V8 Featurelist

System Features – IP302 / IP305 / IP800 / IP2000 / IP6000

For all innovaphone products based on V7 and V8

- Access to exchange via group number
- Access to separate exchange via individual number
- ACD groups
- Active Directory Replication
- Administration via uniform interface
- Alarm and error events
- Alternate route selection
- Announcement before pickup (*)
- Announcements (*)
- Authorisation groups (class of service)
- Automatic break through for call protection for preferred numbers
- Automatic call distribution (ACD)
- Automatic night switching with week clock
- Automatic route selection (ARS)
- Bootcode with minifirmware (not supported with IP3000)
- Branches (virtual central PBX)
- Bunched circuit
- Call back
- Call back also if internal call
- Call detail recording (CDR)
- Calling line identification restriction (CLIR)
- Calling party identification (CLIP)
- Calling to E.164
- Calling via ENUM (H.323 and SIP)

- Call Detail Records (CDR) external and internal
- Circle call with circle call query (#)
- Clock synchronization via IP
- Config templates for PBX User
- Configurable tones
- Configure / change / delete call diversion via web
- Conversation return to operator
- Cross connection via IP
- Cross connection via ISDN
- Different call (external, internal)
- Direct dial (with and without number reassessment)
- Door bell interface with door opener function (#)
- DTMF detection disabling
- DTMF-post dial/direct dial
- Emigration of endpoint
- Enhanced error reporting (SIP, QOS parameter, router error reporting)
- Enhanced H.323 features according to H.450.x
- Extended DTMF Object (feature set via DTMF dial)
- External communication
- Firmware upload in DRAM (not supported with IP3000)
- Floating license model
- Flexible dial plan
- General night switching
- Group call (all, circular)
- Hide Objects from LDAP
- Home office integration
- HTTPS
- Incoming direct call to subscriber or group
- Interactive call acceptance (*)
- Interface for external announcement construction (#)

- Internal communication
- Internal music on hold
- Least cost routing (LCR)
- Message center (H.450.7 or DTMF)
- Message Waiting Indication (MWI)
- Messaging
- Messaging (SIP and H.323)
- Mixed dial method (overlap and en-block dialing)
- MS-chap V1/V2
- Multi conferencing (IP6000, IP800)
- Multi protocol (H.323, SIP)
- Multilevel admin access (detailed multilevel user rights for the PBX configuration)
- Multiple system admin (multiple admin and viewer rights possible for the whole system)
- Multistage login (multi-user)
- Music / announcement in standby from analogue resource (#)
- Music / announcement in standby from HTTP resource (*)
- Network wide features (Master/slave with virtual PBX)
- Only calling to registered subscribers (passive control)
- Overflow of subscribers and cross connection
- Overlapped sending
- PBX directory search (LDAP search via DTMF)
- Phonebook either from internal or from external source (LDAP)
- QSIG
- QSIG inter-working
- QSIG trunking (point to point and point to multipoint)
- Redundancy of the ISDN-lines
- Redundancy of the PBX
- Remote monitoring and remote diagnostics
- Remote software upgrade

- RTP Proxy
- SDP Transit (for video and other codecs)
- Short dial (individual, group, system wide)
- Silent intrusion
- SIP Medial Relay (RTP stream routing via gateway)
- SIP trunking
- SIP via TCP
- SIPS (SIP-Secure) – SIP via TCP/TLS (not supported with IP3000)
- SNMP Traps (alarm events)
- SOAP-API
- SRTP (Voice encryption) for H.323 and SIP (not supported with IP3000)
- Subscriber authorisation
- Standard protocols to the endpoint
- Subscriber groups with progressive call (parallel, circular, sequential)
- Switch between day/night with code digit from subscribers
- System time/date via timeserver (LAN or web)
- TAPI / Multi line TAPI (via several PBX's / locations)
- Telephone number assignment
- Time based subscriber authorisation
- Twin Phones
- US standard (T1 and CAS)
- VoIP-CAPI
- Waiting queue – call busy operators / CFU disable operators
- 802.1x

