Call Setup in the IMS

V.2

Changes against the previous version are marked in pink.

PRECONDITION

This presentation expects basic knowledge

- of the architecture of IMS
- of SIP
- of the architecture of UMTS
- of User Identities
- of SIM, USIM, and ISIM
Table of Contents

- IP Connectivity
  - Location Update
  - Activate PDP-Context
  - IMS Service Registration
- Call Origination to IMS Terminal
  - The Scenario and involved Nodes/Functions
  - The Total Messages Exchange
  - The Messages Exchange in Detail
- Call Origination to PSTN Terminal
  - The Scenario and involved Nodes/Functions
  - The Total Messages Exchange
  - The Messages Exchange in Detail
- Abbreviations
- Appendices

25.02.2007  Call setup in IMS V.2 Seite 3

PRE CONSIDERATION

- IMS allows two different configurations depending on whether the P-CSCF is located in the home or in the visited network.
- When the IP Connectivity Access Network is GPRS the location of the P-CSCF is subordinated to the location of the GGSN, because P-CSCF controls the GGSN via the so called GO interface which is always an network internal interface because otherwise its operation would be complex.
- In the long term vision of IMS the P-CSCF (and GGSN) is located in the visited network (which requires IMS support by the visited network)
- In the short term vision of IMS the P-CSCF (and GGSN) is located in the home network (because it can not be expected that all roaming partners will upgrade their networks at the same time the home network starts with IMS). In this case no IMS support is expected from the visited network (i.e. no 3GPP Release 5 compliant GGSN is provided in the visited network)
- This configuration has the severe disadvantage that it causes tromboning of media streams, as the media plane traverses the GGSN and thus takes a longer way to its destination when GGSN is located in the home network.
- This presentation is based on the long term vision of IMS

25.02.2007  Call setup in IMS V.2 Seite 4
1. IP-Connectivity

1. via DSL, Dial-Up, LAN
2. via UTRAN, WLAN

In case of wireless access, the IP-connectivity procedure happens typically when user terminal (i.e. IMS terminal) is switched to power on. As shown in the next slides, at least 32 messages to be exchanged between different network nodes/functions to successfully authorize the user and register for IMS services after IMS terminal power on.

Assumptions for this section of the presentation

- This presentation describes the usage of an UMTS/IMS UE (User Equipment). The IP-CAN (IP Connectivity Access Network) is GPRS and thus, is composed of SGSN and GGSN nodes.
- The UE is roaming and is equipped with an UICC (Universal Integrated Circuit Card) which include an ISIM application (ISIM is IMS specific while USIM is only UMTS specific). 
- The visited network to which the UE is going to attach provides IMS services. 
- P-CSCF (as well as the GGSN) of the calling party is located in the visited network.
1.1 Location update and setup of logical connection between UE and SGSN (Service GPRS Support Node)

1. When UE is switched on, the ATTACH message is sent via RNC to the SGSN.
   Note: RNC and SGSN are fixed connected to each other.

2. SGSN asks the HLR for authentication by providing the IMSI of the subscriber

3. HLR confirms authentication

4. SGSN asks the HLR for location update, i.e. informs the HLR about the cell_id in which the UE resides

5. HLR responds with the sending of the subscriber profile (subscribed services, QoS profile, static IP-address allocated by the network operator, etc.)

6. SGSN sends ATTACH COMPLETE message to the UE, including the UE IP-address

1.2.1 Activate PDP-Context (request for a particular Access Point Name and either packet connectivity type IPv4 or IPv6 in order to obtain an IP-address of the service node to be used (e.g. address of the P-CSCF).

1. When ATTACH COMPLETE message is received, the UE automatically sends the ACTIVATE PDP CONTEXT REQUEST message to the SGSN. In case the UE is an IMS terminal the APN requested by the UE indicates the IMS network.
   APN (Access Point Name) is is the logical name for a service or network).

2. SGSN provides a crosscheck with the subscriber profile initially received from the HLR and selects the appropriate service node (e.g. the appropriate GGSN Gateway GPRS Support Node).

3. If the subscriber is NOT roaming or the subscriber is roaming, but the visited network provides IMS services as well and thus, the P-CSCF is located in the visited network (SGSN), the SGSN sends the CREATE PDP CONTEXT REQUEST to the appropriate GGSN, which is in the same network to which the SGSN belongs. Otherwise the SGSN has to traverse the message to the GGSN of the home network. Reason: GGSN and P-CSCF to be located in the same network, because their interface is always an intra-operator interface (which makes its operation simpler).
1.2.2 Activate PDP-Context ff

4. GGSN sends CREATE PDP CONTEXT RESPONSE message to the SGSN. This message includes the PDP-address (i.e. the IP-address) of the P-CSCF (the GGSN anchors the PDP-addresses).

5. SGSN sends ACTIVATE PDP CONTEXT ACCEPT message to UE. It conveys the IP-address of the P-CSCF to be used by the UE to continue with SIP services.

A logical tunnel has now been set up between the UE and the GGSN. This is the so-called "integrated procedure".

In the so-called "stand alone procedure" in step 4 not the address of the P-CSCF but the address of the GGSN is returned to the UE and the UE has to contact the DNS (with the help of DHCP server) to get the address of the P-CSCF.

From now on the P-CSCF is allocated to the UE. This allocation does not change anymore until UE is powered off.

