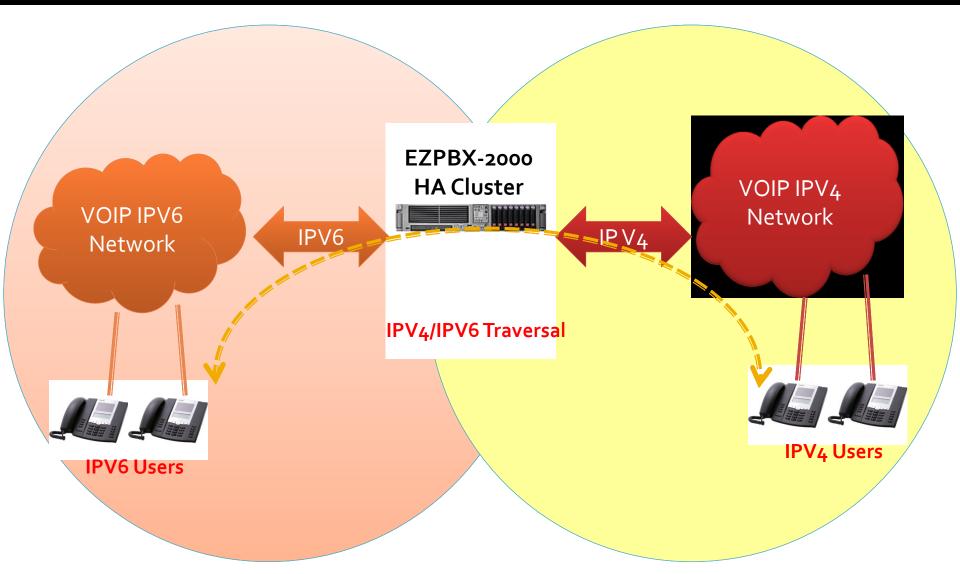
EZPBX-2000 IPV4/V6 Dual IP-PBX

Release 4.0 (2024)



Seamless IPV4 to IPV6 Migration



System Components



Highlights

- Run IPv4 and IPv6 SIP Calls Simultaneously
- High Performance/Reliabilities
- Hitless HA Redundant
- SIP UDP, TCP, TLS Seamless Connection
- Automatic Audio/Video NAT Traversal
- Work as SIP Trunk and SIP Router
- Powerful Digit Manipulation and Call Routing Plan
- Prosperous Telephony Features for Time to Market
- Multi-language Web Management
- Support RFC8599 Push Softphone
- Support SRTP Transcode
- SIP Attack Detection and Prevention
- Country/IP Network Lock

Highlights

- Auto Attendant Service/AA Call Flow Editor
- DID Based Auto Attendant
- Voice Mail Service/MWI/Email Notice
- Up-to 32 Parties Conference Room
 - Meeting Me Conference
 - Dialing Out Conference
 - Adhoc Conference
- Upto 64 parties Broadcasting Service
- Support G.711, G.729A^{*}, GSM and G.722
- Divisional Billing
- Support CPE Auto Provisioning*
- RADIUS, SYSLOG and Call Detail Record
- Running under 64 bits Linux

System Capabilities

- Max Concurrent Extension: 2,000
- Max Concurrent Call: 1,000
- Max NAT/RTP Resource: 1,000
- Max Universal Resource: 256
- Max Voice Logging Resource: 512
- Max BHCC: 200,000
- Max Conference Parties: 16
- Max Broadcasting Parties: 64
- Audio Codec: G.711, G.729A, GSM, G.722
- Hitless HA Redundant

