X-SIP (IP PBX)
VOIP Solution

White Paper
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I. Executive Summary

In today's competitive global economy, the advancement for business and technology moving together at an incredible pace, only the storages companies that best utilizes the informational resource and manage their Hi-tech business infrastructure in correct manner will have the competitive to succeed. Managing data and communication becomes a very daunting task as complexities of telecommunication media increases, and companies that excel in internal communication service between head office and branch office will also possess the VoIP solution that best accommodates newer interconnected technologies of today and tomorrow.

With the X-SIP, a true real-time VoIP solution package enable companies to successful communicate with in any media anywhere anytime. What VoIP does is best communication solution to transmit voice conversations and video streaming over a data network using the Internet Protocol and it also helps you cost saving communication oversea.

With the focus on Increase Productivity, Improved Level of Services, Reduce Operating Expenses, and cost saving, the X-SIP is the perfect VoIP solution package for implementing communication services.

II. VoIP Solution

Evolution Communication

In a past of communication between head office and branch office (oversea or province) that must provide service by telephone operator each country. Companies have to payment for telephone operator per month and service time per minute, then companies must to pay much more of moneys.

In the currently, we have the new technology to support communication and cost saving that called "VoIP" (voice over internet protocol). It is a category of hardware and software that enables people to use the Internet as the transmission medium for telephone calls. Voice conversations and video streaming are sent in packets using rather than by traditional POTS circuits. One advantage of VoIP is that the telephone calls over the Internet do not incur a surcharge beyond what the user is paying for Internet access, much in the same way that the user doesn't pay for sending individual e-mails over the Internet.

X-SIP Solution

Today, the technologies are moving together at an incredible pace such as Wi-Fi, WiMax, RFID, etc. We are moving toward a future in which computing will be ubiquitous, woven seamlessly into the fabric of everyday life, VoIP Technology is one technology to focus and will support in the future. X-SIP solution also is best of VoIP solution package for all companies to manage their Hi-tech business infrastructure in correct manner will have the competitive to succeed. It is support all PBX functional and include specialization functional such as Interactive voice response (IVR), Automatic Calls Distribution (ACD), Conference Room, Mobile gateway, Skype gateway, and also can be customization system integrate with the user's database for adapt improved companies system, so it is best managing calls and keep all calls information into database. The system management status of calls, identify the caller, conference calls, miss calls, call blocking, voice messages, and etc.

III. Architecture Framework

Today, as you know the scalability of the network increase you can send/ received data and sound signal through the networking in real-time, so it helps you improved level of service and cost saving.

Open Architecture

VoIP is an acronym for Voice over IP, or in more common terms phone service over the Internet. If you have a reasonable quality Internet connection you can get phone service delivered through your Internet connection instead of from your local phone company. The standard of VoIP technology have 2 standards are H.323 standard and SIP Standard, we might be called another name is "Call Control Technologies", so It is very important for VoIP Technology to be achieved.

H.323 Standard

H.323 Standard is under ITU-T (International Telecommunications Union) Standard. In the first generation, H.323 standard develop for Multimedia Conferencing standard on LAN, but next generation develop for cover VoIP Technology. H.323 Standard can support Point-to-Point Communications and Multi-Point Conferences. H.323 Standard can support hardware many brand name or many vendors then support common working (Inter-Operate).

