Session Initiation Protocol in 3G

Tuomo Sipilä
Assistant Research Manager
Nokia Research Center
tuomo.sipila@nokia.com

Abstract

This article gives an overview of the 3GPP IP Multimedia subsystem and how SIP protocol is used with it. Also the 3G required changes to SIP are identified. In addition some key problems with SIP protocol are identified with regards to 3G mobile networks.

1   Introduction

3rd Generation Partnership Project started the development of IP based multimedia services to the 3rd generation mobile networks, known as UMTS in Europe and IMT-2000 in Japan, during autumn 1999. The initiation and pressure for the work came through an organisation called 3G.IP which is a group formed to push IP based ideas to 3G networks. The target was to standardise the required enhancements for the 3G network so that IP telephony and multimedia can be provided with equal user perceived quality as with the current mobile network services. Another requirement was that the future 3G network could function fully based on packet and IP connections without the traditional circuit switched domain. Essentially also the IP multimedia would in the future provide via IP a wider and more flexible service set than the current networks. In Spring 2000 IETF defined Session Initiation Protocol (SIP) was selected as the base protocol that shall provide the IP Multimedia sessions to the mobile terminals (UE). This decision tied the 3GPP solution and work into co-operation with IETF.

   Already during year 2000 it became evident that the specification work would take longer that expected since it requires specification of a completely new network subsystem with all required mobile functions. Therefore the specification target was set to the end of 2001 when the 3GPP Release 5 specifications should be finalised.

2   3GPP Rel5 system architecture

The 3GPP Release 5 architecture is illustrated in the figure 1.
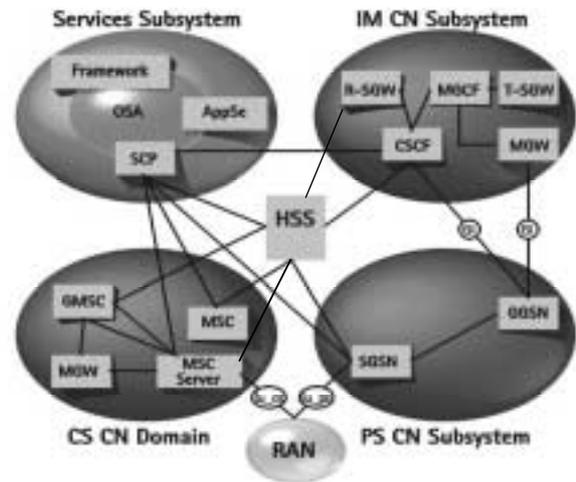
Figure 1: 3GPP Release 5 network architecture and domains (not all elements are shown)

The 3GPP mobile system consists of the following network domains:

-   **Radio Access Network Domain (RAN)** consists of the physical entities, which manage the resources of the radio access network, and provides the user with a mechanism to access the core network. It can be either WCDMA based UTRAN or GSM/EDGE based GERAN
-   **Circuit Switched Core Network Domain (CS CN)** comprises all core network elements for provision of Circuit switched services
-   **Packet Switched Core Network Subsystem (PS CN)** comprises all core network elements for provision of PS connectivity services i.e. guaranteeing the IP flow to and from the mobile terminal (UE)
-   **IP Multimedia Core Network Subsystem (IMSS)** contains all the network elements that are used to provide the IP base multimedia services
-   **Service Subsystem** Comprises all elements providing capabilities to support operator specific services (e.g. IN and OSA)

One of the fundamental principles of the 3GPP network architecture is to maintain to large extent an independence of the network domains and subsystem so that independent evolution is possible. This means that the IP multimedia is seen to large extent as an transparent service through the Radio Access and Packet Core Network domains.

IP Multimedia Core Network Subsystem (IMSS) and its functionality is the main focus of this article. For making the system functionality more understandable the PS CN subsystem functionality is briefly illustrated. Also the IMSS linkage with the service subsystem that provides the open service generation is briefly mentioned.

3 PS CN Subsystem

3.1 Architecture

The PS CN subsystem main functions are to establish and maintain the connection between the terminal and the GGSN, route the IP packets in both directions and do charging. The Packet Switched Core Network subsystem consists of the following GPRS based network elements and functions [4]:

- **Serving GPRS Support Node** which maintains the subscription data (identities and addresses) and follows the location of the terminal within the network
- **Gateway GPRS Support Node** which maintains the subscription information, allocated IP addresses and follows the SGSN under which the terminal is.

The PS CN subsystem in connected to the IMSS via Go and Gi interfaces that are located in the GGSN. The Gi interface is the one that is also used for standard Internet access and it is relatively transparent. The Go interface is used for policy control between IM Subsystem and GGSN and packet core. The reasons for policy control is to allow the operators to limit the utilisation of the best 3G packet QoS classes to their own IP Multimedia services.

