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*Published in:*

Sixteenth Nordic Teletraffic Seminar NTS 16 : Helsinki University of Technology, Espoo August 21-23, 2002 : proceedings (Report / Helsinki University of Technology, Networking Laboratory, ISSN 1458-0322)

2002

[Link to publication](#)

*Citation for published version (APA):*

Andersson, J. K., Aprin, S., & Kihl, M. (2002). Convergence-analysis of the internet and the telecommunication architectures. In P. Lassila, E. Nyberg, & J. Virtamo (Eds.), *Sixteenth Nordic Teletraffic Seminar NTS 16 : Helsinki University of Technology, Espoo August 21-23, 2002 : proceedings (Report / Helsinki University of Technology, Networking Laboratory, ISSN 1458-0322)* Helsinki University of Technology, Networking Laboratory.

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This is an author produced version of a paper presented at  
Sixteenth Nordic Teletraffic Seminar NTS 16 : Helsinki University of  
Technology, Espoo August 21-23, 2002.

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Citation for the published paper:

Andersson, Jens, Aprin, Shabnam, Kihl, Maria, 2002,

"Convergence-analysis of the internet  
and the telecommunication architectures",

*Sixteenth Nordic Teletraffic Seminar NTS 16 : Helsinki University of  
Technology, Espoo August 21-23, 2002 : proceedings (Report / Helsinki  
University of Technology, Networking Laboratory, ISSN 1458-0322).*

Publisher: Helsinki University of Technology, Networking Laboratory.

# Convergence-Analysis of the Internet and the Telecommunication Architectures

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**Abstract.** The convergence of the Internet and Telecommunication Architectures is a key issue in today's telecommunication world. It is foreseen that future versions of UMTS will be all-IP based, and therefore the interest in this area is increasing. The focus in this paper is to show how to access IN-services from the Internet and other IP-based networks. Previous research of the interworking between PSTN and IP-networks will be discussed. The respective advantages of different architectures are presented and some solutions of building a bridge between the protocols used in these networks are described. Further, this paper presents some of the performance problems that may occur in such systems.

## 1. Introduction

Today, both mobile networks and the Internet are widely used and the applications offered are increasing. The present versions of UMTS are based on the GSM system. When a UMTS/GSM user makes a call the user gets connected to a base station. Via some additional steps it is then possible to connect to the PSTN. This is the reason why we today have access to regular PSTN-services from mobile phones. Often, as an additional feature, the mobile operators are able to offer some additional services which are not reachable from the PSTN or from other operators network.

However, future versions of UMTS are foreseen to be all-IP based, which means that telephony and Internet will become converged. This, together with the fact that there is an increasing demand for new services, has lead to an evolution towards an open architecture in which you can access the different networks independent of where you are located. This would also open up the possibilities for the third party to create new services.

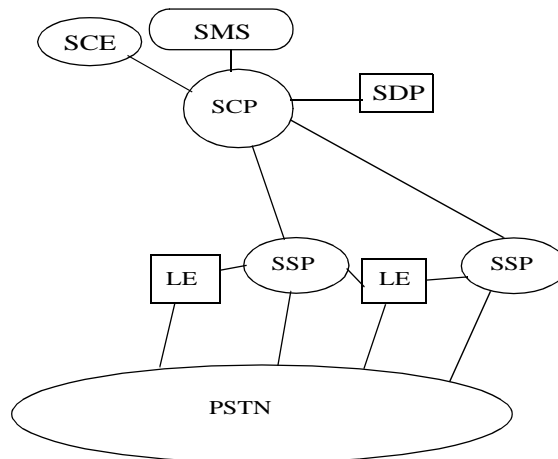
Several architectures have been proposed by different organisations. *Parlay*, developed by the Parlay group [13], is aimed at converging the Internet with the Intelligent Network (IN) in PSTN. By using the Application Protocol Interfaces (APIs) defined in Parlay, IN services may be accessed and controlled from the Internet. *Open Service Access* (OSA) is an architecture developed by the Third Generation Partnership Project (3GPP) [3]. It is based on Parlay, but is developed especially for 3G-networks. The purpose of OSA is to converge all types of communication networks.

