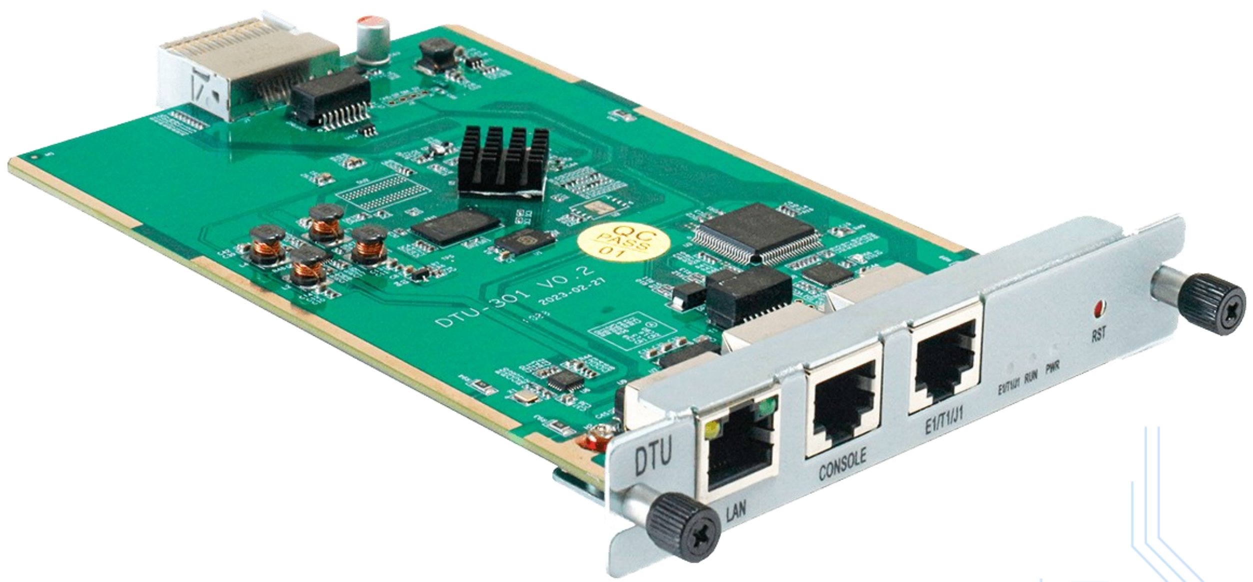
OpenVox



OpenVox DTU-301 Digital Gateway Module

Overview

DTU-301 (Digital Trunk Unit) is an open-source asterisk-based VoIP Gateway Module solution for operators and call centers. It is a converged media gateway product, which could be used with OpenVox UCP Series products. This kind of gateway connects traditional telephone system to IP networks and integrates VoIP PBX with the PRI/SS7/R2 seamlessly. With friendly GUI, customers may easily setup their customized gateway. Also secondary development can be completed through API.

The DTU-301 supports 1 software-selectable T1/E1 interface and supports up to 30 concurrent calls.

Target 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

Features

System Features

- Available in 1 port T1/E1/PRI, up to 30 energy efficiency concurrent processing
- Signaling: PRI/R2/SS7
- Support up to 24 countries' standard R2 signaling
- 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
- 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
- Support ports group management
- Support for custom dialplans
- Firmware update by HTTP
- Support call statistics
- Support auto provision
- Support channel status show dynamically
- Support backup/upload configuration file
- Multiple detailed log output
- Support Chinese language
- Automatically reboot
- Good compatibility, support Asterisk, 3CX, FreeSWITCH and Small and medium IPPBX platform
- Available for OEM/ODM
- 3-month "No Question Asked" Return Policy
- Two-year Warranty

SIP Features

- Support add, modify & delete SIP Accounts
- SIP registration with Domain
- 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

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 & traceroute command on the web
- Support network capture on the web

Technical Specifications

- 1 T1/E1 RJ45
- 2 10/100Mbps Ethernet ports（one on the front panel and one on the back panel)
- Maximum Power Consumption: 3W
- Operating temperature: 0℃~50℃
- Storage temperature: -20℃~70℃
- Operation humidity: 10%~90% non-condensing

DTU-301 E1/T1 VoIP Gateway Module	
Product Name	DTU-301
Interfaces	
T1/E1 Ports	1 RJ45 interface
Concurrent call	30 concurrent calls
Ethernet port	2 * 10/100Mbps ports（one on the front panel and one on the back panel)
Console port	1
USB	-
General Info	
Max Power Consumption	3W
Operation Temperature	0°C ~ 50°C
Operation Humidity Range	10% ~ 90% NON-CONDENSING
Storage Temperature Range	-20°C ~ 70°C

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