



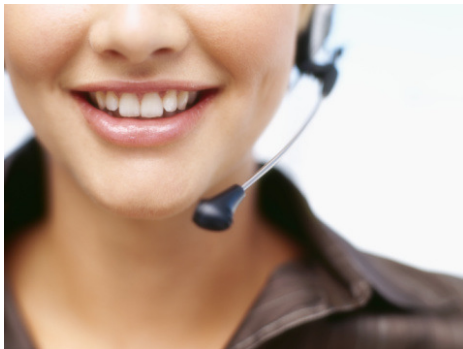
**M** IPPBX



# IPPBX SYSTEM SOLUTION

A new mode to communicate





MIPPBX® is a line of Asterisk®-based IP-PBX products designed to meet the needs of companies from 5 to 500 employees. MIPPBX® is a unified communications system that integrates the best tools available for Asterisk-based PBXs into a single, easy-to-use interface. It also adds its own set of utilities and allows the creation of third party modules to make it the best software package available for open source telephony.

The goals of MIPPBX® are reliability, modularity and ease-of-use. These characteristics added to the strong reporting capabilities make it the best choice for implementing an Asterisk-based PBX.

## MIPPBX ARCHITECTURE

[ It is All in One System ]



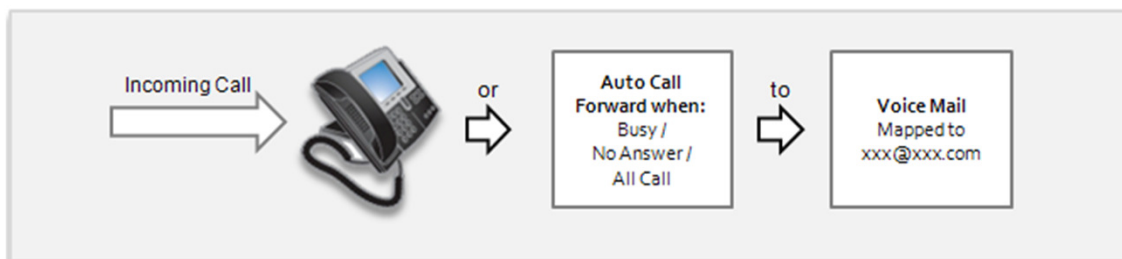
**MIPPBX : Feature : Voice Mail**

**:: Feature**

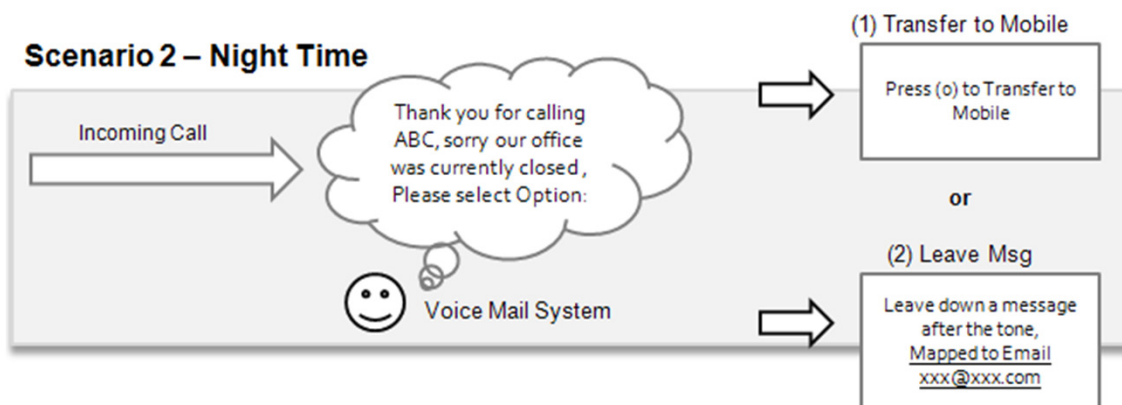
MIPPBX integrated voicemail application to replace traditional voicemail systems with Asterisk-based solutions at a fraction of the cost. MIPPBX voicemail can be implemented as a basic stand-alone system or can act as



**Scenario 1 – Divert to Voice Mail**



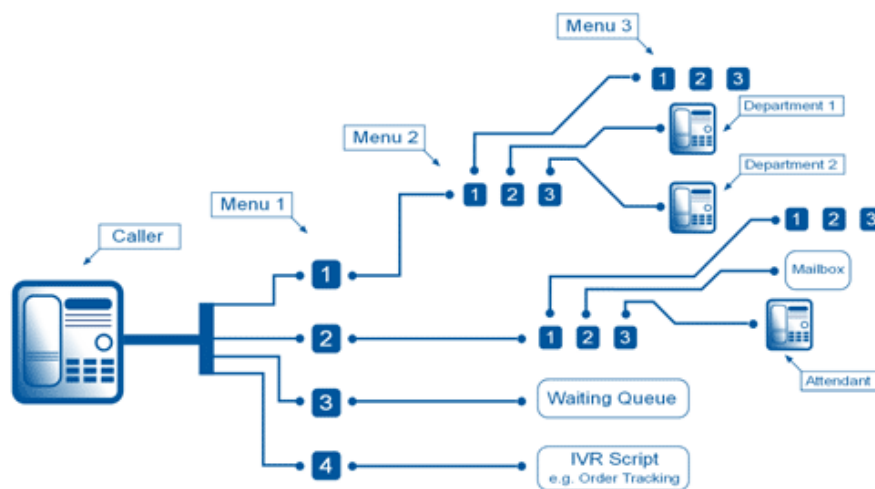
**Scenario 2 – Night Time**



**MIPPBX : Feature : Auto Attendance**

**:: Feature**

Auto Attendance allows companies to save money and eliminates manual repetitive tasks. Asterisk makes it easy to build IVR applications that respond to tone or speech input from the caller. Asterisk's support for data access of over HTTP and ODBC makes integrating the IVR with the data stores simple and reliable. front end



