Mobile Internet telephony: mobile extensions to H.323

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Abstract Internet telephony realizes the transmission of two-way and real-time traffic over IP-based networks. The dominant standard for Internet telephony is ITU-T Rec. H.323. With the current version of H.323, Internet telephony allows interoperability with circuit-switched telephone, but IP host mobility is not supported. In this paper, mobile extensions to H.323 that enable mobile Internet telephony service are proposed. The proposed approach combines the characteristics of both cellular phone system and mobile IP mechanism with Internet telephony, and therefore realizes the transmission of real-time voice traffic for both stationary and mobile hosts over IP-based networks. Two alternatives for mobility extensions to H.323 are described, using the same call signaling procedure. Through proper call setup signaling with the H.323 Gatekeeper, the address of the target endpoint can be resolved before call establishment, enabling the service redirection to be completely handled in the application layer. Therefore, our approach allows mobility support without the need for additional new entities, and with minimal modifications to existing H.323 standard. Such mobility extensions can serve as an add-on feature for the existing Internet telephony systems compliant to the H.323 standard.

Keywords: Internet telephony, H.323, Voice over IP, mobile Internet telephony, mobile extensions to H.323

1. Introduction

Internet telephony, also known as voice over IP or IP telephony, promises to deliver real time, two-way, synchronous voice traffic over the Internet or corporate Intranets. The basic concept behind IP telephony is simple: segmenting voice into a series of packets and transmitting them across an IP network to be reassembled at the receiving end. Dozens of IP telephony applications are available on the Internet, including Microsoft NetMeeting [1], VocalTech IPhone [2], VAT [3], RAT [4], Novet [5], just to name a few. The dominant standard of Internet telephony is the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) Recommendation H.323 [6,7]. H.323 specifies technical requirements for multimedia communications over packet switched networks, including system components, control messages and functions for component communications. Call setup and other call control signaling messages are carried out-of-band, sent through different paths from those for the payload traffic. Such separation creates a variety of opportunities to introduce new services for advanced Internet telephony systems, just like the separation of signaling from voice traffic on the Public Switched Telephone Network (PSTN) creates opportunities for advanced Intelligent Network services.

The telephone was mainly designed to serve real-time voice traffic, and hence the connection-oriented service model is desirable. The connection-oriented service model is basically composed of three phases, namely, call establishment, call transfer, and call termination. The IP networks, however, provides packet switched, connectionless, datagram services. These three phases to make a call on an IP network is therefore simulated through higher level protocols.

Compared to Plain Old Telephone Service (POTS) (i.e., regular phone), Internet telephony allows lower charges and more flexibility in the expansions of versatile capabilities and advanced features, such as voice mail, multipoint conference call, and shared whiteboard, and ease of integration of a variety of media. While Internet telephony enjoys many advantages, the market penetration rate of POTS phones is very high. The interoperability between circuit-switched POTS phone and packet-switched Internet telephony can be realized by the deployment of an Internet telephony gateway defined in H.323.

Another attractive phone service is mobility support, i.e., the ability to maintain communications while moving. This can be seen in the rapidly explosive usage of cellular phones. With the increasing popularity of PDA, lap-top computers, wireless LANs, and the desire for global connectivity, host mobility on the Internet has become an important issue. It may be desirable to integrate such mobility features to Internet telephony as well. For example, when a user originates a POTS phone call to an IP telephone on the Internet, the IP host may not be stationary or fixed. It is likely that the IP host is currently on the move from networks to networks, or the point of attachment of the IP host may have changed. Like cellular phone services, mobile Internet telephony demands seamlessly roaming while conversation is in progress.

Under the current version of H.323 (version 2), however, mobile phone service is forbidden, resulting from the underlying IP mechanism implicitly assuming that a host is fixed. As a result, no signaling messages are defined in H.323 for roaming, handoff, location tracking, or location update. In addition, voice over IP is a real-time connection-oriented service over packet-switched IP-based networks. Neither the circuit-switched cellular phone approach nor the connectionless Mobile IP mechanism is capable of providing mobile Internet telephony service. Mobile IP inherits the connectionless nature from IP and lacks the concept of a connection for a phone call. Thus, connection handoff for mobile phone service should be explicitly performed by H.323. While Mobile IP is an important network layer mechanism to allow host mobility on the Internet, the potential problem involved on the integration of Mobile IP with RSVP [8] renders Mobile IP inadequate for real-time services. The reason is as follows. An RSVP Path message destined to a mobile host may be intercepted by the Mobile IP home agent of the mobile host, which in turn encapsulates and transmits the Path message to the destination through IP tunneling. Being encapsulated in an IP datagram, a Path message is not able to set the intermediary routers along a routing path for resource reservations. Still, Mobile IP has not yet been deployed as widely as IP. Running mobility-enabled H.323 over IP requires less effort than the modifications on the OS kernel to support Mobile IP. Thus, it may be desirable to have a new light-weight approach to permit mobile IP telephony services using IP mechanism.

In this paper, an approach that enables mobile Internet telephony services from mobile extensions to H.323 is proposed. Through proper call setup signaling with the H.323 Gatekeeper, the address of the target endpoint (i.e., callee or called party) can be resolved before call establishment, enabling the service redirection to be completely handled in the application layer. Thus, it realizes mobile IP telephony services with IP rather than Mobile IP. We will demonstrate that H.323 provides an excellent environment for mobility support. Our approach introduces no additional new system entities and requires minimal modifications to the H.323 standard, allowing mobile IP telephony service to be a valued-added feature in the existing H.323-compliant Internet telephony systems.

The rest of the paper is organized as follows. An overview of ITU-T Rec. H.323 is presented in Section 2. The fundamentals and architecture of a mobile Internet telephony system extended from the H.323 standard are described in Section 3. Call signaling procedure for mobility management is proposed in Section 4. An alternative for mobility support from IP multicasting is examined in Section 5. The implementation aspects of mobile IP telephony are mentioned in Section 6. Finally the concluding remarks are included in Section 7.

2. H.323: an overview

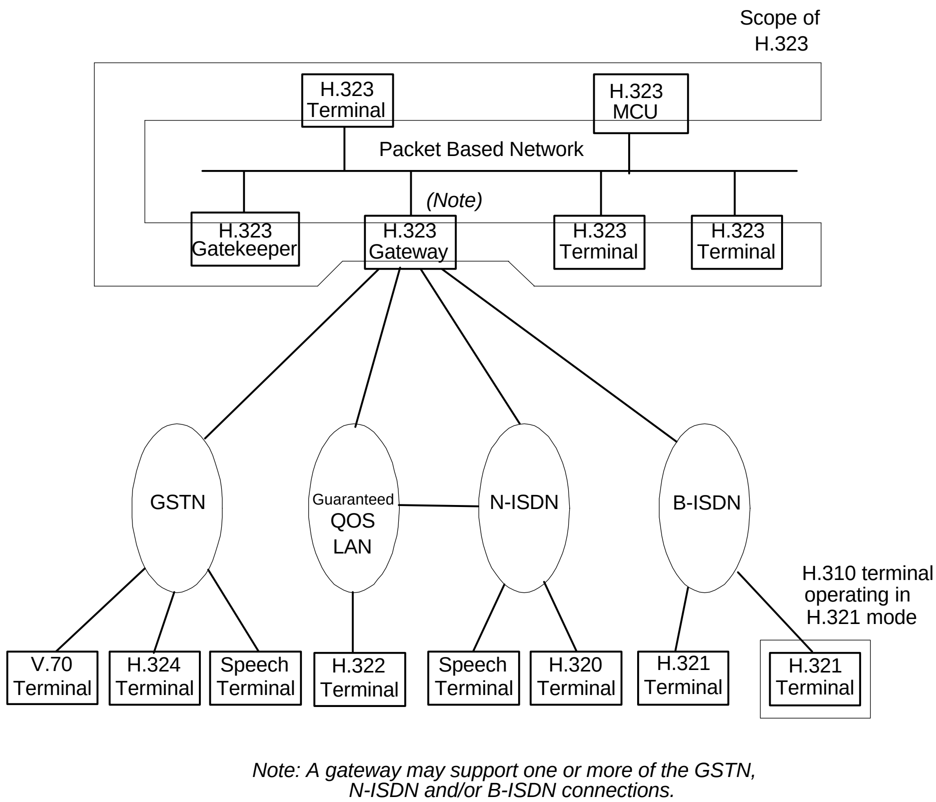


Figure 1. H.323 system architecture [7]

ITU-T Rec. H.323 is an umbrella standard that references many related ITU Recommendations for Internet telephony. H.323 specifies system components, and control messages and procedures that define communications among components. The major system components as shown in Fig. 1 include H.323 terminal, Internet Telephony Gateway (ITG), Gatekeeper (GK), Multipoint Control Unit (MCU), Multipoint processor (MP) and Multipoint Controller (MC). The H.323 terminal is an endpoint in an IP host with the capability of two-way and real-time communications. The ITG bridges the circuit-switched telephone network[1] and the packet-switched data network. It allows a call originating from a POTS phone to be transmitted over an IP network to an IP host, or bridged to another POTS phone. The Gatekeeper provides such call control services as address translation and access control to the system. The MCU, MP and MC are mainly used for a multipoint conference, namely, a conference taking place among three or more terminals. An MCU is required only when a centralized multipoint conference takes place, where all participating terminals communicate via the MCU in a point-to-point manner. To support different conference types, an MCU has been divided into one MC and zero or more MPs, with the MC for control access and the MPs for media multiplexing. The MC is required for all multipoint conference types. It can be located in a terminal, ITG, or Gatekeeper in decentralized multipoint conferences in which the MCU is not in use.

In the call model of H.323, connection setup and torn-down are specified in H.225.0 [9] and end-to-end operation control, in H.245 [10]. Fig. 2 shows the protocol stack for the call management and other signaling of H.323/H.225.0/H.245 [7], where media streams and RAS channels (i.e., for registration, admission, and status messages) are delivered through unreliable UDP transmissions, and call signaling channels and control channels are transported through reliable TCP connections.

