Tender documents

innovaphone PBX

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# General description of the innovaphone PBX telephone system

PBX according to H.323 / SIP standard

A telephone system based on Voice over IP technology (VoIP) should be offered.

In order to guarantee maximum investment protection, the telephone system should be completely standards compliant to the ITU-T H.323 standard with its sub-protocols as well as SIP standard version 2.

The user interface for all platforms (phone display, Web client on computer and smartphone) should be consistent and intuitive to use on the system, allowing users to quickly find their way around the application. A Presence indicator should update itself automatically and in real time on all devices, so that the user can see the availability of other subscribers at any time.

The tender has been drawn up based on the innovaphone IP6010 system. Alternative offers are permissible and welcome. Due to differences in system architectures, all attributes required for proper functioning and necessary components should be included in an additional offer.

## General feature sets

According to H.450

* H.450.1 Call forward / call diversion
* H.450.2 Call transfer, with / without consultation, before/after answering
* H.450.3 Call diversion unconditional, on busy and according to pre-set time delay
* H.450.4 Hold / toggle
* H.450.5 Call Pick-up
* H.450.6 Call waiting, respective signalling of waiting call to caller
* H.450.7 Message Waiting Identification
* H.450.8 Name identification
* H.450.9 Call completion on busy (CCBS) and call completion on no reply (CCNR)

According to SIP

* RFC 1889 RTP: Real-Time Transport Protocol
* RFC 2327 SDP: Session Description Protocol
* RFC 2396 Uniform Resource Identifiers (URI): Generic Syntax
* RFC 2543 SIP: Session Initiation Protocol
* RFC 2616 Hypertext Transfer protocol (HTTP / 1.1)
* RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
* RFC 2782 A DNS RR for specifying the location of services (DNS SRV)
* RFC 2976 The SIP INFO Method
* RFC 3261 SIP: Session Initiation Protocol, SIPS: SIP Security
* RFC 3263 Session Initiation Protocol (SIP): Locating SIP Servers
* RFC 3264 An Offer/Answer Model with the Session Description Protocol (SDP)
* RFC 3265 SIP-Specific Event Notification
* RFC 3326 The Reason Header Field for the Session Initiation Protocol
* RFC 3389 RTP Payload for Comfort Noise
* RFC 3515 The Session Initiation Protocol (SIP) Refer Method
* RFC 3550 RTP: Transport Protocol for Real-Time Applications
* RFC 3551 RTP Profile for A/V Conferences with Minimal Control
* RFC 3555 MIME Type Registration of RTP Payload Formats
* RFC 3578 Mapping of ISDN User Part (ISUP) Overlap Signalling to the SIP
* RFC 3680 SIP Event Package for Registrations
* RFC 3764 enumservice registration for SIP Addresses-of-Record
* RFC 3824 Using E.164 numbers with SIP
* RFC 3891 The Session Initiation Protocol ‘Replaces Header’
* RFC 3892 The SIP Referred-By Mechanism -SIP-aware filtering (to prevent SIP attacks)
* RFC 3842 SIP Message Waiting
* RFC 3311 re-INVITE
* RFC 2833 DTMF via RTP Channel , RTP payload for DTMF
* RFC 3325 Name identification
* RFC 3578 Overlap Dialling
* RFC 3420 Internet Media Type message/sipfrag
* RFC 3711 SRTP: Secure RTP

Other features

* En-block dialling / Overlapped sending
* Music-on-Hold
* Pick-up list, display on telephone, which calls can be picked up
* Calling Line Identification Presentation (CLIP)
* Display name
* Three party conference with internal and external participants
* DTMF tone transmission
* Fax over IP (T.38 real time fax)
* Automatic ring tone generation according to European standard
* Possibility to set up several wait queues with pre-definable time period before pick-up and freely definable announcement function before call pick-up
* Call back on busy
* Automatic time controlled call diversion to voicemail
* Messaging: Send and receive short messages directly from phone to phone
* Send message whilst “Do not disturb” is activated
* Electrical power supply only via Power over Ethernet.
* up to 120 channels in one height unit

Manager-assistant functions

* Number identification for distinctive signalling for numbers or groups of numbers
* Announcement function: Partner function, call partner accepts call automatically with speaker on / off
* Call diversion override for partners
* Busy lamp field, shows partner telephone status

Security

* Minimum information transfer
* Access rights can be configured at will for trunk line, international calls and special prefixes/telephone numbers including recursive filters for branch locations with remote access to the public network
* IP address filter
* Password protected authentication according to H.235
* Encrypted signalling according to SIPS
* Encrypted voice-data transmission according to SRTP
* multi-level administration authorisation
* optional, decentralised authentication service KERBEROS with cross real authentication to Microsoft Active Directory

More security protocols are listed under item 1.2.8.

Administration

* Browser protected administration, password protected, various authorisation levels
* Error search with readable log files, status indicators, PING and Traceroute connection test
* Save and readout configurations
* Firmware update via update server
* Adopt subscriber data from “MS Active Directory” for PBX

Scalability

* The system must be prepared to be able to manage at least 30% more subscribers.
* The system must be scalable

Multiple registration

Multiple registration should be possible on the telephones to enable flexible working at work stations. It should be possible to configure up to 6 extensions to one end device via Hot Desking. A new employee can login by entering name and password. For incoming calls it is possible to set signalling so that it is obvious who the call is directed at. For outgoing calls, it is possible to adjust what sender information is to be transmitted to the call recipient.

Multiple registration should also be possible on various devices. As soon as a user is registered to multiple telephones, all telephones signal an incoming call at the same time.

## Integrating the system in various scenarios

### Smooth migration

The system which is to be set up should provide the possibility of migrating slowly from the existing telephone system to VoIP until the existing maintenance contracts expire.

It should be possible to loop through the new PBX between the trunk line access point and the existing system without having to make changes to the traditional PBX. Instead of immediately having to replace the old PBX, it is possible to extend it gradually using VoIP technology. The number of subscribers can be extended as needed or can changeover from traditional telephony to the modern VoIP environment. Thus after a pre-defined period of time, the traditional PBX can be replaced and the VoIP PBX can take over all functions.

### Integrating locations

It should be possible to install the future system at several sites without additional effort. Every site should work independently and have its own access point to the public network (ISDN). A standard number call plan should be implemented at all locations, whereby the employees can make phone calls across locations just by using the extension number, all PBX features are available. Secure VPN connections should be used for these calls. If an ISDN line should fail or be overloaded, it should be possible to access another location’s trunk line by using a pre-fix.

In addition, it should be possible at will to have a secure access to the telephone system and to make/accept phone calls from any Internet access point via ICE and STUN even without VPN (tunneling) and without additional configuration. Voice connections should always comply with the highest security requirements via the security protocol DTLS-SRTP.

### Redundancy

In order to increase fail safety, the telephone system should be safeguarded by a second identical redundant system. The redundant system should reproduce all settings on the active telephone system simultaneously, so that if the active system should fail, there will be no noticeable configuration differences. Data replication should take place using the standardised LDAP protocol. The active gateway should switch the ISDN connection to the redundant system, making the ISDN access available for the redundant system without any losses in case of power failure.

During normal operation, end devices and gateways only register to the active PBX and do not use any additional licenses for the redundant system. If the active system should fail, the end devices and gateways register to the redundant PBX with the same parameters.

### Home offices

The PBX should be able to provide optimum integration for home offices. The home office needs a flat rate DSL connection without a fixed IP address for this. It should be possible to prepare the full configuration of telephones for home offices without the device being switched on in the home office.

Once the device has been connected, it should connect automatically to the PBX. The first step should be to establish a connection to the internet service provider using the PPPoE protocol. Based on this, the telephone sets up a connection to the company network with the integrated VPN client using the PPTP protocol and then carries out the registration to the PBX with its telephone number and all other attributes.

Internet access should be made available for the home office computer over an internal 2 port Ethernet switch on the telephone. The home office work station is thus connected entirely via the telephone. Other hardware such as separate modems, routers or switches are not required.

At the home office an internal phone number from the PBX should be used for incoming and out-going calls. All system attributes should also be available at the home office. CTI software should support group work and call billing should take place in the same way as for all internal PBX connections.

In addition, it should be possible at will to have a secure access to the telephone system and to make/accept phone calls from any Internet access point via ICE and STUN even without VPN (tunneling) and without additional configuration. Voice connections should always comply with the highest security requirements via the security protocol DTLS-SRTP.

### Integrating branch offices

A location concept should be designed stipulating a Master PBX in every configuration. Each subscriber’s location should be known on the Master PBX. If a subscriber is called from a location that is not configured at this location, the local PBX passes the call to the Master in the central system. The call is then forwarded to the target subscriber - either directly or over another location. The central system’s master function provides an automatic backup version for the local PBX. If the local PBX should fail, but the IP connection remains intact, the central system can immediately takeover the work of the local PBX.

In addition, it should be possible at will to have a secure access to the telephone system and to make/accept phone calls from any Internet access point via ICE and STUN even without VPN (tunneling) and without additional configuration. Voice connections should always comply with the highest security requirements via the security protocol DTLS-SRTP.

### Multi-client capability

The PBX should be multi-client capable if it is to be used as a Hosted PBX. This is understood to mean the capability of operating several clients virtually and entirely separate from each other with individual instances on the same device. The virtual PBXs should be managed entirely separately from each other so that individual applications, announcements and wait music can be used.

### Integrating mobile phones

The PBX should have a module that enables mobile phones to be integrated. Mobile phones should be registered as internal subscribers and all features will be available. Two stage dialling allows mobile phones to dial into the PBX. The subscriber is recognised and can now dial out as an internal subscriber. For the subscriber: incoming calls are signalled simultaneously using the Dual Forking principle on both the internal phone and the mobile phone. As soon as the subscriber picks up the call on a device, the other device stops ringing.

In addition to the simplified communication and the additional features on the mobile phone, this module should also make it easier to control costs.

### Security

* + - 1. Security protocols

**TLS**

Encryption and certificate-based authentication for various applications

**HTTPS**

HTTP over TLS, encrypted Web access to administration

**H. 235**

Authentication with encrypted password

**SIPS**

SIP over TLS, SIP security

**H. 460.17**

H. 323 over TLS, registration and signaling via TLS encrypted

**SRTP**

SDES-encryption of media data (voice, video...)

**DTLS-SRTP**

TLS encryption of media data

**LDAP via TLS**

Data encryption for contact data via LDAP

**IEEE 802.1X**

Access control to the network, also with EAP-TLS

**Kerberos**

Authentication via central server

**X.509 certificates**

Certificate-based authentication for TLS

* + - 1. Session Border Controller

In order to secure SIP trunks, the telephone system should have session border controller functionalities. These should be integrated directly into the innovaphone PBX. The innovaphone Session Border Controller shall either run as a separate process on the innovaphone PBX or be installed on a separate innovaphone VoIP gateway.