**MIPPBX : Feature : Voice Conference Bridge**

**:: Feature**

MIPPBX's Voice Conference bridge application allows you to build sophisticated multi-party conference applications with only a few lines of Dialplan script. Capable of scaling to hundreds of parties, Asterisk-based conference servers represent one of the most compelling values of IPPBX.to a Unified





**Video Conference (Polycom VC to VP2009 Video Phone)**

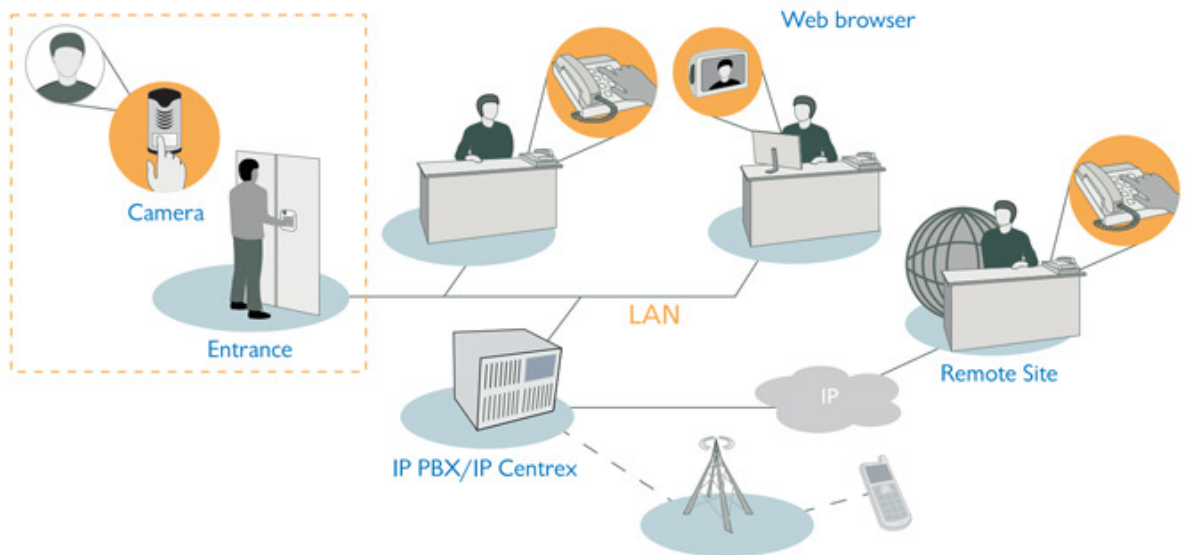
Now, you can make use of MIPPBX Video Conference feature to make a video call from Polycom Video conference system to Yealink VP2009 Video Phone by SIP.



**Video Conference (Counterpath to Counterpath)**

Besides, you can make use of MIPPBX Video Conference feature to make a video call. Video calls can make conversations much more interesting and intimate.





**Camera Viewer and Door Opener (Software)**

The user can easily to open the door by a simple software which can view the visitor on the Computer and open the by one "Click".



**Snom video integration**

The Video door access systems seamlessly integrate with snom phones and the video camera models are ideal for working with Snom 8xx or Yealink 2009P. The real time video is streamed to the colour screen of the 821 or 870 and can be viewed at any time.

There is no need for any 3rd party software or PBX support to view the video stream.



**Web video integration**

Alternatively images can be viewed on a standard PC screen whilst audio and access control are still handled via the IP phone.



**iPhone video integration**

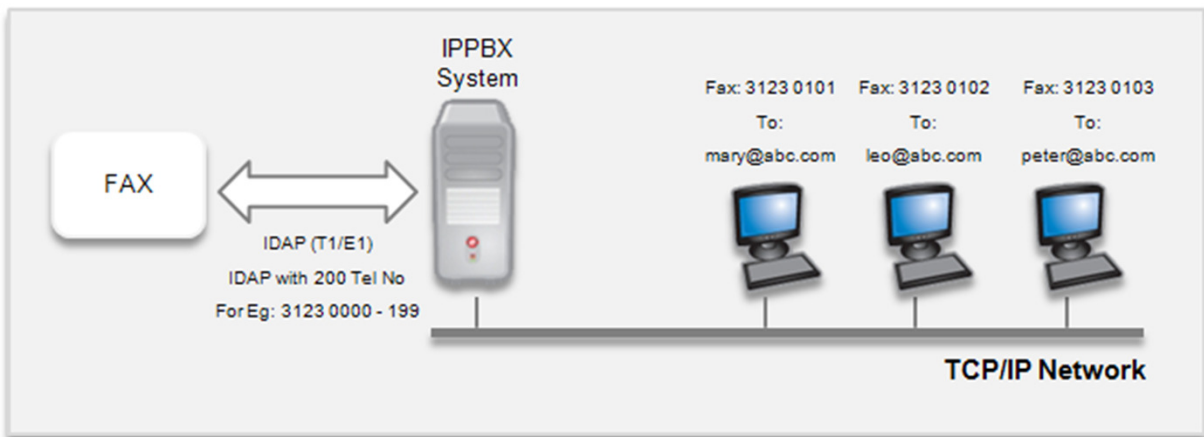
For a bit of fun and as a way of showing how flexible the entry phone can be we have even routed the images to an iphone. As with the PC screen audio and access control would still be via an IP phone.