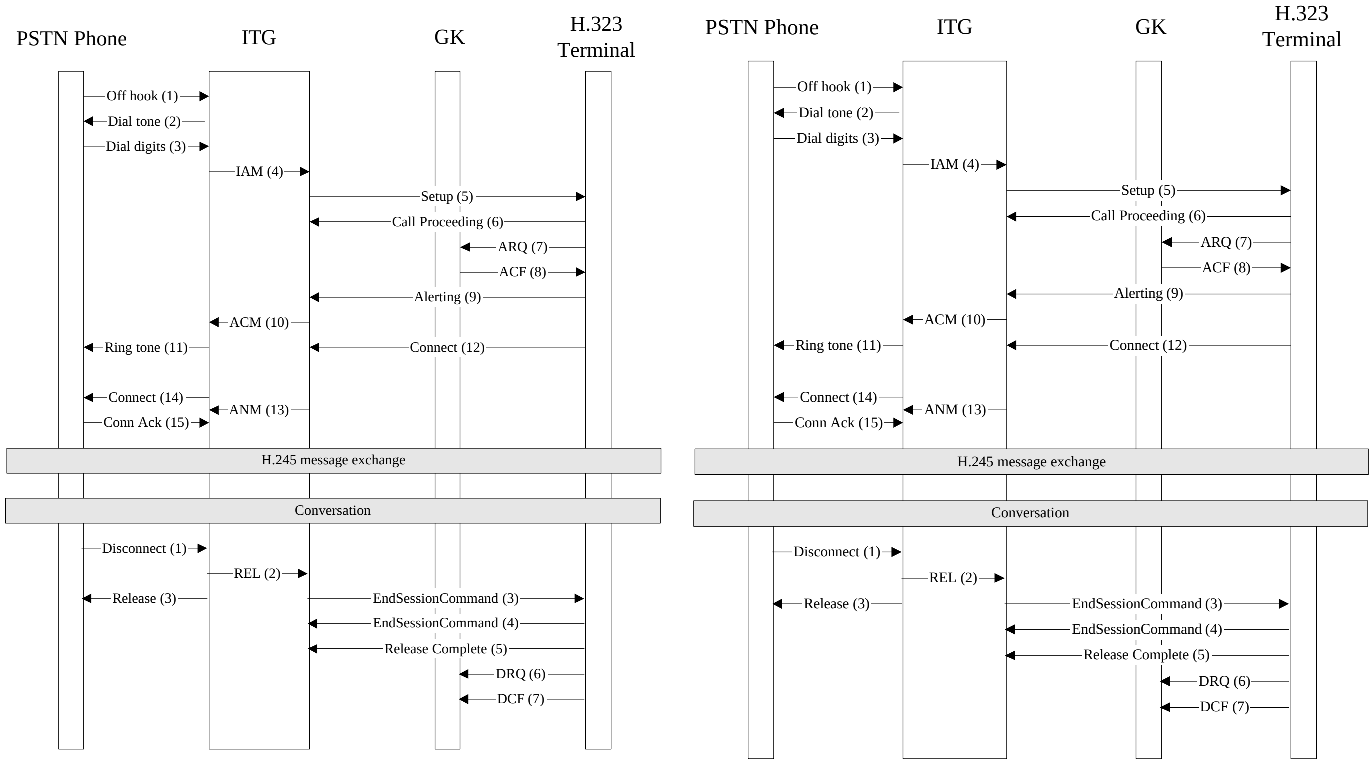
<table>
<tr><td>Audio apps</td><td>Video apps</td><td colspan="4">Terminal control and management</td><td>Data apps</td></tr>
<tr><td>G.711
G.722
G.723.0
G.728
G.729</td><td>H.261
H.263</td><td rowspan="2">RTCP</td><td rowspan="2">H.225.0
RAS
channel</td><td>H.225.0
Call signaling
channel</td><td>H.245
Control
channel</td><td>T.124</td></tr>
<tr><td colspan="2">RTP</td><td colspan="2">X.224 Class 0</td><td>T.125</td></tr>
<tr><td colspan="4">Unreliable transport (UDP)</td><td colspan="2">Reliable transport (TCP)</td><td rowspan="2">T.123</td></tr>
<tr><td colspan="6">Network layer (IP)</td></tr>
<tr><td colspan="7">Subnet layer</td></tr>
</table>

Figure 2. H.323 protocol stack [7]

Call signaling messages may be routed in two ways: (1) direct endpoint call signaling in which the call signaling messages are sent directly between the endpoints, and (2) Gatekeeper routed call signaling in which the call signaling messages are passed through a Gatekeeper between the endpoints. An endpoint here may

[1] Circuit-switched telephone network is not necessarily the Public Switched Telephone Network (PSTN). Here we just use PSTN for ease of explanation.

refer to an H.323 terminal, ITG, or MCU. For both routing approaches, an endpoint should request for admission to the Gatekeeper before connection establishment and request for disengagement after connection termination. As a result, Gatekeepers can keep track of all registered system entities and manage system resources properly. An example of direct endpoint call signaling is shown in Fig. 3, in which call signaling procedures between a POTS phone and an H.323 terminal are demonstrated. Fig. 3 (a) depicts a call originating from a POTS phone to an H.323 terminal, with the call terminated by the caller after conversation, and Fig. 3 (b) shows the same operations in the opposite direction. The messages to the left of ITG follow SS7 signaling, and the messages to the right of ITG, H.225.0/H.245. The acronyms of the signaling messages in Fig. 3 are summarized as follows. IAM is Initial Address Message, ACM: Address Complete Message, ANM: Answer Message, REL: Release, ARQ: Admission Request, and ACF/ARJ: Admission Confirmation/Admission Reject.



(a) POTS phone to H.323 terminal

(b) H.323 terminal to POTS phone

Figure 3. Call signaling for connection establishment and termination

H.323 has introduced the concept of an ad hoc multipoint conference, i.e., a conference capable of expanding from a point-to-point conference involving an MC to a multipoint conference at some time during the call. This expansion may be realized by two alternatives: (1) invite-to-join approach in which a new endpoint *E* is invited into the conference by any participant calling *E* through the MC, or (2) request-to-join approach in which a new endpoint joins the conference by calling a participant in the conference. Again, both direct endpoint call signaling and Gatekeeper routed call signaling can be adopted for such ad hoc multipoint expansions.

3. Mobile Internet telephony

In this section, mobile extensions to H.323 for the provision of the mobile Internet telephony services are proposed. We will demonstrate how H.323 provides an excellent environment for mobile extensions to Internet telephony. No extra system component is introduced. The definition of the H.323 terminal will be extended to refer to both stationary and mobile H.323 terminals, and the rest of the H.323 entities remain unchanged. Some terminology has been borrowed from Mobile IP for the system architecture. We try to reuse the existing protocols in IP networks and stick to those control messages and functions defined in H.323 as much as we can, and show how to seamlessly integrate mobility feature into H.323 compliant Internet telephony systems. We will first explain the concept of how to extend the mobility capability from the existing H.323 Internet telephony systems, and present system architecture and call signaling procedures for mobility management in the following sections.

3.1 Internet telephony: from stationary to mobile terminals

Internet telephony was invented to enable real-time voice services for stationary IP hosts communicating with other stationary IP hosts on IP-based networks, or with circuit-switched telephones through an Internet Telephony Gateway (ITG). To provide host mobility in Internet telephony, roaming should be supported and handoff should be performed if necessary. Roaming refers to the ability to ensure that the global connectivity for an endpoint is still assured while moving. Such reachability can either be discrete or continuous. Discrete reachability is the synonym of portability, implying no on-line reachability and communications taking place while moving. Continuous reachability is the synonym of mobility, allowing seamless roaming while communicating. Obviously, mobility encompasses portability, and requires the connections to be handed off. A *handoff* allows a mobile terminal to be assigned to a new channel in the new location area. Upon crossing a region boundary, a handoff must be initiated; otherwise, the connection is broken and the ongoing conversation is interrupted. The question is what the granularity of the service area is and how a handoff is handled.

In a digital cellular phone system, say GSM [11], a mobile phone requires a handoff upon crossing a cell boundary. The connection is handed off from the old cell to the new one. Once the mobile phone has been given a new channel in the new cell, the old channel in the old cell is released and made available to other users in that cell. Global roaming is realized through the cooperation of Home Location Register (HLR) and Visitor Location Register (VLR) that store the location information of mobile phones. An HLR contains subscription and location information of its “home” users, while a VLR provides temporary services for a user whose primary subscription is in another service area, but presently roaming in its affiliated service area. When a location update is initiated for a visitor, the VLR sends the update to the HLR for further call processing. Note that location update in a mobility agent (HLR or VLR) is implicitly performed after a handoff for roaming across service area boundary in a connection-oriented digital cellular phone system.

In an IP telephony network with wired and wireless subnets, a mobile phone requires a handoff upon crossing a subnet boundary that causes the IP address of the mobile phone to change. Mobile IP [12] is currently the major standard for mobility management on the Internet. With the Mobile IP mechanism, roaming is performed through the cooperative support of home agent and foreign agent, similar to HLR and VLR in the GSM, respectively. When a location update[2] is initiated, the home agent of a visiting host is informed of the *care-of address* temporarily assigned to the roaming host, either by the visited foreign agent or by the host itself. This allows data destined to the mobile host in the foreign network to be routed correctly via IP tunneling. However, Mobile IP does not handle connection handoff because it is based on a connectionless service model and lacks the concept of a connection. IP telephony services, on the other hand, require connection-oriented service over packet-switched IP-based networks. Neither the circuit-switched cellular phone approach nor the connectionless Mobile IP mechanism enables mobile Internet phone service. New approaches are required for mobile IP phone services.

