LinkPro IP PBX

4 Port IP-PBX + SIP Gateway System

The IPG-40XG is an embedded Voice over IP (VoIP) PBX Server with Session Initiation Protocol (SIP) to provide IP extension phone connections for global virtual office of small-to-medium business (SMB) companys. Equipped with 4 x FXO ports, Ethernet LAN and WAN ports plus Life Line features, IPG-40XG integrates the telephony network and the data network into a manageable converged network to provide an efficient and economical PBX for global long distance voice communications.

IPG-40XG IP PBX works with various IP phones (Desktop, WiFi, Bluetooth, and DECT), VoIP gateways, and analog telephone adapters (ATA) to route calls among client phones, analog phones, and PSTN network. Call features such as conferencing, auto attendant, and voicemail can be seamlessly enabled for all phone devices. In addition, it also provides Internet access to all LAN devices through VPN NAT router.

IPG-40XG IP PBX provides call control and media relay services to SIP clients, and it performs many primary functions, such as SIP Registrar, SIP Outbound Proxy with media relay, SIP Gateways (FXO), SIP PBX for extension calls, Auto Attendant Interactive Voice Response (IVR), and Find-Me Conferencing.

IPG-40XG IP PBX has a built-in suite of PBX applications for supplemental services. This lowers down the total cost of a converged network enabled by IPG-40XG IP PBX than building separated infrastructures for legacy telephony network and data network. In addition, with a web-browsable interface to the data network configuration and voice service provisioning, IPG-40XG brings the manageability of both networks together to facilitate administration locally and/or remotely.

Note that IPG-40XG requires an IP address, a subnet mask, and its gateway Router IP address for its own use to connect to Internet. These three are available from your Internet service provider. IPG-40XG may enable PPPoE or DHCP features to automatically get an assigned dynamic IP from the ITSP. Please refer to Web Configurations for detailed information.

IP PBX

Features :

- SIP Server supports 50 user registrations and 25 concurrent calls
- Support 250 voicemail accounts
- Features Web-Calling
- SIP v1 (RFC2543), v2 (RFC3261) with MD5 authentication (RFC2069 and RFC 2617)
- Supports ITU-T G.711a, G.711u, GSM/MS-GSM, G.729A/B, VAD and CNG for Speech Codec
- Configurations by Web Browser
- Embedded NAT/DHCP Server
- PPPoE/DHCP Client for Dynamic IP plus NAT, VPN, DNS, and DDNS Clients
- Support STUN server and DMZ functions for NAT Traversal
- Support VPN function
- Provides Call Detail Record
- Support Call features; Call Forward/Waiting/Transfer/Hold, and Voice Conference Room
- Number Bonding and Call restrictions.
- Extension Pickup for Attendant
- Bill Rate Table with Voice Mail
- Interactive Voice Recording (IVR) Settings by XML
- Programmable Prompt messages
- On-Line Subscriber Status
- Remote Firmware Upgraded by HTTP Web Interface
- Auto Provision Settings
- Out-Band DTMF (RFC 2833) / In-Band DTMF / Send DTMF SIP Info

PBX Features :

- Support call hold, IP Call broadcasting with feature phones
- Built-in in-line call transfer
- Unconditional, unavailable, busy, call forward
- Per-calling-number forward and rejection
- Group-based call pick-up
- Multi-room meet-me conference
- Auto-attendant
- Voice mail system
- Call privilege grouping
- FXO interface for PSTN
- Inbound/Outbound
- FXO disconnection tone detection
- FXO hunt group
- Caller ID detection
- Echo cancellation
- Support 3 SIP trunk
- In-band/RFC2833/SIP-INFO DTMF translation
- Intra-PBX stackable trucking over Ethernet
- FWD/Vantage account sharing for extensions
- Interoperable with Cisco call manager, CCME; IOS SIP gateway; Unity CUE,79XX, ATA
- Call admission control for wire/wireless phones
- Music on hold
- Direct line

Meet-me Conference:

- Multiple rooms with configurable number and PIN
- Hot key to leave conference

Trunk:

- Auto trunk selection
- Size specific trunk
- Call Barring/Transit Call
- PSTN to remote site
- Direct outward dial

Internal Line:

- Call pickup group/ Group hunting/ Hot line/ Group Ringing

Extension:

- Call Transfer/ Call forward/ forward Me/ Call hold/ Call Park/ Do not disturb/ Call pickup/ Conference call

