LAN requirements for CallConnect

<u>Remark</u>: In this document the words: MUST, MUST NOT, REQUIRED, SHALL, 'SHALL NOT, SHOULD, SHOULD NOT, RECOMMENDED, MAY and OPTIONAL has the meaning as defined in RFC 2119.

1. Introduction

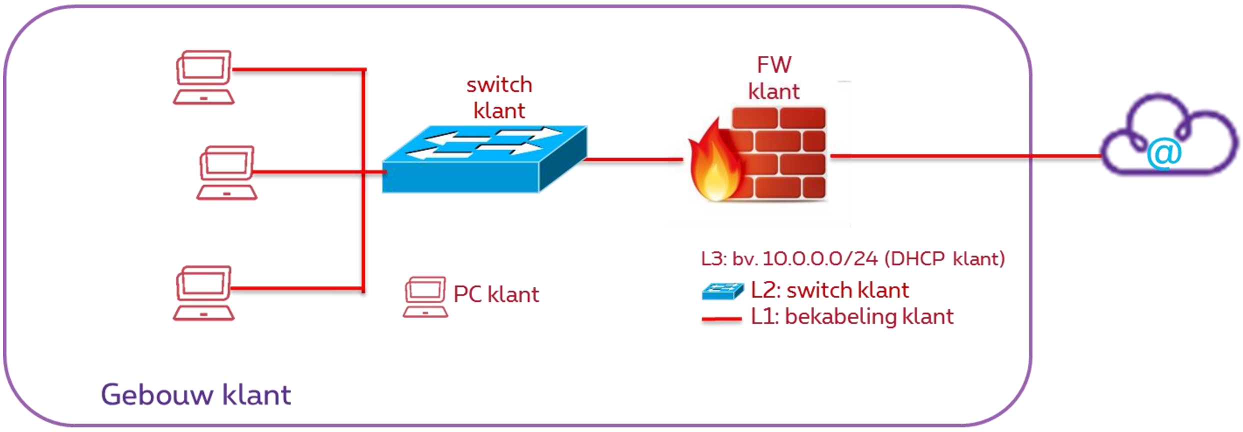
The LAN-environments of customers are rather complex and not maintained by Proximus. The integration of the IP Phones on the existing infrastructure means that this infrastructure must be compatible with the Call Connect service. The purpose of this document is to define the minimum requirements of the LAN to support the Call Connect service of Proximus. This document will highlight the weak points and so inform the customer to make the necessary adaptations before the installation.

2. Inquiry requirements

The customer needs to start-up this inquiry with the help of this document and answers the compatibility questions. If the customer cannot handle this inquiry then he needs to contact his IT responsible. With the agreement of the minimum requirements of Proximus, the customer commits that his LAN network is compatible for the Call Connect service.

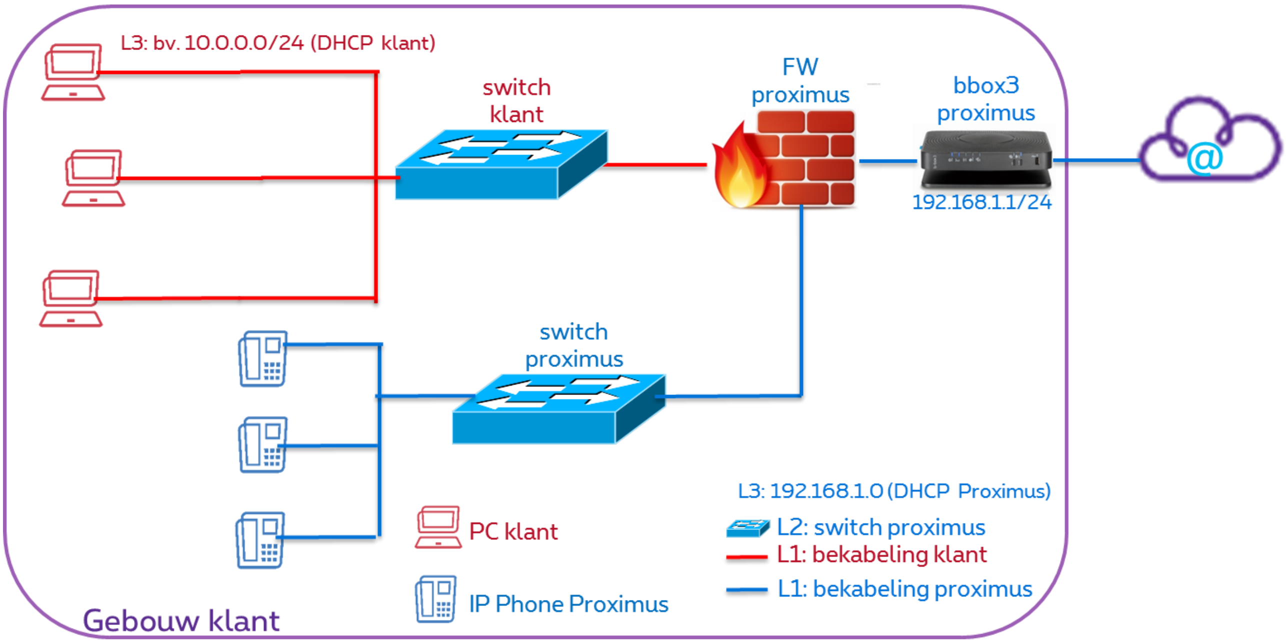
Proximus doesn't validate the received information and takes no responsibility to make the LAN compatible with these minimum requirements. Customer needs also to understand that during the installation process or repair process Proximus asks a tariff for the corrections on the network.

