OpenVox DGW-100X(R) Series E1T1PRI VoIP Gateway

Open Vox T1/E1 Gateway with open-source Asterisk-based VoIP Gateway solution is an operators and call centers. It is a converged media gateway product. This type of gateway connects traditional phone system to IP networks and integrates VoIP PBX and ISDN seamlessly. With user friendly GUI, user may easily setup their customized Gateway. Also, secondary development can be completed through AMI (Asterisk Management Interface). The DGW-100X(R) digital VoIP gateway helps1/2/4 software-selectable T1/E1/PRI interface and helps up to 30/60/120 concurrent calls. The "R" means that the device helps redundant power supply.

Technical Specifications	
♦ 1/2/4 T1/E1 RJ-48 ♦ 2 10/100/1000M Ethernet ports ♦ 2 USB 2.0 ports ♦ DGW-100XR with redundant power supply ♦ Maximum Power Consumption: 20W	♦ Power supply specification: 100-240V/AC ♦ Operation humidity: 5%~95% non-condensing ♦ Operating temperature: 0°C~70°C ♦ Storage temperature: -40°C~85°C

Features	
◆ Available in 1/2/4 port T1/E1, energy efficiency concurrent processing, upto 120 ◆ Signal ling: PRI/R2/SS7 ◆ Support up to 24 countries' standard R2 signal ling ◆ Support new R2 variant ◆ Simple and convenient configuration via Web GUI ◆ Codecs support: G.711A, G.711U, G.729A, G.723.1, G.722, GSM ◆ SIP(RFC3261) compliance ◆ Independent System for Each Module ◆ Customizable IVR	◆ Affordable Price with Superior Performance ◆ Hot-swap Design for both Sim cards and Modules ◆ Compatible with sip server, such as: Asterisk, Elastix, 3CX, Free SWITCH, etc. ◆ Available for OEM ◆ 3-Month "No Questions Asked" Return Policy ◆ One Year Warranty ◆ PIN identification ◆ Mobile number portability (MNP) ◆ Flexible routing settings & unlimited routing rules

SIP Features	
◆ Support add, modify & delete SIP Accounts ◆ SIP registration with Domain ◆ Support multiple SIP registrations: Anonymous, Endpoint registers with this gateway and 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

Routing	
◆ Flexible routing settings ◆ Support 512 routing ◆ Support caller/callee manipulation and filtering	◆ Trunk group support, Trunk priority management ◆ Support add, modify & delete routing ◆ E1/T1 port grouping ◆ Support Failover

Network Features

♦ Network type: Static IP and DHCP ♦ IPv4, UDP/TCP, DHCP, TFTP, SCP ♦ HTTP/HTTPS/SSH	♦ Support DDNS ♦ Support ping & trace route command on the web ♦ Support network capture on the web

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Module Type	GSM	WCDMA	LTE
Band	850/900/1800/1900MHz@GSM	900/2100MHz @UMTS 900/1800MHz @GSM	LTE FDD: B1/B3/B5/B8 LTE TDD: B38/B38/B40/B41 WCDMA: B1/B8 TD-SCDMA: B34/B39 CDMA: BC0 GSM: 900/1800MHz