This specification describes the situation of the Proximus network and services. It will be subject to modifications for corrections or when the network or the services will be modified. Please take into account that modifications can appear at any moment. Therefore, the reader is requested to check regularly with the most recent list of available specifications that the document in one’s possession is the latest version.

Proximus can’t be held responsible for any damages due to the use of an outdated version of this specification.

Whilst every care has been taken in the preparation and publication of this document, errors in content, typographical or otherwise, may occur. If you have remarks concerning its accuracy, please send a mail to the following address proximus.uni.spec@proximus.com and your remark will be transmitted to the right Proximus department.
IMS CORPORATE VoIP
SIP SIGNALING

Wireless Office Extended
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Table 1: document history

1. Scope

This document defines the SIP signalling over the VoIP interface between the Proximus IMS Network and large IP-P(A)BXs, connected as SIP Wireless Office Trunk. The specifications listed in this document are not exhaustive but have to be interpreted as “minimal requirements for compliance to the Proximus IMS Corporate VoIP services”.

The specifications are applicable for the following IMS equipment and software packages:

- Alcatel-Lucent ISC - software package Release 13
- Oracle SBC 4600 – Software Version SCZB.1.0 MR-1 patch 12
- Broadsoft application server – BroadWorks R22

This document is part of a set of documents describing the UNI interface of the Proximus IMS Network, for IP-P(A)BXs. Other documents in this set are:

- PXM IMS Corporate VoIP – UNI specification – General [1]
- PXM IMS Corporate VoIP – UNI specification – Testing
- PXM IMS VoIP – UNI specification – Fax support [36]
2. References

Whenever a date of edition is mentioned, the document with this date should be consulted. If no date is present, the latest version of this document should be consulted.

### 2.1. Normative references

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Table 2: normative references

2.2. Informative references

Table 3: informative references
3. Symbols, Definitions and Abbreviations

3.1. Symbols

For the purpose of the present document, the following symbols apply:

None.

3.2. Definitions

For the purpose of the present document, the following definitions apply:

**IP-P(A)BX:** The IP P(A)BX constitutes an Enterprise’s collection of network elements that provides packetized voice call origination and termination services using the Session Initiation Protocol (SIP) and the Session Description Protocol (SDP) for signalling and the Real-time Transport Protocol (RTP) for media traffic.

**pbxPUID:** The public user identity referring to the IP P(A)BX as a whole. The pbxPUID will/can be used as host part of the SIP URI used by the IP-P(A)BX.

**PBXName:** The same as pbxPUID.

**Dialled-SubB-dn:** destination number as dialled by the originating user. The format can be:

- +32 <MobNSN>
- 0 <MobNSN>
- 0032 <MobNSN>
- <PrivatePBXnumber> (=3 to 5 digits long, starting with 1..9)

Note: Emergency call (i.e 1AB) and shortcode calls (i.e. 1ABC) are not accepted on a wireless office trunk.

**Norm-SubB-dn:** destination E.164 number in international format (i.e. +<Country Code><Area Code><DN> ex: +32475963852).

**Norm-SubA-dn:** originating E.164 number in international format (i.e. +<Country Code><Area Code><DN> ex: +32227970231).
DisplayName: the name of the user.

EnterpriseDomain: the public domain name used by the enterprise. Currently the default domain name is ims.belgacom.be. The possibility for the IP-P(A)BX to use, in the future, as public domain name its own domain name e.g. mycompany.com is under study.

Note: EnterpriseDomain can also be an IPAddress instead of a domain name.

IP-addr-PBXName: the IP address of the PBX with name PBXName.

IP-addr-IMS: the IP address of the Proximus IMS network access point (i.e. the SBC).

1AB: called emergency service e.g. 100,112, etc.

1ABC: called short code service

3.3. Abbreviations

See §2.2 of “PXM IMS Corporate VoIP – UNI specification – General” [1]

Additionally for the purpose of the present document, the following abbreviations apply:

CLIP: Calling Line Identification Presentation
CLIR: Calling Line Identification Presentation Restriction
CFU: Call Forwarding Unconditional
CFB: Call Forwarding on Busy
CFNR: Call Forwarding on No Reply
DN: Directory Number
DTMF: Dual Tone Multi-Frequency
iDN: Individual directory number i.e. a particular DN within the range of the IP-P(A)BX
OCB: Outgoing Call Barring
MobNSN: Mobile national significant number
4. General

4.1. Structure of the document

The general structure of this document mainly covers the following aspects:

- SIP profile: this chapter will contain the list of RFCs and standards to which the Proximus IMS network complies, relevant for IP-P(A)BX interconnection.
- SIP behaviour: besides the protocol support also the expected behaviour is important. E.g. how do supplementary services behave. Main topics to be covered in this chapter:
  - Registration/authentication
  - Basic Call
  - Supplementary services
  - CLI screening
  - Fax support
  - Emergency calls
  - DTMF

4.2. Reference point

See §4.3.1 of [1]: “PXM IMS Corporate VoIP – UNI specification – General”.
5. SIP Profile

5.1. Introduction

The following clauses list the SIP related 3GPP and ETSI standards and IETF RFCs to which the IP-P(A)BX shall be compliant.

5.2. 3GPP standards

The Proximus IMS network implements and supports SIP protocol in accordance with:

- TS 24.229 IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP), stage 3 Release 7 [4]
- TS 23.167 IP multimedia subsystem (IMS) emergency sessions, Release 7 [5]

Although this is a registration configuration, emergency registrations are not applicable.

5.3. ETSI TISPAN standards

The Proximus IMS network implements and supports SIP protocol in accordance with:


5.4. IETF RFCs

The Proximus IMS network implements and supports SIP/SDP protocol in accordance with:

- RFC 3261 SIP: session initiation protocol [9]
- RFC 3262 Reliability of provisional responses in SIP [10]
- RFC 3264 an offer/answer model with SDP [11]
- RFC 3265 SIP-specific event notification [12]

Only the NOTIFY method is supported, the SUBSCRIBE method and the 489 "bad event" response are not supported.

Because the NOTIFY method is a rather generic method usable in many different functions (e.g. message waiting indication) it can not be guaranteed that the behaviour will be correct for every possible functionality using the NOTIFY method.

