

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Proximus can't be held responsible for any damages due to the use of an outdated version of this specification.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

IMS CORPORATE VoIP
General

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1. 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 14, 2011	First version
1.1	April 4, 2011	Correction 1: chapter 6.2.1: the SBC does not contain a child realm Correction 2: chapter 8.2.1: CAC in SBC is removed
1.2	May 16, 2011	Corrections to the cover sheet
1.3	April 22, 2014	
1.4	November 27, 2014	Codec negotiation details added
1.5	May 05, 2015	Clause "Tones and announcements" added
1.6	Januari 8, 2016	Clarification concerning codec lists, change of required bandwidth in chapter on QoS Changed "Belgacom" into "Proximus"
1.7	November 24, 2016	Update due to name change "Bizz IP telephony multi" into "enterprise voice multi"
2.0	May 7, 2019	Update to include aspects for "international business trunking"
2.1	May 25, 2019	Editorial correction in chapter 5.2.3 Belgacom/BGC replaced by Proximus/PXM Document name changed to PXM
2.2	June 06, 2019	Update after comments from IP-PBX team: Enterprise Voice Multi changed into Enterprise Voice / Multiline (aligned with commercial product names) Some corrections/clarifications on connectivity
2.3	September 19, 2019	Update of SW versions in Scope

2.4	May 02, 2024	Update of SW versions in Scope
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Table 1: document history

2. Scope

This document defines general aspects of the VoIP interface, of the Proximus IMS network, for IP-P(A)BXs.

This specification is applicable for Proximus business trunking services offered nationally as well as internationally.

This specification is applicable for the following IMS equipment and software packages:

- Nokia vIMS – software package Release 23.8
- Oracle vSBC – Software Version SCZ9.0.0 Patch 7
- 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 – SIP signalling – Business Trunking with IMS services [1]
- PXM IMS Corporate VoIP – UNI specification – SIP signalling – Wireless Office Extended [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 over IP [36]
- PXM IMS Corporate VoIP – UNI specification – Address templates [37]

3. 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.

3.1. Normative references

1	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Business Trunking with IMS services
2	Proximus	PXM IMS Corporate VoIP – UNI specification – SIP signalling – Wireless Office Extended
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 over IP
37	Proximus	PXM IMS Corporate VoIP – UNI specification – Address templates

Table 2: normative references

3.2. Informative references

11	BGC_D_48_9807_30_04_E.pdf	Information tones
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Table 3: informative references

4. Symbols, Definitions and Abbreviations

4.1. Symbols

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

None.

4.2. 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.

4.3. Abbreviations

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

3GPP	3th Generation Partnership Project
ADSL	Asymmetric Digital Subscriber Line
AKA	Also Known As
ATM	Asynchronous Transfer Mode
CAC	Call Admission Control
CE	Customer edge
Codec	Coder/decoder
PXM	Proximus
DHCP	Dynamic Host Configuration Protocol
DDI	Direct Dialling In
Ethane	Ethernet aggregation network
ETSI	European Telecommunications Standards Institute
FR	Frame relay
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol

IP-VPN	IP Virtual Private Network
LAN	Local area network
LSM	Lucent Session Manager
NAT	Network Address Translation
NGN	Next Generation Network
NNI	Network Network Interface
P(A)BX	Private (Automatic) Branch Exchange
PBX	Private Branch Exchange
PE	Provider Edge
QoS	Quality of Service
RFC	Request For Comment
RTP	Real Time Transport Protocol
SBC	Session Border Controller
SDP	Session Description Protocol
SDSL	Symmetric Digital Subscriber Line
SIP	Session Initiation Protocol
SRST	Survivable Remote Site Telephony
TS	Technical Specification
UNI	User Network Interface
VC	Virtual Circuit
VDSL	Very high bitrate Digital Subscriber Line
VLAN	Virtual LAN
VoIP	Voice over IP
VRF	Virtual Routing and Forwarding
WAN	Wide area network
xDSL	Any type of Digital subscriber Line

5. General

5.1. Structure of the document

The structure of this document mainly covers the following aspects:

- IP-P(A)BX types and operational modes
- IP connectivity: in this chapter necessary information for IP-P(A)BX vendors regarding IP connectivity will be present. This includes ao.
 - Access network types: a short description of the available access types.
 - User network Topology: information about the possible topologies at the user's network side
 - VLAN information
- Codec support: this chapter provides information about the supported codecs
- Quality of Service (QoS) and Call Admission Control (CAC): this chapter contains information regarding the CAC and QoS capabilities and mechanisms of the IMS.
- Naming, Numbering and addressing: this chapter contains information regarding the naming, numbering and (IP) addressing conventions and the capabilities and mechanisms needed in the IP-P(A)BX to comply.
- Load balancing
- Customer SBC

5.2. Corporate VoIP products

The general aspects covered in this document may differ according to the corporate VoIP product. In such case the concerned aspect will be discussed per product separately.

