Toward an All-IP-Based UMTS System Architecture

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Abstract

Looking into the future, two main drivers for the mobile telecommunications market can be identified: third-generation mobile systems (e.g., UMTS) and the Internet (e.g., the introduction of IP technologies like voice/multimedia over IP in mobile networks). UMTS is seen as the enabler of wireless multimedia applications and portability of a personalized service set across network/terminal boundaries, as defined within the virtual home environment (VHE) system concept. In light of these recent evolutions, this article investigates the impact of the evolution toward an all-IP UMTS network architecture on the UMTS service architecture, which is based on the VHE concept. The article discusses two possible scenarios for supporting VoIP services in the UMTS service architecture and analyzes their applicability in an all-IP-based UMTS network. The first is based on the traditional centralized IN service architecture. The second proposes a new decentralized architecture based on direct control of VoIP call control equipment by open service architecture interfaces.

Beginning 1998 six partners — ARIB, T1, TTA, European Telecommunications Standards Institute (ETSI), CWTS, and TTC — started discussions to cooperate for the production of standards for a third-generation mobile system with a core network based on evolutions of the Global System for Mobile Communications (GSM) and an access network based on all the radio access technologies (i.e., both frequency- and time-division duplex modes) supported by the different partners. This project was called the Third Generation (3G) Partnership Project (3GPP) [1]. Almost one year later the American National Standards Institute (ANSI) decided to establish 3GPP2 [2], a 3G partnership project for evolved ANSI/Telecommunications Industry Association (TIA)/Electronics Industry Association (EIA)-41 networks. There is also a strategic group called International Mobile Telecommunications-2000 (IMT-2000) [3] within the International Telecommunication Union (ITU), which focuses its work on defining interfaces between 3G networks evolved from GSM on one hand and ANSI-41 on the other, in order to enable seamless roaming between 3GPP and 3GPP2 networks. Thanks to this worldwide — also called *universal* —roaming characteristic, 3GPP started referring to 3G mobile systems as the Universal Mobile Telecommunication System (UMTS). The rest of this article will focus on topics from a 3GPP standardization viewpoint.

In 3GPP, the UMTS specification work was divided into two phases. For the first phase of UMTS, Release 1999 or R99, standardization work was finished around the end of 1999 and the beginning of 2000. As a result, the first phase of UMTS will be available on the market around 2001. Whereas the first phase of UMTS was more or less a logical *evolution* from the 2nd generation system architecture, the second phase, called Release 2000 or R00, is a complete *revolution*, introducing many new concepts and features. The completion of all standardization work for this second phase is expected around the end of 2001 and the beginning of 2002. This means that commercial operation can be expected around 2004. This article will focus on explaining the UMTS R00 architecture, since this architecture includes the most advanced technologies that will give the user the most complete UMTS multimedia experience.

Since mid-1999 two remarkable trends appeared in 3GPP UMTS standardization which, in the meantime, have greatly influenced all further evolutions of the 3G standards. The first trend [4] was the shift toward an *all-IP UMTS network architecture*. This shift formed the basis for the R00 specifications. More specifically, the R00 all-IP UMTS specifications replace the circuit-switched transport technologies, which were still used in UMTS R99, by packet-switched (e.g., IP [5]) transport technologies and introduce multimedia support in the UMTS core network. Also, outside the official standardization bodies (i.e., 3GPP and 3GPP2) a number of fora and partnerships, between manufacturers and operators (e.g., 3G.IP, MWIF), heavily contributed to the immense level of success in industry of the all-IP UMTS network architecture. The second trend [6] was the evolution toward an *open service architecture* (OSA) which obliged network operators to provide third party service providers access to their UMTS service architecture via open standardized interfaces. Regulatory bodies (e.g., European Commission) all over the world pushed for the opening of network interfaces because it would foster the liberalization of the telecommunications services market by enhancing the portability of telecommunications services between networks and terminals. This concept of service portability was called the *virtual home environment* (VHE) in 3GPP standardization. As regulatory bodies correctly understood, the only way to realize the VHE philosophy — that is, to make it possible for third party service providers to develop UMTS applications that can run on several networks and terminals — is to open/standardize the application interfaces toward these networks (i.e., to standardize OSA). Several standardization bodies (3GPP, etc.) and consortia (Parlay, JAIN, etc.) contributed to the philosophy behind OSA.

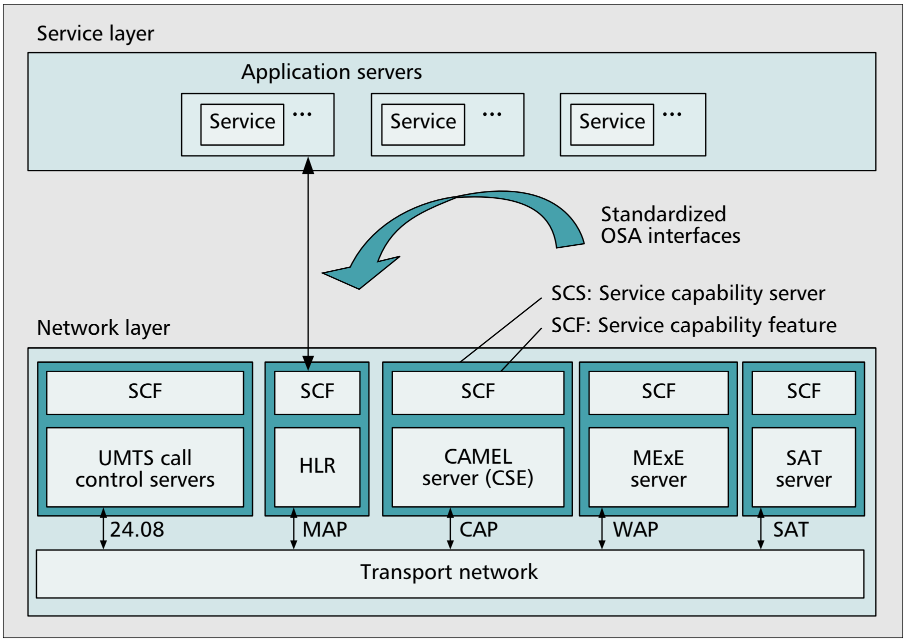


Figure 1. *The concept of the virtual home environment (R99).*

In light of the recent evolutions in 3G standardization, this article further investigates the synergy between the two trends mentioned above: on one hand, the trend in the design of the UMTS *network* architecture to move toward an all-IP approach, and on the other, the trend in the design of the UMTS *service* architecture to standardize open network interfaces. The goal of the article is to clarify the implications of an IP-based core network design on the UMTS service architecture and to analyze possible evolution paths, integrating both network and service aspects, toward a complete all-IP UMTS system architecture.

