Supporting Real-Time IP Multimedia Services in UMTS

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ABSTRACT

UMTS, the successor of GSM, is evolving toward a future wireless all-IP network. In this article we explain how it will support real-time IP multimedia services, as these services are expected to drive the adoption of wireless all-IP networks. We not only focus on the centerpiece of IP multimedia support, the IP multimedia subsystem, but also include the GPRS transport capabilities and OSA middleware capabilities. This helps to explain how the IMS fits in with, and builds upon, other pieces of the UMTS network. We analyze the pros and cons of providing IP multimedia session control capabilities as opposed to just providing basic IP connectivity, and highlight selected features of the IMS design for discussion.

INTRODUCTION

As communications technology develops, user demand for multimedia services is rising. Meanwhile, the Internet has enjoyed tremendous growth in recent years. Consequently, there is a great interest in using IP-based networks to provide multimedia services. One of the most important arenas in which the issues are being debated is in standards development for the Universal Mobile Telecommunications System (UMTS). UMTS is the third-generation (3G) mobile wireless cellular system that is evolving from the Global System for Mobile Communications (GSM). Given the large user base of GSM and the expected large user base of UMTS, the way IP multimedia services are provided in UMTS will significantly influence the success of IP multimedia services and the development of related technologies and protocols, such as IPv6 and Session Initiation Protocol (SIP).

In the past, people's impression of mobile wireless cellular systems was that they mostly provided voice telephony service with a few frills like short message service (SMS). The incorporation of IP multimedia services into UMTS is one of the major features that will transform this image, as many new services like videoconferencing will become available. One of the main reasons multimedia services can be provided in UMTS is the increased data rates possible in 3G mobile wireless networks. Conversely, the potential of multimedia applications helps justify investments in 3G systems.

ON IP MULTIMEDIA SERVICES

As the name multimedia implies, different types of media — video, audio, voice, and text — are involved. When multiple media types are involved in a session (e.g., video and audio), they need to be synchronized for simultaneous presentation to the user. Furthermore, multimedia services may be real-time or non-real-time. Videoconferencing is an example of real-time multimedia service. For real-time services, the time relation between successive data packets must be preserved, with little tolerance for jitter in packet arrival times. An example of a non-real-time multimedia service is multimedia messaging service (MMS) [1], an enhancement of the popular SMS that allows multimedia messages, not just text messages. In this article we focus on real-time multimedia services and the UMTS IP multimedia subsystem (IMS) that enables these services in UMTS.

The voice-centric circuit-switched telephone network and conventional cellular networks are not optimized to offer multimedia services. New network elements and protocols are necessary to provide multimedia services. Although circuit-switched connections could be used for data services, they are inefficient for transporting intermittent or bursty data. Meanwhile, packet-switching technology has been developing rapidly. It allows statistical multiplexing of traffic and can efficiently handle both bursty and nonbursty traffic. Moreover, data traffic volume continues to grow more rapidly than voice traffic volume. For cost and efficiency reasons, and to achieve better service integration, it is more appropriate to use common IP-based technology capable of supporting both multimedia and other kinds of traffic.

The IMS will enable two levels of services. The first level includes basic services like initiation, modification, and termination of multimedia sessions, as well as advanced services like multiparty sessions, session forwarding, blocking, caller ID, and so on. The second level of services

