Description of Services Tele2 Office Phone IP (Status: 09/2011)

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TELE2 BUSINESS

1. Description of the Office Phone IP Service

This performance description defines the performances rendered within the scope of this service and the mutual rights and obligations of Tele2 and the customer in connection with the setting up and provision of the service for the customer. Within the framework of Office Phone IP, Tele2 provides its customers with a central virtual private branch exchange over IP that carries out internal switching functions and also provides access to the public telephone network. This private branch exchange completely replaces the previous system. It is no longer necessary to have a separate telephone line on site. All voice calls are signaled over an IP connection using an IP/SIP. Voice calls are routed internally over the LAN and externally (to other Office Phone IP locations or other telephone numbers outside the Tele2 IP-network) over the IP connection. A virtual private branch exchange can be used for several customer locations. As a result, all functions can be used throughout the company almost seamlessly (e.g. call diverts). It is also possible to use a central switchboard for all sites.

Office Phone IP requires at least one IP connection to the Tele2 IP-network (ViPNet, BizNet xDSL, TopInternet). In special cases, an Internet connection from a different provider (e.g. abroad) may be used as the IP access (see also Section 2 Realization Options). At each site Office Phone IP is always directly connected to the public telephone network over Business Telephony by IP (the business telephony service used hitherto must be re-ordered or changed).

1.1. Online Management and Documentation

Office Phone IP is administered over an existing web browser (e.g. Internet Explorer™). The following options are available via the online access at http://sip.myzone.at:

- Administration of Office Phone IP (as User or Administrator)
- Download of the User Guides

The required access data (user name, pass word) for the administrator will be sent by Tele2 to the e-mail address of the contact person named in the registration form. The access data for the users are administered by the customer administrator.

2. Realization Options

Tele2 offers customers throughout Austria a variety of realization options for Office Phone IP provided that the connection is technically and operationally feasible and makes economic sense for Tele2. Depending on the customer requirement / situation it may not always be possible to realize all types of Office Phone IP connections at a particular customer location. Tele2 reserves the right to refuse to provide an Office Phone IP-access without stating any reason.

2.1. BizNet

The realization option using BizNet provides the connection to the Tele2 IP-network over the Internet. This type of connection is used particularly in cases when the customer access line is located in the unbundling area of Tele2 (more detailed information as to whether the particular customer location is in a Tele2 unbundling area can be obtained by calling 0800-800 882 or from your customer account manager). BizNet with bandwidths of 512 Bit/s or more upstream provides the customer with a high-quality Internet connection that satisfies the requirements for the connection of Office Phone IP.
BizNet xDSL is idea for realizing Office Phone IP for all types of company; from the smallest company with only one location to large companies wishing to link several locations. Tele2 recommends TopNet for customers wishing to connect several locations to form a virtual private branch exchange (see Section 2.3). Further information is available in the Performance Description if required.
Office Phone IP restricts access to the configuration files for IP telephones. In the case of BizNet the assigned IP addresses are initially registered for VoIP. These addresses are private IP addresses from the 172.29.0.0/16 range and are dynamically allocated to a location as a second IP address. It is the responsibility of the customer to change access to the configuration files (see Office Phone IP Administrator Instruction Manual).

2.2. TopInternet

In cases where customers have a higher broadband requirement and a site cannot be unbundled (and a connection over xDSL is not possible), we recommend a TopInternet connection. TopInternet is the premium Internet service from Tele2 that offers maximum flexibility, performance, security and quality. Further information is available in the TopInternet Performance Description if required. TopInternet guarantees a voice bandwidth of 50% of the connection bandwidth with no guarantees for delay and jitter for connection bandwidths lower than 512 kB/s (no link fragmentation).
Office Phone IP restricts access to the configuration files for IP telephones. In the case of TopInternet, the assigned IP addresses are initially registered. It is the responsibility of the customer to change access to the configuration files (see Office Phone IP Administrator Instruction Manual).

2.3. TopNet

TopNet with QoS ensures the appropriate quality guarantees within the framework of the Office Phone IP requirements, provided suitably dimensions connections have been ordered. Any TopNet with QoS can be used as an access to Office Phone IP. The Office Phone IP platform is routed in the customer's network with the IP addresses 62.218.251.0/24. In the case of an IP address conflict, the customer must give up this IP address range, as these are official Tele2 addresses. In addition to the requirements for QoS (which are generally higher), the customer CPE in TopNet must also support NAPT (Network Address Port Translation) for the special destination addresses 62.218.251.0/24, as every location for Office Phone IP must be concealed behind a single address. If the requirements set out in the Performance Description TopNet with QoS are met, the requirements for Office Phone IP with TopNet are also met. The management IP addresses are assigned by Tele2 and must be clear-cut for each location.
Office Phone IP restricts access to the configuration files for IP telephones. In the case of TopInternet, the management IP addresses are initially registered. It is the responsibility of the customer to change access to the configuration files (see Office Phone IP Administrator Instruction Manual).