The article is organized in the following way. The next section gives an introduction to the VHE concept and its realization via OSA interfaces. We then give an introduction to voice over IP (VoIP) in mobile networks and explain how VoIP can be supported in the all-IP UMTS core network architecture. The article then further analyzes the impact of an IP-based core network design on the UMTS service architecture. Two possible scenarios are discussed for supporting VoIP services in the UMTS service architecture based on the principles of the VHE. Finally, the article concludes by evaluating the coexistence of both scenarios: the classical centralized — intelligent network (IN) type — service control architecture and the new decentralized (OSA) type service provisioning architecture.

Introduction to the Virtual Home Environment

The VHE Concept: Standardizing Service Capabilities Instead of Services

In the beginning of the 1990s, UMTS was defined in Europe as the third-generation mobile telecommunications system that would replace the current GSM standard. While GSM booked a major success compared to classical fixed telephony systems mainly thanks to the *mobility* aspect, the main goal of UMTS is to offer a much more attractive and richer set of *services* to the user.

In order to achieve a sufficient degree of service differentiation, UMTS needed three fundamental architectural improvements from GSM:

- *Wideband access*: Higher bit rates over the air open the path toward mobile multimedia applications.
- *Mobile-fixed-Internet convergence*: There is a need for a uniform way to offer users cross-domain services. An example is the tracking of a user's location in the mobile, fixed, and Internet domains and automatically adapting the content of his incoming messages to SMS, voice message, fax, or e-mail. VHE is the enabler of this service portability across networks and terminals in the different domains.
- *Flexible service architecture*: By standardizing not the services themselves but the building blocks that make up services, UMTS shortens the time to market for services from GSM and enhances creativity/flexibility when inventing new services.

3GPP defined the VHE as "a system concept for personalized service portability across network boundaries and between terminals [6]." The aim was to enable end users to access the services of their home network/service provider even when roaming in the domain of another network provider, thus making them feel "virtually at home." VHE allows a user to personalize the set of services for which he/she has a subscription with his/her home network, and provides these home services with the user's personalized "look and feel" across different types of networks — mobile, public switched telephone network (PSTN), Internet — and terminals — mobile, laptop, fixed phone, PDA, PC — he/she might be using. An example of one of the personal service settings of a user could be "from 9h00 to 17h00 I want to be alerted for incoming messages from my boss." The VHE will automatically adapt the type of messaging used to reach the user to the capabilities of the terminal and network the user is using at that time: if the user is using a Wireless Application Protocol (WAP) terminal but is not roaming in a network that supports WAP, the VHE will convert the message into another format (e.g., SMS).

VHE, currently still under standardization in 3GPP [6], promotes the view (Fig. 1) that the UMTS service architecture should be a layered architecture enabling services to be developed independent of the underlying networks. This is achieved by standardizing the interfaces between the so-called network layer, comprising all network elements under the operator's control, and the service layer, comprising third-party servers running service logic. In this way the main goal of the separation between the network and service layers can be achieved: to allow faster, easier, and more flexible creation, deployment, and operation of new personalized applications/services.

The VHE specification [6] introduces some new terminology related to the way this new open interface between the network and service layers, called the OSA interface, is realized (Fig. 1). Service capability servers (SCSs) are defined as all those servers in the network that provide functionality used to construct services. From a software point of view, the OSA interface is defined as an object-oriented application programming interface (API). This means that all the functionality which can be provided by SCSs is grouped into logically coherent software interface *classes*. If we take the mobile switching center (MSC) as an example of an SCS, call control is a class consisting of several call control related functions, for example, "create a new call leg," "connect call leg A to call leg B" … The classes of the OSA interface are called service capability features (SCFs) in the VHE specifications. Practically speaking, the SCFs are not implemented as a new standalone box in the architecture; instead, they are just added as an additional software layer of interface classes on top of existing network elements, which are then called SCSs.

By providing services in the service layer access to the SCFs of all the SCSs in the network layer, OSA aims to offer a secure open standardized interface for service providers toward underlying networks. Security is ensured by additional authentication, authorization, accounting, and management interfaces toward all the SCSs. The service logic constructed according to this OSA principle resides in so-called application servers in the service layer. The SCSs and application servers are interconnected through, say, an IP-based network, which allows for distributed deployment of the SCSs and application servers.

To summarize, the purpose of the SCFs/SCSs is to:

- Raise the abstraction level of the network interfaces toward service providers and simplify application development. The SCFs offer a generalized view of the network functionality to third party application developers via standardized interfaces.
- Hide network-specific protocols and offer connectivity to both circuit-switched and IP networks.
- Protect core networks from misuse via authentication, authorization, accounting, and management interfaces toward all the SCSs

The UMTS Service Architecture: Open Standardized Interfaces on Top of Service Capability Servers

