1. H20-20 PABX System

Product Overview

The H20-20 PABX is an integrated voice/digital exchange system. It is widely used for fields such as private network, public network and integrated service platform. This system is unique among similar products, becomes a famous brand in the communication industry and wins the trust of users at home and abroad due to excellent performance, flexible design and powerful functions. At present, there have been more than 20,000 sets of H20-20 PABX system under operation in the world, including more than 3,000 sets in China. In addition, there have been more than 20 years of stable operation records in China. This system serves the electrical power system, industrial and commercial institutions, government agencies, public institutions and some confidential departments.

System Characteristics

- **High-performance central processing unit (CPU)**

  The new-generation CPU uses the Intel processor. Large-capacity internal storage and high-reliability solid electronic hard disk realize a leap of system performance.

- **Public control devices with full redundancy configuration**

  All the public control devices such as CPU, system clock, exchange network, ringing current unit, signal tone unit, conference mixed tone unit and power supply are with full redundancy configuration and double-set hot-backup to ensure uninterrupted system operation.

- **Unblocked exchange network structure**

  The exchange network of this system uses unblocked structural design. To be specific, every port can bear the telephone traffic of 1 ERL no matter what subscribers or trunks.

- **High-telephone-traffic processing capacity**

  The value of Busy Hour Call Attempt (BHCA) is 1,000,000 times per hour.

- **High-reliability design**

  The cabinets and shelves are under independent power supply. Both the primary and secondary power sources can be with double-set backup configurations to ensure the reliability of power supply. The circuit boards are based on electrical safety design and can realize hot plugging.

- **Powerful networking capacity**

  It is rich in interfaces and complete in signaling. It possesses various analog and digital interfaces, supports various general or private signaling methods and possesses powerful networking functions and signaling coordination capacity.

  It is equipped with a high-precision clock, possesses strong network synchronizing capacity and adapts to multiple network synchronizing methods.

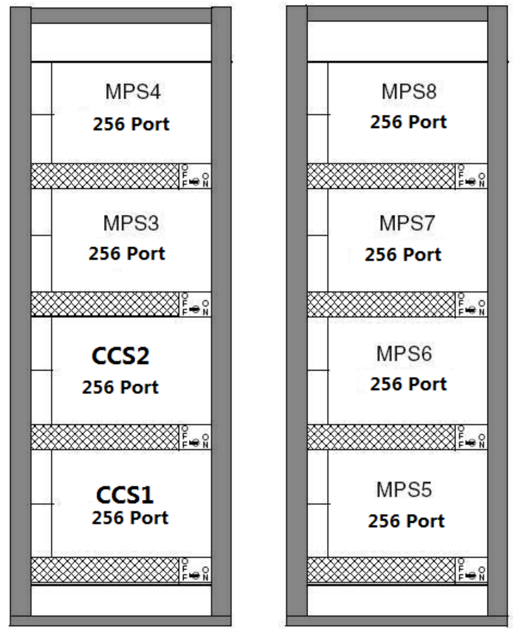
- **Complete system functions**

  It possesses PSTN service and complete PBX functions, including various characteristic functions, convenient to realize different solutions, adapt to administrative communications, dispatching networks, other private networks, hotels, confidential departments, call centers and other value-added services.

- **Hardware/software modular design**

  Building-block-type system structure provides flexible configuration, convenient capacity dilatation to ensure reasonable initial investment and feasible subsequent development.

H20-20 CCS PABX System



The maximum capacity of the CCS System is 2048 ports. The CCS System is made up of 8 layers of shelves at most. Layers 1&2 are two main control shelves (CCS) and Layers 3 to 8 are subscriber shelves (MPS). All the shelves can be configured with two power modules (PSM) used for redundant power source. Every rack is power-supplied with -48VDC. The CCS System can use redundant or non-redundant public control configuration as per application requirement.

System Parameters

- System capacity
  - Maximum port capacity: 2048
  - Signal tone and conference port: 512
  - Telephone trunk line: 1 ERL
  - Telephone subscriber line: 1 ERL
- External dimensions
  - Single main control shelf (CCS) or subscriber shelf (MPS):

    Height: 40cm

    Width: 58.4cm

    Depth: 61cm

    Ground load: 288kg/ m^2

Note: It uses 19" standard EMC cabinet. Every cabinet can hold 4 shelves at most. The weight of cabinet varies from the quantity of port boards installed in cabinet.

- Requirements for power sources
  - Primary power source:

    Noise: < 40dB in-band

    Wave fluctuation: 600mV out-band peak

    Working voltage: -43VDC~-46VDC
  - Maximum power consumption at full load:

    Single shelf: 300W
  - Distribution of power supply for cabinet

    Feeder voltage of bus: -48VDC

    Ringing current: 75/90VAC (adjustable), 25Hz

    Feeder voltage of subscriber line: -48VDC
- Requirements for subscriber line:
  - Loop resistance of subscriber: 2000 Ω (including the telephones) (loop current: 18mA, -48VDC)
  - Insulation resistance among subscriber lines: 20kΩ
  - Capacitance among subscriber lines: 0.7Mf

2. Softswitch platform

2.1 SW9000 Series of Softswitch Core Server

G2S SW9000 series of soft-switching core servers adopts the international advanced software and hardware technologies, which has abundant application capacities and powerful networking capabilities, and is mainly used in NGN network control layer. SW9000 series cover a variety of specifications, meeting the needs of different industries users.

SW9100 soft-switching core servers

SW9500 soft-switching core servers

Features

Large Capacity and High Performance

- The maximum capacity of the system is 100,000;
- BHCA value is greater than 1.44 million, and the call loss is less than one hundred thousandth;

High Reliability Design

- Supporting hot backup redundancy, the change-over time for the application software is as low as 10 milliseconds, the change-over time for system switching is within 15 seconds;
- Supporting the solution of load sharing of high efficiency strategy, which further enhances the usability of the Softswitch system.
- Supporting dual-homing function, which improves the system robustness, reliability and network disaster recovery capabilities.
- Supporting dual power input.

Rich Application Provision Capacity

- It fully inherited the voice application capacity of GHT H20-20 switch, which not only provides various basic voice application and supplementary application that meet the international standard, but also provides various customized new application for users in a variety of industries.
- It supports the dispatch application, allowing cross-network grouping of the dispatch consoles on two sides of the IP and PSTN, where the grouping information is shared, and allowing seamless connection with the traditional circuit dispatch system;
- Providing a complete IP Centrex solution that supports IP Centrex user services.
- Supporting T.38 format for transparent transmission in the packet domain, allowing providing users with high-quality end-to-end fax communication services.
- Supporting SIP and other multimedia communication protocols, allowing providing multimedia services such as video phone, application sharing, electronic whiteboard, and video conferencing.
- Under the support of the application server, it provides the value-added application integrated with voice, multimedia, and Internet and services, such as UC application, IM application, and Presence application.
- It has a complete billing capacity, which not only supports the billing for various applications such as voice, data and multimedia, but also provides a variety of billing methods and bill types; moreover, it can provide complete call list management functions.
- It provides excellent telephone traffic statistics (application statistics) function, supporting a variety of measurement indicators and flexible measurement tasks in high real-timeliness manner, which can fully reflect the equipment application load information and operating conditions.

