



# **FULL VOIP SYSTEM : PREREQUISITES, REQUIREMENTS & RECOMMENDATIONS**

## Document history

Version	Author(s)	Modifications
0.1	Maarten Bossuyt	Initial version of the document
1.0	Maarten Bossuyt	Document release
2.01	Robbe VDG	Chapter reorganization
...		
2.10	Lydia Biey	Internet Explorer when using webscreens
2.12	Robbe VDG	Updated Minimum OS
2.14	Johan Minne	General update

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## 1. Introduction

CTArchitect is a full-blown contact centre solution that:

- Manages all your inbound traffic
- Activates your outbound campaigns
- Ensures that you can monitor your quality and productivity

Thanks to its VoIP implementation, both on agent side and public network side, this system will bring your operational systems to a higher level of flexibility and manageability. From a technical point of view however, VoIP technology brings along new challenges. In order to help you with these technical challenges, we created this document. In what follows, we will focus on:

- The technical impact of a VoIP implementation on your network
- A couple of recommendations and guidelines for the setup of your environment, this on all levels: network layout, server infrastructure and agent equipment
- The quality controls we put in place to ensure that your VoIP telephony provider is acting in accordance with the standards needed in a professional environment

## 2. VoIP impact on infrastructure

Implementing a VoIP setup has an impact on several levels. A (non-exhaustive) list of topics that need to be taken into account when building a similar system can be found below.

First of all, the technology is subject to the constraints of the available bandwidth on the local (LAN) or wide area network (WAN). Every single VoIP call (agent-CTArchitect, supervisor-CTArchitect or CTArchitect-Phone network) requires a certain amount of bandwidth. The needed capacity mainly depends on 2 factors:

- Chosen compression codec (G.711 or G.729)
- Voice activity detection (VAD)

Impact on required bandwidth is as follows:

Codec type	Voice activity detection	Estimated mean bandwidth usage (per call)
G711 A-law	Off	90 Kbps
G711 A-law	On	55 Kbps
G729A	Off	33 Kbps
G729AB	On	24 Kbps

Please bear in mind that the preferred codec choice might have an impact on voice quality. More compression could result in inferior voice/audio quality. This last remark is also valid if additional/external compression is applied to the voice flow.

CoS, or Class of Service, has an impact as well. CoS refers to the process of tagging and labelling network traffic, in order to differentiate, identify and separate various types of traffic signals into various classes or groups. Ideally, CoS 1 must be reached while CoS 2 is just acceptable. Level CoS 3 is not compatible with VoIP (too much delay & packet loss).

Settings	CoS 1	CoS 2	CoS 3
Round Trip Delay (ms)	35	50	65
Packet Loss (%)	0,1	0,3	0,5
Jitter (ms)	10	n.a.	n.a.

Finally, the general network topology (both WAN or LAN) has to correspond to the minimum standards required for VoIP:

- No more hubs, only network switches

- QOS (Quality of Service) activated wherever possible



### 3. Recommendations and guidelines

In order to reach the conditions mentioned above, with optimal voice quality as purpose, we would like to provide a couple of guidelines with regards to the different components implied:

1. General remarks
2. Network layout & implementation
3. Remote Connectivity
4. Server Specifications
5. Server Configuration
6. Client side recommendations

### 4. General remarks

A typical CTArchitect setup can be built in 2 ways, depending on the desired size of the platform:

- If the number of call agents will never exceed 50 simultaneous working positions, CTArchitect's core engine + Telephony Node software can be installed on a single server.
- However, if the number of call agents exceeds this number, a separate Telephony Node will need to be installed per 100 positions. In this case, setup will be as follows:
  - o A software module called "CentralCtarchitect" will be installed on a separate server. This server will steer the separate Telephony Nodes
  - o Per 100 agents, a separate server will be installed, performing as Telephony Node

### 5. Network layout & implementation

#### A. General

As far as network layout & implementation is concerned, different factors play a very important role. First of all, ensuring the Highest Quality of Service is a crucial one. Networks, especially business networks, are dynamic when it comes to used bandwidth & workload/pressure put on them: one moment, the network could be used moderately (eg browsing a couple of websites), the next moment 10 people could start downloading huge files and 200 other people could start a telephone conversation over this very same network. If your network is not very good today (e.g., downloading and/or sending files is slow), then it certainly won't be good enough for VoIP. Things to be checked are:

- Is my cabling CAT5e or more recent everywhere?
- Are all switches capable of handling VoIP traffic?

- Is packet prioritisation done in the right way? (ToS 0xA0; DSCP 40 or DSCP 46)
- ...

Also, optimizing bandwidth for performance is the network equivalent of a “stress test” for your heart. Testing your network can determine how well it will function when all your users are on the phone and simultaneously accessing critical applications internally and externally. It is advised to subject your current network regularly to such performance tests. Depending on the results of these tests, additional (needed) interventions can be identified and put in place for further optimisations. Examples could be:

- Applying an additional voice codec like G.729
- Activation of header compression for point-to-point connections between remote sites
- ...

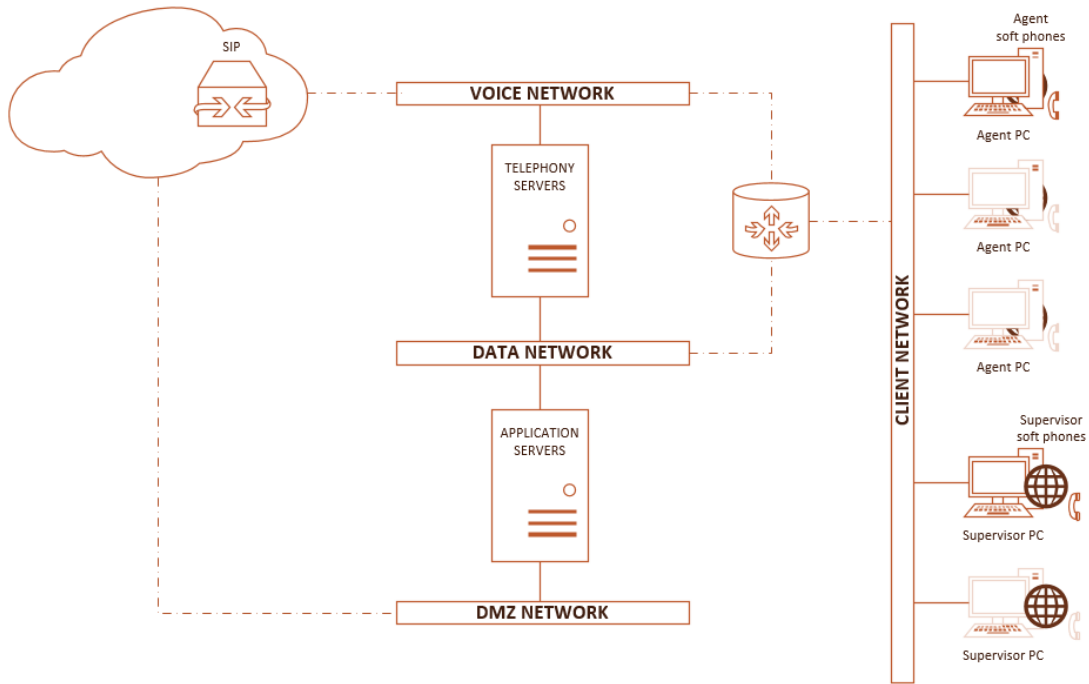
If you consider working with home workers or remote sites, please note that this setup can work very well, but some important remarks need to be taken into account: the stability and quality of the internet connection between the telephony server and the agent PC, needed for VoIP traffic, cannot be assured at all times, since it is dependent on external parties. It is also recommended that every home worker uses a VPN tunnel to connect to the company network (to avoid issues with firewalls & other connections). As backup, we can implement a failover procedure in which we contact the agent on a landline over the regular phone network.

