X&YComm

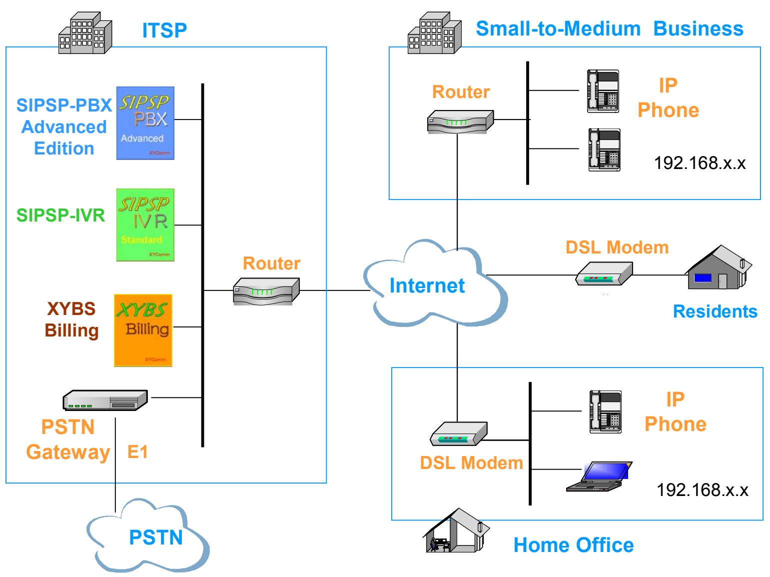
SIPSP-PBX Advanced Edition
---- High Scalability, High Availability, Full-Featured

Data Sheet

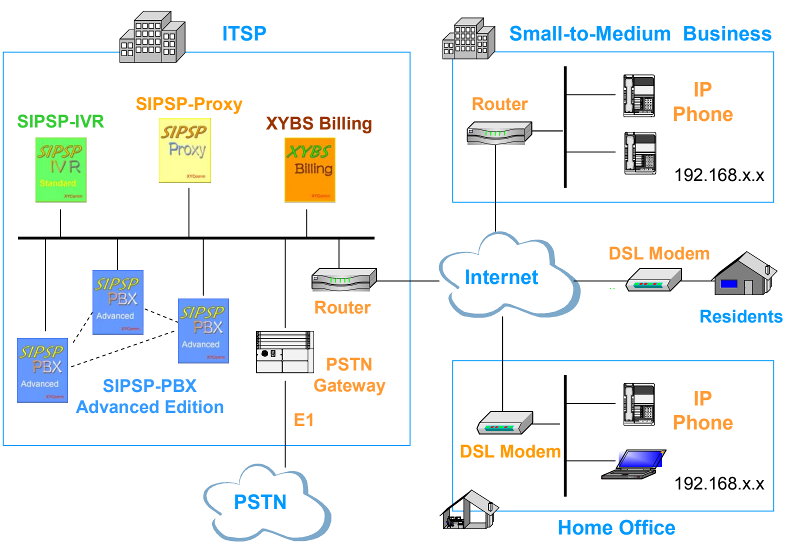
X&YComm SIPSP-PBX is a SIP-based IP-PBX software running on ordinary X86 PC and Windows operating system. Written in pure C language, SIPSP-PBX has proven high performance and stability, with very few consumption of system resources.
SIPSP-PBX Advanced Edition, targeting at **IP Telephony Service Provider**, is a highly powerful and scalable edition with full features including billing, load balance, call routing, and rich call features. Together with **SIPSP-IVR** and **XYBS**(billing system), a whole solution for ITSP is perfect and ready for services.

Solution

A Minimum Solution for Startup VOIP Services



A Scalable Solution for Large Scale VOIP Services



Features

➢ High Capacity & Scalability

At least 2000 concurrent online users are supported on a single ordinary P4 PC server.
Easy to expand to much larger scales, just deploy arbitrary number of SIPSP-PBX servers with a central SIPSP-Proxy server, and make each SIPSP-PBX register to the central SIPSP-Proxy, which automatically routes calls between different SIPSP-PBXs, without complicated point-to-point manual settings.

➢ High Availability

Load balance is available to allow a cluster of arbitrary number of SIPSP-PBX servers to share dynamic running status and backup for each other.
The SIPSP-PBX software is worldwide proved capable of running stably and uninterruptedly without any care. Furthermore, an additional failover watchdog software SIPSP-Startup keeps monitoring the SIPSP-PBX software less than every one minute and will automatically restart it whenever abnormal.
Immediate resumption is available after restarted, no re-registering for online users.