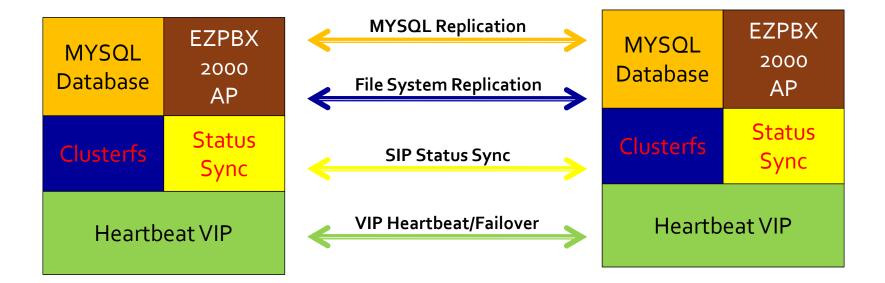
System Service

- Support Multiple SIP Domain
- SIP Proxy/Registrar
- Automatic Audio/Video NAT Traversal
- Support WAN/LAN Interface as Voice Router
- Support RADIUS, Syslog Billing
- Support Extension/Device Monitoring
- Device Allowance/Block Device List
- Country/IP Network Lock
- Missed Call Email Notice
- SIP Attack Detection and Prevention
- INVITE-Initiated Dialog Event (BLF)
- Support SIP Trunk
- PSTN/Mobile Extension

System Service (2)

- ANI Based Routing
- Call Limited By DNIS
- Support RFC8599 Push Softphone
- Support SRTP Transcode
- In Call Service

Hitless HA Redundant



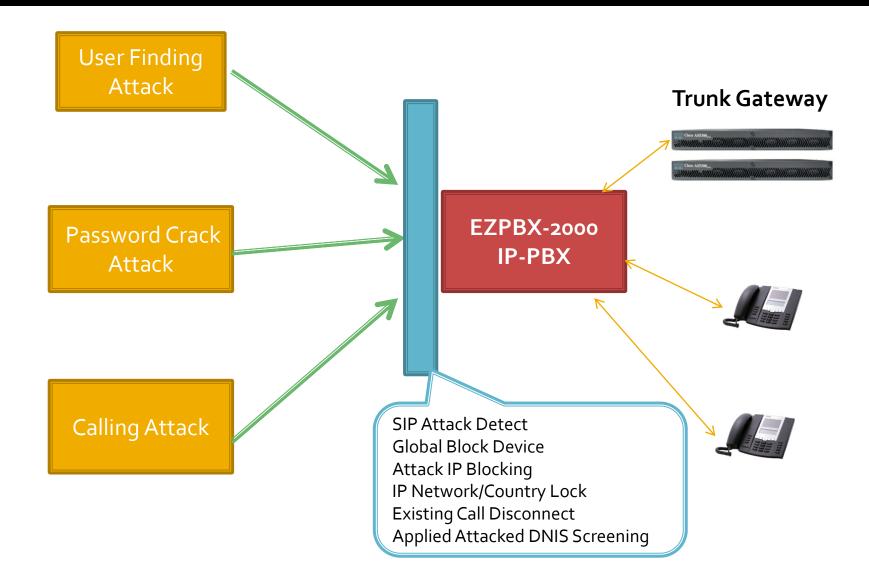
Hitless HA:

- ✓ Connected call will be kept and continue to talk with 2 to 3 seconds silence.
- ✓ Voice recording will be separate into 2 records (before and after failover)
- ✓ Unconnected call will be dropped
- ✓ Calls to AA/VMS will become silence

SIP Security Design

- SIP Attack Detection/IP Blocking (e.g. SIPVicious, Friendly Scanner)
- Disconnect Existing Calls when SIP Attack Detected
- Apply more Secure DNIS Screening Group when SIP Attack Detected
- SIP User Device Restriction
- Global Block Device List (Black List)
- Country/IP Network Lock
- Applied Enhanced Password Option
- Black Routing List For Those Expensive Countries
- Enable CAPTCHA to Protect Web
- Web Access Log

SIP Attack Detection/Protection



Country Network Lock Service

Country Lock:

- Each Extension Have 2 Counties
- Block Register if not from these 2 country
- IP/Network Lock:
 - Each Extension Have 2 IP/Network Define
 - Block Register if not from these 2 IP/Network

Flexible & Powerful Routing Plan

- Group Based Routing
- Time of Day/Weekday Routing
- Preference Routing
- Round Robin Routing
- Load Balancing Routing
- Broadcast Routing
- Unavailable Redirect
- ENUM Routing
- Routing Based No Answer/Response Time
- Black List route
- ANI Based Routing (highest priority)

Telephony Features

- DID/DOD
- Call Transfer
- Call Hold
- Call Waiting
- Call Forward
- Call Display Name
- Camp-On Call
- Call Pickup
- CLIP/CLIR
- DID Routing
- Digit Manipulation
- Call Park/Retrieve
- Missed Call Notice
- Music On-Hold

