Changing the Way the World Communicates

IPNx Analog Gateway – Toll quality voice over IP, Real time fax plus 14:1 compression

Simple VoIP gateway from World Telecom Labs

- ***IPNx Analog Gateway 2, 4 or 8 voice / fax VoIP/SIP connections***
- ***Connects to handsets, pbxs or analog PSTN lines***
- ***Simple to set up and fully remotely manageable***

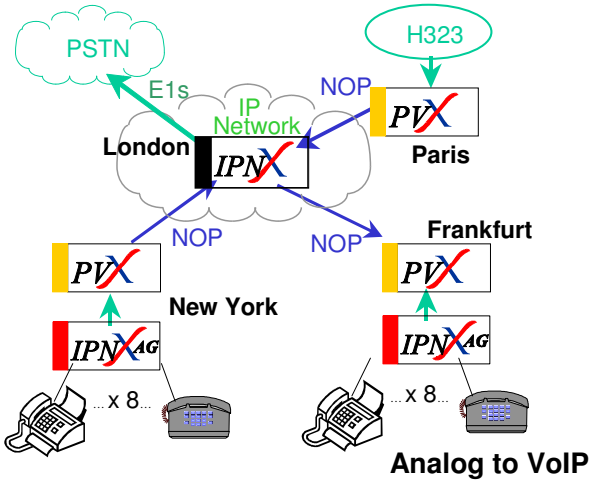
IPNx Analog Gateway used together with the PVx Packet Voice Xcelerator allows high quality, 14:1 compressed VoIP traffic using WTL's patented NOP voice packetisation techniques.

Applications

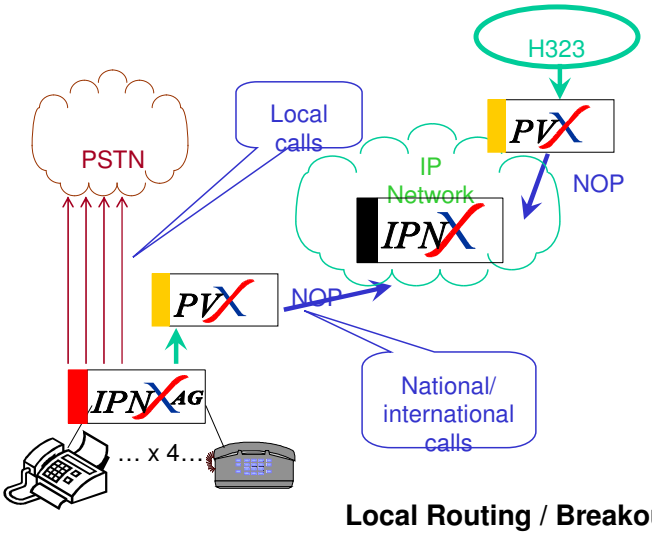
- Ideal for low cost remote PoP (Point of Presence)
- No need for costly voice resources at the edges of the network
- Combine with WTL's WebTL (Call Shop Billing) package to create a cost effective VoIP call shop
- Use of FXO ports means IPNx AG can give low cost analog breakout in remote country where E1 connectivity is not available or too expensive
- Local routing – calls can break out on local PSTN ports as well as route via VoIP.
- PBX interface allows corporate traffic to be collected
- Interoperability – IPNx AG interworks via an IP network with 3rd party SIP gateways

Features

- IPNx Analog Gateway supports 2, 4 or 8 analog ports, fax or phone
- Auto detect and switch over between voice and fax
- Analog ports FXO/FXS
- Remote management by telnet or browser
- Remote software download for field upgrades
- VoIP with G711, G723.1 & G729 plus Fax over IP
- Full DTMF support (detection & generation)
- Echo cancellation (G168), silence suppression and comfort noise generation for excellent voice quality



- True 14:1 compression with WTL PVx Packet Voice Xcelerator
- Comprehensive CDR and statistics generation via PVx
- Breakout to PSTN plus handset support on same unit
- Multiple ring tones supported for different country variants
- Programmable gain control



Local Routing / Breakout

Physical

External wide range AC Power Adaptor: - INPUT: AC100V~240V, 50/60Hz - OUTPUT: DC 12V, 3.0A
Dimensions: W 160 x H 35 x D 242 mm Weight: 0.935kg (unit)
Operating Temp. & Humidity: 0℃~45℃ (32℉~113℉), 10%~85% relative humidity, non-condensing **Storage Temp. & Humidity:** 0℃~55℃ (32℉ ~131℉), 10%~95% relative humidity, non-condensing

Regulatory

CE	UL
FCC Part 15 Class B	FXS/FXO modules compliant with ITU-T G.712

Interfaces

Signalling: Loop Start / DTMF	Telephony ports (RJ-11 2-pin modular jacks) to connect with analog phone (POTS)
WAN port: connect to router, ADSL modem ATU-R), or switch hub directly.	Ethernet port: RJ-45, 10/100 base-T
Ringing Tone: 16.67Hz, 20Hz (default), 25Hz or 50Hz	Reset button (Factory Default)
AC power Jack	Status LEDs:Indicates Ethernet, FXS, and SIP system status

IP Network connection

IPv4 (RFC 791)	DHCP Client (RFC 2131)	ICMP (RFC 792)
MAC Address (IEEE 802.3)	DHCP Server (RFC 2131)	RTP/RTCP (RFC 1889/1890)
PPPoE Client (RFC 2516)	NAT (RFC 1631)	SNTP (RFC 2030)
DNS Client	TCP/UDP (RFC 793/768)	TFTP Client
Telnet Server	HTTP Server	QoS: DiffServ (RFC 2475) / ToS (RFC 791/1349)

IP Telephony (VoIP)

SIPv2 (RFC 3261)	Session Timer (RFC 4028)
Outbound proxy setting for increasing Performance, Productivity, and security	SIP proxy redundancy- Setting Primary and Secondary proxy by IP or URI.
Voice Codecs	G.711(a-Law/u-Law) G.723.1: 6.3k bits G.729A G.729
VAD – Voice Activity Detection with Silence Suppression	CNG – Comfortable Noise Generation
Silence Suppression	Echo Cancelation (G.165/G.168)
Jitter Buffer – Adaptive & Configurable	Packet Loss Compensation - increasing voice quality
DTMF: In-band, Out-of-band (RFC 2833), and SIP-Info	Caller ID Generation – FSK/DTMF/ETSI (Before ringing) ETSI (Between ringing)
FAX transmission: G.711 pass-through T.38 Fax relay	Tone Generation: Ringing Tone, Ring Back Tone, Dial Tone, Programmable Tone

Security

HTTP 1.1 basic/digest authentication for WEB access	MD5 for SIP authentication (RFC 2069/2617)
Password protected Admin access authority	Configure & Update method