## Appendix A

### List of Abbreviations with Full form

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<tr>
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Appendix B

Testing results for IMS Client

In the square bracket "[ ]" show the sub-clause in 3GPP TS 24.229 Release 6 specification.

• Evaluation at the UE

IMS client

[5.1.1] Registration and authentication

[5.1.1.2] Initial registration

On sending a REGISTER request, the UE populate the header fields with an Authorization header, a From header, a To header, a Contact header, a Via header, an Expires header, a Request-URI, a Supported header that accord with the 3GPP TS 24.229 completely. However, there are some parts are not consistent with the standards.

• The UE suppose to associate two parts, a protected client port and a protected server port, but in our situation, we only find the protected server port without association

• We have no Security-Client header

• There is no P-Access-Network-Info header.

On receiving the 200(OK) response to the REGISTER request, as showed in the table, our situation accord with most of the standards, but there are still some differences:

• There is no P-Associated-URI header.

• There is no security association lifetime shows.

When a 401 (Unauthorized) response to a REGISTER is received the UE is barely behave as the standards says. Except that derive the keys CK and IK as described in 3GPP TS 32.203, there is no temporary set of security associations has been set up, no Security-Client header and there is no Authorization header.

SIP client

On sending a REGISTER request, the UE populate the header fields contain an Authorization header, a From header set to the SIP URI containing the public user identity, a To header set to the SIP URI containing the public user identity to be registered, and a Contact header, a Via header, and a Request-URI that are consistent to the standards. But there is no security-client header, no P-Access-Network-Info header, and no Supported header. Besides, the expire parameter in the Contact header set to the value 3600 but
not 600,000 seconds.

On receiving the 200(OK) response to the REGISTER request, as showed in the table, our situation accord with most of the standards, but there are still some differences:

- The UE doesn't store the expiration time of the registration.
- There is no P-Associated-URI header.
- There is no security association lifetime shows.

[5.1.1.3] Initial subscription to the registration-state event package IMS client

On sending a SUBSCRIBER request, the UE populate the header fields with a Request URI set to a SIP URI that contains the public user identity used for subscription, and a From header, a To header, an Event header, an Expires header, a Contact header that are consistent as the standards described. The only difference is that there is no P-Access-Network-Info header.

Upon receipt of a 200 response to the SUBSCRIBE request, the UE stores the information for the established dialog and the expiration time as indicated in the Expires header of the received response.

SIP client

On sending a SUBSCRIBER request, when UE populate the header fields, there are some parts are not consistent of the standards.

- There is no P-Access-Network-Info header
- The Event header set to "message-summary" instead "reg".
- The Expires time set to "300", but not "600000".

2xx response never reached UE, it only forward to the P-CSCF.

[5.1.1.5] Authentication

[5.1.1.5.1] General

The 401 situation is already discussed in 5.1.1.2.

On receiving the 200(OK) response for the protected REGISTER request, for both SIP client and IMS client, there is no security association provided.

[5.1.1.5.2] Network-initiated re-authentication

Since there is no timer F expires at the UE, so we don't consider this situation that described in this sub-clause.

[5.1.1.6] User-initiated deregistration

IMS client
On sending a REGISTER request, the UE populate the header fields with an Authentication header, a From header, a To header, a Contact header, a Via header, an Expires header, and a Request-URI that accord with the description in 3GPP TS 24.229 completely. The differences are: -There is no Security-Client header. -There is no Security-Verify header. -There is no P-Access-Network-Info header.

On receiving the 200 (OK) responses to the REGISTER request, the UE removed all the registration details relating to the public user identity. And since there are no more public user identities registered, the UE deleted the related keys that may towards to the IM CN subsystem.

**SIP client**

The X-Lite does not support deregistration.

**[5.1.1.7] Network-initiated deregistration**

Upon receiving the NOTIFY request on the dialog which was generated during subscription to the reg event package, the UE contains a <registration> element with the state attribute set to "terminated". But the event attribute is a little different from the standards: it is set to "unregistered" but not to "rejected" or "deactivated".

**[5.1.2] Subscription and notification**

**[5.1.2.1] Notification about multiple registered public user identities**

**[5.1.2.2] General SUBSCRIBER requirements**

The UE doesn't receive a 503 response, so we don't need to consider what described in this sub-clause.

**[5.1.3] Call initiation-mobile originating case**

**[5.1.3.1] Initial INVITE request**

For both SIP client and IMS client, our situation is the originating UE does not require local resource reservation.

Upon generating an initial INVITE request, the UE indicates the support for reliable provisional response and the support for the preconditions mechanism by using the Supported header. And it doesn't indicate the requirement for the precondition mechanism by using the Require header mechanism.

- **Evaluation at the P-CSCF**

Generally speaking, the functionality of the P-CSCF is conformant to the specification of 3GPP R6.

**[5.2.1] General**

As the description of 3GPP TS 24.229, the P-CSCF of OpenIMS support the
Path and Service-Route headers, and the Path header is only used in the REGISTER request and its 200 (OK) response, while the Service-Route header is only applicable to the 200 (OK) response of REGISTER request.