In IMS Corporate VoIP three products are envisaged:

- SIP Business Trunking (also known as Business Trunking NNI)
- SIP Enterprise Voice / Multiline (also known as Enterprise Voice Multi)
- SIP International Business Trunking

5.2.1. Business Trunking

Business Trunking is a connection for large IP-P(A)BXs. The IP-P(A)BX can access the IMS network via a so-called Network-to-Network interface aka the NNI mode or Non-registration based mode.

Business Trunking configurations can consist of 1 or more SIP trunks. Reasons to create multiple SIP trunks may be a multi-zonal numbering plan or a redundant access.

Business Trunking has 2 modes of operation:

- Business Trunking with IMS services. This offers advanced services for IP-P(A)BX users using the Broadsoft application server capabilities.
- Business Trunking without IMS services aka Wireless Office. This offers advanced services for IP-P(A)BX users using the Proximus mobile network capabilities. This does not make use of the Broadsoft application server capabilities but corresponds to a peering or transit routing scenario.

The behaviour for SIP Business Trunking with IMS services is the subject of the document "PXM IMS Corporate VoIP – NNI specification – SIP signalling – Business Trunking with IMS services".

The behaviour for SIP Business Trunking without IMS services is the subject of the document “PXM IMS Corporate VoIP – NNI specification – SIP signalling – Business Trunking without IMS services”.

5.2.2. Enterprise Voice / Multiline

Enterprise Voice / Multiline is a connection for smaller IP-P(A)BXs. The IP-P(A)BX can access the IMS network via a User-to-Network interface aka the UNI mode or Registration based mode.

Enterprise Voice / Multiline configurations consist of 1 SIP trunk.

The behaviour for SIP Enterprise Voice / Multiline is the subject of the document “PXM IMS Corporate VoIP – UNI specification – SIP signalling – Enterprise Voice multi”

5.2.3. International Business Trunking

International Business Trunking is a connection for large IP-P(A)BXs. The IP-P(A)BX can access the IMS network via a so-called Network-to-Network interface aka the NNI mode or Non-registration based mode.

International Business Trunking configurations can consist of 1 or more SIP trunks. Reasons to create a multiple SIP trunks may be multiple countries to be covered and/or a redundant access.

International Business Trunking always comes with IMS services. This offers advanced services for IP-P(A)BX users using the Broadsoft application server capabilities.

The behaviour for SIP International Business Trunking with IMS services is the subject of the document “PXM IMS Corporate VoIP – UNI specification – SIP signalling – Business Trunking with IMS services”.

5.3. Reference point

All characteristics described in this set of UNI-specifications are applicable at the reference points indicated in the figures below.