3.2 System architecture

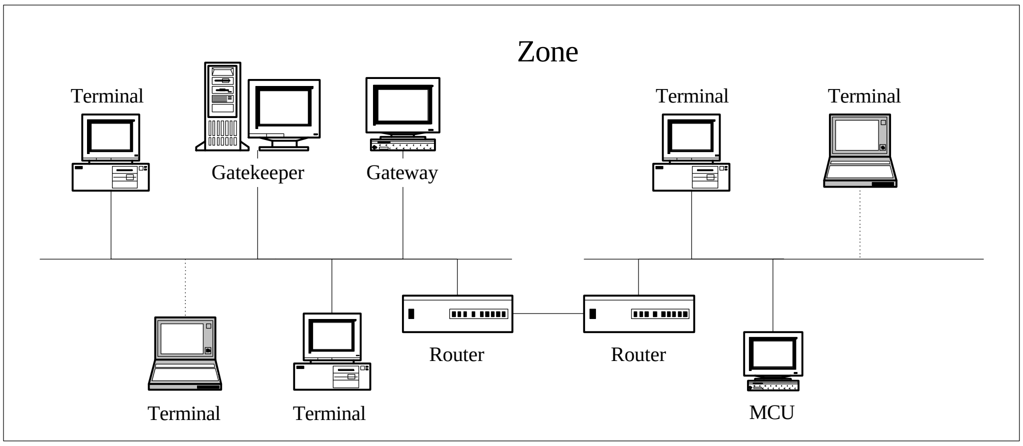


Figure 4. A zone in the H.323

As defined in H.323, a zone is defined as a collection of H.323 entities (H.323 terminal, ITG, or Multipoint Control Unit) managed by a single Gatekeeper, as depicted in Fig. 4. Each zone contains *exactly one* Gatekeeper, but allows entities to be distributed over multiple subnets connected by routers. With the characteristics of only one Gatekeeper and multiple subnets, we observe that the zone of an H.323 system plays a similar role to the service area of a cellular phone system, and the Gatekeeper, a similar role to a mobility agent such as the HLR or VLR. The service area in a cellular phone system may consist of multiple cells and base stations under the management of a mobility agent. Similarly, a zone may consist of multiple subnets under the management of a Gatekeeper. When a mobile IP phone is roaming from subnet to subnet, the IP address of the IP phone is changed accordingly. The connection is broken and the ongoing conversation is interrupted unless a handoff is performed.

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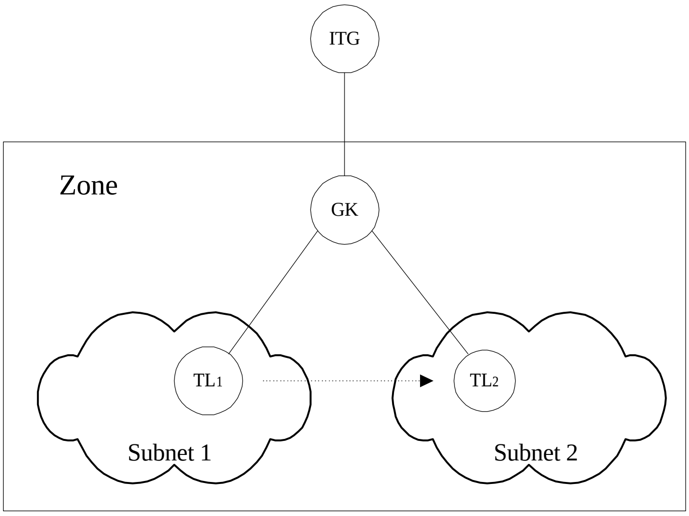
[2] Or mobility binding in the terminology of Mobile IP.

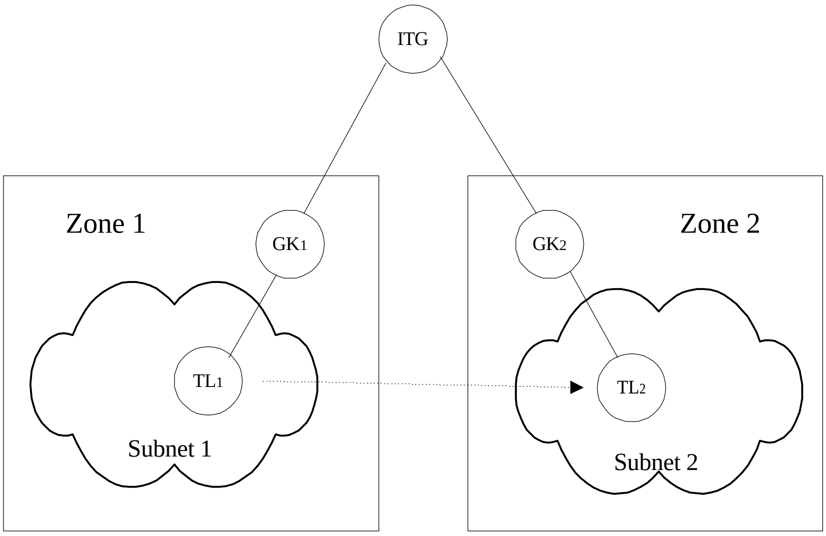
To describe how our approach enables roaming during a conversation, some terminology is introduced. The *home zone* is defined as the zone that an H.323 terminal normally resides, and *foreign zone*, a zone that the H.323 terminal may visit. The corresponding Gatekeepers are called the *home Gatekeeper* and the *foreign Gatekeeper*, respectively. When a mobile host moves within a zone, it is referred to as *intra-zone roaming*. When it crosses the boundary to other zones, it is called *inter-zone roaming*. No location update in a Gatekeeper is required for intra-zone roaming because a mobile terminal is still under the control of the same Gatekeeper. Location update is required only upon crossing a zone boundary. Upon roaming to a foreign zone, a visiting terminal notifies the visited foreign Gatekeeper of its existence with the transport address (a temporary IP address and a port number in a TCP/UDP environment) through registration. A temporary IP address of the visitor may be acquired dynamically from the Dynamic Host Configuration Protocol (DHCP) server [13] in the respective subnet. The foreign Gatekeeper then informs the home Gatekeeper of the reachability information using the address of the affiliated foreign Gatekeeper as the care-of address of the visiting terminal, hiding the actual point of attachment of the visitor behind a coarser destination. The home Gatekeeper performs mobility binding which associates the home address of the moving terminal with the temporary care-of address. Such mobility binding is coarser than Mobile IP and reduces the number of location update messages when a terminal is roaming from subnets to subnets in a zone. As a matter of fact, it will be demonstrated that with our approach, location tracking for a roaming terminal to make a call can be performed even without the support of mobility binding. Note that a Gatekeeper is responsible for address translation and call admission control services and may participate in call signaling, similar to a mobile switching center with the mobility agent in GSM. Recall that a mobile switching center is the heart of a GSM system that is responsible for connection establishment, management, and termination as well as call routing to proper cells. Therefore, global roaming can be realized by the cooperative support of home and foreign Gatekeepers.

3.3 Mobility management

Mobility management is the key to successfully enable mobile Internet telephony service over the connectionless IP networks. The core operations include registration, call establishment, roaming and handoff. In H.323/H.225.0 [9], an endpoint is registered with a Gatekeeper for admission to system access through an unreliable RAS channel. Call establishment and termination are performed before and after a call, respectively, through a reliable call signaling channel. However, as mentioned earlier, no mobility support is addressed in the current version of H.323. Therefore, there is no mechanism for roaming and handoff handling, and no location update and callee location tracking. Using the concept of ad hoc multipoint conference expansions, we will demonstrate how to extend mobility support to an IP phone using only the existing messages defined in the current version of the H.323 standard.



(a) Intra-zone roaming



(b) Inter-zone roaming

Figure 5. Roaming scenarios

When an H.323 terminal roams across different subnets during a call, thereby causing the IP address to change, the connection is broken unless it is handed off. Examine the roaming scenarios shown in Fig. 5. Assume that TL1 denotes an H.323 terminal currently staying in subnet1, and TL2 denotes the same H.323 terminal roaming from subnet1 to subnet2. The subnets are under the management of the same Gatekeeper in the case of intra-zone roaming as shown in Fig. 5 (a) and of different Gatekeepers in the inter-zone roaming as shown in Fig. 5 (b). Either case, originally, contains only one connection created from the ITG to TL1. TL1 is communicating with the ITG while roaming, and while a handoff is taking place during the migration from the subnet1 to subnet2. Once the handoff is completed, the connection from the ITG to TL1 is terminated, and a new connection from the ITG to TL2 is established, leaving only one connection from the ITG to TL2. The resources reserved and allocated for the old connection in subnet1 is released and made available for reuse. Such a handoff simulates the handoff in the cellular phone system, that is, a new connection is first established and co-exists with the old connection, and in turn the old connection is torn down.

Surprisingly, the handoff procedure described above for mobile IP phone roaming may be realized by the dynamic join and departure of an ad hoc multipoint conference in the H.323 compliant Internet telephony systems. An ad hoc multipoint conference is a point-to-point conference that can be expanded to multipoint conference at some time during the call [7]. The steps of dynamically joining and leaving an ad hoc conference are demonstrated in Fig. 6.

