NEW GENERATION IP PBX

KAREL IPV50

IPV50 IS A NEW GENERATION IP PBX DESIGNED FOR SMALL OR MIDDLE SIZED BUSINESSES.

THANKS TO THE AUDIO, DATA AND VIDEO MONITORING AND MANAGEMENT FEATURES PLUS VALUE ADDED APPLICATIONS IPV50 OFFERS, IT MEETS THE CONTINUOUSLY CHANGING AND DEVELOPING NEEDS OF THE BUSINESSES

IPV50 PROVIDES A USER FRIENDLY COMMUNICATION INFRASTRUCTURE WITH ITS WIDE CAPACITY OPTIONS AND APPLICATIONS SUCH AS IP COMMUNICATION, UNIFIED COMMUNICATION, REMOTE OFFICE AND MOBILE COMMUNICATIONS

IPV50 CREATES EFFICIENCY, SAVINGS AND A GREAT DIFFERENCE IN WORK PROCESSES OF THE BUSINESSES WITH ITS FLEXIBLE AND INTEGRATED STRUCTURE.

IPV50 ALLOWS USING IP, DIGITAL AND ANALOG EXTENSIONS AND TERMINALS THAT SUPPORT THESE STRUCTURES IN THE SAME SYSTEM WITHOUT THE NEED FOR ADDITIONAL INFRASTRUCTURE. THEREFORE, IT IS POSSIBLE TO ACCESS AND COMMUNICATE WITH PBX EXTENSIONS, FROM ANYWHERE AND AT ANY TIME.

SYSTEM SOFTWARE RUNS ON LINUX OPERATING SYSTEM, PROVIDES OPEN SOURCE SOFTWARE ARCHITECTURE, AND TAPI AND CSTA SUPPORT. THUS, MANY APPLICATIONS ARE PROVIDED AS STANDARD ON THE PBX, AND ADDITIONAL APPLICATIONS MAY BE INTEGRATED TO THE PBX EASILY.

KAREL IPV50 PBX SOLUTION ALLOWS AUDIO-DATA-VIDEO COMMUNICATION THROUGH SINGLE INFRASTRUCTURE.

EXTENSIONS OF KAREL IPV50 PBX CAN CONTINUE TO COMMUNICATE WITH THEIR EXTENSION NUMBERS REGARDLESS OF THE LOCATION.

EXCHANGE TECHNOLOGY

IP COMMUNICATIONS

IPV50 provide advanced communication solutions such as Voice-over-IP, Fax-over-IP, voice recording, conference call, video call, voicemail to e-mail, monitoring presence, instant messaging, remote extension and remote office. Voice messages, call records and system fault notifications can be forwarded to the specified e-mail addresses with the aid of unified voicemail and e-mail applications. It is possible to access such data and voicemails from any location. Smart mobile phones with mobile extension software and VoIP support may be defined as an extension to the system. Thus, IPV50 users may reply incoming calls with these mobile phones, and listen to their voicemail and use all IP services provided by the PBX.

UNIFIED COMMUNICATION APPLICATIONS

Unified communication applications allow employees to communicate with each other, with customers and suppliers easily, efficiently and rapidly and optimize the business processes.

PRESENCE

Presence allows other extensions to see whether the extension is available, busy, online/offline and the special status that is set by the person himself/herself with a text entry. Thanks to the Presence feature, it becomes possible to access the relevant person with the quickest and most suitable method.

INSTANT MESSAGING

Extensions can communicate with each other instantly in written form with instant messaging feature; thus, the communication is performed very quickly.

VIDEO CALL

All IP extensions connected to IPV50 can make video calls between themselves or through external IP lines using videophones, proprietary sets or smart phones.

CONFERENCE

The conference feature is a conversation platform that provides multiple participation option for extensions. 4 users may make 4 different conference calls simultaneously. This platform allows making conference calls with participants from remote locations and thus reduces the costs and time spent for meetings.

IP FAX SUPPORT

T38 Fax-Over-IP support allows high-quality facsimile transmissions over the IP lines and eliminates the requirement of using PSTN lines.