The difference in our case is: there is not P-Charging-Function-Addresses header. Therefore, the functionality of P-CSCF with P-Charging-Function-Addresses header is not considered.

The other difference is without P-Media-Authorization header in our case, because what we concentrate on is just OpenIMS Core, which the AS is not included.

Both IMS Client and SIP Client get the same situation.

[5.2.2] Registration

In the registration, the P-CSCF is preparing to receive only the initial REGISTER requests on the SIP default port values or on the port advertised to the UE during the P-CSCF discovery procedure.

Most procedures in registration are conformant with TS 24.229. But, we don't consider the security, so, the REGISTER request is not protected. And the parameter "integrity-protected" is inserted with the value "no".

Although the REGISTER request is not protected in our cases, the Security-Client header is not existed. The reason is that, the architecture of the OpenIMS Core in our case is too simple to include the security, because all the components are fixed in a single domain.

For the state that P-CSCF receives a 401 (Unauthorized) response to a REGISTER request, the P-CSCF perform almost the same as the specification, but we could not evaluate the security around it, because there are not security associations, Security-Server, reg-await-auth timer in our case.

For the state that P-CSCF receives a 200 (OK) response to a REGISTER request, some of the functionality is different. At first, there is no Contact header can be checked. And then, there is no P-Asserted-Identity header. Next difference is P-CSCF cannot store the values received in the P-Charging-Function-Address header for the reason that in our case, there is no P-Charging-Function-Address header. The last difference is a term-oi parameter is not received in the P-Charging-Vector header, the security association is not considered.

[5.2.3] Subscription to the user’s registration-state event package

For the situation that upon receipt of a 200 (OK) response to the initial REGISTER request, the different cases for P-CSCF performs as following.

The P-CSCF will generate a SUBSCRIBE request but the From header is not set to the P-CSCF's SIP URI. It set as: sip:alice@open-ims.test which is a Public User Identity's SIP URI. And the Expires header is still set to 600000 which is the same as the Expires header indicated in the 200 (OK) response to the REGISTER request.
[5.2.5] Deregistration

For the SIP Client, it doesn't support the functionality of deregistration. For the IMS Client, there are some functionalities of deregistration are different from the specification.

[5.2.5.1] User-initiated deregistration

When the P-CSCF receives a 200 (OK) response to a REGISTER request sent by the UE, the Expires header will be checked, in the situation that the expires parameter equal zero, the difference for the P-CSCF of OpenIMS does not remove the Public User Identity found in the To header field.

[5.2.6] General treatment for all dialogs and standalone transactions excluding the REGISTER method

[5.2.6.3] Requests initiated by the UE

When the P-CSCF receives an initial request for a dialog or a request for a standalone transaction, the request of IMS client contains a P-Preferred-Identity header, so the P-CSCF shall identify the initiator of the request by that public user identity. As to the SIP client, the situation is different. The request of SIP client doesn't contain a P-Preferred-Identity header, so, the P-CSCF shall identify the initiator of the request by a default public user identity.

There is no Service-Route header in our situation, therefore, we don't consider the related cases.

Both of the IMS and SIP client add its own address to the Via header which the situation is conformant to the specifications.

When the P-CSCF receives a 1xx or 2xx response to the before request, the P-CSCF shall not store the values received in the P-Charging-Function-Address header, cause we don't have this header in our cases.

[5.2.6.4] Request terminated by the UE

When adding P-CSCF’s own SIP URI to the top of the list of Record-Route headers and save the list, the P-CSCF build the P-CSCF SIP URI in a format that contains the report parameter is not conformant to the specifications.

In the situation that P-CSCF receives a 1xx or 2xx response to the request, the P-CSCF performs mostly conformant to the specification. But the case is different for SIP client and IMS client when P-CSCF verifies the list of URIs received in the Record-Route header.

[5.2.7] Initial INVITE

[5.2.7.1] Mobile-originating case

When the P-CSCF receives from the UE an INVITE request, the P-CSCF shall respond to all INVITE requests with a 100 (Trying) provisional response that is conformant to the specification. But the P-CSCF doesn't insert
the P-Media-Authentication header containing that media authorization token.

5.2.8 Call release
5.2.8.1.2 Release of an existing session
The situation is conformant to the specification, but it is different from IMS client to SIP client here. For IMS client, the P-CSCF serves the calling user of the session it shall generate a BYE request based on the information saved for the related dialog. And for SIP client, the P-CSCF serves the called user of the session it shall generate a BYE request based on the information saved for the related dialog.

And we don't consider the situation about security association.

• Evaluation at the I-CSCF
[5.3.1] Registration procedure
Generally speaking, the I-CSCF behaves as a stateful proxy during the registration procedure.
[5.3.1.2] Normal procedures
The I-CSCF decides which HSS to query, and possibly as a result of a query to the Subscription Locator Functional (SLF) entity. But in the OpenIMS Core, the SLF is not included.
[5.3.2] Initial requests
The I-CSCF behaves as a stateful proxy for initial requests.
[5.3.2.1] Normal procedures
All components in our situation are in a signal domain, therefore, we don't consider the IP connective access network. That's the reason why we don't have P-Access-Network-Info headers.