➢ Billing Features

Detailed CDR records are stored to database which can be shown on the web monitoring pages of SIPSP-PBX.
Radius client is also supported to report Radius Authentication, Authorization & Accounting Requests to Radius servers, containing both standard Radius attributes and necessary Cisco VSA attributes, thus it is easy to work with almost any kinds of Radius servers, including **X&YComm XYBS**. Both prepaid & postpaid billing modes are supported.
Together with X&YComm billing system **XYBS**, which is focused on VOIP billing requirements and also supports Cisco VSA, a complete billing solution is provided for ITSP.

➢ Easy Setup and Maintenance

A step-by-step wizard is provided to implement integrated installation and basic settings of all the required components, including SIPSP-PBX software & web pages, MYSQL/MSSQL database, JavaSDK, Tomcat, etc.
Web-based configuration and monitoring are supported, as well as Telnet console, which make the whole system easily under control and management.

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➤ Intelligent NAT Solutions

SIPSP-PBX provides intelligent NAT traversal and RTP relay according to the actual network status and per user settings. Also SIPSP-PBX itself can be located in either public or private networks, supporting various network situations such as DMZ, Dynamic DNS, Port Forward etc.

➤ Rich and Customizable Call Features

SIPSP-PBX supports rich call features like Call Forward, Call Transfer, Call Block, Call Screening, Fax, Instant Message, etc.
When working with **SIPSP-IVR Advanced Edition**, more features are available like ACD, Voicemail, Customer Self Care, Echo-Test etc.
More customized features can be easily defined and supported just by adding new feature description data.

➤ Powerful Inward / Outward Call Routing

SIPSP-PBX Advanced Edition provides various means to support flexible call routing policies for both outgoing and incoming calls, including powerful Dial Plan, Trunk users, ACL, and ACD (together with **SIPSP-IVR**), etc.

➤ Tested Interoperability

IP Phones

Cisco Grandsteam Budgetone, Linksys PAP, PA168s-based IP Phone, SIPURA, Snom, Zyxel

Soft Phones

Nortel SIP Client, SJ-Phone, X-lite

Gateways

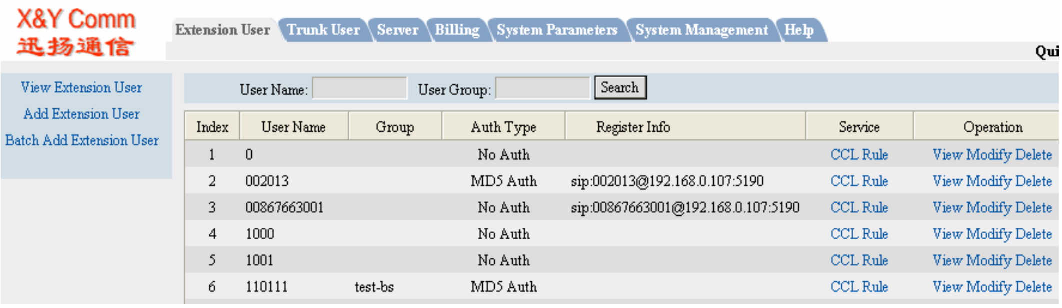
Audiocodes MPxx, Cisco 53xx/26xx/17xx/ATA18x, Netgear, Quantum, Tainet, Welltech