- RFC 4566 SDP: session description protocol [13]
- RFC 2976 SIP INFO method [14]
Because the INFO method is a rather generic method usable in many different functions it can not be guaranteed that the behaviour will be correct for every possible functionality using the INFO method. For example DTMF transport in the INFO method doesn’t work properly. Therefore it is mandatory to support the mechanisms for DTMF transport listed in § 6.9

- RFC 3311 SIP UPDATE Method [15]
- RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP) [16]
- RFC 3325 Private Extensions to SIP for Asserted Identity within Trusted Networks [17]
- RFC 3326 SIP reason header [18]
- RFC 3455 3GPP P-headers [19]
- RFC 3515 SIP REFER Method [20]
- RFC 3891 The Session Initiation Protocol (SIP) “Replaces” Header [21]
- RFC 3892 The Session Initiation Protocol (SIP) Referred-By Mechanism [22]
- RFC 3960 Early Media and Ringing Tone Generation in SIP [23]
- RFC 4028 Session Timers in the SIP [24]
- RFC 2617 HTTP Authentication: Basic and Digest Access Authentication [25]
- RFC 3350 RTP: A Transport Protocol for Real-Time Applications [27]
- RFC 3551 RTP Profile for Audio and Video Conferences with Minimal Control [28]
- RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals [29]
- RFC 4734 Definition of Events for Modem, FAX and Text Telephony signals [30]
- RFC 5806 Diversion indication in SIP [31]
- RFC 6044 Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP) [32]
- RFC 4244 An Extension to SIP for Request History Information [33]
- RFC 3966 The Tel URI for Telephone Numbers [34]
- Draft Applying Loose Routing to Session Initiation Protocol (SIP) User Agents (UA)” (expired) [35]
- RFC 5621 Message Body Handling in SIP [37]

Since more and more SIP applications use multipart message bodies, the correct support and handling of multipart message bodies is very important.

- RFC 5876 Updates to Asserted Identity in the Session Initiation Protocol (SIP) [38]

5.4.1. Supported methods

The following methods shall be supported on the Wireless Office Extended interface:

- INVITE according to [4] and [9]
- ACK according to [4] and [9]
- BYE according to [4] and [9]
- CANCEL according to [4] and [9]
- REGISTER according to [4] and [9]
- OPTIONS according to [4] and [9]
- PRACK according to [4] and [10]
- NOTIFY according to [4] and [12]
- REFER according to [4] and [20]
- UPDATE according to [4] and [15]
- INFO according to [4] and [14]
5.4.2. **Supported responses**

The following responses shall be supported on the Wireless Office Extended interface:

Provisional responses (1xx):
- 100 Trying according to [4] and [9]
- 180 Ringing according to [4] and [9]
- 181 Call Is Being Forwarded according to [4] and [9]
- 182 Queued according to [4] and [9]
- 183 Session Progress according to [4] and [9]

Successful responses (2xx):
- 200 OK according to [4] and [9]
- 202 Accepted according to [4] and [12]

Redirection responses (3xx):
- 302 Moved Temporarily according to [4] and [9]

Request Failure responses (4xx):
- 400 Bad Request according to [4] and [9]
- 401 Unauthorized according to [4] and [9]
- 402 Payment Required according to [4] and [9]
- 403 Forbidden according to [4] and [9]
- 404 Not Found according to [4] and [9]
- 405 Method Not Allowed according to [4] and [9]
- 406 Not Acceptable according to [4] and [9]
- 407 Proxy Authentication Required according to [4] and [9]
- 408 Request Timeout according to [4] and [9]
- 410 Gone according to [4] and [9]
- 413 Request Entity Too Large according to [4] and [9]
- 414 Request-URI Too Long according to [4] and [9]
- 415 Unsupported Media Type according to [4] and [9]
- 416 Unsupported URI Scheme according to [4] and [9]
- 420 Bad Extension according to [4] and [9]
- 421 Extension Required according to [4] and [9]
- 423 Interval Too Brief according to [4] and [9]
- 480 Temporarily Unavailable according to [4] and [9]
- 481 Call/Transaction Does Not Exist according to [4] and [9]
- 482 Loop Detected according to [4] and [9]
- 483 Too Many Hops according to [4] and [9]
- 484 Address Incomplete according to [4] and [9]
- 485 Ambiguous according to [4] and [9]
- 486 Busy Here according to [4] and [9]
- 487 Request Terminated according to [4] and [9]
- 488 Not Acceptable Here according to [4] and [9]
- 491 Request Pending according to [4] and [9]
- 493 Undecipherable according to [4] and [9]
Server Failure responses (5xx):

- 500 Server Internal Error according to [4] and [9]
- 501 Not Implemented according to [4] and [9]
- 502 Bad Gateway according to [4] and [9]
- 503 Service Unavailable according to [4] and [9]
- 504 Server Time-out according to [4] and [9]
- 505 Version Not Supported according to [4] and [9]
- 513 Message Too Large according to [4] and [9]

Global Failures responses (6xx):

- 600 Busy Everywhere according to [4] and [9]
- 603 Decline according to [4] and [9]
- 604 Does Not Exist Anywhere according to [4] and [9]
- 606 Not Acceptable according to [4] and [9]

5.4.3. Supported headers

The following headers shall be supported on the Wireless Office Extended interface:

- Via according to [4] and [9]
- To according to [4] and [9]
- From according to [4] and [9]
- CSeq according to [4] and [9]
- Call-ID according to [4] and [9]
- Contact according to [4] and [9]
- Max-Forwards according to [4] and [9]
- Route according to [4] and [9]
- Record-Route according to [4] and [9]
- Content-Type according to [4] and [9]
- Content-Disposition according to [4] and [9]
- Content-Length according to [4] and [9]
- MIME-Version according to [4] and [9]
- Supported according to [4] and [9]
- Require according to [4] and [9]
- Expires according to [4] and [9]
- Unsupported according to [4] and [9]
- Allow according to [4] and [9]
- Retry-After according to [4] and [9]
- In-Reply-To according to [4] and [9]
- Authorization according to [4] and [9]
- Proxy-Authenticate according to [4] and [9]
- WWW-Authenticate according to [4] and [9]
- Min-Expires according to [4] and [9]
- Proxy-Authorization according to [4] and [9]
- Alert-Info according to [4] and [9]
- Warning according to [4] and [9]

As recommended in RFC 3261 [9] clause 20.22 the IP-PBX shall use as initial value of the Max-Forwards header the value 70.
5.4.4. Unsupported headers

Proprietary SIP headers (i.e. headers starting with “X-” or “x-”) are never supported and shall not be sent on the interface.