Powerful and Flexible Networking Capabilities

- It supports H.248 media gateway control protocol, allowing docking with gateway devices such as IAD, AMG, and TMG, and allowing the access by the terminal devices such as H.248 packet terminals.

- It supports SIP/SIP-T protocol, so it not only enables interoperability with other Softswitch and SIP application server, but also allows the direct access by the SIP packet terminals.
- It supports MTP and ISUP protocols, allowing docking with devices such as SP and STP in No.7 signaling network, which also provides No.7 trunk for docking with PSTN switch.
- It supports M3UA protocol, allowing not only being used as a signaling point (docking with SG), but also being used as an integrated signaling gateway device.

It supports the RADIUS protocol, allowing the access to the billing server.

Convenient and Practical Operation and Maintenance Functions

- The management adopts flexible and diversified ways, so the user can construct the network management network with flexibility based on the network composition, management needs and investment scale.
- It provides complete functions including call tracking, signaling tracking, interface tracking and message interpretation, providing users with powerful fault analysis and positioning capabilities.
- It supports real-time fault management. The system receives and displays the fault report on the network device in real time, so that the maintenance personnel can quickly and accurately diagnose the fault source and take corresponding measures to restore the normal service.
- It supports the functions of online software patching, upgrading, online debugging, remote maintenance, and data dynamic settings.

Product Specifications

Model / Parameter	SW9112	SW9113	SW9500
Platform type	Special server platform		ATCA platform
Platform specification	2U	2U	3U
Support redundancy or not	√	√	√
Support dual-homing or not	√	√	√
Maximum number of registered users	5,000	20,000	100,000
Maximum number of concurrent users	1,000	2,000	10,000
Maximum number of gateways	100	200	300
Maximum number of application servers	50	100	200
Call processing capability (CAPS)	200	300	400
Telephone Traffic load (BHCA)	720,000	1,080,000	1,440,000
Maximum recovery time	Less than 3 minutes		
Maximum failure time per year	Less than 3 (99.999%) minutes		
Call setup time	In-domain users calling in-domain users <60ms; In-domain users calling PSTN users <120ms; PSTN users calling in-domain users <120ms; PSTN users calling PSTN users via Softswitch <170ms		
Working power supply	AC220V dual redundancy (Standard) DV-48V dual redundancy		DC-48V dual redundancy (Standard) AC220V dual redundancy

2.2 SW8100 Series of Integrated Platforms

SW8100 series of integrated dispatch communication platforms is a comprehensive platform released specifically for users of small capacity (512). The series includes two product models, which are SW8101 and SW8102, the difference between them is the support of high-definition video application or not.

SW8100 series of integrated dispatch platforms

Features

- High integration design, with functions of core switching, multimedia application, and dispatching application;
- Supporting hot backup redundancy, the change-over time for the application software is as low as 10 milliseconds, the change-over time for system switching is within 15 seconds;
- Supporting the solution of load sharing of high efficiency strategy, which further enhances the usability of the Softswitch system.
- Supporting dual-homing function, which improves the system robustness, reliability and network disaster recovery capabilities.
- Built-in WEB-based management unit, providing functions of configuration, fault management, and telephone traffic statistics.

Product Specifications

Model / Parameters	Integrated dispatch communication platform	
	SW8101	**SW8102**
Max user accounts	512	512
Max links of trunks	4	4
Max dispatch groups	3	3
Max dispatch consoles	16	16
HD video conferencing support	-	Yes
Max members per conference	16	16
Max voice conferences	16	16
Max voice conference members	64	64
Video resolution	D1	720P
Redundancy support	Yes	Yes
Power supply	AC 220V/DC -48V	AC 220V/DC -48V

3. Application server

3.1 MCS5501 HD Video Conference Server

MCS5501 is a HD video conferencing server released by GHT, which provides customers with high-definition video services to upgrade the original standard definition conference to 720P HD conference.

MCS5501 has advanced architecture design and powerful media processing capabilities, which enables 720P high-definition multiple-screen, and supports SD and HD terminal mixed-access conference. MCS5501 provides powerful conference functions, flexible and efficient meeting management modes, and easy and convenient conference applications

Features

- Supporting high definition, standard definition mixed meeting;
- Supporting standard SIP protocol;
- Supporting high-level conference room, and multiple-screen mode;
- Supporting up to 4 common meeting rooms;
- Supporting data dual-stream meeting;
- Supporting video round display in the conference;
- Supporting anytime and anywhere conference;
- With the conference control APP on mobile devices , it is easy to achieve control and management of all conferences via the network;

Product Specifications

Technical	Description

indicators	
Video parameters	• Supporting H.264 video encoding protocol • Supporting up to 1080p@30fps HD video encoding and decoding • Supporting video resolution: QCIF, CIF, D1, 720P, 1080P • Supporting multiple-screen mode up to 1080P • Supporting 16:9 aspect ratio
Audio parameters	• Supporting audio coding G.711 • Supporting multiple mix
Conference capacity	• Support one high-level conference room with up to 16 conference members, providing video screen synthesis of up to 8 members • Supporting 4 regular meeting rooms, and up to 16 members for each conference room
Management tools	• Web page-based multiple-language management interface • Complete monitoring of system running status • Monitoring the network conference resource, status, and member status
Protocol support	• Supporting SIP protocol
Network parameters	• Network interface 100M/1000M adaptive, RJ45 interface
Physical parameters	• 2U19 inch standard chassis size • Width 482.6mm * depth 335.5mm * height 88mm • Net weight: 9kg
Power supply characteristics	• Pluggable power module • Supporting AC and DC redundant power supply • AC power supply: 220V • DC power supply: -48V • Maximum power consumption: 120W
Working environment	• Operating temperature: 10 ° to 45 ℃ (50 °F to 104 °F) • Storage temperature: -40 ° to 70 ℃ (-40 ° to 158 °F) • Relative humidity: 10% to 90% (non-condensing)

3.2 MS5200 Series of Media Resource Server

The media resource server is a device that provides network-specific media resources in a Softswitch system with the capabilities of audio, video, and multimedia processing. GHT MS5200 series media resource server is deployed in application layer of the G2S system, including function modules such as conference mixing, and voice file playback etc.

Features

- With the dedicated industrial COM-E hardware platform architecture from the independent research and development , the overall performance of the machine has been improved in all aspects , which is stable and reliable, and easy to expand;
- Powerful audio and video processing capacity: supporting up to 256-channel audio, and 16-channel video conferencing;
- An individual device can provide up to 100 IVR resources;
- Supporting overlay use of n sets of MS5200 server , which makes it easy to enhance the media processing capabilities of the entire Softswitch system;
- When multiple MS5200s are used , it supports the resource occupation in the manner of load sharing;
- 1+1 Dual power supply configuration, increasing the equipment reliability;