5.3.1. Business Trunking



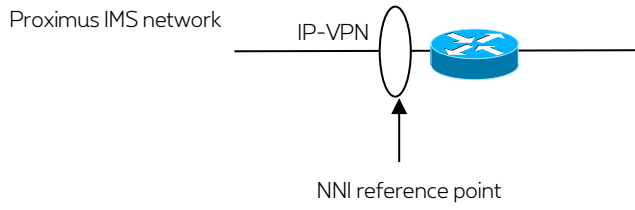


Figure 1 : Business trunking reference point

5.3.2. Enterprise Voice multi

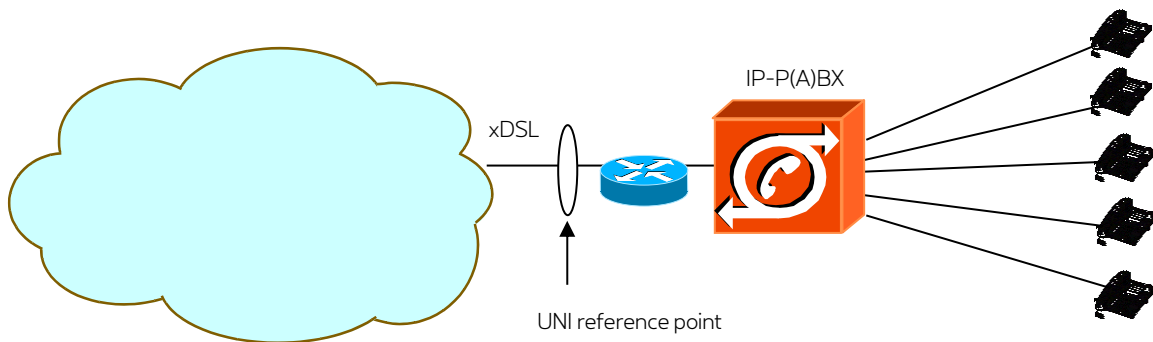


Figure 2 :Enterprise Voice / Multiline reference point

5.3.3. International Business Trunking

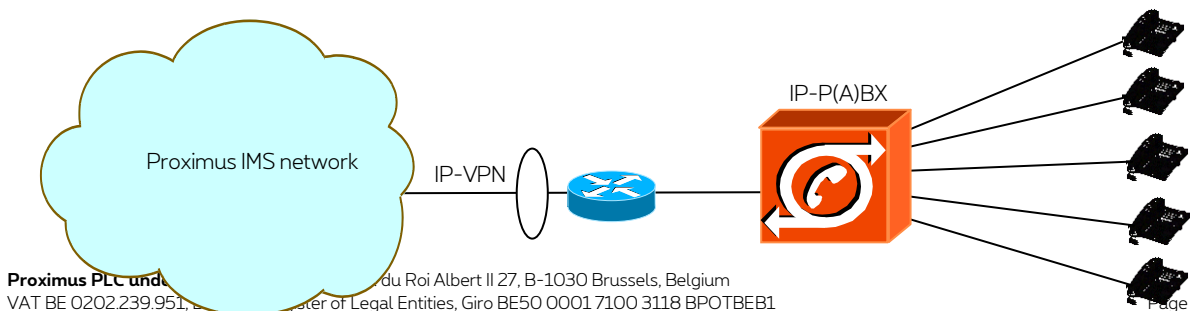




Figure 3 : International Business trunking reference point

6. IP-P(A)BX types and operational modes

6.1. IP-P(A)BX types

The terminology of PBX and PABX refers back to the PSTN/ISDN world, where the automation level of the user's PBX was an important difference with regard to interconnection. The following types of customer equipment can be distinguished.

1. An **IP-PABX** (IP Private Automatic Branch Exchange) is used in case we are dealing with larger equipment, able to handle all telephonic features and services we can expect from an advanced technology. Important in this case is the handling of DDI functionality, where the IP-PABX is connected by means of SIP trunk(s). The identification of an IP-PABX is done by making use of the one to one mapping of (fixed) IP-addresses.
2. **IP-PBX** (IP Private Branch Exchange). This mnemonic is used in case we are dealing with smaller equipment, able to handle most important telephonic features and services we can expect from a lower range technology.

Note: in general, we use the mnemonic **IP-P(A)BX**.

6.2. Operational modes

The following operational modes for IP-P(A)BX can be distinguished. Both are allowed, but with preference for the Non-Proxy case.

1. **Non-Proxy IP-P(A)BX** (IP-PABX without NAT). Concretely this means that the IP-PABX acts as a relay only for the signalling part of the phone call, the voice traffic between the sender (IP) phone and the receiver (IP) phone is not relayed through the IP-P(A)BX (except for some cases where media resources need to be allocated in IP-P(A)BX).
2. **Proxy IP-P(A)BX**. Signalling and media are always passing the IP-P(A)BX. This way of working can have negative impact on the bandwidth of IP network, and consumes extra unnecessary resources in the IP-P(A)BX.

7. IP Connectivity

7.1. Introduction

In order to gain access to the Proximus IMS Network IP-P(A)BXs can use different types of IP connectivity. It is obvious that depending on the size of the IP-P(A)BX some access types are unusable, e.g. a very large IP-P(A)BX shall not be connected over one xDSL line.

Different topologies (e.g. single IP-P(A)BX or multiple IP-P(A)BXs) at the user's side can be supported, depending on the access network.

2 types of access network are possible:

- Explore VPN, for Business Trunking and International Business Trunking, addressed to large IP-P(A)BXs
- Public Internet for Enterprise Voice / Multiline, addressed to smaller IP-P(A)BXs

Note: Enterprise Voice / Multiline is limited to 1 IP-P(A)BX per connection.

Access Network versus Product	Enterprise Voice / Multiline	Business Trunking
Explore VPN	Supported	Supported
Internet xDSL	Supported	Not supported

7.2. Access network types

7.2.1. Explore VPN

The following IPVPN access methods are available:

- **Ethernet** access with the adequate end-2-end Quality of Service;
- **Leased line** access with the adequate end-2-end Quality of Service;
- **SDSL** access with the adequate end-2-end Quality of Service;
- **ADSL/VDSL Office** are possible by the implementation of IP QoS prioritisation but without bandwidth guarantees.