Features

* Security offloading
* Media pinholding
* Header manipulation
* Media relay/media anchoring
  + - 1. Reverse Proxy

The Reverse Proxy shall serve as a central authority for all inbound connections from the public Internet. Among other things, it should be possible to detect attacks and to display this information.

H.323

* H. 323 registrations can be accepted via TCP or TLS. Call forward per TCP or TLS, not necessarily as well as incoming.
* Registration call forward registration based on the gatekeeper ID
* Certificate verification for TLS
* For TLS, optional verification that the certificate name matches the registration name

SIP

* SIP registrations can be accepted via TCP or TLS. Call forward per TCP or TLS, not necessarily as well as incoming.
* Registration call forward registration based on the domain
* Certificate verification for TLS
* For TLS, optional verification that the certificate name matches the registration name
* SIP invites can be accepted via TCP or TLS. Call forward per TCP or TLS, not necessarily as well as incoming.

HTTP

* HTTP requests can be accepted via TCP or TLS. Call forward per TCP or TLS, not necessarily as well as incoming.
* Forwarding of requests based on host name and URL

LDAP

* LDAP connections can be accepted via TCP or TLS. Call forward per TCP or TLS, not necessarily as well as incoming.
* Connection forwarding based on the domain in the authentication

Features

* Different forwardings can be configured for each protocol
* Intrusion detection based on the repeat rate of unsuccessful connection attempts
* Display the current sessions and statistics
* Successful and unsuccessful connection attempts are counted and displayed
* Creation of events for potential attacks when detected
* IP blacklist
  + IP address is automatically added to a blacklist when an attack is detected
  + Removal from the blacklist after timeout
  + Manual removal or addition of an IP address from or to the blacklist by the administrator
  + Accessible blacklist
* IP whitelist
  + IP addresses that are explicitly recognised to be trustworthy by the fact that for example there are a lot of successful logins
  + Manual addition of a trusted IP address to a whitelist
  + Accessible whitelist
* Limitation to special IP networks can be configured

### Conference server

The system should include a conference server. It should be possible to support conference rooms with up to 60 participants. It should be possible to configure conference rooms dynamically or statically. It should be possible to reserve the number of seats for a conference room.

### Call list (call history/log)

The phone system should have a detailed call log, which lists the last incoming and outgoing calls. It should be possible to trace previous communication for up to 1 month. It should be possible to call up structured lists to show all calls, only incoming calls, only outgoing calls and active callbacks.

Each individual log entry should show information about the call status (e.g. grey icon = successful connection, red icon = no connection), the call direction (e.g. by an arrow), name (if available) and the number of the remote device as well as date, time and duration of the call. The call log should be able to offer all other telephony features such as call forward, parking, etc., allowing a user after absence and upon his return to his workplace to have available detailed information of all missed calls (who took the call? Where was the call forwarded to?)

Furthermore, it should be possible to perform new functions with the respective subscriber via the individual entries in the call log. It could be a callback or a chat, it should offer the possibilities of adding the subscriber to the Favourites list and storing local contacts, launching a linked application and triggering a “please call me back” via email.

### Presence status

Each subscriber should be able to set his Presence status on the device and also via a Unified Communications client. This Presence status should display the predefined partners directly over the corresponding icons. This information should also be indicated on the display to all subscribers of the PBX when a call is placed. Examples for Presence status include vacation, away or at lunch. In addition to the status, it should also be possible to create an individual Presence note, which is automatically sent to the caller. This could for example include the vacation dates. In addition, it should be possible to extract and process calendar information to become Presence information, a Presence note should then be generated and added automatically. In Office applications, this Presence information should also be displayed with corresponding icons and notes. Updating Presence should take place automatically in real time.

With the so-called SIP Federation it is possible to transmit the Presence status and text notes to partners outside of one’s own telephone system. Thus, it is possible to extend communication to business partners and to add value to a collaboration.

### Rollout

Configuration templates and specific templates should be stored for fast, individual delivery and initial installation that can be used immediately and automatically after registration by the central system. This enables company-wide standard function keys to be pre-set.

Once approved, the modified user settings can stay stored on the PBX. Individual settings for voice, ring tones and function keys are kept independent of the end device. Users using different phones at different times (hot desking) thus get their familiar telephone settings back.

## Interfaces (API)

In order to ensure optimal integration of software solutions, the necessary interfaces (API) are to be made available.

### XML-API

The XML-API should enable the PBX to be controlled easily. The interface should thus communicate with the PBX over the SOAP protocol (Simple Object Access Protocol).

### TAPI

Standard applications should use the TAPI interface to communicate with the PBX. It must be possible to be able to use the TAPI service provider in both 1st party configuration (every computer controls a line) and in 3rd party configuration (a server controls several lines). The TAPI interface must be available without limitations at all PBX locations.

## Application platform under Linux

The open-source Linux operating system should be operated in parallel to the firmware on the hardware of the central system. There should be the possibility of operating 3rd party applications or existing programmes in parallel via the Linux Application platform. A separate server should not be necessary, instead the applications are to be operated directly on the PBX’s Compact Flash card.

# Central devices

Gateways with ISDN interfaces should be used to connect to the public telephone network. The gateways should be stand-alone devices and geared to be used in 19” technology. In order to guarantee the highest possible security, the gateways should not be based on variations of the most well-known operating systems Windows or Linux.

All core components (gateways, analogue adapters etc.) should be set out in robust 19” technology, if possible without failure prone components with moving parts (e.g. fans, hard disks etc.) Power supply for all components must, in the future, be secured over PoE (according to IEEE 802.3af). It should be possible to store configuration files and announcement texts for wait queues and voicemails on the gateway on commercial and cost-effective Compact Flash or SSD memory cards.

Administration should take place over a WEB browser with HTML without requiring additional software. It should be password protected with a secure authentication. In case of rebooting, the system should be operational again after 10 seconds.

At the same time, the gateway should be able to schedule the VoIP traffic with established SIP providers such as SIPGate, Toplink, HFO, Tele2 etc.

## ISDN VoIP gateway innovaphone IP6010

It should be possible to use the central unit as a loop through variant for scenarios such as smooth migration and also as a fully-fledged PBX without hardware exchange being necessary. The system should be freely scalable and should be recommended for PBX and UC installations up to 500 users. In addition, it should also give users the possibility of registering directly with a VoIP provider. A maximum of 60 calls to ISDN should be made in parallel on one system. The gateway should also represent the conferencing unit for up to 60 subscribers. It should also be possible to use it as a Session Border Controller (SBC) and Reverse Proxy, in combination with other features or as a dedicated device.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the Compact Flash card.

Interfaces

* S2M Interfaces (PRI):
  + 2 x in TE mode for connection to trunk lines **or**
  + 2 x TE mode and 2 x NT mode for loop-through operation
  + Interfaces can be selected at will and activated individually with a license key
* 1 x S0 Interface (BRI): ISDN connection for Routing, Admin, Synchronisation, Backup etc. TE or NT Mode
* 2 x Fast-Ethernet: 10/100-BASE-TX (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet according to IEEE 802.3af (class 3), Energy Efficient according to IEEE 802.az
* Compact Flash: 1 x Compact Flash card type I slot

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP(Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for up to 10 channels
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP- TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## ISDN VoIP Gateway innovaphone IP3011

It should be possible to use the central unit as a fully-fledged PBX or as a media gateway without hardware exchange being necessary. It should be freely scalable and should be recommended for PBX and UC installations up to 500 users. A maximum of 30 calls to ISDN should be made in parallel on one system. In addition, it should also give users the possibility of registering directly with a VoIP provider. The gateway should also represent the conferencing unit for up to 30 subscribers. It should also be possible to use it as a Session Border Controller (SBC) and Reverse Proxy, in combination with other features or as a dedicated device.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the internal SSD memory. In addition to the commonly used codecs, the device should also support the modern Opus codec that realises HD voice at low bandwidth.

Interfaces

* 1 x S2M interface (PRI): for connection to ISDN-S2M trunk lines, use on E1 or T1 connection, RJ-45 (modular Jack 8P8C), TE or NT mode
* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az
* SSD: 1 x internal slot for mSATA SSD hard drives, power consumption up to 2.5W, compatible with the JEDEC MO-300 standard with 50.8 x 29.85 mm external dimensions

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for up to 7 channels
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* Opus-NB
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## ISDN VoIP gateway innovaphone IP1130 (media gateway)

It should be possible to use the central unit as a media gateway without hardware exchange being necessary. A maximum of 60 calls to ISDN should be made in parallel on one system. The media gateway should also represent the conferencing unit for up to 30 subscribers. The media gateway should also represent the conferencing unit for up to 30 subscribers. It should also be possible to use the device as a Session Border Controller (SBC) and Reverse Proxy, in combination with other features or as a dedicated device.

Interfaces

* 1 x S2M interface (PRI): for connection to ISDN-S2M trunk lines, use on E1 or T1 connection, RJ-45 (modular Jack 8P8C), TE or NT mode
* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for up to 7 channels
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## ISDN VoIP gateway innovaphone IP0011

The central unit shall represent a PBX with direct connection to the IP provider without a link to the traditional telephone network or with a link to the traditional telephone network through a remote media gateway. It should also be possible to use the device as a Session Border Controller (SBC) and Reverse Proxy, in combination with other features or as a dedicated device.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the internal SSD memory. The device should be recommended for structured PBX and UC installations up to 25,000 subscribers (possibly in combination with other gateways).

Interfaces

* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az
* SSD: 1 x internal slot for mSATA SSD hard drives, power consumption up to 2.5W, compatible with the JEDEC MO-300 standard with 50.8 x 29.85 mm external dimensions

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for up to 7 channels
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## ISDN VoIP gateway innovaphone IP811

It should be possible to use the central unit as a loop through variant for scenarios such as smooth migration or as a fully-fledged PBX without hardware exchange being necessary. The system should be freely scalable and should be recommended for PBX and UC installations up to 200 users. In addition, it should also give users the possibility of registering directly with a VoIP provider. A maximum of 10 calls to ISDN should be made in parallel on one system. The gateway should also represent the conferencing unit for up to 10 subscribers.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the internal SSD memory. In addition to the commonly used codecs, the device should also support the modern Opus codec that realises HD voice at low bandwidth.

