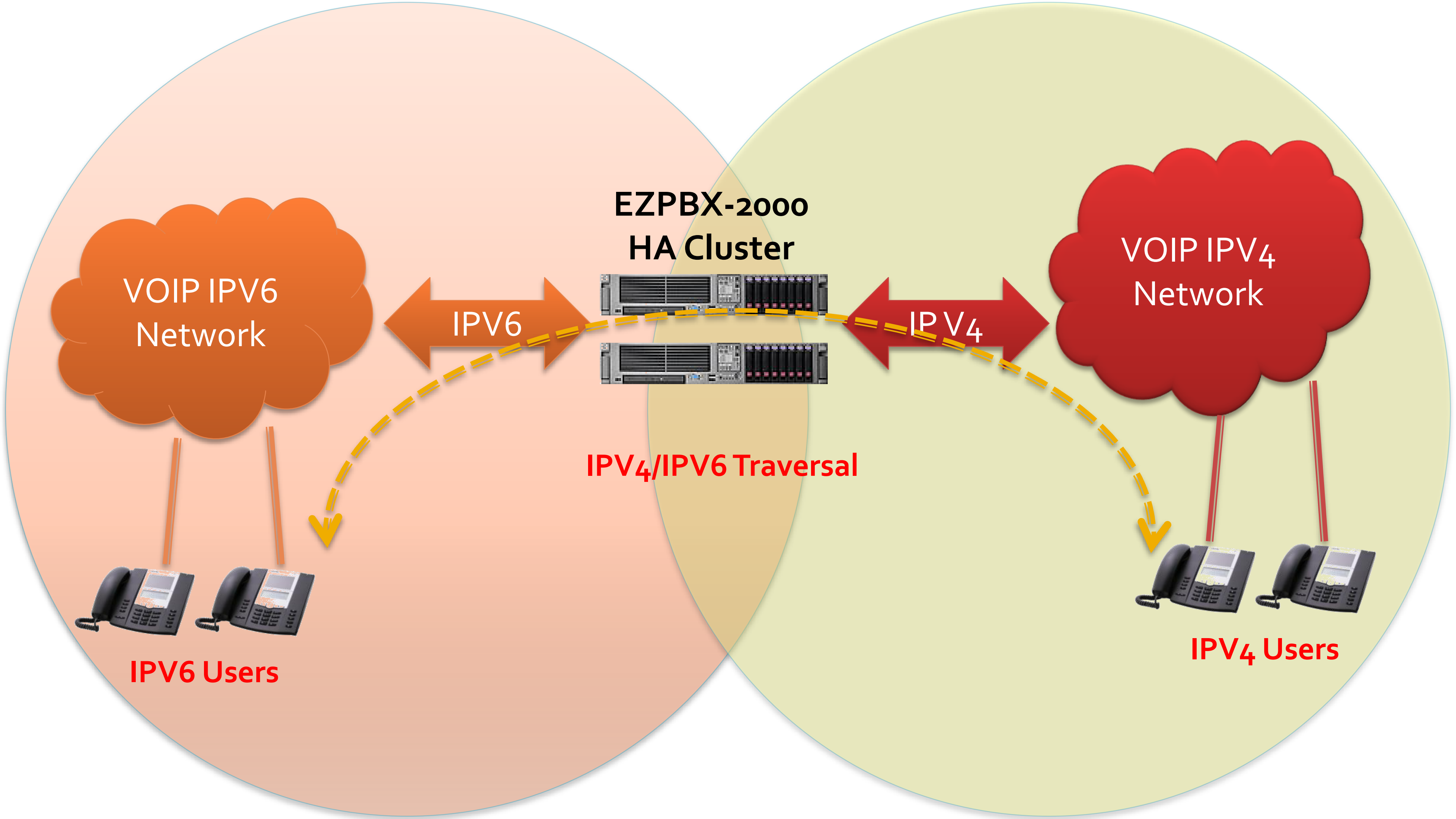
EZPBX-2000
IPV4/V6 Dual IP-PBX

Release 3.0 (2023)

Ezvoicetek
www.ezvoicetek.com

Seamless IPV4 to IPV6 Migration



System Components

Highlights (1)

- 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 (2)