REMOTE OFFICE APPLICATIONS

Multiple IPV50 PBXs can be connected to each other over IP so that company communication networks with higher capacities can be established. Any extension in a branch office can get connected to the exchange in the head office directly over internet with an IP phone, softphone, WiFi phone, IP DECT phone or a VoIP enabled smart phone without the need for additional equipment. Thus, extensions registered in the PBX can call each other with their extension numbers.

IPV50 SUPPORTS UP TO 100 IP EXTENSIONS AND EXTERNAL LINES. TDM CHANNEL CAPACITY IS 56 FOR A SINGLE SYSTEM. CAPACITY EXPANSIONS ARE PERFORMED BY ADDING MODULES AND LICENSES.

IPV50

CENTRAL PHONEBOOK

IPV50 has a common memory, on which the phone numbers are recorded. Common memory is open to all authorized extensions and access to it can be restricted on the basis of authorization levels.

MOBILE COMMUNICATION APPLICATIONS

IPV50 provide mobility in business life. IPV50 provide users the flexibility to continue communication and remain available at any location with Internet connection as an exchange extension by means of a softphone or VoIP enabled smart phone over 3G, WiFi or XDSL connections.

And, there is a single numbering plan covering the entire system. With an IP DECT system that can be integrated to the exchange, it is possible to use the same handset without changing its extension number both in the head office and in any other location containing a base station integrated with the PBX.

New generation VoIP enabled smart phones with mobile extension applications allow making calls as an IP extension of the exchange, answering incoming calls, listening to voicemail and utilizing all exchange services offered to IP extensions.

AUDIO QUALITY AND SECURITY

HD AUDIO QUALITY

IPV50 features advanced audio algorithms. Thus, perfect audio quality far beyond the expectations is obtained with standard terminals with HD audio quality support and Karel HD terminals.

UNINTERRUPTED VoIP COMMUNICATION AND STABLE AUDIO QUALITY

PBX has the capability to detect package losses caused by external conditions and to divert the communication to an alternative route (e.g. PSTN) until satisfactory quality is obtained in order to provide a stable audio quality. This ensures uninterrupted communication and stable audio quality in VoIP applications. Alternative route definitions can be configured according to VoIP for outgoing calls or quality of service or both.

QoS SUPPORT

If the same network infrastructure is used for both audio and data communication, QoS support provided by IPV50 PBX ensures continuity of the audio quality by monitoring the performance on the network and taking necessary precautions.

SUPERIOR SECURITY IN COMMUNICATIONS WITH THE ENCRYPTION SUPPORT

IPV50 provides high level of security in audio communications with the standard SRTP/TLS encryption without the need for VPN installation.

VALUE ADDING UNIFIED SOLUTIONS

IPV50 also provides value added solutions for increasing the efficiency and effectiveness of the business.

AUTO ATTENDANT AND VOICEMAIL APPLICATIONS

IPV50's advanced auto attendant/voicemail structure ensures that incoming calls are distributed to the extension or department called interactively without any intervention by the operator.

Voicemails can be directed to e-mail addresses of the extensions. Thus, extensions can access their voicemail from anywhere with e-mail access.

Incoming calls can be optionally directed to the operator's voicemail box with the caller ID information in specified intervals, i.e. out of working hours.
There are various scenarios available such as Greeting Message, Busy, Missed, Operator Busy, Not Recorded in the System, Closed to Calls from Auto Attendant etc; and also it is possible to prepare different scenarios in addition to the greeting messages on the PBX using specially developed tools.

IPV50 provides auto attendant features to the user without any need for any additional equipment thanks to the special IVR application available in its software. This feature provides basic auto attendant features to the user free of charge, and ensures advantage in the investment costs of the system when there is no need for advanced applications.

CALL RECORDING CAPABILITIES

Calls of a required extension can be automatically recorded thanks to the automatic call record feature integrated to the system without any need for additional equipment. Calls can be sent as an e-mail to the e-mail address of the extension or any other e-mail address. Besides that, extensions can record calls by marking the call they want to record during the call, and receive this record as an e-mail at the end of the call. This system is designed to manage the call records conveniently and to access the records from anywhere with e-mail access, and does not require any additional equipment.

