Consertis Telephony is a service that provides a high-performance telephone system and enables business customers to make high-quality voice calls. The telephone system is maintained by Consertis and hosted in a state-of-the-art data center to ensure the highest level of fail-safety.

The customer connects his end devices via an existing broadband Internet connection.

The system can be managed by the customer or partner via a web interface.

DESCRIPTION OF FUNCTIONS

Some of the functions mentioned below are dependent on the support of the end device.

CALL MANAGEMENT

- Hold call (park)
  The caller is put on hold by pressing a button and hears the system's music on hold. The held call can be resumed at any time. This function is important for transferring calls and for making inquiries with other employees.
- Queries
  The active call is put on hold, a second call (internal or external) is set up. The first call can be transferred to the second caller or resumed by the second caller after consultation.
- Toggling
  Switch between two or more simultaneous calls.
- Pick up held call
  Resume a call that has been put on hold.
- Pick up call (take over)
  Pick up a call from another ringing phone to your own phone.
- Redial of outgoing, received and missed calls
  Call lists of outgoing, received and missed calls are stored on the phone.
  At the touch of a button, a call can be set up to any of these numbers stored in the lists.
- Call waiting
  If a call comes in while a call is already in progress, this is signaled by a knocking sound in the active call and a flashing LED on the phone.
- On-hold music
  While a call is on hold or transferred, the caller hears music on hold. Music on hold is played into the telephone system in self-administration via web browser in .wav or .mp3 format and linked to the dial plan.
- Busy on Busy
  If a call is already active, further incoming calls will be rejected.
- Switching (internal, external) with and without queries
  It is possible to transfer a call within the telephone system or to any external number. This can be done directly or after prior consultation with the subscriber to whom the call is to be transferred.
- Outgoing DTMF
  Dial tones are transmitted when the telephone keys are pressed. This is necessary for controlling interactive menus of automatic switching systems.

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- VIP-List
  Calls from numbers entered in the VIP list are transferred according to separately defined rules. For example, they can be put through directly to the managing director or given priority in waiting loops.
- Address Book
  Most SIP phones offer comprehensive address book features
- Devices per extension
  Any number of devices can be registered per extension.
- Busy lamp
  Signaling the call status of individual extensions on the terminal device.
- Parallel call
  Freely definable parallel call to other internal extensions or external subscribers.

CLIP/CLIR, FILTER LISTS, TIMER

- Caller ID of incoming calls (CLIP)
- Suppress or display own phone number (CLIR)
- Filter lists
- Separate dialing plans can be created for phone numbers entered in filter lists, e.g.
  - Whitelisting
    Only numbers entered in a whitelist are put through, others are rejected or diverted.
  - Blacklisting
    Numbers entered in the blacklist are rejected or redirected.
  - Prioritizing
    Numbers entered in a prioritized filter list are delivered from a queue to a free terminal before the other numbers. Usually, the "first in, first out" principle applies.
- Timer
  Any time slots can be created and provided with their own dialing plan, e.g.
  for business hours, holidays, vacation...

CALL FORWARDING / DND (DO NOT DISTURB)

- Forward all calls (internal, external)
- Forwarding when busy (internal, external)
- Forwarding after timer (internal, external)
- DND „Do not disturb“
  This circuit is activated and deactivated on the device. Incoming calls are rejected, the telephone signals "busy".

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CONSERTIS

ANSWERING MACHINE

- Professional voicemail system for extensions assigned to a telephone
- Individual welcome text & voicemail for each extension number
- Voicemail delivery
  Messages are sent as a .wav file by e-mail to a definable e-mail address
- Remote query with PIN code

GROUPS / QUEUES

- Any number of call groups per system
- Formation of call groups of any size possible
- Groups
  Groups enable the simultaneous ringing of several devices. In some systems, this function is also referred to as parallel calling.
- Waiting Queues
  Calls held in a queue ring at the terminals assigned to the queue according to three possible principles.
- Parallel call
  - All devices in the queue are ringing
  - It rings in a specific sequence, one after the other, at each terminal in the queue.
  - It always rings randomly at a device in the queue.
- Prioritize
  Within waiting loops, call numbers entered in a filter list can be delivered to a free terminal before other waiting calls. The "first in, first out" principle usually applies.

AUTOMATED SWITCHING

Switching by dial tone control: The caller navigates through a switching menu with the help of dial tones. The design of the announcement texts and menu control is free.

- Any number of voice dialogs per system
- Customized selection texts
- Forwarding, hang up and busy
- Flexible handling of incoming calls
- Any chaining depth for voice dialogs possible

FAX

- Fax2Mail
- Incoming faxes are attached to emails in pdf format and sent to an email address specified by the customer.
- For outgoing faxes, the customer uses the web interface. Fax logs can also be viewed via the web interface.
- A classic fax machine can also be connected via special adapters. The compatibility must be tested, however.

CTI (COMPUTER TELEPHONY INTEGRATION)

- SIP-TAPI
  The telephone system can be integrated into the IT environment via SIP using a TAPI interface. For example, dialing from Outlook contacts is possible.

SITE CONNECTIVITY

- Cross-site virtual telephone system
- End devices can be integrated into the telephone system regardless of their location. Therefore, the telephone system is particularly suitable for
- Companies with distributed locations
- Integration of home office workstations and field service employees
- Support for multiple phone numbers
  Multiple numbers - both geographical and mobile - can be used within a telephone system. The only limitations are those imposed by RTR (Rundfunk und Telekom Regulierungs GmbH, see http://www.rtr.at/de/tk/Nummerierung).