2.4. Internet Access

Office Phone IP can also be used over other Internet connections (e.g. abroad) for individual locations and teleworkers. In this case, the customer acknowledges that Tele2 can make no predictions or guarantees regarding the quality of the access line or the Internet service itself and therefore no demand may be made for faultless service. Tele2 recommends using one of the services listed under Sections 2.1. to 2.3. The quality of Office Phone IP over a different Internet service corresponds to a "Best Effort". The realization of this Internet access is not part of the Office Phone IP service. Therefore, if a fault occurs on the Internet access, it must be reported to the relevant Internet provider. Office Phone IP restricts access to the configuration files for IP telephones. As Internet IP-address are initially blocked, the initial configuration of the IP telephones can only be carried out after the Office Phone IP administrator has activated the IP addresses. It is the responsibility of the customer to change access to the configuration files (see Office Phone IP Administrator Instruction Manual).

3. Connection to the Public Telephone Network

A direct central connection to the public telephone network at each customer location is provided by the service "Tele2 Business Telephony". The number of simultaneous voice channels (outside lines) to the public telephone network will be given a customer-specific configuration.
The physical connection is provided over existing or new IP connections. If the connection is provided over the following services, Tele2 can guarantee the required voice quality by means of prioritization (QoS):

- BizNet (upstream bandwidth of 512 kBit/s or more)
- TopInternet
- TopNet

If the connection types listed below are used, the customer acknowledges that Tele2 cannot take preventative measures to meet the required voice quality criteria for telephony (delay, jitter, bandwidth, throughput) and cannot offer the customer support to meet the criteria. Furthermore, Internet service faults must be reported to the relevant Internet provider.

- Tele2 Internet connection, with the exception of BizNet xDSL and TopInternet
- Internet connections from other providers

3.1. Geographic Assignment of Extensions to Subscriber Numbers

Extensions are assigned to the individual geographical numbers during installation (see Section 6 Installation) over the MAC addresses of the respective end devices. The customer is responsible for assigning the extensions to the individual geographical numbers with both the Self Installation and Installation options. In the event of relocation, the customer must make the necessary adjustments to the assignments.
In general, geographical subscriber numbers can only be assigned if there is a place of business in this area code. This is to ensure that emergency calls can be returned (in accordance with the Communications Parameters, Fees and Value-Added Services Ordinance - KEM-V). This is especially important for the delivery of emergency calls. No claims whatsoever may be asserted against Tele2 arising from the customer's failure to assign or reassign extensions to their geographical subscriber numbers.

3.2. Use of 0720 Numbers

Subscriber numbers with the prefix 0720 are not strictly tied to a location. However, their use outside Austria is only permissible, if they are mostly used within Austria.

3.3. Differences to the Performance Description Business Direct over an ISDN Basic or Multiple Line

The Office Phone IP service expands the scope of the previous telephony service Business Direct. Office Phone IP provides the customer with a fully managed IP private branch exchange from Tele2, the functions of which are similar to those of a traditional private branch exchange.

4. End Devices

4.1. Overview of End Device Parameters

Tele2 provides a variety of end devices and a switchboard software. The end devices and its functions in detail:

Technical Data	snom 300	snom 320	snom 360	snom 820	snom m3	Yealink VP-2009
Dimensions in mm (W x D x H)	200x185x120	250x200x120	250x200x130	250x200x135	125x50x20	286x89x45
Maxim Air Humidity	10% - 85%, non-condensing	10% - 85%, non-condensing	10% - 85%, non-condensing	10% - 85%, non-condensing		10% - 95%, non-condensing
Ambient Temperature	0℃ - 40℃	0℃ - 40℃	0℃ - 40℃	0℃ - 40℃		0℃ - 60℃
Power Supply	5V, external plug-in power supply	5V, external plug-in power supply	5V, plug-in power supply	5V, external plug-in power supply	5V, external plug-in power supply	Power over Ethernet (optional) or power supply
LAN Interface	RJ45, 10Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T
PC Connection over integrated switch	RJ45, 10Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T	RJ45, 10/100Base-T	-	RJ45, 10/100Base-T
Telephone Receiver	RJ14	RJ14	RJ14	RJ11	-	RJ11
Headset	-	RJ14	RJ14	RJ11	-	2,5mm Headset port
Display	2x16 characters	2x16 characters	128x64 Pixel	240x128 Pixel		800 x 480 Pixel
Function Keys	6	12	12	12		6 fixed function keys

						and 4-way navigation
Expansion module (42 keys)	-	-	yes, up to 3	yes, up to 4	-	-
Codec	G.711, G.723.1, G.729a	G.711, G.723.1, G.729a	G.711, G.723.1, G.729a	G.711, G.723.1, G.729, G.726, G.722	G.711, G.729, G.726, G.722	G.711, G.723.1, G.729AB, G.722
Connection Analogue Telephone	-	-	-	-	-	-
Fax Support	-	-	-	-	-	-
PoE	no	yes	yes	yes	yes	yes (optional)