The IP connections between terminal (UE) and GGSN are provided by PDP contexts . At PDP context establishment the used QoS profile and terminal IP address are allocated.

3.2 SIP related procedures

The PS CN Subsystem is strongly linked with the IP Multimedia since it shall provide the bearer through radio and the packet core network for the IP Multimedia Signalling (SIP) and also for the IP media streams. Thus co-operation is required on some level. To Packet Core network PDP context activation procedures have been defined for transportation of SIP signalling traffic. This is provided by having a separated signalling PDP context or using a generic PDP context for SIP transfer. For incoming calls either the PDP context is considered to be active constantly or it can established by the network [1]. In addition the establishment of the secondary PDP context for the session data flow has to be synchronised with the SIP signalling negotiation and callee alerting.
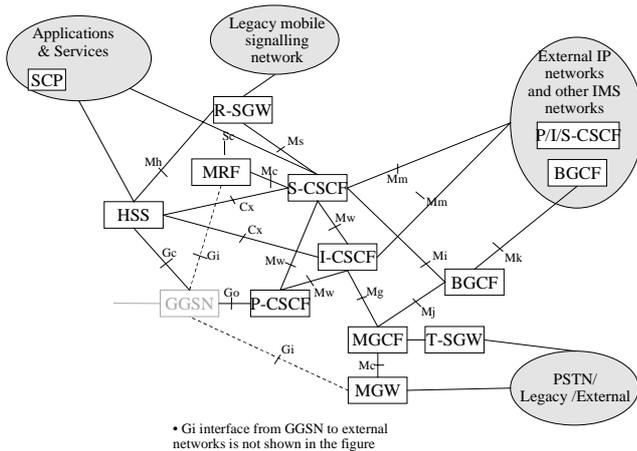
4 IP Multimedia Subsystem

4.1 Requirements

The 3GPP TS 22.228 specifies the following main service requirements for the IP Multimedia Subsystem [6]:

- at least equal end-to-end QoS for voice as in circuit switched (AMR Codec based) wireless systems
- equal privacy, security or authentication as in GPRS and circuit switched services
- QoS negotiation possibility for IP sessions and media components by both ends
- access independence
- applications shall not be standardised
- IP policy control possible
- automated roaming with the services in home and visited network
- hide the operator network topology
- identification of the entities with either SIP URL or E.164 number
- procedures for incoming and outgoing calls, emergency calls, presentation of originator identity, negotiation, accepting or rejecting incoming sessions., suspending, resuming or modifying the sessions
- the resources shall be made available before the destination alerts
- user shall have the choice to select which session components reject or accept

4.2 Architecture

The IP Multimedia subsystem current architecture showing the functions (March 2001) is in figure 2. Note that several of the illustrated functions can be merged into real network elements. The functions and their purposes are clarified in the following subsections. It should be noted that the standardisation for the system is still ongoing so changes can be expected.

- • Gi interface from GGSN to external
  networks is not shown in the figure

Figure 2: IP Multimedia Subsystem

4.3 HSS

Home Subscriber Server (HSS) is a combination of the currently existing UMTS/GSM HLR and the needed register functions for IP Multimedia Subsystem. HSS will provide the following functions:

- user identification, numbering and addressing information.
- user security information: Network access control information for authentication and authorisation
- user location information at inter-system level; HSS handles the user registration, and stores inter-system location information, etc.
- the user profile (services, service specific information…) [3]

4.4 P-CSCF

Proxy Call State Control Function (P-CSCF) is the first contact point within the IM subsystem. Its address is discovered by "Procedures related to Local CSCF Discovery ". The P-CSCF behaves like a proxy (as defined in RFC2543). The functions performed by the P-CSCF are:

- forward the SIP register request received from the UE to an I-CSCF determined using the home domain name, as provided by the UE.
- forward SIP messages received from the UE to the SIP server (e.g. S-CSCF) whose name the P-CSCF has received as a result of the registration procedure.
- as part of processing of the request and before forwarding, the P-CSCF may modify the Request URI of outgoing requests according to a set of provisioned rules defined by the network operator (e.g. Number analysis and potential modification such as translation from local to international format.)
- forward the SIP request or response to the UE.
- detect an emergency call and select a S-CSCF in the visited network to handle emergency calls.
- the generation of CDRs for charging

- Authorisation of bearer resources and QoS management and Security issues are currently open in standardisation [3].