**MIPPBX : Feature : Fax Server**

**:: Feature**

To improve competitiveness, reduce costs and meet the fast and changing market demand, companies have implemented office automation, including fax system has become the most important part. However, the traditional fax machine and can not be integrated with other information systems, but not the best use of information resources and to achieve true office automation. system, storing message using

**FaxServer (Incoming Fax)**



**FaxServer (outgoing Fax)**

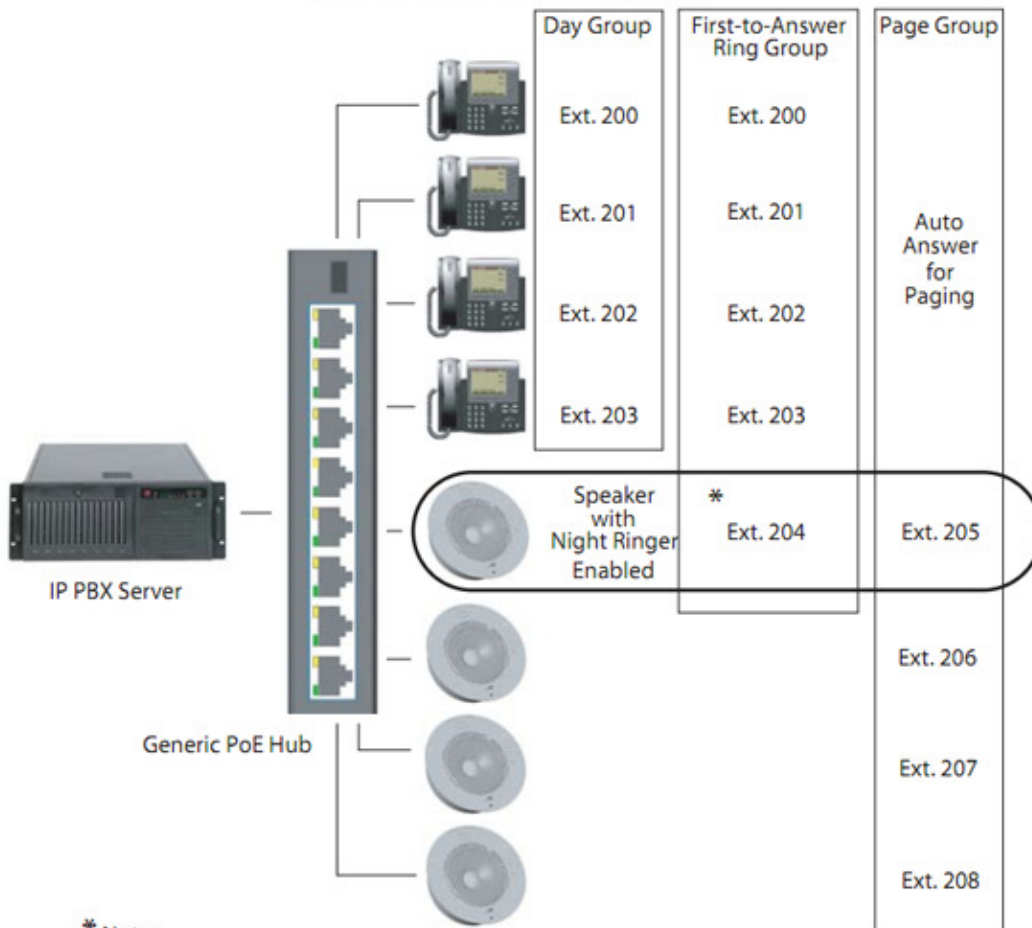


MIPPBX : Feature : VOIP Paging

:: Feature

SiP-enabled V2 Ceiling Speaker is a Power-over-Ethernet (PoE 802.3af) and Voice-over-IP (VoIP) public address loudspeaker (IP Speaker) that easily connects into existing local area networks with a single CAT5 cable connection. The speaker is compatible with MIPPBX. In a non SiP environment, the speaker is capable of broadcasting audio through multicast. Its small footprint and low height allows the speaker to be discreetly mounted almost anywhere.

Typical Installation - Ringer Mode



\* Note:  
An audio ring file is activated when this SIP extension of the Ceiling Speaker is dialed.

## :: Connectivity

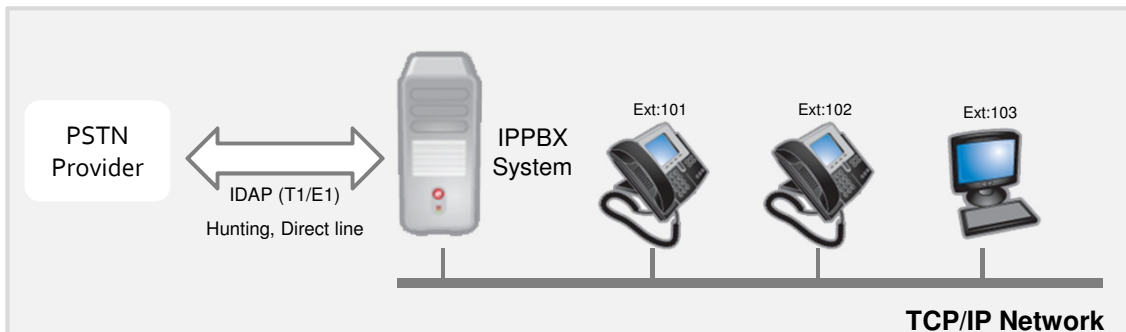
### MIPBX Solution Provides

Regardless of which Base Solution you choose, because they are all based on MPBX you can be assured that they all enable you to:

- Buy what you need and grow – *Flexible and modular system*
- Empower all user – *Enterprise class feature and applications*
- Guarantee long term investment – *Robustness, reliability and high performance*
- Benefit from industry innovation – *Open standards base solution with IP at the core*

### New Site / Full IP Solution

#### Head Office

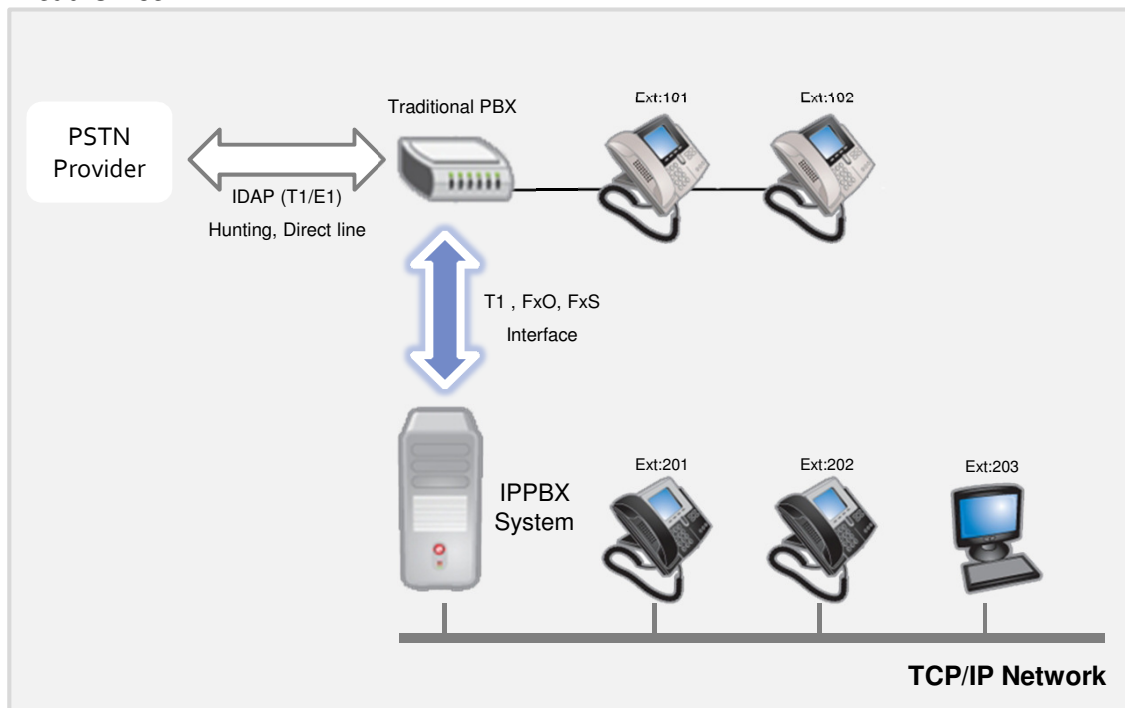


#### Key Benefits

- Single infrastructure for voice and data
  1. High Degree of mobility
  2. Simplified management
- Easy to integrate a wide variety of application on phones
- Soft phone provide flexible , low cost option

Traditional PBX Enhancement

Head Office



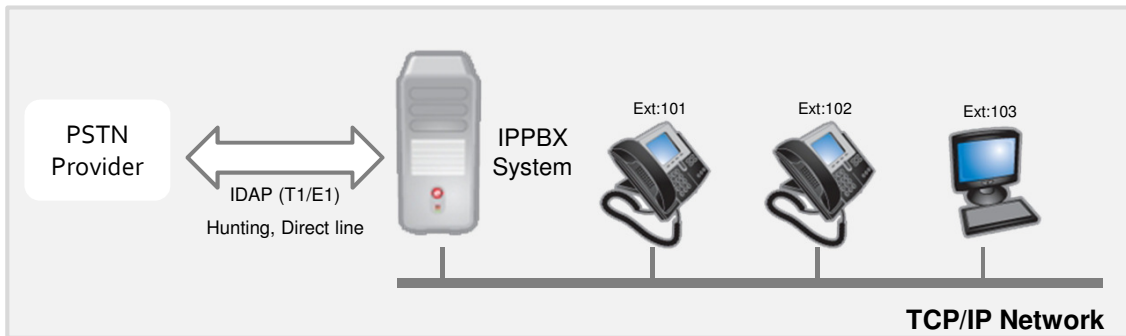
**Key Benefits**

- Enhance the current PBX to IP connectivity
- Break the limitation of extension / Ports capacity
- With a less changeover affair and affect
- Investment protection for upgrade and replace the old system