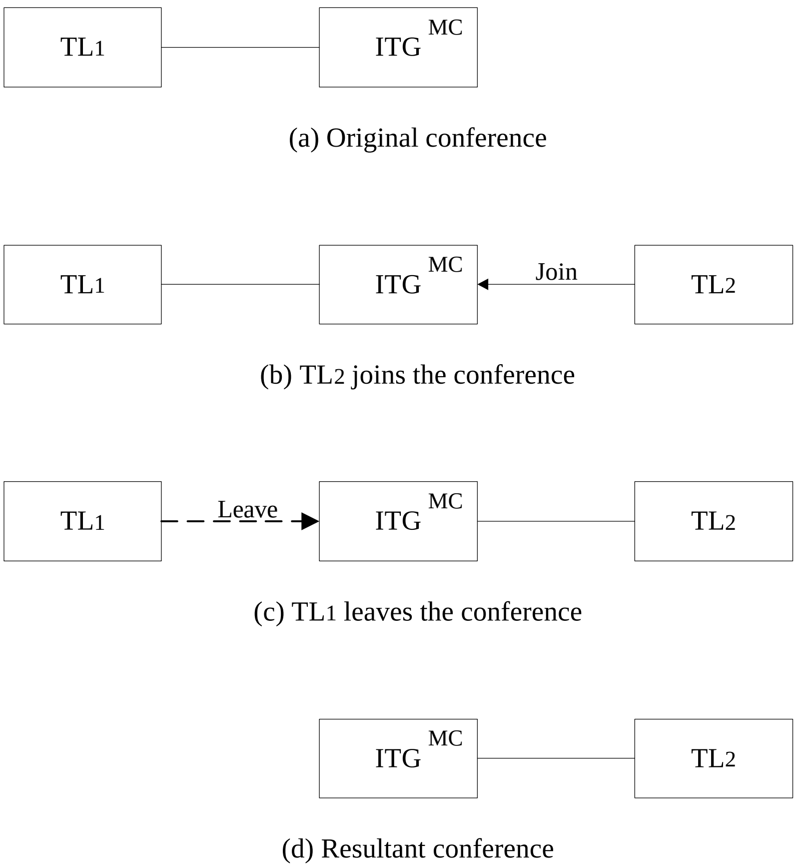


Figure 6. Dynamic join and leave in a multipoint conference

Originally, a point-to-point conference is first created between the ITG and TL1. TL2 then joins the conference either by being invited or by its own request through the MC in the ITG. A new connection from the ITG to TL2 is required to be setup when TL2 joins the conference, with the same media streams in the same conference. Finally, TL1 leaves the conference after TL2 has joined, resulting in the connection from the ITG to TL1 being torn down without affecting the media stream of the ITG to TL2 connection. This shows the effect of the proposed handoff for the mobile Internet telephony equivalent to the dynamic join and departure of an ad hoc multipoint conference. Fig. 6 (a) depicts the ITG communicating with TL1 while roaming. Fig. 6 (b) depicts the handoff from subnet1 to subnet2 while the conversation continues. Fig. 6 (c) shows that the handoff is nearly finished and the connection from the ITG to TL1 is broken, without affecting the connection between the ITG and TL2. Finally, the handoff succeeds, and only the connection from the ITG to TL2 is left, as depicted in Fig. 6 (d).

The correspondence between handoff handling and dynamic join and departure of an ad hoc conference strongly suggests the possibility of the support of mobile IP telephony service from H.323. Strikingly and interestingly, this approach extends stationary Internet telephony systems to seamlessly support mobile IP phone services, without any extra cost for such mobile extensions.

4. Call signaling procedures for mobility management

Because logical channel establishment for media transmission defined in H.245 [8] remains unchanged once a call is made, the major focus of the following will be on the call signaling procedures for registration, connection establishment, and roaming. Due to the limited space, only a call made between a POTS phone and an H.323 terminal is demonstrated. The procedure is applicable for an H.323 terminal to H.323 terminal call as well.

Call handling can be divided into three parts: POTS phone to the ITG (on the PSTN), inside the ITG, and the ITG to H.323 terminal (on the Internet). Here the procedures for both POTS phone to the ITG and inside the ITG remain unchanged for the mobile extensions to Internet telephony services. Thus only the Internet part (i.e., between the ITG and H.323 terminal) is considered in this section. Again, due to the space limitation, only direct endpoint call signaling is demonstrated. The Gatekeeper routed call signaling can be derived in a similar manner.

4.1 Registration

An RAS (i.e., registration, admission, and status) channel defined in H.225.0 conveys messages for Gatekeeper discovery and endpoint registration. Gatekeeper discovery is a process in which an endpoint learns to ascertain which Gatekeeper to register with. As defined in H.225.0, an endpoint may send a Gatekeeper Request (GRQ) message to the Gatekeeper well-known discovery multicast address to determine the Gatekeeper, followed by the response from the Gatekeeper either with the Gatekeeper Confirmation (GCF) message to accept the registration, or with the Gatekeeper Reject (GRJ) for refusal. In a mobile environment, a mobility agent may advertise its presence and availability. Gatekeeper discovery will be extended and performed in two different ways: (1) a Gatekeeper may advertise its availability for mobility services, or (2) a mobile terminal may solicit the availability of a Gatekeeper. In the first case, an extra message called Gatekeeper Advertisement (GAD) will be introduced. A Gatekeeper may advertise its availability with a GAD message through multicasting to the respective zone. In the second case, a mobile terminal may send a GRQ message to the Gatekeeper well known Discovery Multicast Address to ascertain the availability of a Gatekeeper, followed by the corresponding Gatekeeper’s reply to the solicitation request with a GCF or GRJ.

As part of the configuration process, an endpoint registers with the Gatekeeper identified through the discovery process. Registration here refers as to the process by which an endpoint joins a Zone and informs the corresponding Gatekeeper of its existence with its address. Registration must occur before any call attempt. As defined in H.225.0, an endpoint shall register with a *single* Gatekeeper via a Registration Request (RRQ) message. The Gatekeeper then sends either a Registration Confirmation (RCF) or a Registration Reject (RRJ) message in reply, indicating the registration status. Each mobile terminal has a finite registration life and requires a re-registration on timer expiry. We will defer the discussion of registration to roaming in Sec. 4.3. Note that authentication and other security issues are important to registration as well. These issues are outside the scope of this paper.

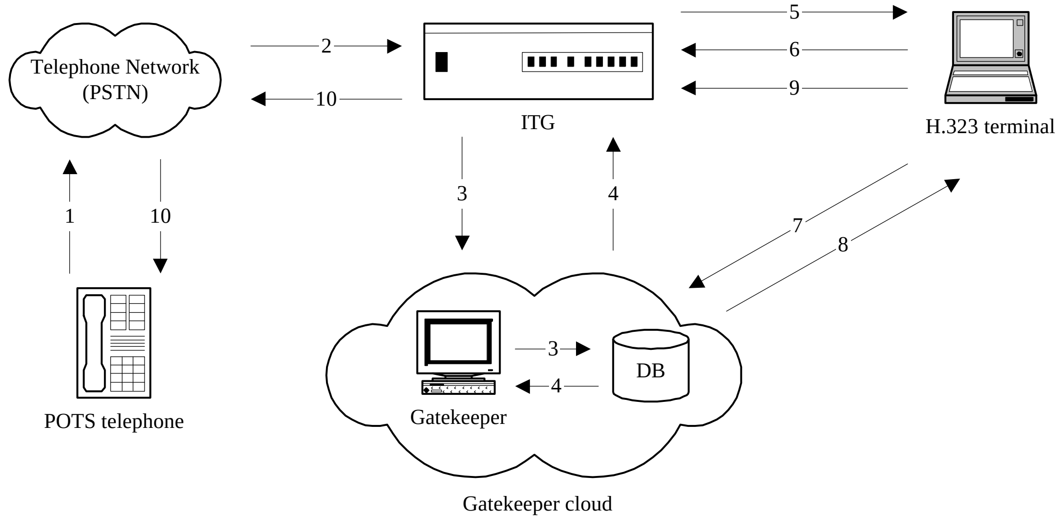
4.2 Call establishment

A call between a POTS phone and an H.323 terminal can be made in either direction, i.e., from a POTS phone to an H.323 terminal or an H.323 terminal to a POTS phone.

(1) ITG (caller) to H.323 terminal (callee)

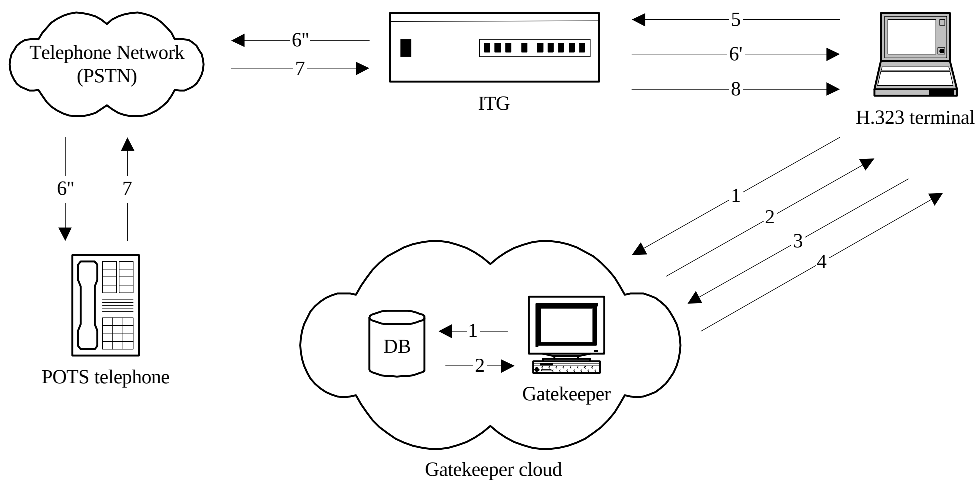
Registration is performed first before any call attempt. Assume each H.323 terminal is registered and managed under a single Gatekeeper in the respective zone. When a call is destined to an H.323 terminal (either stationary or mobile) from the ITG, a paging message of location discovery for the H.323 terminal is sent by the ITG. The caller has no idea as to whether the callee is stationary or mobile. If the called H.323 terminal is a fixed IP host, it always stays in the home zone, under the management of the home Gatekeeper. If the called H.323 terminal is a mobile IP host and has roamed to a foreign zone, it is under the management of a foreign Gatekeeper. The caller, however, does not know the identity and location of the corresponding Gatekeeper of the callee. As defined in H.225.0, when an endpoint would like to determine the contact information of other endpoint, it may multicast a Location Request (LRQ) message to the Gatekeeper well-known discovery multicast address. If the endpoint is registered, then the corresponding Gatekeeper responds with a Location Confirmation (LCF) message containing the contact information of the endpoint; otherwise a Location Reject (LRJ) is returned.