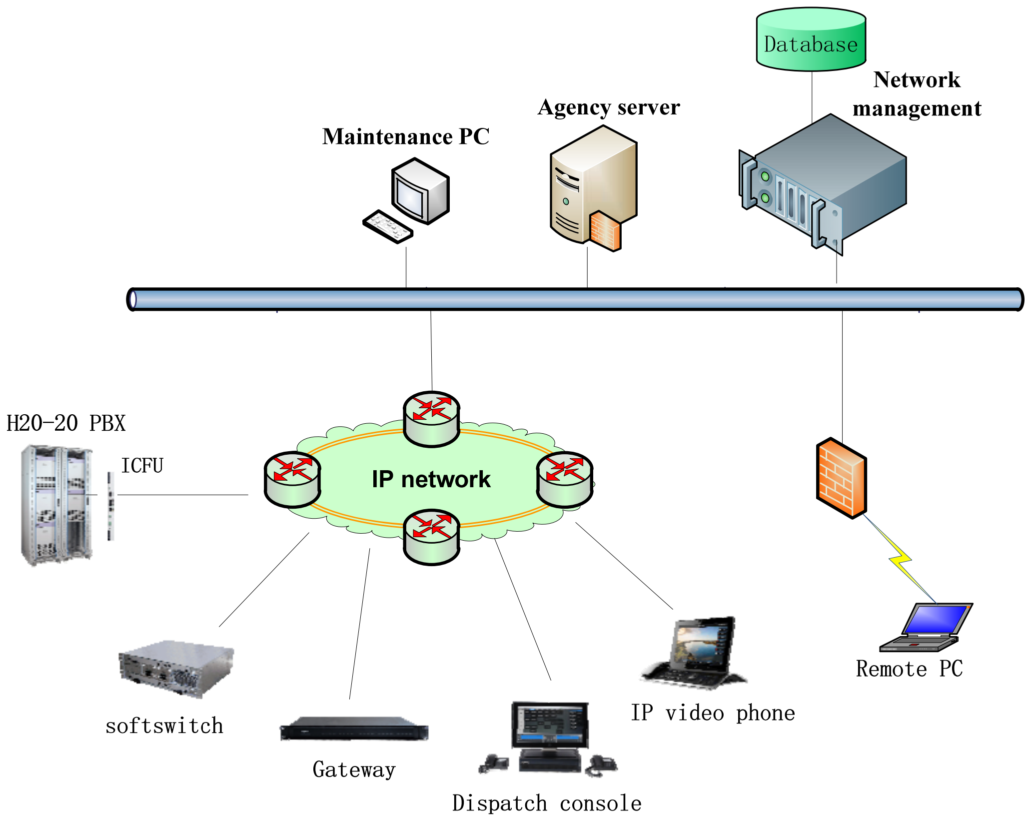
Product Specifications

Model / Parameter	MS5201	MS5202	MS5203
Maximum supported number of conference	64	128	256

members			
Maximum supported number of video members	8	16	16
Maximum number of conferencing members supported in a single conference (pure voice)	64	64	64
Maximum number of conferencing members supported in a single conference ((voice + video)	32+8	32+16	32+16
Maximum supported number of IVRs	100	100	100
Hard disk	SATA Solid State Drive (60G)		
Ethernet interface	Three, 100/1000M adaptive		
Network protocol	TCP/IP、UDP、SSH、TELNET TCP/IP, UDP, SSH, TELNET		
Size (height X width x depth)	482mm(L) X 322.5mm(W) X 88mm (H)		
Net weight	7 Kg		
Gross weight	8 Kg		
Operating temperature	0 °C to 50 °C		
Working humidity	10% To 90% (non-condensing)		
Working power supply	Dual 220V AC		
Power consumption	120W		

3.3 G2M 5012 Network Management System

GHT integrated network management system is a comprehensive management system for the networking communications equipment based on the C/S architecture to meet the needs of management and maintenance of the existing product line equipment of GHT, which takes references to the existing network management software, maintenance platform software and graphical resource management system of the company. The network management system provides the basic functions including network topology management, alarm management, performance management and security management.



Features

- Constructed according to the three-tier structure of standard network management;
- Supporting standard SQL database management system;
- Conforming to the C/S architecture, and providing a graphical network hierarchy;
- Supporting abundant TCP/IP network communication protocols;
- Adopting object-oriented design and implementation;
- Complying with the modular design, of strong scalability.
- Design of fractional de-centralized function

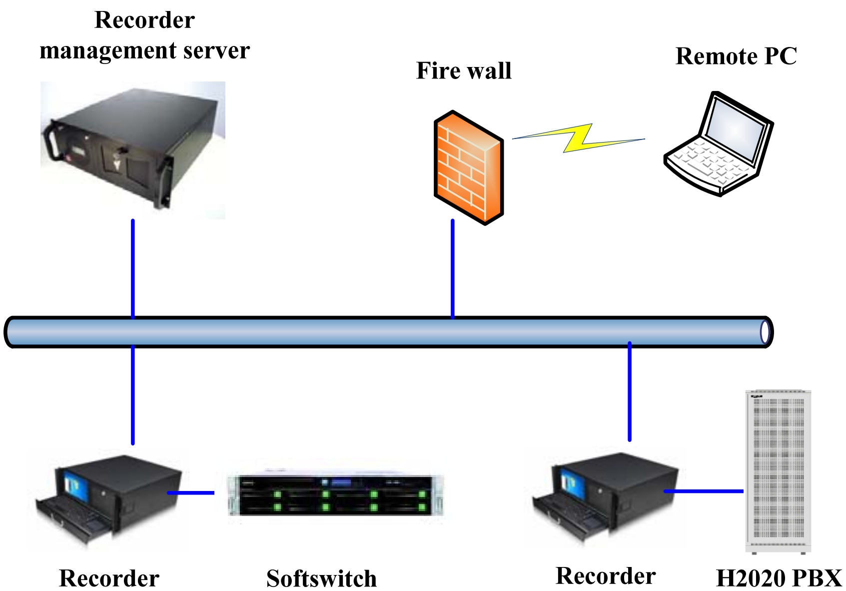
Feature Description

Functional module	Feature description
User management	Adding. modifying, deleting, and viewing the user groups
	Setting up and viewing user group permissions
	User log-in and authentication
	Region adding. modifying, deleting, and viewing users
	Modifying user password
	Viewing online users
	Forcing users to go offline
Region and devices management	Adding, modifying and deleting regions
	Adding, modifying and deleting devices
	Device alarm status statistics
	Tree navigation of regions and device on the left side
Alarm management	Alarm information filter-based2 query
	Alarm information processing
	Alarm statistics
	Warning notification
	Real-time alarm display
	Real-time alarm device-based screening
System log management	Filter-based query system log
Topology management	Region-based topology display
	Full network topology graphical display
	Device TIPS information display
	Device online and alarm status display
	Adding, modifying, and deleting links
	Link status display (including 2M status query)
	Topology zoom in, zoom out and full screen
Supporting network element types	H20-20 digital switch, SW core switching platform, ET series dispatch console, TG2600 series trunk gateway, AD1500 analog gateway, IP video phone, IP voice telephone (Note: H20-20 must be equipped with ICFU board, in order to communicate with the G2M.)
System settings	license
	help

3.4 Audio/Video Recorder System

3.4.1 Audio/Video Recorder Management Server

MR4801 audio/video recording management server adopts a dedicated server hardware structure, embedded with the latest audio/video recording management software platform of GHT. The software platform allows remote management, backup, and data statistics for the subordinate recording terminal devices. It provides customers with a powerful recording network management platform, providing complete functions including maintenance management, backup, and alarming.



The audio/video recording management server means the unified management functions for the subordinate recording system, which can collect information from the subordinate recording system, and do corresponding statistics according to the requirements, which can also dock with the general network management data, so that the general network management can collect the data and do data statistics on the audio/video recording server.