Besides, as the same reason, we can not see the procedures about I-CSCF shown in the Wireshark log messages.

There is a situation is different on IMS Client and SIP Client:
When the I-CSCF receives an initial request for a dialog or standalone transaction, we trace the log messages about IMS Client, and found that, the I-CSCF remove its own SIP URI from the topmost Route header, and route the request based on the Request-URI header field. While the trace on SIP Client, the situation is different. I-CSCF contains more than one Route header, and I-CSCF at first remove its own SIP URI from the topmost Route header, and then forwarding the request based on the topmost Route header.

[5.3.3] THIG functionality in the I-CSCF
We don't consider the situation about THIG, as the reason that the visited network and the home network are the same in our case.
• Evaluation at the S-CSCF

[5.4.1] Registration and authentication

[5.4.1.1] Introduction

The S-CSCF acts as the SIP registrar for UA belonging to the IM CN subsystem.

IMS client

For IMS client situation, the S-CSCF supports the Path header, the Service-Router header, the Require header, and also the Supported header. But it still cannot accord with the standards completely. Because according to the standard, the Path header should only applicable to the REGISTER request and its 200OK, and the Service-Router header should only applicable to the 200OK of REGISTER, but in our situation, both of the header also appears when S-CSCF receiving the "401 Unauthorized-Challenging the UE".

SIP client:

In accordance with the 3GPP TS 24.229, the S-CSCF supports the Path header (only applicable to the REGISTER request and its 200OK), the Service-Router header (only applicable to the 200OK response of REGISTER), and also support the Require header. However, it does not support the Supported header.

[5.4.1.2] Initial registration and user-initiated reregistration

[5.1.1.2.1] Unprotected REGISTER

As says in NOTE 2, if a REGISTER request with Expires header value equal to zero should always be received protected, but for both SIP client and IMS client, the Expires header value are not equal to zero, so our REGISTER request is unprotected.

IMS client

When receiving a REGISTER request with the "integrity-protected" parameter set to "no", the IMS client accord with the standards better than SIP client. Except the timer reg-await-auth haven't been started, others are consistent to 3GPP TS 24.229.

SIP client:

Upon receipt of a REGISTER request without the "integrity-protected" parameter, the S-CSCF behave almost as the standards says, but there is no IK, CK parameters in the WWW-Authenticate header, and because is SIP client, so the security mechanism is MD5 but no AKA V1-MD5. Besides, in
normal case, the S-CSCF doesn't start the timer reg-await-auth.

[5.4.1.2.2] Protected REGISTER
Since our REGISTER request is unprotected, so we don't consider this sub-clause.

[5.4.1.3] Authentication and re-authentication
This situation we already discussed in 5.4.1.2.

[5.4.1.4] User-initiated deregistration IMS client
Since the "integrity-protected" parameter in Authorization header set to "no", according to the standard, S-CSCF apply the procedures described in sub-clause 5.4.1.2.1

SIP client
X-Lite cannot been deregistered by user.

[5.4.2] Subscription and notification
[5.4.2.1] Subscriptions to S-CSCF events
[5.4.2.1.1] Subscription to the event providing registration state
When an incoming SUBSCRIBE request addressed to S-CSCF arrives containing the Event header with the reg event package, the S-CSCF shall check if a subscriber who is authorized to subscribe to the registration state of this particular user generated the request. For both SIP client and IMS client, the S-CSCF can find the identity for authentication of the subscription in the P-Asserted-Identity header received in the SUBSCRIBE request. And the S-CSCF stores the value of the orig-ioi parameter received in the P-Charging-Vector header.

IMS client
When generate a 200 response to the SUBSCRIBE request, the S-CSCF populate an Expires header set to the same value as the Expires header in SUBSCRIBE request which is accord with the standards.

SIP client
When generate a 200 response to the SUBSCRIBE request, the S-CSCF populates an Expires header set to a value that is higher than the Expires header in SUBSCRIBE request, this is the opposite as described in the 3GPP TS 24.229.
[5.4.2.1.2] Notification about registration state

IMS client
For each NOTIFY on all dialogs which have been established due to subscription to the reg event package of the user, the S-CSCF set the Request-URI and Router header to the saved route information during subscription, and set the Event header to the "reg". In the body of the NOTIFY request contains a <registration> elements and for each <registration> element, the S-CSCF set the or attribute to one public user identity, and set the<uri> sub-element inside the <contact> sub-element of the <registration> element to the contact address. Under this situation, if the public user identity has been deregistered, then S-CSCF sets the state attribute in the <registration> element to "terminated", sets the state attribute in the <contact> element to "terminated" and set the event attribute in the <contact> element to "unregistered".

However, there is no P-Charging-Vector header for the NOTIFY request which is different as the standard says.