4.5 S-CSCF

Serving Call State Control Function) (S-CSCF) performs the session control services for the endpoint. It maintains session state. Within an operator's network, different S-CSCFs may have different functionality. The functions performed by the S-CSCF during a session are:

- registration: Acts like a Registrar defined in the RFC2543, i.e. it accepts Register requests and makes its information available through the location server (e.g. HSS).
- session flows: Session control for the registered endpoint's sessions; Interaction with Services Platforms for the support of Services;
- on behalf of an originating endpoint (i.e. the originating subscriber/UE); Forward the SIP request or response to the I-CSCF
- on behalf of a destination endpoint (i.e. the terminating subscriber/UE): Forward the SIP request or response to a P-CSCF for a MT session to a home subscriber within the home network, or for a subscriber roaming within a visited network where the home network operator has chosen not to have an I-CSCF in the path; Forward the SIP request or response to an I-CSCF for a MT session for a roaming subscriber within a visited network where the home network operator has chosen to have an I-CSCF in the path.
- charging and resource utilisation: Generation of CDRs
- Security issues are currently open in standardisation [3]

4.6 I-CSCF

Interrogating Call State Control Function (I-CSCF) is the contact point within an operator's network for all connections destined to a subscriber of that network operator, or a roaming subscriber currently located within that network operator's service area. There may be multiple I-CSCFs within an operator's network. The functions performed by the I-CSCF are:

- Registration: Assigning a S-CSCF to a user performing SIP registration; Cryptography mechanisms may be used by I-CSCF to hide the S-CSCF when S-CSCF name is provided to the P-CSCF in the Visited network
- Session Flows: Route a SIP request received from another network towards the S-CSCF; Obtain from HSS the Address of the S-CSCF; Forward the SIP request or response to the S-CSCF determined by the step above;
- charging and resource utilisation: Generation of CDRs.

- in performing the above functions the operator may use I-CSCF to hide the configuration, capacity, and topology of the its network from the outside
- additional functions related to inter-operator security are for further study.

4.7 MGCF

Media Gateway Control Function (MGCF) Provides the following functions:
- protocol conversion between ISUP and SIP
- routes incoming calls to appropriate CSCF
- controls MGW resources [3]

4.8 MGW

Media Gateway (MGW) provides the following functions:
- Transcoding between PSTN and 3G voice codecs
- Termination of SCN bearer channels
- Termination of RTP streams [3]

4.9 T-SGW

Transport Signalling Gateway provides the following functions
- Maps call related signalling from/to PSTN/PLMN on an IP bearer
- Provides PSTN/PLMN <-> IP transport level address mapping [3]

4.10 MRF

Multimedia Resource Function provides the following functions:
- Performs multiparty call and multimedia conferencing functions [3]

4.11 BGCF

The S-CSCF, possibly in conjunction with an application server, shall determine that the session should be forwarded to the PSTN. The S-CSCF will forward the Invite information flow to the Breakout Gateway control function (BGCF) in the same network.
The BGCF selects the network in which the interworking should occur based on local policy. If the BGCF determines that the interworking should occur in the same network, then the BGCF selects the MGCF which will perform the interworking, otherwise the BGCF forward the invite information flow to the BGCF in the selected network. The MGCF will perform the interworking to the PSTN and control the MGW for the media conversions
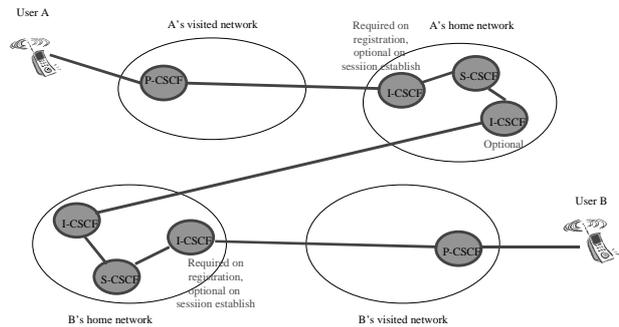
5 IMSS model in the case of internetwork roaming

Figure 3: Call model in roaming case [5]

Figure 3 shows the call model in mobile to mobile call case when both callee and caller are roaming. In the roaming case i.e. when a user roams to network that is outside his home network the IP multimedia services are provided by the S-CSCF in the home network. The P-CSCF in the visited network forwards the service request to the home network. However in some cases some services can be provided directly via the visited network i.e. by the P-CSCF. The local services can be an emergency call or other localised services such as services related to geographical location of the user or local numbering plans [6].