Features

- Centralized management

- Centralized maintenance
- Remote monitoring
- Remote access
- Time synchronization

3.4.2 Audio/video Recorder Server

GHT audio/video recorder terminal includes four models standard definition terminal MR4101 (4U all-in-one type), MR4102 (4U standard type), high-definition terminal MR4111 (4U all-in-one type), and MR4112 (4U standard type), which a variety of interfacing modes including high impedance parallel recording, E1 digital recording (DRCU), G2S dedicated IP recording, and universal IP recording, which adopts advanced CTI technology to provide users with a stable and practical audio and video operating platform with user-friendly operation interface, which provides the perfect audio/video recording quality and convenient file query function.

Features:

- Multiple-channel simultaneous recording: an individual device can achieve up to 1200-way recordings.
- A variety of recording methods: Providing E1 relay recording, centralized recording with recording board, dispatch console distributed recording, and network distributed recording.
- Recording networking function: Supporting networking of multiple devices for data backup, unified management, and centralized maintenance.
- Data quick query: The system conducts management in database manner, and can quickly query the complicated call records. The query can set filters such as file name, channel number, call date, and phone number, allowing fuzzy search on non-deterministic conditions.
- Rich monitoring information: It can display the channel number, channel status, name, start mode, line number, caller ID, and dial-out number; disk remaining capacity, remaining recording time; number of recordings per hour; supporting full screen display, large icons, small icons or detailed information display.
- The system audio/video recording is clear and smooth, which allows playback, and

adopts standard audio WAV and video AVI format.

- The audio recording/video recording is displayed on different pages to show the information such as caller/called party number, name, and interval.
- A call process generates an audio recording and two video recordings (caller and called party respectively).
- Parameter configuration: The main: parameters of the system can be configured locally or remotely.
- User account management: The system account password allows multiple-level user management, and the system administrator can manage the user account and permissions locally or remotely.
- Audio recording/video recording object setting: It allows setting the audio recording/video recording terminal number, which takes effect immediately without restarting.
- Caller/called party number: the audio recordings/video recordings can bear number of the calling and called parties, which supports *#0-9 characters.
- Audio/video recording system can be synchronized with the software switching system.
- The monitoring interface can monitor calling state of each audio recording/video recording object in real time; It can display the distribution diagram of 24 hours of audio recording/video recording quantity.
- Local/remote monitoring of the recording: It allows monitoring the voice call in real time through the monitoring interface; It allows monitoring the voice call remotely in real time through the client monitoring interface; The calling and called parties' video can be displayed during monitoring.
- Batch query for audio recording/video recording: It allows fuzzy search according to the filters such as caller/called party number, and call time interval.
- Supporting phone book function, if the audio /video recording object is listed in the phone book, the audio/video record will show the Chinese name, and it allows fuzzy search based on the Chinese name.
- The audio/video records can be played back by double-clicking on the query screen,

or be played one by one if multiple records are selected.

- The digital video coding device has the functions of video signal acquisition, compression, coding and real-time transmission, which adopts MPEG4 or H.264 as the video compression standard, which adopts the TCP/IP network transmission protocol, where the image effect is 25/frame/sec/channel, while the resolution supports VGA, QVGA, CIF, QCIF, and 720P formats.
- Automatic segmentation of ultra-long recording: The record is segmented according to the actual situation to avoid too large files.

Product Specifications:

Model **parameters**	MR4101	MR4102	MR4111	MR4112
Product Image				
Product form	4U Integrated	4U split-type	HD 4U HD Integrated	HD 4U HD split-type
Supporting high impedance parallel recording mode or not	√	√	√	√
Supporting E1 tandem recording or not	√	√	√	√
Supporting E1 centralized recording mode or not	√	√	√	√
Supporting E1 wiretapping recording mode or not	√	√	√	√
Maximum number of recording channels supported	128	128	128	128
Maximum	600	600	600	600

number of recording channels supported by E1				
Maximum number of recording channels supported by IP	600	600	600	600
Maximum number of video recording channels supported by IP	16	16	16	16

4. IP Gateway

4.1 SIPU Trunk Gateway

SIPU is an E1 trunk gateway board, which is plugged into the user slot of the H20-20 switch; and supports one E1, which is connected to the Ethernet through the front panel network port. SIPU lies between the circuit switching network and Softswitch network, providing conversion of the signaling flow and media streaming of both sides. In addition, it also supports a variety of dispatch functions, conveniently realizing the seamless interoperability of the dispatching application of the circuit switching network and Softswitch network.

Features

- Rich Dispatch Functions

In addition to the basic voice application, SIPU also supports a variety of dispatch functions, conveniently realizing the seamless interoperability of the dispatching application of the circuit switching network and Softswitch network. In addition, SIPU provides flexible routing configuration, E1 trunk group configuration, master/called translation, etc., meeting the needs of a variety of engineering networks.

- Guarantee of complete equipment reliability

SIPU has the following functions: N+1 backup of the device, IP route backup, server dual homing, built-in firewall, IP filtering, and anti-Telnet. There are various means adequately ensuring the reliability of equipment and application, completely avoiding single-point failures of the equipment.

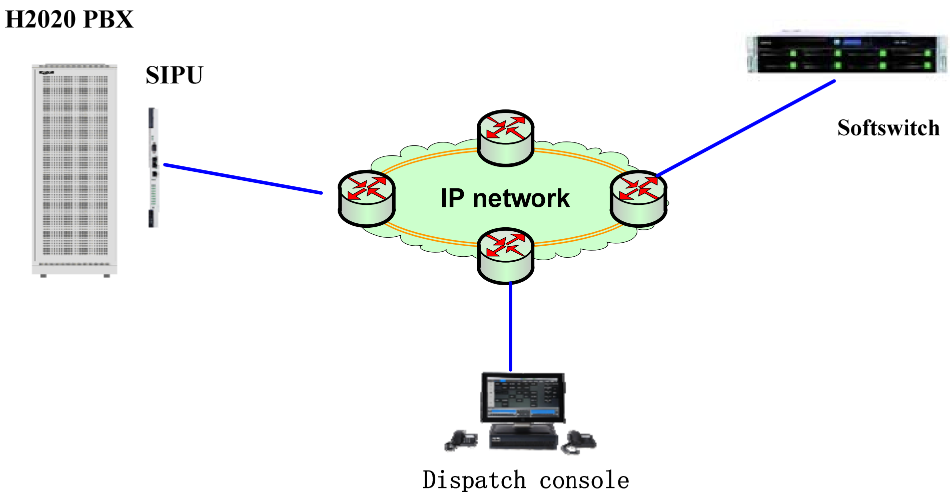
- Simple and convenient management and maintenance

SIPU is plugged in the user slot of the H20-20 switch and, which is powered by the switch back plane, stable and reliable.