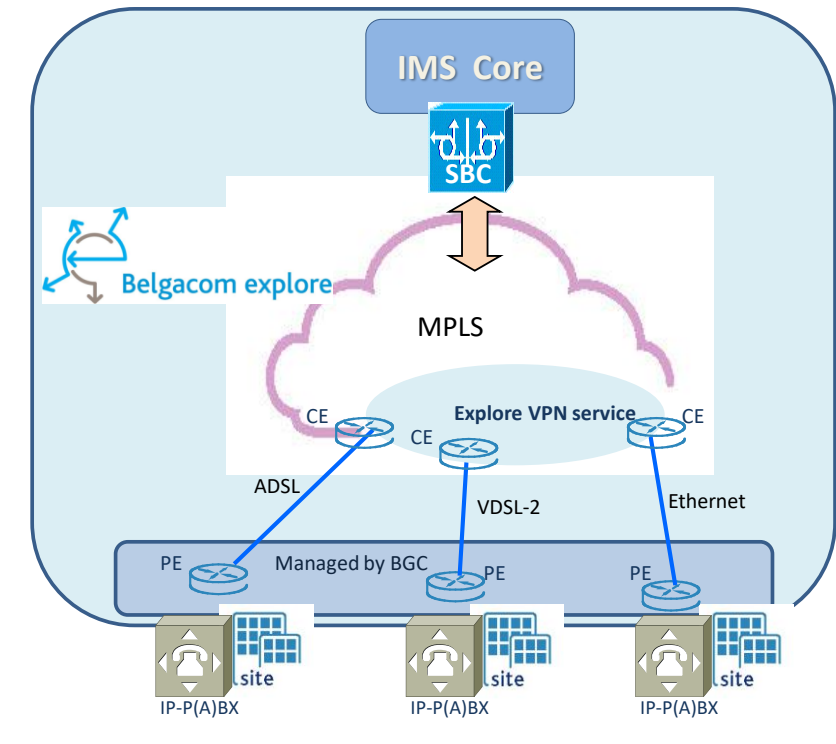


Figure 3 : Explore VPN access methods

Each customer (defined by an IP VPN VRF) will have a dedicated VLAN between the Explore IP VPN PE's and the IMS Catalysts.

Each customer's IP VPN will also be identified at the access side (aka the south side) of the Acme SBC by the following element:

- A Parent Realm (or context) in the SBC defining the customer's VPN and ensuring its privacy and security;

7.2.2. Internet Access

The following Multi access methods are available :

- ADSL
- VDSL2

7.2.2.1. ADSL

When IP –P(A)BX is connected to Public Internet via ADSL, the ADSL modem/router will receive a dynamic IP address using DHCP. Between the ADSL modem/router and the Proximus IMS, a dedicated QoS PVC is used : this dedicated PVC provides QoS treatment of the voice traffic over the internet traffic, which uses another VC named Internet PVC.

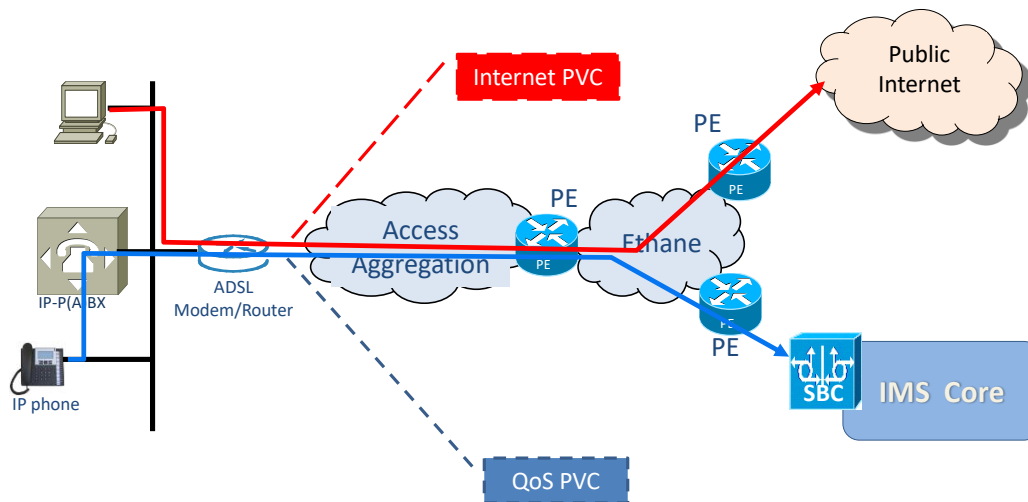


Figure 4 : ADSL internet access method

On IP QoS level, each voice packet is marked as priority by IP-PBX (via DSCP value of the Type of Service field of IP header), then forwarded towards the priority queue (expedited forwarding).