MyForce does not support NAT for agent PC's because we cannot guarantee it will work with SIP protocol and we cannot troubleshoot. Instead, we recommend to have a VPN or direct connection to the servers so each agent IP address is unique.

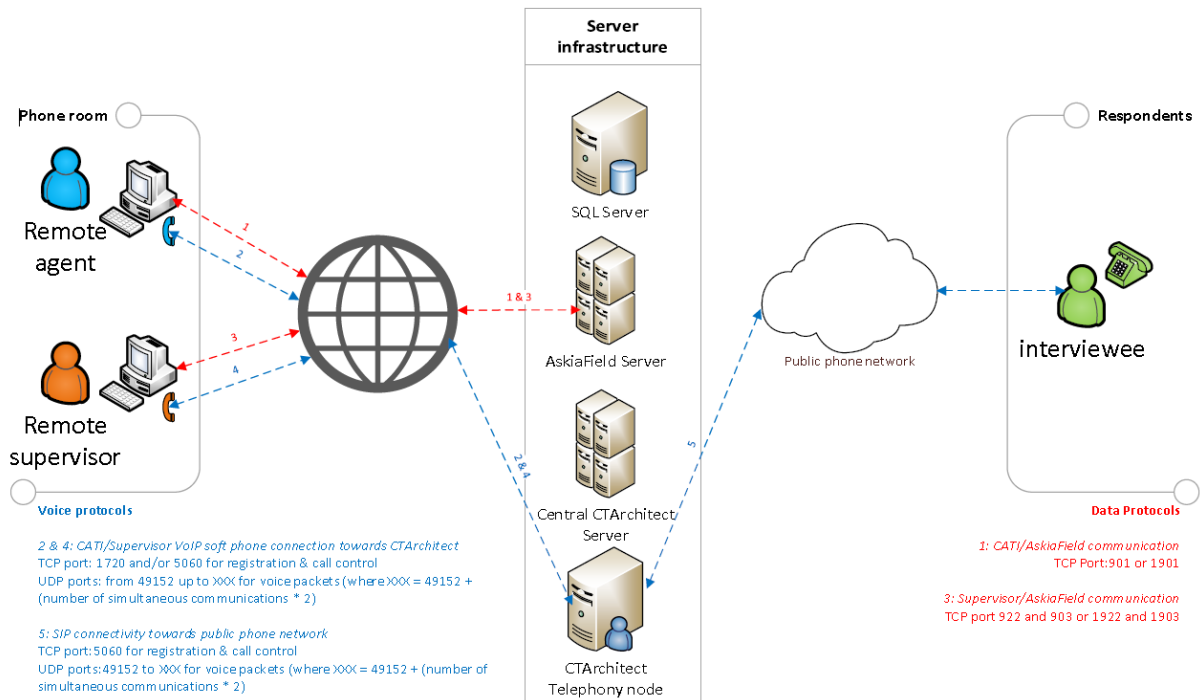
Best practice is to build a network infrastructure that is very well monitored and is capable of handling all traffic even in peak moments. This network should have proper configuration of prioritisation with quality of service.

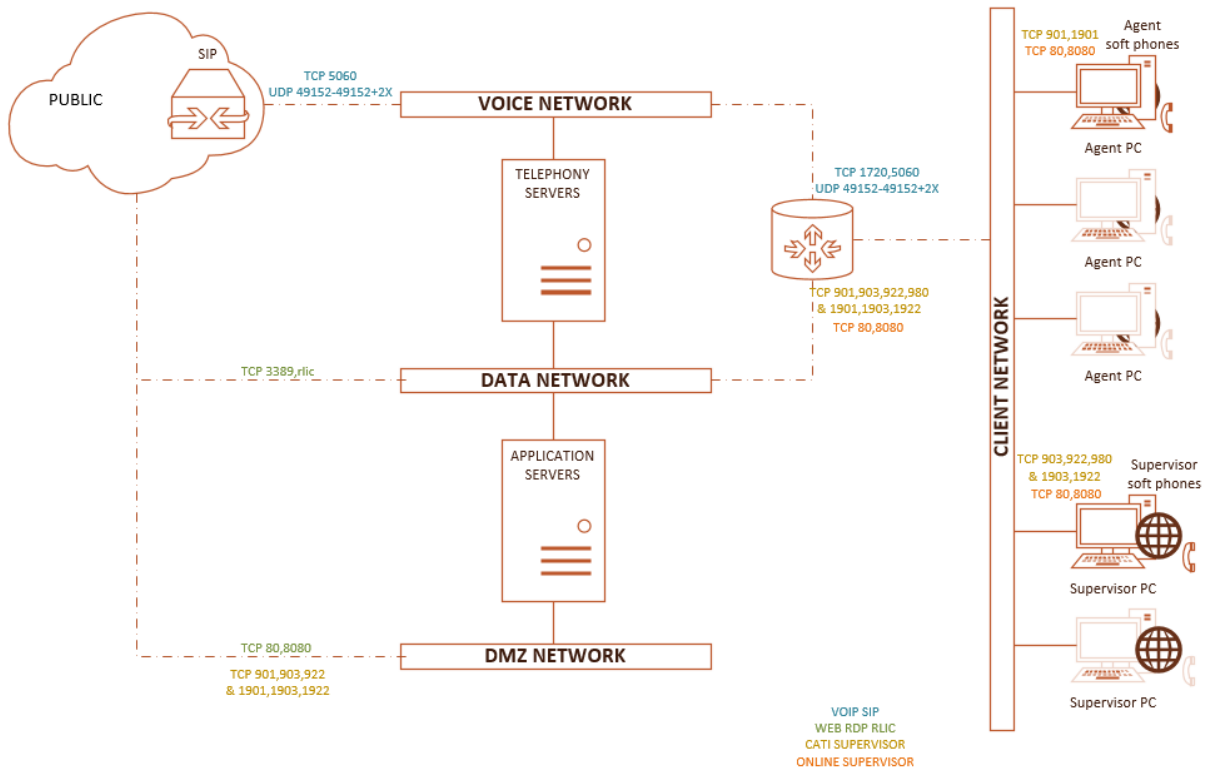
If you cannot guarantee the above, you could also choose to build at least two or three independent networks. One network for VOICE, one for DATA and one for DMZ, and maybe a fourth network for remote management. Keep in mind that windows network settings like gateways and default routes should be configured correctly.