Interfaces

* 5 x S0 interfaces (BRI): for connection to digital S0 trunk lines or end devices, RJ-45 (modular Jack 8P8C), TE or NT mode, additional "power-off-loop" loops through 2 unpowered ISDNs
* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az
* SSD: 1 x internal slot for mSATA SSD hard drives, power consumption up to 2.5W, compatible with the JEDEC MO-300 standard with 50.8 x 29.85 mm external dimensions

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for up to 2 channels
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* Opus-NB
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## ISDN VoIP gateway innovaphone IP411

The compact system should be a combination of an ISDN gateway with an integrated gateway as well as 2 analogue ports to connect fax machines or other analogue end devices. It should be freely scalable and should be recommended for PBX and UC installations up to 50 users. In addition, it should also give users the possibility of registering directly with a VoIP provider.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the internal SSD memory. In addition to the commonly used codecs, the device should also support the modern Opus codec that realises HD voice at low bandwidth.

Interfaces

* 2 x S0 Interface (BRI): for connection to digital S0 trunk lines or end devices, RJ-45 (modular Jack 8P8C), TE or NT mode
* 2 x a/b interfaces (FXS): for connection to analogue end devices, RJ-11 (modular Jack 6P2C)
* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az
* SSD: 1 x internal slot for mSATA SSD hard drives, power consumption up to 2.5W, compatible with the JEDEC MO-300 standard with 50.8 x 29.85 mm external dimensions

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for 1 channel
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* Opus-NB
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## Analog-VoIP-Gateway innovaphone IP311

The central unit should represent an analogue VoIP gateway, which has 4 analogue FXO interfaces and 2 analogue FXS interfaces. Analogue trunk line connections should be converted as a result in an IP line. It should be possible to use the unit as an analogue media gateway as well as a PBX. It should also be possible to use the device as a Session Border Controller (SBC) and Reverse Proxy, in combination with other features or as a dedicated device.

The central unit must be prepared to operate an application platform that runs separately on Linux. Additional applications that have been developed for Linux operating systems should be installed directly on the gateway and should make use of the internal SSD memory. In addition to the commonly used codecs, the device should also support the modern Opus codec that realises HD voice at low bandwidth.

Interfaces

* 4 x a/b interfaces (FXO): for connection to analogue trunk lines, RJ-11 (modular Jack 6P2C)
* 2 x a/b interfaces (FXS): for connection to analogue end devices, RJ-11 (modular Jack 6P2C)
* 2 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 (modular jack 8P8C), LED display for activity and 100 Mbit mode, must be possible to address both connections separately, Ethernet redundancy over RSTP, Power over Ethernet pursuant to IEEE 802.3af (class 3), energy efficient pursuant to IEEE 802.3az
* SSD: 1 x internal slot for mSATA SSD hard drives, power consumption up to 2.5W, compatible with the JEDEC MO-300 standard with 50.8 x 29.85 mm external dimensions

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS (SIPS)
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711 for 1 channel
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* Opus-NB
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## Analogue VoIP Gateway innovaphone IP38

The central unit should represent an analogue VoIP gateway, which has 8 analogue FXO interfaces. Analogue trunk line connections should be converted as a result in an IP line. It should be possible to use the unit as an analogue media gateway as well as a PBX.

Interfaces

* 1 x Ethernet: RJ-45 interface (modular jack 8P8C), 10/100-BASE-TX (auto negotiation) with “Power over Ethernet” according to 802.3af (Class 1), Energy Efficient according to IEEE 802.3af (Class 3)
* 8 x a/b interfaces (FXO): for a connection with at least one RJ-11 (modular jack 6P2C) plug per analogue channel, configuration on the middle PIN

Protocols

* H.323 version 5 including H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP version 2 (including HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP (real time protocol for voice data transfer)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* DTLS-SRTP
* H.460.17 / ICE
* T.38 real time fax with possibility to fall back to G.711 for 1 channel
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p / 802.1q
* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation according to G.168
* PPPoE DSL access
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support (including DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

The central unit should schedule up to 32 tunnels to PPTP in parallel. Encryption should take place with MPPE. External H. 323 and SIP entries should be terminated according to ENUM.

## Application of the PBX software on the virtualisation platform VMware (IPVA)

As an alternative to using a gateway, it should also be possible to operate the PBX software virtually in a VMware environment. Despite the virtualisation of the PBX the complete scope of the communication solution should be available. The VMware environment makes it possible to set up multiple virtual servers on a single hardware platform. This should make it possible for phone systems to be centrally managed with a variable number of different PBX entities. The installation of a new VoIP phone system should be limited to setting up a PBX in a new VMware entity. In doing so, it should be ensured that the VoIP telephone systems of the individual subscribers are safely separated and can be easily managed and billed.

Solutions of any installation size should be possible. Should a VMware environment already exist, it should be possible to run the corporate communication via this infrastructure by setting up a new VMware entity. The use of an appropriate gateway should make it possible for the virtual appliance to also manage ISDN and analogue connections.

Features

* Hardware independent
* Runs directly on VMWare
* Recommended platform for structured PBX and UC installations up to 25,000 subscribers (possibly in combination with other gateways)
* ENUM for H. 323 and SIP entries
* Playback of announcements that are stored as a file on a virtual CF/ SSD card or a Web server
* PBX hosting - multiple PBXs on a HW, customer separation
* Infinitely scalable
* Managed services

Protocols

* H.323 Version 5 inclusive H. 225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450
* SIP Version 2 (inclusive HTTP digest authentication) conform RFC 3261
* SIP over UDP, TCP, TLS (SIPS)
* RTP (real time protocol for voice data transmission)
* SRTP (Secure RTP)
* RTCP (real time control protocol, first level of “Quality of Service”)
* T.38 real time fax
* DTLS-SRTP
* H.460.17/ ICE
* TOS and DiffServ for prioritising IP packets
* VLAN Priority according to IEEE 802.1p and q
* PPPoE support
* VPN Tunnelling with PPTP, up to 32 parallel tunnels, encryption with MPPE
* NAT Network Address Translation, H.323 NAT, STUN, TURN
* IEEE 802.1x (EAP-TLS/ EAP-MD5)
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/ MED support
* NTP support (inclusive DNS for NTP)
* Reverse proxy, various SBC functionalities (see 1.2.8 Security)

# VoIP phones

It should be possible to use various telephone devices. A larger number of system telephones that support the full PBX feature set, a basic telephone to satisfy lower requirements and a telephone application for Windows computers, especially designed for work places where the computer screen is highly used usually with a headset.

The table-top telephones should use both the H.323 and the SIP standard as transmission protocols, should have an internal Ethernet switch in order to connect further devices such as a computer to the network. Multiple registration on the telephone allows several registrations to take place at the same time. In addition, parallel operation of several SIP and H.323 registrations should be possible. It should be possible to import firmware updates automatically and simultaneously to all end devices.

## VoIP phone innovaphone IP111

Interfaces

* 1 x Fast Ethernet: 10/100-BASE-TX (auto negotiation), RJ45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient Ethernet” according to IEEE 802.3az
* 1 x Fast Ethernet: 10/100-BASE-TX (auto negotiation), RJ45 interface (modular jack 8P8C) for PC connection
* Connector for external power supply

Hardware

Power supply

* Power supply: external power unit: 12 V, 6 W
* Power over Ethernet according to 802.3af, Class 1

Storage

* 128 MB DDR3, 16 MB Flash

Environment

* Operating temperature: 0 °C to +45 °C,
* Storage temperature: -10 °C to +70 °C
* Humidity: 10% to 90% (non-condensing)

Display

* Colour display, 320 x 240 pixels (3.5 inch)

Keypad

* Telephone keypad
* 16 function keys
* 32 partner keys

Housing

* Measurements: 21 x 15.5 x 3.5 cm (base)
* Weight: ca. 630 g

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729 (available as software license)
* Opus-NB, Opus-WB
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* IEEE 802.1x (TLS-EAP), 802.1X Proxy
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free and open listening
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

The phones have 12 buttons to the left and right of the display, 8 of which can be programmed at will as function keys. A total of 16 functions can be assigned as there are two pages on the display. In addition, 32 partner keys can be assigned.

Programmable functions (excerpt):

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP112

Interfaces

* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient Ethernet” according to IEEE 802.3az
* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ45 interface (modular jack 8P8C) for PC connection
* Connector for external power supply
* 1 x USB 2.0 port for headset connection

Hardware

Power supply

* Power supply: external power unit: 12 V, 6 W
* Power over Ethernet according to 802.3af, Class 1

Storage

* 128 MB DDR3, 16 MB Flash

Environment

* Operating temperature: 0 °C to +45 °C,
* Storage temperature: -10 °C to +70 °C
* Humidity: 10% to 90% (non-condensing)

Display

* Colour display, 320 x 240 pixels (3.5 inch)

Keypad

* Telephone keypad
* 16 function keys
* 32 partner keys

Housing

* Measurements: 21 x 15.5 x 3.5 cm (base)
* Weight: ca. 630 g

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729 (available as software license)
* Opus-NB, Opus-WB
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* IEEE 802.1x (TLS-EAP), 802.1X Proxy
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free and open listening
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

The phones have 12 buttons to the left and right of the display, 8 of which can be programmed at will as function keys. A total of 16 functions can be assigned as there are two pages on the display. In addition, 32 partner keys can be assigned.

Programmable functions (excerpt):

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP232

Interfaces

* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ 45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient Ethernet” according to IEEE 802.3az
* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ 45 interface (modular jack 8P8C) for PC connection
* 4 x USB 2.0 for headsets and extension modules (up to 2 units in series)
* Connector for external power supply

Hardware

Power supply

* Power supply: external power unit: 12V, 6W
* Power over Ethernet according to 802.3af, Class 2

Storage

* 128 MB DDR3, 16 MB Flash

Environment

* Operating temperature: 0 °C to +45 °C,
* Storage temperature: -10 °C to +70 °C
* Humidity: 10% to 90% (non-condensing)

Display

* Colour display, 480 x 272 pixels (4.3 inch)

Keypad

* Telephone keypad
* 16 function keys
* 32 partner keys
* Touchscreen
* 4 direction navigation key

Housing

* Measurements: 21.5 x 15 x 3 cm (basic)
* Weight: ca. 430 g

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729A
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/ MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate / detect DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

The display has 8 function keys that can be programmed at will. A total of 16 functions can be assigned as there are two pages. In addition, 32 partner keys can be assigned.