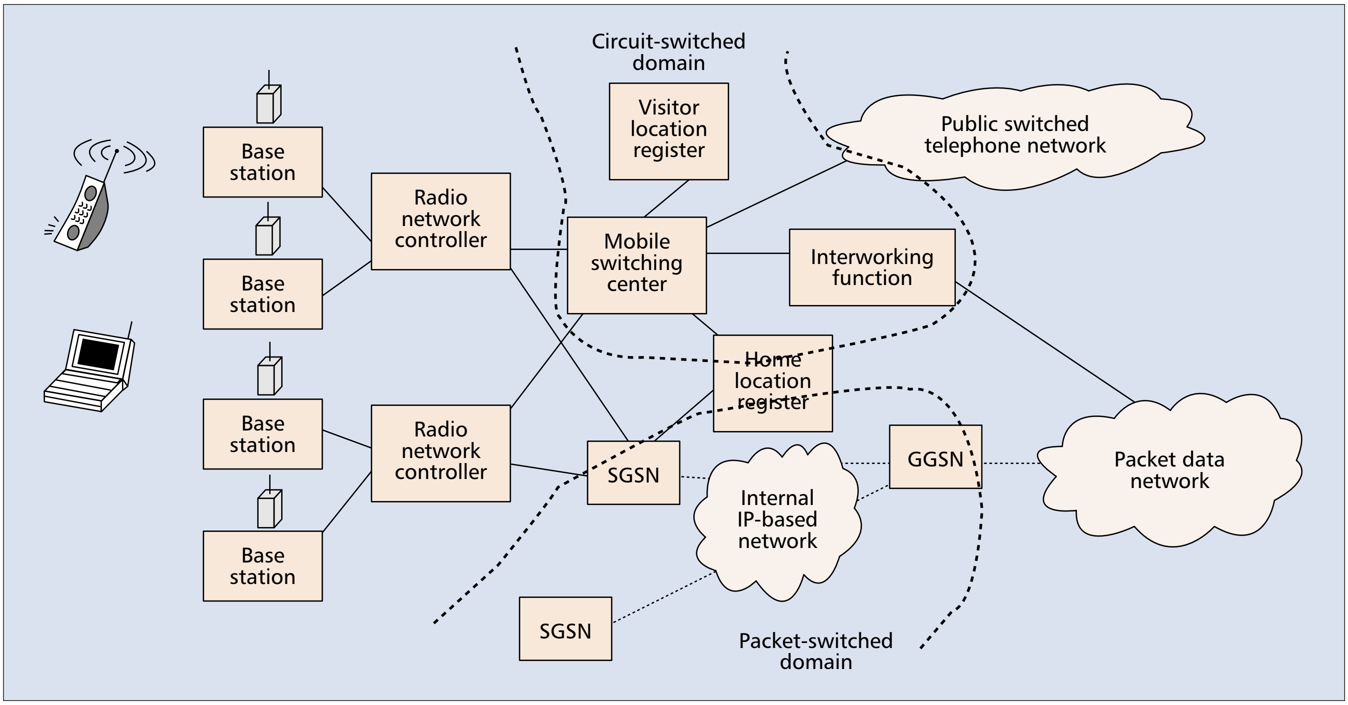


Figure 1. *GPRS architecture.*

includes services that are not being standardized but could be provided by third-party providers using capabilities provided by the network. For instance, location-based advertising (e.g., receiving a video advertisement when walking past a store) could be enabled using location information from the UMTS network.

IP CONNECTIVITY IN UMTS

The introduction of Generalized Packet Radio Service (GPRS [2], Fig. 1) was a major step in the evolution of GSM to UMTS. GSM Phase 1 and Phase 2 provided only circuit-mode connections. GPRS provides an end-to-end connectionless packet service and includes a packet-mode transfer over the air and a packet-switched backbone. Subscribers benefit from higher peak data rates and charging based on traffic volume rather than hold time, while operators benefit from the efficient use of spectrum by allowing multiple users to dynamically share resources.

Resources over the air as well as the radio access network (the base stations and radio network controllers) are dynamically shared between circuit-mode and packet-mode traffic. Circuit-mode and packet-mode traffic take different paths only in the core network behind the radio access network. On one hand, circuit-mode traffic uses the circuit-switched domain, comprising traditional entities like mobile switching centers and visitor location registers that were present in the GSM architecture from the beginning. On the other hand, packet-mode traffic uses the new packet-switched domain. The packet domain functionality in the core network is implemented in gateway GPRS support nodes (GGSNs) and serving GPRS support nodes (SGSNs). SGSNs and GGSNs are IP routers with additional capabilities to control access and track location of mobile stations. The SGSN provides security functions, packet switching, and routing, and keeps track of the location of individual mobile stations. The GGSN provides interworking with external packet-switched networks and communicates with SGSNs via a packet-switched network.

A mobile station desiring packet access must attach to the GPRS network. In order to use a specific packet protocol like IP with specified quality of service (QoS), security, and other requirements, it then activates one or more Packet Data Protocol (PDP) contexts. A PDP context is a set of configuration and usage settings related to the transport of data packets over a GPRS network. GPRS can support PDP contexts for various data network protocols, including X.25 and IP. After the establishment of an IP PDP context, for example, when an IP packet destined for the mobile station arrives at the GGSN, the GGSN tunnels it to the SGSN that currently serves the mobile station.

The GPRS network was initially developed to provide packet routing to external packet networks for the GSM radio access network. For UMTS Release 5 and beyond, GPRS takes on added significance, as it also provides transport for IP multimedia traffic within the mobile network.

MOTIVATIONS FOR THE IP MULTIMEDIA SUBSYSTEM

In GSM/GPRS and even up to Release 4 of UMTS, IP connectivity is a service provided by the network. However, only *basic* IP connectivity is standardized, so even if the wireless operator offers nonstandard enhanced IP services, it is difficult for the operator to offer services that

The network provider provides both the transport and network control elements for IP multimedia services. The network architecture also allows for third-party offering of additional IP multimedia services.

will work across network boundaries. Often, the wireless network operator may be like a plain vanilla Internet service provider, as far as IP services are concerned. In particular, the wireless network operator does not provide IP multimedia services. Instead, just as with most Internet service providers, subscribers are free to try a variety of third-party applications and use third-party providers of IP multimedia services.

