FIBERME E1/T1 VoIP Gateway

FEG4301

FEG4301 is unparalleled cost effective and compliant with PRI ISDN, R2 and SS7 packets, and adopt the equivalent hardware architecture, with dedicated DSP chipsets for processing IP/TDM signaling and optimizing voice quality. Compared with rival products, FEG4301 features high reliability and unmatched cost, and delivers a perfect alternative option for enterprises, operators and system integrators

User-Friendly Management & Toolkits

The Web graphical user interface (WebUI) is a real-time web toolkits to configure and monitor and perform real-time monitoring and maintenance. Flexible SIP and Protocols configuration help to configure SIP, SIP trunking, SIP Mediation, PCM, SS7 and ISDN, Routing and more.

Flexible Scalability

30 Simultaneous SIP sessions with multimedia transcoding provides high performance in a small footprint to help lower ownership cost and operational cost; a broad range of gateway toolkits help gateway's maintenance and software upgrading for Web UI, gateway services and firmware.

Any-to-Any Signaling and Transcoding

Provide any-to-any network connectivity through its ability to interwork multiple protocols to deliver services. It can provide interworking between SS1, R2, ISDN, SS7, SIP formats and any-to-any media transcoding for popular voice codecs, T.38 and G.711 fax interworking, RTP, INBAND and SIPINFO.

Integrated Transcoding for Voice, Tone and Faxing

Eliminates the need to add separate hardware to support transcoding requirements helping to reduce CAPEX and number of platforms deployed, and support a range of Codecs, including G.723, G.729, G.711, iLBC, SIPINFO, RFC2833, RF3261, INBOUND and more

Flexible and efficient Gateway Solution

Provide a clear migration path to an all-IP network. It can scale up to 30 simultaneous IP sessions and at the same time provide media transcoding and impressive sessions per second performance; Support SS7 signaling, call routing, call automated failover from IP to TDM for outbound routing.

	FEG4301
E1/T1 Ports	1
Sessions	30
Routing Features	Call routing and translation (from PCM to IP or reversely)
IP Interfaces	Dual redundant 2 *100 Base-T Ethernet for VoIP payload and signaling
IP protocols	TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN, etc.
Coder support	G.711A,G.711U, G.729
Power Supply	Single
TDM Signaling Protocols	ISDN PRI/MF R2/SS7 ISUP/SS7 MTP1~3/SS7 SIGTRAN/SS7 TCAP
Power Requirements	12V DC
Mounting	Desktop
Compatibility	Interoperable with most IP-PBX and UC Platform, and field-proven by SMB and Carriers Worldwide
Dimensions H*W*D(mm)	30*190*120
Environment	Operating temperature range :0 to +55 °C, 8-90% relative humidity non-condensing Storage temperature range:-20 to +85 °C, 8-90% relative humidity non-condensing
Routing	Call Routing and translation (PCM↔IP)
Safety	Compliant with most international standards, please ask FIBERME or its sales representatives worldwide. FIBERME would comply all new safety standard to for different regions around the world while needed.
EMC/EMI	Compliant with most international standards. For compliance documents, please contact FIBERME's sales representatives.
OAM&P	Network Time Protocol(NTP) Web User interface (WebUI) supports configuration via browser SNMP MIBs
Dedicated DSP-Empowered Capability	Telecom-style DSP algorithm has been optimized for over decades, assuring seamless compliance with any network environment. Plentiful DSP resources are allocated for signaling, media processing, bandwidth optimization, Telco redundancy