Product Specifications

Application functions	
Voice application	Supporting voice connection from PSTN to IP, and from IP to PSTN.
Dispatch application	Supporting a variety of dedicated network dispatch application: call forwarding, diversion, forced insertion, forced breakup, cross-network monitoring, and conference etc.; Supporting cross-network grouping of circuit dispatching station and Softswitch dispatching station, realizing seamless integration of the two network dispatch application;
Call protocol	
PSTN side call protocol	ISDN PRI
Softswitch side call protocol	SIP
Voice/fax features	
Voice coding	G.711 A/U, G.729 A/B
Fax coding	T.30 pass-through, T.38
DTMF	In-band pass-through, RFC2833
Interface	
Trunk interface	Providing an E1 trunk the switch back plane
Network management	
Maintenance mode	Remote WEB page, serial command line, and Telnet command line
Physical characteristics	
Working power supply	Switch back plane power + 5V 1.5A
Rated power	10W
Slots	It is inserted into the telephone interface rack of the H20-20 switch, where the slot limit is the same with DTU board; that is, it is inserted in the even-numbered slot of the 16-line back plane, and the third one of every four slots of the 8-line back plane.

Typical Application


H2020 PBX
SIPU
IP network
Softswitch
Dispatch console

4.2 TG2600 Series of Media Trunk Gateway

TG2600 series of products is a new external media trunk gateway product of GHT G2S system, which lies between PSTN network and IP switching network, providing the conversion of signaling flow and media streaming of both sides, which realizes the seamless interoperability between the PSTN network and Softswitch networks.

The TG2600 is designed for telecom operators, value-added service providers and large and medium-sized enterprises. Compared with other similar products, it has very obvious advantages in terms of performance, system reliability, and compatibility. Efficient hardware and software design and powerful DSP processing capabilities ensure that the TG2600 still maintains a clear voice quality at full capacity

TG2600 supports ISDN PRI signaling on the PSTN side, and uses standard SIP signaling on the IP side, which allows inter-operating with other mainstream SIP switching platforms. TG2600 series contains three products, which are TG2601/TG2602/TG2603; provide three trunk port capacities, which are 2E1, 4E1, and 8E1.

Features

➢ It is based on the Com-e hardware architecture under independent research and development of the company, so the overall performance has been improved in all aspects, which is stable and reliable, and easy to expand;

➢ Excellent voice quality: supporting G.711a, G.711µ, G.729 and other voice encoding & decoding, encoding & decoding dynamic switching function for better use of the network bandwidth, so that the calling quality can be guaranteed at low bandwidth and high error rate;

- Supporting up to eight E1s (can be connected to different PBX), and maximum 240 concurrent calls;
- Supporting one way of E1 power failure escape;
- The power supply is PnP supported, optional 1+1 dual AC, dual DC or 1 AC +1 DC dual power supply, which increases the device reliability, and can be configured flexibly according to the user power supply in the machine room;

Features

Model / Parameter	TG2601	TG2602	TG2603
Number of E1 supported	2E1	4E1	8E1
Ethernet interface	Three, 100/1000M adaptive		
Trunk interface	RJ48 BNC		
Network protocol	TCP/IP, UDP, SSH, TELNET		
Signaling protocol	Supporting PRI QSIG signaling		
	Supporting SIP protocol		
Voice processing	Supporting G711 and G729 voice encoding and decoding;		
	Supporting silence detection, echo cancellation and RTP packet length setting;		
	Supporting comfort background noise;		
	Supporting pass-through fax mode;		
	Supporting global voice coding and outgoing routing preferred voice coding;		
Call processing	Supporting the configuration of global incoming and outgoing routing, and operation on missing code and extra code;		
	The incoming route supports two routing matching modes, which are "dialing rules" and "caller IP";		
	Supporting trunk grouping, so 240 circuits of eight 2M channels can be assigned to the different trunk groups;		
	Supporting outgoing master/standby routes;		
	Supporting DTMF code transmission in INBAND mode;		
	Supporting DTMF transmission in RFC2833 mode;		
	Supporting H2020-triggering QSIG route prediction function		
Communication function	Call hold		
	Call diversion (including no answer, busy, and unconditional diversion)		
	Call forwarding		

	Barge in
	Override
	Ring tones
	Conference
	Supporting authentication, of incoming IP, and supporting configuration of incoming IP white list and blacklist
Online protection	Time server can be configured, which can be time synchronized;
	Supporting system log management
	Allowing real-time output of QSIG and SIP signaling messages;
	Supporting dual network port backup;
Maintenance management	Supporting local maintenance: The gateway supports maintenance via the serial port that supports command line mode
	Remote maintenance: The gateway supports maintenance via telnet or web. Telnet supports command line mode;
	Two-level log-in system permissions: ADMIN and USER
	Providing packet capturing function, including all the packets, covering signaling voice, which can be exported for viewing;
	Providing a record of the call list, which can be exported for viewing;
	Supporting restoration of factory settings;
	WEB maintenance supports toggling between Chinese and English.
	Supporting to display the alarm information on the digital tub in the front panel;
Size (height X width x depth)	482mm(L) X 322.5mm(W) X 88mm (H)
Net weight	7 Kg
Gross weight	8 Kg
Operating temperature	0 °C to 50 °C
Working humidity	10% to 90% (non-condensing)
Working power supply	Dual 220V AC/Dual 48V DC/220V AC+48V DC
Power consumption	120W

4.3 AD1551 Access Gateway

AD1551 integrated access gateway lies in the access layer of the G2S Softswitch system, responsible for connecting PSTN network and network terminal devices to the packet switching network. It can provide traditional voice application with data and voice processing features, which can process analog voice and has media streaming functions to support data application, and existing and to-be-developed Softswitch application.

Features

- Single-board card plugging design, supporting flexible optional user capacity from 32 to 256;
- Two power supply modes : 220V AC and -48V DC power supply, suitable for different installation environment;
- Standard 4U height, for mounting on standard 19 inch rack;
- Supporting all FXO power failure escape;
- Supporting a variety of IP switching registration modes : non-registration, switchboard registration and extension registration;
- Powerful DSP processing capabilities, supporting G.711A/U, G.729, G723 encoding and decoding standards, and supporting silence detection and echo cancellation, optional IP packet length;
- Supporting two fax modes: Pass-through fax and T.38 fax (rate configurable);
- Flexible routing configuration of dialing rules , and supporting called number conversion;
- Providing traditional voice data application, such as voice calls, fax, etc., as well as various supplementary applications, such as call diversion, call pickup, caller ID, hot line call, call forwarding, and call waiting etc.;
- Allowing dialing according to the port limit;
- Configuration of device IP address on extension;
- Three maintenance configuration modes: Web (http), serial port and Telnet.
- Product specifications

Features

FXS port	Maximum number of analog user ports	256
System memory		256M
Main control board	Reset/Restore factory settings	Press the RST key 1s, the system will restart; Press the RST key for more than 3s, the system will restore the factory settings.
DTMF		In-band pass-through, RFC2833
Voice characteristics	Voice encoding and decoding	G.723.2, G.729A/B, G.711A/U
	Echo cancellation	ITU-T G.168 2000 8/48/128ms
	Fax	T.30 pass-through, T.38
	Packet cycle	Supporting customized RTP packet size, 10/20ms
	Other features	Supporting silence detection
Call control protocol		SIP
Network protocol		TCP/IP, UDP, RTP/RTCP, ARP, TFTP, Telnet, ICMP
Size (width × height × depth)		482Mm (length) x308 mm (width) x177 mm (height), mounting on standard 4U19 inch rack;
Net weight		14 Kg
Gross weight		16.5 Kg
Operating temperature		0 ℃ to 55 ℃
Working humidity		≤ 80% (Non-condensing)
Working power supply	AC	220V AC (range 90V AC - 265V AC)
	DC	-48V DC (range -36V DC to -72V DC)
Power consumption		264W

4.4 AD1500 Series of Analog Gateway

AD1500 is a series of IAD (Integrated Access Device) products of GHT for the growing demand for IP switching networks in the sectors of industrial dedicated network and telecom operations, which combine terminal equipment (such as analog telephones, fax machines etc.) into the packet switching network to provide users with flexible, convenient and rich voice data application. AD1500 adopts the industry's advanced voice processing technology, featuring clear voice, easy installation, and high reliability. At the same time, AD1500 has strong networking capacity, which both can accept the authentication and management of soft-switching servers, also can be used independently as a small PBX, leading to cost reduction for users on one hand, and providing traditional and value-added PSTN application as good as PSTN does.