- 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

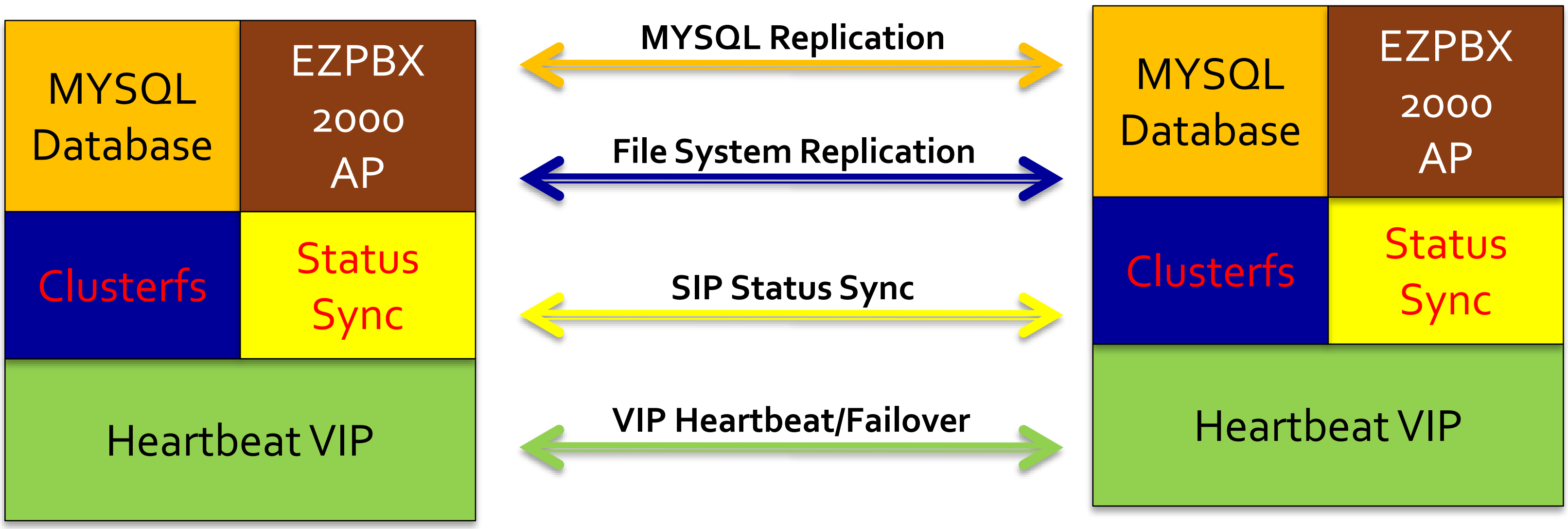
System Service

- 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



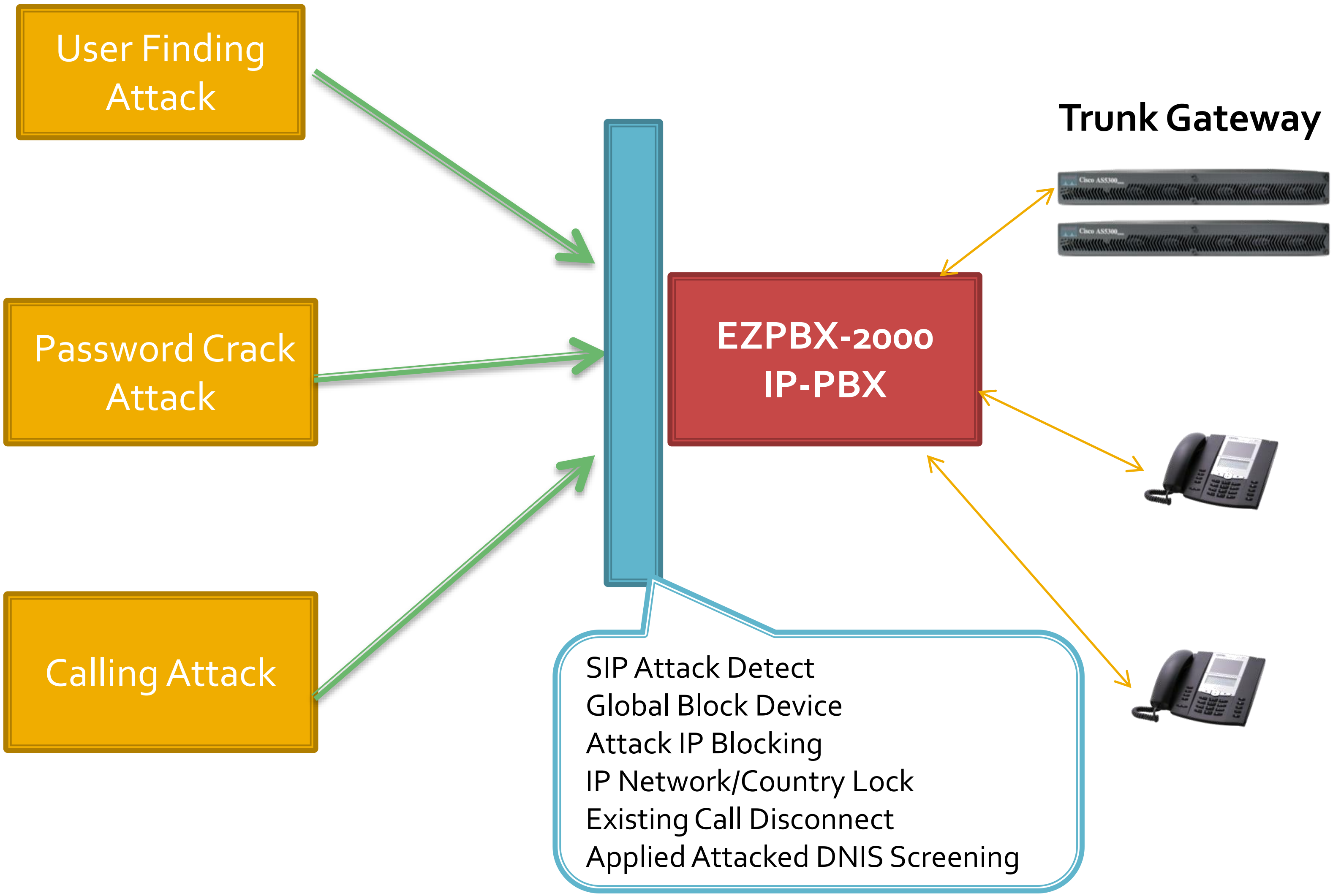
Hitless HA:

- ✓ 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

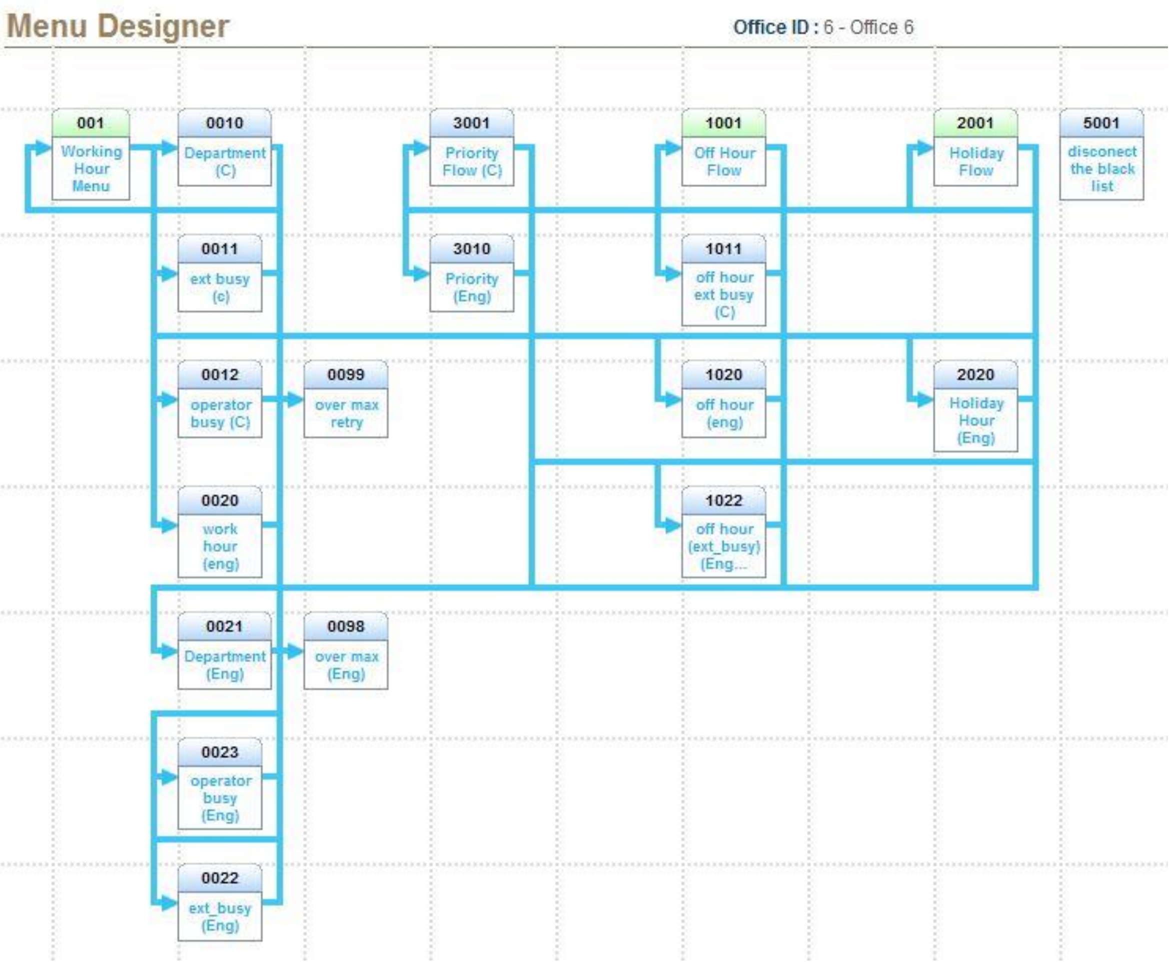
Statistic Reports

- 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 | Search

Extension Number: 6002

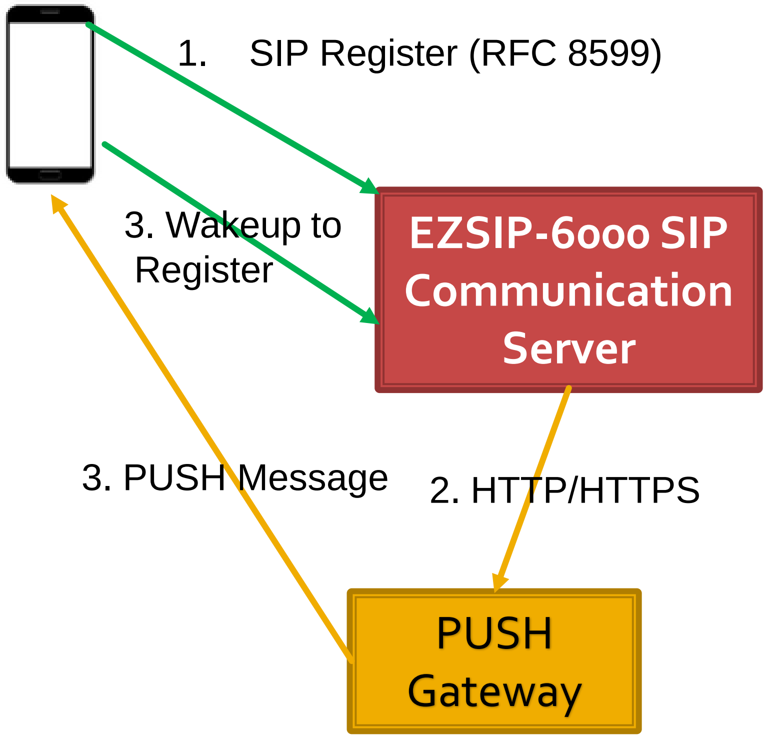
Calling Time	Calling From	Status
2011/07/26 09:57:31.668	6006	