We analyze the advantages and disadvantages of providing only basic IP connectivity as follows:

Subscriber perspective:

The advantages of providing only basic IP connectivity include:

- They have the freedom and flexibility to choose applications and service providers for IP multimedia services.
- They can reuse many of their existing IP multimedia applications from the non-wireless world that assume basic IP connectivity.

The disadvantages include:

- They may not want to have to go to the trouble of choosing their own applications and service providers for IP multimedia services.
- They may prefer to have all the basic services they need come in one package, from one source like the network operator, and to pay one bill.
- Careless reuse of existing IP multimedia applications from the non-wireless world may result in complications like loss of service quality because of differences in error rates for wireless access.

Network operator perspective:

The advantages of providing only basic IP connectivity include:

- Network operators may not have experience or expertise in IP multimedia applications. This approach allows them to focus on their core competency of providing connectivity.
- Even if network operators want to enter the applications business, it is very hard to know which services will be profitable and popular with customers, so the network should be open to third-party applications. Rather than attempt to guess which services will succeed, this approach provides connectivity and leaves tremendous flexibility for different services to be used.

However, drawbacks include:

- Traditional circuit-switched voice systems provide both the transport (bearer service) and teleservices (call origination, call forwarding, etc.). Providing only IP connectivity means that the teleservices are not provided. Providing only transport for IP multimedia services may not be a good return on investment, especially as prices keep dropping and are projected to continue declining.
- They may lose potential revenue by not providing even basic IP multimedia services like voice-over-IP call capability. It is reasonable to expect basic voice over IP to be desired by most subscribers.
- Careless reuse by subscribers of IP multimedia applications originally written for a non-wireless environment may lead to quality issues and customer dissatisfaction with the operator, although not the network operator's fault. In addition, stereotypes about the lack of reliability of wireless networks may be perpetuated, leading to loss of customers and/or revenue.

Third-party application provider perspective:

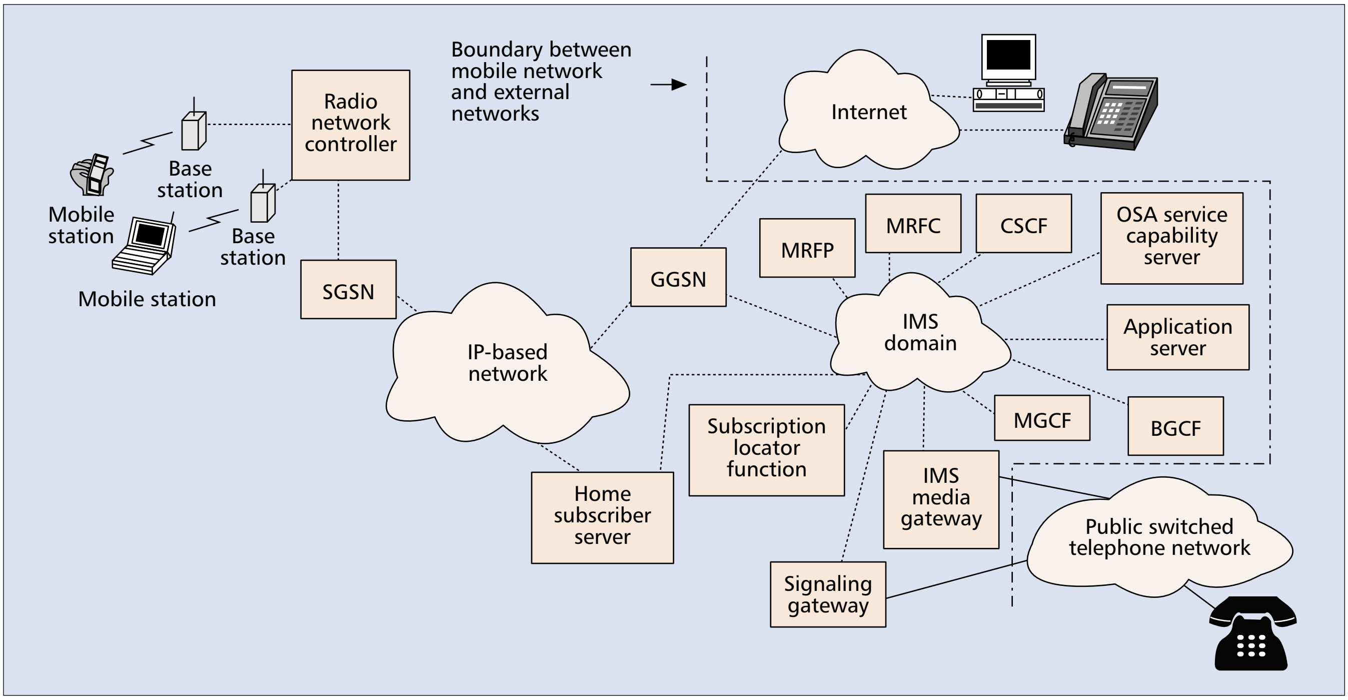
The advantages of providing only basic IP connectivity include:

- They do not have to bother with peculiarities of the wireless network, and can focus on developing IP multimedia applications for both wireless and wireline platforms.
- They do not need extensive knowledge of wireless telecommunications network infrastructures or protocols.