1.3 IMS Service Registration (request authorization to use IMS services). This is mandatory before a session can be established

Purpose of Service registration:
1. User binds his public user identity to a contact address
2. Home network authenticates the user
3. User authenticates the home network
4. Home network authorizes the SIP registration and usage of IMS resources
5. Home network verifies that there is a roaming agreement with the visited network
6. UE and P-CSCF negotiate the security mechanism in place for subsequent signalling
7. P-CSCF and UE establish a set of security associations for the integrity of SIP messages exchanged
8. UE and P-CSCF upload to each other the algorithms used for SIP message compression (SIP message compression is essential for the air-interface, as SIP is a text-based protocol and thus, needs high bandwidth)
1.3.1 IMS Service Registration

1. UE sends SIP REGISTER message to P-CSCF. This message includes:
   - URI that identifies the home network (e.g., ims-mobilkom.com)
   - Public User Identity (i.e., the SIP address of the user, which is e.g., is on his business card)
   - Private User Identity (compatible with the ISIM in GSM, and used for authentication only)
   - Contact address (IP address of the UE or the host name where the user is reachable)
   The first three IDs are stored in the ISIM of the UE.

2. P-CSCF contacts the DNS to locate an entry point into the home network by sending the home domain name.

3. DNS provides the address (the SIP URI) of the I-CSCF of the home network.
   Note: If the home network wants to keep its configuration private, I-CSCF is the entry point. Otherwise, the DNS provides directly the SIP URI of the S-CSCF.

4. P-CSCF sends SIP REGISTER message to I-CSCF. This message includes:
   - SIP URI of the P-CSCF
   - Public User Identity
   - Private User Identity
   - visited network ID (for the check of existence of a roaming agreement)

   25.02.2007 Call setup in IMS V.2

---

1.3.2 IMS Service Registration

5. I-CSCF is not aware of whether or not an S-CSCF is already allocated to the user and what the address of this S-CSCF is. To check this and additionally to carry out a first step authorization, I-CSCF sends a Diameter (=Cx-Interface) message UAR (User-Authentication-Request) to the HSS. This message includes the:
   - visited network ID
   - Private User Identity
   - Public User Identity

6. HSS checks if a roaming agreement exists and validates Private User ID and Public User ID. It sends the Diameter message UAA (User-Authentication-Answer) to I-CSCF. This message includes either the SIP URI of the S-CSCF if already allocated previously or a set of S-CSCF capabilities.
   If this is the first registration after UE is powered on, HSS usually returns a set of mandatory (e.g., SIP calling) and optional (e.g., charging) S-CSCF capabilities.
   Note: The standard does not indicate what these capabilities are and how they are specified. They are network individually defined.

25.02.2007 Call setup in IMS V.2
1.3.3 IMS Service Registration ff

I-CSCF has a configurable table of S-CSCFs and their capabilities and selects the appropriate S-CSCF. Then it sends the SIP REGISTER message to this S-CSCF, including:
- subscriber ID
- visited network contact name
- home network contact point
- P-CSCF name.

S-CSCF contacts the HSS by means of the Diameter message MAR (Multimedia Auth-Request) to get authentication data of the subscriber and to inform the HSS about the S-CSCF URI allocated to that user. Note: S-CSCF needs user data to authenticate the user (initial registrations are always authenticated by the S-CSCF, other SIP messages like INVITE are never authenticated in the IMS).

HSS returns the user authentication data, which will be stored in the S-CSCF.

25.02.2007 Call setup in IMS V.2 Seite 13

1.3.4 IMS Service Registration ff

S-CSCF sends SIP 401 UNAUTHORIZED message back to the UE. This message traverses the I-CSCF, the P-CSCF, and the GPRS-nodes.

Upon receipt of the SIP 401 the UE realizes that there is a challenge included and contacts its USIM to build up the credentials. Note: The actual credentials depend on the IMS network and are derived from UICC. In fact, authentication information is stored in the UICC, which hold the credentials used for security purposes.

25.02.2007 Call setup in IMS V.2 Seite 14
12 UE sends again a SIP REGISTER message to the P-CSCF.

13 The P-CSCF does the same procedure as already done after reception of the initial SIP REGISTER message: It contacts DNS to get the address of the I-CSCF of the home network. 
Note: Due to DNS load balancing mechanisms the address of the I-CSCF in the home network may not be the same as derived upon reception of the first SIP REGISTER message

14 P-CSCF traverses the SIP REGISTER message to the I-CSCF, which now will run the same procedure as already run initially

15 As the (eventually new) I-CSCF is not aware of whether or not an S-CSCF is already allocated to the user and what the address of this S-CSCF is and ... eventually - to carry out a first step authentication, I-CSCF sends a Diameter (=Cx-Interface) message UAR (User-Authentication-Request) to the HSS. This message includes again the visited network ID, the Private User Identity, and the Public User Identity
HSS returns the Diameter message UAA (User-Authentication-Answer) to I-CSCF, which includes now the SIP URI of the S-CSCF already allocated to the user.

25.02.2007 Call setup in IMS V.2 Seite 15
1.3.7 IMS Service Registration

16 I-CSCF forwards the SIP REGISTER message to the S-CSCF, which validates the credentials received against the authentication data already stored.

17 The S-CSCF sends the Diameter message SAR to inform the HSS that the user is now registered. In response by means of the Diameter message SAA the HSS returns the user profile to the S-CSCF, which will be stored locally in the S-CSCF. Note: The user profile includes all the Public User Identities and indicates which of them are automatically registered in the S-CSCF. It also includes the initial filter criteria, i.e. a collection of triggers used to determine the application server providing the service when a SIP request arrives.

S-CSCF has stored now the contact URI for the user as well as the list of URIs along the path to the UE. The S-CSCF will route initial SIP request to the UE along this list of URIs.

18 S-CSCF sends the SIP 200 OK back to the UE to indicate the successful registering. The S-CSCF anyway and may be also the I-CSCF add its address to the Record-Route Header (dependent on the strategy of the home network operator). The UE is now registered in the IMS for the duration of the expires parameter of the SIP 200 OK.