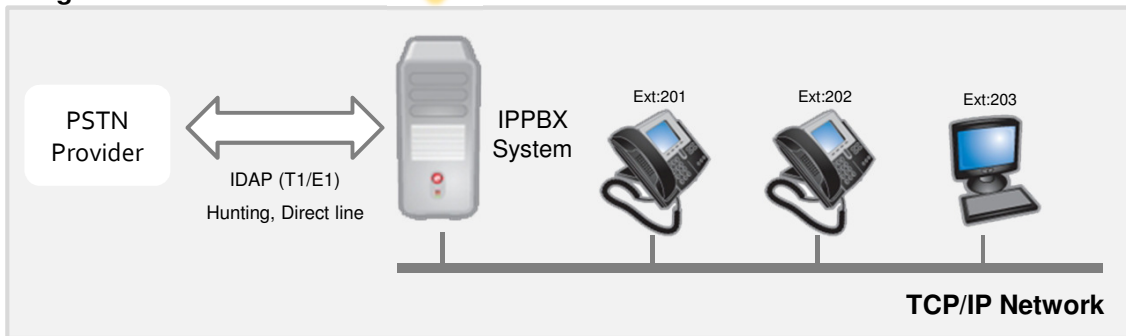
:: Connectivity

Multi-Site Solution

Head Office



Regional Office



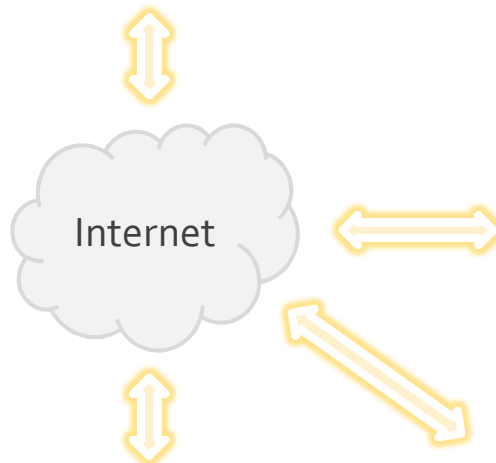
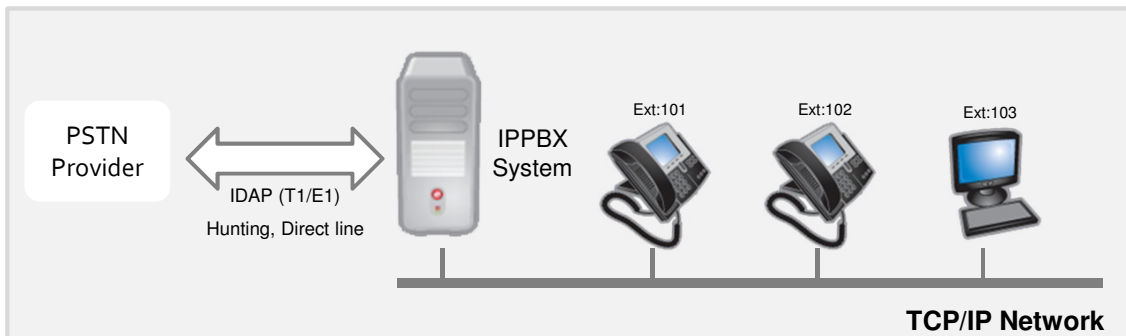
Key Benefits

- Create a virtual enterprise, break the boundary between the Head office to regional office.
- Both office can share the PSTN trunk with others to reduce the IDD cost

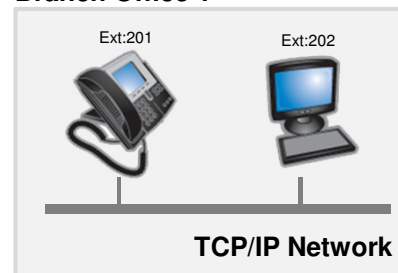
:: Connectivity

Multi-Site Solution

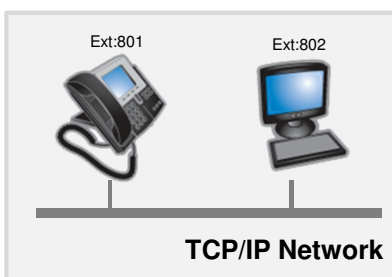
Head Office



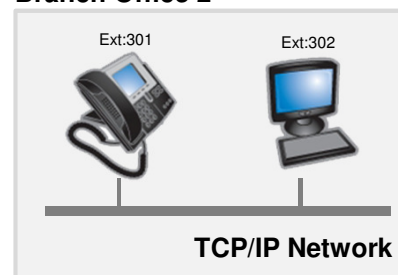
Branch Office 1



Home Office



Branch Office 2



Key Benefits

- Create a virtual enterprise, break the boundary between the Head office to regional office.
- Integrated Voicemail and Voice logging system for ease to manage
- Flexible and mobility for staff or branch office relocation

## :: Feature

Feature	IPPBX
<b>Unified Communications</b>	
<a href="#">Voice Over IP (SIP/IAX)</a>	✓
<a href="#">Conferencing</a>	✓
<a href="#">Call Queues</a>	✓
<a href="#">Send and receive faxes</a>	✓
<a href="#">Video Calling</a>	✓
<b>Extensions</b>	
<a href="#">IP phone extensions</a>	Unlimited
<a href="#">Virtual extensions</a>	✓
<a href="#">Auto-attendants</a>	✓
<a href="#">Set Distinctive Ring Rules</a>	✓
<a href="#">Control permissions for each extension</a>	✓
<a href="#">Extension groups</a>	✓
<b>Calling Methods</b>	
<a href="#">VoIP</a>	✓
<a href="#">Analog phone lines - Hunting Line</a>	✓
<a href="#">T1/E1 phone lines</a>	✓
<a href="#">Connecting multiple IPPBX</a>	✓
<b>Call Control</b>	
<a href="#">Hold</a>	✓
<a href="#">Transfer</a>	✓
<a href="#">Call Parking</a>	✓
<a href="#">Do Not Disturb</a>	✓
<a href="#">Call Forward (All/Busy/Busy)</a>	✓
<a href="#">Directed and Group Pickup</a>	✓
<b>Recording and Monitoring</b>	
<a href="#">Call Recording</a>	✓
<a href="#">Call Monitoring</a>	✓
<b>Administration</b>	
<a href="#">Phone Setup</a>	✓
<a href="#">Disk-space quotas</a>	✓
<a href="#">Access Control</a>	✓
<a href="#">Comprehensive Monitoring</a>	✓
<a href="#">Distinctive Ringtones</a>	✓
<b>Logging and Reporting</b>	
<a href="#">Scheduled Reports</a>	✓
<a href="#">Current Calls</a>	✓
<a href="#">Call Logs</a>	✓
<a href="#">Call Reporting</a>	✓
<b>More Features</b>	
<a href="#">Flexible Language</a>	✓
<a href="#">Dial By Name</a>	✓
<a href="#">Upgradeable Hardware</a>	✓
<a href="#">Organized Phonebook</a>	✓