SIP client
For SIP client X-Lite, we got "487 Event Package Not Supported".

[5.4.3] General treatment for all dialogs and standalone transactions excluding requests terminated by the S-CSCF

[5.4.3.1] Determination of mobile-originated or mobile-terminated cases
For both IMS client and SIP client, upon receipt of an initial request or a target refresh request or a stand-alone transaction, the S-CSCF perform the procedures for the mobile-originating case as described in 3GPP TS 24.229 sub-clause 5.4.3.2, and the S-CSCF remove the "orig" parameter from the topmost Route header.

[5.4.3.2] Requests initiated by the served user IMS client
When S-CSCF receives an initial request for a dialog or a request for a standalone transaction from the served user, the S-CSCF first determines whether the request contains a barred public user identity in the P-Accessed-Identity header field of the request or not. For our situation, there is non-barred public user identity.

Our example accord with most of the situations as described in standards, but there are still some differences:

- The S-CSCF stores the value of the orig-ioi parameter received in the P-Charging-Vector header, but it doesn't remove it from the forwarded request.
- The S-CSCF doesn't insert a P-Charging-Function-Addresses header and have no knowledge that the SIP URI contained in the received P-Asserted-
Identity header is an alias SIP URI for a tel URI (We didn’t use tel URI).

- Since the networking is not needed, so the S-CSCF doesn’t put the address of the I-CSCF to the topmost route header.
- The S-CSCF doesn’t remove the P-Access-Network-Info header based on the destination user (Request-URI) or when it receives a target refresh request from the served user.
- There is no access-network-charging-info parameter in the P-Charging-Vector header field.

**SIP client**

Almost all the situations are have the same result as IMS client example except that there is no original dialog identifier that the S-CSCF previously placed in a Router header is present in the topmost Route header of the incoming request.

[5.4.3.4] Original dialog identifier

As described before, our SIP client example doesn't show the original dialog identifier.

[5.4.4] Call initiation

[5.4.4.1] Initial INVITE

For both SIP client and IMS client, when the S-CSCF receives an INVITE request, the S-CSCF processes the initial INVITE request without examining the SDP.

[5.4.4.2] Subsequent requests

[5.4.4.2.1] Mobile-originating cases

According to the 3GPP TS 24.229, when the S-CSCF receives 1xx or 2xx response, the S-CSCF shall insert a P-Charging-Function-Addresses header and store the access-network-charging-info parameter in it when receiving the request containing the access-network-charging-info parameter in the P-Charging-Vector. But in our situation, for both SIP client and IMS client. The S-CSCF doesn’t insert the P-Charging-Vector header.

When the S-CSCF receives any request or response (excluding ACK requests and CANCEL requests and responses) related to a mobile-originated dialog or standalone transaction, the S-CSCF may insert save value into P-Charging-Vector and P-Charging-Function-Addresses headers before forwarding the message within the S-CSCF home network, however in our testing, the S-CSCF didn’t insert it.

[5.4.4.2.2] Mobile-terminating case

For both SIP client and IMS client, our situation is not consistent to the standards. When S-CSCF receives the any 1xx or 2xx response, the S-CSCF
doesn't insert te P-Charging-Function-Addresses header, and when the S-CSCF receives 180(Ringing) or 200OK(to INVITE) response, the response are not contain the access-network-charging-info parameter, and not contain the P-Charging-Vector.

• **Evaluation at the Cx**

In the square bracket "[ ]" show the sub-clause in 3GPP TS 24.229 Release 6 specification.

[6] **Diameter application for Cx interface**

[6.1] **Command-Code values**

In our situation, there are several commands appear which are User-Authorization-Request (UAR), User-Authorization-Answer (UAA), Server-Assignment-Request (SAR), Server-Assignment-Answer (SAA), Location-Info-Request (LIR), Location-Info-Answer (LIA), Multimedia-Auth-Request (MAR), Multimedia-Auth-Answer (MAA). For both IMS client and SIP client, our examples are mostly accord with the 3GPP TS 29.229. We have all the mandatory AVPs and most optional AVPS in those commands. However, there are no "Registration-Termination-Request (RTR)" , "Registration-Termination-Answer (RTA)" , "Push-Profile-Request (PPR)" and "Push-Profile-Answer" commands in our examples.

[6.2] **Result-Code AVP values**

[6.2.1] **Success**

For both IMS client and SIP client in our example, there are two values stand for success that are "DIAMETER_RIRST_REGISTRATION" (2001) and "DIAMETER_SUBSEQUENT_REGISTRATION"(2002).

The "DIAMETER_RIRST_REGISTRATION"(2001) is appeared in MAA, SAA and LIA commands while the "DIAMETER_SUBSEQUENT_REGISTRATION"(2002) is appeared in UAA command during the registration process.

[6.2.2] **Permanent Failures**

When we use GXP-2000 as SIP client to register, there are "DIAMETER_ERROR_USER_UNKNOWN" (5001) stand for permanent failures. It appears in the last UAA command in the process of registration.