2. Call Origination to IMS Terminal

This chapter of the presentation describes how the UE sets up a session to another IMS terminal while both IMS terminals are roaming in different visited networks and both IMS terminals belong to different home networks.
Assumptions for this section of the presentation

- This presentation describes the usage of UMTS/IMS UE (User Equipment) on originating and terminating side
- The network to which the UEs are attached provide IMS services
- The originating UE is roaming and is equipped with an UICC (Universal Integrated Circuit Card) which include an ISIM application (ISIM is IMS specific while USIM is only UMTS specific)
- Call origination describes the session set up initiated by UE1 to another IMS terminal (UE2) roaming in another visited network.
- Both UE have registered in their home IMS
- The Home Network of UE1 does not involve an I-CSCF, the Home Network of UE2 does
- Neither at the calling side nor at the called side an IMS Application Server is involved

2.1 The Scenario and involed Nodes/Functions

![Diagram showing the scenario and involved nodes/functions]
2.2 The total Messages exchange (SIP messages are black, Diameter messages are red)

1. UE1 sends the SIP INVITE message to its P-CSCF1. It includes:
   a) the Public User Identity of the called party (sip: bob@home2.net)
   b) the IP address and port number where the UE1 expects a response as well as the info for signal compression and the transport protocol used to the next hop (e.g. UDP, TCP, SCTP). Note: Every node in the chain is free to choose its appropriate transport protocol
   c) the IP address and port number where the UE1 expects subsequent responses after the response to the INVITE as well as the info for signal compression
   d) a route list (list of SIP proxies which serve the UE1 and which to be traversed, e.g sip: PCSCF1@visited1.net and sip:SCSCF1@home1.net)
   e) the preferred identity of the UE1 user ("Alice Smith" sip: alice@home1.net) if the user has more than one Public User Identity, to indicate which one to be used for this session (to be included in the charging record, to be shown to the called party, to trigger different services)
   f) end-to-end info explaining who is calling (sip: alice@home1.net), who is called (sip: bob@home2.net), and the Call-ID
   g) additional information like e.g. SIP-extensions to be used/supported and the audio and/or video codec format supported by the UE1. Note: The complete message content is shown in APPENDIX A of this presentation!
25.02.2007 Call setup in IMS V.2 Seite 23

2.3.2 The Messages Exchange in Detail

Upon reception of the INVITE by means of sending the TRYING message the P-CSCF1 returns an acknowledgement back to the UE1 (to inform the sender of the INVITE that his message has been reliably received by the next hop in the chain).

Next, the P-CSCF1 provides some internal checks and procedures:
- check if Route Header is correct and includes the S-CSCF in the home network
- check the requested media parameters against the policy of the visited network operator (e.g. G.711 codec not allowed because of 64 Kbit-bandwidth necessity)
- check the P-Preferred Identity against the list of all the Public User Identities received during the Terminal Registration Process and replace the P-Preferred Identity against the P-Asserted Identity in the INVITE message sent further to the S-CSCF if there is no match, P-CSCF selects one Public User ID out of its list. If there is a match, it puts the received Public User ID into the INVITE message (which includes a list of media types, codecs and other SDP-parameters which are allowed) to UE1.

The S-CSCF1 sends the modified INVITE message to the I-CSCF2.

2 Trying
1 INVITE
3 INVITE
4 Trying
5 INVITE
6 INVITE

25.02.2007 Call setup in IMS V.2 Seite 24

2.3.3 The Messages Exchange in Detail

Upon reception of the INVITE the S-CSCF1 (it was allocated to the UE1 during Service Registration procedure) identifies the user by means of the value in the P-Asserted-Identity header and retrieves the User Profile which was already downloaded during Terminal Registration. Next the S-CSCF1:
- evaluates the filter criteria stored in the User Profile (to find out if — and which — Application Servers to be involved
- check SDP-parameters against local network policy (e.g. codec format, because user has a cheap subscription which does not allow some media or high speed codecs)
- analyses the called address which can be a SIP URI or a TEL URI. In case of
  - a SIP URI (sip:bob@home2.net) or a SIP URI with mapped telephone number (e.g. sip:+1-212-555-0239@home2.net)
  - a TEL URI (telephone leading to an PSTN user (e.g. tel:+1-212-555-0239) or SIM user) S-CSCF1 contacts ENUM to get a SIP URI. If there is no SIP URI available, S-CSCF1 will contact the BGC (Breakout Gateway Control Function).
- adds a TEL URI of the caller to the P-Asserted-Identity header of the INVITE message. This is used in the case the call terminates in the PSTN to enable the PSTN to identify the caller.

S-CSCF1 sends the modified INVITE message to the I-CSCF2.

2 Trying
1 INVITE
3 INVITE
4 Trying
5 INVITE
6 INVITE

25.02.2007 Call setup in IMS V.2 Seite 24
2.3.4 The Messages Exchange in Detail

6 I-CSCF2 acknowledges the message reception by sending back the TRYING message to S-CSCF1.

7 The I-CSCF2 queries the HSS2 about the called SIP URI to get informed, which S-CSCF is already allocated to that user (during the Register Terminal procedure the address of the S-CSCF2 was stored in the HSS). It sends the Diameter message LIR (Location Information-Request), which includes the value ip: sip:bob@home2.net.

8 HSS2 returns the address of the allocated S-CSCF2 to the I-CSCF2 by means of the Diameter message LIA (Location Information_Answer).

9 I-CSCF2 forwards the INVITE message to the S-CSCF2. In this message the address of the I-CSCF2 may or may not be inserted by the I-CSCF2, which is dependent on the configuration made by the network operator (either he wants to hide the address of the S-CSCF2 or not. If it shall be hidden, all SIP signalling will pass the I-CSCF2, so the I-CSCF2 adds its address to the Record-Route Header before sending the INVITE message to the S-CSCF2).

25.02.2007 Call setup in IMS V.2 Seite 26

10 S-CSCF2 sends the TRYING message back to the I-CSCF2.

11 Upon reception of the INVITE message the S-CSCF2 evaluates the initial filter criteria (same evaluation as the S-CSCF1 has already done) to check if Application Services to be involved at the called side. As the S-CSCF2 typically remains in the signalling path (in 3GPP R6 always, in 3GPP R6 in some cases) the S-CSCF2 adds its own SIP URI to the Record-Route Header in the INVITE message. Additionally, it inserts the address of the P-CSCF2 (sip:pcscf2@visited2.net) in the Route Header (learned during Terminal Register Procedure when UE2 was powered on). Thus all SIP signalling will now traverse I-CSCF2 as well as P-CSCF2.

