innovaphone
Pure IP-Telephony

Technical Specification

IP phone IP230

Interfaces

Ethernet**:**	2 x RJ 45 interface 10/100-BASE-TX (auto negotiation) with internal 2 port switch and Power over LAN, Class 1 (IEEE 802.3af)
Headset:	RJ-45 interface (DHSG)

Hardware

Power supply:	Power over LAN, IEEE 802.3af Class 1 (with extension module Class 2) or Mains adapter (primary: 110-240 V, 50 Hz, 45mA, secondary: 12V DC 800mA)
Memory:	8 MB DRAM, 4 MB Flash
Operation environment:	Operation temperature: 0°C to +45°C, Humidity: 10% to 90% non-condensing Storage temperature: -10°C to +70°C
Display:	128 x 64 Pixel - or - 7 lines with 21 characters
Keyboards:	numeric, 4 arrow keys, 7 free programmable function keys 10 partner keys, 3 color LED 9 special control keys, volume control + and –
Add-on:	Key block 30 partner keys, 3 color LED

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

ts_IP230_en.doc, Issue: August 2008

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) Dialoge 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 Adresses-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 Dialoge State monitoring (partner Key) RFC 3428 Extension for Instant Messaging
Quality of Service:	priorisation 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:	7 – 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

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