Features

- Strong networking capabilities

Both can accept the authentication and management of IP switching platform; also it can be used independently as a small PBX; also can be connected to the PSTN switch to realize application interoperability. The strong networking capabilities and sufficiently meet the requirement of different users.

- Rich application functions

Supporting internal self-switching and Hairpin function, and having complete traditional and value-added PSTN application.

- Complete product line

AD1500 provides a wide range of FXS and FXO port specifications, and a full range of analog gateway devices, meeting the requirements of various applications and solutions.

- Guarantee of complete equipment reliability

AD1500 has the following features: N+1 backup of the device, IP routing backup, server dual homing, built-in firewall, IP filtering, and anti-Telnet. There are various means adequately ensuring the reliability of equipment and application, completely avoiding single-point failures of the equipment.

- Simple and convenient management and maintenance

AD1500 is a 19-inch, 1U height construction, can be mounted in a standard cabinet. Both supporting the -48V DC power supply in the telecom machine room, also optionally using 220V AC power supply, adaptive to a variety of machine room conditions.

Product Specifications

Model / Features	AD1508	AD1516		AD1524	AD1532	
Analog interface						
Number of FXS	8	8	16	24	24	32
Number of FXO	--	8	--	--	8	--
Application functions						
Basic voice application	Supporting voice connection between internal extensions, between extension and PSTN, between extension and IP, between PSTN and IP					
Value-added application	Supporting a variety of supplementary services: internal short number, caller ID, hot line call, call hold, call back ringing, call waiting, call pickup, call diversion, call forwarding, call barring, switchboard, and power failure escape					
Call protocol						
Softswitch side call protocol	SIP					
Voice/fax features						
Voice coding	G.723.1, G.729 A/B, G.711 A/U					
Fax coding	T.30 pass-through, T.38					
DTMF	In-band pass-through, RFC2833					
Network management						
Maintenance mode	Remote WEB page, serial command line, and Telnet command line					
Physical characteristics						
Working power supply	220VAC or -48VDC					
Rated power	40W					
Structure size	Standard 19 inches width and 1U height: 44CM × 4.44CM ×					

	29CM (W × H × D)

5. ET Dual-mode Dispatch Consoles

ET series dual-mode dispatch consoles is the latest generation of dispatch console products of GHT following the D series and PM series dispatch consoles, which is in three major categories of product form, including the touch screen dispatch console, Integrated dispatch console and the keyboard dispatch console, covering a variety of scenarios for professional command and dispatch applications. ET dispatch console inherited the high reliability as the constant feature of GHT dispatch console products, while it has absorbed the experience of the earlier dispatch consoles, resulting in the prettier appearance, more powerful functions, and more user-friendly interface, which has got the praises from all users.

<table>
<tr><td rowspan="7">ET series of dispatch consoles look-up table</td><td>Keyboard dispatch console</td><td>ET200 series</td><td>ET210</td><td>Keyboard
Without video</td></tr>
<tr><td>Integrated dispatch console</td><td>ET800 series</td><td>ET820</td><td>Integrated host,
Screen and camera</td></tr>
<tr><td rowspan="5">Touch screen dispatch console</td><td rowspan="2">ET300 series</td><td>ET320</td><td>Small host,
On-desk or under-desk installation
(With video)</td></tr>
<tr><td>ET310</td><td>Small host,
On-desk or under-desk installation
(Without video)</td></tr>
<tr><td rowspan="3">ET600 series</td><td>ET630</td><td>Touch Screen
(720P video)</td></tr>
<tr><td>ET622</td><td>Touch Screen
(With video)</td></tr>
<tr><td>ET611</td><td>Touch Screen
(Without video)</td></tr>
</table>

5.1 ET600 Series Touch Screen Dispatch Console

Touch screen dispatch console uses a host and monitor in separate mode, similar to a desktop computer, which ensures a clean desktop while a larger display screen is obtained. ET600 dispatch console is generally deployed in the machine room, while the monitor, camera, telephone and other human-computer interaction devices are deployed on the dispatch desktop, for which signal extension is realized through extenders.

ET612/ ET622

ET631

Features

- Dual modes are simultaneous online, which improves the system reliability. ET dispatch console has both 2B + D_U port and RJ45 network port, which can be registered to the traditional circuit switches and SIP servers at the same time; the dual-mode interface uses redundancy hot standby, there for, in case of a failure in any transmission channel, or an interface failure, it can automatically switch to the normal channel and interface, ensuring the normal operation of dispatch communications application.
- It adopts GHT Linux professional operating system, bringing higher security.
- It has a 22-inch industrial touch screen as a standard configuration, which adopts SAW surface wave touch technology, of 1920*1080 HD resolution, in LED back light design, more advanced, lighter, low power consumption, as well as longer monitor service life.
- It adopts the industry's popular interactive operation, supporting swiping, and touch and hold operations.
- Interface customization based on user experience. Hot key number and color, tab links, function keys, title bar logo, etc. can be customized by the user.
- Supporting local recording and playback; Supporting remote recording query and playback on the dispatch console.
- As the industry's first in supporting 720P HD visual dispatching and high-definition

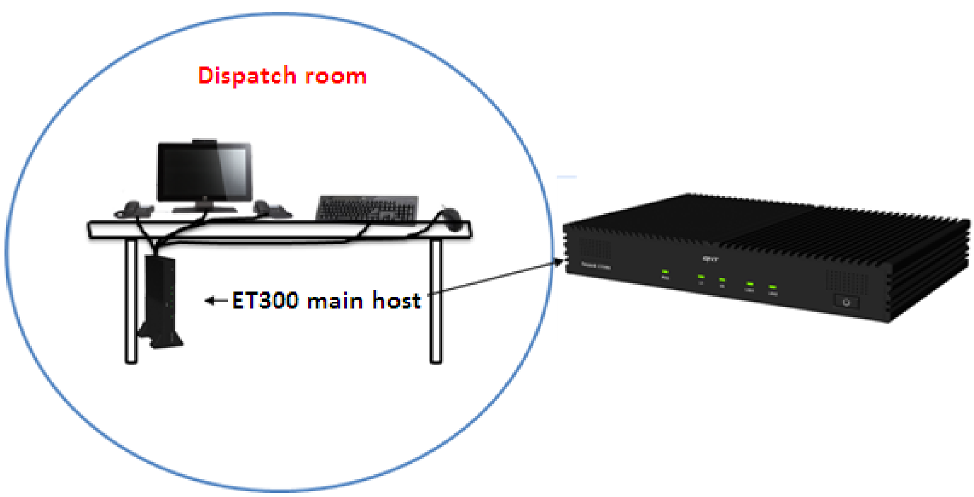
video conferencing. (ET630)