Furthemore, the S-CSCF2 will probably change the Request-URI in the INVITE message. This depends on the Public User Identity the called user has registered during Terminal Register procedure. Note: A user can have several Public User Identities and can register subsequently one after the other by sending new REGISTER messages to the S-CSCF (e.g. sip:bob@home2.net and sip:bob-business@home2.net). In the REGISTER messages the Contact Header always includes the SIP URI of the UE2 terminal as well. Depending on the Public User Ids registered by now, the S-CSCF2 retargets the Public User Identity of the called party but adds the initial called party ID as well because of the following reason: The called party has set his terminal configuration as such that it alerts with different tones if called for business or private reason.

S-CSCF forwards the modified INVITE message to the P-CSCF2.
2.3.6 The Messages Exchange in Detail

Upon reception of the INVITE message the UE2 sends the TRYING message and

follows the call flow model stated in the require-header: precondition (i.e. it has to respond with a SESSION PROGRESS message that contains an SDP answer to communicate the media streams and codecs the UE2 is able to handle for this session).

Next, the UE2 inspects

- the P-Asserted-Identity header to extract the identity of the caller and
- the P-Called-Party-ID header to determine to which of the several identities of the user the INVITE is addressed.

The combination of both identities may be used now or in a later stage to play a personalized ringing or display a picture of the caller, etc.

The called terminal is now in the pre-alert stage.

The SESSION PROGRESS message sent by the UE2 back to P-CSCF-2 includes an advice for UE1 to send an updated SDP when terminal resource reservation on calling side is made (only when resource reservation has been completed on both sides, the calling and the called side, the called party will be alerted!).

25.02.2007 Call setup in IMS V.2 Seite 28
2.3.8 The Messages Exchange in Detail

The SESSION PROGRESS message traverses step by step all the nodes back to the UE1. Except the I-CSCF1, all other nodes provide some checks and functions before sending the message out.

E.g. P-CSCF2 inserts a P-Asserted-Identity header whose value is the same as that included in the P-Called-Party-ID header of the former INVITE (by this way the other nodes get the public user identity of the called party being used for this session).

E.g. the S-CSCF-1 removes the P-Asserted-Identity header if privacy requirements indicate so.

Finally, the SESSION PROGRESS message arrives at the UE1.

16 Upon reception of the SESSION PROGRESS message (which includes the IP-address of UE2) the UE1 is informed - whether or not the UE2 accepts a session with the media streams proposed (or only audio but no video)
- about the codecs supported and desired at the called side

Then the UE1 starts resource reservation, i.e. a procedure which is dependent on the underlying IP Connectivity Access Network and will require
some dialog with the packet and radio nodes (GGSN, SGSN, RNC).

Finally, the UE1 forwards the PRACK message (including the final SDP) to the UE2.

Note: At this time the resource reservation of UE1 most probably will not be completed!

This message traverses all the nodes in the chain (the path is derived from the Record-Route header of the SESSION PROGRESS message).

2.3.9 The Messages Exchange in Detail

Upon reception of the SESSION PROGRESS message (which includes the IP-address of UE2) the UE1 is informed - whether or not the UE2 accepts a session with the media streams proposed (or only audio but no video)
- about the codecs supported and desired at the called side

Then the UE1 starts resource reservation, i.e. a procedure which is dependent on the underlying IP Connectivity Access Network and will require
some dialog with the packet and radio nodes (GGSN, SGSN, RNC).

Finally, the UE1 forwards the PRACK message (including the final SDP) to the UE2.

Note: At this time the resource reservation of UE1 most probably will not be completed!
2.3.10 The Messages Exchange in Detail

18 Upon reception of the PRACK message the UE2 starts resource reservation involving GGSN, SGSN, RNC at the called side. The OK message traverses all the nodes in the chain to UE1.

19 When the necessary resources have been reserved at the called side, UE1 sends the UPDATE message to UE2 (traversing all the nodes in the chain).

20 As any other message with SDP-content the reception of the UPDATE message will be acknowledged by the UE2 with an OK message (traversing all the nodes in the chain).

At this time the UE2 may still be engaged in resource allocation.

2.3.11 The Messages Exchange in Detail

21 Once resource reservation has been completed at the called side (as well as at the calling side - these are independent processes which can be completed in any order) the UE2 starts alerting the called party and generates the RINGING message back to UE1.

22 Upon reception of the RINGING message the UE1 applies locally stored ringing tone to the caller and sends the PRACK message to UE2.

23 The PRACK message will be acknowledged by the UE2 by sending the OK message to UE1.

At this stage the called party gets ringing and the calling party hears ringing tone.
2.3.12 The Messages Exchange in Detail

24 When the called party answers (i.e. accepts the session) the UE2 sends an OK message which completes the INVITE transaction at the called side.

25 When the OK message has arrived, the UE1 stops ringing tone and forwards the ACK message to UE2 to acknowledge the establishment of a session.

26 The session set up is now completed and both parties can generate their audio and video streams. These media streams are sent end-to-end (UE1<->UE2) via the media plane.