<b>Voicemail</b>	
<a href="#">Multiple Custom Greetings</a>	✓
<a href="#">Voicemail to your Email Inbox</a>	✓
<b>Conferencing</b>	
<a href="#">Simple Conference Room</a>	Unlimited
<a href="#">Meet Me Conference Center</a>	✓
<a href="#">Listen Only Conference Calls</a>	✓
<b>Paging and Intercom</b>	
<a href="#">2-way Intercom</a>	Unlimited
<a href="#">1-way Paging</a>	Unlimited
<a href="#">Overhead Paging</a>	✓
<a href="#">Direct Paging and Intercom</a>	✓
<b>Music On Hold</b>	
<a href="#">Custom Music on Hold</a>	✓
<a href="#">Music on Hold</a>	✓
<a href="#">Queue specific Music on Hold</a>	✓
<b>Auto-attendants</b>	
<a href="#">Play Sound</a>	✓
<a href="#">Record Sound</a>	✓
<a href="#">Play Recorded Sound</a>	✓
<a href="#">Email Recorded Sound</a>	✓
<a href="#">Record Digits</a>	✓
<a href="#">Say Digits</a>	✓
<a href="#">Say a number</a>	✓
<a href="#">Say date/time</a>	✓
<a href="#">Dial Extension</a>	✓

## :: 10 Reason for MIPPBX

### •Reason #1: Much easier to install & configure than a proprietary phone system

An IP PBX runs as software on a computer and can leverage the advanced processing power of the computer and user interface as well as Windows' features. Anyone proficient in networking and computers can install and maintain an IP PBX. By contrast a proprietary phone system often requires an installer trained on that particular proprietary system!

### •Reason #2: Easier to manage because of web/GUI based configuration interface

An IP PBX can be managed via a web-based configuration interface or a GUI, allowing you to easily maintain and fine tune your phone system. Proprietary phone systems have difficult-to-use interfaces which are often designed to be used only by the phone technicians.

### •Reason #3: Significant cost savings using VOIP providers

With an IP PBX you can easily use a VOIP service provider for long distance and international calls. The monthly savings are significant. If you have branch offices, you can easily connect phone systems between branches and make free phone calls.

### •Reason #4: Eliminate phone wiring

An IP PBX allows you to connect hardware phones directly to a standard computer network port (which it can share with the adjacent computer). Software phones can be installed directly onto the PC. You can now eliminate the phone wiring and make adding or moving of extensions much easier. In new offices you can completely eliminate the extra ports to be used by the phone!

### •Reason #5: Eliminate vendor lock in

IP PBXs are based on the open SIP standard. You can now mix and match any SIP hardware or software phone with any SIP-based IP PBX, PSTN Gateway or VOIP provider. In contrast, a proprietary phone system often requires proprietary phones to use advanced features, and proprietary extension modules to add features.

### •Reason #6: Scalable

Proprietary systems are easy to outgrow: Adding more phone lines or extensions often requires expensive hardware modules. In some cases you need an entirely new phone system. Not so with an IP PBX: a standard computer can easily handle a large number of phone lines and extensions – just add more phones to your network to expand!

### •Reason #7: Better customer service & productivity

With an IP PBX you can deliver better customer service and better productivity: Since the telephone system is now computer-based you can integrate phone functions with business applications. For example: Bring up the customer record of the caller automatically when you receive his/her call, dramatically improving customer service and cutting cost by reducing time spent on each caller. Outbound calls can be placed directly from Outlook, removing the need for the user to type in the phone number.

### •Reason #8: Twice the phone system features for half the price

Since an IP PBX is software-based, it is easier for developers to add and improve feature sets. Most VOIP phone systems come with a rich feature set, including auto attendant, voice mail, ring groups, advanced reporting and more. These options are often very expensive in proprietary systems.

### •Reason #9: Allow hot desking & roaming

Hot desking – the process of being able to easily move offices/desks based on the task at hand, has become very popular. Unfortunately traditional PBXs require extensions to be re-patched to the new location. With an IP PBX the user simply takes his phone to his new desk – No patching required!

Users can roam too – if an employee has to work from home, he/she can simply fire up their SIP software phone and are able to answer calls to their extension, just as they would in the office. Calls can be diverted anywhere in the world because of the SIP protocol characteristics!

### •Reason #10: Better phone usability: SIP phones are easier to use

Employees often struggle using advanced phone features: Setting up a conference, transferring a call – On an old PBX it all requires instruction.