5.4.5. Supported bodies

Multipart message bodies are supported and used in the Proximus IMS network. Therefore message body handling according to RFC 5321 [37] is mandatory. In any case correct support and usage of SIP response 415 “Unsupported Media type” as specified in RFC 3261 [9] is mandatory.

5.4.6. Timer values

The Proximus IMS network supports the timers described in [4] and [9] with the following (default) settings:

- T1 = 500 msec
- T2 = 4 seconds
- T4 = 5 seconds
- Timer D = 32 seconds
- Timer H = 32 seconds

All other timers are derived from these, according to [4] and [9].
5.4.7. Transport protocol

The Proximus IMS network supports SIP over UDP only.

Important remark: In case SIP message length approaches the MTU size, [4] and [9] specify that UDP transport should be replaced by TCP transport. This shall NOT be applied by the IP-P(A)BX.

5.4.8. Handling of the Retry-After header

In some cases the IP-PABX may receive a final (error) response containing a Retry-after header. A typical example is a 500 “Internal server error” with Retry-after header with a value of X seconds. A typical value of X would be 1.

The IP-PABX shall correctly handle the Retry-after header according to IETF RFC 3261 [9] and 3GPP TS 24.229 [4] and hence not send new traffic during the interval indicated in the Retry-after header.

In some cases of failure the IP-PABX might send a final (error) response containing a Retry-after header. A typical example is a 500 “Internal server error” with Retry-after header with a value of X seconds. In this case the Proximus IMS network (i.e. the SBC) applies the normal handling of the Retry-After header and consider the SIP link with the IP-PABX as out-of-service for the duration of the Retry-After header.

Warning: The SBC will not sent/receive any traffic over the SIP link during this out-of-service period causing a temporary total outage!
6. SIP Behaviour

6.1. Introduction

Besides the standards and RFCs describing the protocol supported, also the expected behaviour is important. E.g. how does registration exactly take place, which information is expected in the “From” header, etc.

**IMPORTANT:**

SIP signalling serves the call set-up/teardown of calls/sessions as well as the description of session parameters (through the use of SDP) and the invocation of features and services. This means that the exchange of SIP signalling between an IP-PBX and the Proximus network shall serve one of the above purposes. In other words, SIP signalling shall be meaningful and the sending of useless SIP messages shall to be avoided!

Example of useless SIP signalling: A re-INVITE in an established session without any SDP included is often useless because typically re-INVITE is used to re-negotiate/change the session parameters. So, if there is no need to re-negotiate/change the session parameters no re-INVITE should be transmitted.

6.2. Registration/Authentication

IP-P(A)BX devices connected to the Proximus IMS network by use of Wireless Office Extended shall not use registration. The identification and authentication of the IP-P(A)BX is done by making use of the one-to-one mapping of fixed IP-addresses. This is done through configuration in the Session Border Controller (SBC) of the Proximus IMS network.

Internal in the corporate network SIP end devices e.g. SIP phones, may register with the IP-P(A)BX itself. These registrations with the IP P(A)BX shall remain invisible to the Proximus IMS network.

Each IP-P(A)BX will be assigned one PBX public user identity (pbxPUID), aka Pilot ID aka PBXName. This is a non-dialable public identity.

An IP-P(A)BX has usually allocated to it one or more contiguous DN ranges.

Authentication procedures will not be requested for INVITE (session authentication).

6.3. Basic call

In Wireless Office extended several call types are possible:

- Private on-net calls are completely treated by the IP-P(A)BX and hence out of scope of this document.
• Public off-net calls are **transited** by the Proximus IMS network. This is a call between an IP-P(A)BX extension and a mobile IP-P(A)BX extension or a normal external (mobile) user.

Basic call set-up and tear down complies with normal SIP behaviour as described in the relevant Standards and RFCs [4][9][10][11][13].

Basic call set-up in the SIP Wireless Office extended case happens without authentication. Since fixed IP addresses are used the Proximus IMS network identifies and authorizes the IP-P(A)BX based on the IP address.

Basic call set-up may use the mechanism for reliable transport of 1XX responses according to IETF RFC 3262 "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)" [10], but it is not mandatory. The procedures for announcing the capability and use are described in [10].

Basic call set-up may use the mechanism for session timer according to IETF RFC 4028 "Session Timers in the Session Initiation Protocol (SIP)" [24], but it is not mandatory. The procedures for announcing the capability and use are described in [24].

The IP-P(A)BX shall use the following Request-URI to originate a call:

• Request-URI:
  o sip:Dialled-SubB-dn@ims.belgacom.be;user=phone

  Note: using an IP address instead of a domain in the host part of the URI will result in call failure.

The IP-P(A)BX shall use the following own identity to originate a call:

• From header:
  o sip:Norm-SubA-dn@EnterpriseDomain;user=phone

  The From header **shall** always contain a valid PUID, even when the CLIR service is to be invoked.

• Contact header
  o sip:Norm-SubA-dn@IP-addr-PBXName, or
  o sip:PBXName@IP-addr-PBXName

  The IP address of the IP-P(A)BX in the Contact header **MUST** be the same as used in the configuration of the Proximus SBC because the Proximus IMS network uses this for admission control.

• P-Asserted-ID header:
  o sip:Norm-SubA-dn@EnterpriseDomain;user=phone

  According to SIP Connect 1.1 [6], it is recommended that the IP-P(A)BX includes a P-Asserted-ID header in the INVITE request. The Proximus IMS network does not expect to receive this header and if received this header will be overwritten by the Proximus IMS network with the correct P-Asserted-ID.

Note: as "Norm-SubA-dn" any DN within the range of the IP-P(A)BX can be used.
6.3.1. Public off-net originating call set-up

Figure 1 shows the expected message flow for a public on-net originating call set-up.

Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

![Diagram showing message flow for public off-net originating call set-up]

To set-up a public off-net call the IP-P(A)BX sends an INVITE message to the Proximus IMS network with the following headers:

- Request-URI = sip:Dialled-SubB-dn@ims.belgacom.be;user=phone
- To = <sip:Dialled-SubB-dn@ims.belgacom.be;user=phone>
- From = DisplayName <sip:Norm-SubA-dn@EnterpriseDomain;user=phone>
- Via = IP-addr-PBX
- Contact = DisplayName <sip:Norm-SubA-dn@IP-addr-PBX>
- P-asserted-id = DisplayName <sip:Norm-SubA-dn@EnterpriseDomain;user=phone>

Note: The DisplayName in the From, Contact and P-Asserted-id header is optional

Live example

Request-Line: INVITE sip:0477143104@ims.belgacom.be;user=phone SIP/2.0
Message Header
- Route: <sip:10.127.249.190;lr>
- Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
- Supported: 100rel,from-change,timer
- User-Agent: XXXX
- Session-Expires: 43200
- P-Asserted-Identity: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
- To: <sip:0477143104@ims.belgacom.be;user=phone>
From: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>; tag=796b7734f7742acdf213d7c07618d32
Contact: "TEL 027979380" <sip:+3227979380@10.127.249.4; transport=UDP;user=phone>
Content-Type: application/sdp
Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
CSeq: 1465254309 INVITE
Via: SIP/2.0/UDP 10.127.249.4;rport;
branch=z9hG4bK088b463ea7b71dd2edac3b21b9cf5b1
Max-Forwards: 70
Content-Length: 276
Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): default 1293026893 1293026893 IN IP4 10.127.249.4
Owner Username: default
Session ID: 1293026893
Session Version: 1293026893
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.127.249.4
Session Name (s): -
Connection Information (c): IN IP4 10.127.249.4
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 10.127.249.4
Time Description, active time (t): 0 0
Session Start Time: 0
Session Stop Time: 0
Media Description, name and address (m): audio 32000 RTP/AVP 18 106 4 8 0
Media Type: audio
Media Port: 32000
Media Proto: RTP/AVP
Media Format: ITU-T G.729
Media Format: 106
Media Format: ITU-T G.723
Media Format: ITU-T G.711 PCMA
Media Format: ITU-T G.711 PCMU
Media Attribute (a): sendrecv
Media Attribute (a): fmtp:18 annexb=no
Media Attribute Fieldname: fmtp
Media Format: 18
Media format specific parameters: annexb=no
Media Attribute (a): rtpmap:106 telephone-event/8000
Media Attribute Fieldname: rtpmap
Media Format: 106
MIME Type: telephone-event
Media Attribute (a): fmtp:106 0-15
Media Attribute Fieldname: fmtp
The INVITE contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the IP-P(A)BX user wants to receive RTP).

The Proximus IMS network first returns a 100 Trying response. When the destination is reached a 180 Ringing response is returned to the IP-P(A)BX.

Remark: Because the initial INVITE sent by the IP-P(A)BX announced support for 100rel in the Supported header, the Proximus IMS network agrees to apply this mechanism. The 180 Ringing response contains SDP information regarding the accepted call parameters and the following headers:

- **Require = 100rel**
- **Rseq = <Rseqvalue>

### Live example

**Status-Line:** SIP/2.0 100 Trying  
**Message Header**

- **To:** <sip:0477143104@ims.belgacom.be;user=phone>  
- **From:** "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>;  
  tag=796b7734f7742acdf213d7c07f618d32  
- **Call-ID:** 49414ca83c459797570631be577686ab@10.127.249.4  
- **Via:** SIP/2.0/UDP 10.127.249.4;received=10.127.249.4;rport=5060;  
  branch=z9hG4bk088b463e17b71dd2edac3b21b9c5f5b1  
- **CSeq:** 1465254309 INVITE  
- **Content-Length:** 0

**Status-Line:** SIP/2.0 180 Ringing  
**Message Header**

- **To:** <sip:0477143104@ims.belgacom.be;user=phone>;  
  tag=1B5D32463135364139520000  
- **From:** "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>;  
  tag=796b7734f7742acdf213d7c07f618d32  
- **Call-ID:** 49414ca83c459797570631be577686ab@10.127.249.4  
- **Via:** SIP/2.0/UDP 10.127.249.4;received=10.127.249.4;rport=5060;  
  branch=z9hG4bk088b463e17b71dd2edac3b21b9c5f5b1  
- **CSeq:** 1465254309 INVITE
Contact: <sip:10.127.249.190:5060;transport=udp>

Require: 100rel
Content-Type: application/sdp
Allow: INVITE, ACK, OPTIONS, BYE, CANCEL, REGISTER, INFO, COMET,
       UPDATE, PRACK, REFER, SUBSCRIBE, NOTIFY, MESSAGE
RSeq: 43981
Content-Length: 207
Server: Alcatel-Lucent-HPSS/3.0.3
Message body

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): - 4000100463 1293026888 IN IP4 10.127.249.190
   Owner Username: -
   Session ID: 4000100463
   Session Version: 1293026888
   Owner Network Type: IN
   Owner Address Type: IP4
   Owner Address: 10.127.249.190
Session Name (s): SDP Data
Connection Information (c): IN IP4 10.127.249.190
   Connection Network Type: IN
   Connection Address Type: IP4
   Connection Address: 10.127.249.190
Time Description, active time (t): 0 0
   Session Start Time: 0
   Session Stop Time: 0
Media Description, name and address (m): audio 10000 RTP/AVP 18 106
   Media Type: audio
   Media Port: 10000
   Media Proto: RTP/AVP
   Media Format: ITU-T G.729
   Media Format: 106
Media Attribute (a): ptime:20
   Media Attribute Fieldname: ptime
   Media Attribute Value: 20
Media Attribute (a): maxptime:60
   Media Attribute Fieldname: maxptime
   Media Attribute Value: 60
Media Attribute (a): rtpmap:106 telephone-event/8000
   Media Attribute Fieldname: rtpmap
   Media Format: 106
   MIME Type: telephone-event
Media Attribute (a): fmtp:106 0-15
   Media Attribute Fieldname: fmtp
   Media Format: 106 [telephone-event]
   Media format specific parameters: 0-15
The IP-P(A)BX shall send a PRACK request to the Proximus IMS network in order to acknowledge the receipt of the 180 response.