Programmable functions (excerpt)

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP222

Interfaces

* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation) RJ 45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient Ethernet” according to IEEE 802.3az
* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation) RJ 45 interface (modular jack 8P8C) for PC connection
* 4 x USB 2.0 for headsets and extension modules (up to 2 units in series)
* Connector for external power supply

Hardware

Power supply

* Power supply: external power unit: 12V, 6W
* Power over Ethernet according to 802.3af, Class 2

Storage

* 16 MB SDRAM, 8 MB Flash

Environment

* Operating temperature: 0 °C to +45 °C,
* Storage temperature: -10 °C to +70 °C
* Humidity: 10% to 90% (non-condensing)

Display

* Colour display, 320 x 240 pixels (3.5 inch)

Keys

* Telephone keypad
* 16 function keys
* 32 partner keys
* 4 direction navigation key

Housing

* Measurements: 21.5 x 15 x 3 cm (basic)
* Weight: ca. 430 g

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729A
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate / detect DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

The phones have 12 buttons to the left and right of the display, 8 of which can be programmed at will as function keys. A total of 16 functions can be assigned as there are two pages on the display. In addition, 32 partner keys can be assigned.

Programmable functions (excerpt)

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP240

Interfaces

* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient” according to IEEE 802.3az
* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 interface (modular jack 8P8C) for PC connection
* Headset: RJ-45 interface (DHSG)
* Connector for external power supply
* AUX interface for IP230-X extension module

Hardware

Electricity power supply

* Optional power supply via external power unit: 12 V, 6 W
* Power over Ethernet according to IEEE 802.3af, Class 2

Storage

* 16 MB DRAM, 8 MB Flash

Environment

* Operating temperature: 0°C to +45°C
* Humidity: 10% to 90% (non-condensing)
* Storage temperature: -10 °C to +70 °C

Display

* 128 x 64 pixels - equivalent to - 7 lines with 21 characters

Keys

* Numerical keypad
* alphanumeric keyboard
* 4 direction navigation keys
* 7 freely programmable function keys
* 8 partner keys with 3-colour LED
* 9 control keys, volume control + and -

Extension

* Extension keypad, stand-alone or clip-on
* 30 partner keys with 3 colour LED

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729A
* G.723.1 (5.3)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free and open listening
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate / detect DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search separately or in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

Programmable functions (excerpt)

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP241

Interfaces

* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 interface (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient Ethernet” according to IEEE 802.3az
* 1 x Gigabit Ethernet: 1000-BASE-T (auto negotiation), RJ-45 interface (modular jack 8P8C) for PC connection
* RJ-45 interface (DHSG) for headset
* Connector for external power supply
* AUX interface for IP230-X extension module

Hardware

Electricity power supply

* Optional power supply via external power unit: 12 V, 6 W
* Power over Ethernet according to IEEE 802.3af, Class 2

Storage

* 16 MB DRAM, 8 MB Flash

Environment

* Operating temperature: 0 °C to +45 °C,
* Storage temperature: -10 °C to +70 °C
* Humidity: 10% to 90% (non-condensing)

Display

* 320 x 240 Pixel, High Colour (16 Bit)

Keys

* Numerical keypad,
* alphanumeric keyboard
* 4 direction navigation keys
* 7 function keys
* 8 partner keys with 3-colour LED
* 9 control keys, volume control + and -

Extension:

* Extension keypad, stand-alone or clip-on
* 30 partner keys with 3 colour LED

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729A
* G.723.1 (5.3)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

VoIP protocols

* H.323 Version 5
* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS, SIPS
* RTP, SRTP, RTCP, DTLS-SRTP
* H.460.17 / ICE

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/ MED support
* NTP support including DNS for NTP

Administration

* Password protected access via web browser, HTTPS
* Save and readout configurations
* Automatic update via Update server

Features (excerpt)

* Dialogue guidance in 19 languages, can be extended
* International character set (UTF-8)
* Partner keys showing Presence and call status
* Partner functions to external PBXs via SIP Federation
* Announcement function, for authorised subscribers
* Multiple registration for up to 6 subscribers
* Call Completion on busy(CCBS) and no reply (CCNR)
* Display message waiting
* Hands-free and open listening
* Mute, deactivates microphone for a short time
* Three party conference, also with external participants
* Separate login and logout in call groups
* Pick up general calls or explicit calls directed at other subscribers
* Lock and unlock via PIN
* Call diversion: unconditional, on busy and on no answer
* Park and pick up calls
* Generate / detect DTMF tone
* Call transfer, with/without consultation
* Hold, supported by Music-on-Hold
* Transmit subscriber name along with signalling
* Call waiting, respective signalling of waiting call to caller

Multilevel telephone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search separately or in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

Function key setting

Programmable functions (excerpt)

* Call preparation
* Quick dial
* Call diversion
* Call diversion off
* Lock phone
* Ring tone off
* Ring tone on
* Alternative ring tone
* Call waiting off
* Call waiting once
* Call presentation off
* Call presentation on
* Joint user login
* Joint user logout
* Change user
* Partner
* Pickup list
* Headset off
* Headset on
* Calls (incoming)
* Announcement
* Login
* Logout

## VoIP phone innovaphone IP150

Interfaces

* 1 x Fast-Ethernet: 10/100-BASE-TX (auto negotiation), RJ-45 interfaces (modular jack 8P8C) with “Power over Ethernet” according to IEEE 802.3af and “Energy Efficient” according to IEEE 802.3az
* 1 x Fast-Ethernet: 10/100-BASE-TX (auto negotiation), RJ-45 interfaces (modular jack 8P8C) for PC connection

Hardware

Electricity power supply

* Power over Ethernet according to IEEE 802.3af, Class 1

Storage

* 8 MB DRAM, 2 MB Flash
* Firmware update via web interface

Environment

* Operating temperature: -20°C to 60°C

Display

* 128 x 64 pixels - equivalent to - 7 lines with 21 characters

Keys

* Numerical keypad,
* 4 directional navigation keys
* 5 other control keys

Housing

* Protection type: IP 65 (over IEC60529), explosion-protected zone 2 (gas) and Zone 22 (dust)
* Size: 29,3 x 19,1 x 12,8 cm
* Weight: ca. 2,3 kg

Ring tone volume

* ca. 95 dB(A) at a distance of 1 m

Protocols

* DHCP dynamic host configuration protocol – IP interface settings
* ICMP internet control message protocol - for Ping tests
* SNTP simple network time protocol – to receive date and time
* LDAP lightweight directory access protocol – access for LDAP compatible database
* PPPoE PPP over Ethernet – for direct access to the DSL modem
* PPTP point to point tunnelling protocol – to set up a VPN Tunnel to the company network
* MPPE Microsoft point to point encryption - encryption in PPTP
* NAT network address translation to translate official IP addresses in unofficial addresses
* and vice versa (including SIP and H. 323)
* HTTP configuration with web browser
* SNMP sends Traps to report errors

Voice over IP

H.323

* H.323 Version 5
* H.245 fast connect
* En-block dialling
* Overlapped sending
* RAS Gatekeeper routed signalling (support for external gatekeeper)
* Security (encrypted password authentication according to H.235)

Supplementary Services

* H.450.2 Call transfer
* H.450.3 Call diversion
* H.450.4 Call hold
* H.450.5 Call pick-up
* H.450.6 Call waiting
* H.450.8 Name identification
* H.450.9 Call Completion busy (CCBS) and Call Completion no Reply (CCNR)

SIP

* SIP Version 2, conform RFC 3261
* SIP over UDP, TCP, TLS (SIPS)
* MWI (rfc3842/rfc3265 "Subscription for message-summary")
* DTMF (rfc2833 "RTP payload for DTMF")
* Name Identification (Display String, rfc3325 "Asserted Identity")
* Hold/Retrieve (rfc3264 "Offer/Answer Model for SDP")
* Transfer (rfc3515 "REFER Method", rfc3891 "Replaces Header)
* Coder Change, T.38 (rfc3264 "Re-Negotiation")
* Call Forwarding (PBX internal, "183 Call Is Being Forwarded", "Diversion Header")
* Overlap Dialling (rfc3578)
* Dialogue State monitoring, partner Key (rfc 4235)
* Instant Messaging (rfc 3428)RAS protocol
* Support for external gatekeeper

DTMF

* H.245 “Alphanumeric” or “Signal Type”

Quality of Service

* Prioritise IP packets via TOS and DiffServ
* VLAN Priority according to IEEE 802.1p / 802.1q
* RTP real time protocol for voice data transmission
* SRTP secure real time protocol
* RTCP real time control protocol first level of “Quality of Service”
* DTLS-SRTP

Voice codecs

* G.711 A-law, μ-law
* G.722
* G.729A
* VAD (Voice Activity Detection),
* CNG (Comfort Noise Generation),
* Dynamic Jitter Buffering
* Acoustic echo compensation

Network

* PPPoE protocol, manual/automatic call connection after start
* PPTP up to 4 parallel tunnels, encryption with MPPE
* NAT, H.323-NAT, STUN, TURN
* RSTP, EAP-TLS/ EAP-MD5 according to IEEE 802.1x
* VLAN Priority according to IEEE 802.1p and q
* VLAN-ID according to IEEE 802.1q
* DHCP and LLDP/ MED support
* NTP support including DNS for NTP
* H.460.17 / ICE

Administration

* Password protected access via web browser, HTTPS
* Error search: Log and Trace files
* Interface and connection status display
* Ping – connection test for internet protocol
* Sending SNMP traps
* Save and readout configurations
* Update boot code and firmware via HTML upload
* automatic update via Update server

Features (excerpt)

* Call transfer before / after answering
* Call diversion: unconditional, on busy and on no answer
* Call Hold / Retrieve, supported by music-on-hold
* Call waiting with respective signalling to caller
* Transmit subscriber name along with signalling
* Call Completion, with all usual variations such as callback on busy and callback on no reply
* Conference with three parties, also with external participants
* Number identification for distinctive signalling for numbers or groups of numbers
* Breakthrough call diversion partner function, specific subscribers can reach target subscriber despite activated call forwarding
* Generate / detect DTMF tone
* Password protected configuration
* Lock and unlock via PIN

Ring tones

* general MIDI compatible synthesizer / interpreter (RTTTL)
* 24 sequences can be downloaded
* Selective ring tones for internal, external and special phone numbers in the local phone directory

Call lists

* last 100 outgoing calls,
* last 100 incoming calls,
* combines lists for incoming and outgoing calls
* Information about date, time and success of call
* Multiple registration up to 6 subscribers

Phone directory

* Local: Store up to 500 private entries, only on this phone, max. 5 definable call tones for special users
* Global: automatic telephone directory entries of all PBX subscribers
* External: Integration LDAP compatible database as telephone directory
* Search separately or in all telephone directories, character by character resolution for name entries, name resolution of incoming calls

## Extension module IP2X2-X

Interfaces

* 2 x USB 2.0 for connection to the phone and an additional extension module
* Data and power supply exclusively via USB max. 2 extension modules per telephone
* Power supply is via the phone with POE Class 3 - configure on the switch.
* Alternative gateway power supply unit on phones with extension modules

Housing

* Colour display, 480 x 272 pixels (4.3 inch)
* Measurements: 90 x 150 x 30 mm

Administration

* Firmware comes directly from the phone
* Programmable via the touchscreen or the PBX

Features

* 2 x 16 freely programmable function keys
* 2 x 16 favourites buttons
* Favourite display with Presence and busy status pursuant to Release of the user
* Favourites are synchronized automatically with the PBX
* Different favourite lists can be displayed on the phone and extension module

## myPBX for Android

The unified communications client should be available via an app on Android smartphones. Within the app it should be possible to access on all contacts from the central phone directory of the telephone system as well as from the contacts stored on the smartphone. The call lists of the unified communications clients and the smartphone should be synchronized. It is possible to select for each call whether to call the contact via the smartphone and GSM or via myPBX and WLAN. Pre-settings should ensure automatisms are also available, which always select IP connections if WLAN is available or which prioritise GSM for external calls.

