IMS at mobilkom austria group

An overview

Revision history

Version	Date	Name	Change
0.0.3	5.6.2008	Günther Pospischil	Section 4 extended
0.0.2	5.6.2008	Helmut Hofstetter	Section 3.1 extended
0.0.1	30.5.2008	Günther Pospischil	Initial version

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1. Introduction

IMS is a standardized IP-based communication environment, allowing the development of any kind of multimedia communication services, including for example Voice-over-IP, instant messaging, collaboration, and gaming applications.

The IMS architecture includes user devices/clients, a core network infrastructure and various enablers, like a presence server. The core network infrastructure is standardised by 3GPP [1] and IETF [2], while enablers are mainly standardized by OMA [3]. A good overview is provided in Tech-Invite [4] and Wikipedia [5].

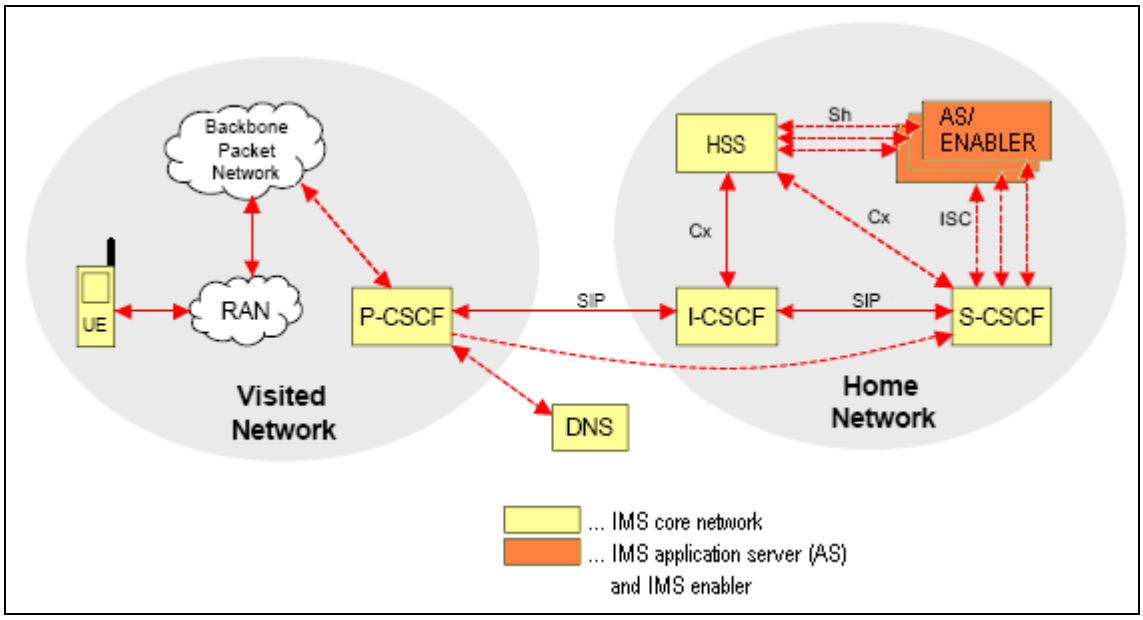

Figure 1: IMS architecture.

For communication in IMS, the SIP protocol [6] is used. It is quite similar to the HTTP protocol, but designed for client to client communication (whereas HTTP is used between a client and a server). For specific tasks, like group management related to presence, additional protocols may be used. More details can be found in section 3.4.

Addressing in IMS is based on the URI format, commonly used in the internet, using a protocol identifier and the actual address. In the internet, it is common to use `http://server_address`, `https://server_address`, or `mailto:mailaccount@mailserver`.

In IMS, users can be identified in two ways:

- For pure IMS communication, an alphanumeric SIP identifier, similar to an email address, is used.
  Example: `sip:testuser@a1.net` (SIP URI).
- The existing phone number can be used (typically when a user has an IMS subscription and a conventional mobile telephony subscription in parallel and interworking between both networks is required).
  Example: `tel:1234567` (TEL URI) or `sip:1234567@a1.net` (SIP URI).

Before IMS can be used, a registration process is required. Usually clients automatically register at the IMS infrastructure by sending the appropriate registration credentials (user ID and password). For details see section 3.1.