[6.3] **AVPS**

There are several AVPs that are showed in the table 6.3.1 of 3GPP TS 29.229 appeared in our examples. We describe them individually below.
[6.3.1] Visited-Network-Identifier AVP (600)
For both IMS client and SIP client, it appears in the UAR command, and the values is: open-ims.test.

[6.3.2] Public-Identity AVP (601)
IMS client
The Public-Identity appears in UAR, MAR, SAR and LIR commands when using IMS client to register. The value is "sip:alice@open-ims.test".
SIP client
When using X-Lite as SIP client, it appears in UAR, MAR, SAR and LIR commands and the value is "sip: user2@open-ims.test".
When using GXP-2000 as SIP client, it appears in UAR command and the value is "sip: user1@open-ims.test".

[6.3.3] Server-Name AVP (602)
When using IMS client and using X-Lite as SIP client to register, the Server-Name AVP appears in UAA, MAR, SAR and LIA commands and the value is "sip:scscf.open-ims.test:6060".
When using GXP-2000 as SIP client to register, it only appears in UAA command and the value is also "sip:scscf.open-ims.test:6060".

[6.3.7] User-Data AVP (606)
For both IMS client and X-Lite as SIP client, the User-Data AVP appears in SAA commands. When using GXP-2000 as SIP client, there is no User-Data AVP appears.

[6.3.8] SIP-Number-Auth-Items AVP (607)
For both IMS client and X-Lite as SIP client, the SIP-Number-Auth-Items AVP appears in MAR, MAA commands. When using GXP-2000 as SIP client, there is no SIP-Number-Auth-Items AVP appears.

[6.3.13] SIP-Auth-Data-Item AVP (612)
For both IMS client and X-Lite as SIP client, the SIP-Auth-Data-Items AVP appears in MAR, MAA commands. The value for IMS client is" Digest-AKAv1-MD5" while the value for X-Lite is “Digest-MD5”. When using GXP-2000 as SIP client, there is no SIP-Auth-Data-Item AVP appears.

[6.3.15] Server-Assignment-Type AVP (614)
For both IMS client and X-Lite as SIP client, the Server-Assignment-Type
AVP appears in SAR command. When using GXP-2000 as SIP client, there is no Server-Assignment-Type AVP appears.

[6.3.19] Charging-information AVP (618)
For both IMS client and X-Lite as SIP client, the Charging-information AVP appears in SAA command. When using GXP-2000 as SIP client, there is no SIP-Number-Auth-items AVP appears.

[6.3.24] User-Authorization-Type AVP (623)
Only when using GXP-2000 as SIP client to register the User-Authorization-Type appears. And the value is "REGISTRATION (0)".

[6.3.25] User-Data-Already-Available AVP (624)
For both IMS client and SIP client, it appears in SAR command.
Appendix C

Messages from call session for "client-based" solution

• (1) INVITE
  Session Initiation Protocol
  Request-Line: INVITE sip:user1@imstestbed.net SIP/2.0
  Message Header
  Via: SIP/2.0/UDP192.168.1.13:2668;branch=z9hG4bK-d87543-8409b2001c6c9c59-1 --d87543--;report Max-Forwards: 70
  Route: <sip: user2@imstestbed.net:4060;lr>
  Contact: <sip:user3@192.168.1.13:2668>
  To: "user1@imstestbed.net"<sip: user1@imstestbed.net >
  From: "user2"< user2@imstestbed.net >;tag=ce7c1c2f
  Call-ID: ZWZiZjVIMTJkM2E3ZWJkMDI5ZmUxOTZiNTM1 MzhhNDY.
  CSeq: 1 INVITE Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO
  Content-Type: application/sdp User-Agent: X-Lite release 1006e stamp 34025
  Content-Length: 325

• (10) 300 Redirect
  Session Initiation Protocol
  Status-Line: SIP/2.0 300 Redirect
  Message Header
  Via: SIP/2.0/UDP192.168.1.13:2668;branch=z9hG4bK-d87543-8409b2001c6c9c59-1 --d87543--;rport=2668
  To: "user2"< user2@imstestbed.net >;tag=b27e1a1d33761e85864fc9855f3a7e58.fc09
  From: "user2"< user2@imstestbed.net >;tag=ce7c1c2f
  Call-ID: ZWZiZjVIMTJkM2E3ZWJkMDI5ZmUxOTZiNTM1 MzhhNDY.
  CSeq: 1 INVITE
  Contact: sip: user2@imstestbed.net
  Server: Sip EXpress router (0.9.6 (i386/linux))
  Content-Length: 0
  Warning: 392 128.39.145.104:5060 "Noisy feedback tells:
  pid=12415 req_src_ip=128.39.145.250 req_src_port=51836
  in_uri=sip: user2@imstestbed.net out_uri=sip: user1@imstestbed.net
via cnt==2"