---

3. Call Origination to PSTN Terminal

This chapter of the presentation describes how the roaming UE (an IMS terminal) sets up a session to a PSTN terminal.
Assumptions for this section of the presentation

- This presentation describes the usage of an UMTS/IMS UE (User Equipment) for call set up.
- The network to which the UE is attached provides IMS services.
- The UE is roaming and is equipped with an UICC (Universal Integrated Circuit Card) which include an ISIM application (ISIM is IMS specific while USIM is only UMTS specific).
- Call origination describes the session set up initiated by UE to a PSTN terminal.
- The Home Network of UE does not involve an I-CSCF.
- Based on the destination address and operator agreements the session is handled by the BGCF of the home network.
- Based on the home network configuration the BGCF does not remain in the signalling path after MGCF has been selected.
- No IMS Application Server is involved at the calling side.
3.2 The total Messages exchange (SIP messages are black, CS-messages are red)

The Messages Exchange in Detail

1. UE1 sends the SIP INVITE message to its P-CSCF. It includes:
   a) the Public User Identity of the called party, which is a TEL-URI (tel: +1-212-555-0293)
   b) the IP address and port number where the UE1 expects a response as well as the info for signal compression and the transport protocol used to the next hop (e.g. UDP, TCP, SCTP). Note: Every node in the chain is free to choose its appropriate transport protocol
   c) the IP address and port number where the UE1 expects subsequent responses after the response to the INVITE as well as the info for signal compression
   d) a route list (list of SIP proxies which serve the UE1 and which to be traversed, e.g. PCSCF1@visited1.net and SCSCF1@home1.net)
   e) the preferred identity of the UE1 user ("Alice Smith" alice@home1.net) if the user has more than one Public User Identity, to indicate which one to be used for this session (to be included in the charging record, to be shown to the called party, to trigger different services)
   f) the type of access network used by UE1 (UTRAN) for service customization and determination of available bandwidth as well the radio cell ID which implicitly contains some location info to be used for local services like e.g. list of local dentists. Note: This info is transferred down to the home network but not further!
   g) end-to-end info explaining who is calling (tel: +1-222-666-1234, who is called (tel: +1-212-555-0293), and the Call-ID
   h) additional information like e.g. SIP-extensions to be used/supported and the audio and/or video codec format supported by the UE1.
3.3.2 The Messages Exchange in Detail

2 Upon reception of the INVITE, by means of sending the TRYING message the P-CSCF1 returns an acknowledgement back to the UE1 (to inform the sender of the INVITE that his message has been reliably received by the next hop in the chain).

3 Next, the P-CSCF provides some internal checks and procedures:
- check if Route Header is correct and includes the S-CSCF in the home network
- check the requested media parameters against the policy of the visited network operator (e.g. G.711 codec not allowed because of 64 kb/s bandwidth necessity)
- check the P-Preferred Identity against the list of all the Public User Identities received during the Terminal Registration Process – and replace the P-Preferred Identity against the P-Asserted identity in the INVITE message sent further to the S-CSCF if there is no match, P-CSCF selects one Public User ID out of its list, if there is a match, it puts the received Public User ID into the P-Asserted identity header. This check is providing authentication of the Public User ID.
- remove/modify some Headers (e.g. which relate to security or signal compression) and insert charging headers in the INVITE message
- record the route together with its own SIP URI

If all the checks were o.k., the P-CSCF forwards the modified INVITE to the S-CSCF (or to the i-CSCF of the home network operator).

25.02.2007 Call setup in IMS V.2 Seite 39

3.3.3 The Messages Exchange in Detail

4 S-CSCF sends TRYING message back to P-CSCF

5 Upon reception of the INVITE the S-CSCF allocated to the UE1 identifies the user by means of the value in the P-Asserted-Identity header and retrieves the User Profile which was already retrieved during Terminal Registration. Next the S-CSCF:
- evaluates the filter criteria stored in the User Profile to find out if and which Application Servers to be involved
- check SDP-parameters against local network policy (e.g. codec format, because user has a cheap subscription which does not allow some media or high speed codecs)
- analyses the called address which can be a SIP URI or a TEL URI. In case of a TEL URI which could belong to a PSTN user (e.g. +1-212-555-1234) or GSM user, S-CSCF contacts ENUM to get a SIP URI. If there is no SIP URI available, ENUM returns a negative response which triggers the S-CSCF next to contact the BGCF (Breakout Gateway Control Function, specialized in routing SIP requests based on telephone numbers)
- adds a TEL URI of the caller to the P-Asserted-identity header of the INVITE message. This is used to enable the PSTN to identify the caller

S-CSCF forwards the modified INVITE message to the BGCF

25.02.2007 Call setup in IMS V.2 Seite 40
3.3.4 The Messages Exchange in Detail

Upon reception of the INVITE the BGCF returns the TRYING message and analyses the destination address (i.e. the TEL URI). Based on agreements the home network operator may have for call termination in the PSTN, the BGCF decides whether the session should be handled by a local MGCF or by a remote MGCF. If the session to be handled locally, the BGCF further decides if it wants to stay in the chain of nodes traversing the further message flow or not (i.e. insert the own SIP ID into the Record Route header of the INVITE message or not). Then the INVITE message is routed to the MGCF (in our case: to a local MGCF, and announcing not to remain in the signalling path for the rest of the session).

Upon reception of the INVITE message the MGCF returns the TRYING message and selects the SGW as well as the MGW to be used for this session (one MGCF can control many SGW and MGW). Then it follows the call flow model stated in the require-header of the INVITE message precondition (i.e. it has to respond with a SESSION PROGRESS message that contains an SDP answer to communicate the media streams and codecs the MGW is able to handle).

Next, the MGCF inspects the PAsserted-Identity header to extract the identity of the caller. The SESSION PROGRESS message sent by the MGCF back to UE1 includes an SDP as well as an advice for UE1 to send an updated SDP and to communicate when terminal resource reservation on calling side is completed (only when resource reservation has been completed on both sides, the calling and the called side, the called party will be alerted).

3.3.5 The Messages Exchange in Detail
3.3.6 The Messages Exchange in Detail

Upon reception of the SESSION PROGRESS message (which includes the IP-address of the reserved PCM channel at the MGW as well as the Route Header field without the address of the BGCF) the UE1 is informed:

- whether or not the UE2 accepts a session with the media streams proposed (for the time being only audio is specified)
- about the codecs supported and desired at the called side

The UE1 now decides for one audio codec which is supported at both ends.