Not so with an IP PBX – all features are easily performed from a user friendly Windows GUI. In addition, users get a better overview of the status of other extensions and of inbound lines and call queues via the IP PBX Windows client. Proprietary systems often require expensive 'system' phones to get an idea what is going on on your phone system. Even then, status information is cryptic at best.

**:: Certified IP Phones**



**SIP – T28P (Executive IP Phone)**

320x160 graphic LCD with 4-level grayscales  
 6 VoIP accounts, Broadsoft validated  
 HD Voice: HD Codec, HD Handset, HD Speaker  
 48 keys including 16 programmable keys  
 BLF/BLA, SMS, Voicemail, Intercom  
 Localized language, XML phonebook  
 FTP/TFTP/HTTP, PnP Auto-provision  
 SRTP/HTTPS/TLS, VLAN, QoS  
 PoE, Headset, 2xRJ45, Expansion module



**SIP – T26P (Advanced IP Phone)**

132x64 graphic LCD  
 3 VoIP accounts, Broadsoft validated  
 HD Voice: HD Codec, HD Handset, HD Speaker  
 45 keys including 16 programmable keys  
 BLF/BLA, SMS, Voicemail, Intercom  
 Localized language, XML phonebook  
 FTP/TFTP/HTTP, PnP Auto-provision  
 SRTP/HTTPS/TLS, VLAN, QoS  
 PoE, Headset, 2xRJ45, Expansion module



**SIP – T20P (Professional IP Phone)**

TI TITAN chipset and TI voice engine  
 3-line LCD(2 x 15 characters and an icon line)  
 2 VoIP accounts, Broadsoft /Avaya/Broadsoft validated  
 HD Voice: HD Codec, HD Handset, HD Speaker  
 31 keys including 9 function keys  
 Voicemail, Intercom  
 Localized language, Local phonebook  
 FTP/TFTP/HTTP, PnP Auto-provision  
 SRTP/HTTPS/TLS, VLAN, QoS  
 PoE, Headset, Wall-Mounted



**EXP 38 IP Phone 38 Key Console**

38 programmable keys each with a dual color LED  
 Daisy-chain 6 modules for 228 programmable keys  
 BLF/BLA, Speed dial, Call pickup, Call park, Intercom  
 Applies to Yealink IP Phone SIP-T28P and SIP-T26P  
 Expansion module(<2) is powered by the host phone







## :: Certified IP Phones



### **VP-2009 IP Video Phone**

TI DaVinci chipset  
 300K CMOS sensor camera  
 7" 800x480 digital LCD, Touch screen  
 H.264 and H.263 video codec  
 Full-duplex speakerphone  
 Online Advertisement  
 2xLAN, 1xUSB, 1xSD  
 A/V out, Headset, PoE(pending)



### **SIP-GW3CM SIP Gateway**

Make/Receive both VoIP and PSTN calls  
 1xWAN, 1xLAN, 1xFXS, 1xPSTN lifeline  
 Support 3 SIP user accounts  
 Support auto-provision via TFTP, FTP and HTTP  
 Support configuration via web browser, console and IVR  
 Support Caller ID/Name display or block  
 Dial plans, dial tone, busy tone, ring back tone, alert tone can be set flexibly



### **IP Phone Wireless Head Adapter**

Full Compatible with Jabra & Plantronics  
 Control phone through wireless headset  
 Plug and play, Easy to use



### **Professional Call Center Headset**

Ultra noise cancelling microphone  
 Quick disconnection cord  
 Ultra light weight only 50g  
 comfortable wearing for all day use  
 330°rotatable microphone boom  
 Pliable steel headband, large size ear pad



### snom 870 - SIP Touchscreen phone

- TFT Touch screen with 24 bit color depth; 10,9cm (4,3 ")
- Intuitive user interface
- 5-way conferencing (bridge on phone)
- 2 x type A USB - WLAN ready
- snom CTI, snom OCS edition ready
- Power over Ethernet
- National Language Support
- XML Mini Browser



### snom Vision - the expansion module

- 16 programmable keys
- High resolution TFT touch screen color display
- 4.3", 272 x 480 pixels, 24 bits color depth
- Ethernet 10/100 Mbps
- USB: 2x type A, Hi/Full/Low Speed host interface
- USB: 1x type B
- Built in Web server



### snom 870 MeetingPoint

- OmniSound® Full duplex broadband sound
- Management of up to 4 external participants
- Recording range up to 30 m<sup>2</sup> room area or 10 participants
- Add-on microphones for a larger range.
- Microsoft Office Communication server 2007 R2 ready
- Multiple SIP registrations



### snom m9 SIP DECT Phone

- Display: 128 x 128 pixels, 65536 colors, backlit
- Li-Ion battery for 20 hours of calls or 100 hours standby
- Range: 50 meters indoors, 100 meters outdoors
- 12 numerical keys, 5 navigation keys, 2 function keys
- Speakerphone on mobile handset
- 8 handsets per base station
- Three-way conference

## :: Other SIP Software



### Softphone (X-Lite)

The new X-Lite is designed to showcase some of the feature rich capabilities available with our commercial softphone such as superior audio and video quality, zero-touch configuration, IM & Presence, and a comprehensive personal address book.



### IPHONE SIP Phone APP (vnet version 4.3)

SIP/VOIP application on iPhone has come to the [iTunes App Store](#). (US\$5.99 Only)

All calls are made directly to the SIP provider, no middle-man like customized proxy involved, so no voice delay and call quality is excellent. iSip 3.0 now supports Apple's push notification service, you can receive the notification message about new incoming call anywhere. You do not need to run the iSip to wait for the incoming call.



