IMS CORPORATE VoIP
SIP SIGNALING

Wireless Office Extended

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0. Document History

Every update of this document results in a complete new version with new version number and release date.

Version	Date	Main or important changes since previous version
1.0	February 15, 2011	First version
1.1	February 16, 2011	Corrections regarding emergency call number formats
1.2	March 02, 2011	Addition emphasizing the importance to use a domain in the Request-URI (rather than an IP address)
1.3	May 16, 2011	Corrections to the cover sheet
1.4	June 29, 2011	References added for support of multipart message bodies Updates regarding P-Asserted-ID handling
1.5	December 20; 2011	Domain replaced by IP-address of IP-PBX for call towards IP-PBX
1.6	March 26, 2012	Addition regarding proprietary SIP headers
1.7	March 08, 2013	Additional info on Retry-After header Guidelines for sending re-INVITE
1.8	Januari 13, 2016	Changed "Belgacom" in "Proximus"
1.9	September 05, 2016	Made support of both DTMF transport methods mandatory
1.10	November 24, 2016	Update due to name change "Bizz IP telephony multi" into "Enterprise voice multi"
1.11	June 28, 2018	Clarification concerning the initial value of the Max-Forwards header The use of Diversion header added. Correction: mobile.ims.belgacom.be changed in woe.proximus.be
2.0	May 25, 2019	Document name changed from BGC to PXM
2.1	June 06, 2019	Small update based on comments from IP-PBX team
2.2	September 19, 2019	Update of SW versions in Scope

2.3	November 20, 2020	Update concerning the use of the Privacy header

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 SCZ8.1.0 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 – SIP signalling – Business Trunking with IMS services [2]

PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice Multi [3]

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

1	Proximus	PXM IMS Corporate VoIP – UNI specification – General
2	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Business Trunking with IMS services
3	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice Multi
4	3GPP TS 24.229	IP Multimedia call control protocol based on SIP and SDP, Stage 3 Release 9
5	3GPP TS 23.167	IP Multimedia Subsystem (IMS) emergency sessions, Release 9
6	SIP forum	The SIP connect 1.1 technical recommendation (draft)
7	ETSI TS 182 025	Business Trunking; architecture and functional description v2.1.1
8	ITU-T E.164	The international telecommunication numbering plan
9	IETF RFC 3261	SIP: Session Initiation Protocol
10	IETF RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
11	IETF RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
12	IETF RFC 3265	Session Initiation Protocol (SIP)-Specific Event Notification
13	IETF RFC 4566	SDP: Session Description Protocol
14	IETF RFC 2976	The SIP INFO Method
15	IETF RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
16	IETF RFC 3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
17	IETF RFC 3325	Private Extensions to SIP for Asserted Identity within Trusted Networks
18	IETF RFC 3326	The Reason Header Field for the Session Initiation Protocol (SIP)
19	IETF RFC 3455	Private header extensions for SIP for 3GPP
20	IETF RFC 3515	The Session Initiation Protocol (SIP) REFER Method
21	IETF RFC 3891	The Session Initiation Protocol (SIP) "Replaces" Header
22	IETF RFC 3892	The Session Initiation Protocol (SIP) Referred-By Mechanism

23	IETF RFC 3960	Early Media and Ringing Tone Generation in SIP
24	IETF RFC 4028	Session Timers in the Session Initiation Protocol (SIP)
25	IETF RFC 2617	HTTP Authentication: Basic and Digest Access Authentication
26	IETF RFC 1321	The MD5 Message-Digest Algorithm
27	IETF RFC 3550	RTP: A Transport Protocol for Real-Time Applications
28	IETF RFC 3551	RTP Profile for Audio and Video Conferences with Minimal Control
29	IETF RFC 4733	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
30	IETF RFC 4734	Definition of Events for Modem, FAX and Text Telephony signals
31	IETF RFC 5806	Diversion indication in SIP
32	IETF RFC 6044	Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP)
33	IETF RFC 4244	An Extension to SIP for Request History Information
34	IETF RFC 3966	The Tel URI for Telephone Numbers
35	IETF draft	Applying Loose Routing to Session Initiation Protocol (SIP) User Agents (UA)" (expired)
36	Proximus	PXM IMS VoIP – UNI specification – Fax support
37	IETF RFC 5621	Message body handling in SIP
38	IETF RFC 5876	Updates to Asserted Identity in the Session Initiation Protocol (SIP)

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 it's 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:

- TS 182 025 Business Trunking; architecture and functional description", v2.1.1 (2008-09) [7]

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 1321   The MD5 Message-Digest Algorithm [26]
- RFC 3550   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]

  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.