Applications in IMS can either be developed on the client side or on top of an application server (see section 4 for details):

- Client sided applications are extremely simple. It is sufficient to take a SIP stack and build the required business logic around it.
- Server based applications can be more powerful, but they are also more complex to implement. Application development is typically done on top of dedicated application servers, for instance based on SIP servlet technology [7]. Specific configuration of the IMS environment is needed to integrate the application servers (and forward the appropriate SIP requests to the application servers).

Most of the described functionality is provided by the service "A1 over IP", i.e. the internet telephony service of mobilkom austria. Service details can be found at http://www.a1.net/privat/a1overip.

2. Features of IMS at mobilkom austria

2.1. General

- Login at IMS via SIP ID and password (SIP REGISTER).
- Alphanumeric SIP URI for communication between PC clients (free).
- Phone number for interworking between PC client and mobile phone (charging according to A1 price plan).
- Interworking with external SIP communities, e.g. Sipgate (www.sipgate.at), Voipbuster (www.voipbuster.com)[1].
- Automatic client configuration (for clients provided by A1).

2.2. Telephony

- Voice telephony (VoIP) and video telephony between PC clients.
- Voice telephony between PC and phone.
- Parallel ringing of phone and PC client (if Multiring is activated).
- Voice conference between 3 parties (client based), including PC clients and phones.
- Video conference between 3 parties (client based), currently only between PC clients.
- Alphanumeric SIP URI for communication between PC clients (free).

2.3. Messaging

- Instant messaging (IM) between PC clients.
- Messaging between PC client and mobile phone (Instant message is forwarded as SMS).
- Messaging between mobile phone and PC client (forwarding SMS as instant message, currently only as response to PC client originated message).
- SMS follow me: Instant message to PC client that is offline is forwarded as SMS to the mobile phone.

2.4. Presence

- Presence status of individual users (on their PC clients) with access control (user-defined privacy settings).
- Presence status of user groups/resource lists.

2.5. Contact management

- Presence status of individual users (on their PC clients) with access control (user-defined
- Central VoIP phone book, currently accessible via HTML/HTTP.

2.6. Planned extensions

- Video telephony between PC and mobile phone.
- Messaging between mobile phone and PC client (also if no message was initially sent from the PC client).

[1] Voice interworking using G.711 codec works with most external SIP providers. Other features (video telephony, presence) may not work in all cases due to different interpretations of standards or blocking by other providers.

3. Using IMS at mobilkom austria

3.1. General

- Registration of clients at IMS:
  User: A1.net user identifier
  Note: User is case sensitive and must be entered without domain (i.e. without @a1.net).
  Password: A1.net password
  Domain: a1.net
  Outbound proxy: sip.a1.net

  Example:

  User: `testuser`
  Password: `testpwd`
  Domain: `a1.net`
  Outbound proxy: `sip.a1.net`

- Firewall settings:

  Allow * to 80.75.55.109 port 5060 udp, tcp
  Allow * from 80.75.55.109 port 5060 udp, tcp
  Allow * to 80.75.55.93 port 50000-50500 udp
  Allow * from 80.75.55.93 port 50000-50500 udp

- Supported SIP methods:

SIP Method	Reference
ACK request	[RFC3261]
BYE request	[RFC3261]
CANCEL request	[RFC3261]
INVITE request	[RFC3261]
MESSAGE request	[RFC3428]
NOTIFY request	[RFC3265]
OPTIONS request	[RFC3261]
PRACK request	[RFC3262]
PUBLISH request	[RFC3903]
REFER request	[RFC3515]
REGISTER request	[RFC3261]
SUBSCRIBE request	[RFC3265]
UPDATE request	[RFC3311]

- Supported multimedia protocols (SDP types):
  The used multimedia protocol depends on the negotiation between the two involved clients and no protocols are blocked by the core. However, there is an average rate limit set for the following codecs: G723 (15kbit/s), G729 (26kbit/s), PCMU (80kbit/s), and PCMA (80kbit/s) which must not be exceeded.

3.2. Telephony

- Supported codecs for calls between PC clients: Any codec is allowed, successful call establishment depends only on a common codec between both involved clients. It is recommended to include widespread/free codecs, like G.711 (a-law or μ-law) and iLBC for voice, and H.263 for video.
- Supported codecs for calls between PC client and mobile phone: G.711 (a-law and μ-law), G.729 (plain plus annexes A and B).