- Local Emergency Call
- Abbreviate Dialing
- PSTN Number
- Parallel Hunting
- Follow Me Always
- Time of Day Follow Me
- Incoming Call Blocking
- Outgoing Call Blocking
- Time Based Screening
- Outgoing Privilege Calling
- Do Not Disturb
- Anonymous Call Blocking
- Distinct Ringing

PSTN/Mobile Extension



- 1. Ring Extension and Mobile at same time
- 2. Mobile will work as an extension after access to PBX

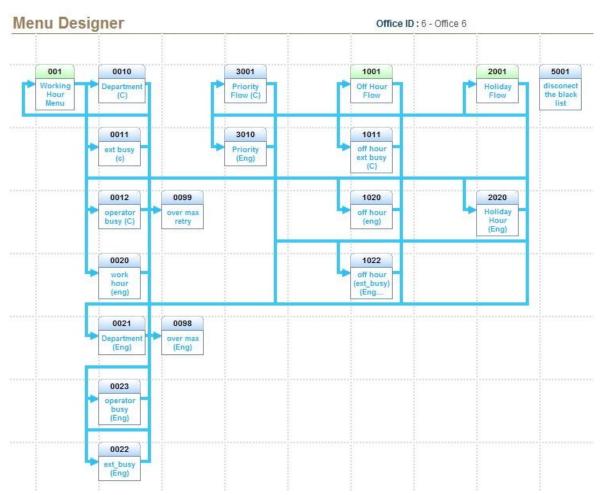
Statistic Reports

- Call Statistic Report
- Extension Statistic Report
- Extension Status Detail Report
- NAT Resource Statistic Report
- Representative Number Report
- Trunk Report

Auto Attendant Service

- Support Multi-languages/Multi-Offices
- Company or DID based Call Flow
- Graphic Attendant Flow Editor
- Office Oriented Call Flow
- Up-to 3 Time Segments
- Working/Off-Time/Holiday/Special Working Day/Priority Flow/Operator
- Black List Filter & Flow/Forward without Answer
- Support Abbreviate Dialing/Schedule Called Group
- Access to Voice Mail
- Outgoing Calling (Password Protect)
- Access to Meeting Me Conference

Call Flow Editor



- Office Based Flow
- Menu Oriented
- Easy Setting
- Template Copy
- Office Copy
- Menu Copy

Voice Mail Service

- Support Multi-languages
- Incoming Calls Limitation
- Support Message Detail
- Voice Mail to Email (MP3)
- Access Voice Mail via Web or Phone
- SIP MWI (RFC 3842)
- Personal Greeting

Access VMS from WEB

Voice Mail Access		Calling Time	▼ 🔍 Sea
Extension Number: 6002			
Calling Time 🔽	Calling From	Status	
2011/07/26 09:57:31.668	6006	0	

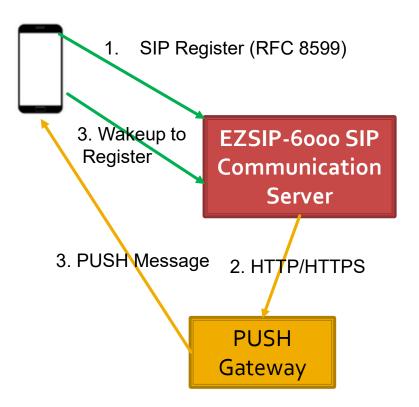
Page 1

Total Record: 1

Delete | Delete All | Back

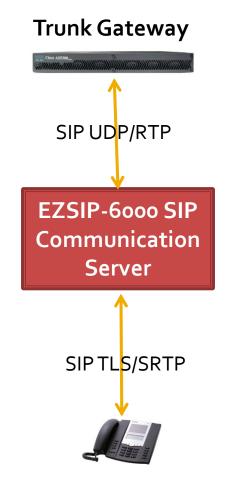
RFC-8599 Push Softphone

- RFC 8599 Compliance
- Customized PUSH Gateway is Required
- Max Time to Wait Softphone Register and Call



SRTP Transcode Support

- SIP TLS and UDP Transparent
- Support SRTP Optional/Mandatory
- SRTP and RTP Transcode