Therefore, this paging message is sent to the Gatekeeper well-known discovery multicast address with an LRQ message so that all Gatekeepers are able to receive the page. This is similar to broadcasting a paging message to all base stations in the cellular phone systems. Only the Gatekeeper that the called party registers with acknowledges the paging message with the callee's IP address in an LCF message, either the real IP address (from the home Gatekeeper) or the care-of address (from a foreign Gatekeeper). Once the callee is located, the connection is established accordingly. The caller initiates the connection establishment through the normal creation procedure for a point-to-point conference. The procedure of call establishment is



1 = call from POTS phone to an H.323 Terminal
2 = call routing by PSTN
3 = H.323 terminal location request
4 = H.323 terminal location return
5 = initial setup
6 = call proceeding
7 = admission request
8 = admission request acknowledgement
9 = connection establishment done on the Internet
10 = call establishment

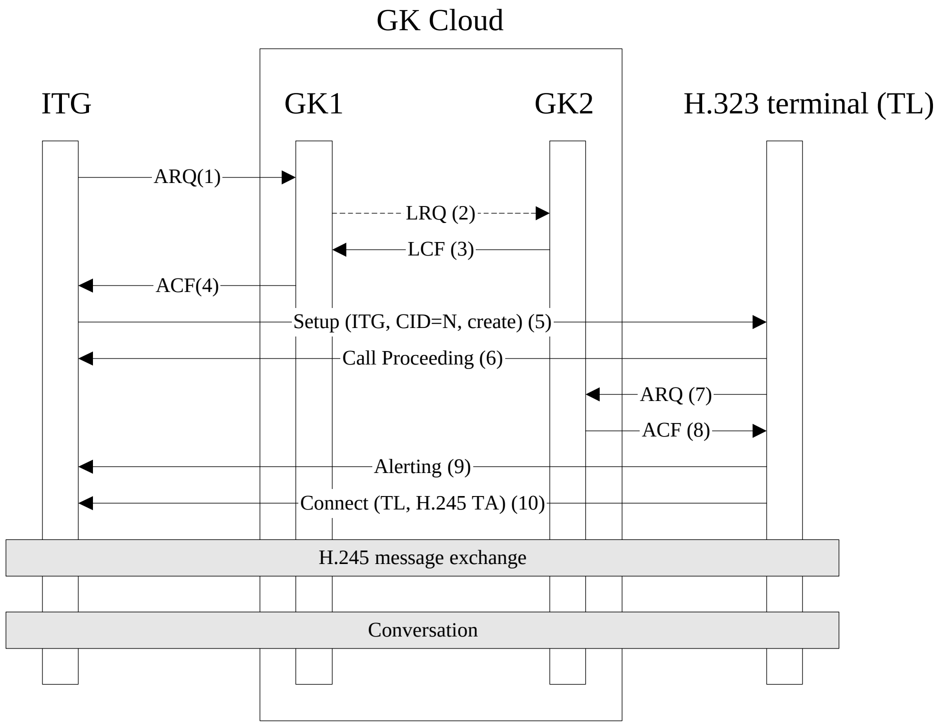
(a) POTS phone to H.323 terminal



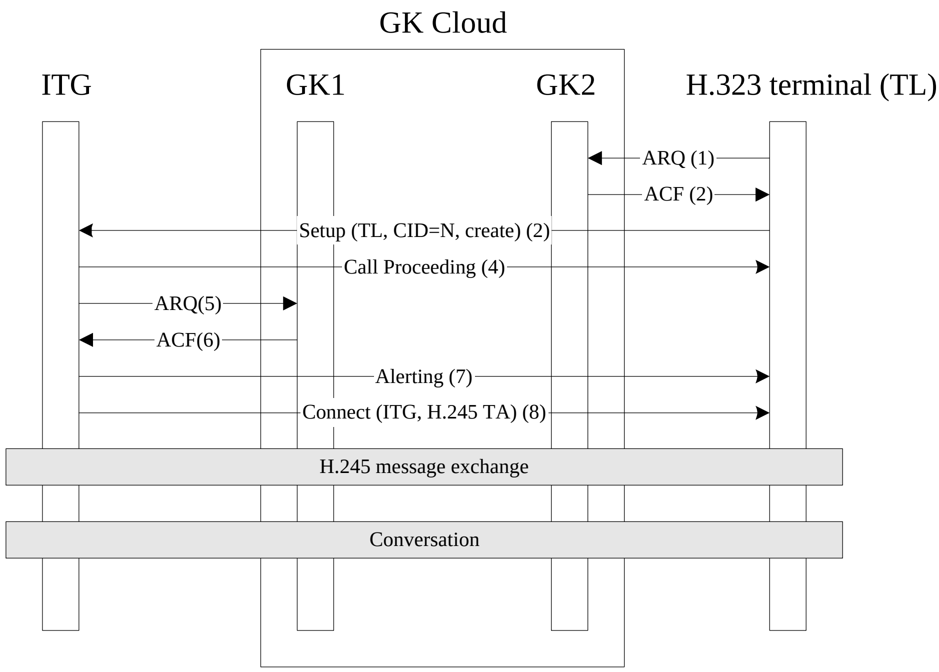
1 = address translation request
2 = translated address return
3 = admission request
4 = admission request acknowledgement
5 = initial setup
6' = call proceeding
6" = call setup on the PSTN
7 = connection establishment done on the PSTN
8 = call establishment

(b) H.323 terminal to POTS phone

Figure 7. Call establishment



(a) ITG to H.323 terminal



(b) H.323 terminal to ITG

Figure 8. Call signaling for call establishment

summarized in Fig. 7 (a), and the corresponding call signaling is shown in Fig. 8 (a). Note that CID means Conference ID, and TA, Transport Address. The number associated with a message, e.g., ARQ (1), indicates the signaling order for a call setup. Assume that GK1 and G2 are the registered Gatekeepers for the ITG and TL, respectively. A dotted line denotes that a message is sent in multicast, while a solid line denotes to a unicast transmission. For example, --- LRQ (2) ----→ within the GK cloud (Gatekeeper cloud, i.e., a collection of Gatekeepers) denotes that the LRQ message is sent by GK1 to the all Gatekeeper well-known discovery multicast address, and ← LCF (3) –, the LCF message sent by GK2 to GK1 in unicast. If GK2 is the home Gatekeeper of TL, TL's permanent home IP address is sent in reply in the LCF; otherwise, a temporary care-of-address of TL is returned in the LCF. GK1 then forwards the current location address of TL in the ACF message back to the ITG, using the information exchanged in the location tracking procedure. A connection is built between the ITG and TL directly, hiding the roaming nature of the mobile terminal.

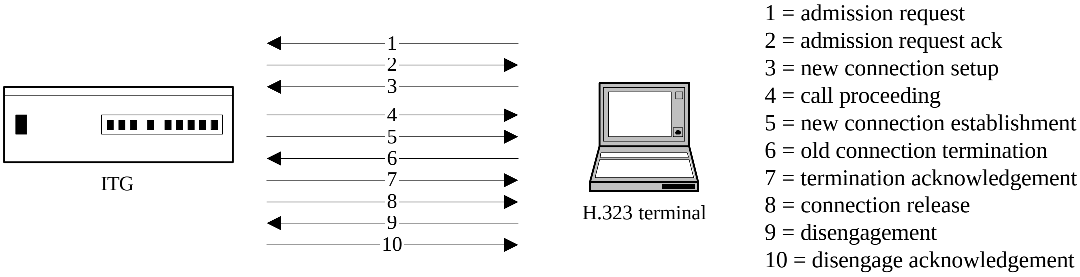
(2) H.323 terminal (caller) to ITG (callee)

When an H.323 terminal calls a POTS phone, no paging message for location discovery is required, but the address translation for ITG location lookup performed through the directory service by the corresponding Gatekeeper is necessary. ITG location discovery is another important issue for the success of Internet telephony. It is outside the scope of this paper. A variety of strategies discussed in [14], such as SLP [15] and LDAP [16], may be used for locating the ITG. Once the translated address of requested ITG is return, the normal creation procedure for a point-to-point conference is performed. Connection establishment is summarized in Fig. 7 (b) and the corresponding call signaling is shown in Fig. 8 (b). Note that the address translation for the ITG is not shown in Fig. 8 (b).

