DGW-L1 Gateway Datasheet

DGW-L1 Gateway

OpenVox DGW-L1 T1/E1 Gateway is an open source asterisk-based VoIP Gateway solution for operators and call centers. It is a converged media gateway product. This kind of gateway connects traditional telephone system to IP networks and integrates VoIP PBX with the PSTN seamlessly. With friendly GUI, users may easily setup their customized Gateway. Also secondary development can be completed through AMI (Asterisk Management Interface).

There is one port on DGW-L1 supporting 30 channels most. It is developed with a wide selection of codecs and signaling protocol, including G.711A, G.711U, G.729A, G.722, G.723 and GSM. It supports PRI/R2/SS7 protocol. OpenVox T1/E1 Gateway has good processing ability and stability. The DGW-L1 gateways will be 100% compatible with all kind of SIP servers, such as Asterisk, Elastix, trixbox, 3CX, FreeSWITCH and other VoIP operating platforms.

Appearance

Parameter

- Size: DGW-L1 31cm*5cm*17cm
- Weight: DGW-L1 1314g

Feature List

General Info	
Storage temperature	-40~85℃

Operating temperature	0~70℃
Operation humidity	5%~95% non-condensing
Power supply specification	12V/AC
Maximum power	12W
WAN interface (Eth0)	1
LAN interface (Eth1)	1
VGA interface	1
USB 2.0 interface	2
Physical Interfaces and Properties	
Ethernet connectors	RJ45
ISDN interface	E1/T1, BNC (G.703), RJ-48 120Ω
Signal	PRI, SS7, R2
System Features	
Available in 1 port T1/E1/PRI/R2/SS7, energy efficiency concurrent processing, up to 30	
Simple and convenient configuration via Web GUI	
Codecs support: G.711A, G.711U, G.729A, G.723.1, G.722, GSM	
Support protocols：SIP、IAX、TCP、UDP、RTP、SSH、HTTP、HTTPS	
Support NTP time synchronization and client time synchronization	
Support SSH access for background management, Asterisk CLI command operation	
Open API interface (AMI)	
Support ports group management	
Support for custom dialplans	
Firmware update by HTTP	
Support call statistics	
Support auto provision	
Support backup/upload configuration file	
Multiple detailed log output	
Support Chinese language	
Automatically reboot	
Good compatibility, support Asterisk, Elastix, Freeswitch and Small and medium IPPBX platform	
Available for OEM	
3-month “No Question Asked” Return Policy, and Two-year Warranty	
Sample Applications	
Connect legacy PBX systems to low-cost VoIP services	

Connect legacy PBX systems to remote sites over private VoIP links
Connect IP PBX systems to legacy TDM services
Phased transition from legacy PBX to IP PBX
Connect virtualized systems to legacy TDM services
Transcoding by connecting systems using varying codecs
Lync connectivity to SIP or legacy TDM providers and SIP or Legacy PBX
SIP Features
Support add, modify & delete SIP Accounts
SIP registration with Domain
E1/T1 port grouping
Support multiple SIP registrations: Anonymous , Endpoint registers with this gateway, This gateway registers with the endpoint
SIP accounts can be registered to multiple servers
Combine different SIP Trunks into group
SIP(RFC3261) compliance
DTMF: RFC2833, SIP INFO, INBAND
Support T.38 /Pass-through Fax
IAX Features
Support IAX trunk
Support encrypted transmitting
Routing
Flexible routing settings
Support 512 routing
Support caller/callee manipulation and filtering
Trunk group support, Trunk priority management
Support add, modify & delete routing
Network Features
Network type: Static IP and DHCP
IPv4, UDP/TCP, DHCP, TFTP, SCP
HTTP/HTTPS/SSH
Support DDNS
Support ping & traceroute command on the web
Support network capture on the web