INTAGate

IG 500

IP-PBX

- 100 Concurrent Calls, Dual Power Supply, 1,000+ Users
- IVR/Voice Mail/Fax/Conference/Recording/CDR/NAT Transverse
- High Level of Integrated Security Like SBC(SRTP, TLS & HTTPS)
- Seamless Interoperability With PSTN: E1/T1/FXO/FXS/Wireless
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The IG500 is an IP PBX appliance designed to bring enterprise-grade unified communications and security protection to all levels of business-es at an unprecedented price point without any licensing fees, costs-per-feature, or recurring fees. The IG500 enables enterprises to unify multiple communication technologies, such as comprehensive voice, fax, calling, conferencing, video/audio surveillance, data tools, security surveillance, mobility, and facility access management into one commonly managed or accessible network.

With an advanced hardware platform, dual power supply and software functionalities, the IG500 can support up to 500 registered users and offer effortless setup and deployment via the web-browser user interface. Besides auto-discovery of diverse endpoints and auto-provision-ing, the IG500 series offers a set of comprehensive features, including customizable call-routing, multi-level IVRs, call queues, auto-atten-dant, call detail records (CDR), multi-site peering, voicemail/fax forwarding to email and more.

Key Features

- Supports up to 500 users, 80 SIP trunk accounts, up to 80 concurrent calls, 50 conference attendees
- Integrated 2 PSTN trunk FXO ports, 2 analog telephone FXS ports with lifeline capability
- Supports up to a limitless-level IVR (Interactive Voice Response)
- Built-in call recording server; recordings accessed via web user interface
- Supports call queue for efficient call volume manage ment
- Built-in Call Detail Records (CDR) for tracking phone usage by line, date, etc.

- Supports voicemail and fax forwarding to email
- Integrated LDAP and XML phonebooks, flexible dial plan
- Zero configuration provisioning of Mainstream SIP end points
- Highest level of security protection using SRTP, TLS and HTTPS encryption
- Multi-language auto-attendant to efficiently handle incoming calls
- 1.5GHz ARM Quad-core processor, 1GB DDR RAM, 8GB EMMC Flash

Unique Selling Points

Hi-Interoperability with Network

IG500 has the super NAT network adaptability. In the system deployment, the remote SIP extension registered to the IG500 need not any NAT traversal setting.

Excellent Compatibility

Without NAT traversal setting, IG500 could be compliant with other mainstream SIP endpoints or components with changeable IP addresses, which effectively reduces complexity of configuration.

Intelligent Resource Allocation

IG500 optimizes system resource utilization and system efficiency via stochastic algorithm, effectively minimizing hitting over processor resource and improving reliability in any scenarios.

High User-Friendliness

IG500 leverages autoclip intelligent inbound routing mechanism. With call records, IG500 can intelligently match in-bound call number with historic called one in autoclip. Moveable extension, call forwarding, DND, etc are available.

IG 500

Multiple High-Security Modes

Multiple security mechanisms in IG500 are available, including password, ACL, data filtering, etc. Besides, outbound routing, DISA, conference, voice mail and other applications support PIN code setting to customize dynamic firewall.

Flexible Surveillance

IG500 adapts flexible multiple-layer monitoring modes to protect privacy at maximum level and ensure high-level of security and reliability in most conditions.

Technical Specifications

Interfaces

Analog Telephone FXS Ports	2 ports (both with lifeline capability in case of power outage)
PSTN Line FXO Ports	2 ports (UC500)
Network Interfaces	Dual (UC500) 10/100 RJ45 ports
NAT Router	Yes (UC500)
Peripheral Ports	USB, TF
LED Indicators	Power/Ready, Network, PSTN Line, USB, TF
Reset Switch	Yes

Voice/Video Capabilities

Voice-over-Packet Capabilities	LEC with NLP Packetized Voice Protocol Unit, 32~128ms-tail-length carrier grade Line Echo Cancellation, Dynamic Jitter Buffer
Voice and Fax Codecs	G.711 A-law/U-law, G.722, G.723.1 5.3K/6.3K, G.726, G.729A/B, GSM, AAL2-G.726-32; T.38
Video Codecs	H.264, H.263, H263+
QoS	Multiple Layers

Signaling & Control

DTMF Methods	In Audio, RFC2833, and SIP INFO
Provisioning Protocol & Plug-and-Play	TFTP/HTTP/HTTPS, auto-discovery & auto-provisioning of various IP endpoints with no configuration
Network Protocols	TCP/UDP/IP, RTP/RTCP, ICMP, ARP, DNS, DDNS, DHCP, NTP, TFTP, SSH, HTTP/HTTPS, PPPoE, SIP(RFC3261), STUN, SRTP, TLS, LADP
Disconnect Methods	Call Progress Tone, Polarity Reversal, Hook Flash Timing, Loop Current Disconnect, Busy Tone

Security

Media Encryption	SRTP, TLS, HTTPS, TELNET with Fail2ban, Whitelist, Blacklist, alerts and more to protect against attacks

Physical

Dual Power Supply	Output: 12VDC, 2A; Input: 100 ~ 240VAC, 50 ~ 60Hz
Dimensions	UC200: 186mm L x 108mm W x 30mm H
Weight	UC200: Unit weight 0.83kg, Package weight 1.1kg
Environmental	Operating: 32 ~ 113ºF / 0 ~ 45ºC, 8 ~ 90% (non-condensing); Storage: -4 ~ 185ºF / -20 ~ 85ºC
Mounting	Desktop

Additional Features

Multi-Language Support	English/Chinese/Traditional Chinese for Web UI; Customizable IVR/voice prompts for English, Chinese, British English; Customizable language pack to support any other languages
Caller ID	Bellcore/Telcordia, ETSI-FSK, ETSI-DTMF
Polarity Reversal/Wink	Yes, with enable/disable option upon call establishment and termination
Call Center	Multiple configurable call queues, automatic call distribution (ACD) based on agent skills/ availability/ busy level, in-queue announcement
Customizable Auto Attendant	Unlimited layers of IVR (Interactive Voice Response)
Maximum Call Capacity	Up to 50 even in SRTP encrypted
Conference Bridges	Up to 25 simultaneous PSTN or IP participants
Call Features	Call park, call forward, call transfer, DND, ring/hunt group, paging/intercom etc.