UMTS Releases

The reader may notice our references to various releases of UMTS (Releases 4, 5, and 6). There are multiple releases of UMTS because work on UMTS continues as the specifications are continually enhanced. Each release provides a stable version of specifications for vendors and operators to use in implementation and deployment. There used to be an annual release from 1996, where changes to GSM were incorporated (Release 96, Release 97, etc., by year). However, what was to become Release 2000 was split to become Release 4 and Release 5. Release 4 was frozen in early 2001 and contains the groundwork for the future all-IP wireless network. Many features, including the IP multimedia session control features, were postponed and first specified in Release 5, frozen in mid-2002. 3GPP currently is working on Release 6.

We mentioned that UMTS Release 4 provides IP connectivity, but neither IP multimedia services nor IP multimedia session control. A possible point of confusion may result from awareness that in Release 4, a common IP-based packet-switched transport for voice and data may be used within the mobile network, unlike in previous releases where voice and data use separate circuit-switched and packet-switched transports, respectively. However, this is an innovation only in the circuit-switched domain in the core network, allowing voice to be packetized at media gateways controlled by mobile switching center servers and packet-switched internal to the provider network. Packet-switched voice in the circuit-switched domain is not end-to-end, and the control signaling is still based on circuit-centric GSM-evolved control protocols. For data traffic, however, the network does not provide capabilities for IP multimedia session control (setting up sessions, tearing down sessions, etc.). It only provides IP transport, not end-to-end IP multimedia services. In contrast, from Release 5 onward, end-to-end IP multimedia sessions can be managed.



■ **Figure 2.** *Network architecture.*

The disadvantages include:

- They are unable to take advantage of the wireless network and its underlying network capabilities to obtain user location information, presence information, and so on.

UMTS Release 5 is designed to overcome the drawbacks while keeping the advantages of providing only basic IP connectivity. In addition to basic IP connectivity, session initiation capabilities are provided by the new IMS. Open service access (OSA) is being introduced, allowing subscribers to choose between service providers for enhanced IP multimedia services beyond basic services by opening the network to third-party application providers in a nonchaotic way.

OSA is a middleware framework that abstracts the capabilities of the underlying network (e.g., providing user location information). Third parties can then access these capabilities in a controlled and secure way without requiring detailed understanding of how the capabilities are provided. This breakthrough will allow many innovative new multimedia services to be provided.

IP Multimedia Subsystem

The IMS comprises the network elements for control of multimedia sessions, and is therefore essential for the provision of IP multimedia services in UMTS. IP multimedia services utilize the GPRS network for transport. The network provider thus provides both the transport and network control elements for IP multimedia services. The network architecture also allows for third-party offering of additional IP multimedia services.

IMS Concepts

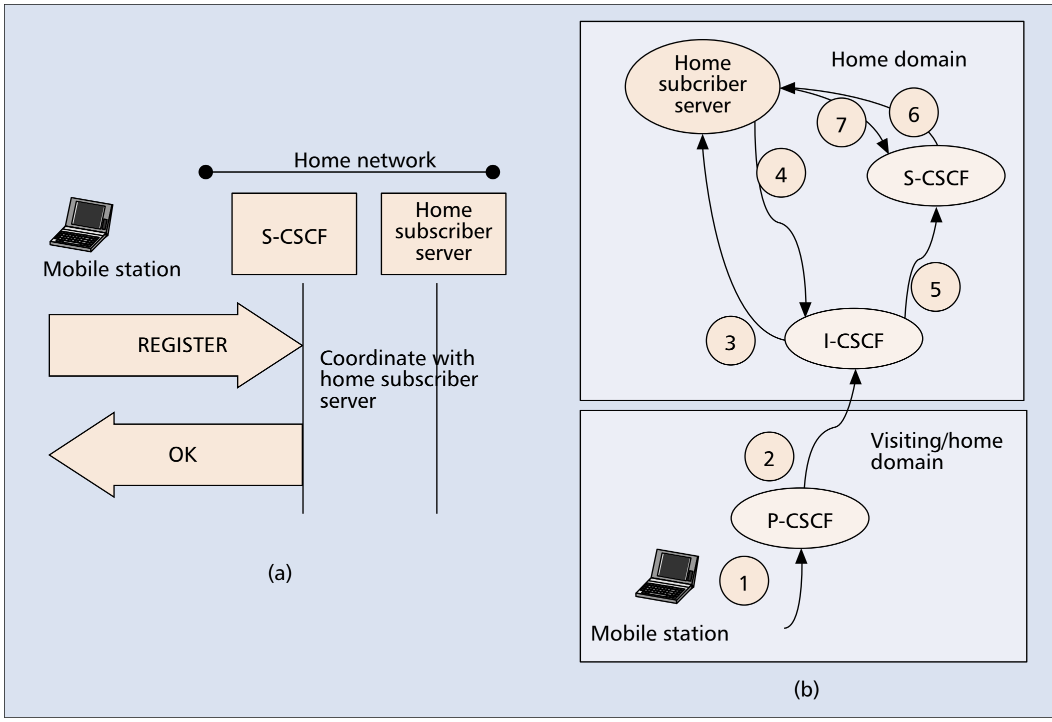
A major addition to the UMTS network appearing in Release 5 and beyond is the IMS, which adds two main new capabilities. First, new network elements like the call state control function (CSCF) are added to allow provisioning of basic IP multimedia services, such as initiation of multimedia sessions between two subscribers. Second, hooks are being developed to abstract the capabilities of these network elements as OSA service capability functions. OSA service capability functions are expected to stimulate innovative new IP multimedia services by third parties, taking advantage of basic IP multimedia services and capabilities provided by the network.