- 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]

- Accept   according to [4] and [9]
- Call-Info   according to [4] and [9]
- RSeq   according to [4] and [10]
- RAck   according to [4] and [10]
- Allow-Events   according to [4] and [12]
- Event   according to [4] and [12]
- Subscription-State   according to [4] and [12]
- Refer-To   according to [4] and [20]
- Privacy   according to [4] and [16]
- P-Asserted-Identity   according to [4] and [17]
- Reason   according to [4] and [18]
- P-Called-Party-ID   according to [4] and [19]
- P-Access-Network-Info   according to [4] and [19]
- P-Associated-URI   according to [4] and [19]
- Replaces   according to [4] and [21]
- Referred-By   according to [4] and [22]
- Session Expires   according to [4] and [24]
- Min-Se   according to [4] and [24]

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-negociate/change the session parameters. So, if there is no need to re-negociate/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:
  - 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:
  - 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
  - sip:Norm-SubA-dn@IP-addr-PBXName, or
  - 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:
  - 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.
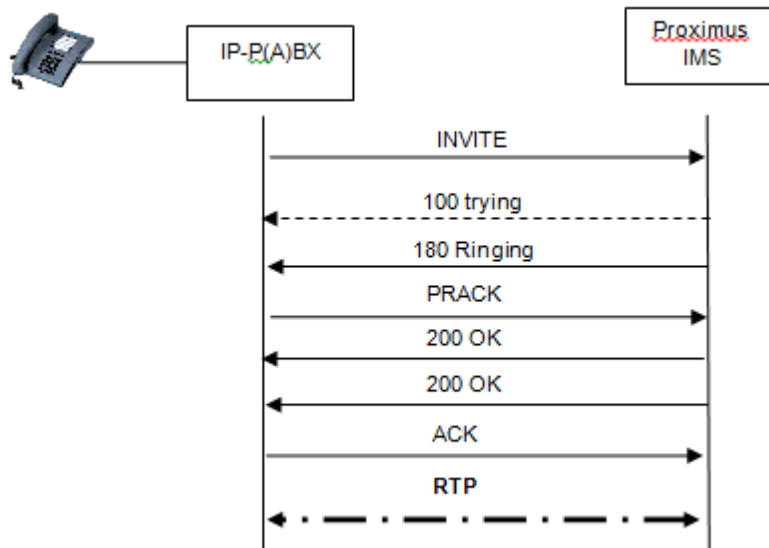

Figure 1 : Basic call/session set-up example public on-net

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

<u>Live example</u>

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=796b7734f7742acdf213d7c07f618d32
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=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): 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

Media Format: 106 [telephone-event]
    Media format specific parameters: 0-15
Media Attribute (a): fmtp:4 annexa=no
    Media Attribute Fieldname: fmtp
    Media Format: 4 [telephone-event]
    Media format specific parameters: annexa=no
Media Attribute (a): maxptime:90
    Media Attribute Fieldname: maxptime
    Media Attribute Value: 90
Media Attribute (a): silenceSupp:off - - - -
    Media Attribute Fieldname: silenceSupp
    Media Attribute Value: off - - - -

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>

<u>Live example</u>

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=z9hG4bK088b463e1a7b71dd2edac3b21b9cf5b1
  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=z9hG4bK088b463e1a7b71dd2edac3b21b9cf5b1
  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=1B5D32463135364139520000
  From: <sip:+3227979380@10.127.249.4;user=phone>;
      tag=796b7734f7742acdf213d7c07f618d32
  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=1B5D32463135364139520000
  From: <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=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.