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New in V8

- Announcing the queue position in a waiting queue
- Callback on busy/free to ISDN
- Customized Visual Appearance (individually designable user interface)
- Dual Forking (OCS Integration)
- Dynamic PBX (more PBXs on one Device) PBX hosting / PBX virtualization
- Kerberos based authentication and Cross Realm authentication
- Mobility (GSM integration) - Call-waiting - hold - transfer - park - group login/logout - mobility on/off - setting presence status
- Off load Dialtone (external gateway)
- Off load Media Relay (external gateway)
- Off load MOH (external gateway)
- Presence
- Recording at the ISDN and analogue interfaces
- Roaming Phone Profiles
- RSTP (Rapid Spanning Tree)
- Send Message to group
- SIP error detection (alarm/event updated)
- SIP Federation
- SIP video multisite support

Features IP Phones – IP110 / IP200 / IP230 / IP240 / IP240-1000

- Adjustable loudness
- Adjustable ringing tones also possible, import RTTTL and midi files
- Alphanumeric phonebook
- Announcement via loudspeaker
- Automatic callback on busy and free
- Automatic redialing
- Automatic redialing to busy bunched circuit
- Blind transfer

- Call diversion to internal phone number
- Call hold / retrieve call diversion to external phone number
- Call intrusion
- Call list (outgoing, incoming, combined, not reached, not answered)
- Call park and enquiry from other subscribers
- Call prevention
- Call transfer / pickup (selective and from a group)
- Call waiting (acoustical and optical signaling of waiting calls)
- Caller identification (name/number)
- Changeover of the assigned partners
- Conference (3 way)
- Configuration of the voice recording via HTTP
- Connect to busy and free subscribers without notification
- Counter and list of missed calls
- Date and time display (automatic summer/winter time adjustment)
- Delete call back
- Delete variable call diversion
- Dialing an IP address (with and without RAS)
- Dialing with code key from an unauthorized phone
- Direct dialing (immediate, internal/external)
- Display duration of call
- Display calling number
- Emigration of subscriber/endpoint
- Fixed call diversion (unconditional, no answer (time adjustable) on busy)
- Forced announcement (priority call)
- Function keys – multitude possible – exact list can be found in the technical specification of each IP phone
- Hands free
- Headset
- Individual configuration

- Individual phonebook
- Intercom with automatic call pick up
- Interlinked and limitless multiple conference
- Manager/assistant feature
- Message waiting indication (MWI)
- Messaging
- Multi language
- Multi-protocol and multi-registration (SIP and H.323 concurrent)
- Multiple registrations with selective de-registration
- Mute (switch off the microphone)
- Name dialing
- Open listening
- Over plugging (connect call to a busy subscriber)
- Phone lock
- Preferred subscribers
- Project code
- Redialing of the last 100 dialed numbers
- Silent Intrusion
- SIP via TCP
- SIPS (Secure SIP via TCP/TLS)
- SRTP (SIP and H.323)
- Status indication of the subscriber (free/busy/call)
- Subscriber with several lines
- Subscriber with several numbers
- Transfer with consultation
- Variable call diversion
- 802.1x

Features: **Analogue phones – IP302 / IP22 / IP24 / IP28**

DECT phones – IP52 / IP54 / IP55 / IP65

ISDN phone – IP800

- Call completion on busy
- Call completion on free
- Call diversion to internal/external number
- Call hold/retrieve
- Call park and enquiry from other subscribers
- Call protection
- Call transfer/pickup (selective and from a group)
- Call waiting (acoustic and optical signal of the waiting call)
- Clip to an analogue phone
- Confirmation tone (positive and negative)
- Connect to a free and busy subscriber without notification
- Delete variable call diversion
- Dial with code key from an unauthorized phone
- Direct dial immediate (internal, external)
- Do not disturb (internal and external calls)
- Fixed call diversion; unconditional, no answer (time adjustable), on busy
- Lock phone
- Message Waiting Indication (MWI) on DECT Phones
- Preferred subscriber
- Project code
- SIP via TCP
- SIPS (Secure SIP via TCP/TLS)
- Special tone when active feature
- SRTP (SIP and H.323)
- Transfer with consultation
- Variable call diversion
- 802.1x

Features innovaphone Voice mail (*)

Languages supported: German / English / Italian / French / Dutch / Spanish / Norwegian / Danish / Polish (in preparation) – additional languages on demand