The purpose with this article is to clarify the present proposals of making different networks communicate with each other, with the main aim to examine the possibilities of accessing the IN-services from future versions of UMTS. Section 2 explains the Intelligent Network (IN) service architecture and how we access the IN-services from the PSTN. In section 3, the architecture of UMTS and the foreseen evolution is introduced. Some of the solutions of how to access IN-services from an IP-network are introduced in section 4. Parlay is described in section 5 and OSA in section 6. An example of an OSA application is described in section 7. Section 8 contains a short discussion about some performance aspects of these service architectures. Finally, section 9 provides some conclusions.

## 2. The Intelligent Network Service Architecture

The objective of the Intelligent Network (IN) is to provide value-added services to the users in a PSTN. The Intelligent Network (IN) is built on the simple idea of separating the service logic from the switching nodes. In this way the operators are offered a much more scalable service platform. Earlier, the service logic was implemented directly into the network switches and each time a service was created all the switches had to be updated.

The network architecture is shown in figure 1. The nodes hosting the services are called Service Control Points (SCPs) and the switching nodes dealing with the switching and the call control are called Service Switching Points (SSPs). The Service Management System (SMS) contains functions to enable management of the IN system. Another important entity in the IN architecture is the Service Creation Environment (SCE) that contains service development tools which the operator uses when creating new services. If a service needs to work with any kind of data there is a Service Data Point (SDP) which is a database that holds information related to the services.



**Figure 1.** Overview of the Intelligent Network Service Architecture

Typical IN services are local number portability, credit card calling, toll-free numbers and so forth. When any of the supported services are requested, the local exchange (LE) connected to the user recognizes this. The LE contacts the nearest SSP which opens up a dialogue

with the SCP that executes service logic corresponding to the requested service. The communication between different nodes such as SCP and SSP is performed via the Common Channel Signalling System No. 7 (SS7) which is a global standard defined by the ITU-T (see [1] for more details). The SS7 protocol stack contains a couple of different protocols with different advantages. In cases when IN service control is needed, the SSP can trigger the SCP by use of the IN Application Part (INAP) protocol. When an SSP communicates with other SSPs they use the ISDN User Part (ISUP).

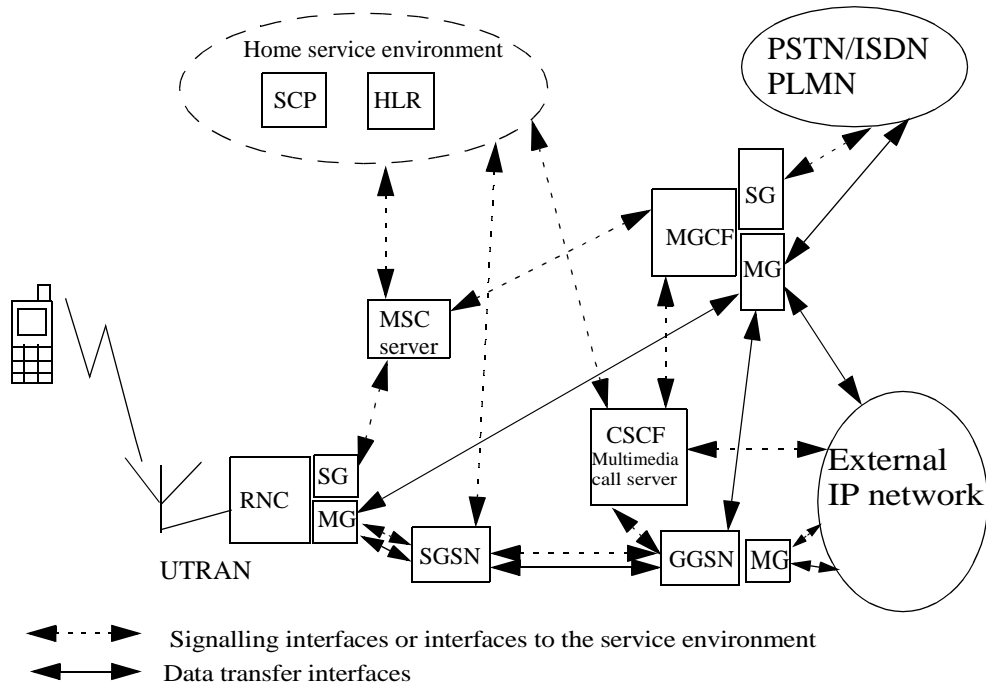
A mobile version of IN has also been designed. It is called Customized Applications for Mobile network Enhanced Logic (CAMEL). The equivalent to INAP in IN is CAMEL Application Part (CAP) in CAMEL. E Geulen and J Hartmann [10] has described the architecture and signalling a bit deeper.