Product Specifications

Model	ET600 series		
	ET612	**ET622**	**ET631**
Host size (Width × depth × height) Unit: mm	430×336×88	430×336×88	430×336×88
Thermal design	Fan		
Power supply	Mixed dual power supply		
Video	-	SD	HD
Operating system	GHT Linux		
Hot key users	The number of hot-key per screen can be customized for up to 255 screens, and each hot key user storage numbers are up to 10		
Display screen	22-Inch industrial-grade surface wave touch screen		
Dispatching interface	Two IP network interfaces (RJ-45), 10/100/1000M adaptive, two ISDN U interfaces (RJ-11)		
Audio port	Two standard telephone interfaces (RJ11), a 3.5mm external speaker/headset jack, a 6.35mm microphone jack		
Recording interface	Recording interface (RJ-10), supporting voltage control and voice control (with GHT recording system), and supporting centralized and distributed recording		
USB interface	4 USB ports, supporting standard USB devices such as camera, mouse, keyboard, and storage device etc.		

5.2 ET300 Series Touch Screen Dispatch Console

ET300 Series dispatch console is a small Touch screen dispatch console that is designed to meet the needs of some users who wish to deploy it at the dispatching seats. While fully inheriting the functions of ET600 dispatch console, ET300 is in no-fan design, lightweight, compact, and zero-noise for, the lowest noise, which can be easily deployed on the desk or under the desk in the dispatcher's office, thus avoiding the problem of introduction of interface extension line when the console host is deployed in the machine room.



Product Specifications

Model	ET300 series	
	ET312	**ET322**
Host size (W×D×H) unit: mm	350×230×51	
Thermal design	Natural cooling (no-fan)	
Power supply	Single AC 220V or DC -48V	
Video	-	SD
Operating system	GHT Linux	
Hot key users	The number of hot keys per screen can be customized for up to 255 screens, and each hot key user storage numbers are up to 10	
Display screen	22-Inch industrial-grade surface wave touch screen	
Dispatching interface	Two IP network interfaces (RJ-45), 10/100/1000M adaptive, two ISDN U interfaces (RJ-11)	
Audio port	Two standard telephone interfaces (RJ11), a 3.5mm external speaker/headset jack, a 6.35mm microphone jack	
Recording interface	Recording interface (RJ-10), supporting voltage control and voice control (with GHT recording system), and supporting centralized and distributed recording	
USB interface	4 USB ports, supporting standard USB devices such as camera, mouse, keyboard, and storage device etc.	

5.3 ET820 Integrated Dispatch Console

The all-in-one dispatch console adopts one-piece design that integrates the screen and host, suitable for direct placing on the desk of the dispatcher. The dual-mode all-in-one ET820 released by GHT are integrated with the host, 15-inch touch screen, 500-megapixel camera, professional microphone, and handle, etc. in one set, featuring small size and pretty appearance, combined with a new interactive UI design, providing users with a perfect operating experience.

Features

- Dual modes are simultaneous online, which improves the system reliability. ET dispatch console has both 2B + D_U port and RJ45 network port, which can be registered to the traditional circuit switches and SIP servers at the same time; the dual-mode interface adopts redundancy hot backup, there for, in case a failure in any transmission channel, or an interface failure, it can automatically switch to the normal channel and interface, ensuring the normal operation of dispatch communications application.
- It adopts GHT Linux professional operating system, bringing higher security.
- Interface customization based on user experience. Hot key number and color, tab links, function keys, title bar logo, etc. can be customized by the user.
- Supporting local recording and playback; supporting remote recording query and playback on the dispatch console.

- Full-angle view high-definition Sharp 15-inch touch screen, supporting four-level adjustable angle, so the user can get a clear view when being seated or standing up.
- Aviation plug, making the power access more secure.
- Supporting power failure escape, the left handle can be used as the emergency telephone for normal communication in case of power failure to the dispatch console, which enhances the emergency response capability of the dispatch command system.

Product specifications

Display screen	Built-in 15" industrial touch screen with a resolution of 1024*768
Operating system	GHT Linux
Dispatching interface	2 IP network interfaces (RJ-45), 10/100/1000M adaptive 2 ISDN U interfaces (RJ-11)
Voice interface	2 Handles (one of which can be used as the emergency telephone port for the right handle); 1 External speaker/headset; 1 Microphone
Recording interface	Recording interface (RJ-10), supporting centralized and distributed recording
USB interface	4 USB interfaces, supporting standard USB devices such as mouse, keyboard, and storage device etc.
Hot key users	The number of hot keys per screen can be customized for up to 255 screens, and each hot key user storage numbers are up to 10 entries
Host size	Width 580mmx height 238 mmx depth 305mm
Operating Voltage	AC220V/DC48V
Video parameters	Built-in camera: sampling rate of 25 frames/sec, 500 million pixels Video encoding format: H.264 Video resolution: QCIF/CIF/D1

5.4 ET210 Keyboard Dispatch Console

The Keyboard dispatch console gets some users' favor with its simple user physical button operation, which is in contrast to the traditional Keyboard dispatch console that is of bulky size. The dual-mode Keyboard type dispatch console ET210 adopts physical keys plus electronic tags for greatly reduction of the product size, which is more suitable for direct placement on the dispatching desk. At the same time, ET210 adopts modular combination design, which is composed of the host module (with the left handle), user hot key expansion module and the right handle module, supporting optional user capacity from 64 to 320 according to the number of expansion modules selected by the user.

Features

- Dual modes are simultaneous online, which improves the system reliability. ET dispatch console has both 2B + D_U port and RJ45 network port, which can be registered to the traditional circuit switches and SIP servers at the same time; the dual-mode interface uses redundancy hot standby, there for, in case a failure in any transmission channel, or an interface failure, it can automatically switch to the normal channel and interface, ensuring the normal operation of dispatch communications application.
- It adopts GHT Linux professional operating system, bringing higher security.
- Supporting local recording and recording playback.

- Using the original German Cherry ultra-thin mechanical keyboard to meet the 20 million times of long-term use.
- Adopting modular combination design, with maximum expansion of four user modules, and 64 user hot keys supported each module.
- Supporting power failure escape, the left handle can be used as the emergency telephone for normal communication in case of power failure to the dispatch console, which enhances the emergency response capability of the dispatch command system.
- 7-Inch LCD color display, LED back light board design, brightness adjustable according to the needs of the scene.

Aviation plug, making the power access more secure.