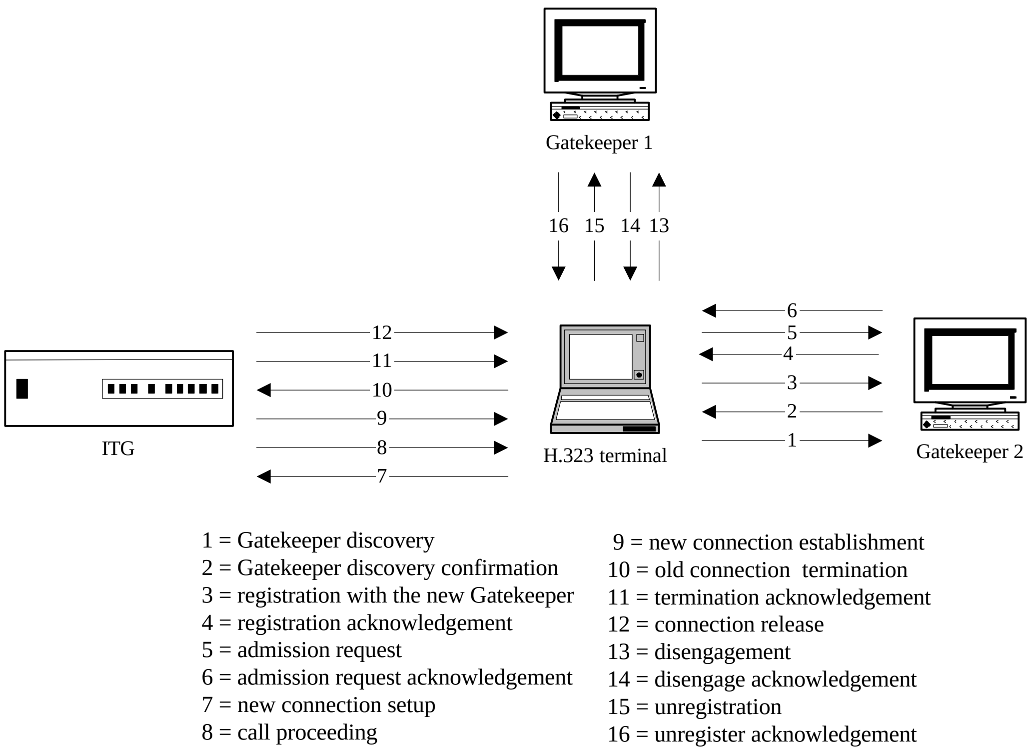
4.3 Roaming

When an H.323 terminal is roaming, it may be in intra-zone roaming or inter-zone roaming. A handoff occurs when terminal TL1 at subnet1 roams to subnet2 as shown in Fig. 5. As mentioned earlier, a handoff can be handled as dynamically joining and leaving an ad hoc multipoint conference. With ad hoc multipoint conference expansions, there are two alternatives for a new endpoint to join the conference through the MC (Multipoint Controller): request-to-join (by requesting on its own) and invite-to-join (being invited by any participant). Once the new endpoint TL2 has joined the conference in either case, TL1 leaves the conference without affecting the ongoing call. In the following, only the request-to-join approach is demonstrated. Invite-to-join can be performed in a similar manner.

For the intra-zone roaming, a handoff can be handled using the normal procedures for a new endpoint joining an ad hoc multipoint conference, followed by an old endpoint leaving the conference. For inter-zone roaming, Gatekeeper discovery should be performed first. This is necessary because the visiting mobile host has no idea as to where the foreign Gatekeeper in the foreign zone is. As mentioned in Sec. 3.1, there are two



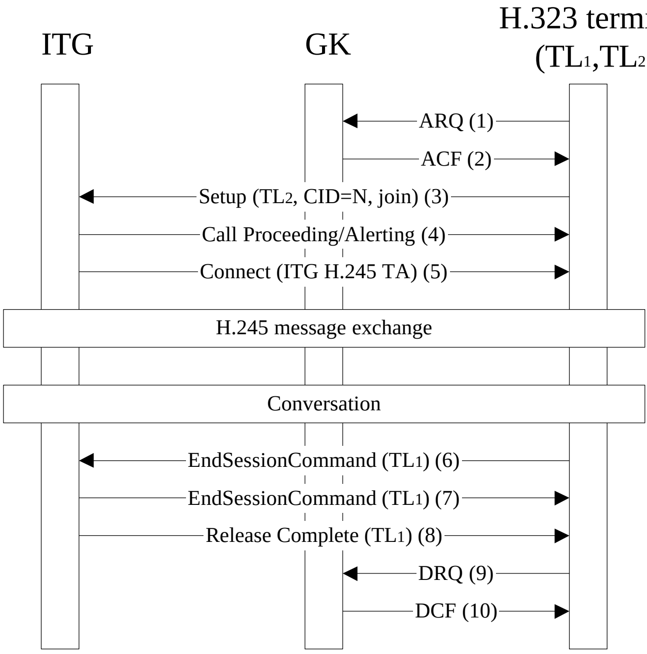
(a) Intra-zone roaming



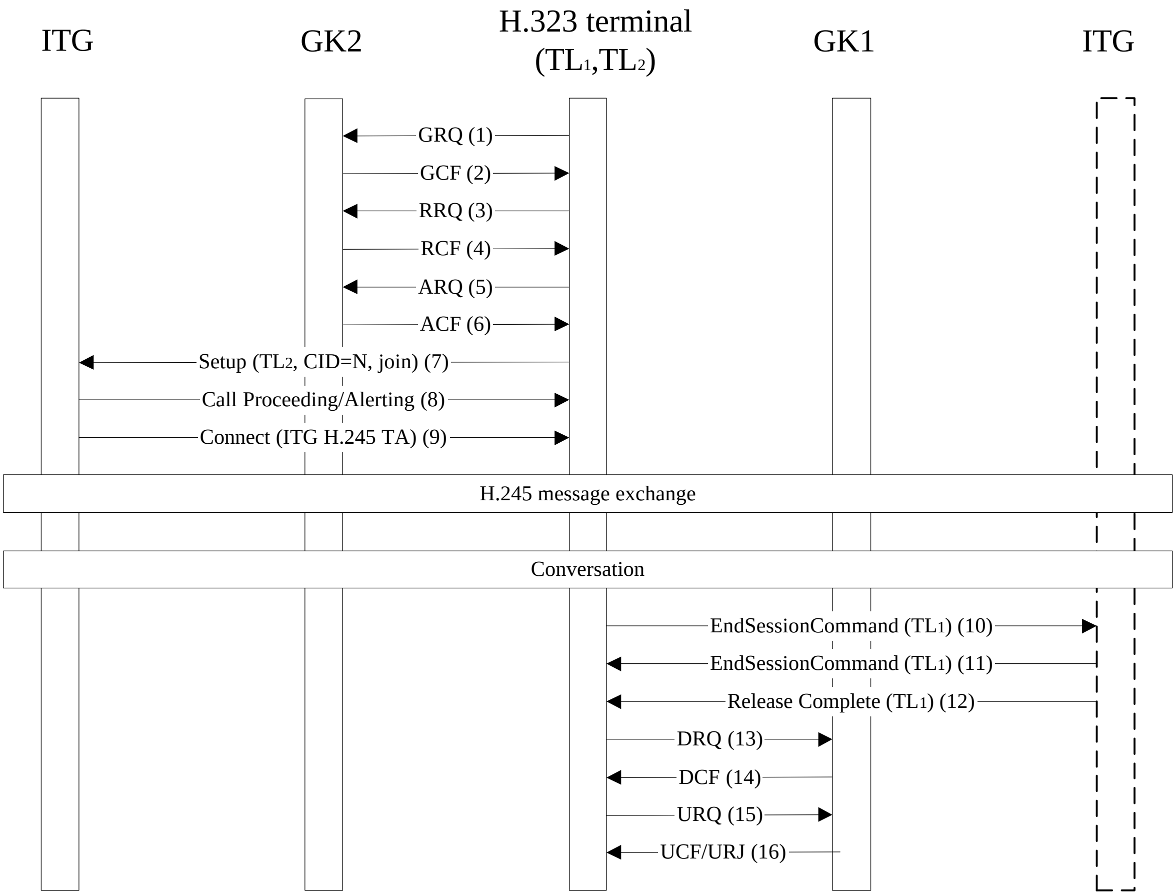
(b) Inter-zone roaming

Figure 9. Roaming

different ways for Gatekeeper discovery: (1) Gatekeeper advertisement, or (2) Gatekeeper solicitation. Once the foreign Gatekeeper has been identified through the discovery process, registration occurs. Then the handoff is handled the same way as in intra-zone roaming, i.e., with the normal procedures for a new endpoint joining an ad hoc multipoint conference, followed by the departure of an old endpoint. With H.225.0, an endpoint issues a Disengage Reject (DRQ) message to terminate a call, followed by a Disengage Confirmation (DCF) message sent in reply by the Gatekeeper. To cancel a registration, an endpoint may either (1) send an Unregister Request (URQ) message to the Gatekeeper, which then responds with either an Unregister Confirmation (UCF) message or an Unregister Reject (URJ) message, or (2) wait until the registration lifetime expires. Of course, H.245 logical channels for the connection should be closed first. Therefore, to break the old connection, the mobile host should de-register from the previous Gatekeeper with a DRQ and URQ messages to complete the handoff.



(a) Intro-zone roaming



(b) Inter-zone roaming

Figure 10. Call signaling for roaming

Signaling procedures for roaming and handoff handled by the "request-to-join" conference approach are depicted in Fig. 10, equivalent to ad hoc multipoint conference join and departure. Note that in Fig. 10 (b), (1) to (4) are for registration to the new Gatekeeper GK2 and (15)-(16) are for de-registration from the previous Gatekeeper GK1. Only Gatekeeper solicitation for discovery is shown. The rest of the procedures remain the same as in Fig. 10 (a).

4.4 Discussion

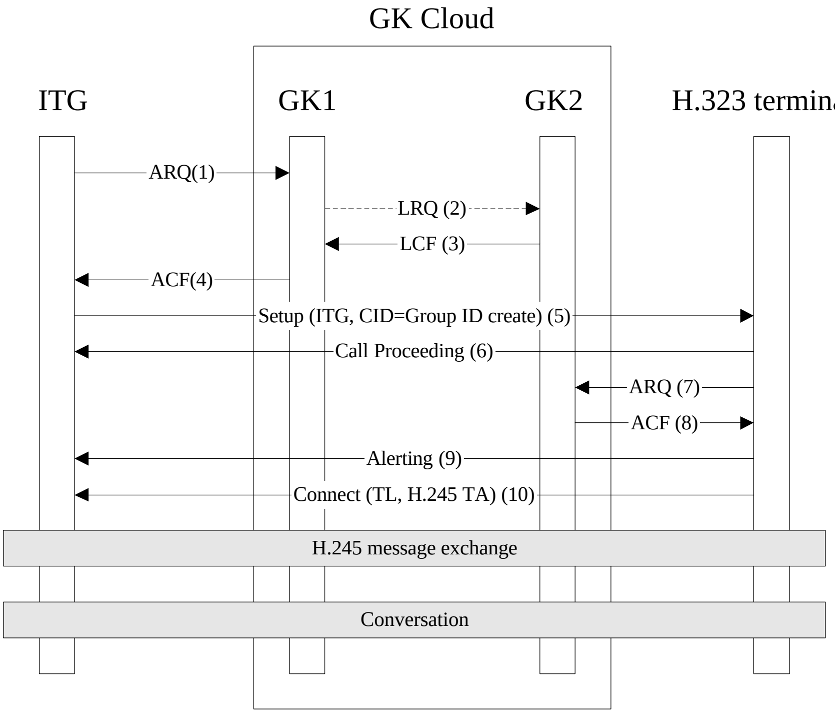
Using a "multicast LRQ" to track a roaming H.323 terminal eliminates the need of location update (or mobility binding) as employed by both GSM and Mobile IP, thereby reducing the operational complexity for mobility management of mobile IP telephony service. For the sake of performance efficiency, a binding cache for mobility binding may be used by a Gatekeeper to maintain a list of moves of its "home" subscribers. A location tracking message LRQ is sent to the home Gatekeeper of a called terminal in unicast delivery by the registered Gatekeeper of a caller, from where the home Gatekeeper in turn forwards the request (LRQ) to corresponding foreign Gatekeepers indicated in the binding list if the callee is currently away from home. The care-of address is returned to the home Gatekeeper of the callee in unicast by the foreign Gatekeeper that the callee registers with in the LCF message, which is in turn sent to the Gatekeeper of the caller.