### 3. UMTS

The evolution of the mobile telephony has accelerated for the last years. Today the 3rd party is no longer pleased by only using the phone for calling. The demands are focused to communication requiring much more bandwidth than provided by GSM. This has led to the evolution of UMTS. UMTS exists in several different shapes. The global wireless industry has created two big partnership projects, namely 3rd Generation Project Partnership (3GPP) and 3GPP2. So far these groups are the sources of the UMTS releases. The first release of UMTS resembles the GSM/GPRS system, only differing in the first radio access network stage where Universal Terrestrial Radio Access Network (UTRAN) was introduced (see Terjesen [21]). UTRAN offers improvement of the bandwidth and the number of possible simultaneous users. Since 1999, when 3GPP released the R99 standardisation work, the trend of the evolution of UMTS has been towards an all-IP based architecture. The next release R00 that 3GPP produced introduced a lot of new concepts and the architecture could not be compared to the precursors.

Figure 2 shows a simplified picture of the R00 architecture (see Bos and Leroy [6]). The architectures of the following releases (release 4 and 5) are almost the same. In R00 the *MSC server* is one of the new components. It controls all calls from a PSTN/ISDN/GSM network and is a component to support the compatibility requirements with earlier versions. Due to the MSC in earlier specifications, the MSC server only takes care of the call control and service part, while the switch is replaced with a *Media Gateway (MG)*. The MGs have different tasks depending on where they are placed in the network. The MG at the UTRAN side transforms VoIP packets into radio frames and the MG at the PSTN side translates all calls coming from PSTN into VoIP calls for transport in UMTS core network. The control of the MGs is managed by the *Media Gateway Control Function (MGCF)* via the H.248 protocol (often referred to as MEGACO). The MGCF also performs translation at the call control signalling level between ISUP signalling, used in the PSTN and SIP signalling used in the UMTS multimedia domain. The *Call State Control Function (CSCF)* is a SIP server that provides/controls multimedia services for packet switched (IP) terminals. The *Signalling Gateway (SG)* just relays all the call-related signalling between UTRAN and PSTN on an IP bearer and sends the signalling data to the MGCF.

However, it implies some difficulties to integrate an all-IP based system with the old PSTN/IN. For example in PSTN the signalling is terminated, examined, sometimes operated and finally reinstated at each intermediate node. This is a fundamental difference from Internet



**Figure 2.** A simplified all-IP release 2000 architecture.

where the execution takes place at the destination application. If the all-IP network is introduced it will, during the first period, keep the capabilities to serve 3G mobile terminals that are circuit-switched. This is because of the backward compatibility requirement [6].

Despite of the difficulties with an all-IP based architecture there are many reasons why the evolution, however, has pointed towards an all-IP based architecture. If the QoS can be arranged, the IP telephony may replace the PSTN. This because of better utilization of network, lower equipment cost and less management of the network, which in turn might lead to lower communication costs for the subscribers. It will also become an improvement to only have one connection point to a global network which take care of all of the communication.

## 4. Accessing IN-services from IP-networks

There are mainly two competing architectures for accessing IN-services from an IP-network. The first type of architecture uses a Soft-Switch that translates the protocols in order to communicate with an SCP. The second type uses APIs, for example Parlay, JAIN or OSA.