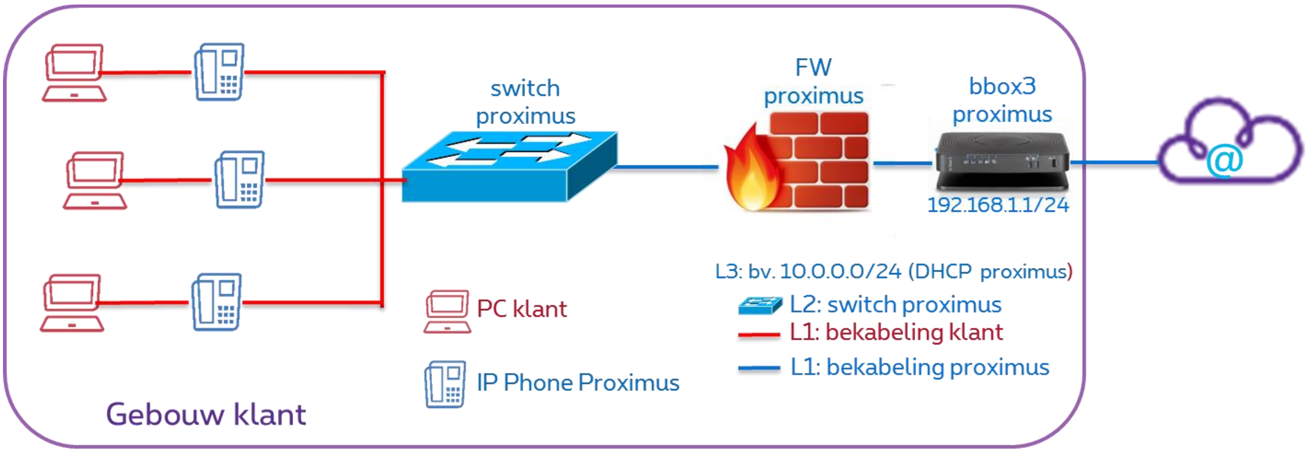
3. Topology before the order



4. The topology with firewall and IP-telephony from Proximus

There are 2 possibilities:





<u>The minimum requirements for Proximus are:</u>

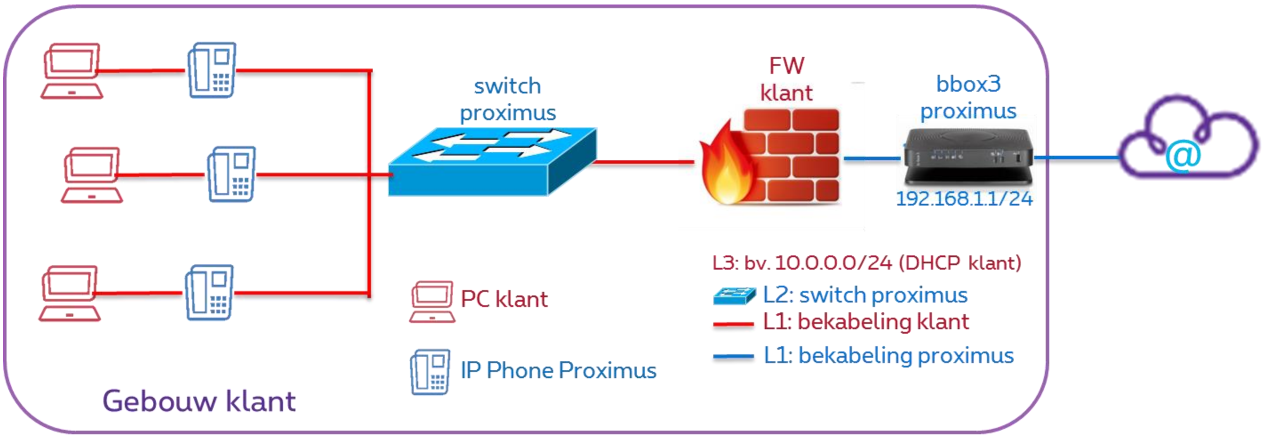
- The cable used between IP Phone and the LAN switch must be a certified CAT5-cable (max cable length of 100 meter).
- The PC must only use a best effort classification (IP precedence = 0) for his data traffic. The value 3 is reserved for voice signalling and 5 for RTP.
- When a Soft Phone application is used on PC then the application will do the right colouring.

- The DHCP server is from Proximus.
- Only IP-traffic allowed on the network if the customer has his own firewall. The IP address range of the customer must not be in conflict with the Proximus IP phones (see annex).

There needs to be 240V AC power outlets available (at least 3) and also a physical place for the switch and/or IP Phones.

5. Topology with the firewall of the customer

5.1 With NAT active on the FW



<u>Minimum requirements for Proximus:</u>

- **Use always the SIP ALG function of the FW. Only when you have problems with it then you can try the different alternatives described below.**

- The cable used between IP Phone and the LAN switch must be a certified CAT5-cable (max cable length of 100 meter).

- The PC must only use a best effort classification (IP precedence = 0) for his data traffic. The value 3 is reserved for voice signalling and 5 for RTP.

- When a Soft Phone application is used on PC then the application will do the right colouring.

- DHCP-leasetime on the firewall must be 24 hour
- The DNS-servers of Proximus needs to be configured (195.238.2.21 and 195.238.2.22).
- No multicast allowed.
- Only IP-traffic allowed on the network. The IP address range of the customer must not be in conflict with the Proximus IP phones (see annex).

- If the customer uses a VLAN trunk (802.1q) then the VoIP VLAN needs to be handled with the highest priority.
- Firewall requirements:
  - If the firewall uses the NAT-functionality then the customer confirms that he uses one of the families defined in RFC3489.
  - Can prioritize the voice in case of congestion
- There needs to be 240V AC power outlets available (at least 3) and also a physical place for the switch and/or IP Phones.

Rules to adapt on the firewall

1/ SIP-traffic from IP Phone to the BBOX			
Source port	Destination IP-address	Destination port	Protocol
UDP 5060 - 5069	81.247.28.224/28 80.200.255.208/28	UDP 5060	SIP
All	81.247.28.224/28 80.200.255.208/28	TCP 5061	SIPS

2/ RTP-traffic from IP Phone to the BBOX			
Source port	Destination IP-address	Destination port	Protocol
UDP 16384 – 16399	81.247.28.224/28 80.200.255.208/28	All	RTP, RTCP

