

MTPBX5100 SIP IP PBX



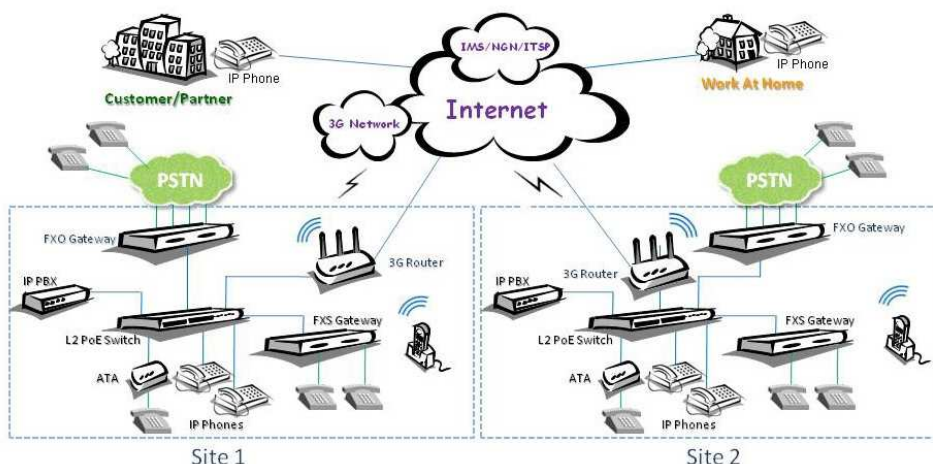
Product Overview

The MicroTell PBX5100 is a feature rich SIP based IP PBX telephony system. Designed for the small and medium business the MTPBX5100 is easy to install and manage. The MTPBX5100 integrates SIP Proxy Server, Registrar, Media Relay, and SIP Lines/Trunks into one unit. It provides users with legacy PBX functionalities, as well as VoIP functions such as free on-net calls and cost-saving off-net calls via ITSP subscription. Using MT's FXO/FXS Gateway or any other brand of T1 and E1 Gateway, the users can easily fit into all sort of telephony environment. Furthermore, standard SIP trunks and trusted peers make the MTPBX5100 seamlessly integrated with ITSP services.

Features include a built-in Auto Attendant service, Voice mail server and all the supplementary services you would expect to find in a small and medium business PBX, such as call forward, call transfer, DND, music on hold, call conference**, intercom, etc. and with the modularized ASD (Application Service Device) support, various advanced features and applications can be supported such as, VoIP broadcasting, WEB call, IP-MVPN, etc. The users may start from the least cost investment, and expand advanced feature sets later for better investment protection and optimized ROI.

FEATURES	BENEFITS
SIPv2 compliant	The MTPBX5100 can connect with any SIP based Internet Telephony Service Provider (ITSP), and build in a SIP registrar and proxy server can work with any SIP based client devices to make and receive low cost VoIP calls.
Supports 100 user accounts	The MTPBX5100 supports up to 100 user accounts and provides non-blocking service, it can support 100 con-current calls.
Supports 30 SIP Trunks	MTPBX5100 supports up to 30 SIP Trunks to connect to ITSP or FXO/E1 Gateway connecting to PSTN.
5-port 10/100Mbps switch	The five 10/100Mbps Ethernet ports allow you to connect the MT's FXS/FXO Gateway, Desktop SIP phones or PCs.
Auto Attendant service	Configurable with 3 levels of Interactive Voice Response (IVR) the Auto Attendant provides an automatic attendant service for both internal and external calls. For example the Auto Attendant can prompt the caller to input an extension number and automatically transfer the call.

Application Diagram



Key Features and Benefits

PBX FEATURES

- Build-in SIP Proxy Server, Registrar, Media Relay
- Intercom
- Timed Alarm
- Call Pickup Group
- Direct Call Pickup
- Group Hunting
- Automatic Call Back
- Hot Line
- Group Ringing
- Trunk Grouping
- Auto. Trunk Selection
- Automatic Routing Selection
- Specific Trunk Seize
- Trunk Class
- Call Barring
- Transit Call
- Trunk Booking
- Speed Dial
- Direct Outward Dial
- Flexible Numbering Plan
- E.164 numbering plan support
- Build-in Auto-Attendant

CALL FEATURE

- Call Transfer
- Call Forward
- Call Park
- Call Hold
- Music on Hold
- Conference Call(using MT's Desktop SIP phone)
- Do Not Disturb

SYSTEM CAPACITY

- Max. Extensions: 100
- Max. Trunks: 30
- Con-current call: 100

MANAGEMENT

- Web-based Management for System
- Console port CLI-like command
- Telnet CLI-like command
- Telephone command
- FTP for firmware/software remote upgrade
- Call Detail Record (CDR)
- NMS

VOICE CODEC Pass-through

- G.711 a-law
- G.711 μ -law
- G.723 (5.3/6.3Kbps)
- G.729A/B

TEMPERATURE

- Operating Temp.: 0 to 50 °C
- Storage Temp.: -10 to 70 °C

HUMIDITY

- Operating Humidity: 10~90% Non-Condensing
- Storage Humidity: 10~90% Non-Condensing

POWER CONSUMPTION

- 4 Watt

DIMENSIONS

- 128.8 x 79.3 x 23 (mm)

WEIGHT

- 0.5 Kg

POWER INPUT

- 9VDC 1.11A / 50~60 Hz

PROTOCOL

- SIP (RFC3261)
- DTMF (RFC2833, SIP INFO)
- IEEE 802.3, IPv4(RFC791), ARP, DNS, DHCP Client, ICMP, RTP, RTCP, TCP, UDP, SNTP, Port Forwarding
- STUN

NETWORK INTERFACE

- 5 10/100Mbps ports
- Support Static IP/DHCP

WARRANTY

- Limited Lifetime

Contact

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