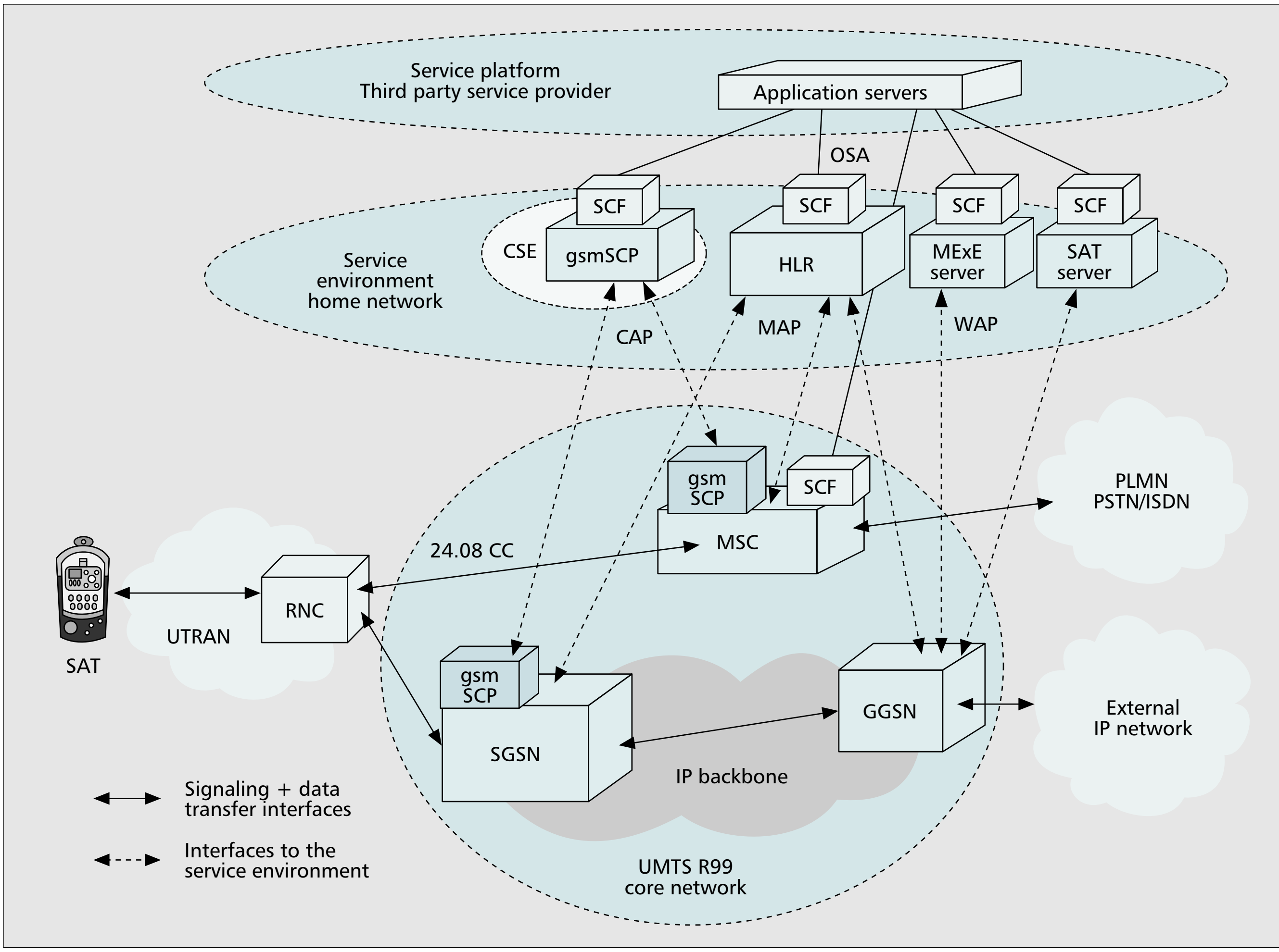
As explained previously, VHE defines SCSs and standardizes SCFs that the SCSs can provide to third party service providers to design new services (Fig. 1). Examples of SCFs are call control, location/positioning, and notifications. The functionality represented by the SCFs is offered via an open standardized interface, the OSA interface, toward the service layer above and is implemented by the underlying transport networks using GSM/UMTS protocols. Examples of such GSM/UMTS protocols are Mobile Application Part (MAP), CAMEL Application Protocol (CAP), and WAP.

As identified in R99 of the 3GPP VHE specification [6], the SCSs and their roles in service provisioning are:

- *UMTS call control servers*: As SCSs they offer mechanisms for applications to access basic bearer/call control capabilities. Since R99 only supports circuit-switched telephony, the only call control element is the MSC. The 24.08 CC protocol is the UMTS call control protocol.
- *Home location register (HLR)*: The HLR is an intelligent database that contains the location and subscriber information of all subscribers of the network to which it belongs. The MAP protocol allows the exchange of location and subscriber information between different network elements.
- *Mobile execution environment (MExE) server*: The mobile execution environment is the execution environment, which can be a Java Virtual Machine or a WAP browser, in the terminal. Value-added services are offered through a client/server relationship between the MExE server in the network and the MExE client in the terminal. WAP is a protocol designed to provide services to mobile terminals taking into account their limited capabilities; display, processing power, and so on. Wireless Telephony Application (WTA) is an extension to WAP that allows WAP applications to use telephony related functionality in the terminal and the network.
- *SIM application toolkit (SAT) server*: SAT is a mechanism that offers additional capabilities to the communication protocol between the subscriber identity module (SIM) card and mobile terminal. A SIM card is the smart card inserted in the mobile terminal. The SIM card contains on one hand certain subscriber and security related information used by the mobile network to authenticate the user and, on the other, some small applications (e.g., phone book, calendar, electronic wallet). The most important additional capabilities supported by SAT are the pro-active commands from SIM card to terminal; for example, the SIM card can instruct the terminal to download information. Via the SAT server the operator can control existing SAT applications on the SIM card and download new SAT applications to the SIM card.
- *Customized Application for Mobile Networks Enhanced Logic (CAMEL) server*: CAMEL extends the scope of IN [7] service provisioning to the mobile environment. CAMEL allows the provisioning of certain IN services (e.g., prepaid) to mobile networks and enables the exchange of mobile-specific service information, for example, related to SMS or GPRS, between the CAMEL network elements, the service switching point (SSP) and service control point (SCP). CAMEL services are invoked via triggers, which are contained in the SSP inside the MSC, to an SCP residing in the CAMEL service environment (CSE).

We must remark here that there is not necessarily a relationship between the different SCSs. Some simple services only require a UMTS bearer. For other services, like WAP, an MExE server is essential. For location-based services it is necessary to consult the HLR, and if you want to provide IN services to mobile phones CAMEL is needed.

To give a practical example, Fig. 2 illustrates how services can be delivered in the UMTS R99 architecture; by the home network operator as well as third party service providers. Traditionally network operators provide services via servers (e.g., MSC, SCP, HLR, MExE server, SAT server,) and protocols (e.g., MAP, CAP, WAP, SAT) controlled completely from inside the operator's private network/service environment. The novelty of the UMTS service architecture is that, via the OSA interfaces toward the operator's SCSs, third party service providers can also start offering services. In this case the actual service logic is run on application servers in the third party domain, but it uses capabilities of the underlying network that it can access via the OSA interfaces toward the operator's SCSs.



■ Figure 2. *Mapping of SCFs to the Release 99 network architecture.*

An Introduction to VoIP in Mobile

Why Is IP Technology Interesting for Operators as Well as End Users?

In the mid-1970s experiments were conducted for ARPANET to encode voice through a packet switch. A pioneer in the field of "packet voice" was Danny Cohen, at that time working for the Information Sciences Institute (ISI). Since 1992 audio conferencing experiments have been conducted on the MBone. In 1995 work started on the development of the H.323 protocol suite within ITU. Approximately two years later the IETF started developing its own VoIP protocol, SIP. Since then interest in VoIP has spread worldwide across academia as well as industry. Nowadays even the most traditional telecom operators are taking VoIP seriously.

