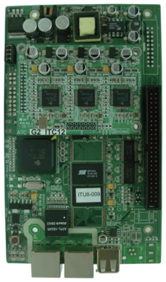
IP Trunk Unit

Model：G1E-ITU

Product Overview

The G1E-ITU is a Trunk Interface Unit designed to be installed in the G1E+ or G1E+M Converged IP-PBX Series.
It enables networking capability between systems over a SIP Based VoIP infrastructure, allowing direct station to station dialing, enhanced call routing and avoidance of costly toll charges.
The G1E- ITU also can be used as a replacement for traditional telephone lines, offering additional features and significant cost savings on monthly telephone bills.



Specifications

Model	VoIP Channel	Applicable System
G1E-ITU4	4	G1E+, G1E+M
G1E-ITU8	8	G1E+, G1E+M
G1E-ITU16	16	G1E+M only

Note: One G1E-ITU per cabinet for both G1E+ & G1E+M
G1E+ supports ITU4 or ITU8 only
G1E+M supports ITU4, ITU8 or ITU16

System Type	VoIP Channel		Cabinet Configuration
	Min.	Max.	
G1E+	4 or 8	8	One cabinet
G1E+M (1)	4, 8 or 16	16	One cabinet
G1E+M (2)	4, 8 or 16	32	Two cabinets
G1E+M (3)	4, 8 or 16	48	Three cabinets
G1E+M (4)	4, 8 or 16	64	Four cabinets

VoIP

- Codec: G.711A, G.711u, G.723, G.729
- Audio quality:
  VAD (Voice Activity Detection)
  CNG (Comfort Noise Generation)
  PLC (Packet Loss Concealment)
  AJB (Adaptive Jitter Buffer)
- Echo cancellation: G.168
- Gain Control: Input / Output: +/- 14db
- DTMF signal: In-Band, SIP-Info, RFC 2833
- DTMF payload: 96/101
- Communications protocol: SIP RFC-3261
- Support the features of the Outbound Proxy Server.
- Support IP address or Domain name setup.
- RTP/RTCP (Real-Time Transport Control Protocol) RFC 3550, RFC 3551, RFC 2198
- Network Management Features
  Web browsing setting: HTTP
  Password-protected
  Support Static, DHCP, and PPPoE IP address assignments
  Software update: TFTP
  Support backup and restoration of the config file.
  Event Log File: TFTP
- QoS (Quality of Service): RTP(TOS/Diffserv), UDP(TOS/Diffserv)
- Phone book: 20 sets of telephone numbers and their respective IP addresses.

＊ Specifications are subject to change without prior notice.

Features

ITU

- 20 sets of Telephone Number to IP Address Conversions for peer-to-peer connections.
- Each VoIP trunk can be configured its primary and a secondary SIP Server respectively.
- Each VoIP trunk can be configured to connect to other VoIP devices in peer-to-peer mode or through the SIP Proxy Server.
- Incoming Trunk Hunt group capability
- Caller ID message: Telephone Number and Name
- Each VoIP trunk can be set individually to different SIP & Proxy servers. (4 sets of servers could be selected)
- Support peer-to-peer VoIP communication under NAT environment.
- ITUs can be integrated together by registering with one another to provide station to station dialing on inter-system connections.
- Auto-detect and auto-adjust fax signal for the T.38 relay standard.

G1E+ / G1E+M System Features

- DID (Direct Inward Dialing)
- MSN (Multi-Subscriber Number)
- LCR (Least Cost Routing)
- All existing standard outside line features in G1E+ and G1E+M are included, such as Incoming Call, Outgoing Call, Day & Night Service, DISA (Direct Inward System Access), Toll Control, Call Accounting (with SMDR), CO Line Ringing, Automatic Transfer, Line Group Assignment, etc.