The PRACK request contains the following headers:

- **Rack** = `<Rseqvalue as received> <Rackvalue> INVITE`

**Live example**

```
Request-Line: PRACK sip:0477143104@ims.belgacom.be;user=phone SIP/2.0
Message Header
  Route: <sip:10.127.249.190;lr>
  RAck: 43981 1465254309 INVITE
  RSeq Sequence Number: 43981
  CSeq Sequence Number: 1465254309
  User-Agent: OxO_GW_710/133.001
  To: <sip:0477143104@ims.belgacom.be;user=phone>;
      tag=1B5D3246315364139520000
  From: <sip:+3227979380@10.127.249.4;user=phone>;
       tag=796b7734f7742acdf213d7e07f618d32
  Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
  CSeq: 1465254310 PRACK
  Via: SIP/2.0/UDP 10.127.249.4;rport;
       branch=z9hG4bKd99d0b399988d43bf3b2657889205e44
  Max-Forwards: 70
  Content-Length: 0
```

The Proximus IMS network shall send a 200 OK response to the IP-P(A)BX in order to finalize the PRACK transaction.

**Live example**

```
Status-Line: SIP/2.0 200 OK
Message Header
  To: <sip:0477143104@ims.belgacom.be;user=phone>;
      tag=1B5D3246315364139520000
  From: <sip:+3227979380@10.127.249.4;user=phone>;
       tag=796b7734f7742acdf213d7e07f618d32
  Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
  Via: SIP/2.0/UDP 10.127.249.4;rport=5060;
       branch=z9hG4bKd99d0b399988d43bf3b2657889205e44
  CSeq: 1465254310 PRACK
  Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,
        UPDATE,PRACK,REFER,SUBSCRIBE,NOTIFY,MESSAGE
  Content-Length: 0
```

When the call is answered the Proximus IMS network returns a 200 OK response for the INVITE transaction.
Remark: Because the initial INVITE sent by the IP-P(A)BX announced support for session timer in the Supported header and Session-Expires header, the Proximus IMS network reacts with a Session-Expires header granting the proposed expiration time and indicating that the user agent server (i.e. the Proximus IMS network) will be responsible to refresh the session.

Live example

Status-Line: SIP/2.0 200 OK
Message Header
   To: <sip:0477143104@ims.belgacom.be;user=phone>; tag=1B5D3246313564439520000
   From: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
      tag=796b7774f7742adcfd2137607f61d32
   Call-ID: 49414c83c459797570631be577686ab@10.127.249.4
   Via: SIP/2.0/UDP 10.127.249.4;received=10.127.249.4;rport=5060;
      branch=z9hG4bKo88b463ea7b71dd2edac3b21b9c5f1b1
   CSeq: 146524309 INVITE
   Contact: <sip:10.127.249.190:5060;transport=udp>
   P-Asserted-Identity: +32477143104 <sip:+32477143104@10.127.69.39:5061;user=phone>
   Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,
      UPDATE,PRACK,REFER,SUBSCRIBE,NOTIFY,MESSAGE
   Supported: 100rel,timer,rebuses,diversion
   Session-Expires: 43200;refresher=uas
   Server: Alcatel-Lucent-HPSS/3.0.3
   Content-Length: 0

The IP-P(A)BX sends an ACK message to the Proximus IMS network in order to properly close the INVITE transaction.

Live example

Request-Line: ACK sip:10.127.249.190:5060;transport=udp SIP/2.0
Message Header
   Route: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
   User-Agent: OxO_GW_710/133.001
   To: <sip:0477143104@ims.belgacom.be;user=phone>; tag=1B5D3246313564439520000
   From: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
      tag=796b7774f7742adcfd2137607f61d32
   Call-ID: 49414c83c459797570631be577686ab@10.127.249.4
   CSeq: 146524309 ACK
   Via: SIP/2.0/UDP 10.127.249.4;rport;
      branch=z9hG4bK7ce8a6c77c13b84a6dd3b1c6023ea27
   Max-Forwards: 70
   Content-Length: 0
6.3.2. Public off-net terminating call set-up

Figure 2 shows the expected message flow for a public off-net originating call set-up. Note: depending on the call scenario (e.g. whether reliability of provisional responses is used or not) differences may occur.

To set-up a call to the IP-P(A)BX, the Proximus IMS network sends an INVITE message to the IP-P(A)BX with the following headers:

- Request-URI = sip:Norm-SubB-dn@IP-addr-PBXName;user=phone
- To = <sip:Norm-SubB-dn@EnterpriseDomain;user=phone>
- From = DisplayName sip:Norm-SubA-dn@woe.proximus.be;user=phone or DisplayName tel:Norm-SubA-dn
- Via = IP-addr-IMS
- Contact = <sip:IP-addr-IMS>
- P-asserted-id = DisplayName sip:Norm-SubA-dn@woe.proximus.be;user=phone or DisplayName tel:Norm-SubA-dn

Note: The DisplayName in the From, Contact and P-asserted-id header is optional

Remark: Any incoming basic call to an IP-P(A)BX may have undergone diversion before it reaches the IP-P(A)BX. Therefore a diverting number can be present in the INVITE message in the Diversion header (the Diversion Top header contains the Redirecting Number, while the Diversion Bottom header contains the Original Called Number). The Diversion header has been documented in IETF RFC 5806 Diversion indication in SIP [31].

The INVITE contains SDP information regarding the proposed call parameters (e.g. codec, IP address and port number on which the Proximus IMS network wants to receive RTP).