It is believed that IP will be capable of carrying all types of data, including real-time data like voice. Using VoIP has several advantages over traditional telephony. For network operators it means lower equipment cost and management of the network. Using VoIP with techniques like silence suppression can result in a bandwidth gain of a factor of four compared to 64 kb/s PCM connections [8]. This, in turn, can result in lower communication costs to users.

Last but not least, the use of end-to-end IP sessions with higher bandwidths as in UMTS opens the path for mobile end users to a whole new set of multimedia over IP services such as videoconferencing, personal guidance systems, and network games. These services are believed to be some of the main drivers for UMTS. Using the same technology (i.e., IP services) in fixed and mobile networks facilitates interworking between both types of networks; also, the development and creation of new services is provided in a consistent way.

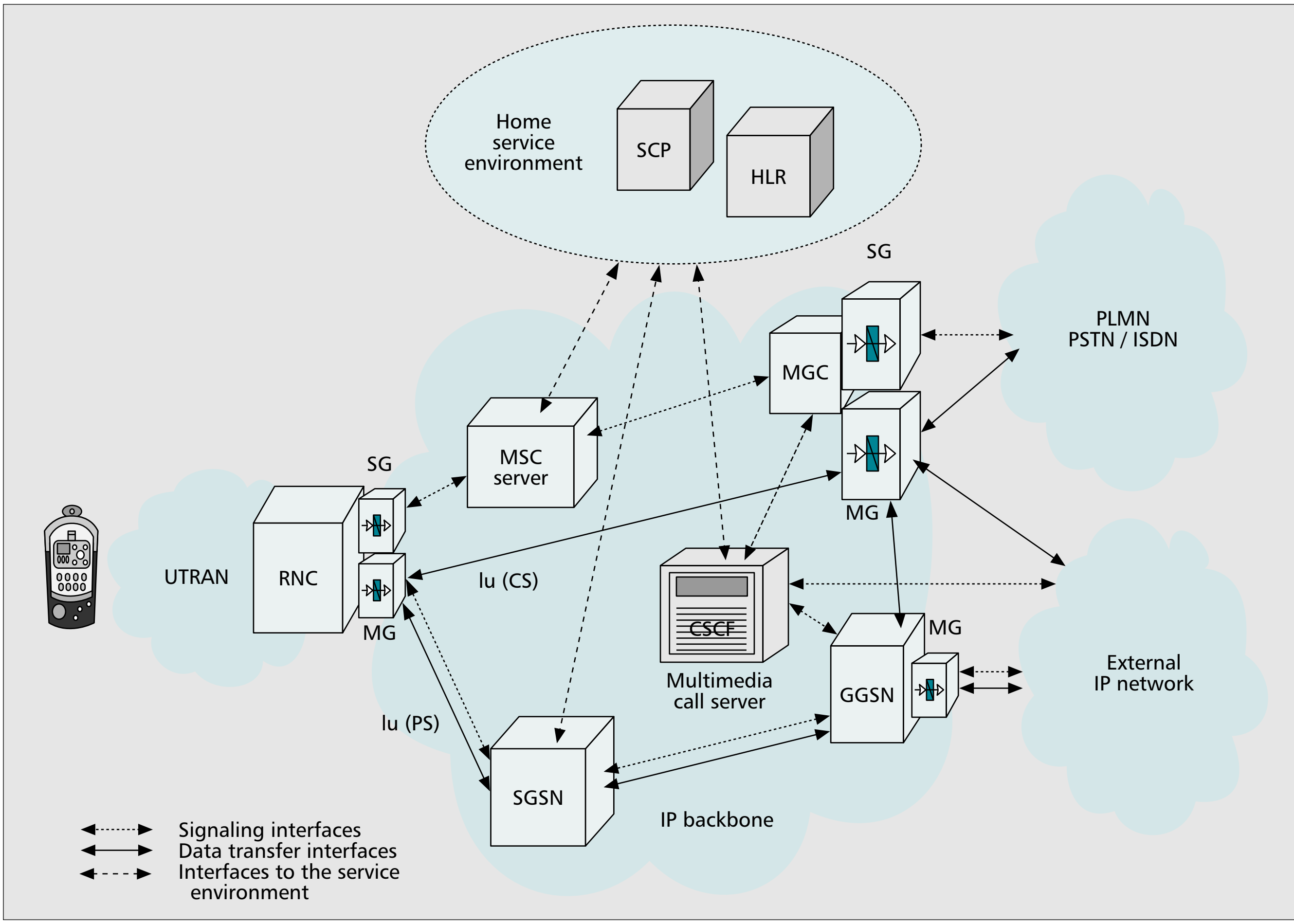
A big technological challenge still to be solved in the context of real-time VoIP services is the provisioning of sufficient QoS, especially in the context of mobile networks, to control delays introduced by handover, manage scarce radio resources, and perform admission control.

The All-IP UMTS Solution

The first release of UMTS will be based on the R99 specifications. The second release, R00, is an evolution of R99. The major innovation of R00 is the introduction of the IP multimedia domain. The following section concentrates on R00. The following new features are introduced in Release 2000:

- Provisioning of IP-based multimedia services as an extension of the packet-switched services.
- Enabling a bearer-independent circuit-switched network architecture. Circuit-switched transport is replaced by packet-based network transport.
- IP transport within the UTRAN (i.e., on Iub and Iur).
- Network architecture is independent of the transport layer, which can be based on either ATM or IP.

In the context of this article, an all-IP solution for UMTS refers to an all-IP core network (Fig. 3). The use of IP in the UTRAN will not be discussed in this article. In the all-IP core network, all data is transported on IP, including even traditional circuit-switched voice data. R00 supports two types of real-time services. The first is a circuit-switched voice service, and the second is an IP-based multimedia service. In the R00



■ Figure 3. *A simplified Release 2000 all-IP architecture.*

specification, the classical MSC is split into a control part, the MSC server, and a transport part, the media gateways.

The requirements for an all-IP core network are summarized as follows [4]:

- Support of roaming and handover to 2G networks (e.g., GSM, GPRS).
- Support of 3G circuit-switched terminals in a full IP UMTS core network, providing backward compatibility with R99 terminals.
- Support of new (e.g., IP and multimedia) as well as existing services, such as speech, SMS, and supplementary IN services. Support of legacy services is required since subscribers accustomed to the services in GSM may not be willing to sacrifice these services when migrating or roaming toward the new UMTS system.

