Her research interests include multimedia communications and wireless Internet. A member of the Phi Tau Phi scholastic honor society, she was a recipient of the outstanding research paper award at USC in 1997.

JEN-CHI LIU (ccliu@ccl400.ccl.itri.org.tw) received his B.S. and Ph.D. degrees from National Chiao Tung University, Taiwan, in 1991 and 1997, respectively. Since August 1997 he has been a section manager at the Computer and Communication Laboratories, Industrial Technology Research Institute, Hsinchu, Taiwan. His research interests include multimedia networking, wireless Internet, high-performance transport system, ASIC design, and embedded system design.