The classification criteria for VoIP traffic on the ADSL modem/router is based on destination IP address written in each IP packet.

In case the QoS PVC is unavailable, a fallback mechanism has been implemented enabling to route the voice traffic via Internet PVC (without QoS).

The ADSL modem/router is doing NAT without a SIP ALG active.

The IP Phone (System Phone) takes contact with the IP P(A)BX. The IP P(A)BX translates then this call to a SIP call towards IMS and its group SIP account, located on the IP P(A)BX itself.

IP-P(A)BX uses :

- port 5060 for SIP signalling of the VoIP calls,
- fixed range of UDP ports 16384 – 16399 for the RTP stream of the VoIP calls.

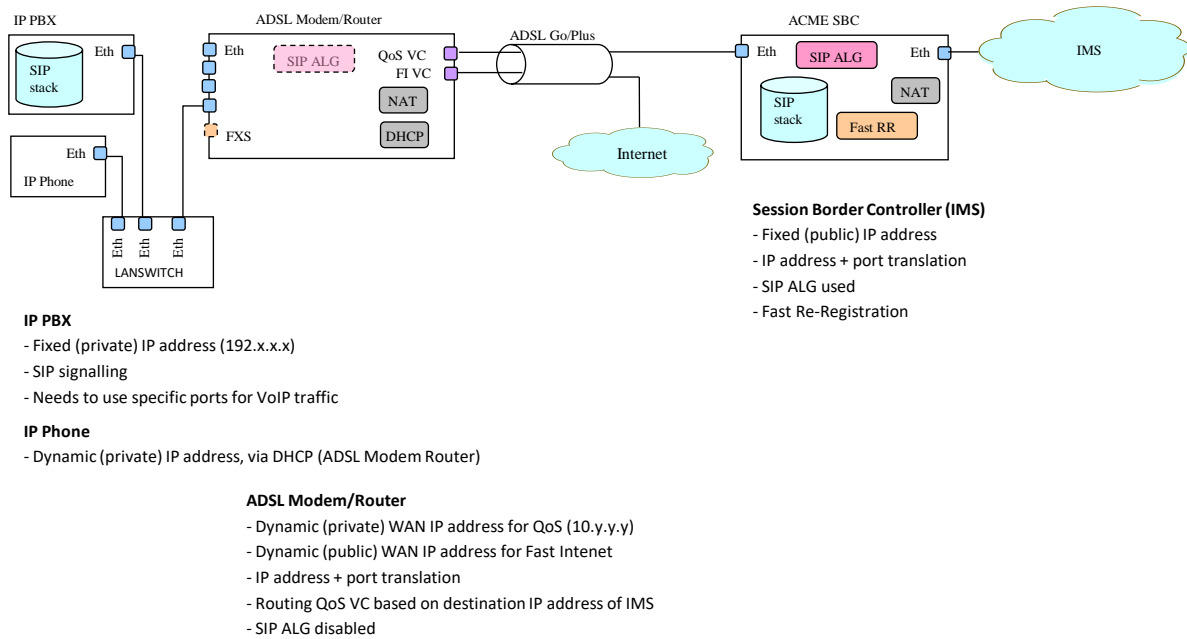


Figure 5 : ADSL internet access functional overview

7.2.2.2. VDSL

When IP –P(A)BX is connected to Public Internet via VDSL, the VDSL modem/router will receive a dynamic IP address using DHCP. Between the VDSL modem/router and the Proximus IMS, a dedicated QoS VLAN is used : this dedicated VLAN provides QoS treatment of the voice traffic over the internet traffic, which uses another VC named Internet PVC.