<u>Live example</u>

```
Status-Line: SIP/2.0 200 OK
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=z9hG4bK088b463e1a7b71dd2edac3b21b9cf5b1
    CSeq: 1465254309 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,replaces,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.

<u>Live example</u>

```
Request-Line: ACK sip:10.127.249.190:5060;transport=udp SIP/2.0
Message Header
    Route: <sip:10.127.249.190;lr>
    Contact: "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=1B5D32463135364139520000
    From: "TEL 027979380" <sip:+3227979380@10.127.249.4;user=phone>;
        tag=796b7734f7742acdf213d7c07f618d32
    Call-ID: 49414ca83c459797570631be577686ab@10.127.249.4
    CSeq: 1465254309 ACK
    Via: SIP/2.0/UDP 10.127.249.4;rport;
        branch=z9hG4bK7cef8a6c77c13b84a6dd3b1c6023ea27
    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.
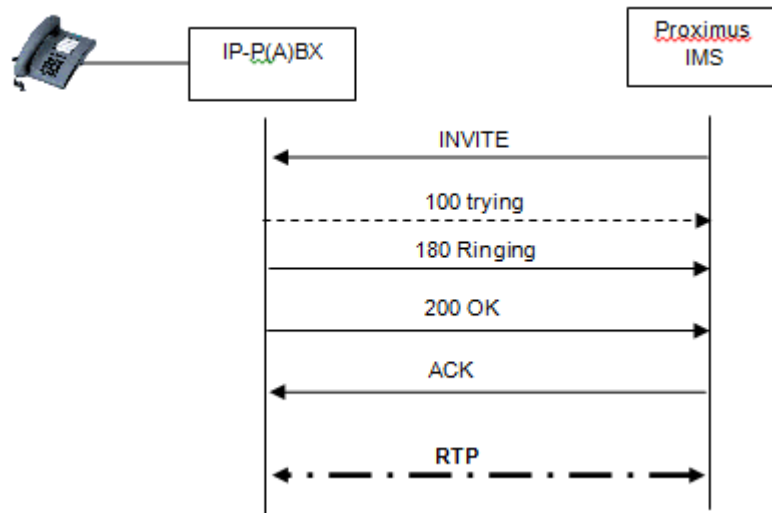

Figure 2: Basic call/session set-up example public off-net

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=z9hG4bKkqotd0nduhrdr90k3nauqe15l4
  Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.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:10.127.249.190:5060;transport=udp>
  P-Asserted-Identity: <sip:+32477143104@woe.proximus.be;user=phone>
  P-Charging-Vector: icid-value=S9C80-20101222152015-00000909;
    icid-generated-at=149.204.0.1;orig-ioi=sub.alcatel-ngnvoice.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

Media Attribute (a): rtpmap:101 TELEPHONE-EVENT/8000
   Media Attribute Fieldname: rtpmap
   Media Format: 101
   MIME Type: TELEPHONE-EVENT
Media Attribute (a): ptime:20
   Media Attribute Fieldname: ptime
   Media Attribute Value: 20
Media Attribute (a): maxptime:30
   Media Attribute Fieldname: maxptime
   Media Attribute Value: 30

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: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.sel.alcatel.de
  CSeq: 1 INVITE
  Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqotd0nduhrdr90k3nauqe15l4
  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=b43cffbe9aaa92a65b38e934ff10d37b
  From: <sip:+32477143104@woe.proximus.be;user=phone>;
    tag=D65E3246313536419B6E0000
  Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.sel.alcatel.de
  CSeq: 1 INVITE
  Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqotd0nduhrdr90k3nauqe15l4
  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=b43cffbe9aaa92a65b38e934ff10d37b
  From: <sip:+32477143104@woe.proximus.be;user=phone>;
     tag=D65E3246313536419B6E0000
  Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.sel.alcatel.de
  CSeq: 1 INVITE
  Via: SIP/2.0/UDP 10.127.249.190:5060;branch=z9hG4bKkqotd0nduhrdr90k3nauqe15l4
  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

Media Format: 101 [telephone-event]
Media format specific parameters: 0-15
Media Attribute (a): silenceSupp:off - - - -
Media Attribute Fieldname: silenceSupp
Media Attribute Value: off - - - -
Media Attribute (a): ptime:20
Media Attribute Fieldname: ptime
Media Attribute Value: 20
Media Attribute (a): maxptime:30
Media Attribute Fieldname: maxptime
Media Attribute Value: 30