INTEGRATED GATEWAY BOARD

Media-Gateway boards that can be optionally installed on IPV50 serve for converting the voice signals between IP (SIP/H323) and analog/digital lines of the PBX and optimizing the audio quality by using special algorithms. Thus, there is no need to use an external gateway, and you can save the costs for additional devices.

SOLUTIONS THAT PROVIDE COST ADVANTAGE

WEB BASED MANAGEMENT

IPV50 can be managed conveniently from anywhere by using a standard web browser thanks to its web server. Remote management feature provides savings in support and management costs.

Extensions are authorized by user, operator and administrator authorization levels, and connect to the "Personal Assistant" web interface with their passwords. They can change the parameters, personal forwarding settings, voicemail settings and personal phonebooks within their authorization levels.

It is always possible to manage both DDIVR applications and EIVR applications, which are integrated to the system and free of charge, using the web based PBX management interface remotely, and perform all settings and installations from anywhere and at any time. Thus, these modifications are performed more quickly and with less cost.

CALL RECORD SOFTWARE

NetCM and web based WebCM software ensures keeping call records and generating extension specific reports. It is possible to keep the audio traffic of your company under your control and create economic solutions suitable for your requirements.

REMOTE OFFICE SUPPORT AND IP EXTENSION APPLICATIONS

IP phones on a remote branch office become an extension of the central PBX over internet without the need for any additional devices. Calling the branch offices over the extensions provide economy in communication costs.

Making calls over the PBX as an extension from anywhere using devices like softphones, mobile phones with VoIP support etc. decreases the communication costs.

IP TRUNK/OPERATOR APPLICATIONS

With IPV50, it is possible to connect to the operators that provide VoIP service directly over IP Trunk. As the VoIP rates are cheaper than PSTN rates, this allows a great saving in communication costs.

SPECIFIC CALL TIME CONFIGURATIONS

Calls made from any extensions may be restricted. It is possible to specify the total time for external line calls. Line is deactivated when this time is exceeded and directed to an alternative route. This feature allows control on the usage of advantageous rates of GSM operators defined by a minute limit. When the defined total call time is exceeded, the line is forwarded to another selected line.

LEAST COST ROUTING

LCR feature, which is flexible and programmable based on the extension, provide savings in communication costs.

CONFERENCE

Conference feature helps to shorten the meeting times, and decreases transport costs.

GSM TERMINALS

These terminals act as a bridge between the PBX and GSM network; thus ensures that calls made from phones connected to the PBX in the company to the mobile phones are charged with cheaper GSM call rates.

OTHER PERIPHERALS

Net-Console:

This CTI (computer to telephony integration) software integrates two devices that PBX users use in office environment (telephones and computers) and allows control of audio communication over computers.

GT SMS server:

This Windows based server software communicates with Karel GSM Gateway (FCT) products and allows sending SMS from the computers on the network that the PC running this software is connected to, and receiving incoming SMS and operator broadcast messages.

IPV50 also provides value added solutions for increasing the efficiency and effectiveness of the business.

KAREL IPV50 PBX PROVIDES VARIOUS FEATURES TO ALLOW YOU MEET YOUR DAILY COMMUNICATION REQUIREMENTS IN THE EASIEST AND MOST EFFECTIVE WAY.

IPV50 EXTENSIONS CAN AUDIO-VISUALLY COMMUNICATE USING IP VIDEOPHONES.

AESTHETIC EQUIPMENT THAT CAN BE USED WITH IPV50

• Operator/Proprietary Phone Sets

Proprietary sets that are designed for operators and users, and that provide convenience and speed in using the system features:
NT10D, NT30D, ST30, ST26, NT32I, NT42I, NT62I.

• Units for Direct Access to Extension

Phone units that provide convenience and speed in accessing PBX features, and allow monitoring of external line and extension statuses. DSS3L-24, DSS3K-24 unit for direct access to extension for NT30D, NT32I, NT42I and NT62I. DSS25-28 unit for direct access to extension for ST26, ST30.

• Standard Telephones

Fulya, Ladin.

• Caller ID Telephones

TM120

• Wall Type Telephones

Standard TM900 and TM910 caller ID telephones.