### 4.1 VoIP

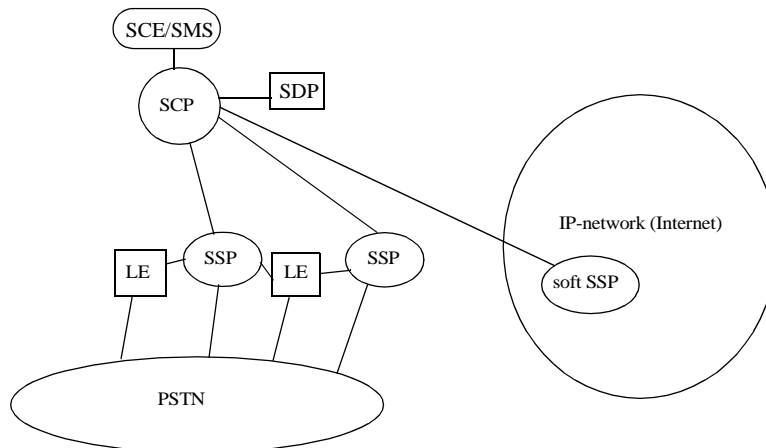
The purpose of the Voice over IP (VoIP) protocols is to enable telephony services over the Internet. Since long, much work has been performed on how to access IN-services from an

VoIP network. Thus, there are a lot of documentation and many ideas of accessing the IN-services from the VoIP protocols SIP and H.323.

In 1997 a working group named PSTN/Internet Interworking (PINT) was created with the aim to investigate how Internet applications can request and enrich telecommunication services. To reach the IN from an IP network a PINT client forwards a service request to a PINT server. The PINT server then relays the request to a relevant node in a correct way so that the destination can adopt it. Probably the destination node is a SCP that executes the requested service. However, PINT has not been as successful as foreseen. Brennan *et al.* [9] discuss some limitation of PINT that might have been conducive to this. However, PINT is only one of many solutions that have been presented.

The most common solution is a so called Soft-Switch (soft SSP) which acts like a gateway, communicating with the SCP in the IN, see figure 3. This Soft-Switch must appear just like an ordinary SSP for the SCP although it is using IP based protocols instead of SS7. The service requests arriving at a Soft-Switch may be coded using higher layer IN-protocols, like INAP or TCAP, but one or more of the lower layers will be replaced with TCP/IP (see Ciang *et al.* [4]).

Wang *et al.* [7] propose a Soft-Switch that can access IN-services from a SIP network, denoted an Intelligent Services Gateway (ISG). Dagiuklas and Galitos [20] have a proposal of how to access from an H.323 network. Chapron and Chatras [16] investigate how to map INAP signalling from an SCF to SIP and H.323 signalling. Wang and Hao *et al.* [8] present a good overview of which protocols that are used when two entities are communicating both within an IN and within an IP-network, but also between an IN and an IP-network.



**Figure 3.** Accessing IN from an IP-network using a soft switching point.

## 5. Parlay

The Parlay group [13] is a collaboratorium that were founded by British Telecom, Microsoft, Nortel Networks, Siemens and DGM&S Telecom in 1998. This group has focused on the creation of the Parlay specification which is an alternative to the Intelligent Network (IN). The specification specifies a set of standard application interfaces (APIs), that will enable applications residing outside the network to access and control a lot of resources and devices.