Use the below infrastructure as a guideline for your network setup.



Finally it is important to understand the flows between stations and servers to configure routing and firewalls. The below images show the different steps in the communication flow and which ports are being used. Please keep this in mind when securing your networks.





## B. Port overview

A port overview is available in the below table. Use this table as a reference to configure your firewall settings.

From application	To application	Type of traffic	Protocol	Port range
CCA	(Central)CTArchitect	data	TCP	1812
CCA	(Central)CTArchitect	signalling	TCP	902
CTDesign	(Central)CTArchitect		TCP	9799
CTArchitect "GUI"	(Central)CTArchitect		TCP	9072
GlobalCCA	(Central)CTArchitect		TCP	904
SQL	(Central)CTArchitect		TCP	3389
WebService	(Central)CTArchitect		TCP	1802
Cati	Agent Monitoring Service	see-in	TCP	15500
Supervisor	Agent Monitoring Service	see-in	TCP	15900
CCA	CAPI		TCP	940
CCA	CAPI		TCP	1940
CTA/CTNode	Cati	SIP call signalling	SIP UDP	5060
CTA/CTNode	Cati	call stream	RTP	49152 to 49152+(2*#Agents)
Cati	CCA		TCP	901
Cati	CCA		TCP	1901

Cati	CCA	data	TCP	923
Cati	CCA	data	TCP	1923
Cati	CCA	beacon	UDP	1918
Cati	CCA	packages	TCP	1991
Supervisor	CCA		TCP	903
Supervisor	CCA		TCP	1903
Supervisor	CCA	data	TCP	922
Supervisor	CCA	data	TCP	1922
Supervisor	CCA	beacon	UDP	1918
Supervisor	CCA	packages	TCP	1991
Agent Monitoring Service	CCA	see-in	TCP	990
CCA-API	CCA		TCP	980
CCA-API	CCA		TCP	1980
(Central)CTArchitect	CCA		TCP	902
Export Service	CCA		TCP	960
Export Service	CCA		TCP	1960
CCA-WEB-API	CCA-API		TCP	808
Cati	CTA/CTNode	SIP call signalling	SIP UDP	5060
Cati	CTA/CTNode	H323 signalling	UDP	1720
Cati	CTA/CTNode	call stream	RTP	5000 to 6000
Supervisor	CTA/CTNode	SIP call signalling	SIP UDP	5060
Supervisor	CTA/CTNode	H323 signalling	UDP	1720
Supervisor	CTA/CTNode	call stream	RTP	5000 to 6000
CTDesign "GUI"	CTDesign		TCP	9797
CTNode "GUI"	CTNode		TCP	9072
(Central)CTA	External Licensing server		TCP	9798
(Central)CTA	HWClient		TCP	9070
(Central)CTA	HWClient		TCP	9071
CCA	Reporting Service		TCP	970
CCA	Reporting Service		TCP	1970
CTA/CTNode	SIP provider	SIP call signalling	SIP UDP	5060
CTA/CTNode	SIP provider	call stream	RTP	49152 to 49152+(2*#Agents)
CTA/CTNode	Supervisor	SIP call signalling	SIP UDP	5060
CTA/CTNode	Supervisor	call stream	RTP	49152 to 49152+(2*#Agents)

## C. Testing the network port

Run the program scan.exe (<ftp://ftp.remote.myforce.be/pub/Tools/PortScan/>) to test if the necessary ports are open and communication is possible. The program will scan each port and send a small amount of data over it. The ports which need to be scanned can be configured in the services.conf file. The syntax to use the

program is simple. Call upon the program and give as parameter the IP address of the server which needs to be tested. A result list will be shown afterwards.

## **D. Monitoring**

It is strongly advised to have a network monitoring tool in place. Especially when you have leased lines or vpn tunnels or remote locations.

When voice quality issues arise, network capturing capabilities (eg WireShark) are essential to investigate the problems.

## 6. Recordings

Our system provides multiple ways of recordings:

- Server side recording
  - mono
    - requires one voice channel per call
    - more or less 0.5MB/minute speaking time
  - stereo
    - requires two voice channels per call
    - more or less 2.5MB/minute speaking time
- Whole survey recording
  - requires one voice channel during recording
- Agent based stereo recording
  - requires local pc resources and extra bandwidth between agentpc's and servers to upload recordings at the end of the call
  - more or less 2MB/minute speaking time
- Open question recording
  - requires a voice channel during recording

## 7. Remote agent connectivity

When agents are working on a remote site it is essential to have a secure site-to-site vpn connection which assures bandwidth and QOS. The used technology can be IPsec, PPTP, L2TP or SSL.

## 8. Remote management connectivity

The support team of MyForce needs to have access to the different servers, this can be done using the RDP or VNC protocol. These servers should be accessible directly from an external IP address or using a VPN connection. We support VPN technologies from different providers like Microsoft, Cisco, Juniper, Citrix, ...

We do not support Teamviewer connections towards production servers. Teamviewer can only be used for troubleshooting purposes. In these cases, Teamviewer can be downloaded from our website <http://support.myforce.be>.

## 9. Server Specifications

### A. Overview

The number of required servers is directly dependant on the amount of agents. See below the guidelines of server roles depending on the size of the platform.