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

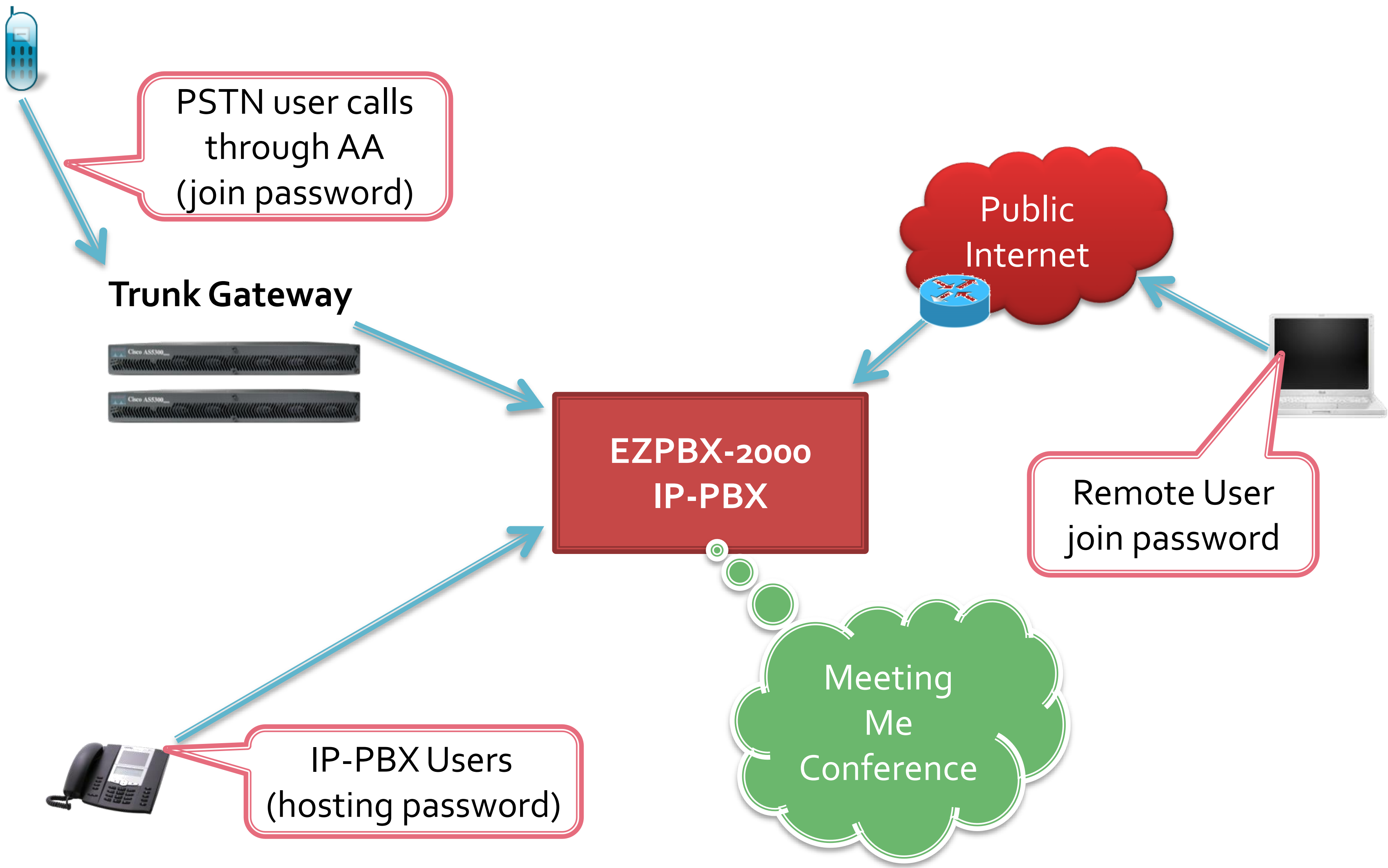
In Call Service

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

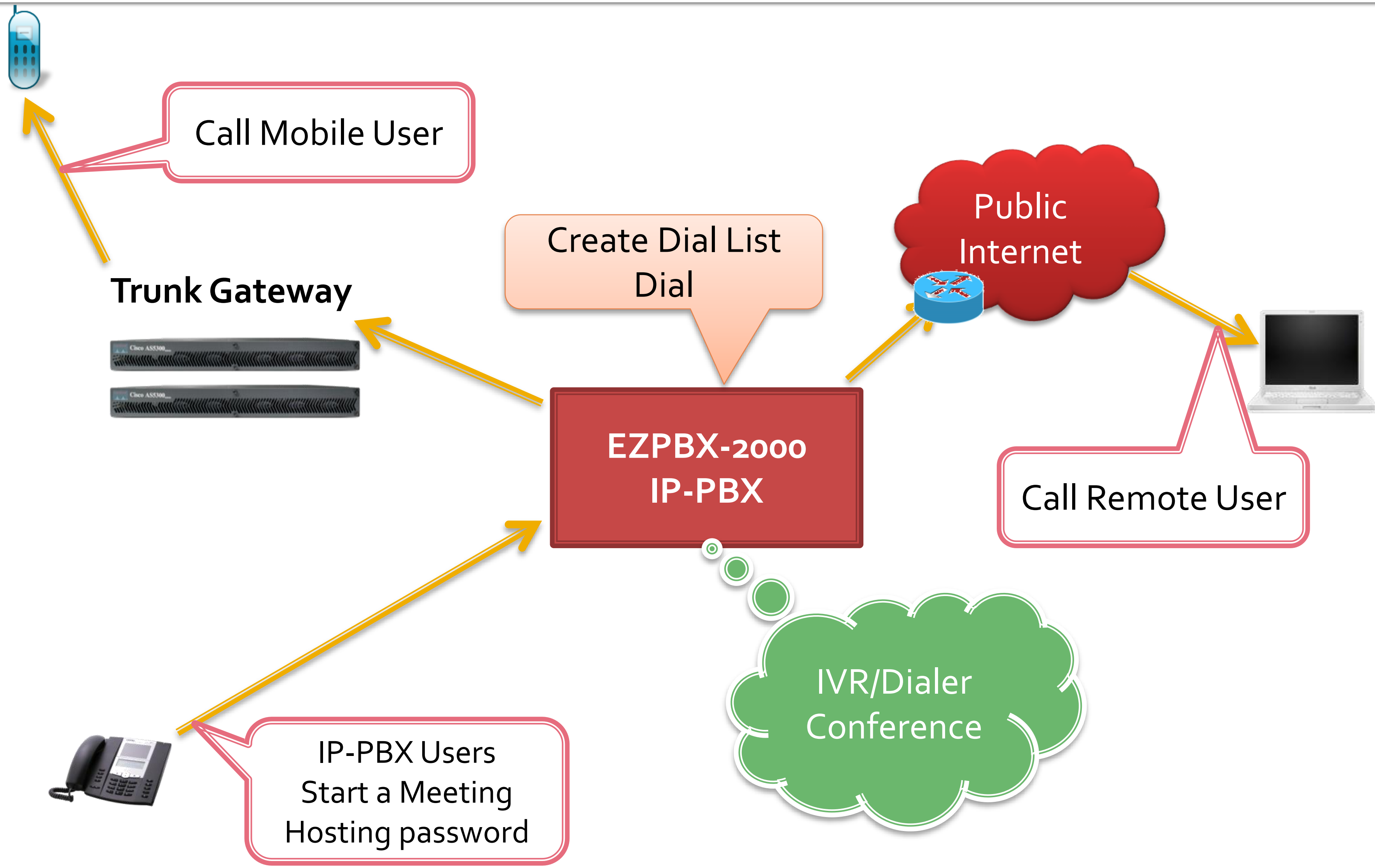
Conference Service

- 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

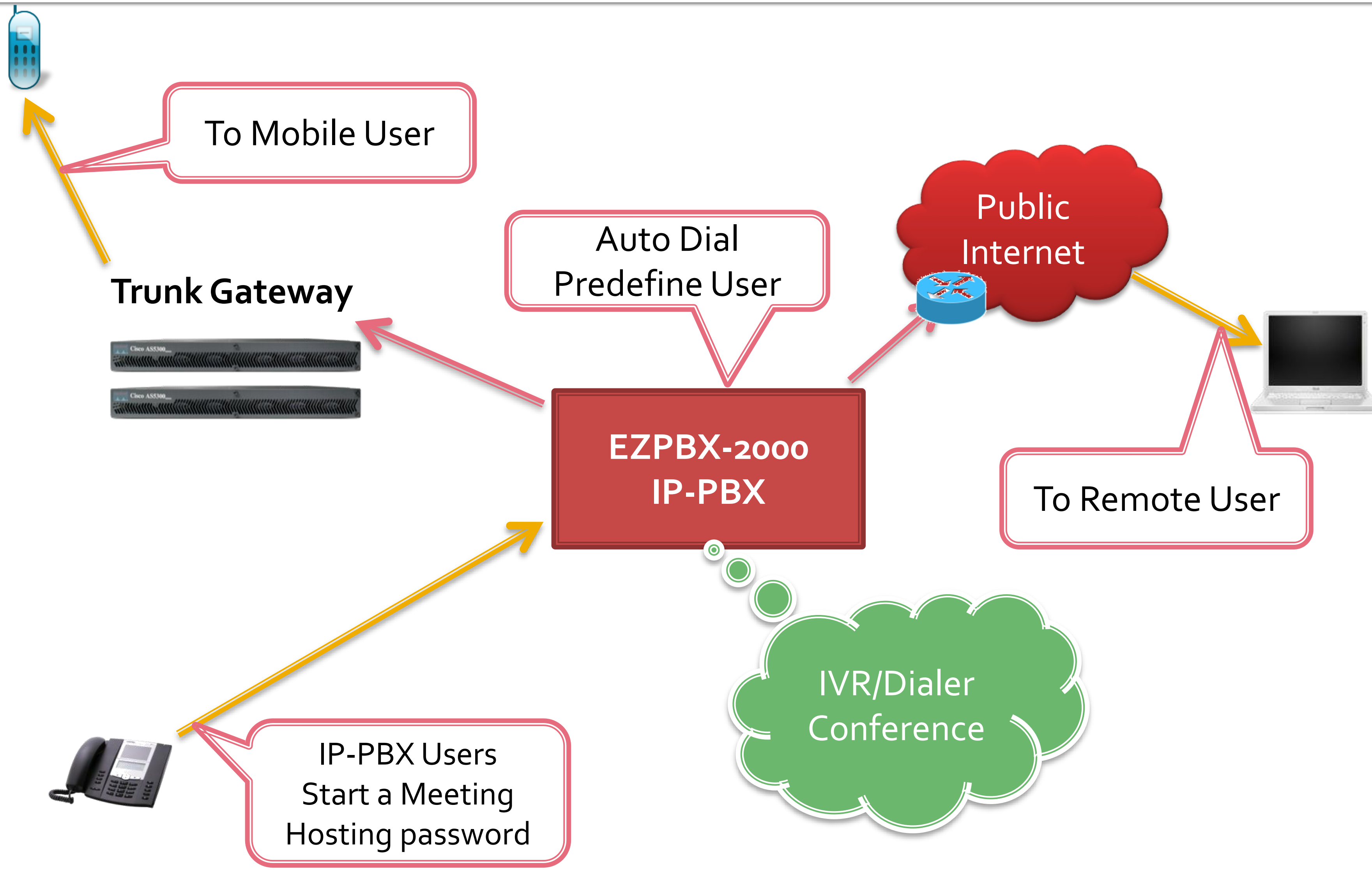
Meeting Me Conference



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