Live example
Request-Line: INVITE sip:+3227979380@10.127.249.4:5060;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqotdonduhrd90k3nauqe15l4
Call-ID: o1FF26AEc281400000000002@impmsilab5-sig.stgl.sel.alcatel.de
To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>
From: <sip:+32477143104@woe.proximus.be;user=phone>; tag=D65E324631536419B60000
CSeq: 1 INVITE
Max-Forwards: 57
Content-Type: application/sdp
Contact: <sip:10.127.249.190:5060;transport=udp>
P-Asserted-Identity: <sip:+32477143104@woe.proximus.be;user=phone>
P-Charging-Vector: icid-value=S9C80-20101222152015-0000000909;
icid-generated-at=149.204.0.1;orig-oi=sub.alcatel-ngvoice.com
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,
UPDATE,PRACK,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Supported: 100rel,timer,replaces,diversion
Expires: 155
Session-Expires: 36000
Min-SE: 90
Alcatel-Service-Data: Profile-Service-Data=COLP-request
Timestamp: 28315
Content-Length: 192
User-Agent: Alcatel-Lucent-HPSS v3.0.3
Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): 4000400462 1293027615 IN IP4 10.127.249.190
Owner Username: -
Session ID: 4000400462
Session Version: 1293027615
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.127.249.190
Session Name (s): SDP Data
Connection Information (c): IN IP4 10.127.249.190
Connection Network Type: IN
Connection Address Type: IP4
Connection Address: 10.127.249.190
Time Description, active time (t): 0 0
Session Start Time: 0
Session Stop Time: 0
Media Description, name and address (m): audio 10002 RTP/AVP 8 18 101
Media Type: audio
Media Port: 10002
Media Proto: RTP/AVP
Media Format: ITU-T G.711 PCMA
Media Format: ITU-T G.729
Media Format: 101
The Proximus IMS network expects a 100 Trying response, followed by a 180 Ringing, when the destination user is reached.

Live example

Status-Line: SIP/2.0 100 Trying
Message Header
To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>
From: <sip:+32477143104@woe.proximus.be;user=phone>
   tag=D65E3246313536419B6E0000
Call-ID: 01FF26AEC28140000000002@impmgsilab5-sig.stgl.sel.alcatel.de
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bK2qotdonduhrd90k3nauqe15l4
Timestamp: 28315
Content-Length: 0

Status-Line: SIP/2.0 180 Ringing
Message Header
Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Contact: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
Supported: from-change
User-Agent: OxO_GW_710/133.001
P-Asserted-Identity: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>
   tag=b43efbe9aa92a65b38e934ff10d37b
From: <sip:+32477143104@woe.proximus.be;user=phone>
   tag=D65E3246313536419B6E0000
Call-ID: 01FF26AEC28140000000002@impmgsilab5-sig.stgl.sel.alcatel.de
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bK2qotdonduhrd90k3nauqe15l4
Content-Length: 0

When the call is answered the Proximus IMS network expects a 200 OK response containing SDP information regarding the accepted call parameters (e.g. codec, IP address and port number on which the destination wants to receive RTP).

Live example
Status-Line: SIP/2.0 200 OK
Message Header
   Content-Type: application/sdp
   Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
   Contact: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
   Require: timer
   Supported: 100rel,timer,from-change
   User-Agent: OxO_GW_710/133.001
   Session-Expires: 36000;refresher=uac
   P-Asserted-Identity: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>
   To: <sip:+3227979380@ims.belgacom.be;5060;user=phone>;
   tag=b43cfbe9aaa92a65b38e9934ff0d37b
   From: <sip:+32477143104@woe.proximus.be;user=phone>;
   tag=D65E3246313536419B6E0000
   Call-ID: 01FF26AEC28140000000002@impmgsilab5-sig.stgl.sel.alcatel.de
   CSeq: 1 INVITE
   Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqotdonduhdr90k3nuqe15l4
   Content-Length: 247
Message body
Session Description Protocol
   Session Description Protocol Version (v): 0
   Owner/Creator, Session Id (o): default 1293027627 1293027627 IN IP4 10.127.249.4
   Owner Username: default
   Session ID: 1293027627
   Session Version: 1293027627
   Owner Network Type: IN
   Owner Address Type: IP4
   Owner Address: 10.127.249.4
   Session Name (s): SDP Data
   Connection Information (c): IN IP4 10.127.249.4
   Connection Network Type: IN
   Connection Address Type: IP4
   Connection Address: 10.127.249.4
   Time Description, active time (t): 0 0
   Session Start Time: 0
   Session Stop Time: 0
   Media Description, name and address (m): audio 32000 RTP/AVP 8 101
   Media Type: audio
   Media Port: 32000
   Media Proto: RTP/AVP
   Media Format: ITU-T G.711 PCMA
   Media Format: 101
   Media Attribute (a): sendrecv
   Media Attribute (a): rtpmap:101 telephone-event/8000
   Media Attribute Fieldname: rtpmap
   Media Format: 101
   MIME Type: telephone-event
   Media Attribute (a): fmtp:101 0-15
   Media Attribute Fieldname: fmtp
The Proximus IMS network sends an ACK message to the IP-P(A)BX in order to properly close the INVITE transaction.

Live example

Request-Line: ACK sip:+3227979380@10.127.249.4;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.249.190:5060;
branch=z9hG4bKb7ekt8dekqmq15u0edlcc7oe1-g0g5
P-Charging-Vector: icid-value=S9C80-20101222152015-00000909;
icid-generated-at=149.204.0.1; orig-oi=sub.alcatel-nginxvoice.com
CSeq: 1 ACK
To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>;
tag=b43cffbe9aa92a65b38e934ff10d37b
From: <sip:+32477143104@woe.proximus.be;user=phone>;
tag=D65E3246313536419B6E0000
Call-ID: 01FF26AEC2814000000000002@impmsglab5-sig.stgl.sel.alcatel.de
Max-Forwards: 69
Timestamp: 28396
Content-Length: 0

6.3.3. Forward call tear down

Message flow example
Figure 6: Basic call/session teardown example

To end a public off-net call the IP-P(A)BX sends a BYE message to the Proximus IMS network.

- **CallID** = matches the CallID of the concerned call
- **Cseq** = is incremented with regard to the previous transaction related to this call

**Live example**

Request-Line: BYE sip:10.127.249.190:5060;transport=udp SIP/2.0
Message Header
   Route: <sip:10.127.249.190;lr>
   User-Agent: OxO_GW_710/133.001
   To: <sip:0477143104@ims.belgacom.be;user=phone>;tag=1B5D32463135364139520000
   From: <sip:+3227979380@10.127.249.4;user=phone>
   Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
   CSeq: 1465254311 BYE
   Via: SIP/2.0/UDP 10.127.249.4;rport;
       branch=z9hG4bKebfeabbb1781f85942ca544c7b5ba1ec
   Max-Forwards: 70
   Content-Length: 0

The Proximus IMS network returns a 200 OK response.