Number of agents	Application servers	Telephony servers
10-25	<ul style="list-style-type: none"> <li>SQL + CCA</li> </ul>	<ul style="list-style-type: none"> <li>CTArchitect</li> </ul>
25_50	<ul style="list-style-type: none"> <li>SQL</li> <li>CCA</li> </ul>	<ul style="list-style-type: none"> <li>CTArchitect</li> </ul>
50-100	<ul style="list-style-type: none"> <li>SQL</li> <li>CCA</li> <li>Central CTArchitect</li> </ul>	<ul style="list-style-type: none"> <li>Telephony Node 1</li> </ul>
100-200	<ul style="list-style-type: none"> <li>SQL</li> <li>CCA</li> <li>Central CTArchitect</li> </ul>	<ul style="list-style-type: none"> <li>Telephony Node 1</li> <li>Telephony Node 2</li> </ul>
200-300	<ul style="list-style-type: none"> <li>SQL</li> <li>CCA</li> <li>Central CTArchitect</li> </ul>	<ul style="list-style-type: none"> <li>Telephony Node 1</li> <li>Telephony Node 2</li> <li>Telephony Node 3</li> </ul>
300-400	<ul style="list-style-type: none"> <li>SQL</li> <li>CCA</li> <li>Central CTArchitect</li> </ul>	<ul style="list-style-type: none"> <li>Telephony Node 1</li> <li>Telephony Node 2</li> <li>Telephony Node 3</li> <li>Telephony Node 4</li> </ul>

### B. Telephony Node (or CTArchitect) server

As described above, each CTArchitect platform will include one or more Telephony Nodes, depending on the number of agents. For systems smaller than 100 agent seats, the CTArchitect application will cover the functionality of the Telephony Node.

We strongly recommend using dedicated servers, solely to be used by the Telephony Node. If this recommendation is not followed, voice quality can be impacted by third-party applications running on the same server. Also pay attention with snapshots and its impact on performance.

Dedicated servers can be real physical servers, or Virtual Machines. When virtual machines are used for Telephony Nodes it is mandatory to use VMWare ESX 4.1 or higher because this is the only virtualisation



software that is supported by Dialogic software. For virtual telephony nodes it is also highly recommended to use dedicated CPU assignment. Dynamic assignment of CPU frequency introduces a high risk for performance impact when the system is under heavy use.

Also very important to know is that the NIC's used for voice traffic on the Telephony Node Servers require a fixed MAC address because of licensing.

The Telephony Node application requires local administrator permissions. It is running as a graphical user interface and thus requires a windows user to be logged on at all times. An autologon is essential to guarantee a minimum downtime and allow automatic server boots.

### Physical servers

When it comes to technical specifications, we recommend a server with similar or better specifications as the **DELL Poweredge R530**. A possible PowerEdge R530 configuration can be as follows:

- 2 x Intel Xeon E5-2620 2.00GHz
- 3.5" Chassis with up to 8 Hard Drives
- 2 x 4GB 1333 MHz RDIMMs
- 16X DVD Drive SATA
- Dual, Hot-plug, Redundant Power Supply (1+1)
- RAID 1
- iDRAC

General **Telephony Node** (or CTArchitect) recommendations are as follows:

	Minimum Requirements	Recommended Requirements
<b>OS</b>	64bit Windows Desktop 64bit Windows server	64bit Windows server
<b>Processor</b>	4 cpu cores	8 cpu cores
<b>RAM</b>	4GB	8GB
<b>HD1</b>	40GB	80GB
<b>HD2</b>	60GB	250GB
<b>USB</b>	2 + 2 usb 2.0*	2 + 2 + 2 usb 3.0*
<b>Network</b>	2 x 1Gbps	3 x 1Gbps (redundant network + management connection)

\* Not required for virtualized servers

NOTE: Minimum requirements are only valid for small configurations.

As stated above, your equipment should perform equal or better than the recommended hardware. Special attention should go to low latency of all server components: even small delays can cause huge impact on VOIP quality because all data is handled in “real time”.

Also it is important to prevent big kernel time jumps caused by network time synchronisation. Time adjustments under 1 second are handled fine but bigger corrections will lead to call and service interruptions.

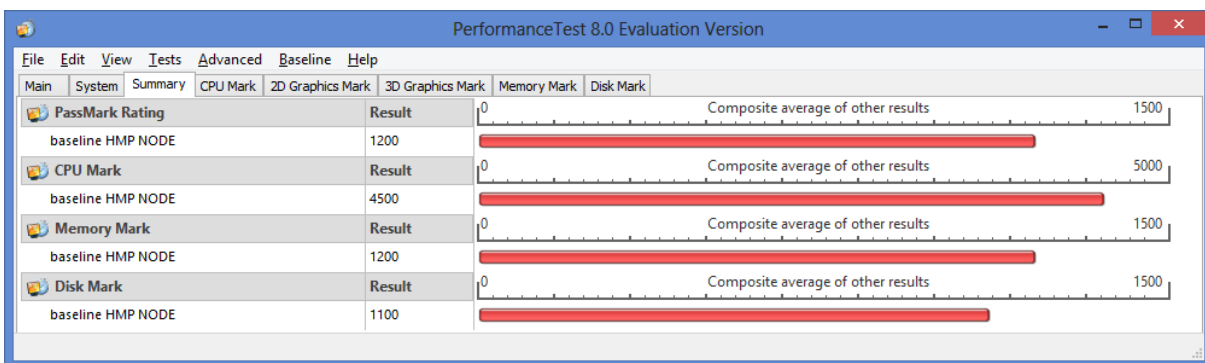
### Virtual Servers

A telephony node can be installed on a Virtual Machine as well, but the problem is that VM’s theoretical specs are a bit tougher to compare with the specs described for physical servers above. This is why we have created a benchmark, setting the baseline for the server’s essential components.