The scope of delivery of the phones and the analogue adapter includes a CAT5 cable (2m, UTP, 1:1) to be connected to the LAN. If required other cable lengths must be provided by the customer. If required, Tele2 will provide the customer with switchboards (software solution for Windows PCs). This software may only be used in combination with Office Phone IP and may not be duplicated. Once Tele2 has granted a license, the switchboard software may be downloaded from http://sip.myzone.at with the login data of the assigned user. Within the framework of the Office Phone IP service, Tele2 provide its customers with the necessary end devices which are to be used correctly by them. At the end of the contractual relationship, all the end devices provided must be returned to Tele2. Tele2 reserves the right to charge the customer the current market price for devices which are no longer in an optical and technical condition consistent with normal wear and tear in an office environment. This does not apply to end devices that were purchased.

4.2. Use of Fax Machines

If required, a suitable adapter (analogue adapter, see Section 4.1) can be connected to a conventional fax machine (Group 3) provided by the customer. The Tele2 analogue adapter supports Group 3 fax machines with up to 14.4 kBit/s. Tone dialing is the only dialing method that is supported. Pulse dialing is not supported. For further details see Section 4 End Devices

4.3. Other End devices and Soft Clients

In addition to the Tele2 end devices listed in Section 4, the customer may connect and operate its own end devices and soft clients provided they comply with the technical requirements of Office Phone IP (see Section 13 Technical Parameters and Appendix 5: End Devices Compatible with Office Phone IP). The customer has the rights of disposal over and responsibility for the end devices and soft clients. Tele2 assumes no warranty for the proper functioning of end devices and soft clients not provided by Tele2. However, the customer may not operate any end devices or soft clients that have a disruptive effect on Office Phone IP. Tele2 reserves the right to close IP accesses behind which such clients are operated. In case of doubt, the customer must obtain the agreement of Tele2 in this regard. Office Phone IP does not support the operation of modems.

5. Configuration

Configuration by Tele2 is carried out on the basis of a standard configuration. In this configuration, the number of extensions offered in the solution proposal is set up for the customer together with the options.

- Standard Configuration: 2-digit numbering plan
- Standard prefix for voice mail and fax: 9 before the extension number
- The numbers 111 to 144 may not be used, as they are reserved for emergency numbers which can called without obtaining an outside line.
- Extension numbers never begin with “0”.
- The switchboard will be configured by Tele2 during installation.
- All extensions are initially in the group “All.”

Special requests for routing rules such as speed dialing for mobile numbers (in one number group) or other number groups will be carried out by Tele2 free of charge. The extent to which these are possible

will be verified on a case by case basis and a solution proposed and provided. Special requests for routing rules must be made in writing and sent by e-mail to voip-support@at.tele2.com.
An operator (account/switchboard extension) cannot register and requires real extensions that carry out the switchboard function. These real extensions are generally reached with parallel ringing of the switchboard extension.
In general main numbers without an extension number cannot be reached. If the customer wishes to reach an extension on the main number without an extension number, an alias must be set up with the main number in the format <CountryCode><Prefix><Main Number> in the desired extension (e.g. switchboard or group call distribution) (for example 43190123).
Example: The switchboard has the extension number 0. The switchboard can then be distributed between the real switchboard extensions 10, 11 and 12 (if there are several of them) over the parallel ringing function. An alias of 43190123 in the extension 0 (switchboard) would mean that a call to the main number without an extension number would appear like a call to the extension 0. Generally, any extension with any available main number may be used as an alias for this purpose. Incoming calls can therefore be distributed to different phones without an extension number

6. Installation

6.1. Hardware Installation by Tele2

Tele2 activates the customer's private branch exchange in accordance with the information provided in the registration form and with a general configuration (see Section 5). The Tele2 customer service or a Tele2 partner subsequently installs the phones supplied by Tele2 and assigns user names to the phones. The assignment of direct outward dialing, speed dialing numbers, group assignments and all other customer-specific settings is carried out by the customer. Installation by Tele2 includes brief training in the use of Office Phone IP at the customer's premises with a description of the most important functions and the use of web administration lasting 10 to max. 15 minutes per telephone. This training is held in small groups. During the training session, the following points are explained:

- Description of the keys
- Making simple calls (dialing, connection)
- Announcement of the links for the personal settings and the documentation

Any additional necessary work or training will be charged to the customer separately by Tele2. The prerequisite for correct installation by Tele2 is that the customer makes the installation site available in a condition suitable for this work, e.g.:

- Tele2 Internet connection (or ViPNet) must be available and functioning
- Power supply must be available for the phones
- LAN-socket/plug must be available and functioning
- DHCP-server must be available (if it is not available, the information regarding the IP addresses, netmask, default gateway must be provided)
- Firewall, DHCP-server, router etc. must be properly configured (see also Section 13 Technical Parameters)
- Access data (user name and pass word) for the management access must be provided by the customer (will be e-mailed to the customer by Tele2 in advance)

Furthermore, the customer must have fully configured Office Phone IP (extensions, functions, groups etc.) by the time the installation is carried out. Otherwise the installation may be aborted and charged separately by Tele2 at a later date.