3.3. Messaging

- Messaging between PC client and mobile phone: Important is the use of the appropriate SIP method `MESSAGE` and the correct content type `text/plain; charset=UTF-8` [11][12][13].

```
MESSAGE sip:436646000000@a1.net SIP/2.0
Via: SIP/2.0/UDP 10.241.131.64:5052;branch=z9hG4bK-d87543-
8a00c77fb24f2025-1--d87543-;rport
Max-Forwards: 70
From: "testuser1"<sip:testuser1@a1.net>;tag=1
To: "testuser2"<sip:+436641234567@a1.net>
Call-ID: 1-6176@10.247.128.17
Cseq: 1 MESSAGE
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE,
SUBSCRIBE, INFO
User-Agent: eyeBeam release 1003s stamp 31159
c: text/plain; charset=UTF-8
Content-Length:   40

This is a test SMS over SIP using sipp
```

3.4. Presence

- Initially peer-2-peer presence is supported, i.e. presence information is exchanged directly between two clients. The presence document structure is described below.

online:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:unknown/>
    </rpid:activities>
</dm:person>

<tuple id='t2f5f9c51'>
    <status>
        <basic>
            open
        </basic>
    </status>
</tuple>
```

busy:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:busy/>
    </rpid:activities>
    <dm:note>
        Beschäftigt
     </dm:note>
</dm:person>
<tuple id='t2f5f9c51'>
    <status>
        <basic>
            open
        </basic>
    </status>
</tuple>
```

on-the-phone:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:on-the-phone/>
    </rpid:activities>
    <dm:note>
        Telefoniert
    </dm:note>
</dm:person>
<tuple id='t2f5f9c51'>
    <status>
        <basic>
            open
        </basic>
    </status>
</tuple>
```

inactive:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:unknown/>
    </rpid:activities>
    <dm:note>
        Inaktiv
    </dm:note>
</dm:person>
<tuple id='t2f5f9c51'>
    <status>
        <basic>
            open
        </basic>
    </status>
</tuple>
```

away:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:away/>
    </rpid:activities>
    <dm:note>
        Abwesend
    </dm:note>
</dm:person>
<tuple id='t2f5f9c51'>
    <status>
        <basic>
            open
        </basic>
    </status>
</tuple>
```

offline:

```
<dm:person id='p125f5027'>
    <rpid:activities>
        <rpid:unknown/>
    </rpid:activities>
</dm:person>
<tuple id='t2f5f9c51'>
    <status>
        <basic>
            closed
        </basic>
    </status>
</tuple>
```

- It is already planned to establish a presence server infrastructure. In this case, presence information will be handled via a central presence server, together with a XDMS/RLS server (storing contact information, presence/resource lists, and privacy settings. The necessary SIP methods and presence documents are described below.

Used SIP methods:
- SIP SUBSCRIBE
- SIP NOTIFY
- SIP PUBLISH

XDMS communication via XCAP:
- HTTP PUT
- HTTP GET

The following standards are supported for Presence:
- RFC3856
- RFC3857
- draft-ietf-simple-partial-publish
- draft-ietf-simple-partial-notify
- RFC4745
- draft-ietf-simple-presence-rules
- RFC3863
- draft-ietf-simple-partial-pidf-format
- RFC4662
- RFC4479
- RFC4480
- RFC4481
- RFC4482
- RFC4660
- RFC4661
- OMA Presence SIMPLE V1.0.1 Technical Specification
- OMA Presence SIMPLE V1.0.1 Architecture Document

The following standards are supported for XDMS/RLS:
- RFC4825
- draft-ietf-simple-xcap-diff
- draft-ietf-simple-xml-patch-ops
- draft-ietf-simple-presence-rules
- RFC4745
- RFC4826
- OMA XML Document Management Core V1.0.1 Technical Specification
- OMA XML Document Management Shared V1.0.1 Technical Specification
- OMA XML Document Management V1.0.1 Architecture Document
- OMA Presence SIMPLE XDM V1.0.1 Technical Specification
- OMA Presence SIMPLE RLS XDM V1.0.1 Technical Specification

- The SIP communication is in all cases routed over the Session Border Controller, there is no need for special presence settings in the SIP stack. The XDMS/RLS server will be accessible by using an appropriate DNS name/public IP address that mobilkom austria will provide as soon as the system is established.
- Note: Clients should support both presence methods to be future proof!

3.5. Contact management

- The VoIP phone book can be accessed via browser: http://www.a1.net/a1overiptelefonbuch/
- Currently there is no public API to the VoIP phone book, but in the future it is planned to integrate the VoIP phone book with the existing A1 address book (see http://www.a1.net/adressbuch and http://www.a1.net/privat/adressbuchinfo). The integrated address book will provide a SOAP API for application access (plus an LDAP interface for fast read access).

4. Application development guidelines

The IMS architecture includes several components (see Figure 2):

- The user accesses the system either with soft clients running on a PC or a mobile phone, or via device clients (SIP phones).
- Clients connect to IMS via some kind of IP access network, e.g. ADSL, UMTS, HSDPA.
- To protect the IMS Core from unwanted traffic and to allow NAT/firewall traversal, an Access Session Border Controller (A-SBC) is used. From a high level perspective, this device is similar to a firewall.
- The IMS Core consists of several Call Session Control Function (CSCF) nodes and the Subscriber Repositories, including User Mobility Server/Home Subscriber Server (UMS/HSS) and the Domain Name Server/Electronic Number Mapping Server (DNS/ENUM). The DNS/ENUM is used to match SIP IDs and phone numbers. The UMS/HSS stores the service profiles for a subscriber, defining which applications shall be included during session establishment, which is handled by the CSCF.
- For enhanced IMS services, dedicated Enablers and SIP Application Servers (SIP AS) are connected to the IMS Core.
- IMS also allows interworking with classical mobile phones, connected to the classical network elements, like Mobile Switching Center (MSC) and Short Message Center (MSC). To enable this kind of interworking, protocol conversion between SIP and the classical Signaling System Number 7/Mobile Application Part (SS7/MAP) protocols is required. SMS/IM interworking is achieved by a Short Message Service Gateways (SMS-GW), voice call interworking between VoIP and classical voice is done via Media Gateway Controller/Media Gateway (MGC/MGW).