6 SIP protocol in 3GPP Rel5

6.1 SIP in IMSS and Service SS

SIP and SDP as a protocol has been selected to some and IPv6 as the only solution to all of the IP Multimedia Subsystem interfaces.
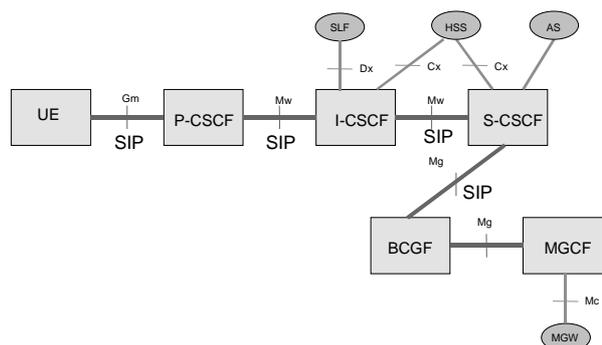

Figure 4: SIP protocol in IM SS [5]

As shown by the figure 4 the basic SIP (RFC2543) has been selected as the main protocol on the following interfaces:
- Gm: P-CSCF - UE
- Mw: P-CSCF – S-CSCF and P-CSCF – I-CSCF
- Mm: S/I-CSCF - external IP networks & other IMS networks

- Mg: S-CSCF – BCGF
- Mk: BCGF – external IP networks & other IMS networks

Eventually there may be differences in the SIP procedures of Gm and Mw reference points. This implies that there is a difference in UNI and NNI interfaces [3].

The following procedures have been defined for the 3GPP IM subsystem in [3]:

- Local P-CSCF discovery: Either using DHCP or carrying address in the PDP context
- S-CSCF assignment and cancel
- S-CSCF registration
- S-CSCF re-registration
- S-CSCF de-registration (UE or network initiated)
- Call establishment procedures separated for
    - Mobile origination; roaming, home and PSTN
    - Mobile termination; roaming, home and PSTN
    - S-CSCF/MGCF – S-CSCF/MGCF; between and within operators, PSTN in the same and different network
- Routing information interrogation
- Session release
- Session hold and resume
- Anonymous session establishment
- Codec and media flow negotiation (Initial and changes)
- Called ID procedures
- Session redirect
- Session Transfer

6.2   SIP in Service SS

The service subsystem and its connections to IM subsystem is shown in the figure 5. The S-CSCF interfaces the application development servers with SIP+ protocols. The SIP application server can reside either outside or within operators network [3]. The OSA capability server and Camel refer to already standardised 3G and GSM based service generation elements.
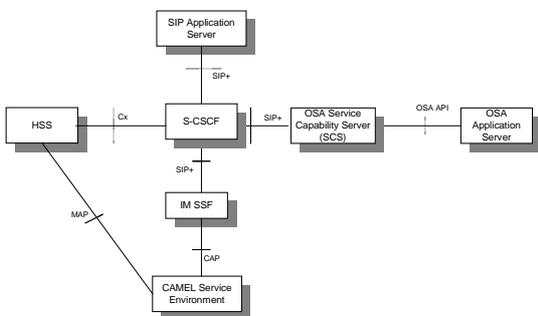
Figure 5: Service Subsystem connections with IMSS

SIP+ is used to interface the Application servers on the following interfaces:

- S-CSCF- SIP Application server
- S-CSCF- Camel Server
- S-CSCF-OSA Service Server

The plus sign implies here that the standard SIP may need to be modified. The modifications have not been identified yet [3].

7   3GPP SIP requirements

3GPP is in its specifications referring to IETF specifications and the target has been to minimise the changes. 3GPP is currently dependent on completion of the following SIP WG items [5] :
- draft-ietf-sip-rfc2543bis: SIP: Session Initiation Protocol
- draft-sip-manyfolks-resource: Integration of resource management and SIP
- draft-ietf-sip-100rel: Reliability of Provisional Responses in SIP
- draft-ietf-sip-privacy: SIP extensions for caller identity and privacy
- draft-ietf-sip-call-auth: SIP extensions for media authorization
- draft-roach-sip-subscribe-notify

 3GPP has found out that significant amount of the features are already provided by the SIP protocol and therefore very few candidates 3GPP originated enhancements have been identified [7]. The following SIP enhancements have been recognised so far [1][8]:
- **addition of routing PATH header** to the SIP messages to record the signalling path from P-CSCF to S-CSCF
- **location information in the INVITE message** to carry the location of the terminal (for instance Cell ID)
- **emergency call type** is needed to indicate the type of emergency call i.e. is it police, ambulance etc.
- **filtering of routing information** in the IM SS before the SIP message is sent to the terminal to hide the network topology from terminal
- **refresh mechanism inside IM SS**
- **Network-initiated de-registration**
- **183 Session Progress provisional response** for INVITE to ensure that the altering is not generated before PDP contexts for session are activated
- **Reliability of provisional responses** – PRACK method to acknowledge the 183 message
- **Usage of session timers** to keep the SIP session alive
- **Indication of resource reservation status** – COMET method
- **Security for privacy**
- **Extensions for caller preferences and callee capabilities**

Discussions are currently ongoing on the changes between 3GPP and IETF.