The second requirement implies that there will be three types of 3G mobile terminals: circuit-switched, packet-switched (IP), and those that support both modes. Both circuit- and packet-switched modes are supported at the radio interface. The circuit-switched mode is used for traditional circuit-switched terminals and makes optimal use of the radio resources for voice services. Circuit-switched voice is optimized in terms of both bandwidth (small frame protocol overhead) and quality (the codec rate is adapted to the radio link quality, and every voice sample is split into three streams, each with a different level of error protection/correction). The packet-switched mode is more flexible in terms of services supported and allows the introduction of multimedia applications, but is less efficient in terms of bandwidth consumption due to the IP header overhead over the radio.

There are two major protocols for supporting VoIP: SIP, standardized by the IETF [9], and H.323, standardized by the ITU [10]. Recently, it was decided in 3GPP to use only SIP as the call control protocol between terminals and the mobile network. Interworking with other H.323 terminals (e.g. fixed H.323 hosts) will be performed by a dedicated server in the network. Figure 3 shows the proposed 3GPP all-IP UMTS core network architecture [4].

New elements in this architecture are:

- *MSC server*: The MSC server controls all calls coming from circuit-switched mobile terminals and mobile terminated calls from a PSTN/ISDN/GSM network to a circuit-switched terminal. The MSC server interacts with the media gateway control function (MGCF) for calls to/from the PSTN. R00 introduces the functional split of the MSC, where the call control and services part is maintained in the MSC server, and the switch is replaced by an IP router (MG). This functional split reduces the deployment cost and guarantees the support of all existing services.
- *Call state control function (CSCF)*: The CSCF is a SIP server that provides/controls multimedia services for packet-switched (IP) terminals, both mobile and fixed.
- *MG at the UTRAN side*: The MG transforms VoIP packets into UMTS radio frames. The MG is controlled by the MGCF by means of Media Gateway Control Protocol H.248. The media gateway is added to fulfill requirement two. In Fig. 3, the MG is drawn at the UTRAN side of the Iu interface, so the Iu interface, between the core network and UTRAN, is IP-based. The MG can also be located at the core network side of the Iu interface. In this case, the Iu interface remains unmodified from R99, without impact on the UTRAN.
- *MG at the PSTN side*: All calls coming from the PSTN are

translated to VoIP calls for transport in the UMTS core network. This MG is controlled by the MGCF using the H.248 protocol.
- *Signaling gateway (SG)*: An SG relays all call-related signaling to/from the PSTN/ UTRAN on an IP bearer and sends the signaling data to the MGCF. The SG does not perform any translation at the signaling level.
- *MGCF*: The first task of the MGCF is to control the MGs via H.248. Also, the MGCF performs translation at the call control signaling level between ISUP signaling, used in the PSTN, and SIP signaling, used in the UMTS multimedia domain.
- *Home subscriber server (HSS)*: The HSS is the extension of the HLR database with the subscribers' multimedia profile data.

For the transport of data traffic, UMTS uses the General Packet Radio Service (GPRS) network. For voice calls, there are two options: for packet switched mobile terminals, voice data is transported over the GPRS network using the GPRS Tunneling Protocol (GTP) on top of IP. All mobility is solved by the GPRS protocols. For circuit-switched mobile terminals, voice samples are transported over IP between the MGs using the Iu Frame Protocol (FP). In the latter case there is no GTP tunneling; hence, mobility has to be solved in a different way, namely by MG handovers.

Two Possible Scenarios for Providing VoIP Services in VHE

For several years the boundary between mobile operators and Internet service providers has been blurring due to cross-area expansion (VoIP, mobile IP, GPRS, WAP). The requirement to open the UMTS network to service providers will accelerate even more the development and deployment of services that combine telecom and datacom features (e.g., VoIP, MMoIP).

As explained above, the UMTS architecture is enhanced in R00 to also cover VoIP/MMoIP services. Let us again study the R00 all-IP architecture (Fig. 3), as explained previously, and try to map this to the concept of VHE (Fig. 1), which was explained earlier. Note that Fig. 1 represents the R99 view of VHE and Fig. 3 the R00 all-IP architecture. Comparing Fig. 1 and Fig. 3 we can easily detect that, in order to derive the R00 VHE picture, a new additional call control element needs to be added to Fig. 1 to incorporate the novelties of the R00 all-IP architecture shown in Fig. 3. In Release 1999 the only call control element was the MSC providing circuit-switched telephony services. In Release 2000 there are two call control elements: the MSC server for delivering traditional circuit-switched telephony services, and the CSCF or SIP server for delivering the new VoIP/MMoIP services. Since the R99 MSC is simply split into two parts (MSC server and MG) in R00, without any major functional changes, circuit-switched services can be provided in exactly the same way as in R99: via the CAMEL platform. The CSCF, on the other hand, is a call control element not present in R99 at all, introducing totally new multimedia capabilities. Since there is no standard way yet in mobile history to provide multimedia services via a CSCF, several possible options could be explored.

As suggested in Fig. 4, there are two possible scenarios for the deployment of VoIP services. VoIP services can be provided based on either classical IN/CAMEL service control [7] via the operator's SCP (A in Fig. 4) or third party call control mechanisms (B in Fig. 4). For the latter case, an open standardized interface directly on top of the CSCF is needed. This OSA interface can be implemented in several ways, using, for example, CGI, CPL [12], or even SIP. In the following paragraphs, the two scenarios and their impact on the UMTS service architecture will be explained in further detail.

Scenario A: The "SoftSSP" Concept; INAP/CAP Support of VoIP

In the old days operators used to implement service logic directly in the network switches. IN is a mechanism designed for operators to control the provisioning of services in their networks from a centralized point, the SCP, outside of the switch network. IN relies on SSPs in the switches to trigger the SCP via the IN Application Part (INAP) protocol when IN service control is needed. The main advantage of IN is that it offers operators a much more scalable service platform, which allows them to introduce new services in a more flexible and faster way. With the success of GSM, a mobile version of IN, CAMEL, was designed. The equivalent of INAP for IN is the CAMEL Application Part (CAP).

IN and CAMEL were developed in several releases, each new IN/CAMEL version supporting new functionality. The power of IN/CAMEL lies in the degree of complexity of the SSP and INAP/CAP. In order to be able to provide the correct triggers to the SCP, the SSP contains a mapping that determines which point in the MSC call state model needs to trigger which point in the state model of the IN/CAMEL service logic. The more complex this mapping, the more complex services can be provided. This means that in order to provide services via IN/CAMEL on a SIP server, all you need to do is develop an SSP on top of the SIP server: a mapping between the SIP call state model and the state model of the IN/CAMEL service logic. This kind of SSP is called a "SoftSSP."