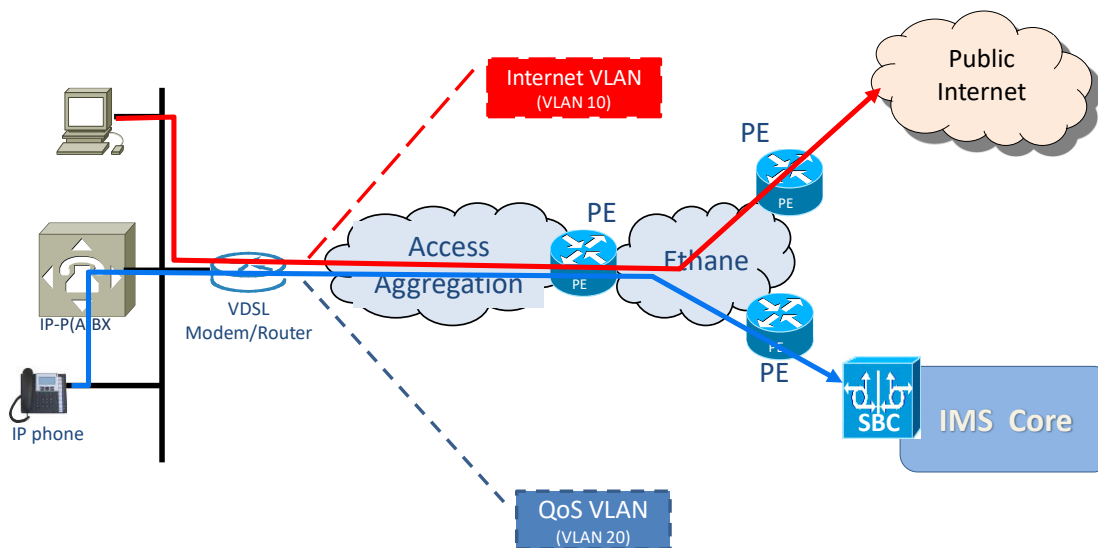


Figure 6 : VDSL internet access method

On IP QoS level, each voice packet is marked as priority by IP-PBX (via DSCP value of the Type of Service field of IP header), then forwarded towards the priority queue (expedited forwarding).

The classification criteria for VoIP traffic on the VDSL modem/router is based on destination IP address written in each IP packet.

In case the QoS VLAN is unavailable, a fallback mechanism has been implemented enabling to route the voice traffic via Internet VLAN (without QoS).

The VDSL2 modem/router is doing NAT with a SIP ALG active.

The IP Phone (System Phone) takes contact with the IP P(A)BX. The IP P(A)BX translates then this call to a SIP call towards IMS and its group SIP account, located on the IP P(A)BX itself.

IP-P(A)BX uses :

- port 5060 for SIP signalling of the VoIP calls,
- fixed range of UDP ports 16384 – 16399 for the RTP stream of the VoIP calls.

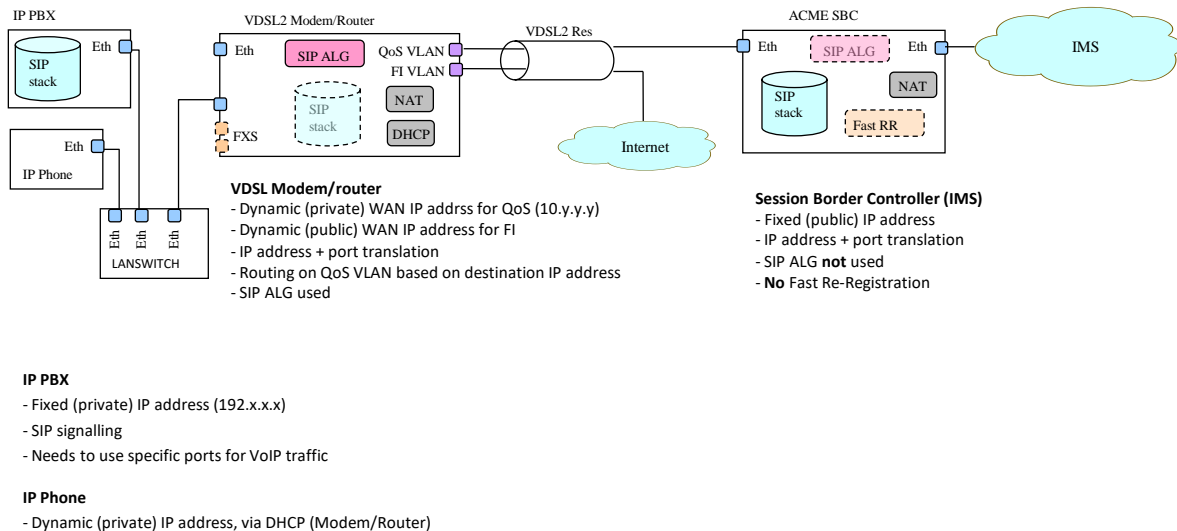


Figure 7 : VDSL internet access functional overview

7.3. User network topology

7.3.1. General

The following cases can be highlighted according to the structure of customer site.

7.3.2. Single IP-P(A)BX

In this case the customer enterprise spans only one site, where the IP-P(A)BX is located; the routing plan is simplified and centralized.