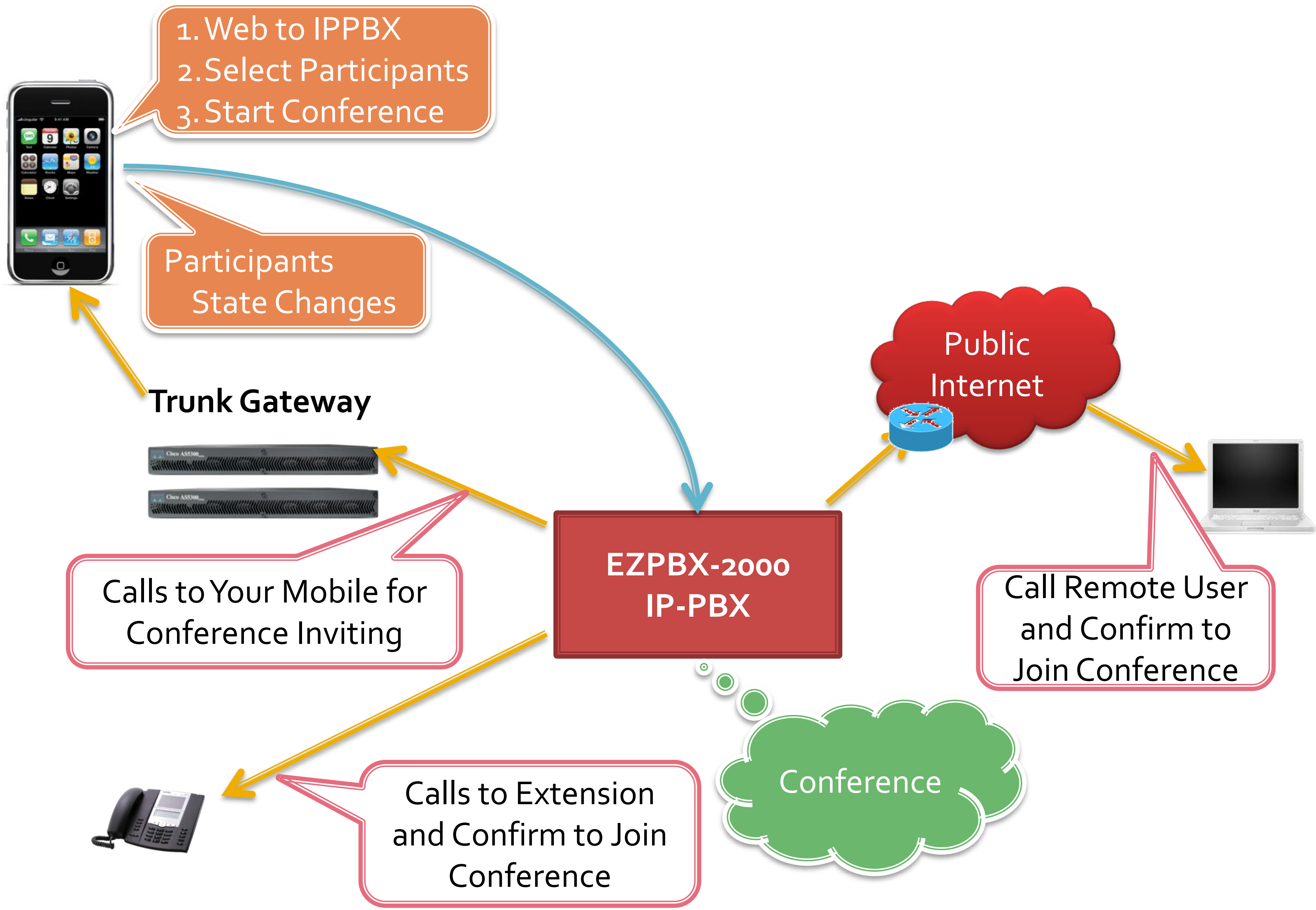
Main Menu

Create Conference

Meeting Me Conference

Call To

Settings


1. Web to IPPBX
2. Select Participants
3. Start Conference
Participants State Changes
Trunk Gateway
Public Internet
EZPBX-2000 IP-PBX
Calls to Your Mobile for Conference Inviting
Call Remote User and Confirm to Join Conference
Conference
Calls to Extension and Confirm to Join Conference

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

Apply 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	19,188.070	

Division : All Division

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
	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
- 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
0003			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

