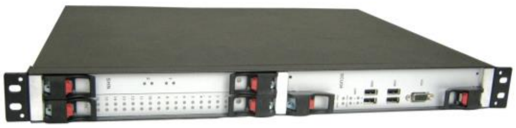
DATASHEET

Integrated IPPBX

TX-2110



Telco Grade Modular IPPBX

Product Overview

TX-2110 Integrated IPPBX Platform is a 1U sized carrier-class IP PBX designed for enterprises and professional customers moving to a Voice over IP (VoIP) architecture. With high reliability, stability, expansibility, and large capability,

TX-2110 ability to scale upto 1000 users and to support standards-based protocols, including Session Initiation Protocol (SIP), provide invest protect for enterprises for current and future telecom needs. With a built-in flexible and optional gateway interface, the platform can connect up to 8 analog loop-start central office (CO) lines or 1-4 E1 digital interface cables directly to PSTN.

The web-based NetSet management utility, hosted by the TX-2110 platform, provides intuitive system configuration that lets organizations avoid the need to train personnel in using command line interfaces or new programming languages.

Key Features

- **Friendly Web Management**
- **Plug and Pay and Auto Configuration**
- **FXS/FXO/E1 Supported**
- **IP Phone/Gateway/Softphone**
- **Voice Mail（Option）/Follow Me**
- **Built-In Automatic Call Distribution**
- **Voice Conference**
- **Conference based on Web Control**
- **Reception Assistance**
- **Call Recording**
- **Supports 30 Sip Trunks**
- **Browser-based NetSet Management**
- **Inbuilt Router,Firewall,Switch**
- **Embedded applications including voice messaging, auto-attendant**
- **Syslog Support**
- **Supports E1 PRI, CAS, SS7**
- **Support Call Center Suite**
- **Supports QSIG**

VOIP

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Technical Specification

Item	TX-2110
Hardware	
Dimension(w*d*h)	489*340*45
Weight	4~6kg
Power Consumption	95-105w
Power	100-240 VAC, 50-60 Hz
Operating temperature	0° to 40°C (32° to 104°F)
Storage temperature:	-40° to +70°C (- 40° to +158°F)
Humidity:	5 to 85% non-condensing
System Capacity	
Analog (FXS/FXO)	Max.32
Lifeline(optional)	Max.32
E1/PRI	Max.04
T1/PRI	Max.04
SIP Trunk	30
IP Phones/users	500
Concurrent calls	> 50
Network Ports	
WAN	1* 100/10M
LAN	1* 100/10M
Protocol	
SIP	RFC 3261, RFC2327
E1	PRI,CAS,SS7
Codec	G.711, G.729a/b, ADPCM, G.723
network:	802.1d, 802.1p, 802.1q, 802.2, 802.3af, 802.11, IP, IP-ToS, DiffServ, TCP/IP, UDP/IP, DHCP, DNS, PPPOE
Features	
Follow me	YES
Voice conference	YES
Automatic Call Distribution (ACD)	YES
Voice mail	YES
Call Recording	YES, Standard 500 Hours
FAX	T.38, T.30
Call park	YES
Call log	YES
Intelligent routing selection	YES
Dial rules	YES
RING GROUP	YES
Incoming call dial out authentication	YES
Call conference	30 users, support multi conference rooms
call center	YES
voice messaging	YES
auto-attendant	YES
BLF	YES
Call Authentication	YES
Auto configuration*	Mac or phone number binding and auto number distribution
Enterprise phone book*	YES
Enterprise billing	YES

VOIP

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CDR (Call Detail Record)	YES
Reset Switch	YES
Pop-on-Screen	YES
CRM	YES
Reception Assistance	YES
Management	
Telnet	Yes
Web	NetSet management
Language	English
SYSLOG	YES

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VOIP