IPPBX Intelligent Convergence System

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I. Product Overview

The IPPBX Intelligent Convergence System is a smart integrated communication system developed based on the actual needs of various industries and organizational users. It integrates traditional TDM hard-switching technology and IP soft-switching technology, adopting the "IP&TDM+IOT+AI" design philosophy to create a new generation of IPPBX systems. It provides users with a simple-to-use, easy-to-install, reliable, and technologically advanced integrated communication solution.

On one hand, this IPPBX Intelligent Convergence System can connect to traditional telephones, fax machines, and traditional analog PBX to provide high-quality voice services. On the other hand, it adopts the standard SIP protocol, making it compatible with mainstream IPPBXs, IMS softswitch platforms, and SIP-based network platforms. Additionally, the system comes with optical ports, enabling fiber-optic remote extension to easily address deployment challenges across multiple floors or buildings. The system also supports intelligent voice recognition features, automatically recording and analyzing calls (including speech-to-text conversion, emotion analysis, key point analysis, service attitude evaluation, response speed assessment, etc.). Furthermore, it offers open secondary interfaces for customization and integration with users' third-party systems.

II. Functional Advantages:

Main Features

- Safe and Reliable: Domestic CPU + Linux system, industrial-grade embedded system, independent operation, safe and reliable.
- Supports dual-system hot backup, automatic switching between primary/backup IPPBX servers in case of failure.
- Rich interfaces: Supports E1/FXO/FXS/SIP/IP phones/gateways, etc.
- External line grouping: One device can be used by multiple departments, supporting external line grouping, allowing each department to use their own external lines without conflict.
- Internal line grouping: Multiple staff members in a department can be set as a port group for incoming calls (enabling group ring, sequential ring, and circular ring).
- Call Queuing: When all agents are busy on incoming calls, the system can notify callers that there are xx people ahead in the queue, with an estimated wait time of xx.
- Intelligent IVR: The system comes with multiple IVR voice segments and can intelligently prompt IVR guidance based on different business departments to enhance the business image.
- Adopts standard SIP protocol, perfectly compatible with IMS/NGN and mainstream softswitch platforms.
- Supports multiple trunk access methods, including analog trunk, SIP trunk, E1 digital trunk, etc.; supports SNMP/TR069 standard network management protocols.
- Supports various terminal access methods, such as IAD voice gateways, SIP phones, analog phones, etc.
- Call Recording: The system supports full-channel call recording for the entire duration, with all calls being recorded; automatically generates WAV format and other formats of recording files, which can be stored locally on the device or on a remote server; through the WEB page and other interfaces, users can remotely query, play, and download recordings at any time.
- Call Detail Record Query: The system can provide all users' call records and account recharge records, which can be queried in the background. It also supports retrieving users' call records at any time via user codes.
- Security Prevention: Supports multiple users accessing simultaneously, and allows restricting user access through various methods such as IP address/IP address range.
- Email Push: Real-time monitoring of device status, with immediate email notifications for any anomalies to alert management personnel.
- Call Detail Record Statistics: Can be analyzed, compared, and summarized by account or extension number.
- Operation logs are not lost; all operations on the device are recorded and can be queried through the backend.
- The system can be connected to a local area network via an Ethernet port, allowing effective remote management of the device through management tools or web pages. It even supports maintenance and management via external networks.

- Provides secondary development interfaces, supports secondary development, and can develop new features to integrate with third-party systems according to different customer needs.

Voice Features

- Voice Codecs: G.711a/μ law, G.723.1, G.729A/B, G.726；
- Silence Suppression;
- Comfort Noise Generation (CNG);
- Voice Activity Detection (VAD);
- Echo Cancellation (G.168), maximum 128ms;
- Dynamic Jitter Buffer;

VoIP Protocols

- Protocols: SIP V2.0 (UDP/TCP), RFC3261, SDP, RTP(RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311;
- RTP/RTCP, RFC2198, RFC1889;
- SIP over TLS;
- RFC4028 Session Timer;
- RFC3266 IPv6 in SDP;
- RFC2806 TEL URL1;
- RFC3581 NAT.rport;