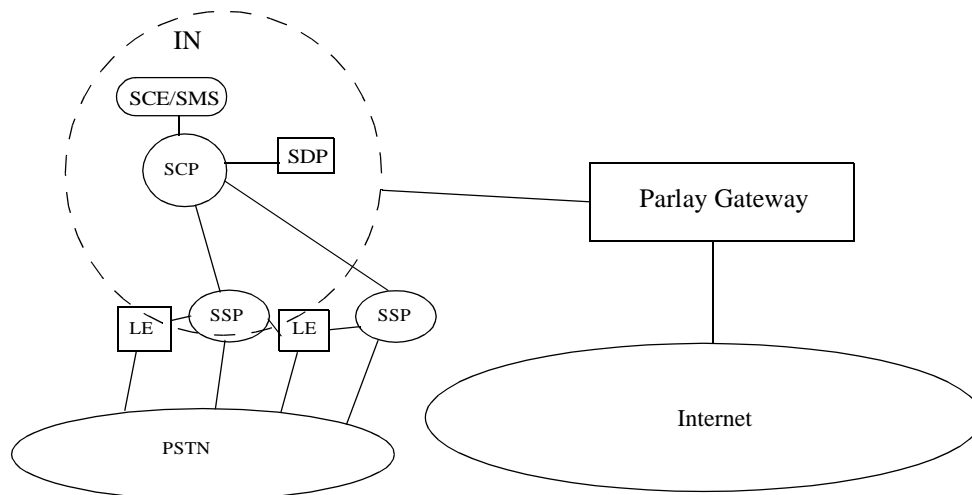
## 5.1 Advantages

The major advantage and one of the reasons why the Parlay project once was created is the ease of creating new services (see C-A Licciardi et al. [19]). The Parlay API specifications are open and technology-independent, so that a wider range of Independent Software Vendors (ISVs) may develop and offer advanced telecommunication services. An API is intended to be simple in order to be applied in all different kinds of networks and can be used from 3rd party application developers. In IN the operator is responsible for the creation and operation of all the applications. Further, Parlay offers better opportunities to test the software before it gets deployed.

## 5.2 Architecture

The Parlay APIs consist of two categories of interfaces, *Service* and *Framework*. The Service interfaces are the interfaces that offer applications access to network capabilities such as call control and messaging. The Framework interfaces are the interfaces that takes care of the security and manageability. The design and implementation of the API and applications using this API supports a wide range of existing distribution middleware such as CORBA, DCOM and JAVA/RMI.

Figure 4 shows the architecture of a system where the PARLAY gateway enables access to the IN services from Internet. The difference between accessing an IN service from a soft SSP (fig. 3) and an open API (fig. 4) is that in the case of a soft SSP it is pretended that the soft SSP is inside the IN. In the case with PARLAY the networks are both connected to the PARLAY gateway and the gateway offers the ability to use services from the other networks.



**Figure 4.** Accessing IN-services from the Internet using a Parlay Gateway.



## 6. Open Service Access (OSA)

From the beginning the OSA was an acronym for Open Service Architecture, but it has been re-termed to Open Service Access. OSA is based on the concept of Parlay and is developed by the 3GPP. OSA differs from Parlay in a way to better satisfy the needs for a 3G network. This means that the concept is the same but some modifications are made.

### 6.1 Advantages

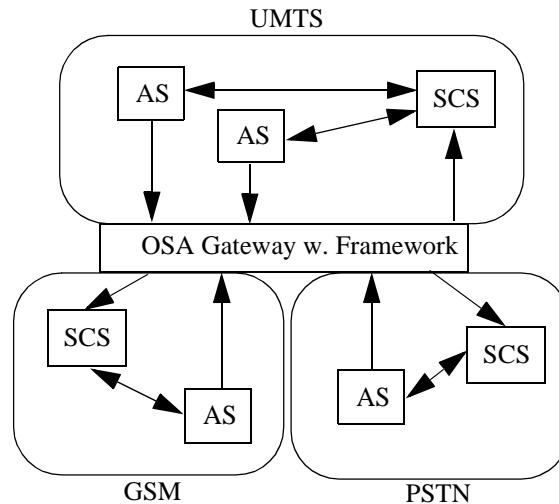
One of the reason why OSA has been evolved is the opportunities seen in the area of services and wireless Personal Digital Assistants (PDAs). It is foreseen that there will be a great demand for services, and to respond to this demand the pace of the development has to speed up. Therefore the OSA has been proposed, in order to make it easier developing and testing new services outside the telecom domain. The number of feasible service providers has increased because of the fact that OSA APIs offers the security and integrity that the operators need in order to open up their networks to independent software developers and service providers (see Lundqvist [2]).

### 6.2 Architecture

Figure 5 shows the general OSA architecture. Each network connected to the OSA gateway enables their users to take part of the applications and services from other networks as well as other networks are enabled to use their applications and services. OSA consists of three parts, namely the *Applications*, the *Framework* and the *Service Capability Servers* (SCSs). Applications are for example Virtual Private Networks (VPNs), conferencing and location based applications. The applications can be executed on one or more external application servers (called AS in the figure). The applications use so called Service Capability Features (SCFs) provided by the Framework and the SCSs.