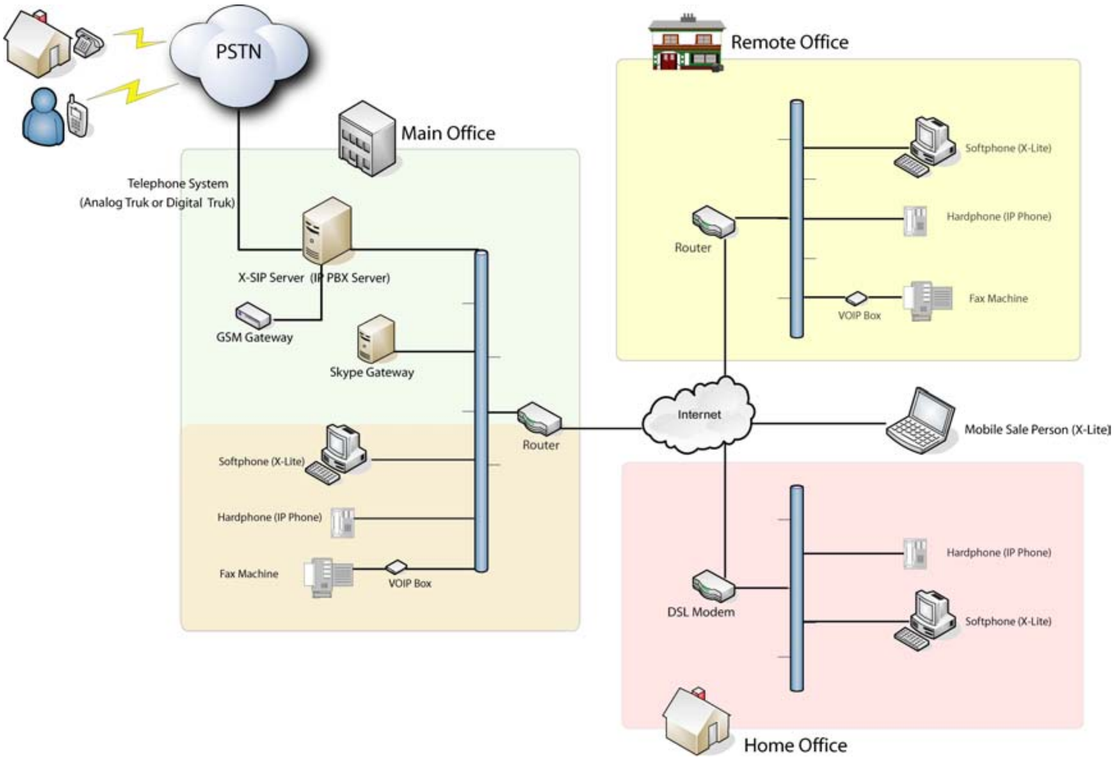
SIP (Session Initiation Protocol) Standard

SIP Standard is new one design used VoIP technology, so SIP Standard special design for support IP networking topology. SIP Standard is IETF Standard, and it Application Layer Control Protocol standard for Creating, Modifying and Terminating of Session. SIP Standard has architecture same as Client-Server Protocol and Strong Reliability

Comparison of H.323 and SIP

H.323	SIP
Complex Protocol	Comparatively Simpler
Binary representation for its messages	Textual representation
Not very modular	Very modular
Not very scalable	Highly scalable
Complex Signaling	Simple Signaling
Hundred of Header	37 Headers
Loop Detection is difficult	Loop detection is comparatively easy

X-SIP Solution



Scalability

In an impermanent economy companies may choose to implement VoIP that meets marginal needs, however it will also have to accommodate future needs as the company grows. The scalability of X-SIP solution package allows for the possibility for future upgradeability. With X-SIP solution package you are able to conveniently upgrade and accommodate various employee sizes to use on system.

X-SIP solution package gives you the capability of integrating and configuration system to able customized solutions and to deploy it for your future need. X-SIP solution package is main management functional, and some functional customize by our team.

The X-SIP is carefully designed for maximum flexibility. Specific APIs are defined around a central PBX core system. This advanced core handles the internal interconnection of the PBX, cleanly abstracted from the specific protocols, codecs, and hardware interfaces from the telephony applications. This allows X-SIP to use any suitable hardware and technology available now or in the future to perform its essential functions, connecting hardware and applications.

The X-SIP core handles these items internally:

- **PBX Switching** - The essence of X-SIP, of course, is a Private Branch Exchange Switching system, connecting calls together between various users and automated tasks. The Switching Core transparently connects callers arriving on various hardware and software interfaces.
- **Application Launcher** - launches applications which perform services for uses, such as voicemail, file playback, and directory listing.
- **Codec Translator** - uses codec modules for the encoding and decoding of various audio compression formats used in the telephony industry. A number of codecs are available to suit diverse needs and arrive at the best balance between audio quality and bandwidth usage.
- **Scheduler and I/O Manager** - handles low-level task scheduling and system management for optimal performance under all load conditions.

Loadable Module APIs:

Four APIs are defined for loadable modules, facilitating hardware and protocol abstraction. Using this loadable module system, the X-SIP core does not have to worry about details of how a caller is connecting, what codecs are in use, etc.

- **Channel API** - the channel API handles the type of connection a caller is arriving on, be it a VoIP connection, ISDN, PRI, Robbed bit signaling, or some other technology. Dynamic modules are loaded to handle the lower layer details of these connections.
- **Application API** - the application API allows for various task modules to be run to perform various functions. Conferencing, Paging, Directory Listing. Voicemail, In-line data transmission, and any other task which a PBX system might perform now or in the future are handled by these separate modules.
- **Codec Translator API** - loads codec modules to support various audio encoding and decoding formats such as GSM, Mu-Law, A-law, and even MP3.
- **File Format API** - handles the reading and writing of various file formats for the storage of data in the file system.