Then the UE1 starts resource reservation, i.e. a procedure which is dependent on the underlying IP Connectivity Access Network and will require some dialog with the packet and radio nodes (GGSN, SGSN, RNC).

Finally, the UE1 forwards the PRACK message (including the final SDP) to the MGCF.

Note: At this time the resource reservation of UE1 most probably is not been completed!

The PRACK message traverses all the nodes listed in the Record-Route header of the SESSION PROGRESS message.

The MGCF confirms final codec format by means of the OK message.

The OK message traverses all the nodes in the chain back to UE1.

When the necessary resources have been reserved at the calling side, UE1 sends the UPDATE message to MGCF (traversing all the nodes in the chain).

Upon reception of the PRACK message the MGCF starts resource reservation in the MGW. The messages between the MGCF and the SGW are typically transported over SCTP; the interaction with the MGW is based on H.428.

As any other message with SDP-content the reception of the UPDATE message will be acknowledged by the MGCF with an OK message (traversing all the nodes in the chain).

The MGCF triggers the SGW to establish a speech path through the PSTN down to the called party.

This request is transcoded by the SGW into SS7 signaling (the Initial Address Message). In parallel the PCM speech channel is set up between MGW and PSTN.
### 3.3.8 The Messages Exchange in Detail

#### When the CS-path through the PSTN is set up the SGW receives on SS7 the ACM (Address Complete Message) from the PSTN, which indicates that the called party is alerted.

#### The SGW traverses the ACM to the MGCF (packed into SCTP).

#### The MGCF performs the mapping of SS7 signalling and SIP and thus, sends the RINGING message back to UE1.

#### Upon reception of the RINGING message the UE1 applies locally stored ringing tone to the caller and sends the PRACK message back to the MGCF.

#### The PRACK message will be acknowledged by the MGCF by sending the OK message to UE1.

At this stage the called party gets ringing and the calling party hears ringing tone.

---

### 3.3.9 The Messages Exchange in Detail

#### When the called party answers the SGW receives the SS7 message ANC (Answer Call, Charge).

#### The reception of the ANC is communicated to the MGCF (by means of the SCTP).

#### The MGCF sends an OK message to the UE1 which completes the INVITE-transaction at the called side and requests the MGW to activate the PCM channel in forward direction (ACH=Activate Channel).

#### When the OK message has arrived at the UE1 it stops ringing tone and forwards the ACK message to MGCF to acknowledge the establishment of a session.

#### The session set up is now completed and both parties can generate their audio and video streams. These media streams are sent end-to-end (UE1<->UE2) via the media plane.
4. ABBREVIATIONS (1)

**AS** Application Server, interfacing the S-CSCF and handling and executing services. The AS can be located either in the home network or in an external third-party network. There are 3 different types of AS:
- SIF AS: A native application server for IP multimedia services based on SIP.
- OAS: Open Service Access Service Capability Server. It inherits all the OSA capabilities, e.g., those to access the IMS securely from external networks and interfaces, it acts as an interface between the OSA Application Server and the OSA Application Programming Interface as well as an Application Server.
- Enriched Logic developed for GMN.

**BGCF** Broadcast Gateway Control Function (providing routing based on telephone number and/or subscriber ID, e.g., if called as a circuit switched network, it selects the appropriate network where the call will be transferred to; in case of an appropriate PSTN gateway).

**GGSN** Gateway GPRS Support Node, responsible for IP address management and GGS and the provision of external gateway functions.

**GSM** Global System for Mobile Communication. The second generation of mobile networks, incorporating in 1991, is designed for circuit switched mobile voice traffic and the low rate data service (20-20 kbps).

**HSS** Home Subscriber Server. As an evolution of the HLR of the GSM, it is a database for user-related information including location information, security, authentication and authorizations information, user profile, etc.

**ID** Identity, used to uniquely identify users and services. In the PSTN telephone numbers are used to identify users (or services like e.g., 086 for faxphone). In the IMS there is also a deterministic way to identify users and services: the Public User Identity, used to uniquely identify the subscriber, but stored onto the UICC. In 3GPP Release 6 more than one Private User Identities can be hosted in an Application Server. It has the Programm Interface as well as an Application Server.

**IP** Identity, used to uniquely identify users and services. In the PSTN telephone numbers are used to identify users (or services like e.g., 086 for faxphone). In the IMS there is also a deterministic way to identify users and services: the Public User Identity, used to uniquely identify the subscriber, but stored onto the UICC. In 3GPP Release 6 more than one Private User Identities can be hosted in an Application Server. It has the Programm Interface as well as an Application Server.

** IMS** IP Multimedia Subsystem: Based on technical specification of the 3GPP working groups, IMS combines the latest trends in technology, provides a common platform to develop multimedia services, supports GGSN, is interfacing with the internet and circuit switched networks, takes care on roaming and a multitude of charging rules and principles. Release 5 focuses on the circuit switched networks only (i.e., IMS) whereas Release 6 is an enhanced access independent RMI; i.e., it provides support for different access networks.

**IMSI** International Mobile Subscriber Identity, used to uniquely identify users and services. It is stored in the SIM card and is only needed when calling a subscriber who is not currently on an operator's network.

**IP Multimedia Subsystem (IMS)**: Based on technical specification of the 3GPP working groups, IMS combines the latest trends in technology, provides a common platform to develop multimedia services, supports GGSN, is interfacing with the internet and circuit switched networks, takes care on roaming and a multitude of charging rules and principles. Release 5 focuses on the circuit switched networks only (i.e., IMS) whereas Release 6 is an enhanced access independent RMI; i.e., it provides support for different access networks.

