

# Digital Module SFP-E1 for IP PBX systems

Digital telephone module E1 in SFP form factor for integrating open software IP-PBX systems (Asterisk, FreeSWITCH, SipXecs, Yate) with PSTN or legacy PBX

## Features

### Network interfaces

- TDM – 1x E1 interface (G.703)
- IP – Ethernet 1000 BASE-X SFP, 1 SIP trunk
- SIP and RTP (up 30 VoIP channel)

### Signaling protocols

- IP – SIP, RTP, TCP, UDP
- PSTN – ISDN PRI (DSS1, QSIG, Q.931), R2 MFC R1.5

### Feature support

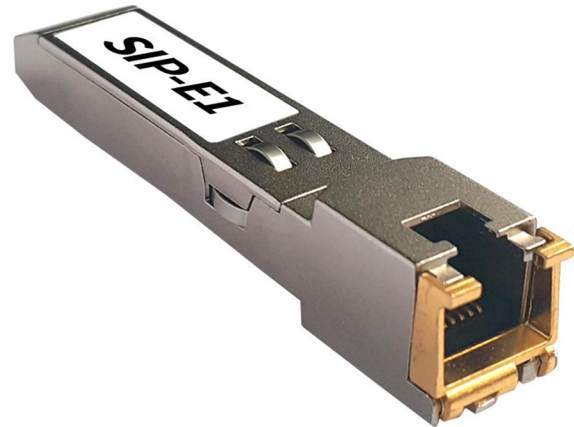
- Voice codec – G.711
- Echo Cancellation – G.168, 32 msec tail length
- Fax – fax over G.711, T.38 fax relay

### Control & Management

- Embedded Web Server
- SNMP Monitoring

### Power

- From the main device (DC 3.3V)
- Power consumption up to 1 W

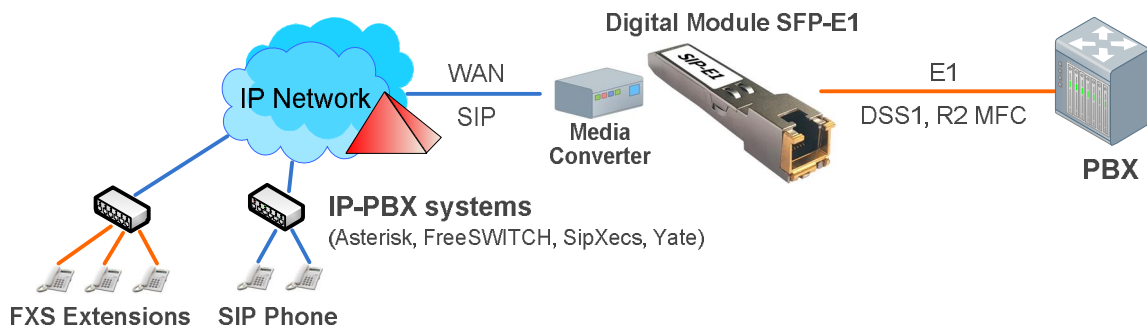


### Dimensions

- SFP form factor (14X14x67 mm)
- Weight – 0,025 kg

### Operating environment

- Operating temperature: 0°C to +50°C
- Humidity: up to 80% at +25°C



## Overview

Digital Telephone module in SFP form factor (or SFP VoIP Gateway) with support for the SIP protocol and E1 interface can be used to integrate open software IP-PBX systems (Asterisk, FreeSWITCH, SipXecs, Yate, Kamailio, CallWeaver, etc.) with the PSTN or legacy PBX.

Support for ISDN PRI (DSS1, QSIG, Q.931), R2 MFC, R1.5 signaling protocols for E1 interface used in the PSTN network, also SIP protocol, allows the use of equipment for reconstruction and modernization of various outdated Telephone Exchange.

Characteristics of E1 interfaces: Transfer rates, 2048 kbit/s and Number of channels at 64 kbit/s – 30.