System:

- Music on hold
- Call detail recording
- Built-in dialer
- External voice mail
- Networking &stackable
- Support private IP network

Voice Mail:

- User PIN
- Multilingual
- Multi-folder archive
- Fast-forward/Rewind/ Undelete
- Personal reception on unavailability/ busy
- Voicemail forward
- Reply call or new call in voicemail menu

Special Support:

- Built-in WEB Call
- Support CBCOM

Support Standards:

RFC 3261, RFC 3311, RFC 3515, RFC 3265, RFC 3892, RFC 3361, RFC 3842, RFC 3389, RFC 3489, RFC 3428, RFC 2327, RFC 2833, RFC 2976, RFC 3263, RFC 3264

Network Management:

- DHCP/PPPoE/Static IP
- LAN IP and net mask specification
- Firewall on predefined services
- DNS and dynamic DNS

NAT:

- Auto NAT discovery and traversal
- RTP proxy
- RTP port range designation

SIP Proxy:

- Proxy server
- Call-based MD5 authentication
- NAT traversal for clients
- Outbound proxy with or without WAN
- Inter-proxy call hand-off

SIP Registration:

- Static/ Dynamic registration
- MD5 authentication
- Authentication proxy to external registrars
- Configuration PBX Caller ID
- User profile
- Handle loose RFC-compliant phones
- Resilient message retry mechanism
- Seeding historical registrations

Relational Provision:

- Logical partition/relation on user and trunks
- Logical provision on outgoing and incoming calling search scopes
- Rich dial-plan expressiveness through route patterns
- Object-oriented provisioning paradigm administration
- Web-based configuration
- CDR
- Extension status display
- Network time protocol time synchronization
- Real time clock setting
- DHCP server
- Configuration time zone
- Firmware upgrade through web interface

Auto Attendant:

- Configuration greeting
- Key to reach operator
- Timeout interval and timeout action
- Music on ringing extensions
- Forward to voice mail on no-answer

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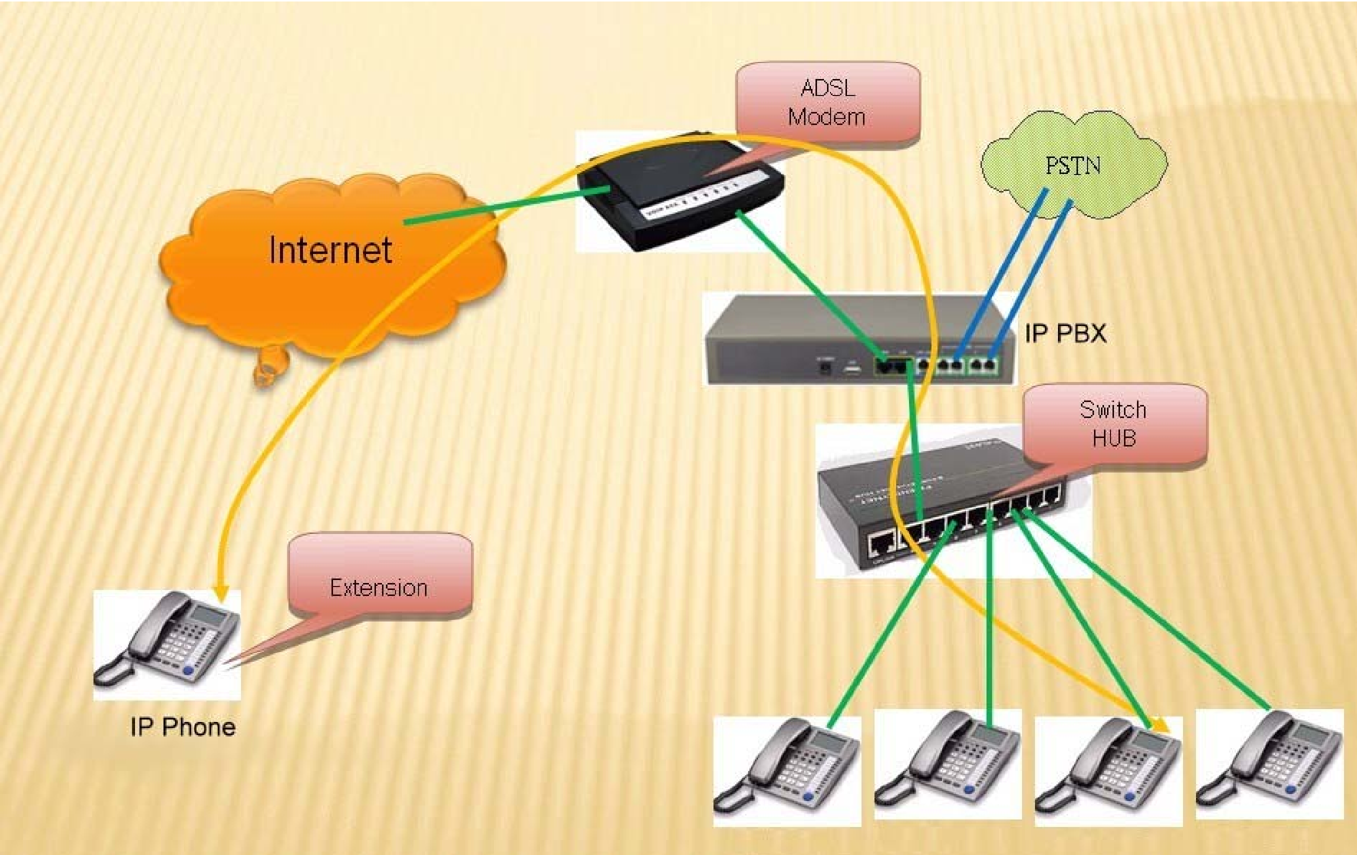
Embedded IP-PBX Features List