- **(11) ACK**
  Session Initiation Protocol
  Request-Line: ACK sip:user1@imstestbed.net SIP/2.0
  Message Header
  Via:SIP/2.0/UDP192.168.1.13:2668;branch=z9hG4bK-d87543-8409b2001c6c9c59-1 --d87543--rport
  Route: <sip:orig@scs2f2.open-ims.test:4060;lr>
  To: "user2@imstestbed.net" <sip:user1@imstestbed.net>; tag=b27e1a1d33761e85846fc98f5f3a7e58fc09
  From: "user2"<sip: user2@imstestbed.net>;tag=ce7c1c2f
  Call-ID: ZWZiZjVIMTJkM2E3ZWJkMDI5ZmUxOTZiNTM1MzhhNDY.
  CSeq: 1 ACK Content-Length: 0

- **(12) INVITE**
  Session Initiation Protocol
  Request-Line: INVITE sip: user2@imstestbed.net SIP/2.0
  Message Header
  Via:SIP/2.0/UDP192.168.1.13:2668;branch=z9hG4bK-d87543-5a2372669626033f1 -d87543--rport Max-Forwards: 70
  Route: <sip: user2@imstestbed.net:4060;lr>
  Contact: <sip:user3@192.168.1.13:2668>
  To: "user1@imstestbed.net" <sip: user1@imstestbed.net>
  From: "user2"<user2@imstestbed.net>;tag=ce7c1c2f
  Call-ID: ZWZiZjVIMTJkM2E3ZWJkMDI5ZmUxOTZiNTM1MzhhNDY. CSeq: 2 INVITE Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, NOTIFY, MESSAGE, SUBSCRIBE, INFO Content-Type: application/sdp User-Agent: X-Lite release 1006e stamp 34025 Content-Length: 325

- **(13) 300 Redirect**
  Session Initiation Protocol
  Status-Line: SIP/2.0 300 Redirect
  Message Header
  Via:SIP/2.0/UDP192.168.1.13:2668;branch=z9hG4bK-d87543-8409b2001c6c9c59-1--d87543--rport=2668
  To:"user1@imstestbed.net" <sip: user1@imstestbed.net>; tag=b27e1a1d33761
From: "user2"<user2@imstestbed.net>;tag=ce7c1c2f
Call-ID:ZWZiZjVIMTJkJkM2E3ZWJkJkMDI5ZmUxOTZiNTM1MzhhNDY.
CSeq: 1 INVITE
Contact: sip: user1@imstestbed.net
Server: Sip EXpress router (0.9.6 (1386/linux))

Warning: 392 128.39.145.104:5060 "Noisy feedback tells: pid=12415
  req_src_ip=128.39.145.250 req_src_port=51836
  in_uri=sip: user1@imstestbed.net out_uri=sip: user2@imstestbed.net
  via cnt==2"
Appendix D

AddUser.java file

public final class AddUser {

    public static void main(String[] args) {

        System.out.println("use hssdb;\n");

        for (int i = 1; i <= 100; i++) {

            String num = "" + i;
            int zeroesToAdd = 4 - num.length();

            System.out.println("insert into imsu(name) values ( 'alice" + num + "_imsu');\n");

            System.out.println("insert into impi(impi_string, imsu_id, imsi, scsdf_name, s_key, chrg_id, sqn) values(\n    'alice" + num + "@open-ims.test', (select imsu_id from imsu where imsu.name='alice" + num + "_imsu'), \n    'alice" + num + "_ISDN_User_part_ID', \n    'sip:scsdf2.open-ims.test:4060', \n    '616c696365000000000000000000000000', (select chrg_id from chrginfo where chrginfo.name='default_chrg'), \n    '000000000000000000');\n");

            System.out.println("insert into impu(sip_url, tel_url, svp_id) values (\n    'sip:alice" + num + "@open-ims.test','tel:00491234 " + num + "', \n    (select svp_id from svp where svp.name='default_sp'));\n");

            System.out.println("insert into impu2impi(impi_id, impu_id) values ((select impi_id from impi where impi.impi_string='alice" + num + "@open-ims.test'), (select impu_id from impu where impu.sip_url='sip:alice" + num + "@open-ims.test'));\n");

            System.out.println("insert into roam(impi_id, nw_id) values((select impi_id from impi where impi.impi_string='alice" + num + "@open-ims.test'), (select nw_id from networks where networks.network_string='open-ims.test'));\n");
        }
    }
}
}
Appendix E

XML file of REGISTER using in SIPP

<?xml version="1.0" encoding="ISO-8859-1" ?>
<!DOCTYPE scenario SYSTEM "sipp.dtd">

<scenario name="sip-to-sip call">

<send retrans="500">
<!CDATA[
REGISTER sip:open-ims.test SIP/2.0
Via: SIP/2.0/[transport] [locaUp]:[local_port]
Route: <sip:pcscf.open-ims.test:4060;lr>
Max-Forwards: 70
From: "user2" <sip: user2@imstestbed.net:4060>
To: "user2" <sip: user2@imstestbed.net:4060>
P-Access-Network-Info:3GPP-UTRAN-TDD;utran-cell-id-3gpp=C359A3913B20E
Call-ID: [call_id]
Contact: <sip:user1@[local_ip]:[local_port]>;transport=[transport]
Content-Length: 0
Supported: path
Expires: 300
CSeq: 1 REGISTER
User-Agent: Sipp v1.1 -TLS, version 20061124
]]> </send>