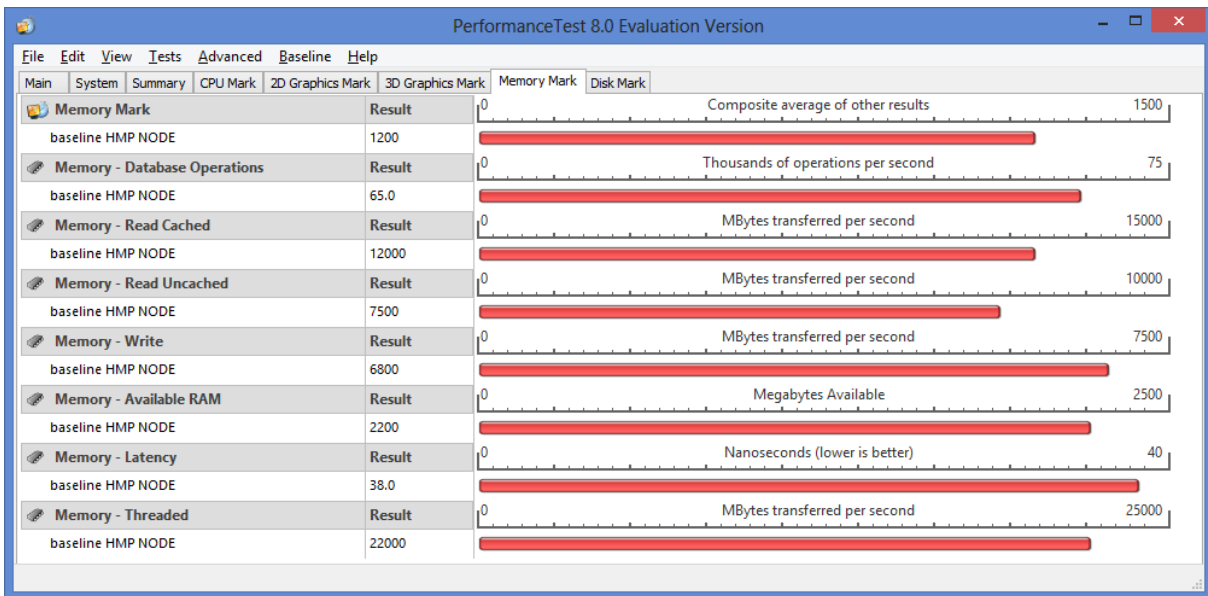
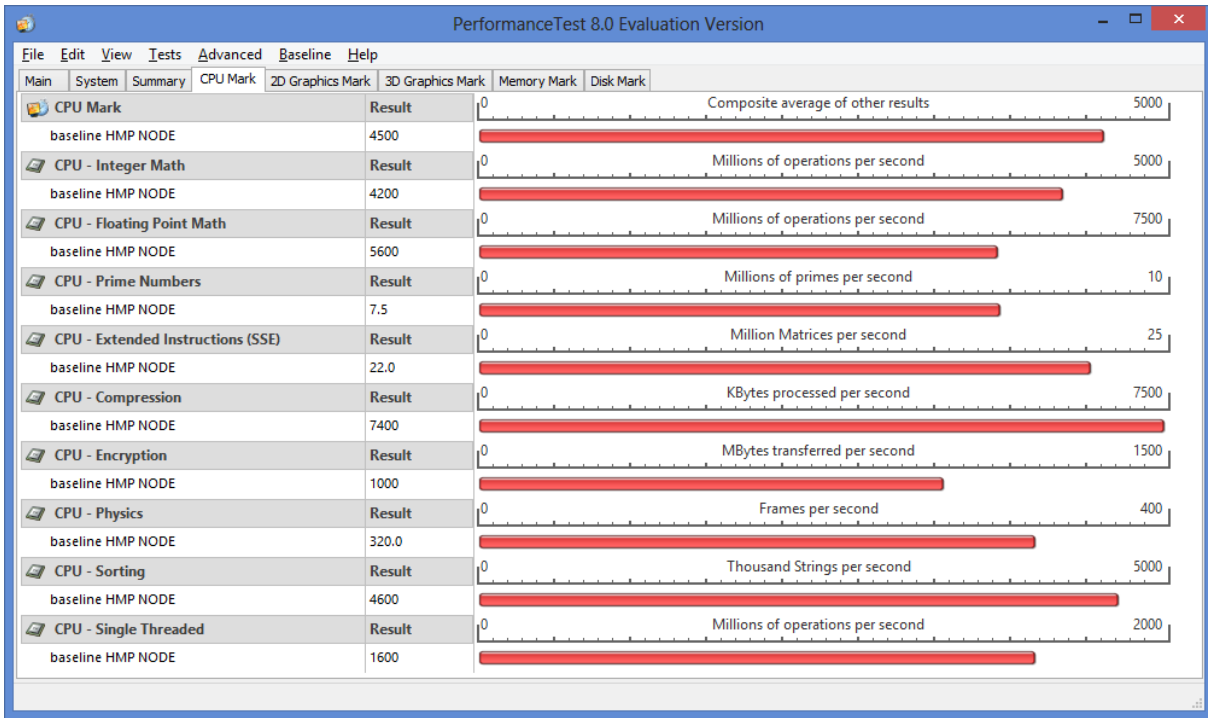
This benchmark can be found on our ftp site: <ftp://ftp.remote.myforce.be/pub/Tools/Benchmark/>. Please contact our support department to receive login information.

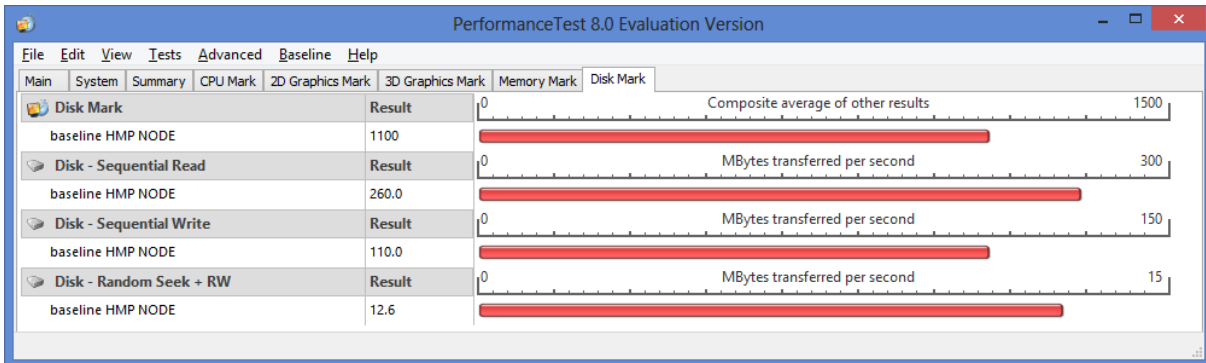
After logging in, download the trial version of “Performance Test 8.0”, install this on the server, run all CPU-, memory- and disk tests and compare to the information below.

The performance of your server should meet the following benchmark results:



Benchmark details:





### C. CentralCTArchitect server

The (Central)CTArchitect application requires local administrator permissions. It is running as a graphical user interface and thus requires a windows user to be logged on at all times. An autologon is essential to guarantee a minimum downtime and allow automatic server boots.

When the CTArchitect is not acting as a “Central CTArchitect”, the server will have to meet the requirements of a telephony node because it will be the server that manages the voice traffic.

