MicroTEL

MTSPG211

SIP Personal Gateway (Analog Terminal Adaptor)

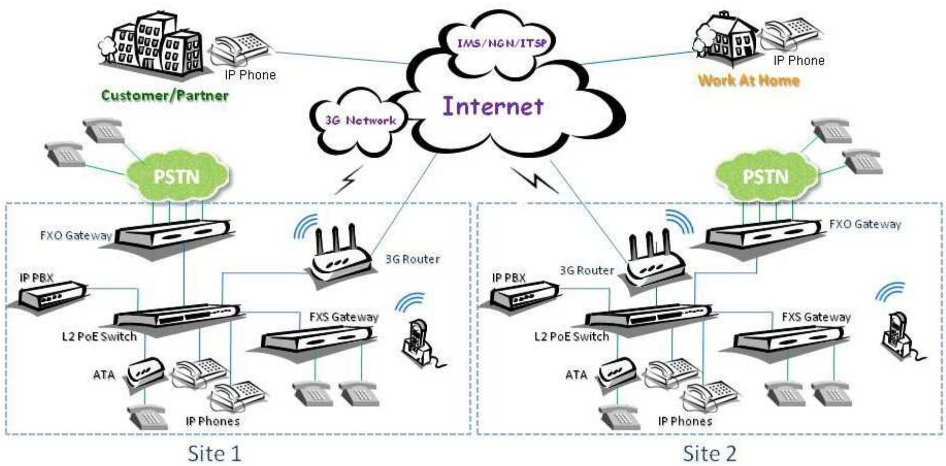
Product Overview

The MicroTell SIP Personal Gateway(MTSPG211) is compliant with the SIPv2 Voice over Internet Protocol (VoIP) and can be used with a SIP based Internet Telephony Service Provider (ITSP), IP PBX and other SIP based client devices to make and receive VoIP calls anywhere you have internet access. When used with an ITSP both home and business users can take advantage of reduced rate calls to the traditional telephone network, and generally free calls over the Internet.

For small businesses the MTSPG211 can be used with MT's IP PBX (MTPBX5100). This allows you to take advantage of free calls between branch offices located around the world as well as reduced rate calls to the traditional telephone network. Increased functionality is added with supplementary services such as Call Waiting, Call Hold, Call Resume, Call Transfer, Call forwarding and 3-way conference calling. You can also centrally manage VoIP user accounts and automatically provision these settings to phones.

FEATURES	BENEFITS
SIPv2 compliant	The MTSPG211 can be used with a SIP based Internet Telephony Service Provider (ITSP), IP PBX and other SIP based client devices to make and receive low cost VoIP calls.
Supplementary services	Supplementary services such as Call Waiting, Call Hold, Call Resume, Call Transfer, Call forwarding and 3-way conference calling provide increased functionality at no extra cost.
Superb voice quality	Advanced Digital Signal Processing (DSP), Silence suppression, VAD, CNG and AEC provide superb voice quality.
2-port 10/100Mbps switch	The two 10/100Mbps Ethernet ports allow you to connect the MTSPG211 and a PC using a single network point.
Auto provisioning**	MTSPG211 support remote auto provisioning via HTTP or FTP.

Application Diagram



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Key Features and Benefits

CALL FEATURES**
- Call Hold/Resume
- Call Forward
- Call Park
- Call Transfer
- Call Pickup
- 3-Way Conference Call
- Caller ID
- Call History
- Redial
- DND
- Abbr. Dial

VOICE FEATURES
- SIPv2 (RFC3261)
- SIP INFO (RFC2976)
- SIP RPORT(RFC 3581)
- SDP (RFC2327)
- RTP (RFC1889)
- RTCP (RFC1890)
- G.711 a-law
- G.711 µ-law
- G.726
- G.729A/B
- G.723.1
- Voice Activity Detection (VAD)
- Comfort Noise Generation (CNG)
- G.168 Acoustic Echo Cancellation (AEC)
- Adaptive Jitter Buffer (AJB)
- T.38 Fax Relay

DTMF METHOD
- In-Band
- RFC2833
- SIP INFO

NAT TRAVERSAL
- STUN (RFC3489)
- Outbound Proxy

MANAGEMENT
- Keypad configuration
- Web based management interface (HTTP)
- Auto provisioning**
- Remote firmware upgrade**

SECURITY
- HTTP 1.1 Basic/Digest Authentication for web setup
- MD5 for SIP authentication (RFC2069/RFC2617)

Hardware Overview
- 2 port 10/100Mbps
- Phone port: FXS interface for connecting a analog phone
- Line port: FXO interface for connecting to PSTN

IP Assignment
- Static IP
- DHCP
- PPPoE

TEMPERATURE
- Operating: 0°C to 40 °C
- Storage: -20°C to 70 °C

HUMIDITY
- Operating: 10% to 85% Non-condensing
- Storage: 5% to 90% Non-condensing

DIMENSION
- 122 × 92 × 31mm

WEIGHT
- 0.2Kg

POWER INPUT
- 12VDC, 1A

CERTIFICATIONS
- FCC
- CE

SYSTEM REQUIREMENTS
- Available 10/100Base-T Ethernet port
- A SIP based VoIP user account
- We recommend using this product with the MTPBX5100

PACKAGE CONTENTS
- One SIP ATA (MTSPG211)
- One power adapter
- One RJ-45 Ethernet cable
- Quick Start Guide

WARRANTY
- Limited Lifetime

**Requires network support

Contact

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