Concentrating again on mobile networks, it is clear that the main advantage of this SoftSSP approach (A in Fig. 4) is that the operator's R99 investments in CAMEL can be reused to provide VoIP services on a CSCF. When designing an R00 SoftSSP for SIP call control, many traditional auxiliary processes, such as database handling and billing, can be reused from the R99 SSP for circuit-switched call control. Either they can be retained completely as they are, or they need some enhancement to reflect functionality specific to multimedia.

The interface between the CSCF and the SoftSSP call control processes must:
- Carry sufficient call data for the SoftSSP to function correctly and to deliver the necessary information to the SCP so that service logic decisions can be made.
- Allow the SCP in combination with the SoftSSP to control VoIP calls (e.g., change "B" party address, add/subtract media components) and to manipulate call information (e.g., presentation number)

This scenario — with SCP control of both existing CAMEL services and new VoIP services — can offer some advantages for *existing* operators since they already own a traditional (UMTS R99 or even GSM) circuit-switched network controlled by a CAMEL service platform. In the SoftSSP scenario all applications, for legacy as well as new VoIP/MMoIP services, can be created according to the same proven CAMEL service creation environment methods. Based on a dedicated mapping between CAP and SIP call control, VoIP/MMoIP services are, just like traditional CAMEL services, under control of the operator's SCP.

In this scenario, third party service providers can get access to the operator's network via the OSA interfaces only via a central access point, the SCP. Third party service providers cannot get direct access to the CSCF. The fact that in this scenario third party service provisioning always relies on the operator's underlying CAMEL network implies that, in the development of new VoIP/MMoIP services, third party service providers are inevitably limited by the capabilities of the CAMEL version supported by the network operator. This is a

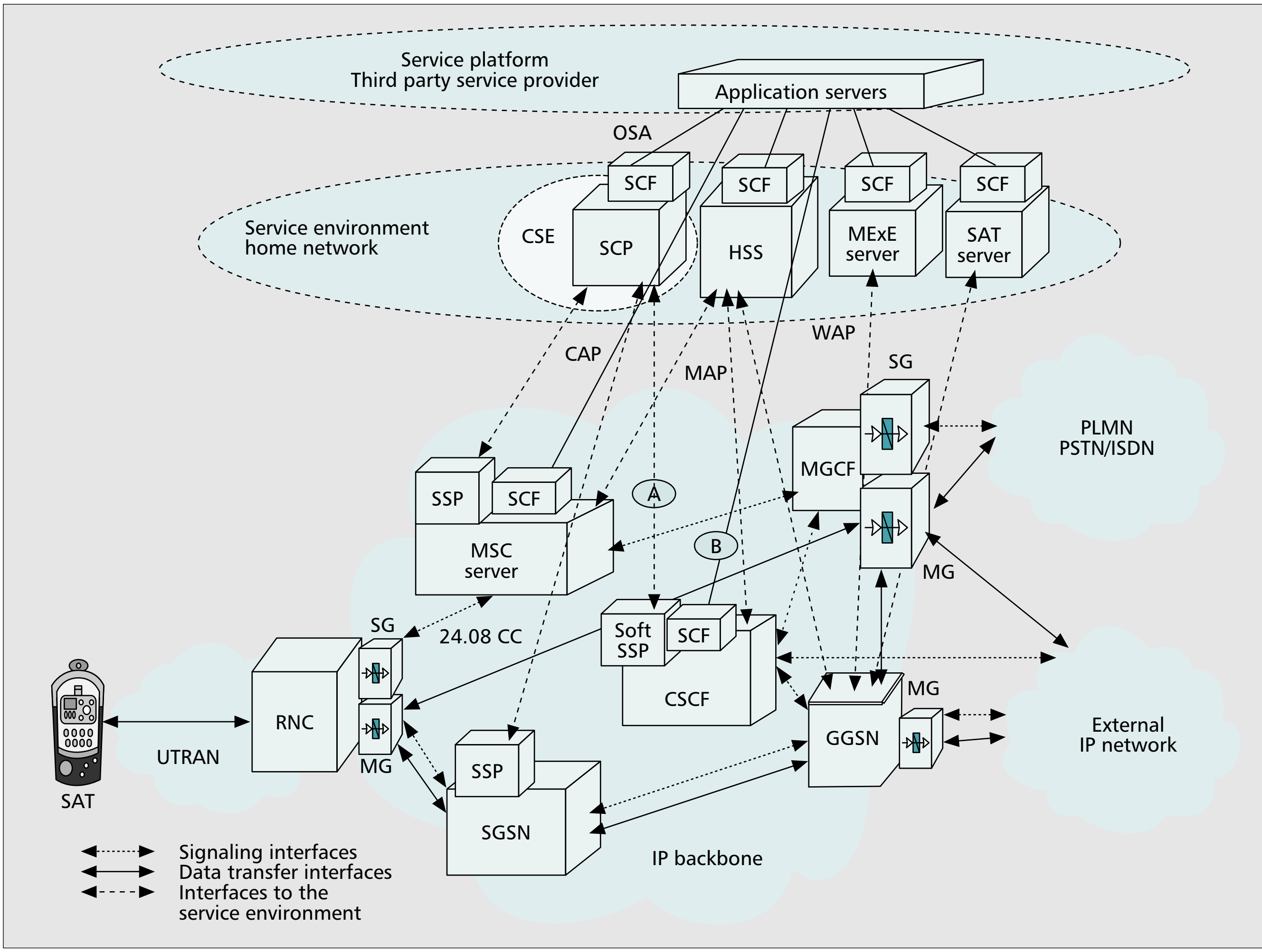


Figure 4. *Mapping of SCFs to the Release 2000 network architecture.*

serious drawback of this approach since it slows down the introduction of new VoIP/MMoIP services to the speed of CAMEL standardization.

Scenario B: Direct "Third Party Call Control"; OSA Support of VoIP (via CGI/CPL or SIP)

SIP allows for new services to be defined through a few powerful third-party call control mechanisms. There are two mechanisms, other than SIP itself, that allow a third party to instruct a network entity to create and terminate calls to other network entities: Common Gateway Interface (CGI) or Call Processing Language (CPL).

Both CGI and CPL are based on the separation of the service logic from the SIP server (Fig. 5), a concept already used in the IN world. This separation enables rapid development of new services and opens ways for third party service providers. SIP's textual approach makes it easy to write CGIs and use text-processing languages such as Perl. Both CGI and CPL are needed to provide a complete service solution [11]. The CGIs are intended for trusted users (e.g., administrators), giving a flexible general-purpose solution; CPL, which gives more limited access to the network, is needed for untrusted users (e.g., subscribers and third parties). If the service logic resides on separate servers, a specific interface, the OSA interface in the context of VHE, should be defined between the CSCF and the application server running the service logic. Many servers, each running specific service logic, can be connected to each other via a distributed service platform such as Common Object Request Broker Architecture (CORBA).