Grandstream Handytone, Linksys PAP, Octel, SIPURA, Vega,

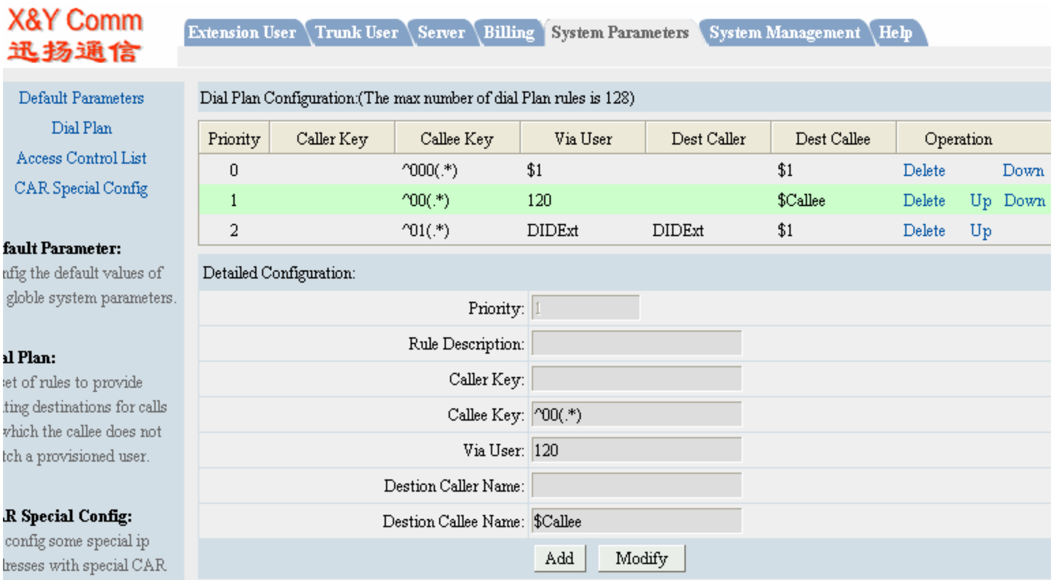
VOIP ISPs

Broadvoice, FWD, Vonger, etc.

WEB Configuration



Extension User Accounts Configuration



Dial Plan Configuration

Specifications

Hardware/ Software	Hardware：PIII 1G/1G HD/128MB Memory at least OS：Windows 2000 / Windows XP / Windows 2003
Protocol	SIP：RFC3261 / RFC2543 , UDP/SIP-URL, Configurable SIP Port
Detailed Extension User Control	Multiple Register Modes: **Dynamic / Static / No Register** Maximum 8 Registration Per User Account Authentication Modes: **MD5 / No Auth** Configurable NAT Traversal Modes Per User : **Off / On / Auto** Configurable RTP Relay Modes Per User : **Off / On / Auto** Maximum Concurrent Calls Restriction Per User Maximum 100 Call Feature Rules Per User
Trunk User	Support Multiple Trunk Users Support Registering to Other SIP Servers Support Routing Calls to / from Other SIP Servers Flexible Inward/Outward Call Feature Control for each Trunk User
Call Routing Inward/ Outward	Flexible Trunk User and Dial Plan settings for Outgoing Calls Flexible Trunk User , ACL and ACD feature (with SIPSP-IVR) for Incoming Calls Maximum Dial Plan Rules: 128
Billing Features	Detailed CDR Records Stored in Database and Shown on the Web Pages of SIPSP-PBX Support Radius Client with Features as Below : --- Report Radius Authentication, Authorization & Accounting Requests to Radius Servers --- Support Both Standard Radius Attributes and Cisco VSA Attributes --- Support Both Prepaid and Postpaid Billing Mode --- Support Multiple Primary / Secondary Radius Servers --- Support Auto Radius Server Availability Detection, Switch & Resumption Provide our own **XYBS** Billing System to Build a Complete Billing Solution
Intelligent NAT/Firewall Traversal	SIPSP-PBX Can be placed in either public or private networks Support DMZ, Dynamic DNS, Port Forwarding Support Configurable NAT Traversal Mode Per User Support Configurable RTP Relay Mode Per User
Rich Features	Call Forward All , Voicemail, (*with **SIPSP-IVR**), Call Forward Busy, ACD, (*with **SIPSP-IVR**) Call Forward No Answer, Echo-Test (*with **SIPSP-IVR**) Call Transfer, Customer Self Care (*with **SIPSP-IVR**) Call Hold, FAX (T.38), Call Screening, Instant Message, Call Blocking, Multimedia (Voice, Video, Data) Services, Emergency Call, Customized Call Features Anonymous Call,

(To be continued)

Specifications Continued:

Run Mode	Support NT Service Mode Support Normal Windows Program Mode
Easy Setup & Maintenance	Step-by-Step Installation Wizard for Integrated Installation & Settings of **ALL** Required Components : --- SIPSP-PBX Software & Web Management Pages --- MYSQL / MSSQL Database --- JavaSDK --- Tomcat Provide WEB-based System Settings Provide Both WEB-based and Telnet-based System Remote Monitoring Provide System Log Record, with Configurable Log Level
High Capacity & Scalability	Max Concurrent Online Users Per Single Ordinary P4 PC Server : at least **2000** (NO Restriction for Configured User Accounts and Concurrent calls) Easy to Support Much Larger Scales as Below: --- Deploy Arbitrary number of SIPSP-PBX Servers and a Central SIPSP-Proxy Server --- Let SIPSP-PBXs Register to the SIPSP-Proxy Server --- SIPSP-Proxy Server will Auto Route Calls between Different SIPSP-PBXs
High Availability	Load Balance -- Allow Arbitrary number of SIPSP-PBX Servers Backup for Each Other SIPSP-PBX Software Itself Running Stably and Uninterrupted without any Care Provide Additional Failover Software SIPSP-Startup to Monitor & Restart SIPSP-PBX Immediate Resumption after SIPSP-PBX Restarting, no Re-Registering for Online Users
Performance	Very High Performance. Totally rely on your hardware.