- Interworking with other SIP/IMS providers in the internet is enabled by the Network Session Border Controller (N-SBC).

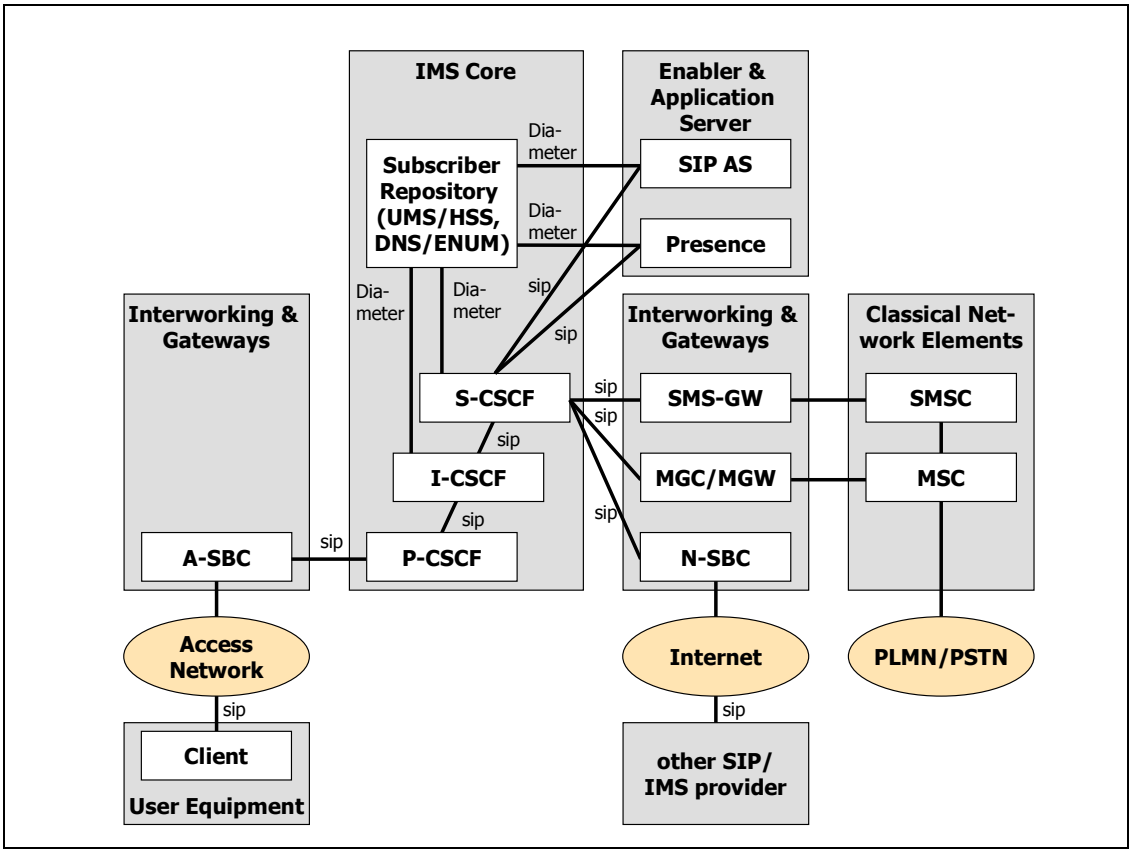

Figure 2: IMS architecture components.

IMS services can be purely client based, like pure client-to-client Instant Messaging, or involve application servers.

4.1. Client-side developments

- Possible functionality: Send all required SIP methods. Receive different SIP methods, directly addressed to the client. Establish media sessions to other clients.
- Restrictions: Obviously client-side logic is only executed if the client is addressed with it's specific SIP address.
- Example:
    - A SIP Client can be configured to forward a call to a voice mail system if the user does not click "accept call" within 30 seconds.
    - A1 Shoutbox, a SIP enabled application which displays SMS/Instant Messages for events ("SMS wall"), which was developed by ftw.
- Tested toolkits: pjSIP [8].
- Testing: Obtain the necessary number of A1 over IP accounts (free or commercial service), register the clients and activate the required features.

4.2. Application server (AS) based developments

- Possible functionality: Manipulate SIP requests (redirect, cancel, duplicate/fork), even if they are not addressed to the AS (but rather to another user).
- Restrictions: The IMS Core needs to be configured specifically for each service running on top of an AS (to send the appropriate SIP methods to the AS).
- Example:
    - A SIP AS can monitor all calls to a specific user. If the user is not registered at all or if the user does not pick up an incoming call within 30 seconds, the call may be forwarded to a voice mail system.
- Tested toolkits: BEA Weblogic SIP Server (BEA WLSS) [9].
- Testing: The BEA WLSS includes a simplified SIP/IMS environment for testing. Additionally testing can be done by downloading and installing "IMS in a bottle" [10].

5. Further information

- Lucent/ITU IMS overview:
  http://www.itu.int/ITU-T/worksem/ngntech/presentations/s1-towle.pdf
- Tech-invite:
  http://www.tech-invite.com
- Fraunhofer-Fokus IMS Lab:
  http://www.fokus.fraunhofer.de/ims/
  http://www.fokus.fraunhofer.de/bereichsseiten/testbeds/ims_playground/at_a_glance/at_a_glance.php?lang=de
  http://www.fokus.fraunhofer.de/ims/opensoaplayground/?lang=de
- ftw. IMS tutorial:
  http://www.ftw.at/ftw/events/tutorials%20a.%20workshops/MemberTutorialFolder/ims050331

6. References

[1] 3GPP Service requirements for the Internet Protocol (IP) multimedia core network subsystem (IMS); Stage 1
http://www.3gpp.org/ftp/Specs/html-info/22228.htm

[2] IETF SIP working group
http://www.ietf.org/html.charters/sip-charter.html

[3] IP Multimedia Subsystem in OMA (IMS in OMA) V1.0
http://www.openmobilealliance.org/technical/release_program/ims_v1_0.aspx

[4] Tech-invite IMS overview
http://www.tech-invite.com

[5] Wikipedia IMS overview
http://en.wikipedia.org/wiki/IP_Multimedia_Subsystem

[6] SIP protocol
http://www.rfc-archive.org/getrfc.php?rfc=3261

[7] SIP servlet specification version 1.0
http://www.jcp.org/en/jsr/detail?id=116

[8] pjSIP
http://www.pjsip.org/

[9] BEA Weblogic SIP Server
http://www.bea.com/sip/
(free evaluation version available)

[10] "IMS in a bottle" (adapted OSIMS Platform)
http://www.a1.net/a1innovations/download/IMS-in-a-Bottle.rar
http://www.openimscore.org

[11] RFC 3428- Session Initiation Protocol (SIP) Extension for Instant Messaging
http://www.rfc-archive.org/getrfc.php?rfc=3428

[12] RFC 3325 - Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks http://www.rfc-archive.org/getrfc.php?rfc=3325

[13] MIME Content-Type 'application/vnd.3gpp.sms'
http://www.iana.org/assignments/media-types/application/vnd.3gpp.sms