**Live example**

Status-Line: SIP/2.0 200 OK
Message Header
   To: <sip:0477143104@ims.belgacom.be;user=phone>
   tag=1B5D32463135364139520000
   From: <sip:+3227979380@10.127.249.4;user=phone>
   Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
   Via: SIP/2.0/UDP 10.127.249.4;received=10.127.249.4;rport=5060;
       branch=z9hG4bKebfeabbb1781f85942ca544c7b5ba1ec
   CSeq: 1465254311 BYE
   Content-Length: 0

### 6.3.4. Backward call tear down

**Message flow example**
To end a public on-net or a public off-net call the Proximus IMS network sends a **BYE** message to the IP-P(A)BX.

- **CallID** = matches the **CallID** of the concerned call
- **Cseq** = is incremented with regard to the previous transaction related to this call

### 6.4. Supplementary services

All supplementary services shall be executed at - and managed by - the IP-P(A)BX. The Proximus IMS network serves as transit network towards the mobile network only. No services are offered nor executed in the Proximus IMS network.

However, some services executed by the IP-P(A)BX involving external users may need particular interaction with the Proximus IMS network. The expected behaviour is discussed in the following clauses.

#### 6.4.1. Calling Line Identity Presentation (CLIP)

##### 6.4.1.1. Incoming call to the IP-P(A)BX

The CLIP service sends the identity of the calling line to the IP-P(A)BX. The calling line identity can be provided by the network or it can include identity information supplied by the caller and validated by the network.

The calling line identity can be contained in the **From** and **P-Asserted-Id** headers in the **INVITE** message sent to the IP-P(A)BX. The presentation information of the calling line identity, which establishes the identity type (Presentation Allowed or Presentation Restricted) is contained in the **Privacy** header in the **INVITE** message.

The format of the calling line identity information in the **From** and **P-Asserted-Id** headers in the **INVITE** message is:
• **From** = DisplayName sip:Norm-SubA-dn@woe.proximus.be;user=phone or DisplayName tel:Norm-SubA-dn

• **P-asserted-id** = DisplayName <sip:Norm-SubA-dn@woe.proximus.be;user=phone> DisplayName tel:Norm-SubA-dn

Note 1: The DisplayName in the From header and P-Asserted-ID header is optional.

Note 2: The Norm-SubA-dn in the From header and P-Asserted-ID header is not necessarily the same. For example in case of interworking with legacy ISDN two calling line identities may be delivered.

The Privacy header in the INVITE message has the value "none". The absence of the Privacy header equally means "no privacy".

**Live example**

Request-Line: INVITE sip:+3227979380@10.127.249.4:5060;user=phone SIP/2.0
Message Header
Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqodunduhrd90k3nauqe15l4
Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stg.sel.alcatel.de
To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>
From: <sip:+32477143104@woe.proximus.be;user=phone>; tag=D65E3246313536419B6E0000
CSeq: 1 INVITE
Max-Forwards: 57
Content-Type: application/sdp
Contact: <sip:+3227979380@ims.belgacom.be:5060;transport=udp>
P-Charging-Vector: icid-value=S9C80-2010122152015-00000909;
   icid-generated-at=149.204.0.1;orig-oui=sub.alcatel-ngvoice.com
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,REGISTER,INFO,COMET,
   UPDATE,PRACK,REFER,SUBSCRIBE,NOTIFY,MESSAGE
Supported: 100rel,timer,replaces,diversion
Expires: 155
Session-Expires: 36000
Min-SE: 90
Alcatel-Service-Data: Profile-Service-Data=COLP-request
Timestamp: 28315
Content-Length: 192
User-Agent: Alcatel-Lucent-HPSS v3.0.3

Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): -4000400462 1293027615 IN IP4 10.127.249.190
   Owner Username: -
   Session ID: 4000400462
   Session Version: 1293027615
   Owner Network Type: IN
6.4.1.2. **Outgoing call from the IP-P(A)BX**

The calling line identity shall be contained in the From and P-Asserted-Id headers in the INVITE message sent by the IP-P(A)BX.

- From = DisplayName <sip:Norm-SubA-dn@IP-addr-PBX;user=phone>
- P-asserted-id = DisplayName <sip:Norm-SubA-dn@IP-addr-PBX;user=phone>

Note: The DisplayName in the From and P-Asserted-Id header is optional.

**Live example**

Request-Line: INVITE sip:0477143104@ims.belgacom.be;user=phone SIP/2.0
Message Header
Route: <sip:10.127.249.190;lr>
Allow: INVITE, ACK, CANCEL, BYE, OPTIONS, PRACK, REFER, NOTIFY, UPDATE
Supported: 100rel,from-change,timer
User-Agent: OxO_GW_710/133.001
Session-Expires: 43200
P-Asserted-Identity: "027979380"

<sip:+3227979380@10.127.249.4;user=phone>
To: <sip:0477143104@ims.belgacom.be;user=phone>
From: "027979380" <sip:+3227979380@10.127.249.4;user=phone>;
tag=796b7742f742acf613c67c07f618d32
Contact: "TEL 027979380" <sip:+3227979380@10.127.249.4;
transport=UDP;user=phone>
Content-Type: application/sdp
Call-ID: 49414ca83e459797507631be577686ab@10.127.249.4
CSeq: 1465243009 INVITE
Via: SIP/2.0/UDP 10.127.249.4; rport;
branch=z9hG4bK088b463e1a7b71dd2edac3b21b9cf5b1
Max-Forwards: 70
Content-Length: 276