- Email notification (with or without attachment)
- Listen to/delete /store voice mails by means of DTMF input
- MWI (according to H.450.7)
- Personal greeting message
- Voice recording
- Voicemail runs on Compact Flash (CF slot in IP800,IP305,IP302,IP2000,IP6000) or on external Web Server

Features Operator (V8) software based

Localized versions: German / English / Italian / Dutch – additional languages on demand

- Busylampfield (divided into groups)
- Call journal
- Call transfer (with/without consultation (blind))
- Drag&Drop or with keypad use
- Integrated help
- Monitor blind transferred calls
- Monitor of waiting queue
- Name display for incoming calls in LDAP directory
- Park/unpark
- Send messages to PBX users
- Searching within a LDAP directory
- Searching for PBX users in all PBX´s
- Set/change call forward for all PBX users
- Set/change presence status for all PBX users
- Show busy state
- Show call forward
- Show current call
- Show presence status

- State-of-the-art user interface
- Support of Master/Slave scenarios

Features Operator Phone – IP230 / IP240 / IP240-1000

Phone can be used as standalone Operator or in combination with the software based Operator

- Adjustable volume
- Adjustable ringing tones also possible, import RTTTL and midi files
- Alphanumeric phonebook
- Announcement via loudspeaker
- Automatic callback on busy and free
- Automatic redialing
- Automatic redialing to busy bunched circuit
- Blind transfer
- Busy indicator lamp
- Call back (enquiry)
- Call back delete
- Call back on busy subscriber
- Call back to free/busy subscriber
- Call diversion to external phone number
- Call diversion to internal phone number
- Call hold/retrieve
- Call intrusion
- Call list (outgoing, incoming, not reached, not answered)
- Call parking and enquiry from other subscribers
- Call prevention
- Call transfer/pickup (selective and from a group)
- Call waiting (acoustical and optical signaling of waiting calls)
- Changeover day/night with status on phone display
- Changeover of the assigned partner

- Conference (3 way)
- Configuration of the Voice recording via HTTP
- Connect to busy and free subscribers without notification
- Date and time on display (automatic summer/winter time adjustment)
- Delete variable call diversion
- Dialing an IP address (with and without RAS)
- Dialing with code key from an unauthorised phone
- Different ringing tones (internal/external)
- Direct dialing; immediate internal/external
- Display duration of call
- Display calling number
- Emigration of the subscriber/endpoint
- Fixed call diversion; unconditional, no answer (time adjustable) on busy
- Forced announcement (priority call)
- Function keys – multitude possible – exact list can be found in the technical specification of each IP phone
- Hands free
- Headset protocol DHSG
- Headset and handset parallel and autonomous
- Identification of the caller (name/number)
- Individual configuration
- Individual phonebook
- Intercom with automatic call acceptance
- Manager/assistant feature
- Message waiting indication (MWI)
- Messaging
- Multi language
- Multi protocol and multi-registration (SIP and H.323 concurrent)
- Multiple conference interlinked and limitless
- Multiple operators

- Multiple registrations with selective de-registration
- Mute (switch off the microphone)
- Name dialing
- Open listening
- Over plugging (connect call to a busy subscriber)
- Phone lock
- Preferred subscriber
- Project code
- Redialing of the last 100 dialed numbers
- Rush display (0-4)
- Silent Intrusion
- SIP via TCP
- SIPS (Secure SIP via TCP/TLS)
- SRTP (SIP and H.323)
- Status indication of the subscriber (free, busy, call)
- Subscriber with several lines
- Subscriber with several numbers
- Telephone book
- Transfer of the subscriber phone number
- Transfer with consultation
- Variable call diversion
- 802.1x

Features Interactive Voice Response (IVR) (*)

- Announcements depending on the caller (e.g. automatic or manual language selection)
- Call handling according to lines and/or calling number (CLI)
- Listen to the menu also from internal subscribers
- Multi level menu
- Overflow from the operator to an automatic operator
- Voice menu at extension (if no answer, busy, unconditional)
- Voice menu with information and automatic dialing
- Voice menu with predefined destinations
- Voice menu with manual dialing of the destination (DTMF)

New in V8

- Announcing the queue position in a waiting queue

Note:

(*) PC with WIN XP (web server) required or via Compact Flash (CF slot in IP800, IP305, IP302, IP2000, IP6000)

(#) innovaphone IP21 required