6.2. DECT Radio Field Measurement

In the case of DECT systems, Tele2 recommends that its customers carry out a radio field measurement for which there is a charge. The customer may choose not to do so, but in this case Tele2 cannot promise that the end devices will have comprehensive reception.

7. Prerequisites for Operation

Office Phone IP can only be used if the following conditions are met:

• A switched network environment in the LAN, CoS (IEEE 802.1p) is recommended to ensure voice quality.
• IP-connection with a bandwidth of at least 512 kbit/s. In the case of asymmetric connection bandwidths, this figure refers to the lower of the bandwidths (generally, the upstream bandwidth).
• QoS (Low Latency Queuing, LLQ) to maintain voice quality with simultaneous use as a data connection, with the exception of BizNet xDSL in the Unbundling (xDSL) version.
• In line with the bandwidth of a telephone call (between 25 kb/s (G.723.1) and 80kb/s (G711)), no more than half the connection bandwidth should be used for voice traffic if voice traffic and data traffic are carried simultaneously.
• Care must be taken that VoIP data packets with an IP precedence value of 5 or TOS (Type of Service) value of 160 are marked. If other data packets are marked with this value, voice quality cannot be guaranteed. The customer is responsible for ensuring that data applications do not use this service quality (IPPrec = 5, TOS = 160)
• The power supply (230 VAC) required for the connecting unit and the end devices are to be provided by the customer. Normally (except in the case of Power over Ethernet capability) a power supply must be provided for each connecting unit and end device.
• The customer must ensure that an operating temperature range of +5°C to +40°C and a relative air humidity of 35 to 75% (non-condensing) are maintained.

8. Security/Privacy of the Office Phone IP Service

Tele2 makes every effort to ensure the security and privacy of the service with all reasonable technical and economic means.
The following methods are used to prevent unauthorized access:
• Firewall with intrusion detection and 7x24h monitoring including on-site standby
• Carrier grade hardware including hardware and software maintenance
• Only encrypted passwords are sent (MD5 digest authentication)
• MPLS Exp Bit Remarking @ Provider Backbone Edge (DoS Prevention)
• MPLS Hub & Spoke IP VPNs (direct data communication between IP-VPN-customers) are not possible

To prevent abuse the customer shall be obliged to keep its access data (user name, password) secret and to store it safely. Tele2 accepts no liability for any damages arising from improper use.

9. Service Management

9.1. Service Availability, Response and Fault Repair Times, Support Level

With regard to service availability, response and fault repair times, and Support Level, the same times basically apply as set out in the Service Level Agreement upon which the Access is based (e.g. TopNet or TopInternet). This does not however apply to the replacement of end devices (see Point 17.1) and individual configuration changes.
If there is no Service Level Agreement available for the underlying Access or one was not concluded, Best Effort will be used for fault repairs.

9.2. General

To maintain service availability, the service will be maintained by Tele2 or a third party appointed by it. This service work includes the rectification of all faults and failures in the sphere of responsibility of Tele2 or third parties appointed by it. The rectification of faults and failures which are attributable to Tele2 or its vicarious agents is free of charge for the customer.
Should Tele2 be called upon to rectify a fault and it is established that either the Office Phone IP service has no fault or that the fault is attributable to the customer, the customer must reimburse Tele2 for the expenses incurred according to the applicable specialist hourly rate. The customer accepts that 100% availability cannot always be guaranteed. Tele2 reserves the right to temporarily suspend or restrict the services for maintenance purposes or for reasons of safety or capacity.

Software-based products (e.g. Switchboard Software, CTI Client, Soft Client, etc.) provided by Tele2 must without exception be implemented solely by the customer. Installation or troubleshooting regarding such

clients will not be performed by our field service engineers. Customers accept that the individual PC configuration (e.g. Firewalls, Virus Scanners, etc.) might interfere with the functionality of the client. In such cases Tele2 cannot provide any on site support. However Tele2 always puts great effort into helping find the cause if traces are provided by the customer.

10. Network Termination Point

The network connection point defines the boundary of responsibility between Tele2 and the customer. All network equipment in front of the connecting unit (on the network side) and also the connecting unit itself fall within Tele2's area of responsibility.
If the end devices are also provided by Tele2, they also fall within Tele2's area of responsibility.
Notwithstanding this, the customer shall be liable to Tele2 for ensuring that the connection units represent the state-of-the-art and meet the requirements described in this document, that they are suitable for the contractual purpose and are free from defects and that he has full powers of disposal over them.