Message body

Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): default 1293026893 1293026893 IN IP4 10.127.249.4
Owner Username: default
Session ID: 1293026893
Session Version: 1293026893
Owner Network Type: IN
Owner Address Type: IP4
Owner Address: 10.127.249.4
Session Name (s): -
Connection Information (c): IN IP4 10.127.249.4
  Connection Network Type: IN
  Connection Address Type: IP4
  Connection Address: 10.127.249.4
Time Description, active time (t): 0 0
  Session Start Time: 0
  Session Stop Time: 0
Media Description, name and address (m): audio 32000 RTP/AVP 18 106 4 8 0
  Media Type: audio
  Media Port: 32000
  Media Proto: RTP/AVP
  Media Format: ITU-T G.729
  Media Format: 106
  Media Format: ITU-T G.723
  Media Format: ITU-T G.711 PCMA
  Media Format: ITU-T G.711 PCMU
  Media Attribute (a): sendrecv
  Media Attribute (a): fomtp:18 annexb=no
  Media Attribute Fieldname: fomtp
  Media Format: 18
  Media format specific parameters: annexb=no
  Media Attribute (a): rtpmap:106 telephone-event/8000
  Media Attribute Fieldname: rtpmap
6.4.2. Calling Line Identity presentation Restriction (CLIR)

6.4.2.1. Incoming call to the IP-P(A)BX

In case the CLIR service has been invoked by the calling user of an incoming call to the IP-P(A)BX, the From header in the INVITE will not contain the identity information of the calling user and the P-Asserted-ID header will not be present in the INVITE. The presentation information of the calling line identity, which establishes the identity type (Presentation Allowed or Presentation Restricted) is contained in the Privacy header in the INVITE message.

- From = DisplayName <sip:Anonymous@anonymous.invalid>
- Privacy = id and/or header and/or User

Note: if present DisplayName will have the value “Anonymous”.

6.4.2.2. Outgoing call from the IP-P(A)BX

In order to prevent the presentation of the calling user’s identity (invoke CLIR) the IP-P(A)BX can include a Privacy header in the INVITE. The value of the Privacy header in the INVITE shall be id and/or header and/or User.

In any case the From header shall always contain a valid URI (not anonymous), even when the CLIR service is to be invoked.
6.4.3. Call Forwarding Services

The call forwarding feature(s) shall be managed and handled by the IP-P(A)BX.

The following behaviour is valid for all flavours (e.g. unconditional (CFU), on busy (CFB), on no reply (CFNR), …) of call forwarding executed by the IP-P(A)BX.

In case an incoming call to the IP-P(A)BX is forwarded to a destination external to the IP-P(A)BX, the IP-P(A)BX shall create the forwarding by setting up a new (forwarded) call. The incoming call and the outgoing (forwarded) call are considered being 2 separate calls. A Diversion header may be included in the outgoing (forwarded) call in which case the calling line identity possibly shown to the forwarded destination will be the original calling line identity, else the calling line identity possibly shown to the forwarded-to destination will be the identity of the forwarding IP-P(A)BX user (or the IP-P(A)BX general number). The History-info header MUST not be included in the outgoing (forwarded) call.

Remark: The IP-P(A)BX shall not use a 302 Moved Temporarily response.

6.4.4. Outgoing Call Barring

The OCB service is not offered to the IP-P(A)BX. No network Outgoing Call Barring can be programmed for the complete IP-P(A)BX and/or for each iDN.

In case the IP-P(A)BX applies Outgoing Call Barring to one of his users, the call shall not be sent to the Proximus IMS network.

6.4.5. Call Hold

The call hold feature shall be managed and handled by the IP-P(A)BX.

An IP-P(A)BX user may place an active call on hold and may retrieve a held call. The IP-P(A)BX shall provide “music on hold” or play an announcement for the held user. For these purposes, the Proximus IMS supports the use of re-INVITE transactions to modify the media description parameters for a call according to the held/retrieved state.

An IP-P(A)BX user involved in an active call may be placed on hold and may be retrieved. Depending on the holding user (external to the IP-P(A)BX) either the Proximus IMS or the holding user’s equipment is responsible to provide “music on hold” or to play an announcement for the held IP-P(A)BX user. For these purposes, the Proximus IMS supports the use of re-INVITE transactions to modify the media description parameters for a call according to the held/retrieved state.

6.4.6. Conference call

The conference feature shall be managed and handled by the IP-P(A)BX. In case external users are involved in a conference call, the incoming and/or outgoing calls to/from the IP-P(A)BX will be treated like normal basic calls by the Proximus IMS.

6.4.7. Call transfer

The call transfer feature shall be managed and handled by the IP-P(A)BX.
The following behaviour is valid for all flavours (e.g. with or without consultation) of call transfer executed by the IP-P(A)BX.

In case a call involving an IP-P(A)BX user is transferred (by that IP-P(A)BX user) to a destination external to the IP-P(A)BX, the IP-P(A)BX shall set-up a new call to the transferred-to destination, and:

- either use a re-INVITE to transfer the RTP stream of the original call to the new destination. No Diversion header nor History-info header shall be included in the outgoing call. The calling line identity possibly shown to the transferred-to destination will be the identity of the forwarding IP-P(A)BX user (or the IP-P(A)BX general number).
- or not use a re-INVITE and handle the transfer of the RTP streams autonomously.

The IP-P(A)BX shall not use a REFER message to redirect the call to the transferred-to-number.

6.5. CLI screening

The Proximus IMS network will not verify the identity of the calling user received from the IP-P(A)BX in the From header and P-asserted-ID.

6.6. FAX support

See “PXM IMS VoIP – UNI specification – Fax over IP” [36]

6.7. Emergency calls

Emergency calls will not be accepted by the Proximus IMS network on a wireless office trunk. The IP-P(A)BX shall route all emergency calls over the Business Trunk with IMS services (aka fixed Business Trunk).

6.8. DTMF

For transport of DTMF the following capabilities exist:

- Usage of RFC 2833 (inband signalling in RTP by use of events)
- Usage of G.711 codec (inband signalling in RTP)

Both methods of transporting DTMF shall be supported.
Evolution: Usage of the SIP INFO message (outband signalling: the info follows the path of signalling) is currently under study and may be announced in a later version of this document.

6.9. Codec changes during a call/session

Codec changes during an established call/session are allowed using the re-INVITE mechanism. However it should be noted that sending a re-INVITE to the Proximus IMS network very fastly after the establishment of the call/session may result in an error response with Retry-after header.

Therefore it is advised to not sent a re-INVITE faster then 50 mseconds after the ACK request finalising the previous INVITE transaction.