7.3.3. Multiple IP-P(A)BXs

in this case the enterprise can span a number of sites, where we can have:

- Multi-IP-P(A)BX centralized: the different sites are interconnected by a WAN, so that all remote sites can reach call handling resources at the main site..
- Multi-IP-P(A)BX distributed: some sites (or all sites) have its own call handling resources; the routing plan to be configured on each IP-P(A)BX is more complex and requires coordination when changes apply (routing between sites for private calls are in charge of IP- P(A)BX).

8. Codecs

In principle the Proximus IMS network does not restrict the codecs that can be used. However, in order to ensure interoperability with all types of VoIP endpoints (e.g. CPEs, IP-P(A)BXs, media gateways,...) every IP-P(A)BX shall support the following codecs:

- G.711 A-law
- G.729

Remark: for the support of T.38 for fax see “PXM IMS VoIP - UNI specification - FAX over IP v1.0.docx” [36]

The recommended and preferred packetisation time is 20 msec. In any case it shall **not** be higher than 30 msec.

8.1. Originating calls

In practice an IP-P(A)PBX is allowed to use any codec (e.g. G.722) as preferred codec as long as the two above mentioned codecs are offered as well. Both mandatory codecs shall be put in the SDP part of the SIP signalling message no later than at the 3th and 4th place, leaving room for 2 other more preferred codecs. The IP-P(A)BX is allowed to send a maximum of 8 codecs per m-line and/or maximum 16 SDP attributes in its SDP offer. Proximus does not guarantee the correct working when using other codecs.

When using another codec than the 2 mandatory codecs it shall be taken into account that the Proximus network will not guarantee the codec is used towards all destinations. In such cases “fallback” to any of the 2 mandatory codecs will happen.

Remark: In order to ensure the correct working of the “fallback” procedure some tests shall be done during the certification of an IP-P(A)BX.

8.2. Terminating calls

When receiving a call the IP-P(A)BX shall select one codec from the SDP offer received and put this codec in the SDP answer. In case more than one codec is sent back in the SDP answer the IMS network may narrow down the codec list via a re-INVITE transaction.

Under no circumstances the IP-P(A)BX shall sent an SDP answer with more than one codec in a reversed order than the SDP offer received.

Any codec, received in the SDP offer, and not supported by the IP-P(A)BX, shall be ignored as described in IETF RFC 3264 [11]. The call **MUST** not be rejected unless there is none of the offered codecs supported by the IP-P(A)BX.

9. Quality of service and Call admission control

9.1. Quality of Service (QoS)

Besides the support of QoS marking of the VoIP packets, one important issue regarding QoS is the dimensioning of the access in terms of available bandwidth. The dimensioning of the number of voice channels depends of customer's choice. Proximus will recommend a maximum number of simultaneous calls supported in function of the available guaranteed bandwidth.

When using the **G.729a codec**, the bandwidth requirement per voice channel (or simultaneous call) is:

- **30Kbps** on a **FR** link which includes Ethernet and Leased line access methods
- **45Kbps** on an **ATM** link which includes xDSL access methods

When using the **G.711 A-law codec**, the bandwidth requirement per voice channel (or simultaneous call) is:

- **100Kbps** on a **FR** link which includes Ethernet and Leased line access methods
- **110Kbps** on an **ATM** link which includes xDSL access methods

Important remark about fax calls: Fax calls can use the G.711 a-law codec or the T.38 codec. For more information see "PXM IMS VoIP – UNI specification – Fax over IP" [36].

It is thus very important to take into account these parameters to properly dimension the bandwidth per site towards the VoIP platform.

Important remark about Proxy/Non-Proxy IP-PABX:

In case of Non-Proxy IP-P(A)BX the voice traffic between the sender phone and the receiver phone is not relayed through the IP-P(A)BX.

In case of Proxy IP-P(A)BX, the signalling and the voice traffic between the sender phone and the receiver phone are relayed through the IP-P(A)BX.

This difference has an impact on the bandwidth of the site where the IP-P(A)BX is located since 1 call between 2 remote phones uses 2 voice channel on the site where the IP-P(A)BX is located.

9.2. Call admission Control

Call admission control means controlling the number of simultaneous calls in an IP-P(A)BX configuration. This can be applied at 2 different locations:

- Call limiter on IMS application server: does not take into account the bandwidth usage but is rather a static configuration
- CAC in IP-P(A)BX.

9.2.1. Call limiter on IMS application server

The call limiter on IMS application server is an administrative parameter.

Broadsoft has parameter settings which can limit the number of calls, for the customer/enterprise in general, and per SIP trunk.

The customer/enterprise call limit parameter determines the number of call licenses for the complete enterprise. It has a direct relation with the license cost.

The SIP trunk call limit parameter determines the number of simultaneous calls allowed on that SIP trunk.

