

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
- Supports the features of an Outbound Proxy Server.
- Supports 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
 - Supports Static, DHCP, and PPPoE IP address assignments
 - Software update: TFTP
 - Supports 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 number to related IP addresses (for peer-to-peer connections).

* Specifications are subject to change without prior notice.

Features

- **ITU**
 - Each VoIP channel can be configured to connect to other VoIP devices in peer-to-peer mode or via a SIP server or service provider.
 - Each VoIP channel can be assigned to use a different SIP server or service provider. There are 4 sets of server details on each ITU card configurable for primary and secondary values of both Proxy server and SIP server.
 - Incoming Trunk Hunt group capability
 - Caller ID message: Telephone Number and Name
 - Supports peer-to-peer VoIP communication under NAT environment.
 - 20 sets of Telephone Number to IP Address Conversions for peer-to-peer connections.
 - 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.