<recv response="401" auth="true" rtd="true">
<action>
<ereg regexp=".*" search_in="hdr" header="Service-Route" assign_to="1" />
</action>
</recv>

<send retrans="500">
<!CDATA[
REGISTER sip:open-ims.test SIP/2.0
Via: SIP/2.0/[transport] [local_ip]:[local_port]
Route: <sip:pcscf.open-ims.test:4060>;lr
Max-Forwards: 70
From: "user2" <sip: user2@imstestbed.net >
To: "user2" <sip: user2@imstestbed.net >
P-Access-Network-Info:3GPP-UTRAN-TDD;utran-cell-id-3gpp=C359A3913B20E
Call-ID:[call_id]
CSeq: 2 REGISTER
Contact: <sip:user2@[local_ip]:[local_port]>
Expires: 300
Content-Length: 0
[authentication username= user1@imstestbed.net password=12345]
Supported: path
User-Agent: Sipp v1.1 -TLS, version 20061124
]]> </send>

<recv response="200">
</recv>

</scenario>
Appendix F
XML file of INVITE using in SIPp

UAC

<?xml version="1.0" encoding="ISO-8859-1" ?>
<!DOCTYPE scenario SYSTEM "sipp.dtd">

<scenario name="sip-to-sip call">

<send retrans="500">
<![CDATA[
INVITE sip:user1@open-ims.test SIP/2.0
Via: SIP/2.0/transport [local_ip]:[local_port];branch=[branch]
Route: <sip:orig@scscf2.open-ims.test:4060;lr>
From: "user2" <sip:user2@open-ims.test:4060>;tag=[call_number]
To: "user1" <sip: user1@open-ims.test:4060>
Call-ID: [call_id]
CSeq: [cseq] INVITE
Contact: <sip: user2@[local_ip]:[local_port]>
Max-Forwards: 70
Subject: Performance Test
Content-Type: application/sdp
Content-Length: [len]

v=0
o=-02IN IP4 192.1 68.1. 3
s=
c=IN IP4 192.168.1. 3
t=00
m=audio [media_port] RTP/AVP 0
a=rtpmap:0 PCMU/8000
]]>
</send>

<recv response="100" optional="true"/>
ACK sip: user1@[local_ip]:[local_port] SIP/2.0
Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
Route: <sip:mo@pcscf.open-ims.test:4060;lr>
Route: <sip:mt@scscf.open-ims.test:6060;lr>
Route: <sip:mt@scscf.open-ims.test:6060;lr>
From: "user2"<sip: user2@open-ims.test>;tag=[call_number]
To: "user1"<sip: user2@open-ims.test>[peer_tag_param]
Call-ID: [call_id]
CSeq: [cseq] ACK
Contact: <sip: user2@[local_ip]:[local_port]>
Max-Forwards: 70
Subject: Performance Test
Content-Length: [len]
]
</send>

BYE sip: user2@[local_ip]:[local_port] SIP/2.0
Via: SIP/2.0/[transport] [local_ip]:[local_port];branch=[branch]
From: "user1"<sip: user1@open-ims.test>;tag=[call_number]
To: "user2"<sip: user2@open-ims.test>[peer_tag_param]
Call-ID: [call_id]
CSeq: [cseq] BYE
Contact: <sip: user2@[local_ip] :[local_port]>
Max-Forwards: 70
Subject: Performance Test
Content-Length: [len]
]]>
</send>

<recv response="200" crlf="true"/>
</scenario>
UAS

<?xml version="1.0" encoding="ISO-8859-1" ?>
<!DOCTYPE scenario SYSTEM "sipp.dtd">
<scenario name="uac-uas(sip-sip call), server-side">
<recv request="INVITE">
</recv>
<send> <![CDATA[ SIP/2.0 180 Ringing
[last_Via:]
[last_Record-Route:]
[last_From:]
[last_To:];tag=[call_number]
[last_Call-ID:]
[last_CSeq:]
Contact: <sip: user2@[local_ip]:[local_port]>
Content-Length: [len]
]]>
</send>

<pause milliseconds="2000"/>

<send retrans="500">
<![CDATA[
SIP/2.0 200 OK
[last_Via:]
[last_Record-Route:]
[last_From:]
[last_To:];tag=[call_number]
[last_Call-ID:]
[last_CSeq:]
Contact: <sip: user2@[local_ip]:[local_port]>
Allow:
INVITE,REGISTER,ACK,BYE,INFO,REFER,NOTIFY,SUBSCRIBE,MESSE
GE,CAN
GEL
Content-Type: application/sdp
Content-Length: [len]}}
v=0
o=- 0 2 IN IP4 [localjp]
s=-
c=IN IP4 [media_ip]
t=00
m=audio 40000 RTP/AVP 8 0 1 8
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
]]> </send>
<recv request="ACK" crlf="true">
</recv>
<recv request="BYE">
</recv>
<send>
<![CDATA[
SIP/2.0 200 OK
[last_Via:]
[last_From:]
[last_To:]
[last_Call-ID:]
[last_CSeq:]
Content-Length: 0
]]> </send>
</scenario>
Appendix G

SIP MESSAGE FLOW

The SIP message flow shown in Appendix A was adapted from the 3GPP TS24.228 Release 6.