Recall that registration refers as to the process by which an endpoint joins a Zone and informs the corresponding Gatekeeper of its existence with its address. This definition can be easily extended to support the above-addressed enhancement for a mobile environment in which registration is defined as a process whereby a mobile phone registers for a Gatekeeper and may exchange its current binding information with its home Gatekeeper. On crossing a service boundary, Gatekeeper discovery is performed first, followed by a registration. During registration, location update takes place in which the foreign Gatekeeper informs the home Gatekeeper of the care-of address of the mobile host and stores such location information in its local directory; the home Gatekeeper performs mobility binding and may instruct the previous foreign Gatekeeper to remove the entry of the mobile terminal. Once registration is completed, the procedure of roaming and handoff described in Sec. 4.3 can be applied directly.

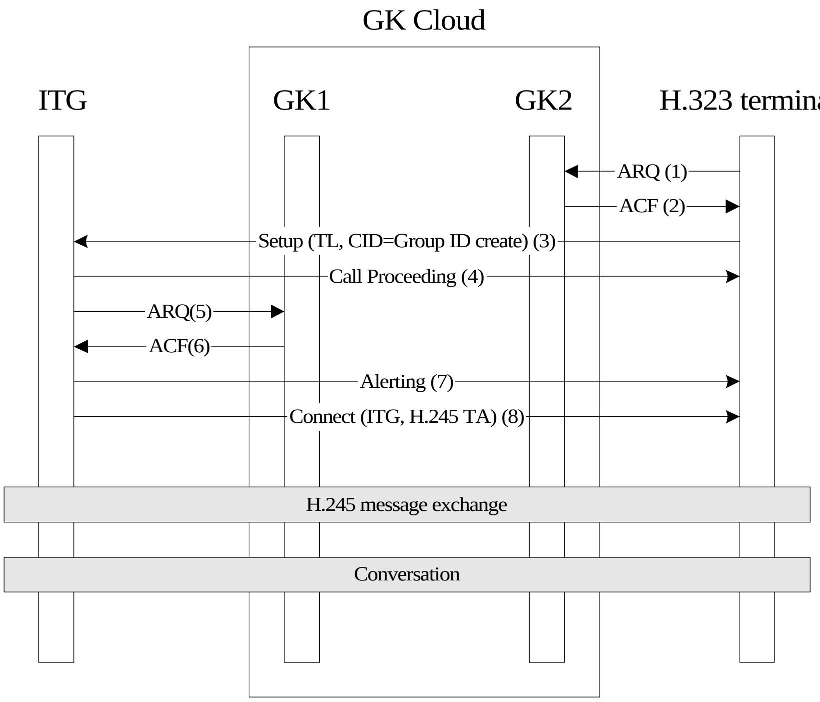
4. An alternative approach: using IP multicast to support mobility

In this section, we will demonstrate an alternative approach that uses IP multicast to support mobility. With IP multicasting, a multicast data packet contains a class D group address in the destination address field of the IP header, and is delivered to destination group members who join the addressed group, with the same best-effort delivery as unicast IP data transmission. Individual hosts are free to join and leave a multicast group at any time. No restriction is placed on the physical location and the number of group members. The provision of multicast data delivery on the Internet is supported through two mechanisms: local group management (IGMP

[17]) and global routing. The local group management mechanism enables multicast routers to learn the presence of group members on their directly attached networks; the global routing mechanism provides the ability for multicast routers to exchange information and to determine multicast delivery trees for the forwarding of multicast datagrams across the Internet. Important multicast routing protocols include DVMRP [18], MOSPF [19], CBT [20], and PIM [21].

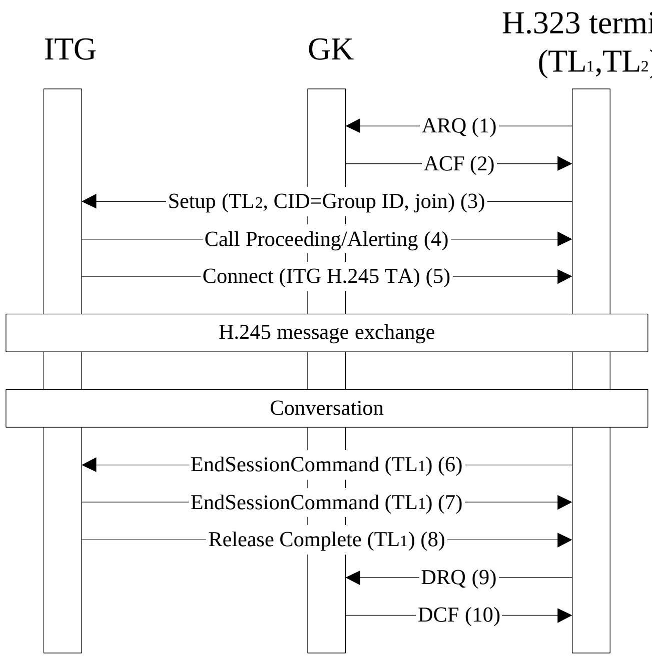


(a) ITG to H.323 terminal with multicasting

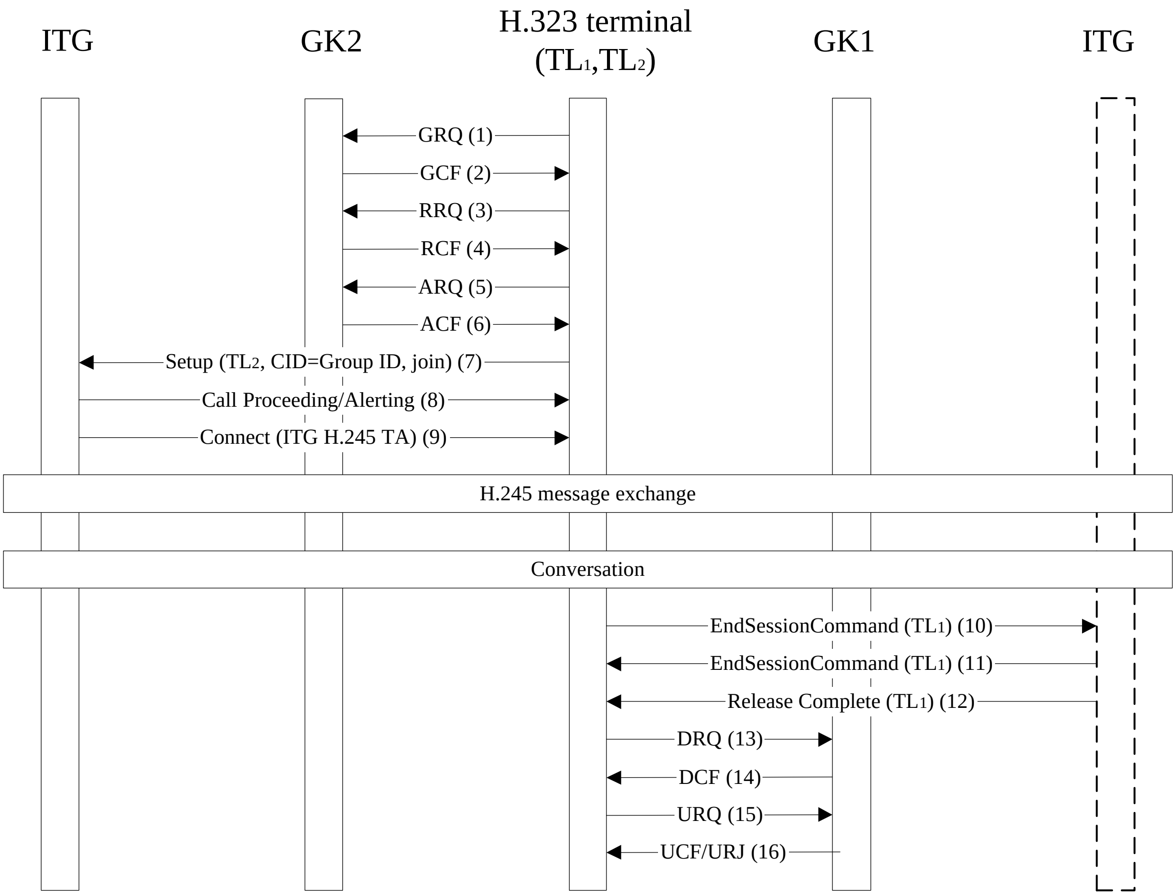


(b) H.323 terminal to ITG with multicasting

Figure 11. Call signaling for call establishment with multicasting



(a) Intro-zone roaming with multicasting



(b) Inter-zone roaming with multicasting

Figure 12. Call signaling for roaming with multicasting

To support mobility through IP multicasting, a mobile host first joins a multicast group. When the mobile terminal moves from an old subnet to a new subnet, it sends an IGMP leave message to the immediately neighboring multicast router to depart the group in the old subnet, and sends a report message to join the group in the new subnet. The nice property of this approach is that IP multicast is in place and ready to use. We will show that H.323 provides an excellent environment to be used in conjunction with IP multicasting to support mobility.