Remark: in case of multiple SIP trunks, the enterprise call limit shall be equal to or greater than the sum of SIP trunk call limits.

9.2.2. Call admission control on IP-PABX

Some IP-P(A)BX types are also supporting the call admission control. The feature implementation is strongly depending on vendor, and does not fit in the scope of this specification. Important to know is that the only point in the chain for Corporate VoIP environment is clearly the point of call handling: IP-PABX. The IP-PABX is aware of all call handling and connectivity. This is the most logic point to apply any kind of CAC.

10. Naming, Numbering and Addressing

10.1. Naming

10.1.1. General

Each IP-P(A)BX has one pbxPUID or PilotID. This is a non-dialable public identity.

Example: MycompanyPBX1@ims.belgacom.be

10.1.2. Business Trunking

to be completed

10.1.3. Enterprise Voice / Multiline

to be completed

10.1.4. International Business Trunking

to be completed

10.2. Numbering

10.2.1. General

An IP-P(A)BX has a series of number ranges containing consecutive numbers or individual numbers, which are all public dialable numbers.

Ranges always start with a number ending with 0 (for 10), 00 (for 100), 000 (for 1000) etc.

10.2.2. Business Trunking

For Business Trunking, there can be ranges of 10, 100, 1000, 10 000 or 100 000 consecutive numbers.

Note: A range of 1 number (=an individual number) can be added to a business trunk

10.2.3. Enterprise Voice / Multiline

For Enterprise Voice / Multiline, only ranges of 10 or 100 numbers are supported (but this could evolve in the future).

Note: A range of 1 number (=an individual number) can be added to an Enterprise Voice / Multiline

10.2.4. International Business Trunking

For International Business Trunking, there can be ranges of 10, 100, 1000, 10 000 or 100 000 consecutive numbers.

10.1. IP addressing

10.1.1. Explore VPN

to be completed

10.1.2. Internet Access

to be completed.

11. Load balancing

Load balancing capabilities are currently under development and will be detailed here in a later version of the document.

12. Customer SBC

This version of the corporate VoIP UNI specifications do not take into account the presence of an SBC function at the customer premises.

For the support and remote management of such customer SBC possibly an update of this specification is needed.

13. Tones and announcements

Proximus specification “BGC_D_48_9807_30_04_E” [I1] specifies the characteristics of tones generated by the Proximus PSTN/ISDN network. These characteristics serve as the basis for tones generated in the Proximus IMS network.

However, these characteristics can not be guaranteed because:

- In IMS tones are provided by playing recorded tones in .WAV files. This involves conversion of the tones generated in PSTN/ISDN to .WAV files.
- Just like in PSTN/ISDN tones may be generated by another network e.g. in case the destination is located in the network of another operator. This other network may use different characteristics for the tones.

A copy of the .WAV files can be obtained on request.

14. Emergency handling

14.1. National Business Trunking

Emergency calls originated inside Belgium, with a Belgian calling line identity, need to be routed by the Proximus IMS network to the correct emergency center based on the geographical location of the IP-P(A)BX.

Emergency calls can be offered by the IP-P(A)BX to the Proximus IMS network in two ways:

- Without emergency access prefix: The call routing is based on **static localization** through the definition of a local area parameter in the Proximus IMS network. The parameter defines the ZIP code related to the installation address of the SIP DDI. The calls are routed based on the ZIP code. This solution is a static way to route the calls to the emergency center. Nomadism of SIP lines is not taken into account.
- With emergency access prefix: The call routing is based on emergency routing by using the **1992** access prefix. The IP-P(A)BX **shall** provide information about the localisation (i.e. the ZIP code) of the user making the emergency call.

Note: Please refer to “PXM IMS Corporate VoIP – UNI specification – Address templates” [37] chapter 6.2.1 for details about the number format to be used for 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).

14.2. International business trunking

Emergency calls originated outside Belgium, with a foreign calling line identity (i.e. with a valid E.164 number in international format in the SIP From header), need to be routed by the Proximus IMS network to the country of origin in order to be delivered to the correct emergency center.

Note: Please refer to “PXM IMS Corporate VoIP – UNI specification – Address templates” [37] chapter 7.1 for details about the number format to be used for emergency calls

14.3. Enterprise Voice / Multiline

Emergency calls originated from an Enterprise Voice / Multiline will be routed by the Proximus IMS network to the correct emergency center based on the geographical location of the IP-P(A)BX. As Enterprise Voice / Multiline includes one and only one physical installation, the ZIP code of the physical installation address is stored in the Proximus IMS network and used for this routing.

Note: the mechanism used for National Business Trunking i.e. with access prefix 1992 is NOT supported for Enterprise Voice / Multiline