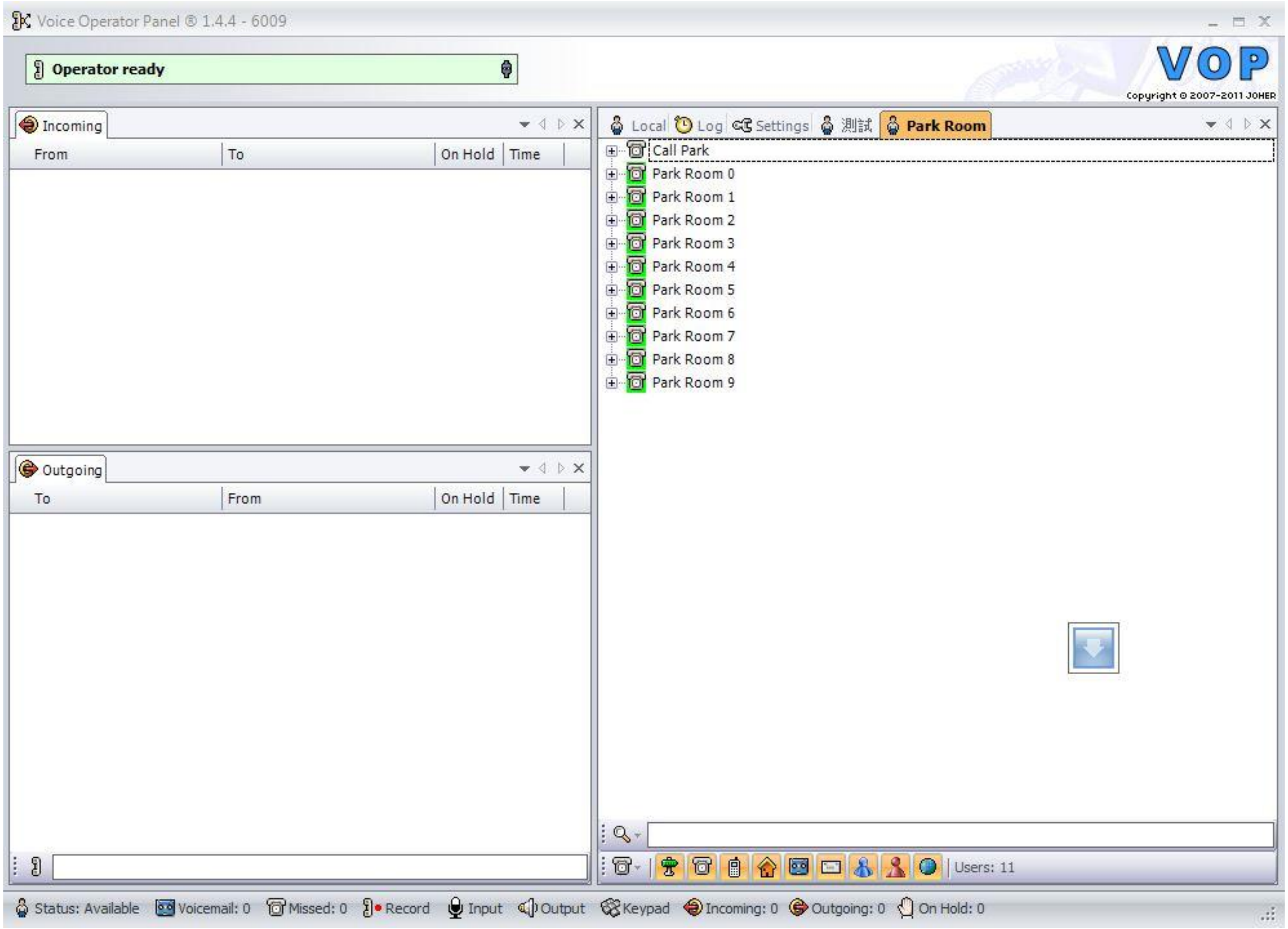
年: 2011 月: 7 日: 25 查詢 列印 匯出 刪除

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%

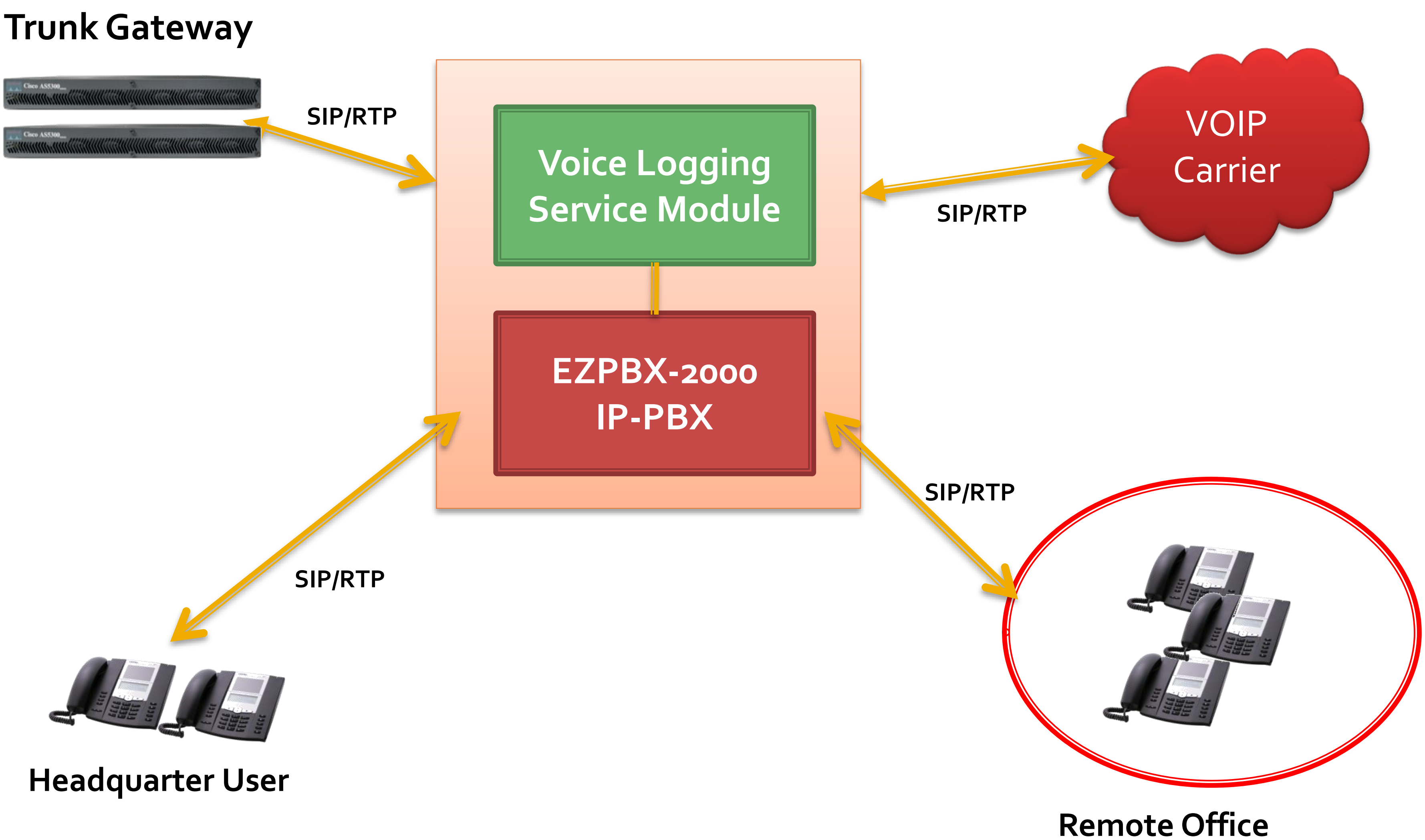
3rd Party Operator Console

- Support Windows 2000, XP, Vista, 7
- Support G.711, G.729A
- Outlook/LDAP/Local Directory
- Phone Presence
- Park Room Status
- Consultant/Blind Call Transfer
- Call Queuing
- Drag and Drop Calling
- Integrated/Tested with EZPBX-2000

Operator Console – Call Park Monitor



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
- 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

Q&A