### Fring

[fring](#) supports SIP, allowing you to easily make cheap mobile calls to landline and regular cell phone (GSM/CDMA) contacts using the Matrix IPPBX of your choice, even when your mobile phone doesn't support SIP. fring's easy-to-use interface allows you to simply login and logout of SIP services, giving you freedom to choose between providers for each "out" call, depending on the provider's features such as better quality or savings.



### Webphone

Flaphone™ can be used on any computer wherever you are.  
 Incoming calls processing.  
 No need to install any software  
 Calling regular phone numbers at minimal cost, choosing the most convenient IP telephony provider.  
 No NAT and firewall problems.  
 Call history.  
 Video calls.  
 Instant messaging (IM).  
 User availability (Presence service).

## :: Other SIP Device

### SIP Door Entry Control Unit (with Camera)



- ~Buttons dial an extension or group when pressed
- ~Options for single/double buttoned which call different numbers/groups
- ~Backlit buttons
- ~Two internal SPDT relays for controlling a door opener or light etc.
- ~Relays controlled individually by DTMF tones
- ~SIP Protocol
- ~Unit with IP camera
- ~Optional IP camera to view real time video on a computer
- ~Talk to callers, look at callers, then open the door
- ~Camera permanently operational
- ~A range of electronically controlled door opener devices to connect to the door phone
- ~12v power supplies with battery back up option
- ~Exit switches
- ~Can be bundled as complete door entry kits

### SIP Ceiling Speaker



- ~SIP (RFC 3261) compatible
- ~Simultaneous SIP and multicast
- ~Web-based configuration
- ~Paging prioritization
- ~Web-based firmware upgradeable
- ~Autoprovisioning
- ~High efficiency speaker driver
- ~PoE 802.3af Enabled (Powered-over-Ethernet)
- ~DTMF-controlled relay
- ~Line-out connection
- ~Network and manual speaker volume control
- ~Can drive one external analog speaker for greater coverage
- ~User-uploadable ring and alert tones

**BUREAU VERITAS**  
Certification



Certification  
Awarded to

**YEALINK NETWORK TECHNOLOGY CO., LTD.**

7/F HUALIAN BUILDING, NO.580 JIAHE ROAD, XIAMEN CHINA

Bureau Veritas Certification certify that the Management System of the above organisation has been audited and found to be in accordance with the requirements of the management system standards detailed below

STANDARD

**ISO 9001 : 2000**

SCOPE OF SUPPLY

DESIGN, MANUFACTURE AND SALES OF NETWORK COMMUNICATION  
TERMINAL EQUIPMENTS.

Original Approval Date: **7 FEBRUARY 2007**

Subject to the continued satisfactory operation of the organisation's Management System, this certificate is valid until: **29 DECEMBER 2009**

To check this certificate validity please call **852 - 2815 2092**

Further clarifications regarding the scope of this certificate and the applicability of the management system requirements may be obtained by consulting the organisation.

Certificate Number: 206962 Date : 7 FEBRUARY 2007 Lin-hua Jiang  
Deputy Local Technical Manager

For BUREAU VERITAS  
CERTIFICATION (S.A.)  
S.A. using the accreditation  
certificate number 008



008

MANAGING OFFICE ADDRESS:  
BUREAU VERITAS CERTIFICATION  
(Holding) S.A.  
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# CERTIFICATE

IQNet and CQM  
hereby certify that the organization  
Yealink Network Technology Co.,Ltd.

Domicile: Unit 402,502, No.63, Wanghai Road, 2nd Softwork Park, Huli District, Xiamen City, Fujian, P.R.China  
Office Add.: Unit 402, 502, No.63, Wanghai Road, 2nd Softwork Park, Huli District, Xiamen City, Fujian, P.R.China  
Production Add.: Hualian Electric Mansion, No.580, Jiahe Road, Xiamen City, Fujian, P.R.China

Postcode.:361012

is in conformity with

ISO 9001:2008 Standard

This certificate is valid to the following product(s)/service:  
The design, manufacture of the USB Phone, SIP phone series  
products

Issued on: 2010-01-19

Validity date: 2013-01-18

Registration Number: CN-00210Q10329R0M



René Wasmer  
President of IQNet

Zhang Wei  
CEO of CQM



**IQNet Partners\*:**

AENOR Spain AFNOR Certification France AIB-Vinçotte International Belgium ANCE Mexico APCER Portugal CISQ Italy  
CQC China CQM China CQS Czech Republic Cro Cert Croatia DQS Holding GmbH Germany DS Denmark ELOT Greece  
PCAV Brazil FONDONORMA Venezuela HKQAA Hong Kong China ICONTEC Colombia IMNC Mexico Inspecta Certification Finland  
IRAM Argentina JQA Japan KFQ Korea MSZT Hungary Nemko AS Norway NSAI Ireland PCBC Poland  
Quality Austria Austria RR Russia SH Israel SIQ Slovenia SIRIM QAS International Malaysia SQS Switzerland SRAC Romania  
TEST St Petersburg Russia TSE Turkey YUQS Serbia

*IQNet is represented in the USA by: AFNOR Certification, CISQ, DQS Holding GmbH and NSAI Inc.*

*\* The list of IQNet partners is valid at the time of issue of this certificate. Updated information is available under [www.iqnet-certification.com](http://www.iqnet-certification.com)*

## A new mode to communicate



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Fax: 2125 4475

<http://www.hk-matrix.com>

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Fax: (86) 0755 8270 2911