CONFERENCE CALLS

- Any number of conference rooms per facility
- Conferences with up to 50 internal and external participants
- PIN authentication separately for conference leaders and participants
- Locking conferences until the conference leader logs in
- Automatic termination of conferences after the conference leader logs out
- Mute all participants of the conference
- Transfer call to conference
- Announcement of additions and departures

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WEBINTERFACE

The telephone system is managed via a web interface for self-administration. The login to the web interface is done by user ID and password. The following functions are available in the web interface:

- User management: create, edit and delete users and extensions
- Devices: create, edit and delete devices
- Automatic configuration of end devices: An automatic configuration file can be created for certain end devices using the web interface.
- Device list: Overview of all devices of the telephone system
- Call groups: Creating, editing and deleting call groups
- Time windows: create, edit and delete time windows
- Voicemail: Create, edit and delete voicemail boxes.
- Faxboxes: Create, edit and delete faxboxes
- Text on hold: upload and delete texts and music for on hold, music on hold etc.
- Announcements: Upload and delete announcement texts
- Menus: Create, edit, and delete interactive, dial tone menus.

FINAL DEVICES

End devices register with the telephone system using SIP. In principle, the use of any SIP-capable terminal device is possible. Consertis can guarantee the proper functioning of the telephone system only with end devices tested and offered or recommended by Consertis. Consertis provides the customer with all information necessary for the connection and operation of any SIP device, such as user ID and password, as well as server addresses and port addresses necessary for operation. There is no entitlement to additional support for end devices not recommended by Consertis.

CONNECTION TO THE PUBLIC TELEPHONE NETWORK

The connection to the public telephone network is provided by Consertis. The connection depends on the bandwidth of the Internet connection, as described in the chapter "Prerequisites for operation". The recommended minimum bandwidth per call channel is 100 kbit/s.

GEOGRAPHIC CALL NUMBERS

The Communications Parameters, Fees and Value-Added Services Ordinance 2009 (KEM-V 2009) defines the public numbering plan and the public dialing plan as a subplan for communications parameters, as well as the fees and general regulations regarding value-added services. For the various

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number ranges, the allocation criteria and usage features are defined and the procedure for obtaining usage rights is regulated. Section 49 of the KEM-V 2009 stipulates that "Geographical numbers are national telephone numbers and are used to address fixed network termination points."

Consertis shall technically ensure that an assigned geographical telephone number can only be used by the subscriber in accordance with § 49 KEM-V. Accordingly, the network termination point is understood to be the network connection at the Consertis modem or at a fixed receiving device (network termination) installed by Consertis at the customer's premises. For the use of extensions at other locations, a location-independent call number is additionally assigned.

LOCATION-INDEPENDENT NUMBERS

Location-independent numbers are identified by the 0720 area code. They are national telephone numbers and are used to address subscribers in connection with public

telephone services that enable the subscriber to maintain his or her telephone number regardless of location in the fixed network or on the Internet.

EXTENSION NUMBER ALLOCATION

The assignment of extensions and terminals to the individual phone numbers is done in the web interface for self-administration and is the responsibility of the customer.

self-administration and is the responsibility of the customer. This is particularly necessary for the correct delivery of emergency calls. No claims can be made against Consertis based on the customer's failure to assign or reassign extensions to their geographical numbers.

EUROPEAN EMERGENCY NUMBER

Attention is drawn to the existence of the European emergency number 112.

The following requirements must be met for the proper operation of a Consertis telephone system:

NETWORK CABLING AND POWER

An Ethernet connection and a power connection are required for each end device. Most end devices draw power either from their own power supply unit, or alternatively via PoE (Power over Ethernet). Consertis recommends the implementation of PoE.

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INTERNET CONNECTION WITH APPROPRIATE BANDWIDTH

A telephone call via VoIP (Voice over Internet Protocol) requires a bandwidth of up to 96 kbit/s, depending on the codec used. This bandwidth, multiplied by the maximum number of simultaneous calls, must be provided in addition to the other Internet traffic. A dedicated Internet connection for VoIP only is ideal. If voice and data are used simultaneously, the bandwidth requirement for voice should not exceed half of the total capacity. Prioritizing VoIP traffic in WAN and LAN helps minimize performance problems in non-dedicated networks.

ANALOG PHONE ADAPTER

Analog telephone adapters (ATA) - also called IP a/b adapters - are cost-effective products that connect conventional analog terminals (telephones, fax machines or door intercoms) with Consertis telephony.

Note: Since fax is a historically evolved protocol, there are in practice diverse protocol deviations, so that an ATA cannot be compatible to all fax devices in the market.

RESTRICTIONS

VoIP technology cannot map all the functions of classic voice telephony (with ISDN or analog connection). In particular, Consertis Telephony therefore does not offer the following functionalities:

- Emergency call function in case of failure of the customer IP network. Consertis Telefonie recommends the use of a mobile device at each location.
- Connection of ISDN standard or special devices such as ISDN PC cards, fire detectors, ATM or credit card systems, franking machines and alarm systems.
- Special numbers and number blocking - Connections to special services are provided by Consertis within the limits of what is legally permissible and within the technical and operational possibilities of Consertis.
- Consertis reserves the right to block individual destination numbers, groups of destination numbers, or specific country codes, taking into account the interests of the customer.
- The use of call-by-call and CPS offers is not possible.

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