

Technical Specification

WLAN phone IP72

Hardware

Dimension (l x b x d):	134 x 60 x 27 mm
Weight:	145g (incl. Battery)
Environmental:	Operating temperature: 0°C to +40°C, Storage temperature: -20°C to +70°C Charging temperature: ≥ +5°C
Display:	128 x 64 Pixel - 7 lines with 21 characters illuminated by white LEDs
Keyboards:	numeric, 4 direction navigation, 3 programmable function keys Off hook, On hook (off), volume + and -

WLAN

Networks:	IEEE 802.11 b / g
Quality of Service:	WMM, IEEE 802.11 e
Security:	IEEE 802.11 i
Encryption:	WEP 64 / 128 bit, TKIP, AES-CCMP
Authentication:	IEEE 802.11 x, WPA-PSK, WPA2-PSK, LEAP, PEAP-MSCHAPv2, EAP-MD5

Protocols

DHCP	dynamic host configuration protocol - IP interfaces settings
ICMP	Internet control message protocol - for Ping tests
SNTP	simple network time protocol – to receive date and time values
LDAP	lightweight directory access protocol – access for LDAP compatible database
PPPoE	PPP over Ethernet – direct access to DSL modem
PPTP	point-to-point tunneling protocol – to setup VPN tunnels to the corporate 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
HTTP	for configuration, using Web browser
SNMP	sending traps to the management software

Voice over IP

H.323:	H.323 version 5, H.245 fast connect Enblock dialing, Overlapped sending RAS Gatekeeper routed signaling (support for external Gatekeeper) Security (encrypted password authentication H.235)
Supplementary Services:	H.450.1 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.450.7 Message Waiting Indication H.450.8 Name identification H.450.9 Call Completion busy (CCBS) and Call Completion no Reply (CCNR) DTMF: H.245 "Alphanumeric" oder "Signal Type"
SIP:	SIP version 2 (including HTTP digest authentication) conform RFC 3261 SIP over UDP, TCP, TLS (SIPS, V 7.0 or higher)
SIP services:	MWI (rfc3842/rfc3265 "Subscription für 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 Dialing (rfc3578) Dialogue State monitoring, partner Key (rfc 4235) Instant Messaging (rfc 3428)
supported SIP-RFCs:	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 (obsolete) 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 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 Dialing

RFC 3420 Internet Media Type message/sipfrag
 RFC 4235 Dialog State monitoring (partner Key)
 RFC 3428 Extension for Instant Messaging

Quality of Service: prioritisation of IP packets via TOS and DiffServ
 VLAN priority (IEEE 802.1p / 802.1q)

RTP real time protocol – for speech transport
 SRTP – secure speech transport (V 7.0 or higher)
 RTCP real time control protocol

Voice Codecs: G.711 A-law / μ -law (64 kbps),
 G.723.1 (5.3 und 6.3 kbps),
 G.729A (16 kbps)
 VAD (Voice Activity Detection),
 CNG (Comfort Noise Generation),
 Dynamic Jitter Buffering

Echo Compensation: G.168

Administration

Access: via Web browser with HTML
 Password protected with secure authentication

Bug tracking: log and trace files
 Display of interface and connection status
 Ping – connection test for Internet Protocol
 sending SNMP Traps

Update: save and reload config files
 boot code and firmware update via HTML-Upload
 automatic update via update server

Supplementary services

Call Transfer with and without consultation

Call Diversion / Redirection

Call Hold / Retrieve with music on hold

Call waiting acoustic and optical signaling of waiting calls

Message Waiting displays a waiting message on the phone

Partner Pickup displays on the phone the possibility for pickup

Pickup-List displays on the phone the list on pickup able calls

Name Display additional name signaling on the phone

Call Completion on busy and no reply

3party conference conference with 3 subscribers, inclusive external subscriber

Calling line identification for different ring tones depending an the call (external, VIP, ...)

Direct speech partner function, the remote phone device accepts the call automatically in loudspeaker mode

with short signal tone before conference start
remote microphone in mute
needs IP 200 phone

Call Diversion break through partner function, authorized subscriber can call an other subscriber with active call diversion

Busy lamp field partner function, shows the actual status of the partner phone
related key supports pickup when ringing and speed dial when free
needs IP 200 phone

partner key: partner status display „free“, „Ringing“ or“ busy“
speed dial in status „free“
pickup in status „Ringing“

DTMF DTMF tone detection & ringing

Security: password-protected configuration
PIN protected phone access

Ring tones: general MIDI compatible synthesizer / interpreter (RTTTL)
24 sequences downloadable
selective ring tones for internal, external and special phone connections specified in phone book

Call lists: last 100 outgoing calls,
last 100 incoming calls,
combined list for incoming and outgoing calls
information about date, time and success of the call

Multiple registrations: up to 6 subscribers can be registered to one phone
Multiple registrations of the same subscriber on different phones supported (supported by innovaphone PBX)

Free speech: free speech in on hook mode

Mute: switch off the microphone

Phone book

internal: for private entities, only for this phone
notes and special ring tone for all phonebook entries

global: automatic access to all PBX subscribers

external: import from LDAP compatible database

Usability: search separate or global in all phone books
Character wise resolution during name entering
Name resolution of incoming phone numbers

Function keys

Number of keys:	3 – free programmable
Number of states:	4 states for every function key
Programmable function:	indirect dialing Quick dialing Call diversion Call diversion off Lock phone Ringing off Ringing on Alternative ringing Call waiting off Call waiting once Calling line presentation off Calling line presentation on Register co-user Unregister co-user Switch user Partner Pickup list Headset off Headset on calls (inbound) announcement Register Unregister