Recall that in Sec. 3.3, the proposed handoff procedure can be realized by the dynamic join and departure of an ad hoc multipoint conference, making use of the existing call signaling procedure and messages defined in H.323. Interestingly, ad hoc multipoint conference expansions share some points with IP multicast. Both allow participants to join and leave a conference without affecting the rest in the group. As a result, the call signaling procedures proposed in Section 3 can be applied to the IP multicasting approach, with some minor modification.

Fig. 11 depicts the call signaling messages for call establishment with multicasting. Again, we use “multicast LRQ” to locate an H.323 terminal, as in Fig. 8. Once the current location address of a callee is identified, the caller initiates the establishment of a point-to-point conference, with the replacement of the Conference ID (i.e., CID) with a group ID. The group ID here is a class D group address, indicating that the caller informs the callee to join the addressed multicast group for communications. Both caller and callee are free to seamless roaming, provided by the dynamic join and departure of a multicast group from IP multicast mechanisms. Fig. 12 illustrates the call signaling messages for roaming with IP multicasting.

5. Implementation aspects of mobile IP telephony

To test the feasibility of the proposed approach for mobile IP phone service, a testbed environment shown in Fig. 13 has been setup in the Internet Research Lab. at National Taiwan University. It consists of a 100 Mbps Fast Ethernet and a 2Mbps Lucent WaveLAN. System components include two Lucent WavePoint-II Access Points, two DHCP servers, some desktop PCs, notebooks, a router, a switching hub, and two Gatekeepers. The two Gatekeepers are located in two subnets connected by a router. Each subnet has a WavePoint-II Access Point, through which wireless terminals can communicate with hosts on wired networks. A DHCP server is employed in each subnet for dynamic IP address assignment.

Fig. 14 shows the functional components of the software framework, running on the OS platform of Windows environment. The framework includes five basic modules: call service, conference, registration, call tracking, and roaming modules. The call service module is the main program, accepting request messages and forwarding requests to the corresponding modules. For example, if a registration request is received, the call

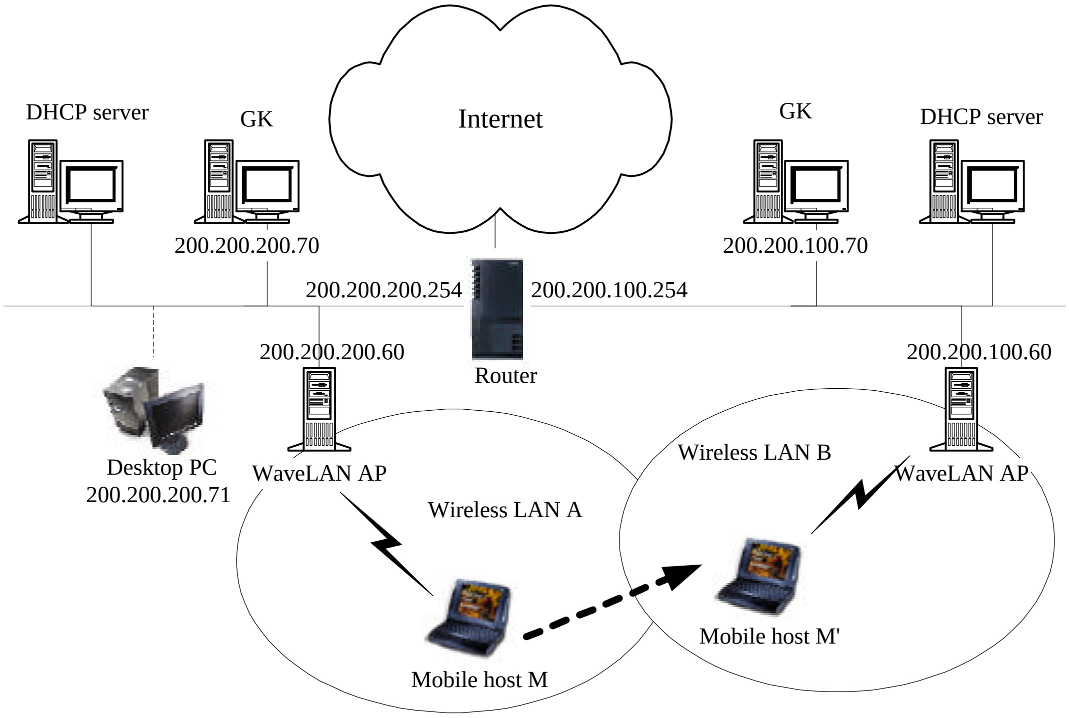


Figure 13. The testbed environment

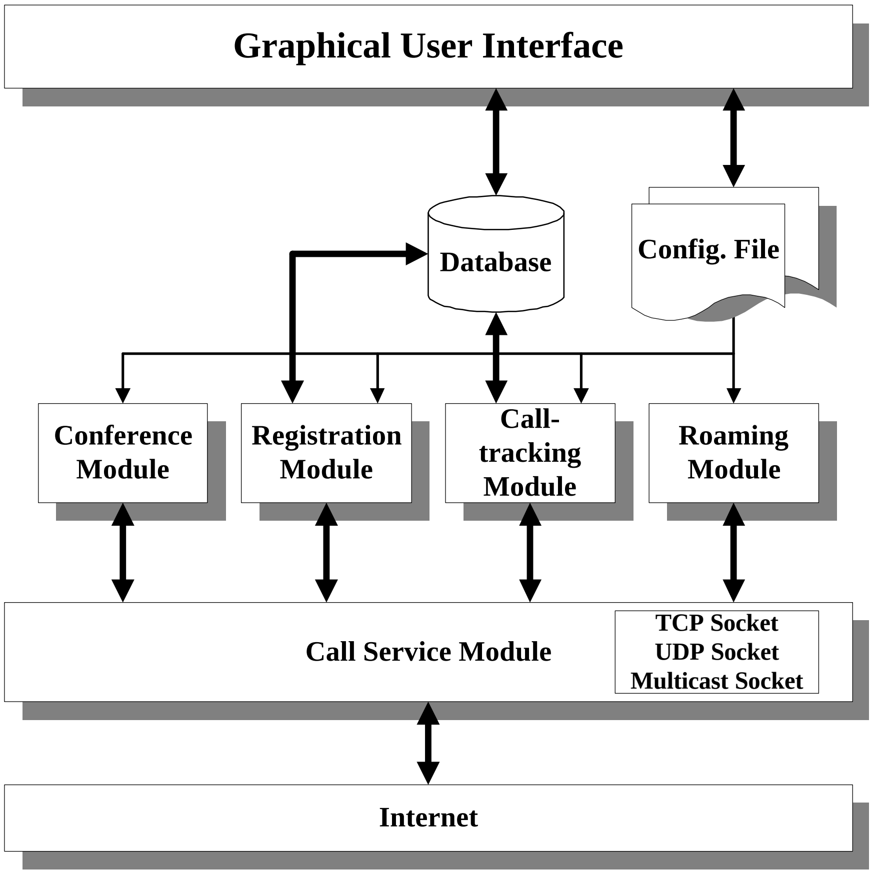


Figure 14. Functional components of a mobile IP telephony prototype

service module forwards the request to the registration module for registration. The conference module performs ad hoc multipoint conference creation and expansion. The registration module handles registration. The call-tracking module tracks the current location of a callee. The roaming module ensures a mobile host to have continuous in conversation while crossing a service boundary. In addition to the five modules, two supplementary components were also designed, namely, subscriber database and configuration file. The subscriber database stores the information of its subscribers, and the configuration file records configuration parameters to assist system function properly. Note that both Gatekeepers (servers) and H.323 terminals (clients) run on the application layer: Gatekeepers wait and listen to requests from H.323 terminals and respond to the requested services accordingly. Please refer to [22] for more details on the lessons learned from the implementation of the proposed approach.

6. Concluding remarks

We have proposed an approach to realize mobile Internet telephony service from the mobile extensions to ITU-T Rec. H.323. The system architecture has been described, and the corresponding mobility management has been presented. No extra system entities are introduced to support host mobility. We try to reuse the existing protocols in IP networks and stick to the control messages and functions defined in H.323 as much as we can, and show how to seamlessly integrate mobility features into H.323 compliant IP telephony.

The proposed approach combines the characteristics of circuit-switched cellular phone systems and packet-switched Mobile IP mechanism with Internet telephony, thereby allowing connection-oriented mobile telephone services over IP-based networks. Through the proposed mobile Internet telephony system, real-time voice services for both stationary and mobile IP hosts are supported. The optimization in the routing protocol in Mobile IP [23] is implicitly offered in the proposed approach, because the paging message for host location lookup is broadcast to all Gatekeepers, followed by the location information returned by the corresponding Gatekeeper for further call routing. No mobility binding as employed in GSM and Mobile IP is required in the proposed approach. Two alternatives for mobility extensions to H.323 are described, using the same call signaling procedure. Finally, it enables the service redirection to be completely handled in the application layer, thereby realizing mobile IP telephony services with IP rather than with the Mobile IP mechanism.

The most striking result of the proposed approach is that we have transformed host mobility management into the ad hoc multipoint conference manipulation. The key operations for mobility management are summarized as follows. (1) Registration can be achieved through the conference registration process, plus one extra message defined for Gatekeeper advertisement. (2) Connection establishment and termination for mobile Internet telephony service can utilize the procedures for creation and termination of a point-to-point conference, plus location discovery for the mobile terminal. (3) Handoff handling for global roaming can be solved through the procedures of dynamic join and departure of the conference during a call, plus registration.

Through the proposed approach, the host mobility can be seamlessly included as a value-added feature in the rapidly growing H.323 compliant Internet telephony systems.

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