Using these APIs X-SIP achieves a complete abstraction between its core functions as a PBX server system and the varied technologies existing (or in development) in the telephony arena. The modular form is what allows X-SIP to seamlessly integrate both currently implemented telephony switching hardware and the growing Packet Voice technologies emerging today. The ability to load codec modules allows X-SIP to support both the extremely compact codecs necessary for Packet Voice over slow connections such as a telephone modem while still providing high audio quality over less constricted connections.

The application API provides for flexible use of application modules to perform any function flexibly on demand, and allows for open development of new applications to suit unique needs and situations. In addition, loading all applications as modules allows for a flexible system, allowing the administrator to design the best suited path for callers on the PBX system and modify call paths to suit the changing communication needs of a going concern.

IV. System Component

X-SIP solutions offer a rich and flexible feature set. The one offers both classical PBX functionality and advanced features, and interoperates with traditional standards-based telephony systems and Voice over IP systems. The one offer the features one would expect of a large proprietary PBX system such as Voicemail, Conference Bridging, Call Queuing, and Call Detail Records. X-SIP has much more features to support all functional, so it have main component as below.

PBX Switching

Part of component is basic function of PBX machines such as Roaming Extensions, Transcoding, Trunking, DNIS and etc, which automates the delivery calls to end-users.

Call Management

Part of component is Call management function, which support management calls on system such as Blacklist, Blind Transfer, Call Forward on Busy, Call Forward on No Answer, Call Monitoring, Call Parking, Call Recording, Call Routing (DID & ANI), Call Transfer, Call Waiting, Caller ID, Caller ID Blocking, Caller ID on Call Waiting, Dial by Name, Conference Bridging, Flexible Extension Logic, Route by Caller ID, and etc.

Fax Transmit and Receive

Part of component is Fax Transmit and Receive function, which help to automate the delivery information out to Fax, however you have to install 3rd Party tool (OSS Package) run successful.

Interactive Voice Response (IVR)

Part of component is the Interactive Voice Response (IVR) system, which automates the delivery of information or pre-recorded messages to callers thus improving customer service response time, to almost instantaneous. Such information is especially vital, when it comes to decision making and responding to market demands. The components enable customization delivery of information from database.

Music On

Part of component is music on hold calls and transferring calls, which can be support MP3 file format. The library music to playing can choose normal play or random play. The music volume also can set by user.

Short Messages (SMS)

Part of component is short messages (SMS) function by integration with user system and automates sending text messages through SMS Gateway public service to mobiles, thus improving customer service or internal service.

Text& Video Streaming

Part of component is text& video streaming, which support data even those voice formats to send/received from source to destination such as video streaming or text real-time thus improving internal communication.

Voice Messages

Part of component is management voicemail, which automates alert messages waiting to users and user stutter dial tone for messages waiting, voicemail to email, and voicemail groups.

Voice Gateway

Part of component is voice gateway function, which enhance improving system.

Billing

Part of component is billing, which report users to know more details of outbound calls, thus to help users easy to management cost of outbound calls.

Reporting

Part of component is System Reporting, which support report users to management over all of system such as report inbound calls, outbound calls, and etc.

Computer-Telephony Integration

Part of component is Integration, which to improved system feature by support integration with other system such as CRM, ERP, etc.

V. Benefit of X-SIP

1. Cost Saving
   Our solution give you benefit to adapt use with current network system such as Router, Switch Hub, etc. However, if you choose our solution will help cost saving of calling cross oversea country or cross province in same country such as call free between head office and branch office, call free in local country, and cost saving 2 baht per minutes to call roaming to international country.

2. Increase Productivity
   Our solution give you benefit to use full capacity of current network system productivity and current telephone system productivity such as Router, Switching hub, PBX, etc.

3. Improved Level of Services
   Our solution give you benefit to improved branch interchange information not serious cost budget communication between branches, thus each branch will get latest information in real-time to improve the operation level and service.

4. Reduce Operating Expenses
   Our solution give you benefit to save cost payment for employer to manage and monitor telephone system because currently your branch will have 1 person to monitor and manage the telephone system, but if you choose our solution you can monitor and manage by 1 person from central office.

5. Easy to Install and Use
   In today's competitive global economy, the advancement of business and technology is moving together at an incredible pace, Wi-Fi technology also improving at an incredible and some companies has implemented Wi-Fi technology adapt use with operation, thus if your company has already implemented Wi-Fi Technology you can use our solution with IP Phone (Wi-Fi), so it very easy to install and use by no need wiring cable and improved service everywhere.