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

<u>Live example</u>

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=z9hG4bKb7ekt8dekqmq15u0eedlcc70e1-g0g5
    P-Charging-Vector: icid-value=S9C80-20101222152015-00000909;
        icid-generated-at=149.204.0.1; orig-ioi=sub.alcatel-ngnvoice.com
    CSeq: 1 ACK
    To: <sip:+3227979380@ims.belgacom.be:5060;user=phone>;
        tag=b43cffbe9aaa92a65b38e934ff10d37b
    From: <sip:+32477143104@woe.proximus.be;user=phone>;
        tag=D65E3246313536419B6E0000
    Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.sel.alcatel.de
    Max-Forwards: 69
    Timestamp: 28396
    Content-Length: 0

6.3.3. Forward call tear down

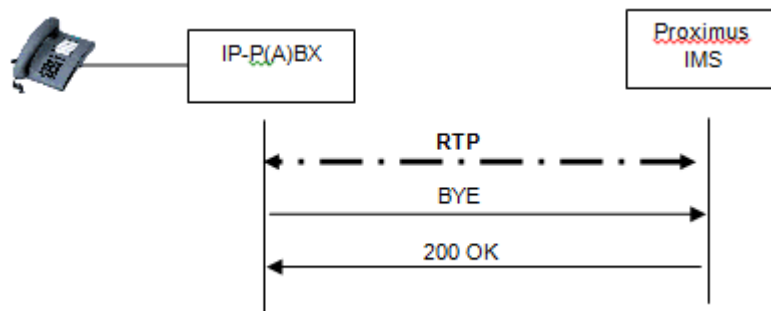<u>Message flow example</u>

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

<u>Live example</u>

```
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>;
        tag=796b7734f7742acdf213d7c07f618d32
    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.

<u>Live example</u>

```
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>;
        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=z9hG4bKebfeabbb1781f85942ca544c7b5ba1ec
    CSeq: 1465254311 BYE
    Content-Length: 0
```

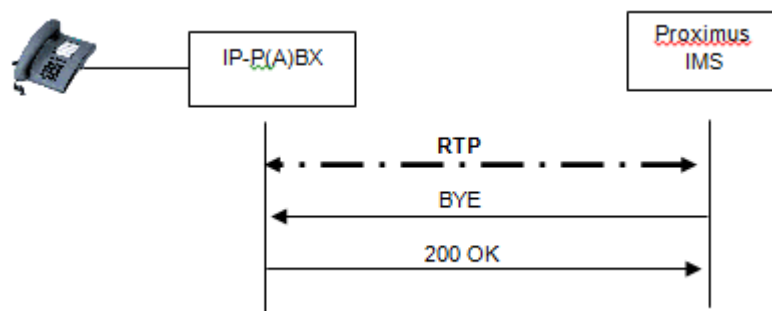## 6.3.4.   Backward call tear down

<u>Message flow example</u>

Figure 7 : Basic call/session teardown 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".

<u>Live example</u>

```
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=z9hG4bKkqotd0nduhrdr90k3nauqe15l4
  Call-ID: 01FF26AEC281400000000002@impmgsilab5-sig.stgl.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:10.127.249.190:5060;transport=udp>
  P-Asserted-Identity: <sip:+32477143104@woe.proximus.be;user=phone>
  P-Charging-Vector: icid-value=S9C80-20101222152015-00000909;
      icid-generated-at=149.204.0.1;orig-ioi=sub.alcatel-ngnvoice.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
Media Attribute (a): rtpmap:101 TELEPHONE-EVENT/8000
Media Attribute Fieldname: rtpmap
Media Format: 101
MIME Type: TELEPHONE-EVENT
Media Attribute (a): ptime:20
Media Attribute Fieldname: ptime
Media Attribute Value: 20
Media Attribute (a): maxptime:30
Media Attribute Fieldname: maxptime
Media Attribute Value: 30

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=796b7734f7742acdf213d7c07f618d32
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=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): 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
Media Format: 106 [telephone-event]
Media format specific parameters: 0-15
Media Attribute (a): fmtp:4 annexa=no
Media Attribute Fieldname: fmtp
Media Format: 4 [telephone-event]
Media format specific parameters: annexa=no
Media Attribute (a): maxptime:90
Media Attribute Fieldname: maxptime
Media Attribute Value: 90
Media Attribute (a): silenceSupp:off - - - -
Media Attribute Fieldname: silenceSupp
Media Attribute Value: off - - - -

Note: 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 but the call will not fail if received.

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-to 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.