G.1: SIP Registration Message Flow

The dotted lines in show messages and procedures to be followed based on the 3GPP specification. These dotted line functionalities are not supported by the INT IMS testbed yet, therefore they will not be implemented by this IMS Client. As the INT IMS testbed is still work-in-progress, such functionalities will be supported in future. The IMS Client will be registered only when the initial SIP REGISTER request has been sent, as shown by the solid lines.
G.2: Session Establishment Message Flow

Appendix G.2 is the sequence diagram indicating the establishment of the session from UE A to the terminating network [135]. The terminating network, which belongs to UE B, is shown in Appendix G.3. An I-CSCF is not shown in the diagram. It is important to note that the terminating network can also be the same network, if the same IMS network operator serves both UEs.
G.3: Session Termination Message Flow

Appendix G.3 is the sequence diagram indicating the termination of the session from the originating network to UE B [135]. The originating network may be similar to the one shown in Appendix G.2, but it can also share the same network with the terminating network.
Appendix H
JAVA CODE TO CREATE A DISPLAY OF J2ME

//Create the commands to be attached to the register
display
protected Command exitCmd = new Command("Exit",
    Command.EXIT, 1);
protected Command registerCmd= new Command("Register",
    Command.OK, 0);

//Create registerFrm: Allow the user to register
private Form getRegisterFrm()
{
    if (registerFrm == null)
    {
        registerFrm = new Form("Unregistered to IMS", new Item[]
        {
            new TextField("You have to register first\nPress
            Register to register\n
            Registrar IP address:"
                ,registrar, 30, TextField.ANY) });
        registerFrm.addCommand(exitCmd);
        registerFrm.addCommand(registerCmd);
        registerFrm.setCommandListener(this);
    }
    return registerFrm;
}
Appendix I
Global Diagram of MoBlog

- create your own blog
- add blog entries with pictures/text
- download/view/search in other blogs
- add comments on blogs
Appendix J

XML MESSAGE STRUCTURE

New Blog
Subject: NBL;
<body>
  <date>date in long format</date>
  <title>blog title</title>
  <keywords> (optional) category of the blog</keywords>
</body>

Delete Blog
Subject: DEL;

New entry
Subject: NBE;
<body>
  <date>date in long format</date>
  <topic>subject of the entry</topic>
  <text>entry text</text>
</body>

New Comment
Subject: NCE;blog_id(int);entry_id(int);
<body>
  <date>date in long format</date>
  <text>comment text</text>
</body>

Request Blog list
Subject: REQ;BL;page_nr(int) ;

Edited entry
Subject: EBE;blog_id(int);entry_id(int);
<body>
Requesting an entry list
Subject: REQ;EL;blog_id(int);page_nr(int);

Requesting a comment list
Subject: REQ;CL;blog_id(int);entry_id(int);page_nr(int);

Requesting an entry
Subject: REQ;BE;blog_id(int);entry_id(int);

Requesting a comment
Subject: REQ;CE;blog_id(int);entry_id(int);comment_id(int);

Requesting multimedia
Subject: REQ;DT;blog_id(int);entry_id(int);

Search
Subject: REQ;SR;page_nr(int);
<body>
  <type>title|date|keywords</type>
  <value>search value</value>
</body>

List of blogs
<body>
  <pages>number of pages</pages>
  <blog>
    <id>identification number of the blog</id>
    <title>blog title</title>
  </blog>
</body>
List of subscribed blogs

<body>
    <pages>number of pages</pages>
    <blog>
        <id>identification number of the blog</id>
        <title>blog title</title>
    </blog>
</body>

List of entries

<body>
    <pages>number of pages</pages>
    <author>name of the author</author>
    <entry>
        <id>identification number of the entry</id>
        <topic>topic of the blog</topic>
        <extra>01 1 2 3</extra>
    </entry>
</body>

List of comments

<body>
    <pages>number of pages</pages>
    <comment>
        <id>identification number of the comment</id>
        <title>title of the comment</title>
    </comment>
</body>

Blog Entry

<body>
    <date>date in long format</date>
    <topic>title of the entry</topic>
    <text>entry text</text>
    <data>flag indicating the presence of pictures</data>
</body>
Comment on an Entry

<body>
    <author>name of the author</author>
    <date>date in long format</date>
    <text>text of the comment</text>
</body>

Search

<body>
    <blog>
        <id>identification number of the blog</id>
        <title>title of the blog</title>
    </blog>
</body>
Appendix K
MoBlog Menu Structure