• IP Phones

Functional and elegant IP phones that can operate as an extension to the PBX in the central office from anywhere over the network and internet, and compatible with the all PBXs with SIP support:
NT32I, NT42I, NT62I, IP111, IP112, IP116, IP118.

• IP Videophones

IP phones that provide video calls besides the audio communication: NT62I

• IP Softphones

These software based IP phones provide solutions for your various communication requirements such as audio call, video conference, instant messaging and presence at any location with a network or internet connection. YT500.

• USB Handset

Mini handset that helps using the features of YT500 and that can be connected to the computer via USB port: UT101.

• Operator Headsets

Aesthetic and ergonomic operator headsets that allow the using the phone and computer simultaneously: GLA101, GLS201, GLS300, GLA200.

• IP DECT

Flexible and high performance solutions, IP DECT Base stations and hand units that use the secure DECT standard and VoIP technology together for the growing wireless communication requirements of your business.

DE260

DB260


WiFi

TECHNICAL SPECIFICATIONS

INITIAL CAPACITY 0/0

NUMBER OF BOARD SLOTS 7

CAPACITY IN A SINGLE SYSTEM (TDM)
56 digital and analog lines in total (max. 24 digital lines)

IP CAPACITY 100 lines

BASIC CAPACITY EXPANSION OPTIONS
EXP50 (0/8), Analog Extension Module for 8 extensions with Caller ID
EXP50 (0/4), Analog Extension Module for 4 extensions with Caller ID
EXP50 (8/0), Analog External Line Module for 8 lines with Caller ID (FSK and DTMF)
EXP50 (4/0), Analog External Line Module for 4 lines with Caller ID (FSK and DTMF)
EXP50 (0/8U), Digital Extension Module for 8 proprietary digital phones:
EXP50 (1S2/0), ISDN PRI Module for E1-PRI interface (30 channels)
EXP50 (2/4+2U), Hybrid Module for 2 external lines and 4 extensions with CID, 2 proprietary digital phones
EXP50 (2/4), Hybrid Module for 2 external lines and 4 extensions with Caller ID
EXP50 (4T0/0), ISDN BRI Module for 4 ISDN-BRI lines
EXP50 MG 12 12-channels VoIP
EXP50 MG 24 24-channels VoIP

OTHER TECHNICAL SPECIFICATIONS
RISC architecture, POWERPC (80 MIPS)
Embedded real time LINUX operating system
Unblocked system with a switching capacity of 256x256
Capacity increase by remote system with VoIP or adding extensions
SIP proxy server
H323 trunk that can operate simultaneously with SIP
IP phone, softphone, videophone, IP DECT and all SIP compliant terminals
Perfect audio quality far beyond the expectations with terminals with HD audio quality support
G.711 A/U, G.729AB, G.723.1 and iLBC and codec support
T.38 fax-over-IP support*
8-channels automatic greeting
2, 4, 6, 8-channels advanced greeting scenarios
e-mailing voicemails
Web based embedded management and maintenance server
Net-CM call record system support
Net-CM web based call record and reporting system support
CTI applications with standard APIs
(CSTA-XML, CCXML, VoiceXML, TAPI)
Application development for Windows NT/2000/XP and Linux
19" metal chassis with a height of 3U
Installation to 19" cabinet or wall

METAL CABINET

AMBIENT CONDITIONS
0 C° to +40 C°, 10% to 80% uncondensed humidity

DIMENSIONS
36.5 cm (d) x 11 cm (h) x 42 cm (w)

WEIGHT 6 kg

PLASTIC CABINET

AMBIENT CONDITIONS
0 C° to +40 C°, 10% to 80% uncondensed humidity

DIMENSIONS
16.2 cm (d) x 59 cm (h) x 30 cm (w)

WEIGHT 4.5 kg

SERVICES PROVIDED BY THE PBX

PBX SERVICES

- Message box for busy or missed calls
- "I'm not available" message record
- Listening to voicemails
- Leaving a message for extension
- Intervention
- Queuing
- Parking
- Call forwarding
- Extensive call forwarding
- Call hunting
- Group call hunting
- Conference
- Removing from conference
- Call recording
- Automatic redialling

IP extensions registered on the PBX can use all standard SIP services.