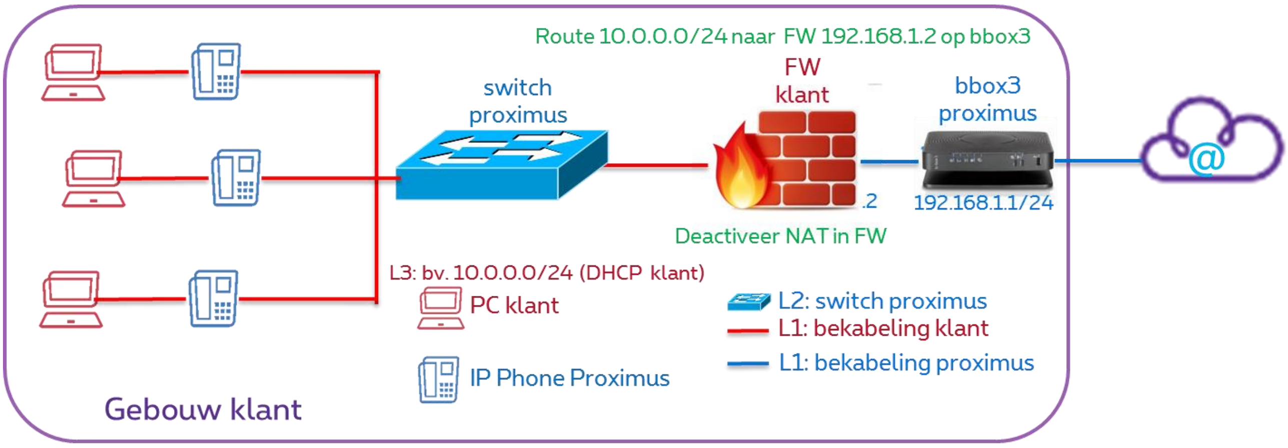
3/ Remote configuration of the IP Phone and extended services (DMS & XSP)			
Source port	Destination IP-address	Destination Port	Protocol
All	rps.yealink.com ccdms.proximus.be ccxsp.proximus.be 81.245.3.234 81.245.3.226 81.240.251.52	TCP 443	https

4/ Remote management van Linksys LAN switch			
Source port	Destination IP-address	Destination port	Protocol
all	195.13.30.54 217.136.223.101 194.78.192.5 80.200.253.149 87.66.1.247 (port forwarding 10443:443 linksys)	443 on LAN-switch	https

6/ NTP			
Source port	Destination IP-address	Destination port	Protocol
All	cn.pool.ntp.org ntp.belbone.be, ntp1.belbone.be.	123	ntp

5.2 Without NAT on the FW

When NAT on the FW gives problems with the SIP registration then the following option can be considered.