Figure 2 is a simplified picture of the UMTS Release 5 network architecture. It shows the essential network elements used in providing real-time IP multimedia services. Since the IMS uses only the packet-switched domain for transport and local mobility management, it can be deployed without a circuit-switched domain. Hence, circuit-switched domain elements like mobile switching centers are not shown in the figure. Other network elements pertinent to the provision of services other than IP multimedia services (e.g., SMS) are also omitted. However, the complete network reference architecture can be found in [3].

The IMS includes one or more CSCFs, media gateway control functions (MGCFs), IMS media gateways, multimedia resource function controllers (MRFCs), multimedia resource function processors (MRFPs), subscription locator functions, breakout gateway control functions (BGCFs), and application servers. The roles of these elements will be explained as we now address how the IMS performs major functions and adds multimedia support features to UMTS.

IPv6 Inside — IMS signaling and session traffic are carried over IP. Moreover, not just any version of IP can be used; IMS boldly requires that only IPv6 must be used in the IMS domain [4]. Although it would make UMTS the first global commercial system with widespread deployment

Many complex issues have arisen in the design and standardization of UMTS. For example, it was finally decided in late 2000 that service control for a subscriber is always performed by its home network. This applies even when the subscriber is roaming in a foreign network.



■ **Figure 3.** *Registration flow: a) the basic idea; b) with addition of I- and P-CSCF.*

of IPv6 (UMTS could be the "killer application" that kick starts IPv6), it is a bold requirement because

- The IMS may be (but does not have to be) a different IP addressing domain than that used for the packet-switched domain backbone (may be IPv4) and the circuit-switched domain backbone, so interworking issues between IPv4 and IPv6 need resolution.
- There is a lack of experience in IPv6.

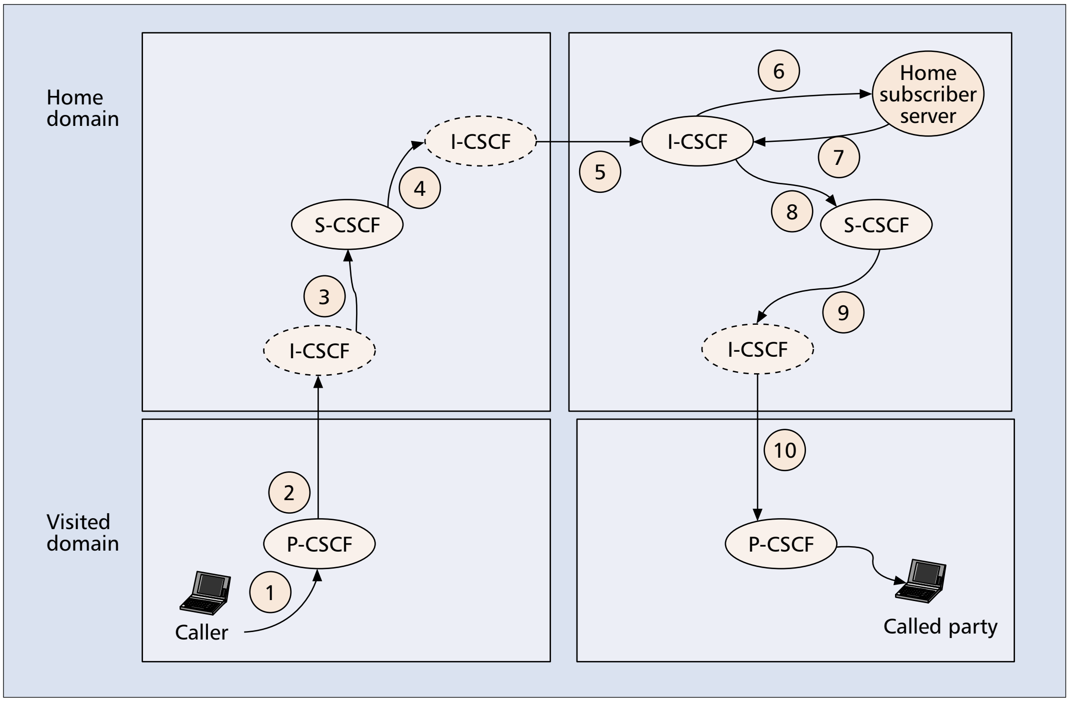
The IP address used by the mobile station to access IP multimedia services must be from the IMS addressing domain, which can be arranged in PDP context activation when setting up IP connectivity. It is considered a possible benefit in terms of routing efficiency that this address be obtained from a GGSN in the serving network rather than the home network. Earlier in the standards process, it was even *required* that this address be obtained from a GGSN in the serving network, but the requirement was removed, allowing more flexibility in choice of GGSN.