The app user interface should be intuitive to use and the same as the phone user interface and the UC client on the computer. Within the app, it should be possible to create Favourites, obtain detailed call information and set the presence status.

## myPBX (WebRTC)

All end devices on the system should be compatible with browsers that support WebRTC. Transfer should take place using SRTP (secure real-time transport protocol) together with a specifically negotiated direct connection and ensure a secure encryption via DTLS. The implementation should not use a WebRTC gateway.

The unified communications client should also be able to use the WebRTC browser support. In doing so, WebRTC is integrated in the UC client as another phone and can be used as an independent extension. If a call is made via WebRTC, a request should be made as to whether the existing webcam and the microphone may be accessed, subsequently the call should be set up directly through the Web browser. In addition to transferring voice and video files, data (file sharing) and applications (application sharing) should be shared. The unified communications client is thus platform independent and can be used on MacOS, Linux and other systems.

## innovaphone software phone

The Software phone should turn the computer into a telephone. Convenient telephone calls should be possible over connected audio devices. Preferably, headsets that are operated over USB or audio connectors should be used. The Unified Communications client myPBX should be used as the user interface for controlling calls.

## myPBX mobile

It shall be possible to integrate mobile devices in the innovaphone PBX. This shall be possible thanks to a combination of innovaphone mobility and the innovaphone UC client myPBX. The solution should be ready for use as a Web application on all browsers and operating systems without installation.

Calls should go to the desk phone and the mobile device at the same time (one-number concept). No matter where the calls are picked up, they should be synchronised on the desk telephone call logs, as well as in the myPBX mobile call list and that of the unified communications client on the PC. It shall be possible to set Presence on the mobile device. It shall be possible to access central directories and personal contacts from Microsoft Outlook from the mobile device. Favourites shall be accessible, and it shall be possible to see their Presence status. Outbound calls from myPBX shall be switched automatically by the innovaphone PBX and sent to the remote device as a call from the PBX. The mobile number should be hidden.

# Analogue adapter

Analogue adapters to connect analogue end devices such as fax, analogue telephones, analogue DECT systems or door intercoms with door openers. The adapter is compatible with SIP and H.323 and is easy to integrate in the PBX environment. It should also be possible to integrate it in a 19” rack system.

## Analogue adapter innovaphone IP29

Interfaces

* 1 x Fast-Ethernet: 10/100-BASE-TX (auto negotiation), RJ-45 interface (modular jack 8P8C), “Power over Ethernet” according to IEEE 802.3af (Class 3), Energy Efficient according to IEEE 802.3az
* 8 x a/b interfaces (FXS): for connection to analogue end devices, RJ-11 (modular Jack 6P2C)

Hardware

Power supply

* Power over Ethernet according to IEEE 802.3af, Class 3

CPU

* RISC CPU for processing protocols
* Digital Signal Processor (DSP) for voice processing on all channels
* 400 MHz CPU

Storage

* 256 MB RAM, 32 MB Flash

Environment

* Operating temperature: 0°C to 45°C
* Storage temperature: -10 °C to 70 °C
* Humidity: 10% to 90% (non-condensing)

Voice over IP

H.323

* H.323 version 5
* H.323 over UDP, TCP, TLS
* RTP (real time protocol for voice data transmission)
* RTCP (real time control protocol, first level of “Quality of Service”)
* RAS (Protocol Support for external gatekeeper)
* H.245 fast connect
* En-block dialling / Overlapped sending
* H.245 “Alphanumeric” or “Signal Type”
* Encrypted password authentication according to H.235

Supplementary Services

* H.450.1 Call forward / call diversion
* H.450.2 Call transfer, with/without consultation, before/after answering
* H.450.3 Call diversion unconditional, on busy and according to pre-set time delay
* H.450.4 Hold / toggle
* H.450.5 Call pick-up
* H.450.6 Call waiting, with respective signalling to caller
* H.450.7 Message Waiting indication
* H.450.8 Name identification
* H.450.9 Call back on busy (CCBS) call back on no reply (CCNR)

SIP

* MWI (rfc3842/rfc3265 "Subscription for message-summary")
* DTMF (rfc2833 "RTP payload for DTMF")
* Name Identification (Display String, rfc3325 "Asserted Identity")
* Hold/Retrieve (rfc3264 "Offer/Answer Model for SDP")
* Transfer (rfc3515 "REFER Method", rfc3891 "Replaces Header)
* Coder Change, T.38 (rfc3264 "Re-Negotiation")
* Call Forwarding (PBX internal, "183 Call Is Being Forwarded", "Diversion Header")
* Overlap Dialling (rfc3578)
* Dialogue State monitoring, partner Key (rfc 4235)
* Instant Messaging (rfc 3428)RAS protocol
* Support for external gatekeeper

Fax over IP

* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711

Quality of Service

* Prioritise IP packets via TOS and DiffServ
* VLAN Priority according to IEEE 802.1p / 802.1q

Voice codecs

* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation
* Opus-NB

Administration

Access

* over web browser with HTML
* Password protected with secure authentication
* Kerberos server

Error search

* Log and Trace files
* Interface and connection status display
* Ping – connection test for internet protocol
* Sending SNMP traps V1

Update

* Save and readout configurations
* Update boot code and firmware via HTML upload
* Automatic update via update server

Features

* Call transfer: Call Transfer, in all usual variations: with/without consultation, before/after answering etc.
* Hold: Call Hold / Retrieve, supported by Music-on-Hold
* Call waiting: Call waiting, with respective signalling to caller
* Number identification for distinctive signalling for numbers or groups of numbers
* Name display: Transmit subscriber name along with signalling
* DTMF: Generate / detect DTMF tone
* Level adjustment: separate volume setting for devices connected externally
* Call diversion: Activate and deactivate, unconditional diversion, diversion on busy, diversion on no answer
* Security: Set, lock and unlock PIN
* Silence in front of the phone: Set and cancel “Do not disturb”, individually for internal or external calls or for all calls
* Pick-Up: Pick up general calls or explicit calls directed at other subscribers
* Park: Park and pickup calls
* Join group: Separate login and logout in call groups

## Analogue adapter innovaphone IP22 and IP24

Interfaces

* 1 x Fast-Ethernet: 10/100-BASE-TX (auto negotiation), RJ-45 interface (modular jack 8P8C), „Power over Ethernet“ according to IEEE 802.3af (Class 3), Energy Efficient according to IEEE 802.3az
* 2 x or 4 x a/b interfaces (FXS): for connection to analogue end devices, RJ-11 (modular Jack 6P2C)

Hardware

Power supply

* Power over Ethernet according to IEEE 802.3af, Class 3
* Mains adapter

CPU

* RISC CPU for processing protocols
* Digital Signal Processor (DSP) for voice processing on all channels
* 125 MHz CPU

Storage

* 30 MB RAM, 16 MB Flash

Environment

* Operating temperature: 0°C to 45°C
* Storage temperature: -10 °C to 70 °C
* Humidity: 10% to 90% (non-condensing)

Voice over IP

H.323

* H.323 version 5
* H.323 over UDP, TCP, TLS
* RTP (real time protocol for voice data transmission)
* RTCP (real time control protocol, first level of “Quality of Service”)
* RAS( Protocol Support for external gatekeeper)
* H.245 fast connect
* En-block dialling / Overlapped sending
* H.245 “Alphanumeric” or “Signal Type”
* Encrypted password authentication according to H.235

Supplementary Services

* H.450.1 Call forward / call diversion
* H.450.2 Call transfer, with/without consultation, before/after answering
* H.450.3 Call diversion unconditional, on busy and according to pre-set time delay
* H.450.4 Hold / toggle
* H.450.5 Call pick-up
* H.450.6 Call waiting, with respective signalling to caller
* H.450.7 Message Waiting indication
* H.450.8 Name identification
* H.450.9 Call back on busy (CCBS) call back on no reply (CCNR)

SIP

* MWI (rfc3842/rfc3265 "Subscription for message-summary")
* DTMF (rfc2833 "RTP payload for DTMF")
* Name Identification (Display String, rfc3325 "Asserted Identity")
* Hold/Retrieve (rfc3264 "Offer/Answer Model for SDP")
* Transfer (rfc3515 "REFER Method", rfc3891 "Replaces Header)
* Coder Change, T.38 (rfc3264 "Re-Negotiation")
* Call Forwarding (PBX internal, "183 Call Is Being Forwarded", "Diversion Header")
* Overlap Dialling (rfc3578)
* Dialogue State monitoring, partner Key (rfc 4235)
* Instant Messaging (rfc 3428)RAS protocol
* Support for external gatekeeper

Fax over IP

* T.38 real time fax (supported rates 9.6k and 14.4k) with possibility to fall back to G.711

Quality of Service

* Prioritise IP packets via TOS and DiffServ
* VLAN Priority according to IEEE 802.1p / 802.1q

Voice codecs

* G.711 A-law / µ-law (64 kbps)
* G.723.1 (5.3)
* G.729A (16 kbps)
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic echo compensation

Administration

Access

* over web browser with HTML
* Password protected with secure authentication
* Kerberos server

Error search

* Log and Trace files
* Interface and connection status display
* Ping – connection test for internet protocol
* Sending SNMP traps V1

Update

* Save and readout configurations
* Update boot code and firmware via HTML upload
* Automatic update via update server

Features

* Call transfer: Call Transfer, in all usual variations: with/without consultation, before/after answering etc.
* Hold: Call Hold / Retrieve, supported by Music-on-Hold
* Call waiting: Call waiting, with respective signalling to caller
* Number identification for distinctive signalling for numbers or groups of numbers
* Name display: Transmit subscriber name along with signalling
* DTMF: Generate / detect DTMF tone
* Level adjustment: separate volume setting for devices connected externally
* Call diversion: Activate and deactivate, unconditional diversion, diversion on busy, diversion on no answer
* Security: Set, lock and unlock PIN
* Silence in front of the phone: Set and cancel “Do not disturb”, individually for internal or external calls or for all calls
* Pick-Up: Pick up general calls or explicit calls directed at other subscribers
* Park: Park and pickup calls
* Join group: Separate login and logout in call groups

# Mobiles solutions

## innovaphone IP DECT gateway & base station IP1202/4, IP1202 and IP1202e

IP-DECT gateway to extend the IP PBX with DECT compatible subscribers. There should be a possibility to set up very complex DECT systems. The multi-cell capability of the base stations means it should be possible to install several devices guaranteeing roaming and automatic handover between these devices. One base station should support up to 4 or up to 8 channels in parallel.