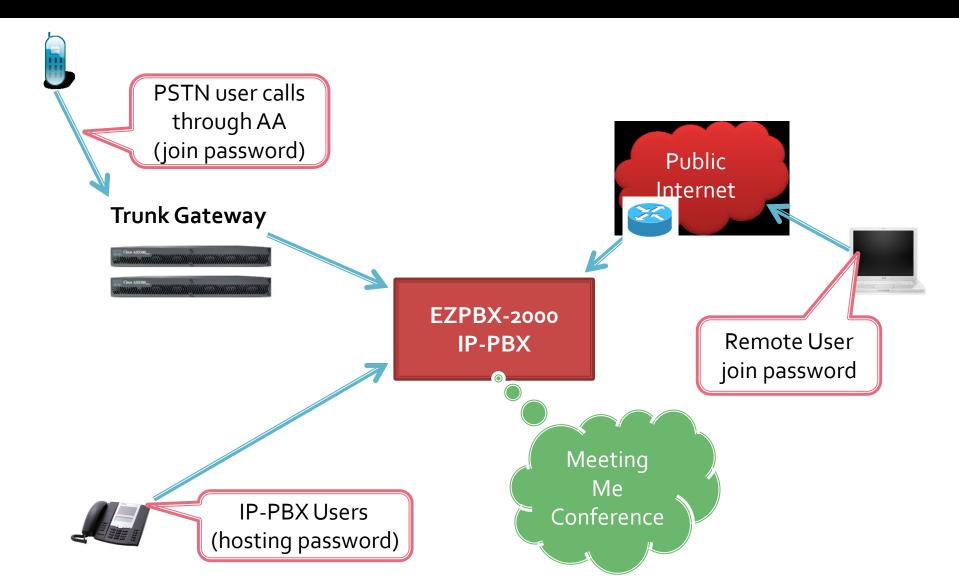
In Call Service

- Support Regular User/Mobile User
- Press ## during the Call
- Call Flip (from Mobile to Extension or vice versa)
- Call Transfer
- Call Hold/ Unhold

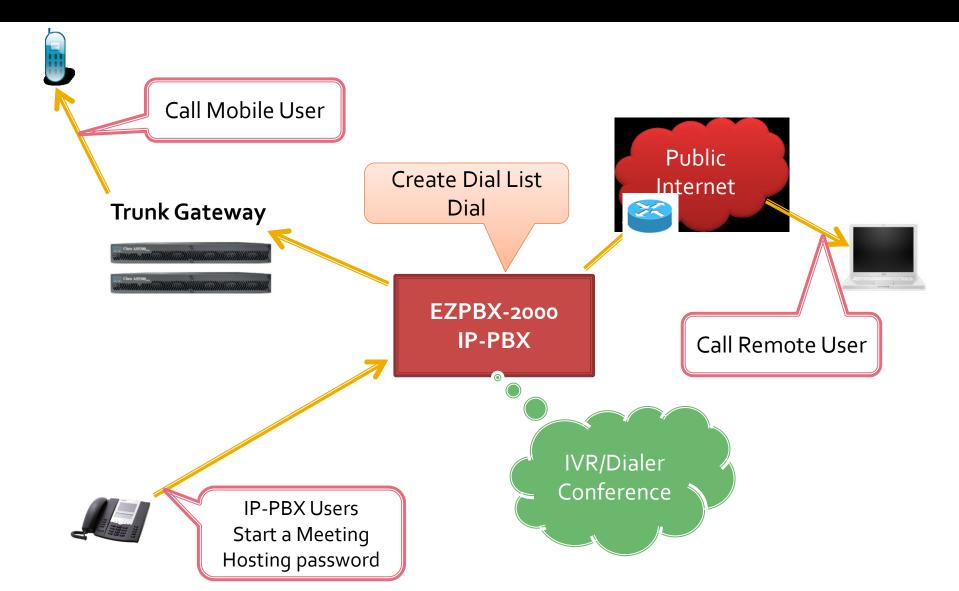
Conference Service

- Up-to 32 Participants Conference
- Support Multi-languages
- Incoming Calls Limitation
- Web Conference Control by Smart Calling (Optional)
- Support Meeting Me Conference
 - Hosting/Participant Password
 - Join/Quit Announcement
- Support Dialing Out Conference
 - Hosting Password
 - Predefine Participant List (Auto Dial Out)
 - Dynamic Participant List Building/Calling (Ad-Hoc)
 - Join/Quit Announcement
 - Add Participant within Conference