When only 1 telephony node is needed, the CTArchitect application can be installed on the Telephony Node server – this does not affect the Telephony Node specs described above.

For larger environments (for systems >50 agent seats), the CentralCTArchitect application needs to be installed on a separate server.

In the second case, the following specs are recommended:

	Minimum Requirements	Recommended Requirements
<b>OS</b>	32bit Windows Desktop 32bit Windows server	64bit Windows server
<b>Processor</b>	4 cpu cores	6 cpu cores
<b>RAM</b>	4GB	6GB
<b>HD</b>	60GB	120GB
<b>USB</b>	2 + 2 usb 2.0**	2 + 2 + 2 usb 3.0**
<b>Network</b>	2 x 1Gbps	3 x 1Gbps (redundant network + management connection)

\*\*In case of virtualized servers, 1 USB port is needed at all times for the HASP license key (<http://www.safenet-inc.com/products/software-rights-management/sentinel-hasp-hardware-keys/>)

\*\* In case of virtualized servers, no USB port is available. Make use of Anywhere USB device. See the separate section about this.

## D. Application server

Next to the CTArchitect/Telephony node(s), a separate application server needs to be installed, running our CCA (Contact Center Administrator) software. This application server can be a physical or virtual server.

	Minimum Requirements	Recommended Requirements
<b>OS</b>	64bit Windows Desktop 64bit Windows server	64bit Windows server
<b>Processor</b>	4 cpu cores	8 cpu cores
<b>RAM</b>	8GB	12GB
<b>HD</b>	60GB	120GB
<b>USB</b>	2 usb 2.0*	2 + 2 usb 3.0*
<b>Network</b>	2 x 1Gbps	3 x 1Gbps (redundant network + management connection)

\* Not required for virtualized servers

The CCA can be installed either as an application or as a service.

## E. Database server

Finally, a database server is needed, running a supported Microsoft SQL Server and having SQL Profiler installed. Depending on the size of the platform and the server's technical specifications, it is also possible to install CCA and SQL on a single server. Please keep in mind that different editions and versions of Microsoft SQL Server provide different functionality and options. Also choose wisely your licensing.

	Minimum Requirements	Recommended Requirements
<b>OS</b>	64bit Windows Desktop 64bit Windows server	64bit Windows server
<b>Processor</b>	4 cpu cores	8 cpu cores
<b>RAM</b>	8GB	16GB

<b>HD1</b>	80GB	200GB
<b>HD2</b>	100GB	400GB
<b>USB</b>	2 usb 2.0*	2 + 2 usb 3.0*
<b>Network</b>	2 x 1Gbps	3 x 1Gbps (redundant network + management connection)

\* Not required for virtualized servers

## F. Operating system

Check the following website to verify if your OS is supported

<https://support.microsoft.com/en-us/lifecycle/search>

For new installations we advise **Windows Server 2016** or **Windows Server 2019**.

## G. Backup and redundancy servers

There are several ideas and strategies on reducing downtime in case of failure. One option is to install all servers on a virtual environment with a full mirror on a different physical location. This is however not always feasible.

An alternative is to install a SQL mirror and foresee a backup server which can cover functionality on demand. If full agent capacity needs to be guaranteed as well, an additional Telephony Node is required to reach n+1.

In case of virtualized backup/redundancy servers, 1 USB port is needed at all times for the HASP license key. (<http://www.safenet-inc.com/products/software-rights-management/sentinel-hasp-hardware-keys/>)

## H. Anywhere USB

If there is no USB port available on the CTArchitect server, there is a possibility to use an Anywhere USB device to connect the USB Hasp key to the server. The device and key can be delivered by Myforce.

If there is a firewall on the internal network or windows firewall, these steps need to be taken:

### To access an AnywhereUSB that is behind a firewall:

- Your firewall must have a well known static IP address, for example, 10.52.48.37.
- The AnywhereUSB must have an IP address on the private subnet, for example, 192.168.1.10.
- Your firewall must be configured to allow TCP/IP and UDP/IP packets to pass through port **3422**.
- The firewall must be configured to send these TCP/IP and UDP/IP packets directed to the IP Address of the AnywhereUSB; in this example the address would be 192.168.1.10.
- You must manually add the address of the firewall to the **Connection List**.

**Note that you can access only one AnywhereUSB through each firewall.**

## I. Monitoring

It is strongly advised to have a server monitoring tool in place. Parameters like cpu usage, memory usage, free disk space, uptime, service availability and service response times should be checked on a regular basis.

Do not hesitate to contact MyForce for further information.

## 10. Server configuration

### A. Telephony Node, (Central)CTArchitect & application server

#### Physical server

BIOS and OS settings need to be optimized for high-performance / low-latency. For example, processor power saving states should be disabled. Also network and disk sleep states should be disabled.

#### Virtual server

When installing the servers on a virtualised environment it is essential to use dedicated CPU cores and RAM. The process of dynamically assigning resources is not compatible with our software setup. An exception can be made when it comes to HDD storage space: this can be assigned dynamically.

E1000 and E1000E virtual NICs are a known source for connection and stability problems on virtual machines. Please select VMX3 as the virtual network adaptor.

#### Third party software

Dialogic Powermedia HMP software will be installed on Telephony Node servers and CTArchitect server. More info about HMP software and how to configure the servers can be found [here](#).

#### *Anti-virus Software Policy*

*Service Update 313 announces Dialogic's general policy regarding third-party anti-virus software. Dialogic understands and acknowledges the desire for customers / end users to install anti-virus software in their environment and is providing this policy statement for guidance in this area. Loading and running any third-party anti-virus software on Dialogic® HMP-based servers, regardless of the operating system (Linux or Windows), may compromise the performance of a real time telephony system. Consequently, it is recommended that if such anti-virus software is required by the customer for security purposes, it should be configured to run during periods of minimal to no-call traffic (i.e., system inactivity) to minimize the risk of interference or disruption. If a problem requiring diagnosis occurs on a server running anti-virus software, and the antivirus software is suspected of causing or contributing to the problem, Dialogic Technical Support will likely ask the customer to remove or disable the anti-virus software before proceeding with further diagnosis. Dialogic does not validate any particular antivirus package nor does it endorse the use of a particular third-party anti-virus software vendor or product. Note: For 64-bit Windows 7, Windows Vista, and Windows Server 2008 operating systems using anti-virus programs, it is recommended that users exclude Dialogic subdirectories and also any application sub-directories from the system.*

### B. Database Server

#### Virtual server

When installing the servers on a virtualised environment it is essential to use dedicated CPU cores and RAM. Dynamic allocation of disk resources is supported.