For the IP side, the system should be both H.323 and SIP compatible. Further requirements include echo cancellation and the support of several codecs for voice compression. GAP and CAP compatibility are necessary for DECT.

Interfaces

* RJ-45: Fast-Ethernet 10/100-BASE-TX (auto negotiation), LED screen for activity and 100Mbit Mode, Power over Ethernet possible (802.3af, class 2, max. 5W)
* RJ-45: External power supply
* DECT
  + GAP / CAP compatible
  + Frequency range: 1880-1900 MHz
  + Simultaneous calls:
    - IP1202/4: up to 4
    - IP1202, IP1202e: up to 8
  + Up to 1000 subscribed handsets per cell
  + Roaming and handover
  + On air synchronization
  + 2 × MCX connectors for external antennas (only IP1202e)
  + RF output power with internal antennas: 23 dBm - 28 dBm (EU), 17 dBm – 21,6 dBm (US)

Hardware

Power supply

* Power over Ethernet according to IEEE 803.2af, Class 2
* Mains adapter 21 - 56 V DC, max. 5 W

Operating environment

* Operating temperature: 10°C to 55°C
* Storage temperature: -25 °C to 55 °C
* Humidity: 15% to 90% (non-condensing)

Features

System capacity

* Min. Number of IP base stations = 1
* Max. Number of IP base stations = 1000000
* Max. Number of simultaneous calls per base station = 8
* Max. Number of registered devices = 1000000

Roaming

* Roaming between several IP DECT base stations

Handover

* Seamless handover (full slot) between multiple IP DECT base stations on 8 channels

Routing

* Integrated IP routing with quality of service, bandwidth management, performance monitoring and Remote Access Service (RAS)

Maintenance

* Integrated web server based operation and maintenance concept

Internet

* IP Internet Protocol - basis for TCP and UDP protocols
* DHCP dynamic host configuration protocol – IP interface settings

Voice over IP

H.323

* Version 4 over UDP including H.225, H.235, H.245 and RAS Gatekeeper routed signalling, H.450

SIP

* H.245 fast connect
* En-block dialling
* Overlapped sending

RTP

* RTP, SRTP (secure real time protocol for voice data transmission)
* RTCP (real time control protocol, first level of “Quality of Service”)

Quality of Service

* Prioritise IP packets via TOS and DiffServ
* VLAN Priority according to IEEE 802.1p / 802.1q

Voice codecs

* G.711 A-law / µ-law (64 kbps),
* G.723.1 (5.3 kbps),
* G.729AB
* VAD (Voice Activity Detection)
* CNG (Comfort Noise Generation)
* Dynamic Jitter Buffering
* Acoustic Echo Compensation according to G.168

Administration

Access

* over web browser with HTML
* Password protected with secure authentication

Error search

* Log and Trace files
* Interface and connection status display
* Ping – connection test for internet protocol
* Sending SNMP traps

Update

* Save and readout configurations
* Update boot code and firmware via HTML upload
* Automatic update via update server

## IP-DECT devices

DECT end devices to connect IP DECT base stations. Various types should be available which support the CAP standard as well as all modern DECT end device features. For example an internal phone directory, vibration alarm, redialling, mute and key locking.

Battery

* Talk time up to 12 hours
* Standby time up to 120 hours
* Capacity indicator

Interface

* Headset connection

Display

* 12 characters x 3 lines, toolbar
* If desired with a full graphical display

Telephony

* Toggle
* Consultation
* Display phone number
* Display name
* DTMF tone transmission

Internal phone directory

* 80 entries
* 24 letters for name and 24 numbers for phone number
* Access to LDAP compatible databases (phone directories) over the PBX

Languages

* German
* English
* French
* Italian
* Spanish
* Dutch
* Danish
* Swedish

Additional functions

* 9 different ring tones
* Key lock
* Volume settings
* Redial for up to 10 numbers
* Mute microphone
* Automatic hook off
* Vibration indicator for incoming calls
* Hands-free mode

### IP DECT phone innovaphone IP61

Housing

* Dimensions: 133 x 53 x 24 mm
* Colour: Black
* Display: S/W, 28 x 35 mm, 112 x 115 pixels, backlight
* LED: to display incoming calls and for charging status
* Keys:
  + 3 function keys
  + Accept call, hang-up
  + 4 direction navigation key
  + Numeric characters
  + Volume + and -
  + Ring tone on / off
* Weight: 115 g (incl. battery and clip)

Battery

* Type: 600 mAh (Li-ion)
* Talk time: 16 hrs
* Stand-by time: 180 hrs
* Charging time: < 4 hrs

Interfaces

* Headset interface (standard 2.5 mm)
* DECT:
  + GAP/CAP standard: EN 301 406, TBR22
  + Frequency range: 1880-1900 MHz
  + Modulation: GFSK
  + Antenna: Integral
  + Sensitivity:-93 dBm
  + Output power: 10 mW (EU)

Environment

* Operating temperature: 0 °C to +40 °C,
* Storage temperature: -20 °C to +60 °C
* Protection class: IP40

Features

* Telephony:
  + Toggle
  + Consultation
  + Display phone number
  + Display name
  + DTMF tone transmission
* PBX features: (feature codes)
  + Call diversion
  + Pick-Up (call transfer)
  + Call waiting on / off
  + Do not disturb (generally, internal, external)
  + Call Park / unpark
  + Callback on busy / no answer
  + Join and leave groups
  + 3-party conference using an
  + LDAP-based phone directory
  + CTI support
* Internal phone directory:
  + 250 entries
  + 48 characters for name and
  + 24 characters for number
  + 24 additional characters

Languages

* Danish, German, English
* Finnish, French, Dutch
* Italian, Norwegian, Spanish
* Swedish, Turkish

Additional functions

* Call history list with the last 25 connections
* 14 Ringtones, LED display, vibration alarm
* Auto-reply
* Speaker ringer volume control in 8 levels

Accessories:

* Charger base, charges batteries over USB 2.0
* Belt clip for easy carrying at work
* Headset with boom microphone

### IP DECT phone innovaphone IP63

Housing

* Dimensions: 134 x 53 x 26 mm
* Colour: Black
* Display: Colour, more than 65000 colours, 28 x 35 mm (128 x 160 pixels), backlight
* LED: to display incoming calls and for charging status
* Keys:
  + 3 function keys
  + Accept call, hang-up
  + 4 direction navigation key
  + Numeric characters
  + Volume + and -
  + Ring tone on / off
* Weight: 130 g (incl. battery and clip)

Battery

* Type: Li-Polymer
* Talk time: 20 hrs (13 hrs with Bluetooth)
* Stand-by time: 120 hrs
* Charging time: < 4 hrs

Interfaces

* Headset interface (standard 2.5 mm)
* Bluetooth (2.4 GHz radio spectrum)
* DECT:
  + GAP/CAP standard: EN 301 406, TBR22
  + Frequency range: 1880-1900 MHz
  + Modulation: GFSK
  + Antenna: Integral
  + Sensitivity:-93 dBm
  + Output power: 10 mW (EU)

Environment

* Operating temperature: 0 °C to +40 °C,
* Storage temperature: -20 °C to +60 °C
* Protection class: IP44

Features

* Telephony
  + Toggle
  + Consultation
  + Display phone number
  + Display name
  + DTMF tone transmission
* PBX features: (feature codes)
  + Call diversion
  + Pick-Up (call transfer)
  + Call waiting on / off
  + Do not disturb (generally, internal, external)
  + Call Park / unpark
  + Callback on busy / no answer
  + Join and leave groups
  + 3-party conference using an
  + LDAP-based phone directory
  + CTI support
* Internal phone directory
  + 250 entries
  + 48 characters for name and
  + 24 characters for number
  + 24 characters for mobile number
  + 24 additional characters
  + Adjustable ringtone per contact

Languages

* Danish, German, English
* Finnish, French, Dutch
* Greek, Italian, Norwegian
* Polish, Portuguese, Russian,
* Swedish, Slovakian, Spanish
* Hungarian, Czech and Turkish

Additional functions

* 9 programmable speed dials
* Call history list with the last 25 connections
* 14 Ringtones, LED display, vibration alarm
* Auto-reply
* Speaker
* Ringer volume control in 8 levels
* Mute

Accessories

* Charger base, charges batteries over USB 2.0
* Belt clip for easy carrying at work
* Headset with boom microphone

## WLAN phone innovaphone IP62

WLAN end device with coverage of up to 100m. It should be possible to extend the coverage at will by using additional access points. Handover capability when changing to another access point should also be guaranteed.

Battery

* New battery technology, economical chip sets
* Supports the Spoofing protocol from the IEEE 802.11e wireless multimedia extensions
* 15 hours of talk time (depending on U-APSD WLAN environment and configuration)
* 100 hours of standby without recharging (without enabled screensaver)

Interface

* Headset connection

Display

* TFT colour display
* 3 function keys, 4 direction key
* LED “Message waiting” indication

Technical data

* Supports H.323 and SIP protocols
* WLAN Standards: IEEE 802.11 a / b / g / n
* Quality of Service: WMM-E, IEEE 802.11e
* Security: IEEE 802.11i
* Encryption: WEP 64 / 128, TKIP, AES-CCMP
* Authentication: WPA-PSK, WPA2-PSK, EAP-TLS, EAP-FAST, PEAP-MSCHAPv2
* 4 Profiles / registration

Accessories

* Charger base, charges batteries over USB 2.0
* Belt clip for easy carrying at work
* Headset with boom microphone
* Headset with ear protection

# Unified Communications solution

The telephone system should represent a complete communications solution, which includes various Unified Communications functionalities in addition to IP telephony. The communication solution shall thereby be realized by a Web client and accessed via a Web browser. This ensures that it is possible to access the application no matter where one is - also when away from the office. Administration should take place via a Web browser. The use of additional software is thus not necessary. The application should clearly show the individual Unified Communications modules on one interface and should be intuitive to use.