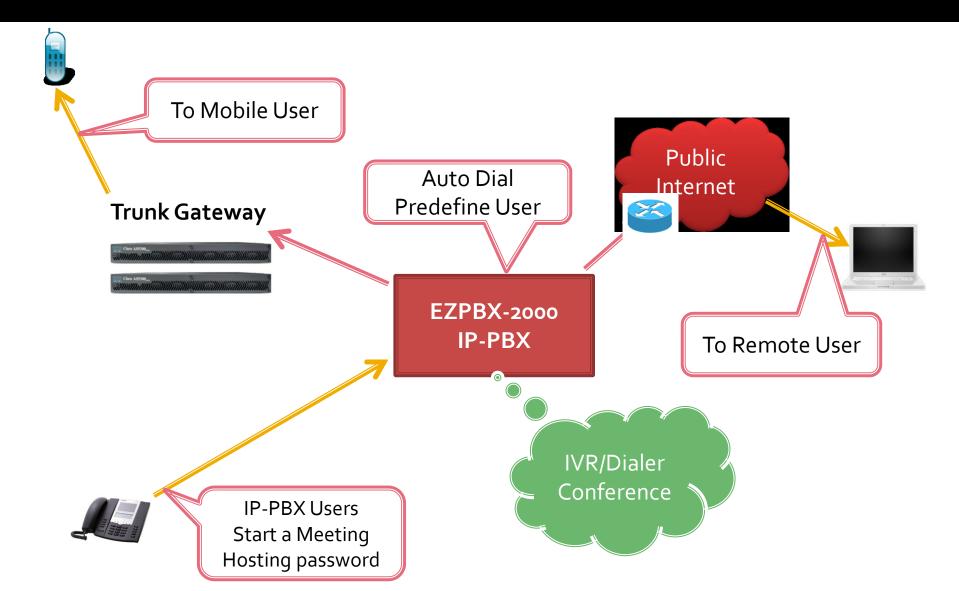
Meeting Me Conference



Ad-hoc Conference



Dialing Out Conference



Broadcasting Service

- Up-to 64 Participants Broadcasting Service
- Support Beginning/Ending Notice
- CPE Auto Answer

CPE Auto Provisioning

Support SIP Multicasting PnP

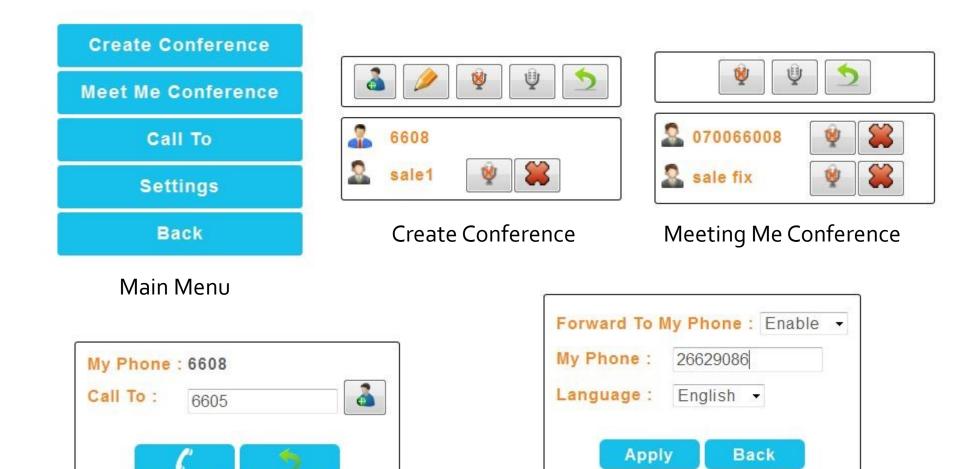
- Auto Show Un-Assigned MAC & User Agent
- Support Add to Extension
- Support Predefine Provisioning URL
- Support http/https MAC File Configuration
- Support http/https Firmware Upgrade
- Support CPE Sync Configuration/Reboot

Smart Calling Module (Optional)