11. Responsibility in the LAN

IP-telephony requires an IP-switch that is IP-accessible. This includes, among other things, the correct function of name resolution (accessible to public or private DNS server), correct allocation and handling of IP addresses (DHCP server, static IP addresses, etc.), which are often the responsibility of the customer. The customer LAN itself is not the responsibility of Tele2 – this also applies to firewalls, DNS and DHCP servers operated by the customer. We therefore ask that the following checks are carried out before a fault is reported to Tele2 (voip-support@at.tele2.com):

- Does the IP-telephone or analogue adapter have a correct IP configuration?
- Is the network area 62.218.251.0/24 accessible?
- Is the power supply working?
- Is the firewall functioning properly (see Section 13.4), are the reply packets of our server being let through into the LAN?

12. Service Handover

The handover of the Office Phone IP service for each access takes place when the handover certificate for the access is handed over. Billing for the services provided by Tele2 for this access commences when the service is taken into service, however, at the latest upon the date of the handover certificate. Once the service has been handed over, all changes to the configuration must be carried out by the customer. If changes to the configuration are carried out by Tele2, charges will be incurred by the customer which will be billed according to the applicable specialist hourly rate.

13. Technical Parameters

Office Phone IP is based on the general standard SIP (Session Initiation Protocol). The standardization of SIP is carried out by the IETF (Internet Engineering Task Force), which results in an open interface and freely accessible documentation (http://www.ietf.org). The RFCs (Request for Comments, (Pre-) Standard documentation) used here are named below.

13.1. Numbering Restrictions

The numbers 111 to 144 are reserved for emergency calls and may not be used as extensions. This makes it possible to make emergency calls without dialing 0 to obtain an external line. The emergency call is then delivered in the area code to which the end devices have been assigned by the customer's administrator. Tele2 supports a subscriber number (0043 + main number + extension) with a maximum of 16 digits (example 0043 5574 908141 - 10 are 16 digits). If these are exceeded, international accessibility can no longer be guaranteed. Extension numbers may differ in length and may also start with the same number group (e.g.: 20 and 201).

13.2. Implementation of PSTN to VoIP Telephone Numbers

The conversion of conventional telephone numbers to Office Phone IP extensions which becomes necessary if an extension is dialed from a public telephone network results in an additional waiting time of approximately 3 seconds while the connection is being set up. During this time the system waits for

additional extension numbers. Digits dialed after this time are ignored. Digits dialed after a call has been set up, are sent as DTFM (Dual Tone Multi Frequency) Information analogously to the PSTN.

13.3. RFCs to Office Phone IP

The definitions in the RFCs below are minimum requirements for SIP end devices (SIP UA, SIP User Agents), that may be operated on Office Phone IP. If these are not met, the use of the end devices on Office Phone IP is prohibited.

- RFC3261: SIP: Session Initiation Protocol. J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler. June 2002. (Updated by RFC3265)
- RFC3262: Reliability of Provisional Responses in Session Initiation Protocol (SIP). J. Rosenberg, H. Schulzrinne. June 2002. (Obsoletes RFC2543)
- RFC3263: Session Initiation Protocol (SIP): Locating SIP Servers. J. Rosenberg, H. Schulzrinne. June 2002. (Obsoletes RFC2543)
- RFC3264: An Offer/Answer Model with Session Description Protocol (SDP). J. Rosenberg, H. Schulzrinne. June 2002. (Obsoletes RFC2543)
- RFC3265: Session Initiation Protocol (SIP)-Specific Event Notification. A. B. Roach. June 2002. (Obsoletes RFC2543) (Updates RFC3261)
- RFC3267: Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs. J. Sjoberg, M. Westerlund, A. Lakaniemi, Q. Xie. June 2002.
- RFC3581: An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing. J. Rosenberg, H. Schulzrinne, August 2003.
- RFC3891: The Session Initiation Protocol (SIP) "Replaces" Header. R. Mahy, B. Biggs, R. Dean. September 2004.
- RFC3892: The Session Initiation Protocol (SIP) Referred-By Mechanism. R. Sparks. September 2004.

