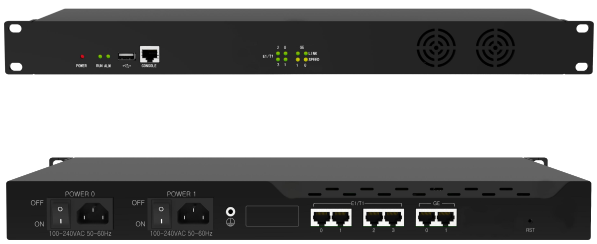
# TSV-IPB200-xE1 VOIP Trunk Gateway

| Ref | Model |
|---|---|
| TSV-IPB200-1E1 | Gateway with one E1/T1 port |
| TSV-IPB200-2E1 | Gateway with two E1/T1 ports |
| TSV-IPB200-4E1 | Gateway with four E1/T1 ports |

## Overview

TSV-IPB200-xE1 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-IPB200-xE1 Trunk Gateway withexisting 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 upfront costs associated with communications infrastructure changes related with the migration to VoIP.

**TSV-IPB200-xE1**



TSV-IPB200-xE1 offering PBX/IP-PBX users a powerful telephony cost saving device and advanced flexible routingcapabilities between ISDN PRI/SS7 to SIP, and also can be situated between the PBX and the PSTN, saving thecost of an additional PBX E1/T1 port. It is ideal for telecom distributors, resellers and service providers.

## Key Features

- Efficient concurrent processing, up to 60 channels
- Support flexible dialing rules and operations, allowing users to customize dialing rules according to different working environments
- Support T.38,Pass-through fax, as well as modem and POS machines
- Support multiple coding standards: G.711A/U, G.723.1, G.729A/B and AMR
- High compatibility, interoperable with PBX of Avaya, NEC and Alcatel, and also leading soft-switch of Huawei, Cisco and ZTE etc.

## 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

## PSTN

ISDN PRI

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

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

Signal 7/SS7

ITU-T, ANSI，ITU-CHINA

MTP1/MTP2/MTP3, TUP/ISUP

R2 MFC

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

## Call Features

**Flexible Route Methods**

PSTN-PSTN, PSTN-IP, IP-PSTN

Intelligent Routing Rules

Call Routing base on Time

Call Routing base on Caller/Called Prefixes

256 Route Rules for each Direction

Caller and Called Number Manipulation

## Voice Capabilities

**Codecs:**

G.711a/μ law,G.723.1,

G.729A/B, iLBC 13k/15k,AMR

Silence Suppression

Comfort Noise

Voice Activity Detection

Echo Cancellation (G.168), with up to128ms

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

## VoIP Protocol

SIP v2.0 (UDP/TCP),RFC3261

SDP,RTP(RFC2833), RFC3262,

3263,3264,3265,3515,2976,3311

RTP/RTCP, RFC2198, 1889

SIP 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

## Environmental

1+1 Redundancy Power Supply

Input 100-240VAC, 50-60 Hz

Power Consumption:15W

Operating Temperature:0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 436*300*44.5mm(1U)

Unit Weight: 2.0kg

Compliance: CE, FCC

## Software Features

Local/Transparent Ring Back Tone

Overlapping Dialing

Dialing Rules，with up to 2000

PSTN group by E1 port or E1 Timeslot

IP Trunk Group Configuration

Voice Codecs Group

Caller and Called Number White Lists

Caller and Called Number Black Lists

Access Rule Lists

IP Trunk Priority

## Maintenance

Web GUI Configuration

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