The Framework provides the applications with basic mechanisms, like authentication before accessing the network functionalities, or discovery of the capabilities needed by an application. Moerdijk and Klostermann [17] describe how the framework is used during the registration phase. The Framework SCFs include Trust & Security Manager, Discovery and Integrity Management SCFs. It is due to these SCFs that OSA can offer the security that is required by the operators. The Framework must be implemented in a server. We will assume that there is an OSA gateway in which the Framework is implemented. The gateway acts a mediation device, offering a uniform and technology independent interface. Another solution is to have one OSA gateway in each network. The gateways will then provide the communication between the networks.

The SCSs, implemented in the underlying networks, provide the applications with all the functionality, called Network SCFs, required to construct services. There may be one or more SCSs in each network. The SCSs communicate with the network entities (HLRs, MSCs, SCPs etc.) needed to perform a specific SCF. In the all-IP based UMTS architecture (figure 2) this means that both the MSC server and the CSCF must communicate with an SCS. The Network SCFs includes the Call Control, User Interaction, Mobility, Terminal Capabilities, Data Session Control, Generic Messaging, Connectivity Manager, Account Management and Charging



**Figure 5.** The OSA architecture.

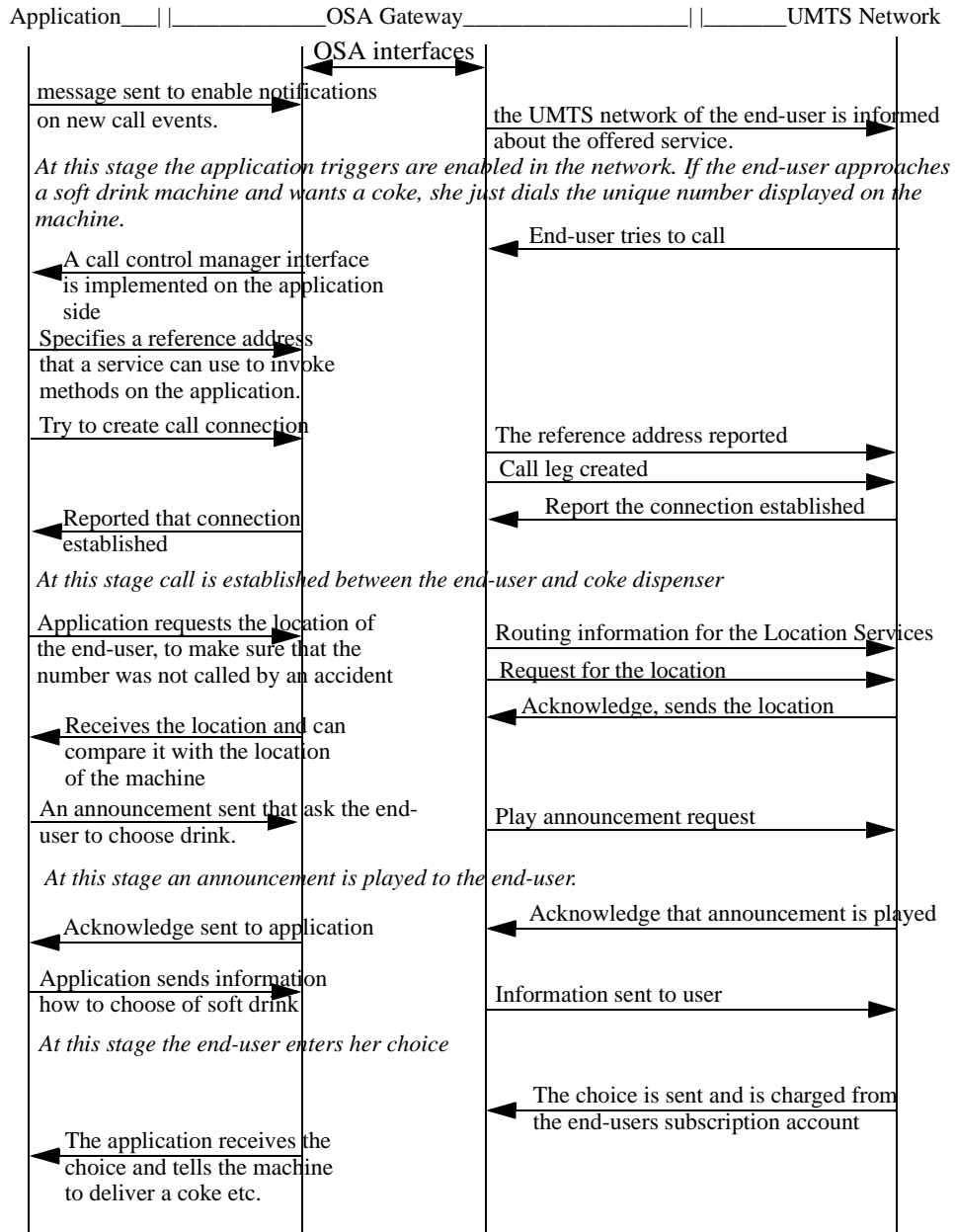
SCFs. These different SCFs is the network capabilities that the service providers can use as building blocks to compose and create new OSA applications. The applications developed can then be executed on several networks and terminals thanks to OSA, that have standardized application interfaces towards these networks.