Moreover, RFC drafts that define switchboard functions are also used. As these have not been definitively established, changes in accordance with the standards will be made over time at a later date. The drafts are being dealt with in the SIP, SIPPING and Network work groups, and can be found at:

- http://www.ietf.org/ID.html
- http://www.ietf.org/ids.by.wg/sip.html
- http://www.ietf.org/ids.by.wg/sipping.html

In particular, reference is made to the following drafts:

- Session Initiation Protocol Call Control - Transfer. R. Sparks, A. Johnston Internet-Draft, Expires: August 15, 2004 draft-ietf-sipping-cc-transfer-02.txt
- A Framework for Conferencing with the Session Initiation Protocol. J. Rosenberg Internet-Draft, Expires: December 28, 2004 draft-ietf-sipping-conferencing-framework-02.txt
- Session Initiation Protocol Call Control - Conferencing for User Agents. A. Johnston, O. Levin Internet-Draft, Expires: August 15, 2004 draft-ietf-sipping-cc-conferencing-03´.txt
- Session Initiation Protocol Service Examples. A. Johnston, R. Sparks, C. Cunningham, S. Donovan, K. Summers Expires: January 14, 2005 draft-ietf-sipping-service-examples-07.txt

13.4. NAT and Firewalls

In principle private IP addresses are supported in the customer LAN and in firewall-protected Internet accesses. The NAT (Network Address Translation) devices and firewalls must support the following connections from the private network to the Internet or external network.

Outgoing connections for the protocols:

DNS (UDP:53), NTP (UDP:123), HTTP (without proxy) (TCP:80), TFTP (UDP:69), RTP (> UDP:10000), SNMP (UDP:161 and UDP:162), SIP (UDP:5082)

All source ports of these protocols must be translated to > 1024 at NAT.

The connections must accept reply packets to outgoing requests after a time-out of up to 40 seconds (UDP time-out 40s).

Where possible SIP NAT protocol support should be deactivated and only used after prior verification by Tele2.
There is generally no need to allow incoming connections in the firewall rules as long as UDP is handled by a statefull firewall.
If additional restrictions in the outgoing connection rules are desired, the address range can be restricted to the IP addresses 62.218.251.0/24. This address range applies until further notice and may be changed by Tele2 without notice.

14. Voice Codec Selection for Voice Quality

Office Phone IP provides 2 different options for the voice codec:

- G. 711: Standard codec without compression, voice quality comparable with ISDN, required bandwidth including overhead per call approx. 80kbit/s, MOS-value of 4.3 (optional)
- G.729a: Alternative codec with compression, quality comparable with GSM, required bandwidth including overhead per call approx. 40kbit/s, MOS-value of 4.1 (standard codec)

Codec	Transmission Speed	MOS	Audio Frequency	Voice Quality Equivalent	Compression
G.711a	approx. 80kbit/s	4,3	300 - 3.400Hz	ISDN	No
G.729a	approx. 40kbit/s	4,1	300 - 3.400Hz	GSM-connection	Yes

15. Parameter Definitions and DHCP Settings

Generally speaking, parameter modifications to the registrar service (registration of the end devices with Office Phone IP) for end devices not administered by Tele2 are not communicated, but must be carried out dynamically in accordance with RFC3263 (via DNS NAPTR and SRV requests)
Access for all Office Phone IP configurations for domain administrators and end users: http://sip.myzone.at
All Office Phone IP end devices require a DHCP service. This must assign at least the following options to the end devices:

- IP-Address
- Subnet Mask
- Default Gateway (Router)
- Domain Name
- Domain Name Server

Registrar (SIP Outbound Proxy Server)

- sip-reg.utanet.at:5082, UDP

Configuration Server

- All snom end devices receive their configuration via HTTP from http://sipcont.utanet.at

The configuration server can also be assigned over DCP. In this case, the option 66 (TFTP server) must include the content “sipcont.utanet.at.” The option 67 (boot file) should be empty. Each type of snom end device automatically enters the correct value. The configuration file for all AudioCodes MP10x type phones is made available on the domain management page http://sip.myzone.at and must then be imported into the phone manually. Certain Cisco phones that were agreed after prior consultation with Tele2, receive their configuration on tftp://sipcont.utanet.at. For IP VPN connections (ViPNet) that have no Internet access, Tele2 offers the following services in VoIP VPN:

- NS Server: 62.218.251.193, 62.218.251.194
- Time Server: siptime1.utanet.at, siptime2.utanet.at
- Management access (only in ViPNet): http://sipmgmt.utanet.at

Furthermore, only authenticated connections are possible. This means that only connections originating from users who have authenticated themselves over digest authentication (RFC3261 and References) are possible.

16. Cancellation of Telephones

The Office Phone IP service and all its components (telephones, options etc.) are concluded for the agreed contractual period by the customer with a waiver of cancellation. If the customer's financial situation makes it necessary to return individual telephones before the end of this period, a commercial agreement must be reached with the responsible Tele2 sales representative. Otherwise, the monthly charges applicable until the agreed termination date can be invoiced at once with the next bill. Cancelled telephones must be returned in proper condition (see Section 4 End devices) and in suitable protective packaging material to the following address:
Head ON Communication GmbH
Marxergasse 25
1030 Vienna

17. Maintenance and Support

The Tele2 VoIP hotline under the number 0800 800 VoIP (0800 800 8647) is available around the clock Mon-Sun. to report technical faults and obtain technical support. Faults in central components of the Tele2 network are proactively monitored and rectified by Tele2 Mon-Sun. 24 hours a day.