Call/Session Control — The CSCF plays a central role in call control and uses SIP. The CSCF is analogous to the signaling and control part of mobile switching centers for circuit-switched voice calls. However, it is capable of supporting multimedia sessions, not just voice calls. SIP is used for signaling between mobile station and CSCF, between CSCFs, between CSCF and MGCF, and between CSCF and application servers. The CSCF plays different roles: as a proxy CSCF (P-CSCF) it is the first point of contact between a mobile station and the IMS; as a serving CSCF (S-CSCF) for session control; and as an interrogating CSCF (I-CSCF), the main contact point in a network for all IMS-related signaling for its mobile stations. Call flows in the next section illustrate these CSCF roles.

PSTN Interworking — While at least one end of the session is an IMS subscriber, the other end for voice-only sessions may often be a public switched telephone network (PSTN) subscriber. Four new functional elements — the IMS media gateway, the MGCF, the signaling gateway, and the BGCF — support scenarios where calls go between the PSTN and the PS domain. The IMS media gateway performs media translation: translation between media signals encoded in one format on one side and signals encoded in another format on the other side. The MGCF controls the IMS media gateway, provides application-level signaling translation between SIP-based and ISUP-based (ISDN User Part, which is used on the PSTN side) signaling, and communicates with the S-CSCF. Transport-level signaling translation between IP-based and SS7-based transport is performed at the SGW. The BGCF identifies the network and the MGCF within that network, where the breakout to the PSTN should occur.

Handling User Subscription and Status Information — The home subscriber server (HSS) is a master database that stores information like subscription and location information. It can be thought of as an enhanced version of the home location register found in GSM. Since there may be multiple HSSs in a network, the subscription location function is queried by CSCFs during registration or session setup to find the HSS that has the desired subscriber information. The subscription location function is not needed in a single HSS environment since the CSCFs would know which HSS to use.

Various application servers are supported, including SIP-based application servers and OSA application servers, where application servers are servers hosting applications that can implement



■ **Figure 4.** *Wireless-to-wireless session setup: call flow.*

Originally, SIP did not have some capabilities needed for commercial use in the many scenarios for which it has been, and continues to be, proposed. However, extensions and modifications of SIP are being developed to address these concerns.

all kinds of services, such as voice announcement services and prepaid billing enforcement. The support of a variety of application servers facilitates the addition of innovative new services. In both cases, SIP is used for signaling with S-CSCFs. However, in the case of an OSA application server, an OSA service capability server is inserted between the OSA application server and the S-CSCF.

Multiparty Conferencing — The media resource function processor (MRFP) and media resource function controller (MRFC) support multiparty multimedia conferencing and media resource (e.g., for playing announcements) capabilities. The MRFC controls the media stream resources of the MRFP, which processes and mixes the actual media streams.