## 11. Client side recommendations

First of all, when working with the integrated soft phone for agents or supervisors, both PC & headset specifications are crucial factors.

### A. PC Specifications

PC should respect following minimum specifications (especially sufficient processing power & good network connection for voice handling):

Item	Minimum Requirements	Recommended
Processor	Windows experience index 1,5	Dual core cpu Windows experience index 3,0
RAM	1 GB Windows experience index 1,5	2 GB Windows experience index 3,0
HD	500 MB free space	2GB free space
Network	100 Mbps wired network connection	2x 100 Mbps wired network connection

Please note that even the most powerful PC can have a negative impact on voice quality if its resources are used by other applications at the same time. This is why a multi-core CPU is recommended as well.

### B. Operating system

For a list of supported operating systems, we refer to

<http://windows.microsoft.com/en-us/windows/lifecycle>

### C. Antivirus and Firewall

All 3th party software installed on the client PCs should be configured to whitelist the CATI software and its connections.

### D. Browser: Internet Explorer

If you want to use webscreens in Cati (or askiaface), we recommend to use a recent version of Internet Explorer (10 or higher).

## E. Softphone

CTArchitect comes with an integrated softphone. Using this softphone is highly recommended. Third party soft- or hard-phones should support the same protocols and features. Please contact us if you consider using another soft- or hard-phone.

## F. Headset

As far as headsets are concerned, we recommend professional USB headsets with (ultra-)noise cancellation and in-line volume and mute controls. The chosen headset needs to be compatible with the installed OS. Of course, we are always available to assist you in the choice of the correct headset (with advice or even with testing).

## G. Hard phone

If you would prefer working with a fixed/hard phone, following types have been approved for inbound traffic (and are in production on a couple of sites):

- Innovaphone IP110 (basic handset): <http://www.innovaphone.com/index.php?id=125>
- Innovaphone IP200 (advanced): <http://www.innovaphone.com/index.php?id=124>

Keep in mind that you need an IP gateway to register your IP phones

## H. Virtual/Thin clients

The Myforce agent software (softphone, cati, ctsript, askiaclient) has not been optimized to run on a virtual/thin client. It is important to keep the basic VOIP concepts in mind and thus required to manage the voice streams locally. This means the Softphone/Cati/Ctsript/Askia app needs to run locally. This is also the case for any other connected application such a task pop-up or an application integrated by means of the agent OCX.

However, in combination with a real hardphone, the Agent Software (without softphone) can run in virtual/thin client mode. Please get in touch with MyForce to discuss the technical details.

## I. Internet/Voip connectivity

Internet/Voip connection performance needs to meet the requirements for VOIP. Pay special attention to round trip latency, packet loss, jitter and Qos. The bandwidth depends on the number of simultaneous calls and their compression methods.

Your sip provider can advise.

We recommend to set up a trusted connection with the network of your sip provider to ensure a proper routing of sip and rtp traffic between the private ip address of our telephony node and the public ip address of the sip provider.

Best practice is to ask your sip provider to provide also a dedicated connectivity (private link). The sip provider sets up a private link between their core network and your server network to enable the sip provider to directly communicate with internal ip's of the telephony nodes.

An alternative is to set up dedicated 1 to 1 NAT between a public IP and the internal IP of the node. One public IP per node will be required. Please be aware of the security risks this solution will cause.

Another solution might be to set up NAT on an existing public IP to forward traffic to the node. This requires flexibility from the operator and will only work for single-node systems.

And last but not least you could also use registration instead of ip-trusting/trunk with the disadvantage that acceptance of calls depends on it. If registration fails, calls will be blocked (this is not the preferred way but is not too bad).

## **J. SMTP**

An smtp server needs to be accessible to send out monitoring email notifications towards the support team. Following details are required: server, port, user, password, ssl.

## **K. FTP**

Allow ftp traffic towards `ftp://ftp.remote.myforce.be/`.

## 12. Software version support

Our lifecycle management is based upon the following idea

[https://en.wikipedia.org/wiki/Long-term\\_support](https://en.wikipedia.org/wiki/Long-term_support)

This strategy has been used by other developers all over the world. For example,

<https://wiki.ubuntu.com/LTS>

## 13. Quality controls VoIP/SIP operator

### A. General

Compared to ISDN (TDM based) technology, SIP technology is relatively young and new. As a result, many (often smaller/newer) telephony provider companies have emerged in the market, with mixed VoIP voice quality and service as a result. This situation, combined with the fact that VoIP is more difficult to manage and set up in your infrastructure, makes it a lot harder to choose a high-quality SIP provider. Certain quality standards that we have grown accustomed to when dealing with ISDN-environments, such as good call quality, adequate reception of call disconnect causes, good call setup... are not always guaranteed in the case of certain SIP providers.

### B. Configuration

To configure the use of your preferred SIP provider we need some details like:

- Trunk IP address
- CLI(s) and CLI format
- DID format
- Which codecs are supported
- Optionally Inbound number and Inbound number format
- Confirmation about the usage of out-of band signalling messages. The usage of in-band signalling messages, although still possible from a pure technical point of view, always require custom development time (extra cost) and are putting a burden on the overall performance of the system.

### C. Validation

In order to assist CTArchitect users with the selection process and validation of a SIP provider, a procedure has been put in place to give you information about your preferred VoIP/SIP operator - even before installing it in a real production setup. The only thing needed is a test account, and after a battery of tests, a detailed report will be provided:

= approved;  = not tested yet or test on-going;  = not approved, requires further modification

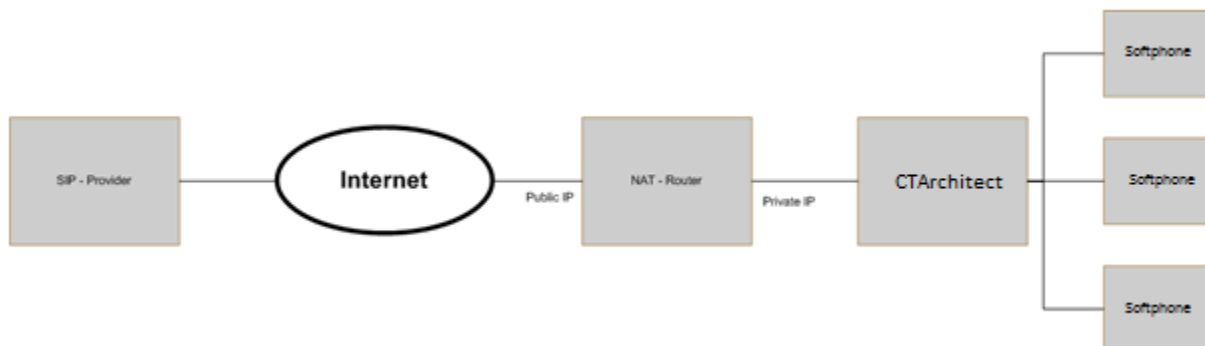
Test Name	Importance	Status
<b>Trunk</b>		
Trusting (trunk) of SIP provider or local SIP/ISDN gateway (IP approval)	KO	<input type="checkbox"/>
Registration (user) to SIP provider or local SIP/ISDN gateway (REALM, Server, Client, Password) (registration can help during validation but is not supported for production)	Optional	<input type="checkbox"/>
<b>Basic Call</b>		
Call using G711A	KO	<input type="checkbox"/>