PBX SERVICES THAT CAN BE USED BY SIP EXTENSIONS

- Message box for busy or missed calls
- "I'm not available" message record
- Listening to voicemails
- Leaving a message for extension
- Queuing
- Parking
- Extensive call forwarding
- Call hunting
- Group call hunting
- Automatic redialling
- Dropping an external line in use
- External line marked access
- Dropping an external line in use
- Do not disturb
- Executive-Secretary feature
- Night mode switching
- Entering a number to the common memory
- Calling from common memory
- Entering a number to the individual memory
- Calling from individual memory
- Programming
- Authorization transfer
- CLIR
- COLR
- Door opening
- Clock setting
- Date setting
- Changing password
- Phone lock
- Reminder
- Reminding by message
- Wake-up
- Headphone user
- Abusing calls
- Total charging list
- Deleting records from call records
- Extension call record list
- Call record stopping
- Call record counter resetting

VARIOUS INTERFACES

- Analog external line and extension modules
- UPN, PRI, hybrid modules
- IP modules
- SIP external line
- SIP extension
- H.323 external line
- Media Gateway

CONFIGURATION MODULES

- Analog extension board (0/8)
- Analog external line board (8/0)
- Analog external line board (4/0)
- Digital extension board (0/8UPN)
- 2 external / 4 analog extension + 2UPN hybrid boards (2/4+2U)
- ISDN PRI (1S2/0)
- 2 external / 4 analog extension hybrid boards (2/4)
- Analog extension board (0/4)
- ISDN BRI board (4T0/0)
- 12 channels media gateway board
- 24 channels media gateway board
- EVM Voicemail - 8 Channels

FEATURES OF CONFIGURATION MODULES

8 Analog Extension Modules
Usage of long lines up to 11 km
Modem/fax tone detection
Sending and receiving DTMF
12khz / 16khz credit signal generation
Sending FSK or MF CallerID
Automatic line tests
Adjustable volume levels

8 Digital Extension Modules
Line length of 1.3 km
Usage of 2-wire cables
Usage of digital phone sets with various features

Analog External Line Module
Modules with 4 or 8 trunk lines
Polarity reversal detection
Type1 or Type2 CallerID support
DP calling support
Credit signal detection for charging
Adjustable volume levels

PRI Module
E1/T1/J1 long haul interface
E1(CEPT) PCM-30/ISDN-PRI
Full compliance with EURO ISDN, QSIG standards:
ETS 300 403, ITU-T Q931
ETS 300 402, ITU-T Q921
ETS 300 011
ISDN additional services

VoIP Media Gateway Module
Standard SIP protocols
IP extension/trunk support
IETF Session Initiation Protocol (SIP), RFC3261
Realtime Transport Protocol (RTP), RFC 3550
Realtime Transport Control Protocol (RTCP)
Session Description Protocol (SDP) RFC 2327
DTMF/Tone support while using media gateway
RFC 2833, SIP INFO or in-band DTMF detection, generation
Media gateway configuration over network

Audio Codecs and Algorithms
G711 A-law, U-law
G729A/B
G723.1
iLBC
Encryption with SRTP/TLS*
T.38 fax-over-IP*
Advanced dynamic jitter buffer
Packet loss compensation (PLC)
Central collection and querying of QoS statistics
Flexible capacity up to 72 channels with modular DSP boards
Echo prevention, G.168 compliant
Audio activity detection
Noise suppression, RFC3389
Automatic gain control

Voicemail Module
Voicemail service up to 8 channels
Audio recording
E-mailing of audio records
Online audio recording to NFS server

* Please contact our sales department for detailed information about these features.