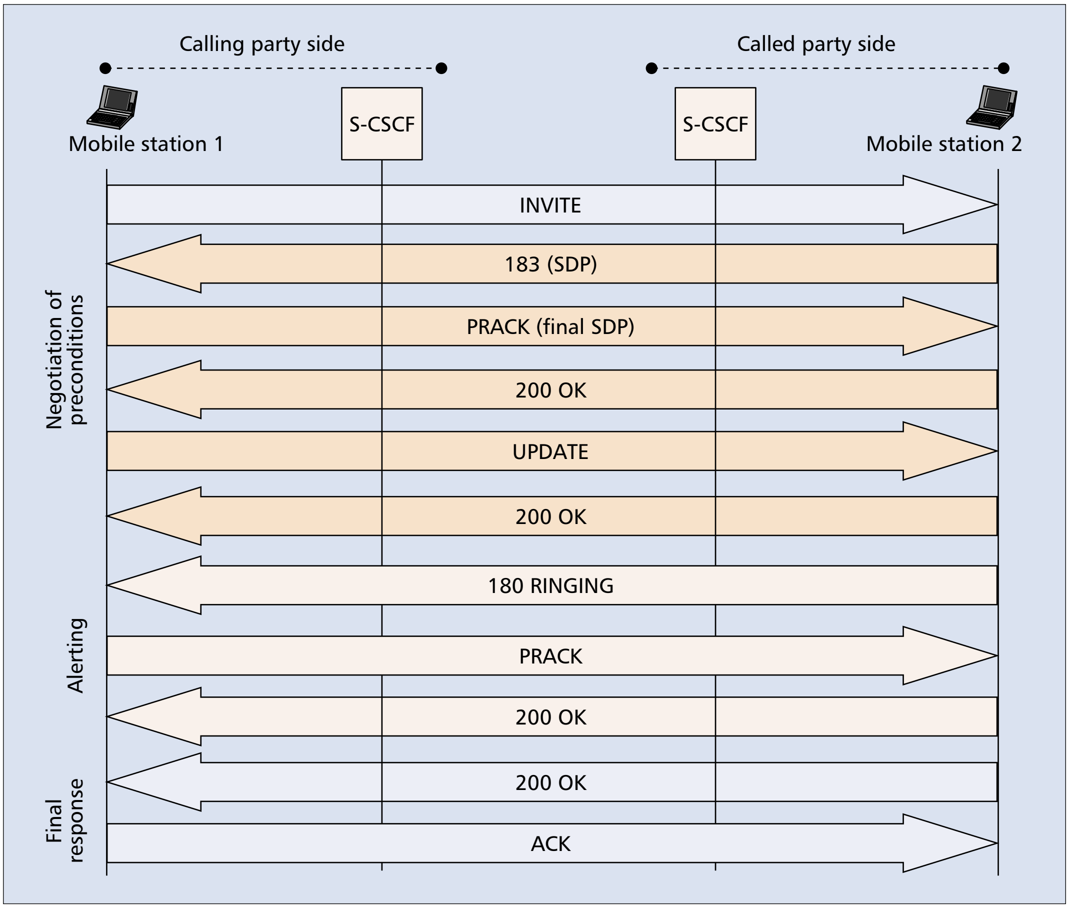
Other Issues — Many complex issues have arisen in the design and standardization of UMTS. For example, it was finally decided in late 2000 that service control for a subscriber is always performed by its home network. This applies even when the subscriber is roaming in a foreign network. Another option, visited network service control, was also considered earlier on, because of potential disadvantages of control by only the home network, such as overloading of the home network and delay in the handling of packets. The 3G Partnership Project (3GPP, the umbrella of standards organizations specifying UMTS) debated for a while the support of both home and visited service control by the IMS. However, support of both models makes the architecture more complex. Therefore, 3GPP eventually decided to work on home network service control, and possibly revisit the issue in the future.

Originally, SIP did not have some capabilities needed for commercial use in the many scenarios for which it has been, and continues to be, proposed. However, extensions and modifications of SIP are being developed to address these concerns. For example, originally SIP did not provide reliable delivery of what is known as provisional responses. These are informational responses (e.g., to a SIP INVITE message) informing the initiating party that the other party is being alerted. One or more provisional responses would be followed by one or more definitive responses. In many cases, it does not matter if some provisional responses are lost as long as the definitive responses are not. However, in an operator-controlled mobile network or other situations where provisional responses are tied to state machine transitions, reliable delivery of these provisional responses is essential. The provision response acknowledgment (PRACK) message has been added for reliable acknowledgment of provisional responses. An example of the usage of PRACK is provided in the next section.

IMS Procedures

In this section we discuss procedures used for the provision of services in the IMS. For illustrative purposes, the typical procedures for application-level registration and session setup are discussed. IMS procedures also allow various other services (session transfer between terminals, media modification, session redirection, etc.) [5]. We consider the general case of roaming mobile stations in visited networks. The visited and home networks are both involved. Procedures when either one mobile station is in its home network, or both mobile stations are in their respective home networks, are straightforward to deduce from the general case. We assume a single HSS environment, so the sub-

The decision to base the signaling of the IMS on standard IP protocols like SIP is the right decision, enabling future inter-operability and network convergence. Wisely, the door is open for 3rd party applications through OSA, which could greatly help the success of UMTS multimedia services.



■ **Figure 5.** *Control signaling for wireless-to-wireless session setup.*

scription locator function (SLF) is not needed.

To obtain IP multimedia services using the IMS, the mobile station authenticates itself, and attaches, to the UMTS network using GPRS attach procedures. The mobile station then activates one or more PDP contexts. The primary PDP context is usually used for signaling. The mobile station may activate secondary PDP contexts, while reusing the same PDP address, but with different QoS profiles as appropriate.