**OAS** Open Application Server, interfacing the S-CSCF and handling and executing services. The OAS can be located either in the home network or in an external third-party network. There are 3 different types of OAS:
- SIF OAS: A native application server for IP multimedia services based on SIP.
- OAS: Open Service Access Service Capability Server. It inherits all the OSA capabilities, e.g., those to access the IMS securely from external networks and interfaces, it acts as an interface between the OSA Application Server and the OSA Application Programming Interface as well as an Application Server.
- Enriched Logic developed for GMN.

**RTP** Real Time Protocol and **RTCP** (Real Time Control Protocol) transports real-time media streams.

**SIP** (used to control sessions)

**SIP AS** Application Server for IP multimedia services. It acts as an interface between the OSA and the S-CSCF of the IMS. It is used to control media planes by transporting media streams between PDPs and PEPs.

**SCS** Open Service Access

**SCCF** Session Control Function. Represents an Application Server which is used in the IMS to control the media plane. It is used to transport media streams between PDPs and PEPs.

**TEL URI** but a Network Access Identifier (NAI) and looks like:
- e.g. tel:+15550293
- e.g. sip:+15550293@telecom.com; user=phone

**User Identity** is a database for user-related information including location information, security, authentication and authorizations information, user profile, etc.

**VoIP URI** or a TEL URI or a combination of both (e.g. sip:+15550293@w3com.com; user=phone)

**VMSC** Visitor Mobile Switching Center

**Voicemail Center**

**WCDMA** Wideband Code Division Multiple Access

**WIMAX** Worldwide Interoperability for Microwave Access

**XML** Extensible Markup Language

4. ABBREVIATIONS (2)

**IMS** IP Multimedia Subsystem: Based on technical specification of the 3GPP working groups, IMS combines the latest trends in technology, provides a common platform to develop multimedia services, supports GGSN, is interfacing with the internet and circuit switched networks, takes care on roaming and a multitude of charging rules and principles. Release 5 focuses on the circuit switched networks only (i.e., IMS) whereas Release 6 is an enhanced access independent RMI; i.e., it provides support for different access networks.

**COPS** (Common Open Policy Service) is used to transfer policies between PDPs and PEPs.

**DIAMETER** (which an evolution of RADIUS is used for Authentication, Authorization, and Accounting and is e.g. used to interact with the HSS).

**H.248** (also referred to as MEGACO, say MEdia GAteway COntrol, i.e., those to access the IMS securely from external networks and interfaces) is an evolution of Q.2931, and is designed for circuit switched mobile voice traffic and the low rate data service (14.4 kbps).

**Dia** Diameter is used to control the media plane. It is used to transport media streams between PDPs and PEPs.

**DIAMETER** (which an evolution of RADIUS is used for Authentication, Authorization, and Accounting and is e.g. used to interact with the HSS).

**GW** Gateway

**H.248** (also referred to as MEGACO, say MEdia GAteway COntrol, i.e., those to access the IMS securely from external networks and interfaces) is an evolution of Q.2931, and is designed for circuit switched mobile voice traffic and the low rate data service (14.4 kbps).

**Dia** Diameter is used to control the media plane. It is used to transport media streams between PDPs and PEPs.

**DIAMETER** (which an evolution of RADIUS is used for Authentication, Authorization, and Accounting and is e.g. used to interact with the HSS).

**H.248** (also referred to as MEGACO, say MEdia GAteway COntrol, i.e., those to access the IMS securely from external networks and interfaces) is an evolution of Q.2931, and is designed for circuit switched mobile voice traffic and the low rate data service (14.4 kbps).

**Dia** Diameter is used to control the media plane. It is used to transport media streams between PDPs and PEPs.

**DIAMETER** (which an evolution of RADIUS is used for Authentication, Authorization, and Accounting and is e.g. used to interact with the HSS).

**H.248** (also referred to as MEGACO, say MEdia GAteway COntrol, i.e., those to access the IMS securely from external networks and interfaces) is an evolution of Q.2931, and is designed for circuit switched mobile voice traffic and the low rate data service (14.4 kbps).

**Dia** Diameter is used to control the media plane. It is used to transport media streams between PDPs and PEPs.
4. ABBREVIATIONS (3)

IP-CAN  IP Connectivity Access Network. There are a multiple types of: Digital Subscriber Line, Local Area Networks, GPRS, WLAN, etc.
ISIM  IP multimedia Service Identity Module. It contains parameters used for user identification and -authentication as well as terminal configuration in the IMS environment. This application can co-exist on the ISC with ISIM and USIM. The IMS-relevant parameters stored in the ISIM are:
- Private User Identity (allocated to the user)
- Public User Identity (one or more SIP URIs of Public User identities allocated to the user)
- Home Network Domain URI (HFN URI of the home network domain name)
- Long-term secret (used for authentication and for calculation of the integrity and cipher keys used between the terminal and the network).

I-CSCF  Interrogating Call/Session Control Function. It is logically located at the edge of an administrative domain (usually in the home network) and its address is listed in the DNS. After retrieval of user location information from SLF/HSS, SIP requests are routed further to the S-CSCF. A network will include typically a number of I-CSCF for scalability and redundancy reason.

MGCF  Media Gateway Control Function. Used for protocol conversion (mapping SIP to ISUP over IP and vice versa) and control of resources in the MGW.

MGW  Media Gateway, interfacing the media planes of the GSM or PSTN and thus, mapping the RTP to PDU time slots and performing transcoding when the IMS terminal does not support the codec used at the PSTN side (typically: B8 or B9 terminal using A5/2 codec; PSTN using ITU G.711 codec).

MRF  Media Resource Function, located in the home network and used to play announcements, mix media streams (e.g. conferences, bridged), transcode between different codecs, provides specific statistics and media analysis. The function is subdivided into a Media Resource Function Controller (MRFC, which acts as a SIP User Agent and contains the SIP interface to the S-CSCF) and a Media Resource Function Processor (MRFP, which provides the media-related functions).