** Karel's right to make changes on the features specified in this booklet is reserved.

RICH APPLICATIONS OFFERED BY KAREL IP PBX

- Supports VoIP (SIP and H.323), analog and digital interfaces in external lines and IP (SIP, PIP), analog and digital terminals in extensions
- Capability to use PBX features in proprietary IP extensions
- Accessing the extensions from anywhere, whether inside or outside the office, at any time and in the shortest time possible
- Uninterrupted mobile communications with mobile users
- Capability to use work phone from may terminals such as DECT, GSM or softphone on a PC
- Unified communication methods such as using audio, video and e-mail together
- Convenient and fast management over internet with web based PBX management software
- User specific settings such as personal management pages and greeting messages

PHONE SETS

PHONE SETS	NT10D	NT30D	ST30	ST26
GENERAL FEATURES	DIGITAL PHONE SETS	OPERATOR/ PROPRIETARY PHONE SET	OPERATOR/ PROPRIETARY PHONE SET	OPERATOR/ PROPRIETARY PHONE SET
Display Type	2 rows x 20 characters	8 rows x 24 characters 160x100 pixels graphic	8 rows x 24 characters 160x100 pixels 4" graphic display	4 rows x 20 characters LCD
Operator Headset Connectivity	-	-	-	•
Wall Installation	-	•	-	-
Adjustable Display Angle	-	•	•	-
Adjustable Body Angle	-	•	-	-
Headphone Support	-	•	•	•
PHONE FEATURES				
Handsfree	Half Dublex	Full Dublex	Half Dublex	Half Dublex
Bluetooth	-	Optional	Optional	-
Phonebook Capacity	50	120	40	50
Displaying Names of Callers Recorded in the Phonebook	•	•	•	•
Calling by Name from the Phonebook	•	•	•	•
Integration with Net Console CTI Application	•	•	•	•
USER INTERFACE				
Display Illumination	-	•	•	-
Incoming Call Lamp	•	•	-	-
Volume Control Keys	Handsfree, muting outgoing audio, handsfree volume up and down	Handsfree, muting outgoing audio, handsfree volume up and down, headphone	Handsfree, muting outgoing audio, handsfree volume up and down	Handsfree, muting outgoing audio, handsfree volume up and down
Special Function Keys	Voicemail, redial	Park, menu, phonebook, call records, function, voicemail, redial, conference, transfer	Park, phonebook, menu, transfer, flash and redial	Park, phonebook, menu, transfer, flash and redial
Number of Fast Access Keys	8	-	16	16
Number of Soft Keys	4	4	4	4
Navigator Key	-	•	-	-
Message Warning Light	•	•	-	-
Handsfree Warning Light	•	•	•	•
Headphone Light	-	•	-	-
DSS Support	-	Optional (max. 5 DSS)	Optional (max. 1 DSS)	Optional (max. 1 DSS)
DSS	-	24 Keys	28 Keys	28 Keys
Supported DSSs	-	DSS3L-24, DSS3K-24	DSS25-28	DSS25-28
Color Alternatives	Black / White	Black / White	Black	Black