So far, these first two steps are not IMS-specific, just the standard way to obtain IP connectivity over GPRS. The next steps are to get set up for IMS services. Step 3 is to discover a P-CSCF in the network the mobile station is in that it can use to perform the IMS registration procedure. Several ways to perform P-CSCF discovery are possible, including the use of DNS. The fourth step is IMS registration.

REGISTRATION

Before multimedia sessions can be established, the mobile station must perform application-level registration with the home network, during which the home network dynamically assigns an S-CSCF that will perform session control and store the P-CSCF information for this mobile station during the registration lifetime. Figure 3 shows a successful registration procedure. SIP registration is used. The mobile station sends a SIP REGISTER message to the P-CSCF (step 1) including its home network domain name. The P-CSCF examines the home domain name to discover the entry point to the home network (i.e., I-CSCF) and forwards the REGISTER message to the I-CSCF (step 2). The I-CSCF queries the HSS (step 3) to request information on S-CSCF capabilities, and the HSS responds (step 4). Based on the required S-CSCF capabilities and availability of S-CSCFs, the I-CSCF selects an S-CSCF for the mobile station. The I-CSCF forwards the REGISTER message to the S-CSCF (step 5), which interacts with the HSS (steps 6 and 7) to store the S-CSCF name and download relevant information from the subscriber profile. The S-CSCF returns a SIP 200 OK message to the mobile station via I-CSCF and P-CSCF.

SESSION SETUP

We consider the establishment of a session between two mobile stations where both are in visited networks, as shown in Fig. 4. Sessions will always involve two S-CSCFs performing service control, one for each mobile station, regardless of whether the mobile stations are in their home networks or in visited networks. With the use of well-known I-CSCFs as contact points with external networks, the internal topology (e.g., S-CSCF information) of a network can be hidden. I-CSCFs between network boundaries are required in the signaling path for session setup only if network topology hiding is required. The exception is the I-CSCF in the called party's home network that queries the HSS to determine the S-CSCF to contact, since this query is always necessary. After the initial INVITE, the I-CSCF in the called party's home network may be left out of the signaling path for subsequent exchange of messages in the session setup, except for provisional and final responses to the INVITE. Otherwise, the subsequent messages (described next) follow the same path taken by the INVITE.

The signaling exchanges during session setup are shown in Fig. 5. In the first phase, negotiating preconditions, mobile station 1 sends mobile station 2 a SIP INVITE request, containing supported multimedia capabilities. Mobile station 2 returns the subset of multimedia capabilities that it finds acceptable in a provisional 183 response. Mobile station 1 sends mobile station 2 a PRACK message to acknowledge receipt of the provisional response, including the mutually agreed session parameters. After securing resources for the session, mobile station 1 sends an UPDATE message indicating success of the resource reservation. With the 200 OK response, negotiation of preconditions is completed. The alerting phase occurs next, where the 180 RINGING response needs reliable delivery; therefore, PRACK is used again. When the user has answered the session, the final response phase is entered and the media flow begins between the two mobiles. The media flow will be between the two mobile stations, with the arranged QoS guarantees, and will not flow through the CSCFs.

If mobile station 2 is in the PSTN, the S-CSCF in the caller's network will determine that the call should go to the PSTN, and will forward the INVITE to a BGCF in the same network. The BGCF will forward the INVITE to an appropriate MGCF. If mobile station 1 is in the PSTN, the MGCF in the IMS acts as the SIP endpoint that initiates requests on behalf of the PSTN, as if the signaling came from an S-CSCF.

Discussion and Conclusions

With 3G wireless, order-of-magnitude higher data rates are available. The development of the IP-based network and related signaling components for IP multimedia services is similarly exciting in that many new services will become available to make use of the higher data rates. The decision to base the signaling of the IMS on standard IP protocols like SIP is the right decision, enabling future interoperability and network convergence. Wisely, the door is open for third-party applications through OSA, which could greatly help the success of UMTS multimedia services.

Latest Developments

The current ongoing work on IMS in UMTS is increasingly on the service aspects (incorporating concepts of presence, messaging, etc.), while updating the basic signaling and network foundations as appropriate (e.g., to keep up with the latest versions of SIP, and refining IPv4/IPv6 interworking). The services being worked on include push services and multimedia broadcast message services.

Summary

This article gives an overview of how real-time IP multimedia services will be provided in UMTS networks with the newly specified IMS. The latest advances in SIP and other Internet technologies like IP QoS and IP security, plus IPv6, come together in the UMTS platform that will support real-time IP multimedia services. Market trends point to its great potential. The IMS may be the platform for killer applications that rescue 3G, kick start IPv6, and usher us into the brave new world of mobile IP multimedia services.

The latest advances in SIP and other Internet technologies like IP QoS and IP security, plus IPv6, come together in the UMTS platform that will support real-time IP multimedia services. Market trends point to its great potential.

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Additional Reading

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Biographies

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