Just like in Parlay, OSA uses an object-oriented technique, like CORBA, which makes it possible to use different operating systems and programming languages in application servers and service capability servers. An SCF corresponds to an object class with some functions. For example, Call Control can be seen as a class consisting of functions like “create call leg”, “delete call leg”, “connect call legs” etc.

3GPP has also introduced an interesting concept called Virtual Home Environment (VHE) [11]. The idea of VHE is to consistently present the same personalised features, User Interface customization and services to the user, independent of which terminal or network that is present for the moment. L Moretti and R Depaoli [15] investigates the support of VHE by means of OSA.

## 7. An OSA service example

In [5] there is an example of the messaging sequence for an application where the end-user can pay for her purchase using the prepaid balance on her mobile subscription account. Here follows the same example with the simplification of not showing the signalling between different interfaces inside the OSA gateway. Of course there are additional signalling at the application side and the UMTS Network side too, but it is only the amount of signalling through the gateway that is of interest to us. More about most of the messages and their arguments can be read with help of [12] and [14]. The sequence diagram can be seen in figure 6 which contains short descriptions of each signal. As seen there is a lot of signalling even for simple service as just described.



**Figure 6.** Message sequence diagram of a soft drink machine.

## 8. Performance issues

One important issue when developing a new service architecture is how to ensure a good QoS for the users. When a user wants to run an application, it is necessary that the delays are short. Therefore, it is necessary that the potential performance bottlenecks are identified before the architecture is deployed.

When studying Parlay and OSA, one potential bottleneck is easily identified, namely the Gateway. Since the Gateway is a centralized service control point, it is sensitive to overload. In the Integrity Management SCF, that is part of the Framework, a Load Manager is specified that may detect and control overload. However, this mechanism has not been further described in the specification. Therefore, overload control mechanisms should be developed in order to maintain a good performance during all traffic situations.

Further, the gateway will provide many types of services. These services may have different priorities that must be taken into account. The services may have so called QoS contracts, that must be kept. High priority services, for example those that generate profits for the service provider, should not be blocked in cases of overload.

Other potential bottlenecks are the SCSs in OSA. These servers act as an SCP in the IN, which means that they too are sensitive to overload. Since each SCS will provide access to SCFs situated on other network entities, distributed processing may cause performance problems.

So far, there has only been one published paper about performance analysis of the service architectures described above. Melen *et al.* [18] use a simple queuing network model to investigate a Parlay gateway supporting multiple services. They develop a scheduling mechanism for the gateway that ensures that each service obtains processing capacity that is proportional to the offered load for that service.

## 9. Conclusions

The 3rd generation mobile network (3G) is foreseen to be all-IP based, which means that telephony and Internet will become converged. Also, the demand for new advanced services is increasing.

Therefore, new service architectures must be developed. Several architectures have been proposed by different organisations. In this article we have described Parlay and OSA, the two most promising service architectures.

Since both Parlay and OSA are based on distributed processing and open APIs, several performance problems may be identified. Therefore, it will be necessary to develop performance models for these architectures and then investigate how they behave during various traffic scenarios.

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