The individual Unified Communications elements should be managed by licenses. It should be possible to enable the licenses individually depending on the needs of the subscriber.

## innovaphone myPBX (Unified Communications Client)

The system to be built should include a Unified Communications client, which supports different Unified Communications functionalities. In the best case, this should involve a Web client, which is accessible using a Web browser and requires no additional installation.

Traditional telephone functions should be available such as dial, hang up, hold, park, 3-party conference, toggle, connect, set call diversions and pickup. A call can be triggered with just one mouse click. Various end devices (fixed or mobile phone or software phone) are available over the Unified Communications client which can be selected individually as needed.

It should be possible to search for contacts using an input box. Detailed data on the identified contact such as address, E-Mail address, etc. should be displayed. This contact data can be sent via e-mail to another person. It is also possible to create a Favourites list with the most frequently used contacts, these can also be combined in different groups. The Presence information of the contacts should be visible.

The Unified Communications client should provide the possibility of setting the Presence status, as well as of adding a Presence note. Outlook calendar entries and changes to the Presence should be gathered and displayed on an end device. Any change to availability is updated in real time in all applications.

One mouse click should make it possible to start a chat with a subscriber. It should also be possible to invite any number of subscribers to an existing chat. Subscribers will be informed if someone writes or as soon as someone enters of leaves the chat. Application sharing should be possible.

All incoming and outgoing calls are stored in a history list. Special icons indicate whether the call was answered, missed or forwarded. More detailed information is available for each individual call. A call or a chat can be started from the history list. Furthermore, a callback request can be triggered from the Unified Communications client in that Outlook opens an email containing all the necessary information to call back.

The Unified Communications client should have an embeddable video functionality. In the settings, the subscribers can choose between audio-video or pure audio telephony. A corresponding icon indicates whether video telephony is enabled. Before each call is setup or accepted, subscribers can decide whether they wish to make calls with or without video.

Call diversions should be available for three different statuses: Always, busy or on no reply. Any change in the call forwarding settings should be synchronized automatically with the respective device. If a mobile phone is also in use, there should also be a call diversion available that can trigger a call on multiple devices at the same time. This should ensure that the subscriber on the road can also be reached.

Subscribers can determine which of his/her information (E.g. Presence, call details, etc.) is to be made visible for which contact.

It should be possible to either place the Unified Communications client anywhere on the desktop screen or to fix it to the edge of the screen. It should also be active when minimized. In the event that the Unified Communications client is not visible or is in the background, pop-up windows at the edge of the screen should appear to inform the user about incoming calls and chats. The calls can be accepted or rejected directly from there. It should be possible to read Chat messages in the pop up window until the Chat is entered.

The Unified Communications client should be able to link all telephony activities with a CRM or ERP system. For incoming calls, the caller ID should be used to establish a connection in real time to the customer file stored in the system even before the call is accepted. The enables the person being called to immediately see all of the current information stored for this caller. This can be started automatically for every call or alternatively by clicking on the incoming call message. If the contact is not found, a new contact should be created automatically with the correct parameters.

Features

* Supports traditional telephone functions such as dial, hang up, hold, park, 3-party conference, toggle, connect, set call diversions and pickup
* Create a Favourites list
* Enter Presence status
* Enter calendar entries to be shown in the Presence note
* Update Presence in real time
* Restrict the visibility of personal activity for certain subscribers
* Chat with multiple participants
* Application Sharing
* History list that stores all incoming and outgoing calls and contains more detailed information
* Ad-hoc Video telephony solution
* Default settings for audio and audio-video telephony can be selected at will.
* On demand choice between audio and audio-video telephony for every call
* Change call diversion and automatic synchronization with all end devices
* Pop up window informing about incoming calls and chats

## innovaphone Video

The phone system should offer the possibility of an ad-hoc video telephony solution. This should require a Windows computer and a webcam in addition to the telephone system. The Video telephony solution should be an integral part of the telephone system, which requires no server or other hardware or software. Configuration for administrators should be kept at a minimum

As soon as a call is accepted, a Video window automatically pops up on the desktop and displays the other party. It should be possible to change the size of this window as required. One’s own picture can be monitored in an extra window. Furthermore, it should be possible to place this window in all four corners of the video window, to ensure that the calling partner is always visible.

An icon should indicate whether video telephony is enabled. In the pre-settings it is possible to do determine whether calls are made with or without video. However, prior to any communication the user should have the possibility of choosing between making a call with pure audio data transfer or a video call with audio-video data transfer. This is done by simply clicking on the video icon. If the user changes the default settings for one call, the settings automatically return to the default settings once the call has finished.

The video telephony solution should support 3-party video conferences without the use of an MCU (Multipoint Control Unit). The third calling partner should be added to an ongoing phone call, and once connected should also be visible in the video window.

The video telephony solution should support the H.264 standard, so that other companies can be reached using Federation. In addition, it should be available on mobile devices. It should be ensured that the mobile user can be reached as an internal subscriber via the company network or a VPN connection when on the road.

Features

* Ad-hoc Video telephony solution
* No server required, an integrated part of the PBX
* Default settings for audio and audio-video telephony can be selected at will.
* On demand choice between audio and audio-video telephony for every call
* Minimal configuration for administrators
* Video window opens automatically
* Ad-hoc 3-party Video conference without additional equipment
* Third caller appears automatically in the video window
* H. 264 compatibility

## innovaphone Application Sharing

The telephone system should have an application that allows the screen contents to be transferred from one person to another. This should require no installation of plug-ins and no sending of URLs by e-mail. The screen transfer to can be started with just one click. Configuration, dial, and authentication should take place in the background using the telephone connection. Data encryption takes place according to the principle of voice data encryption.

On each screen transfer, it is possible to select which application is to be shared. For example, the entire desktop can be shared. All applications/windows called up by the user are visible for the parties. To restrict visibility for the other party, it should also be possible to share just selected applications. During the screen transfer, the control can be passed to and retrieved from the other party. It should be possible to use application sharing and pass control in three-party conferences.

In order to carry out internal webinars, it shall be possible to share an application within a phone conference with up to 60 subscribers. The subscribers should need to make no extra settings and the shared presentation should automatically appear on the screens after dialling into the conference.

Features

* No server required
* Requires no additional software
* No subscriber configuration required
* Share one or more applications
* Share desktop
* Passing control, also in three conferences
* Improving collaboration within distributed work groups and sites

## innovaphone Office integration

The system to be built should bundle and make available all Presence information from different applications (e.g. MS Office, VoIP phone, Unified Communications client). The information should be updated and shown automatically and in real time.

In addition to the manual input of the Presence status on the VoIP phone or the Unified Communications client, calendar entries (e.g. Microsoft Outlook) that automatically change the status of a subscriber depending on availability, should also be processed. Meetings and dates from the calendar should also be displayed as a Presence note. The period to be taken into account can be configured freely.

Presence should be represented by corresponding icons, which immediately show whether the respective person is available. A selection of Presence status should be available for this (e.g. present, absent, busy, vacation etc.), which, depending on the calendar entry, are automatically displayed or can be manually converted. It should also be possible to add a Presence note manually.

Presence status and note should be shown in the Unified Communications client, in MS Office applications and on the device. If the information changes at one place, it is changed accordingly in the other places at the same time.

Features

* Comprehensive display of Presence information in all MS Office applications and on the IP phone
* Outlook calendar entries are shown automatically as a Presence note
* Quick overview of the colleagues’ and business partners’ Presence status thanks to Presence icons in all applications
* Automatic real time update of Presence information and notes
* Individually configurable Presence notes
* The display period for a Presence note can be configured freely

## innovaphone Fax

In order to send and receive faxes via the computer or the personal mail client, the telephone system should provide a network-wide fax solution. It should be possible to implement Mail-to-fax and Fax-to-mail in both directions. Compatibility with analogue fax machines must be ensured and the Fax solution should represent an integrated solution that requires no server or other hardware or software.

It is possible to send a fax to multiple recipients simultaneously. There is no page limit for fax attachments. All Libre Office compatible documents and graphics, such as e.g. doc, jpg or png are to be supported. For outgoing faxes, it is possible to generate a customised cover page with various information (e.g. sender’s address, signature of the sender, company logo, etc.).

The fax solution shall send a confirmation to the sender for successful fax transmission. A message indicating the error reason is required in the case of failed fax transmission. It must be possible to guarantee that faxes can be received at all times, even when the fax machine or computer are not switched on.

In order to be compatible on all lines, Fax transmission shall support the T.38 protocol and G.711.

Features

* Integrated fax solution, no server required
* Mail-to-Fax, Fax-to-Mail
* No additional software or interfaces necessary
* Confirmation of successful fax transmission
* Message indicating the error reason when fax transmission fails
* Easy archiving of digital fax documents
* Fax reception at any time
* Fax transmission and storage on an existing mail system
* No printed documents required
* Support of all Libre Office compatible documents and graphics
* No page limit for fax attachments
* Serial faxes by simultaneously sending to multiple recipients
* Configurable cover pages (for example, logo, signatures etc.)
* Compatibility with all analogue fax machines
* T. 38 with fall back to G. 711

## innovaphone Voicemail

The central unit shall provide an integrated, server-independent, network-wide voicemail solution. Remote Voicemail retrieval should be possible from internal and external phones. Access from an external phone must be protected by a PIN request.

It should be possible to call the caller directly from the voicemail menu. Personalized announcements can be recorded directly over the phone. Any messages should be indicated by text, light or icon. Also an email can be delivered to the recipient, informing the latter by means of a sound file (E.g. wav-file) that there is a voice message. The voicemail menu should be available in major languages.

It should be possible to store announcements and messages without needing external computer servers (e.g. Compact Flash card). The Voicemail should be based on XML, thus making it possible at any time to make changes with the aid of scripts.

