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OpenVox MIU Series Magnet Gateway

Overview

MIU Magnet Trunk Gateway, a powerful voice access device, provides 4/8 ports magnet phone to IP network access capability. Using standard SIP protocol, it can be interconnected with standard SIP softswitch systems to provide VoIP/MoIP solutions for operating magnet phone users and others. The device is easy to deploy quickly, has a high degree of security as well as strong anti-jamming capability, and is widely used in areas such as military or other special command and dispatch systems.

The MIU panel supports 4/8 magnet channel interfaces, Console interface, operation status indicator, powerstatus indicator, channel status indicator and RST restart button. In terms of software interface, MIU uses standard SIP protocol, compatible with mainstream IPPBX and SIP servers, and supports most of the VoIP operating system platforms, such as Asterisk, Issabel, 3CX, FreeSwitch, BroadSoft, VOS and so on.

Technical Specification

- Size:12.4cm×17.3cm×2cm
- Weight：178g
- Max Power Comsuption：6.5W
- Storage Temperature Range：-40~125°C
- Operating Temperature Range：0~50°C
- Operating Humidity Range：10%~90%

Features

- Support 4/8 ports magnet phone interface
- Supports WEB network management configuration function
- Support multiple network protocols and can integrate
- Multiple interface types with the unified communication platform.
- Can be connected to enterprise IP phone system and various unified communication systems to improve communication efficiency
- Calls to different magnet phone interfaces can be triggered by automatic detection, and calls can be established quickly.
- interfaces through automatic detection to quickly establish a call
- Supports automatic switch-off for peer-to-peer calls and automatic switch-off for peer-to-peer hang-up
- The magnet phone interface uses transformer coupling method, strong anti-interference ability
- Adapt to different network environments to ensure stable transmission
- Adaptive codec technology to ensure call quality
- Support SIP v1(RFC2543), v2 (RFC3261) protocol
- Provides Ethernet electrical and optical interfaces, 10/100BASE-T full-duplex, half-duplex,adaptive, and the maximum bandwidth can be up to 100M
- Support DHCP, DNS/DDNS, and NAT networking
- Rich log function
- Open API interface
- DTMF detection
- Support volume adjustment, gain adjustment, call hold, call waiting,call forwarding, caller ID
- Suppor SSH remote operation
- Support configuration file backup and upload
- Compatible with Asterisk, Elastix,3CX, FreeSWITCH SIP Server and VOS VoIP operation