17.1. Replacing Faulty End Devices

Tele2 will replace individual faulty end devices supplied by it once a fault report has been submitted. The replacement device will be shipped within two working days of the fault report. Faulty devices may be replaced in one of 2 ways:

- By post: The customer must return the faulty end device to the following address no later than 14 days after the replacement devices have been delivered:
Head ON Communication GmbH
Marxergasse 25
1030 Vienna
- Immediately and reciprocally: The replacement devices are delivered by a courier service hired by Tele2 which will also take receipt of the faulty devices.

If faulty devices are not received by Tele2 within 14 days of the replacement devices having been delivered, the customer will be invoiced for the replacement devices.

Appendix 1: Standard Functions

iPBX Features	Description	Configuration/ Activation by	Parameters/ Options
Alternative subscriber number (Alias)	Additional numbers or names (dialable from a PC) can be allocated in addition to normal extension numbers, e.g. when switching to a new numbering plan, the old number can be assigned as an alias to ensure continued accessibility.	System administrator	Numerical or alpha numerical subscriber numbers
Call waiting internal/external	When an extension is engaged a called subscriber is made aware of a waiting call by a call-waiting tone	Configured by the system administrator, activated by the user	Call waiting activated or deactivated
Call lists	Log of all calls categorized as missed calls, received calls and dialed numbers with the possibility of dialing the logged numbers	User	Selection directly from the telephone or web interface
Call Pickup	Calls can be answered from another telephone within a local group (e.g. room) using a code or key, any number of groups are possible	Configured by the system administrator, activated by the user	Activation and deactivation by the user
Authorization classes	An individual authorization can be allocated to each extension, whereby 7 classes are offered: - Internal calls only - Calls within Austria - Calls within Europe	System administrator	

	- Global calls - In addition to Austria, Europe and worldwide calls, Austrian premium numbers (e.g. 0930) may be dialed		
Non-blocking internal traffic	All users can telephone with one another simultaneously, external calls limited by the number of outside lines		
Three-party conference	A decision can be made during a call whether an additional caller should be added on to the call	User	Business and Premium telephones
Extension	Callers can dial in directly to the extensions	System administrator	
Itemized call list	In addition to the itemized call list that is provided with the monthly bill, details of all calls, including internal calls, can be retrieved any time over a web browser		Date, time, A-subscriber number, B-subscriber number, duration Customer number allocation (only with convenience packages)
Fax connection	Faxes (Group 3) can be connected to Office Phone IP with an analogue adapter	System administrator	
Remote maintenance and configuration	All central functions can be configured and activated over a web browser	System administrator, user	
Hands-free function	Telephony without the caller having to hold the receiver	User	Business and Premium telephones
Customer-specific feature code prefix	Usually feature codes are initiated and completed with a #, this may be changed if required	Tele2 system administrator	
Customer-specific feature codes	Feature codes are usually assigned by Tele2, they may be changed if required	Tele2 system administrator	
Speed dialing	Up to 100 numbers per telephone, to be imported from a central file	User	Business and Premium telephones
Line occupancy (outside line obtained directly)	To dial out from the private branch exchange (external call), an outside line number, usually 0, must be dialed before the telephone number	User	
Alternation	The user can switch between 2 calls	User	
Night service extension	Special configuration, so that if, for example, the switchboard is not manned, calls are diverted to another extension, an answering machine responds etc.	System administrator	
Name display on telephone	Within the private branch exchange the name of the caller is displayed	System administrator	Business and Premium telephones
Parallel ringing	Up to 5 additional telephone numbers can be called in parallel (e.g. the mobile phone and a telephone in the secretariat also ring), the call is taken by the telephone that responds first, the other telephones then stop ringing	User	Starting and finishing times during which the function should be used can be selected
Parking	A call is parked within a local group (e.g. room) using a destination key and can be taken from any telephone in the group with the destination key (key flashes), similar to switching a call	User	
Group call distribution	Callers are forwarded to a special extension that distributes the calls to the actual extensions. Up to 20 extensions can be simultaneously signaled over group call distribution. The telephone that answers first has the call. Another feature is the “group call distribution bell” (e.g. telephone within a group). All calls that land at the group call distribution bell can be retrieved using the code “pick up group call distribution bell.”	System administrator	
Call forwarding always	All incoming calls are immediately forwarded to registered number (e.g. voice mail)	User	Starting and finishing times during which the function should be used can be