8 Problems

The following problems can be identified in the 3GPP SIP usage:

- **architecture complexity** i.e. with several functions there will be several interfaces, implementation may differ from vendor to vendor thus the multivendor cases may become challenging
- **call establishment delay problems** due to the signalling taking place on multiple levels (RAN, PS CN, IMSS)
- **guarantees of QoS**: Several elements and several IP based interfaces, in addition the packet radio included in the path while the requirements are at the same level as current GSM circuit voice calls
- **lengthy standardisation time:** more issues there are to standardise, more there are opinions and more time it will take
- **suitability of the SIP protocol for the radio interface** i.e. it is a character based protocol with long signalling messages and requires certain transport quality
- **IETF and 3GPP standardisation co-operation:** the operations and the behaviour are different in IETF and 3GPP
- **Terminal complexity:** the terminals become more and more complex with several protocol stacks, only to provide very similar services than today. SIP has to provide true revolution in applications and services !

Conclusion

The major identified differences with the SIP IETF architectures are as follows:

1. the architecture of the IMSS is defined based on 3G model (home and visited)
2. in 3G networks there is a separation of UNI and NNI interface
3. due to radio and packet core functionality there are some change proposals to the SIP and SDP
4. due to the P-CSCF – S-CSCF interface and the 3G roaming mode there are some requirements to the SIP and SDP protocols
5. in 3G SIP is used also to interface the application development elements, they set requirements for SIP and SDP protocols

Despite of the above mentioned differences it seems that the SIP protocol is suitable to suit the needs of the UMTS network. The identified problems can be overcome and some of them have political or architectural nature thus they are more of choices than problems. The current work in 3GPP is still unfinished and the discussion with IETF has just been started. It is likely that the 3GPP Release 5 shall contain some specifications on SIP and IMSS architecture but their maturity is not probably too good by the end of 2001 to guarantee fully functioning network. One major advantage is the SIP changes so far required by 3GPP are not extensive thus the SIP can probably be tailored for 3GPP. However, since the specification work for a new subsystem is a relatively large, it can be expected that the specification work will continue also during 2002 and beyond. When the SIP and IMSS has been finalised for the UMTS network the real-time packet services can provide for the operators a true way to differentiate from each other and thus generate longed for revenues.

References

[1] Ahvonen, Kati: Master's Thesis: IP telephony signalling in a UMTS All IP network, Helsinki University of Technology, 24.11.2000

[2] 3G TS 23.002 version 5.1.0 Network Architecture (Release 5)

[3] 3G TS 23.228 version 2.0.0 IP Multimedia (IM) Subsystem - Stage 2

[4] 3G TS 24.228 version 2.0.0 IP Multimedia (IM) Subsystem - Stage 3

[5] Drage, Keith: 3GPP and SIP. Presentation in IETF #50 (March 18-23, 2001). http://www.softarmor.com/sipwg/meets/ietf50/slides/drage-3gpp-sip.ppt

[6] 3G TS 22.228 V5.0.0 (2001-01) Service requirements for the IP Multimedia Core Network Subsystem (Stage 1) (Release 5)

[7] Meeting minutes of 3GPP TSG-CN/SA SIP ad-hoc, February 12-14 http://www.3gpp.org/ftp/TSG_CN/WG1_mm-cc-sm/SIP_meetings/CN1_SA2_03_(New%20Jersey)/Report/NewJersey0102.zip

[8] Tdoc N1-010233; 3GPP TSG-SA WG2 / TSG-CN WG1 SIP ad-hoc meeting, 13-15 February, 2001, New Jersey, USA; Nokia: Feedback from IETF's interim SIP WG meeting held on week #6 http://www.3gpp.org/ftp/TSG_CN/WG1_mm-cc-sm/SIP_meetings/CN1_SA2_03_(New%20Jersey)/Tdocs/N1-010233%20.zip

[9] Tdoc N1-010429; 3GPP TSG-SA WG2 / TSG-CN WG1 SIP ad-hoc meeting, Routing of SIP requests in call flows http://www.3gpp.org/ftp/TSG_CN/WG1_mm-cc-sm/SIP_meetings/CN1_SA2_02_(New%20Jersey)/Documents/N1-001429.zip