NT10D

NT30D

ST30

ST26

IP PHONES

IP PHONES	IP111	IP112	IP116	IP118	NT32I	NT42I	NT62I
GENERAL FEATURES							
Display Type	3 rows LCD (2 rows x 15 characters + 1 icon row)	132X64 gray toned (4 levels) LCD	320x160 gray toned (4 levels) LCD	4.3" TFT-LCD, 16.7 M Colors	160X100 gray toned (4 levels) LCD	480X272 24 Bit Color touchscreen TFT display	480X272 24 Bit Color touchscreen TFT display
Ethernet Port	2x (LAN/WAN)	2x (LAN/WAN)	2x (LAN/WAN)	2x Gbit (LAN/WAN)	2x (LAN/WAN)	2x (LAN/WAN)	2x (LAN/WAN)
PoE	Optional	Optional	Optional	Optional	Optional	Optional	Optional
Operator Headset Connectivity	•	•	•	•	•	•	•
DSS Module Addition	-	-	•	•	•	•	-
Bluetooth	-	-	-	-	Optional	Optional	Optional
PSTN Output	-	-	-	-	-	-	Optional
USER INTERFACE							
Color Display	-	-	-	•	-	•	•
Touchscreen Display	-	-	-	-	-	•	•
Display Illumination	•	•	•	•	•	•	•
Volume Control Keys	•	•	•	•	•	•	•
Integrated DSS Keys	-	-	10	10	6	-	-
Message Warning Light	•	•	•	•	•	•	•
PHONE FEATURES							
Protocol	SIP v1, SIP v2	SIP v1, SIP v2	SIP v1, SIP v2	SIP v1, SIP v2	SIP v1, SIP v2	SIP v1, SIP v2	SIP v2.0 (RFC3261)
SIP Accounts	2	3	6	6	3	3	10
Handsfree	Full Dublex	Full Dublex	Full Dublex	Full Dublex	Full Dublex	Full Dublex	Full Dublex
Phonebook Capacity	300	300	300	1000	128	128	128
Export/Import Phonebook	•	•	•	•	•	•	•
Black List	•	•	•	•	-	•	-
Installing Additional Ring Tones	•	•	•	•	•	•	•
Installing Logo to the LCD	•	•	•	•	•	•	•
Automatic Update	•	•	•	•	•	•	•
WAN Static IP/DHCP/PPPoE	•	•	•	•	Static IP/DHCP	Static IP/DHCP	Static IP/DHCP
TLS, SIPS, SRTP	•	•	•	•	-	-	-
Web Interface	HTTP/HTTPS	HTTP/HTTPS	HTTP/HTTPS	HTTP/HTTPS	HTTP	HTTP	HTTP
xML Phonebook (Central Phonebook)	-	•	•	•	-	-	-
BLA/BLF, Voicemail, SMS	Voicemail	BLF, Voicemail, SMS	BLA/BLF, Voicemail, SMS	BLA/BLF, Voicemail, SMS	BLF, Voicemail	BLF, Voicemail	BLF, Voicemail
Hotdesking	-	•	•	•	-	-	-
Proprietary Features	-	-	-	-	•	•	•
AUDIO							
HD Audio Quality	•	•	•	•	•	•	•
Codecs	G.722, G.711A, G.711U, G723, G729AB, ILBC	G.722, G.711A, G.711U, G723, G729AB, ILBC	G.722, G.711A, G.711U, G723, G729AB, ILBC	G.722, G.711A, G.711U, G723, G729AB, ILBC	G.722, G.711A, G.711U, G723, G729AB	G.711, G.723, G.726, G729, G722.2, ILBC, G.722	G.711, G.723, G.726, G729, ILBC, G722, SPEEX
QoS	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP	802.1p/Q, TOS/DSCP
VIDEO							
Camera Head	-	-	-	-	-	-	Moveable Camera Head
Video Camera Features	-	-	-	-	-	-	Adjustable 300K pixel CMOS
Analog Video Output	-	-	-	-	-	-	•
Video Codecs	-	-	-	-	-	-	H.263, H.263+, H.264
Color Alternatives	Black	Black	Black	Black	Black / White	Black / White	Black / White