P-CSCF  Proxy Call/Session Control Function. Allocated to the HSS terminal during HSS registrations, it is an outbound/ inbound SIP server traversing all the SIP messages and to from the terminal and provides some basic functionality related to security (integrity protection, key update) ensures that the content of the message have not changed since its creation, user authentication, checksums of SIP request initiated by the IMS terminal. Additionally, it provides compression/decompression of SIP messages, generates charging information, sends a charging collection node and may include a PDF (Policy Decision Functions, which authorizes media plane resources and manages QoS over the media plane). An IMS network includes a number of P-CSCF for scalability and redundancy reason (each P-CSCF serves a number of IMS terminals). The P-CSCF is always located in the same network where the GGSN is located. It is expected that the first IMS networks will have GGSN and P-CSCF in the same network.

### Abbreviations (4)

RNC  Radio Network Controller

SDP  Session Description Protocol (textual format to describe multimedia sessions). It includes e.g. the IP address of the requestor, the port number where the requestor expects to receive audio and video, and the list of audio and video codecs supported.

SGSN  Service GPRS Support Node, responsible for mobility management, security and authorization.

SSG  Signalling Gateway, interfacing the signalling plane of the PSTN or GSM and performing lower layer protocol conversion (ISUP to SIP over IP and vice versa) and control of resources further to the S-CSCF in the home network.

SIP  Session Initiation Protocol (defined in RFC 3261 by the Internet Engineering Task Force in June 2002) is based on HTTP (Hypertext Transfer Protocol) and purposes for basic call control and application signalling for voice and multimedia calls or sessions in a packet switched network. It benefits from simplicity, scalability, robustness, flexibility, and extensibility and has the following values:
- Line (in case of a SIP Request called Request Line, in case of a SIP response called Status Line)
- empty line
- optional Message-Body (a set of Header Fields provide information about the message Body). The Message Body provides e.g. the SDP:
  - Target peers include details of the SDP protocol version used, in case of a Secure Line it includes a status code and a reason phrase (e.g. SIP/2.0 401 Unauthorized). In case of a Request Line, it also includes a method name and a request URI (e.g. INVITE sip:John_Doe@server.com SIP/2.0).
  - Status codes range from 100-199 (Provisional or informational), 200-299 (Success), 300-399 (Redirection), 400-499 (Client Error), 500-599 (Server Error), and 600-699 (Global Failure).
  - Method names are e.g. OPTIONS (probes the establishment of a session), INVITE (establishes a session), CANCEL (terminates a session), BYE (terminates a session), MAX-forwards (acknowledges the receipt of a provisional response), etc.

As SIP is not an efficient protocol regarding message size (because its textual based), users would ton bandwidth access (e.g. radio access) needs to minimize the amount of data transmitted over the access network. For this reason, header compression will be applied for SIP messages exchanged between the UE and the P-CSCF.

25.02.2007 Call setup in IMS V.2
4. ABBREVIATIONS (5)

SIP (part)

SIP extension: The SIP core protocol is relatively simple and encourages future extensions, i.e. allows system designers to subsequently add new features. SIP can be extended in at least three ways:
- defining new message body types
- defining new header types
- defining new message types

Interoperability of extensions: The base protocol includes mechanisms for extension management and rules for how to deal with unknown or unexpected extensions. Extensions are identified by a standardized token that is registered with IANA (Internet Assigned Numbers Authority). Two SIP implementations dynamically assess which extensions are supported and negotiate down to a basic level of operation.

SLF

Subscription Location Function. If a network contains more than one HSS because the number of subscribers is too high to be handled by a single HSS, the SLF maps user addresses to HSSs.

SS7

Signalling System No 7, the signalling used in circuit switched networks

S-CSCF

Serving Call/Session Control Function. It is a SIP server which performs session control and acts as a SIP registrar (mapping of the IP address of the terminal the user is logged on and the user’s SIP address, i.e. the Public User Identity) Mainly it performs routing functionality and, thus, all SIP signalling to and from the IMS terminal traverse the S-CSCF. In the case a telephone number is dialled instead of a SIP URI (Uniform Resource Identifier) the S-CSCF provides translation service based on DNS E.164 Number Translation.

The S-CSCF interfaces to the HSS for user authentication and downloading of the user profile as well as asking the HSS for mapping its address with the user for the duration of the registration. Each S-CSCF serves a number of IMS terminals. The S-CSCF is always located in the home network.

UE

User Equipment

UICC

Universal Integrated Circuit Card. A removable smart card with standardized interface and limited storage capacity used to hold subscription information, authentication keys, phone numbers, messages, etc. It may contain one or several logical applications such as SIM (Subscriber Identity Module), USIM (Universal Subscriber Identity Module) and ISIM (IP Multimedia Service Identity Module). The UICC itself refers to the physical card, whereas SIM, USIM and ISIM refer to applications stored onto the UICC.

UMTS

Universal Mobile Telecommunication System. Standardized by 3GPP, it is referred to as the third generation network, providing data rates of 2 Mb/s and supporting video and audio streaming and location based services.

Release 99 defines the basic architecture consisting of the UMTS Terrestrial Radio Access Network (UTRAN), the Circuit Switched Core Network (CS-CN), and the Packet Switched Core Network (PS-CN).

Release 4 adds new services but does not change the Release 99 architecture.

Release 5 offers both traditional telephony as well as packet switched enhanced multimedia services over a single converged packet based network, using SIP as the basic protocol and IMS as the signalling architecture.

4. ABBREVIATIONS (6)

URI

Universal Resource Identifier (similar to email addresses the URIs are needed to identify users)

USIM

Universal Subscriber Identity Module. Used to access UMTS networks, this application stored onto the UICC includes similar information than the SIM.

UTRAN

UMTS Terrestrial Radio Access Network

25.02.2007 Call setup in IMS V.2 Seite 51
APPENDIX A

SIP INVITE

sent by the calling UE

Message header

Call setup in IMS V.2

Session Description Protocol (SDP) body

APPENDIX B

SIP INVITE

received by the called UE

Appendices