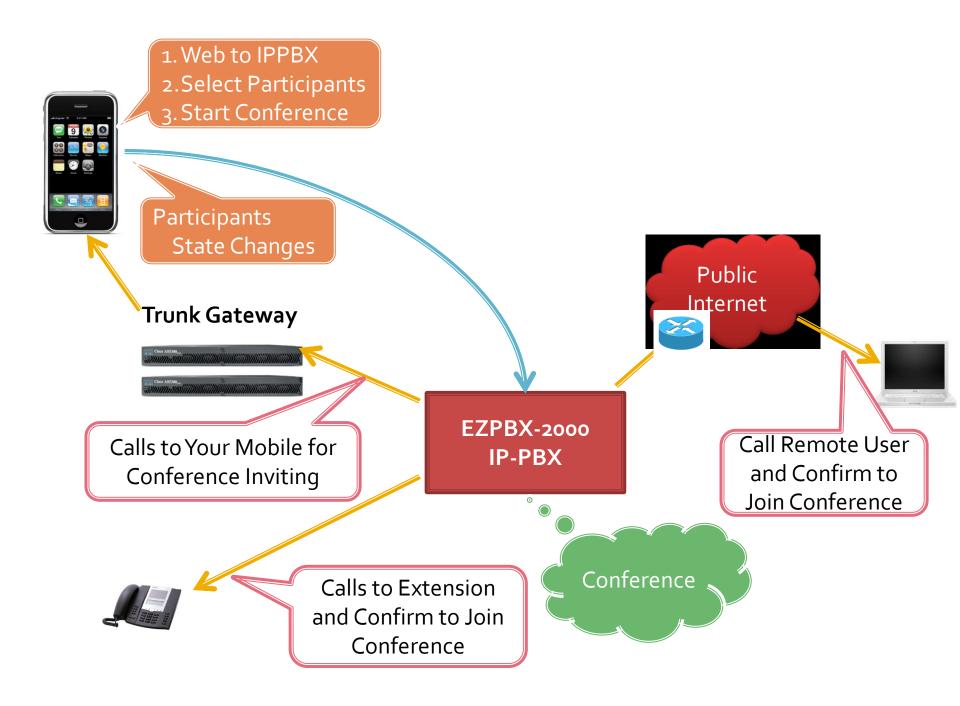
- Worked with iPhone & Android Phone
- Receive Extension Calls from Mobile
- Call Your Customer through IP-PBX
- Create Conference through IP-PBX
- Meeting Conference Monitor
- Conference Control/Monitor
 - Mute, Hand-up, Disconnect, Add

Smart Calling

Call To



Settings



Billing Feature

- Flat Call Detail Record File
- Enterprise Billing Feature
- Support Charge Division/Division Manger
- Top Usage Users Report
- Top Prefix Usage Report
- Prefix Summaries Report
- Division Billing Report
- Division Wide Tariff Plan
- Call History Detail Report

Billing Report Examples

Division Billing Report

Query Condition Period :	2011 🔻 - 08 👻	~ 2011 ▼ - 08 ▼		ely 📑 Print	
Period	Division	Calls	Duration	Charge Amount	Charge Percentage
2011-08	Sales	31	4,274	558.670	2.912%
2011-08	RD	0	0	0.000	0.000%
2011-08	technical support	22	6,248	18,629.400	97.088%
	Total :	53	10,522	19, <mark>1</mark> 88.070	

Division : All Division

l	Ranking	Extension Number	Calls	Duration	Charge Amount
	1	6006	21	6,242	18,611.400
	2	20010	22	3,822	479.520
	3	20018	8	422	73.150
	4	6009	1	6	18.000
	5	20016	1	30	6.000
ľ		Total :	53	10,522	19,188.070

System Diagnostic

- System Real Time Status
- Extension/SIP Trunk/Call Status
- Blocked IP Status
- Ping Test
- Machine Status
 - Disk, Memory, CPU, Network, I/O, Files System
- Debug/Capture Interface

Management

- Multi-Language Support
- Support 2FA based on Google Authenticator
- Administrator, Division Manager, User
- Web Provisioning Access Log
- Easy Web GUI (HTTP/HTTPS)
- On-line Manual & Pop-up Help
- Customizable Web Access Rights
- System Alert by Syslog/Email/HTTP Hook
- Real Time System Monitor & Tracing
- Backup/Restore System
- SOAP Provisioning Interface
- Schedule Update for SSL, License, Patch etc.