Model	IPG-40xG	IPG-80xG	Remarks
Interface	NO Port	4 FXS,4 FXO	
	2 FXS,2 FXO	2 FXS,6 FXO	
	4-FXO	6 FXS,2 FXO	
		8 FXO	
PBX Function			
Auto Attendant	Auto Attendant, IVR, Greeting		
Extension	50	120	SIP Register
SIP Trunk	10	10	Multiple ITSP
Extension DIY management	Y	Y	
Extension group broadcasting	Y	Y	IP Broadcasting
Concurrent Call	25	30	
Conference Room	20 (2* Room)	20	
WebCall	Option		
Support Video	support H.264/H.263/MPEG4		
System billing	Y	Y	
Multiple language prompt	English, Chinese	English, Chinese	
system parameters backup	Y	Y	
system retrieval	Y	Y	
Monitoring on-line users	Extension #/ User name / device type /CPE IP address		
Phone Function			
Caller ID display	DTMF/FSK		
Direct record Prompt voice	Use IP PHONE dial 1605# to record IVR file		
Music on Hold	can change the music file of HOLD		
Call log file	Y	Y	
Call Back	Option		
ring, forward, etc.	call distribution		
Voice mail LED	flash while voice mail is activated		
CALL Transfer	Support blind transfer and attend transfer		
CALL Park	Y	Y	700/701
Call pick-up/ group pick-up	Program group extensions to pick-up		* 8
Multiple off-net route	SIP Trunk, FXO, other IP-PBX		Dail plan setting
extension bunding (3 sets)	Caller ID display and ring		
Follow-me (6 sets)	Y	Y	UMS, one-number for all
Call Forward	Forward All , No Answer Forward , Busy Forward , Offline Forward		
Network Function			
VPN server	Y	Y	
setup wizard page	quick setup for new learner		
CDR log out	can log out for backup, whole system or extension		
QOS	TOS/DSCP		
Internet Link Types	PPPOE , Dynamic/Static IP		
Auto-provisioning (HTTP)	Plug & play : can setup password, number		
Phone Auto Update	Y	Y	
PSTN Life Line	optional for special model		
CBCOM cryption	Prevent SIP-Port block out by carrier		
F/W Update	Support Telnet / WEB		
SIP ENUM	Support SIP ENUM 070		
Web/Telnet interface	Y	Y	
Support QoS, DDNS	Y	Y	
Support DHCP Router	Y	Y	
VLAN,ReInvite	Y	Y	
FXO Auto Busy detection	Y	Y	

IP PBX

Application:



Ordering Information:

IPG-402G : 4 Port IP-PBX + SIP Gateway System, 1WAN+1LAN, (2FXO+2FXS)
IPG-404G : 4 Port IP-PBX + SIP Gateway System, 1WAN+1LAN, (4FXO)

* Product specification subject to change without notice.