

TSV-200-2E VOIP Trunk Gateway

Ref	Model
TSV-200-1E	Gateway with one E1/T1 port
TSV-200-2E	Gateway with two E1/T1 ports
TSV-200-4E	Gateway with four E1/T1 ports

Overview

TSV-200-2E Trunk Gateway enables direct routing of calls between the fixed line ISDN and the cost-effective IP networks for capitalizing on low cost VoIP telephony. By integrating TSV-200-2E Trunk Gateway with existing PBX / PABX telephone systems, businesses of all sizes can benefit from low-cost Voice over IP calls and achieve substantial cost savings without the high up front costs associated with communications infrastructure changes

related with the migration to VoIP.

TSV-200-2E offering PBX/IP-PBX users a powerful telephony cost saving device and advanced flexible routing capabilities between ISDN PRI to SIP, and also can be situated between the PBX and the PSTN, saving the cost of an additional PBX E1/T1 port. It is ideally fit for various access networks of SMEs, call centers, telecom operators and large-scale enterprises.

TSV-200-2E



Key Features

- Up to 120 simultaneous calls with 4* E1/T1 ports
- Support echo cancellation, DJB, CNG, VAD and QoS
- Use of existing E1/T1 ISDN PRI interface of the PBX
- Use of existing VoIP interface of the IP-PBX
- Maintain existing dialing habits and business communication patterns
- Built-in Web-based management, SNMP, command line interface (CLI)
- Standard ISDN compliant and Interoperable with a wide range of IP-PBXs

Physical Interfaces

E1/T1 Ports

1 to 4 E1/T1

Interface Type

RJ48(Impedance 120Ω)

Ethernet Interface

GE1: 10/100/1000 Base-T Adaptive Ethernet

GE0: 10/100/1000 Base-T Adaptive Ethernet

Serial Port

1* RS232, 115200bps

Voice Capabilities

Default codec:G.711a/μ law, AMR

G.723.1, G.729AB, iLBC (License)

Silence Suppression

Comfort Noise

Voice Activity Detection

Echo Cancellation (G.168), up to 128ms

Adaptive Dynamic Buffer

Voice ,Fax Gain Control

FAX:T.38 and Pass-through

Support Modem/POS

DTMF Mode: RFC2833/SIP Info/In-band

Clear Channel/Clear Mode

VLAN 802.1p/q

Environmental

Power Adapter: 100-240VAC@DC12V1A

Power Consumption:15W

Operating Temperature:0 °C ~ 45 °C

Storage Temperature: -20 °C ~80 °C

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 225*150*38mm

Unit Weight: 0.8kg

Compliance: CE, FCC

PSTN

ISDN PRI

23B+D(T1),30B+D(E1),NT or TE

ITU-T Q.921, ITU-T Q.931, Q.Sig

SS7 (optional)

ITU-T, ANSI, ITU-CHINA

MTP1/MTP2/MTP3, TUP/ISUP

R2 MFC (optional)

China and other 22 variants standard

E1 Frame Type :

DF,CRC-4,CRC_ITU

T1 Frame Type :

4-Frame Multi-frame (F4,FT),

2-Frame Multi-frame (F12, D3/4),

Extended Super-frame (F24, ESF) ,

Remote Switch Mode (F72, SLC96)

Line Codes:

E1:NRZ,CMI,AMI,HDB3

T1:NRZ,CMI,AMI,B8ZS

Maintenance

Web GUI Configuration, HTTP/HTTPS

Data Backup/Restore

PSTN Call Statistics

SIP Trunk Call Statistics

Firmware Upgrade via TFTP/Web

SNMP v1/v2/v3

Network Capture

Syslog:

Debug, Info, Error, Warning , Notice

Call History Records via Syslog

NTP Synchronization

Centralized Management System

Software Features

Local/Transparent Ring Back Tone

Overlapping Dialing

Dialing Rules, with up to 2000

Voice Codecs Group

Access Rule Lists

100 SIP Trunks

Route direction:

PSTN-IP, IP-PSTN, PSTN-PSTN

VoIP Protocol

SIP v2.0 (UDP/TCP),RFC3261

SDP,RTP(RFC2833), RFC3262,

3263,3264,3265,3515,2976,3311

RTP/RTCP, RFC2198, 1889

TLS/SRTP

SIP-T,RFC3372, RFC3204, RFC3398

SIP Trunk Work Mode :Peer/Access

SIP/IMS Registration :

with up to 256 SIP Accounts

NAT: Dynamic NAT, Rport