



VoIP gateway

IAD100-4S

The IAD100-4S is a multi-functional voice gateway designed to enable seamless connection between analog telephones, fax machines, or analog Private Branch Exchanges (PBX) and IP voice networks. It is equipped with 4 FXS voice interfaces, boasting excellent voice/fax processing capabilities and high stability, and delivers efficient, high-quality IP voice services for operators, enterprises, residential community users and companies. Supporting the standard SIP protocol, the IAD1000-4S has obtained protocol compatibility certifications from Elastix and Broadsoft, the renowned platforms in the industry. It is fully compatible with operators' IMS/NGN softswitch platforms and call center systems. In addition, it can build a communication network for enterprise branches to realize interconnection among branches, providing enterprises with an efficient voice communication solution.

Highlights

- ✓ Multi-functional cost-effective voice gateway with 4 FXS voice interfaces;
- ✓ Supports voice, fax and POS machine services to meet the diverse needs of different users;
- ✓ Complies with SNMP/TR069 standard network management protocols;
- ✓ Dual-stack support for both IPv4 and IPv6 for convenient future network upgrades;
- ✓ Certified by the open-source platform Elastix and compatible with Broadsoft;
- ✓ Supports G.711, G.729, G.723 and other audio codecs;
- ✓ Adopts the standard SIP protocol, achieving full compatibility with IMS/NGN and mainstream softswitch platforms;

Detailed parameters

VoIP Protocols	Protocols: SIP v2.0 (UDP/TCP), RFC3261 SDP, RTP (RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311	Management & Maintenance	SNMP v1/v2/v3
	RTP/RTCP, RFC2198, RFC1889		TR069, TR181
	RFC4028 Session Timer		Automatic upgrade/configuration
	RFC3266 IPv6 in SDP		Web/Telnet management
	RFC2806 TEL URI		Configuration backup/restore
	RFC3581 NAT, rport		HTTP firmware upgrade
	Primary/secondary SIP servers		Call detail records
	External proxy servers		Syslog
	DNS query / A record query /		Ping/Tracert testing
	SIP trunk creation support		Network packet capture
	Early media/early answer		Line diagnosis (GR909)
	Software Functions		Ring group
Web access control rules		IVR voice maintenance	
Telnet access control rules		Cloud-based centralized maintenance	
Action URL		G.711A/U law, G.723.1, G.729A/B	
PPPoE/IPv4/IPv6		Silence suppression	
Number mapping		Comfort Noise Generation (CNG)	
Flexible call routing strategies		Voice Activity Detection (VAD)	
Supplementary Services	Calling and called number transformation	Environmental Specifications	Echo cancellation (G.168), up to 128ms
	Call waiting		Dynamic jitter buffer
	Blind transfer		Flash hook detection
	Consultation transfer		Automatic Gain Control (AGC)
	Group call pickup		T.38/Pass-through
	Call forward on busy		Modem/POS
	Call forward on no answer		DTMF modes: DiffServSignal/RFC2833/INBAND
	Call forward unconditional		Layer3 QoS and DiffServ
	Hot line		Telephone Interfaces:4*FXS, RJ11
	Call hold		Network Interfaces 1LAN, 10/100Mbps, RJ45 1WAN, 10/100Mbps, RJ45
	Do Not Disturb (DND)		Serial Port:None
	Three-way conference		Power Adapter: 100-240VAC, 50-60 Hz @ DC12V 1A
Voice mail	Device Power Consumption: 5W		
Speed dial	Operating Temperature: 0 °C ~ 45 °C		
FXS Specifications	Connection interface: RJ11	Storage Temperature: -20 °C ~ 80 °C	
	Dialing modes: DTMF and pulse dialing	Humidity: 10% - 90% non-condensing	
	Pulse dialing: 10 and 20 PPS	Dimensions (W/D/H): 19513535mm	
	Caller ID: DTMF/FSK standards	Net Weight: 0.6kg	
	Maximum wiring length: 800 meters	Certifications: CE, FCC	
	Supports reverse polarity signaling		
	Complies with multiple dial tone standards		