Software Features

- Access Methods: Web, Serial Port;
- Call Policies: Call Routing, Digit Mapping, Routing Strategy;
- Number Transformation: Caller Number Transformation, Callee Number Transformation;
- Call Types: Internal Line, Local Call, Long Distance, International Long Distance, Various Custom Types;
- Billing Rules: Multi-Time Period, Multi-Rate, Hybrid Billing Algorithms;
- Permission Control: Extension Levels;
- Number of servers: Supports 4 IMS platforms with automatic primary/backup IP switching;
- Remote networking: Supports networking for more than 16 points simultaneously;
- Automatic deployment: Fully automatic deployment for various well-known brands of IP phones and voice gateways;
- Log management: Operation logs, runtime logs, user logs, and security logs;
- Alarm Management: Real-time alarms, historical alarms, email notifications;
- Local Management: Graphical interface configuration, version upgrade/rollback, data backup and restore, license management;

Partial Functions

- Call Forwarding (Unconditional Forward, No Answer Forward, Busy Forward);
- Call Waiting, Call Hold;
- Call Transfer (Blind Transfer, Consultative Transfer);
- Group Pickup, Dual Number on One Device, Fixed-Mobile Integration, Internal and External Transfer;
- Hotline, Abbreviated Dialing;
- Wake-up service, Do Not Disturb;
- Three-way calling, conference calling;
- Call Forced Insertion, Call Forced Disconnection;
- Voice navigation, automatic callback;
- Voice dedicated line, alternate conversation, selective access, call queuing;
- Call billing, call statistics, prepaid deposit, level control;
- Automatic deployment, one-click update;
- Emergency assistance, one-click alarm;
- Blacklist and whitelist, distinction between internal and external line duplicate prefixes;
- Phone directory, Chinese display;
- Ring-back tone, call message, voice mailbox, call recording;
- Port group (trunk group): group ring, sequential ring, cyclic ring;
- Intelligent work schedule, trunk group;
- Call time limit, smart access control;

Management and Maintenance

- Web management configuration interface;
- Automatic upgrade/configuration;
- SNMP V1/V2/V3;
- TR069;
- Configuration Backup/Restore;
- HTTP/TFTP/FTP Firmware Upgrade;
- Call record query and export;
- Sys log query and export;
- Ping/Tracert test;
- Line diagnosis (GR909);
- NTP/Daylight Saving Time;
- IVR Voice Maintenance;
- Cloud Centralized Maintenance;
- Remote Web Functions (Reliable Transmission, Upload/Download Capability);

III. Product Appearance

SPC6600 HX20 Series

(Rack-mounted/Hard Disk Type, Maximum 256 IP Users and 80 Analog Interfaces (FXS/FXO))

SPC6600 RXX50 Series

(Rack-mounted/Hard Disk Type, Maximum 512 IP Users and 80 Analog Interfaces (FXS/FXO))

SPC6600 RXX100 Series

(Rack-mounted/Hard Disk Type, Maximum 1024 IP Users and 80 Analog Interfaces (FXS/FXO))

IV. Model Description

Parameter	SPC6600 HX20 serials	SPC6600 RXX50 serials	SPC6600 RXX100 serials
IP Extension	256	512	1024
FXS	80	80	80
FXO			
SIP Trunk	256	512	1024
E1(NO.1/NO.7/PRI)	0	2	4
Analog Concurrent Calls	16	32	32
IP Concurrent Calls	64	128	256
Storage Medium	Hard Disk		
Operating Temperature	0-50°C		
Hardware Performance	Processor: 2-core 1.2GHz, ROM: 8GB, RAM: 1GB		
Operating Humidity	10%-95% (non-condensing)		
Storage Temperature	-20 - 80		
Power Supply	AC 220V / DC 48V		
Equipment Weight	≤7KG		
Total Power Consumption	≤50W		
Dimensions	440mm*475mm*45mm (Equipment Dimensions) 510mm*450mm*115mm (Packaging Dimensions)		

V. Physical Characteristics

Physical Interfaces

- FXS Interface RJ45;
- Network Interface RJ45;
- E1 Interface RJ45;
- FXO Interface RJ45;
- Serial Interface RJ11;
- SD Interface SD card;

FXS Parameter

- Interface Type: RJ45;
- Dialing method: DTMF and pulse dialing;
- Pulse dialing: 10 and 20 PPS;
- Caller ID: DTMF/FSK caller ID standard;
- Wiring length: 3 kilometers (custom orders can support up to 10 kilometers);

Operating Environment

- Operating Temperature: 0 °C ~ 50 °C;
- Storage Temperature: -20 °C ~ 80 °C;
- Operating Humidity: 10% ~ 95% (Non-condensing).