IP111


IP112


IP116

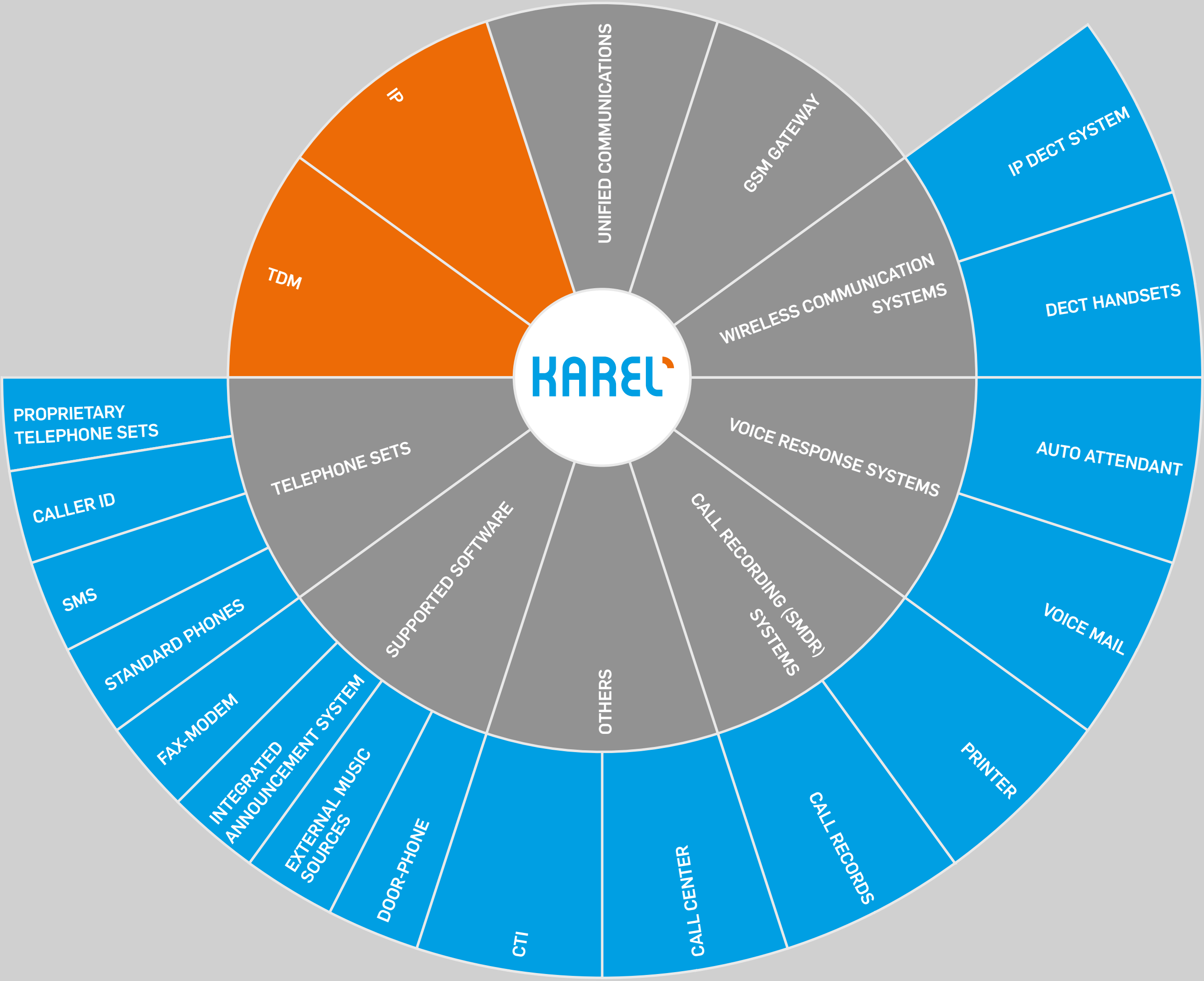

IP118


NT32I


NT42I


NT62I

SOFTPHONE	YT510	YT520	YT530
Voice Call	•	•	•
Video Call	-	•	•
Status Monitoring	•	•	•
GENERAL FEATURES			
Number of Active Accounts	2	10	10
Number of Call Channels	2	3	6
Audio Codecs	G.729, G.711u/a	G.729, G.711u/a, G.722, iLBC, EG711a, EG711u, ISAC, IPCMWB	G.729, G.711u/a, G.722, iLBC, EG711a, EG711u, ISAC, IPCMWB
SIP Compatibility	•	•	•
Audio and Video Prioritization with QoS	•	•	•
Encryption (TLS-SRTP)	-	•	•
USB Plug-and-Play	•	•	•
Internal Phonebook	•	•	•
Microsoft Outlook Integration	•	•	•
Number of Video Conference Channels	0	2	4
Video Codecs	-	H263-1998, H263, LSVX, H264	H263-1998, H263, LSVX, H264
Voice Call Record	•	•	•
Audio Conference Channel Capacity	2	3	6
MINIMUM REQUIREMENTS			
Operating Systems	Windows Vista Business, Windows 7 Enterprise, Windows XP Professional, Windows Server 2008 R2 Enterprise, Windows Server 2003 Enterprise		
Minimum RAM Requirement	512 MB.		
Minimum Harddrive Space	100 MB.		
CD-ROM Drive	Required		
IP Network Connection	Broadband, LAN, Wireless		
Sound Board Specifications	Full-Dublex, 16 bit		



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