			selected
Numbering plan	Any numbering plan can be chosen, 1-5 digits	System administrator	
Call forwarding after time when telephone is free	All incoming calls are forwarded to a registered number (e.g. voice mail) if they are not taken after a configurable time	User	Starting and finishing times during which the function should be used can be selected
Call forwarding if the line is occupied	All incoming calls are immediately forwarded to a registered number (e.g. voice mail) if the line is occupied	User	Starting and finishing times during which the function should be used can be selected
Consultation call	An additional user can be consulted during an active call	User	
Call-back if engaged	If an extension (B) is engaged, the caller (A) can initiate a call-back as soon as the busy extension (B) becomes free, if the possibility of a repeated call attempt is indicated to (A), only internally	User	
Call-back if free	If an extension (B) does not take the call, the caller (A) can initiate a call-back as soon as extension (B) has conducted a telephone call, if the possibility of a repeated call attempt is indicated to (A), only internally	User	
Send number (CLIP)	Irrespective of his telephone the user who receives a call (B) sees the number of the caller (A)		
Telephone number is suppressed (CLIR)	The caller (A) can suppress their own number so that it cannot be seen by the called party (B)	User	
Connect	The user can connect calls to another number with or without prior announcement	User	
Repeat dial	The most recently dialed telephone numbers can simply be redialed	User	
Web administration	Configuration of the personal settings over a web interface	User	
Speed dialing	Important numbers can be dialed with a single-digit code	User	
Selective call divert	All incoming calls are immediately diverted to a number (e.g. voice mail), with the exception of one dialable number (e.g. secretariat)	User	Starting and finishing times during which the function should be used can be selected
Boss/Sec	In a boss/secretariat situation, the boss can divert all his calls to his secretariat, however, the secretariat can call the boss directly and connect calls	User	Starting and finishing times during which the function should be used can be selected
Itemized call list	List of all calls within the private branch exchange, internal and external	System administrator	
Customer number attribution (attribution of telephone calls)	The costs for a telephone call can be attributed to a customer number, evaluation by means of a web itemized call list and incorporation in other applications by means of CSV	User	During a call the customer number is always entered using the telephone keys
Call waiting music	Callers hear a message/music while the party taking the call connects the call to another user, message/music can be changed at any time	System administrator	
Message before response	Callers hear a message/music before the party they have called answers the telephone, message/music can be changed at any time	System administrator	

Appendix 2: Option Unified Messaging

Voice Mail

Every extension has the possibility to use an individual voicemail. A personal greeting can be recorded for the voicemail. Each user may configure the criteria for diverting a call to voicemail by web browser. Messages will be indicated on the telephones. Adequate storage capacity is provided for each user (> 30

minutes) so that even if calls are diverted to voicemail for a lengthy period of time, e.g. while the user is on holiday, as many calls as possible can be recorded. Voicemail can be retrieved from any telephone (e.g. mobile) with a user name and PIN. Voicemail can be ordered as an option. The function is then available to all extensions, depending on the configuration chosen by the customer administrator.

Voicemail2email
Voicemail2email automatically forwards messages left on the voicemail box as an attachment (WAV) to the address of the user defined in the system configuration. The e-mail can be opened and heard in the mail program. The Voicemail will be marked as retrieved in the voicemail box and automatically deleted after 30 days. Forwarded voicemails include the number of the caller in the "subject" to enable the user to gain a rapid overview and priorities them in the mail program.

Fax2email
This option allows incoming fax messages to be forwarded to personal e-mail addresses (the address of the user defined in the system configuration) (TIFF format). The customer dials, for example, an additional fax identification number in front of the subscriber's extension number (e.g. 9). Example: extension number is 17, fax extension number is therefore 917. For system reasons, we cannot be held liable for e-mails (fax, voicemail2email) that cannot be delivered.

Appendix 3: Option CTI/Soft Client

The CTI/soft client makes it possible to dial telephone numbers from any PC application over an Office Phone IP telephone (prerequisite is that the application supports the TAPI interface) Moreover, pop-up windows indicate the call and number of the caller on the PC when a call comes in. The application stores all dialed and received numbers and allows them to be dialed quickly with the touch of a key. The correct number of client licenses must be ordered for the number of PC workstations that are to use this function. System requirements: Operating system Windows 98/2000/XP/Vista/7

Appendix 4: Option Switchboard Software

The switchboard software expands the functions of a normal PC workstation to include those of a telephone exchange. The PC then acts as the hub within the company for incoming calls, switching calls etc. and can also be used for office work. The software supports the following functions:

- Acceptance of incoming calls
- Switching to internal and external subscriber numbers
- Acceptance of specific calls from a queue
- Status display of the selected extension (free, busy, unavailable)
- Status display of extension groups (local groups)
- Dialing of outgoing calls

System requirements: Operating system Windows 98/2000/XP/Vista/7

Appendix 5: Office Phone IP Compatible End Devices

The following products have been tested with Office Phone IP and fulfill the functions (depending on the technical possibilities of the individual products, e.g. with or without display):

Manufacturer	Designation
snom	300
snom	320
snom	360
snom	370
snom	820
snom	m3
Polycom	IP 7000