CGI: CGI is a mechanism already used on the Internet for creating dynamic Web pages in an easy way. In SIP the CGIs will be triggered when the first request arrives at the server.

CPL: The CPL script-language allows users to upload their CPL scripts to network servers. After reading and verifying the script, the service is immediately instantiated. When the controlled party executes the instructions, status messages are passed back to the CPL controller. This allows the CPL controller to take further actions based on some local program execution, much like IN. Services are based on simple standardized mechanisms.

Safe and reliable execution of third party applications such as CGIs/CPL scripts in an operator's network puts some extra requirements on the OSA architecture that will support third party service control [11]:

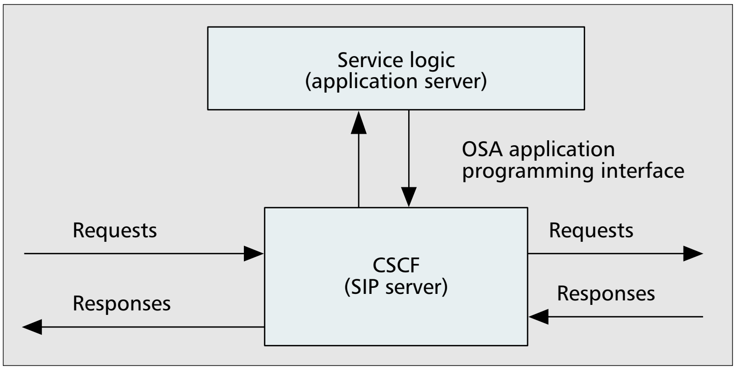
- Standardized representation: A standardized way for creating services should be defined in order to facilitate multivendor implementations. This requirement can be fulfilled by standardizing OSA interfaces on top of the CSCF (SIP server).
- Portability: Messages and service abstraction should be defined at a high level, not SIP-specific, to allow portability across different signaling protocols. This requirement is fulfilled by defining high-level service capability features specified by the OSA interfaces, independent of the underlying protocols that implement them.
- Verifiability: It must be possible to check that the script is well formed and can be executed successfully.

- Completion: Once a service is initiated it must be sure it can be terminated.
- Safety of execution: The service should not be able to initiate unsafe actions, such as modifying the data of other users.

The last three requirements are fulfilled by incorporating specific security, authentication, and verification mechanisms in the OSA interface definition.

The scenario of third party call control (B in Fig. 4), which does not have centralized SCP control of both CAMEL and VoIP services, is very interesting for third party service providers and new UMTS operators. This last category consists of newcomers that recently won a UMTS license; they do not own a legacy circuit-switched GSM or R99 network (or a CAMEL platform), but directly adopt the UMTS R00 all-IP architecture. Using CPL/CGI or SIP, service logic can be downloaded and controlled directly in the operator's SIP server by third party application servers.

In this scenario, VoIP/MMoIP services are created, provisioned, and managed completely independent of the classical CAMEL services, which are still controlled via the SCP. The OSA interfaces on the SCP are used for third party control of legacy CAMEL services. The OSA interfaces on the CSCF itself allow third party service providers to control VoIP services directly via the CSCF in the operator's network. An advantage of an OSA interface directly on the CSCF is that the deployment of VoIP services does not depend on the evolution of future releases of the CAMEL capability sets.



■ Figure 5. *The CGI/CPL services model.*

Toward a Fully Integrated All-IP Service Architecture

Two possible scenarios were explained above that can be used to provide VoIP/MMoIP services in the UMTS all-IP architecture; OSA interfaces on top of the operator's SCP or OSA interfaces directly on top of the CSCF. In this section we will explore in more detail the advantages and drawbacks of these two competing scenarios. To conclude, we evaluate which architecture would finally best suit an operator that wants to provide its customers an integrated package of both legacy and new VoIP/MMoIP services.

The left column of Table 1 investigates the scenario where both legacy services and new VoIP/MMoIP services are provided using only a CAP interface on the CSCF, and the right column of Table 1 explores a scenario where only OSA interfaces are available on top of the CSCF.

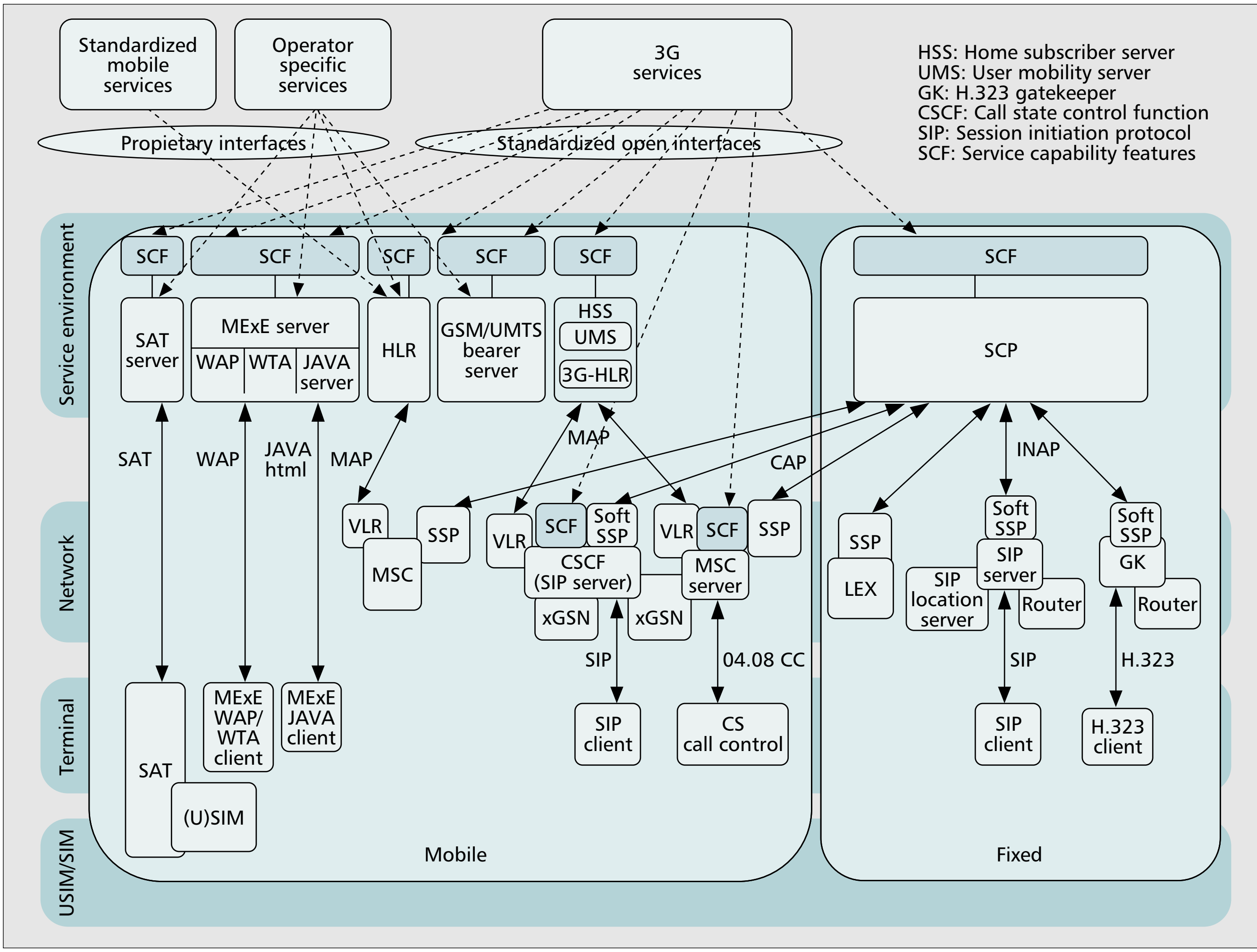
As can be seen from the table, both scenarios have their merits. To support legacy CAMEL services, clearly CAP interfaces are needed on top of the CSCF. On the other hand, creation of new VoIP/MMoIP services would benefit from having OSA interfaces directly on the CSCF. Therefore, both types of interfaces will coexist, and the solution will lie in the optimal selection of the type of interface most suitable to provide that particular service.