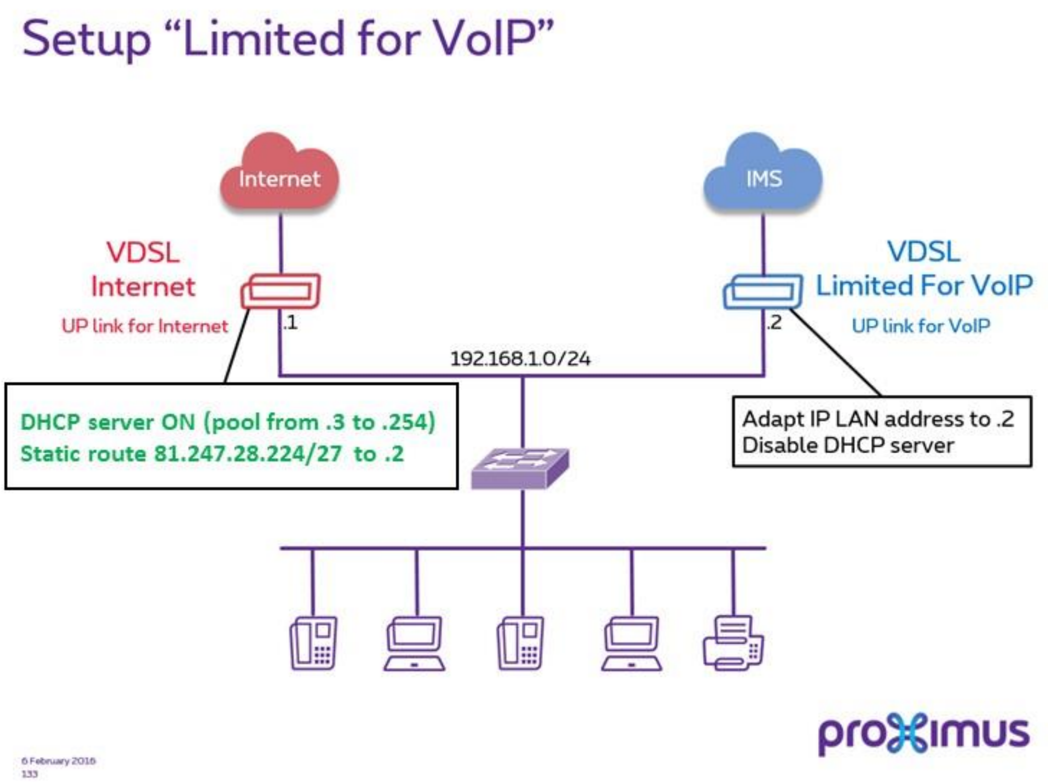


Disable the NAT on the FW and activate a static route on the bbox3 towards the FW. Example: route subnet 10.0.0.0/24 towards the FW.

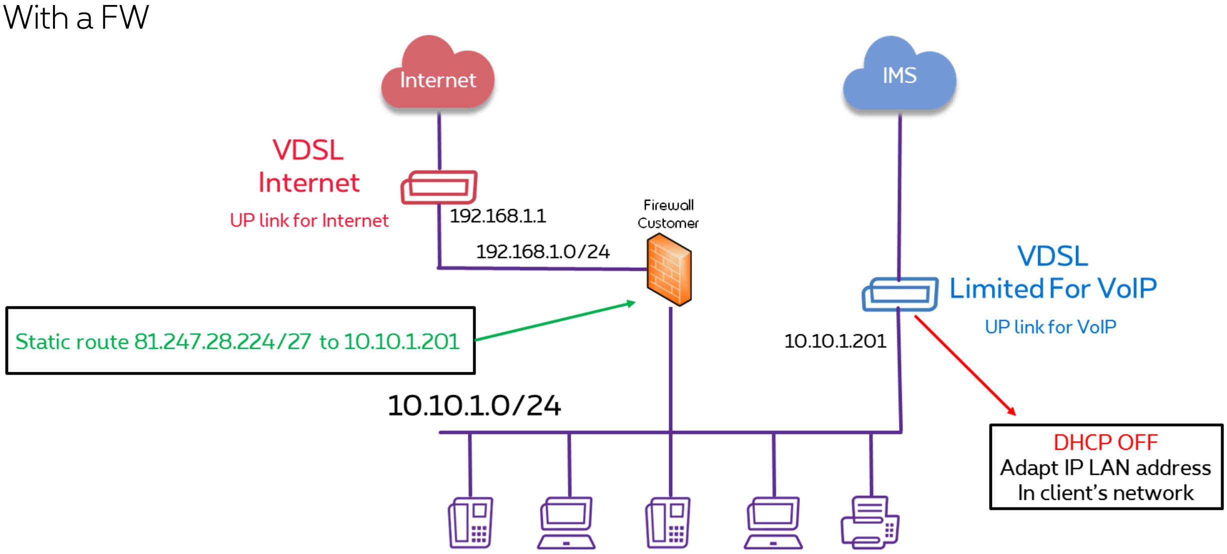
The command on the Technicolor is:
. :ip rtadd dst 10.0.0.0/24 gateway 192.168.1.2
:saveall

6. With a Limited for VoIP line

There is always a connection to Internet needed for CallConnect!



With a FW



The best place to configure this extra route towards the limited for Voip line is the on the firewall itself. NAT problems on the external FW with SIP can be avoided by placing the Limited for VoIP line *on the same LAN* of the IP Phones!

7. End-to-end requirements for VoIP

- The total packet loss must be lower than 1%
- The total jitter must be mower then 20ms.
- The total delay must be lower than 150ms.

8. Annex

The reserved IP address space for Proximus on the bbox3 are:
10.192.0.0 => 10.223.255.255 (/11)
10.88.0.0 => 10.91.255.255 (/14)
10.92.0.0 => 10.95.255.255 (/14)
10.128.0.0 => 10.159.255.255 (/11)

from10.168.0.0 => 10.168.255.255 (/16)
10.169.0.0 => 10.169.255.255 (/16)
...
to 10.239.0.0 => 10.239.255.255 (/16)

The colouring of the IP packets on the Phones are the following:
TOS = 46 for RTP, TOS = 26 for SIP