Call using G729	Optional	<input type="checkbox"/>
CLI can be blocked	Optional	<input type="checkbox"/>
CLI can be altered	Optional	<input type="checkbox"/>
Long-time call possible (>30 min)	KO	<input type="checkbox"/>
Voice Quality	KO	<input type="checkbox"/>
International call set-up	Optional	<input type="checkbox"/>
<b>Outbound Call disconnection</b>		
Delay in receipt of disconnect causes	KO	<input type="checkbox"/>
Reception of busy disconnect cause	KO	<input type="checkbox"/>
Reception of disconnect cause wrong number (complete number provided to SIP-operator)	KO	<input type="checkbox"/>
Reception of disconnect cause wrong number (incomplete number provided to SIP-operator)	KO	<input type="checkbox"/>
Reception of disconnect cause wrong number (too large number provided to SIP-operator)	KO	<input type="checkbox"/>
<b>Inbound Call</b>		
Receive inbound call and redirect internally (Provider->CTArchitect)	Optional	<input type="checkbox"/>
<b>DTMF</b>		
DTMF tones sent correctly	KO	<input type="checkbox"/>
DTMF tones received correctly	Optional	<input type="checkbox"/>
<b>MultiNode Outbound</b>		
Set up a call from each node	Optional	<input type="checkbox"/>
<b>Load balancing</b>		
Receive inbound calls with "round robin" load balancing	Optional	<input type="checkbox"/>
Receive inbound calls with "least used" load balancing	Optional	<input type="checkbox"/>
Receive inbound calls with "overflow" load balancing	Optional	<input type="checkbox"/>

## D. Validation plan details

### Test Scenario

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This scenario describes a setup where CTArchitect and soft phones are in a private network.

### Trunk

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#### IP trusting or SIP Registration

Purpose: Validate digest authentication algorithm (specified by RFC 3261)

Test description:

- Make a test call based on IP Trusting
- OR
- Connect to the SIP provider with the provided credentials.

Result: not tested yet

### Basic Call

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#### Call using G711a (KO)

Purpose: Test of capability to handle G711A RTP - Streams

Test description:

- Soft phone is configured with G711A exclusive coder preference.
- Soft phone calls a phone in the PSTN

Result: not tested yet

### **Call using g729 (optional)**

Purpose: Test of capability to handle G729 RTP - Streams

Test description:

- Soft phone is configured with G729 exclusive coder preference.
- Soft phone calls a phone in the PSTN

Result: not tested yet

### **CLI can be blocked/alterd (optional)**

Purpose: Test if provider accepts anonymous calls

Test description:

- Soft phone is configured to not send his CGPN. 'User Setup'->'Number Presentation' = Off
- Soft phone calls a phone in the PSTN. The display of the PSTN phone should show as Calling Party: 'anonymous'

Result: not tested yet

### **Long time call possible (KO)**

Purpose: Test if provider interrupts call after session timer expires ([RFC 4028](#)). It is possible that the provider awaits Re-Invite after the timer expires.

Test description:

- Soft phone calls a phone in the PSTN
- Call is longer than session expire timeout.(usually 1800 seconds)

Result: not tested yet

### **Voice Quality OK? (KO)**

Purpose: Simple test of overall voice quality during a call.

Test description:

- Soft phone calls soft phone. Soft phone picks up the call.



- Manual test of voice quality by speaking/listening on both ends.

Result: not tested yet

### **International call set-up (optional)**

Purpose: Test of capability to handle the set-up of an international call

Test description:

- Soft phone dials an international number.
- Soft phone calls a phone in the PSTN

Result: not tested yet

### **Outbound call disconnection**

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#### **Delay in receipt of disconnection causes (KO)**

Purpose: validate that disconnect causes(800) are received within an acceptable delay

Test description:

- Soft phone is configured to call a wrong and/or busy number
- Disconnect are received by CTArchitect

Result: not tested yet

#### **Reception of busy disconnect cause (KO)**

Purpose: validate that disconnect cause(17) busy is provided in a correct way

Test description:

- Soft phone is configured to call a busy number
- Disconnect are received by CTArchitect

Result: not tested yet

### **Reception of disconnect cause wrong number (complete number provided to SIP-operator) (KO)**

Purpose: validate that disconnect cause wrong number is provided in a correct way if the complete number is provided to the SIP operator

Test description:

- Soft phone is configured to call a wrong number with a correct number format
- Disconnect are received by CTArchitect

Result: not tested yet

### **Reception of disconnect cause wrong number (incomplete number provided to SIP-operator) (KO)**

Purpose: validate that disconnect cause wrong number is provided in a correct way if the incomplete number is provided to the SIP operator

Test description:

- Soft phone is configured to call a wrong number with an incorrect number format (e.g. not enough digits)
- Disconnect are received by CTArchitect

Result: not tested yet

### **Reception of disconnect cause wrong number (too large number provided to SIP-operator) (KO)**

Purpose: validate that disconnect cause wrong number is provided in a correct way if the incomplete number is provided to the SIP operator

Test description:

- Soft phone is configured to call a wrong number with an incorrect number format (e.g. not enough digits)
- Disconnect are received by CTArchitect

Result: not tested yet

## DTMF

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### **DTMF tones sent correctly (KO)**

Purpose: Test if DTMF signals going from CTArchitect to Provider are received and interpreted correctly.

Test description:

- Soft phone calls phone in the PSTN.
- Check if the DTMF - tones are received at the PSTN phone

Result: not tested yet

**DTMF tones received correctly (KO)**

Purpose: Test if DTMF signals going from Provider to CTArchitect are received and interpreted correctly.

Test description:

- PSTN phone calls soft Phone.
- Check if the DTMF - tones are received at soft phone

Result: not tested yet