Extension List GUI

Extension				Extension Number	▼ ~	Search
Office ID: 1 - office1 😔 Inactive 🝚 Unregister 🝚 Ready 🝚 Talk 🝚 Ringing						
Extension Number	🛆 Name	Belonged Division	SIP Security	RADIUS Call Authorization	Contact Policy	Extension Type
⊖ 000			None	No	Permanent Contact	FXO/Trunk/Proxy
⊜ 00000			Register/Invite	No	Register	IP Surveillance
⊜ 00001	0123456789012	1 - Sales	Register/Invite	No	Register	Phone/ATA
⊜ 00002		1 - Sales	Register/Invite	No	Register	ENUM
⊖ 0001			Invite	No	Permanent Contact	SIP Trunk
⊜ 0002			Register/Invite	No	Register	SIP Trunk
			Invite	No	Permanent Contact	SIP Trunk
☺ 0004			Invite	No	Permanent Contact	SIP Trunk
⊖ 0005			None	Yes	Permanent Contact/NAT	FXO/Trunk/Proxy
⊖ 0006		1 - Sales	Invite	No	Permanent Contact/NAT	Phone/ATA
⊜ 0007		1 - Sales	Register/Invite	No	Register	Phone/ATA
⊕ 0008		1 - Sales	Register/Invite	No	Register	Phone/ATA
⊕ 0009			Register/Invite	No	Register	Phone/ATA
⊖ 001			None	No	Permanent Contact/NAT	FXO/Trunk/Proxy
⊜ 0010		1 - Sales	Register/Invite	No	Register	Phone/ATA

Call Statistic Report

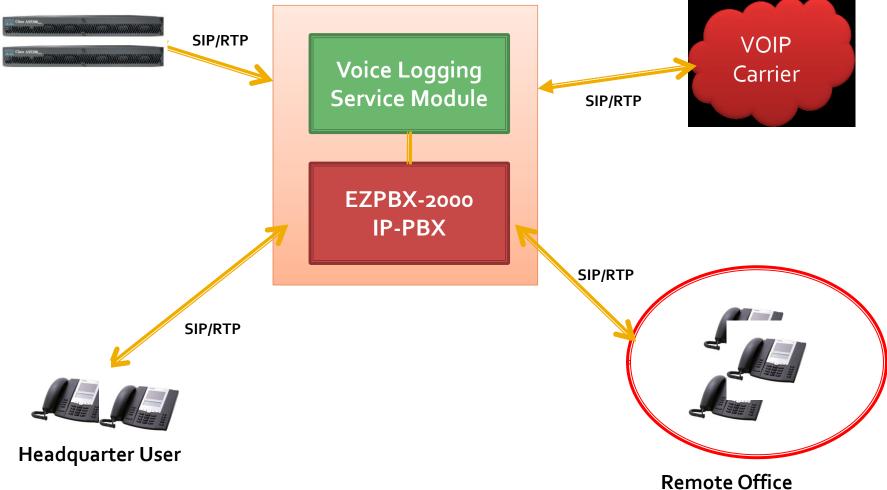
呼叫統計報表



區間	總呼叫	總通話	最高呼叫	最高通話	接通率
00-01	0	0	0	0	0.00%
01-02	0	0	0	0	0.00%
02-03	0	0	0	0	0.00%
03-04	0	0	0	0	0.00%
04-05	0	0	0	0	0.00%
05-06	0	0	0	0	0.00%
06-07	0	0	0	0	0.00%
07-08	0	0	0	0	0.00%
08-09	0	0	0	0	0.00%
09-10	11	10	1	1	90.90%
10-11	0	0	0	0	0.00%
11-12	0	0	0	0	0.00%
12-13	0	0	0	0	0.00%

Voice Logging Module





Voice Logging Module

- Support IPv4/v6 Voice Recording Simultaneously
- Support G.711, G.729A, GSM, G.722, G.723, iLBC Decode
- Support Extension/PSTN Number Recording
- WAV/MP3 Compressed File Format (VBR, CBR)
- Support AES Encryption
- Provides Voice Logging Detail Report
- No High Performance Switch Mirror Required
- Support External NAS or DB