Features

* Voice recording: Callers can leave a voice message
* Based on XML scripts, can be expanded flexibly for other applications
* No server required to operate Voicemail
* Data storage on a CF card or external Web server
* Message waiting is indicated by a MWI lamp (or by text/icon) on the telephone, alternatively, an email message (with or without message)
* Message Waiting Indication (MWI) standard based (according to H.450.7), can be used for 3rd party SIP and H.323 telephones
* Voicemail menu (easy to use with any DTMF telephone)
  + Return call
  + Listen to, save, delete, repeat message
  + Jump to next/previous message
  + Record personal announcement e.g. personalized greeting
  + Change PIN
  + Pickup voicemail messages without PIN

## innovaphone Mobility

The system to be built should provide the possibility to integrate mobile phones into the central PBX. The subscribers should be recognized as internal subscribers. The PBX features should also be available to them. It should be possible to establish calls via the mobile phone over the PBX at any time.

All fixed line and mobile telephones assigned to a subscriber are shown as a single extension number: The subscriber can make calls from different devices, whereby the same number is always displayed. At the same time, the subscriber is always available under the same number within and outside of the company. It should be possible to bundle voicemail messages to a central voice box.

Features

* Call forwarding
* Toggle/Hold/Park
* Pick-up call
* Call waiting
* End or Hold active calls
* Accept held calls or accept a further call
* Define own Presence status
* Login and out of groups
* Set or delete own call forwarding
* Search & dial from the phone directory
* Only one mail box for fixed telephone and mobile phone
* Callback request (via HTTP)

## innovaphone Conferencing

The phone system should provide a conference solution that enables telephone or video conferences with up to 60 participants. Several conference calls or video calls can occur in different conference rooms at the same time. The subscriber can be connected to the respective conference room by entering a code.

Announcements guide all those involved through the menu and inform them about all events. Upon entering a conference room, the subscriber types in his name. When a subscriber enters or leaves, the name is shown to the other subscribers. The Conference menu announcements should be available in major languages, yet can be adjusted at any time via a respective script or can be created from scratch again.

There should be the opportunity to call a predetermined group of subscribers using a special number. Subscribers who answer the call, are in the Conference.

In a video conference, the participant who is talking shall be seen in the video window. If another participant speaks, this is automatically detected and the video window adapts.

innovaphone application sharing should be possible during a telephone or video conference.

External participants should be able to take part in the phone or video conference via WebRTC and shall be able to use all the features without installing special software.

Features

* No server required
* Conference call or video conference with up to 60 participants
* Password protected conference possible
* Conferences in different conference rooms
* Conference menu announcements in different languages
* Create and customize the conference menu announcements over the same script

# myPBX Toolbox

A JavaScript library should allow users to use communication features of the innovaphone PBX in their own Web applications.

## Call Me Button

Website visitors should be able to connect easily to their contact persons. This should be realised via a call me button. The call me button should be a WebRTC Widget, which consists of a myPBX Toolbox JavaScript. It will be stored and configured on a Web server. The widget should be loaded and started by the browser in real time. There shall be two versions: as a business card and as a sidebar. Audio and video calls should be possible, and Presence of the subscriber stored to the PBX should be recognisable. It should be possible for internal subscribers to change own visibility.

The website should register on the PBX as a virtual subscriber and immediately set up a connection when the call button is pressed. Registering to the PBX shall be stored via SHA process. The voice connection via WebRTC to the innovaphone PBX shall take place via DTLS and SRTP. External subscribers shall be configured on the PBX in such a way that they can only call certain counterparts. He should get no access to the trunk line.

Business card

* With photo or avatar
* In responsive Web design - number of cards next to each other is determined by the width of the browser
* Any number of business cards on a page
* Video image is created in the business card
* Presence is shown
* Button for call, video call and email

Sidebar

* Slowly appears on the side once the page has loaded
* With photo or avatar
* Multiple contacts can be created
* The first available contact is offered if there are several possible contacts. If he is no longer available, it switches immediately to the next contact.
* If the last contact in the list is no longer available, this is indicated and only email is possible.
* Click to expand the widget and it offers more contact information
* Video image is created in the expanded view of the Widget, you can still see the page content
* Warning when changing pages as otherwise the connection is interrupted

# Software solutions

It should be possible to connect various software products over the following interfaces: XML SOAP-API, Microsoft TAPI and CAPI.

The PBX should also make available Call Details Records in order to integrate them in a tariff solution.

CDR

Call Details Records (CDR) should be compiled for every external call for both analysis and billing. It should be possible to transmit CDRs via various methods:

* based on the SYSLOG protocol (RFC 3164)
* as raw data over TCP/IP protocol
* in http format of the web for setting up HTML pages automatically

For the analysis it should be possible to transmit CDR data to two different destinations at the same time. This should render it possible to collect CDR data redundantly. CDR information should be either transferred once at the end of the call or an update should be transferred whenever the call status changes.

## innovaphone PBX Operator

It should be possible to access the PBX wait queue via a computerized switchboard and forward the calls. It is operated via keyboard or mouse click. A busy lamp field indicates whether a target extension number is available.

It should be possible to accept a call by pressing one key or with a double-click. Once a call has been accepted, the focus automatically jumps to a search field that allows the corresponding target extension number to be established. The operator should be able to see who is free or busy, which Presence has been enabled and whether calls are being forwarded. Calls can be switched either with or without consultation (blind transfer).

For incoming calls, the application should be shown in the front. All incoming and outgoing calls are to be recorded in a history list. It should also be possible to Park calls. These parked calls are to be marked graphically.

If a contact is not available, the operator should have the opportunity to send an email from the application requesting a return call. This email should already contains all of the necessary information for the return call. Outgoing calls can be made by the operator despite incoming calls.

Features

* Multiple location capability
* Support master/slave scenarios
* Call transfer with consultation
* Call transfer without consultation (blind transfer)
* LDAP functionalities: Backward and forward searches
* Can be operated via shortcuts
* Drag & drop
* Call journal; can be filtered according to outgoing or incoming calls
* Waiting Queue Monitoring
* Monitor blind transfer calls When mistakes are made, any calls that are switched wrongly can be recalled
* Park and unpark calls
* Integrated help (inline help) printable
* Send Instant Messages
* Send E-mails
* Live search results: After the search the Busy status is shown live
* Shows active calls of any PBX subscribers
* Shows Presence status
* Shows Presence note
* Shows permanent call diversions
* Set / change Presence status of all subscribers
* Set / change call diversions of all subscribers
* Recording of calls possible (also for 3rd party products)
* Night service extension

## innovaphone Reporting

The system that is to be built should contain an analysis application that provides information about the call and response situation within the telephone system. The data should be available in real time and it should be possible to evaluate, store, export or print it at any time.

All calls that have passed through the telephone system during a certain time, can be listed by entering an object name (e.g. a person) and an evaluation period. All calls on the PBX are listed if a query is made without specifying a particular object name. It should also be possible to generate individual filters in order to be able to perform queries with multiple objects, numbers or even pre-determined groups. It is also possible to evaluate calls with a specific call status (no answer, connected, busy, no connection) or a certain call direction (incoming, outgoing, switched or forwarded calls).

The analysis application should provide dedicated user access and filters, so that multiple clients can access the same application independently of each other and be able to see only the information relevant to them. Encrypted evaluation queries should be possible as well as the option to make the evaluated data anonymous.

Features

* Call queries for individual subscribers or groups
* A list of all calls; the results list can be grouped in any way by date or object
* Reports can be rendered anonymous
* Generating individual filters, especially for frequent queries
* Query by call status (no answer, connected, busy, no connection)
* Question by call direction (incoming, outgoing, switched or forwarded calls)
* Evaluations can be stored at any time via a PDF or XML file
* Reports are updated in real time
* Multi-client capability: multiple clients can use the same analysis application independently
* Both scheduled and manual backup of the Reporting database and web server configuration files are possible via the general Web Server administration

## innovaphone Queue Monitor

An application is required to determine the load on a telephone service system. This application should collect the information in real time and represent it graphically and in a clear form.

A call indicator should show how many calls are currently in the wait queue, how long the oldest call has been waiting and how many callers have abandoned their call before it was answered.

An early-warning and alarm system should make it possible to determine the current wait time, the waiting calls and prematurely abandoned calls. An optical and acoustic alarm indicates these pre-set limits have been reached or exceeded. A counter indicates how often a limit has been exceeded and when the last alarm occurred.

A period counter summates the waiting times, incoming calls and prematurely abandoned calls in individually adjustable periods. The achieved peak and average values should also be displayed.

An agent indicator should display how many employees are assigned to a system and how many of them are logged in or active. The indicator should also show the various employee status (ready, on the phone, incoming call, in the post-call process time).

After a completed call, it should be possible to assign a post-call or wrap-up period, in which no calls are allocated. Once this time has expired, the employee is available again and a new call can be allocated. This post-call processing time can be the same for all employees or it can be adjusted individually. If there is a bottleneck, it should be possible to completely cancel this post-call period.

Features

* Optionally, multiple applications can run on a single computer and monitor different wait queues.
* The application can be displayed in two different formats - a detailed one and a more compact one. The compact format is reduced to the call indicator, the re-settable time counter, the limit settings and a system setup display. The detailed format shows more time period values.
* The application is intuitive to use and needs no training.
* All counters are logged in the year log files. This makes it possible to make historical analyses and evaluations directly or with the help of external programs such as Excel or access.

## innovaphone Voice Recording

The system should have an application that enables the recording of incoming and outgoing calls. It should not matter which end devices are used, i.e. it should be possible to record calls with IP phones, analogue phones, DECT telephones and mobile phones.

It should be possible to record entire calls or just selected parts of a call. The recording can be done automatically or manually from one’s own computer. Even after exiting a call, this should be possible for up to 5 minutes. Recordings take place in stereo mode. The audio files are stored as .wav or .mp3 files and can also be AES encrypted, if required. A security copy should be created automatically.

It should be possible to record calls in several small branches over a central recorder. Recording takes place locally in the branch while the data transfer to the central recorder is time controlled. It should be possible to customise the time of the data transfer.

All call records should be managed and edited by means of an intuitive system. Recordings can be played back, fast-forwarded and reminded, merged, archived, deleted, marked as important and copied to global and private lists. Detailed information of the call e.g. forwarding, wait queue, 3-party conference should be displayed.

It should also be possible to add important information such as date, time, name of the calling parties to the recordings. This is to ensure that recordings can be found quickly with extensive filtering and search capabilities. Each and every editing should be logged to ensure the entire audit trail. An interface to external applications should allow remote control and database queries.

Features

* Call recordings (incoming and outgoing calls), no matter from what source
* Recording in different file formats
* Automatic or manual recording on your own computer
* Recording in stereo mode
* Encrypted call recordings
* Possibility of recording calls via a central computer
* Local real time recording per branch
* Individual configuration of the length of time and timing of data transfer
* Management and editing of the recordings
* Adding of important information
* Extensive filtering and search capabilities
* Log function for full audit trail