According to the concept of VHE, users' access to their personalized set of services depends on the capabilities supported by the terminal and networks involved in service delivery. For a roaming user, it is sensible to assume that there is a difference in the capabilities supported by the home network and the visited network. In such a case, the home network should compare the differences in the supported capabilities of the home and visited networks. Based on this comparison, the home network should make the selection of the most suitable environment and/or interfaces to be used for service delivery.

For example, if the service requested by the roaming user is a legacy service (e.g., prepaid) which can be perfectly supported by CAMEL and the visited network supports the necessary version of CAMEL, the home network may decide to leave call control to the visited network. In the case of a third party service provider, the service logic, which can be provided by an OSA interface on top of the CAMEL SCP in the home network, communicates with the call control in the visited network by means of the standardized CAP protocol between the

Only CAP interfaces on CSCF		Only OSA interfaces on CSCF	
+	–	+	–
Maximum reuse of existing CAP standards	Development of new VoIP/MMoIP services depends on capabilities supported by future CAP versions	Development of new VoIP/MMoIP services does not depend on capabilities supported by future CAP versions	Little reuse of existing CAP standards
No need to standardize new OSA interface on top of CSCF			Need to standardize new OSA interface on top of CSCF
Seamless support of 2G legacy services in 3G (reuse of CAMEL service logic possible)	New service logic for VoIP/MMoIP services has to be developed	Easy service creation mechanisms (CGI/CPL) exist for developing new VoIP/MMoIP services	Would require service logic for legacy services to be redeveloped using CGI/CPL
Integrated billing and management for legacy and new VoIP/MMoIP services is done via well-known CAMEL mechanisms	Need to develop new functional entity called "SoftSSF"	No need to develop new functional entity called "SoftSSF"	Integrated billing and management for legacy and new VoIP/MMoIP services is done via new OSA mechanisms (extra security level necessary)

■ Table 1. *Advantages and drwabacks of the two service provisioning scenarios.*



■ Figure 6. *UMTS service architecture.*

home and visited networks. If, on the other hand, the service requested by the roaming user is a new VoIP/MMoIP service (e.g., multimedia conferencing) , which cannot be supported by the CAMEL capabilities of the visited network, the home network may decide to handle call control in the home network. In the case of a third party service provider, the service logic, which can be provided by an OSA interface on top of the CSCF in the home network, communicates with the call control in the home network by means of an OSA interface directly on top of the CSCF in the home network.

From the previous analysis we can conclude the following. An operator that wants to provide its customers an integrated package of both legacy and new VoIP/MMoIP services needs an architecture that allows him to flexibly switch between both mechanisms; service control via CAP as well as service control via OSA interfaces directly on top of network elements. To conclude, Fig. 6 presents an overview of such a "fully integrated" UMTS service architecture for the provisioning of both legacy CAMEL and new VoIP/MMoIP services in line with the principles of the VHE. The top left corner of Fig. 6 illustrates how in 2G mobile systems, services — standardized or operator-specific — were created and operated using proprietary interfaces toward network elements. The middle of Fig. 6 shows how the third-generation UMTS service architecture promotes the provisioning of 3G services through open standardized interfaces between network and applications by standardizing service capability features provided by underlying network servers. Finally, Fig. 6 clearly demonstrates how both scenarios of OSA interfaces on top of SCP and OSA interfaces directly on top of the CSCF, can coexist to ensure optimal service delivery according to the principles of VHE. Legacy services (e.g., prepaid) are provided by reusing CAMEL, while new VoIP/MMoIP services (e.g., multimedia conferencing) via the new OSA interfaces directly on top of the CSCF. Remark also that the mix of CAP/INAP and OSA interfaces allows operators and third party service providers to offer combined services to a user that has both a mobile and fixed subscription; for example, "if I am not reachable on my fixed phone, try my mobile phone."

Conclusion

UMTS is seen as the enabler of wireless multimedia applications and portability of a personalized service set across network and terminal boundaries, as defined within the virtual home environment system concept. This article investigates the evolution toward an all-IP UMTS system architecture, clarifying the impact of an IP-based core network design on the UMTS service architecture. Two possible scenarios are discussed for supporting VoIP services in a UMTS service architecture based on the principles of the VHE. On one hand, there are the classical centralized IN-type service control architectures, which remain very important for continuing support of legacy IN services. On the other hand, there is the new decentralized OSA-type service provisioning architecture which appears to be very interesting for flexible deployment of future innovative multimedia services. Because each of the two scenarios has its merits in certain situations, both types of interfaces will coexist. and the solution will lie in the optimal selection of the type of interface most suitable to provide a particular service.

Acknowledgments

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Additional Reading

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