Product Specifications

Display screen	7-Inch industrial display with a resolution of 800*480
Operating system	GHT Linux
Built-in speaker	2 pieces, rated power 2W, impedance 4Ω
Keyboard	Cherry ultra-thin mechanical keyboard, 20 million times of button life.
Voice encoding and decoding	711 Rate A and rate U
Network protocol	Supporting SIP v2 (RFC3261) protocol
DTMF transmission	Transparent transmission/RFC2833
hot key users	The host module supports 64 user hot keys, up to four expansion modules, and up to 320 user hot keys.
Host size	Width 580mm * height 63 mm * depth 255mm
Power supply	Supporting two power input modes, AC220V and DC48V (Providing two AC and DC power supply adapter, 220V to 12V, 48V to 12V)

5.5 Dispatch console Function List

Feature category	Features	Feature description	Remarks
Call function	One-touch call-out	Calling out by clicking on the screen hot key or by pressing the physical hot key	
	Dialing call	Calling by using the phone keypad or by clicking the virtual key pad on the screen	
	Video call	In the SIP registration mode, video chatting is available for both sides, which is the default call function, where the resolution is subject to their agreement	Supported only on video dispatch console
	Video On/Off	Supporting manual shutdown of video streaming under poor network conditions	Supported only on video dispatch console
	Call hold	Holding the ongoing call while making a new call operation, and the held party will listen to the prompt sound or music.	
	Call forwarding	Transferring the ongoing call to a third party.	
Dispatching function	Number of hot keys is customized	Each tab can support to customize the number of hot keys, and the font size is adaptive	
	The tabs link to hot keys	Define a hot key type link to a tab to realize quick switching of tabs.	
	One key for multiple number	A user hot key can be configured with up to 10 numbers	
	Single call/selected call/round call	Single call: call the first number under the user hot key; Selected call: secondary selected call of the number under user hot key; Round call: call all numbers under the user hot key until they answer;	Keyboard type dispatch console supports single call and round call
	Caller queue	The system queues the incoming calls for the same dispatching group according to the time, while emergency calls go first automatically, distinguished in different color.	
	Answering incoming calls	The dispatcher picks up the phone to answer an incoming call from the queue.	
	Selected	The dispatcher can choose to answer any	

	answering of incoming calls	incoming call in the queue based on the importance of incoming calls priority.	
	Group answering to the incoming calls	In the event of calls from multiple users at the same time, group answering allows answering to all of them at the same time, then it enters the conference state	
	On-duty extension	The system will forward the incoming call to the backup extension or the on-duty extension, then the dispatcher can answer the it on the on-duty extension.	
	Barge in	When the dispatch console is busy to answer the incoming calls, the forced insertion key allows a three-way call	
	Override	When the dispatch console is busy to answer the incoming calls, the forced breakup key allows answering the new call while breaking up the existing call	
	Merger	When a handle is talking, press the "Merge" button, it allows dispatching for the same user with the two handles at the same time.	
	Loudspeaker	The local and opposite talking can be played on the loudspeaker	
	Blacklist	A number can be add to the blacklist, so no prompt is given then it calls in, but a missed call is left in the history	
	Caller address display	The caller field and call history can display the caller address	
	Smart callback	The dispatch console can recognize local calls and remote calls, so the prefix number can be added automatically when dial out.	
	Dispatch console grouping	Multiple dispatch consoles can be defined as a dispatch console group, where the members share the dispatching information and status, achieving all consoles ringing on a call, and answering to all calls by any member.	
	In-group listening	It allows monitoring the talking of members of the same group	
	Absent/restoration	In the state of absence, the dispatch console status is displayed as "absent", and all incoming calls are handled by other dispatch consoles in the same group.	
Conferenc	Voice	Supporting voice conference	

e function	conference		
	Video conference	Supporting HD/SD mixed video conference	
	General broadcast/equalization	General broadcast: the master host talks, while members listen; Equalization:: All members of the conference can talk and listen.	
	Conference control	Supporting conference control such as adding and deleting members, and screen selection	
Recording function	Local recording and playback	The dispatch console allows local recording, and recording playback	
	Server recording query and playback	The dispatch console allows query of recording on remote server, and playback the recording	
	Audio/video recording interface	The recordings are sent directly to the audio/video recording system by the dispatch console, which need not be transmitted through the server.	
Dual-mode function	Dual-mode registration	Supporting ISDN and SIP online at the same time	
	Automatic selection of preferred call-out	Hot key users can call out preferably ISDN or SIP mode independent of the current active mode status	
	Quick switching of working modes	Quick switching of current active work mode in the main interface	
Local setting	Customized ringing tone selection and pre-playback	Supporting WAV and MP3 format and ring tone uploading	
	Volume adjustment	Supporting volume adjustment for the speaker and microphone	
	Customized display of handle function keys	The function key position can be adjusted according to the user's habit	
	Replacement of the LOGO at the upper left corner of the interface	It is allowed to replace the logo at the upper left corner of the main interface according to user requirements	Supported by Touch screen and all-in-one type dispatch consoles

6. IP Phone

6.1 IP Video Phone

VGS2000 dual-mode video phone is enterprise-grade video phone integrated with voice, video and rich applications, which provides a better voice quality, faster response speed and integration of more value-added application functions, greatly improving user experience of video communication and enterprise productivity. VGS2000 integrates many new application functions such as BLF (Busy Lamp Field), and voice mail etc., which is an ideal phone for multimedia communications, virtual office and IP video conversations (business or home users).

Features

- Supporting 6-way SIP accounts and 1-way PSTN accounts online at the same time;
- Supporting PSTN power failure escape;
- 10.1" Digital color LCD multiple-touch capacitance screen, with resolution up to 1290 × 800
- 500 million pixel camera, multiple-angle adjustable
- Supporting up to 720P video encoding and decoding
- Supporting conference initiation and conference control
- Local call recording and management
- Dial smart match, efficient call
- Android system, mass compatibility, and fast customization
- Supporting Wi-Fi access
- Supporting Bluetooth

Technical Specifications

Video characteristics	• Video encoding and decoding: H.264 High profile • Picture format: JPEG, PNG, BMP • Video format: 720P/VGA/QCIF/CIF • Supporting frame rate 15 frames/sec and 30 frames/sec • Adaptive bandwidth adjustment • Local video control
Audio characteristics	• Broadband encoding and decoding: G.722 • Narrow band encoding and decoding: G.711A/U, G.729 • DTMF: SIP info, SIP info relay, RFC2833 and Mix • Supporting full duplex hands-free, with automatic echo cancellation • Comfortable background sound • Adaptive jittery buffer • Adaptive packet loss recovery
Contact list	• Supporting local and remote contact list • Phone book (1000 contacts), call records (200 entries); • Contact avatar • Supporting LDAP remote contact list and localization • Custom group, 10 level display • "Speed Dial" application and address book search • Editor of VCF format local address book
Telephone characteristics	• 6 SIP accounts, 1-way PSTN line • Dual mode online at the same time • Supporting 6 SIP accounts at the same time • 36 Custom soft keys • Self-supported 3-party voice conference • The phone is still available in PSTN mode in case of an unexpected power failure • Audio call, video call, call hold, and call transfer • Call queuing capability, including incoming calls, outgoing calls, and ongoing calls • Breathing light prompt on incoming calls, and for event reminding • Voice mail • Busy Line Indicator (BLF) • Default 100 hours of local call recording, allowing longer hours with TF card extension • Custom ring tones, wallpaper • Android4.4.2 operating system • Supporting unified communications (UC) linkage • Software customization based on user requirements

Network and security	• SIP protocol v1 (RFC2543), v2 (RFC3261) • NAT traversal: UPNP • IP address acquisition mode: static/DHCP/PPPoE • Time and date synchronization: SNTP • QOS supporting - IEEE 802.1p/Q tag (VLAN), Layer 3 TOS and DSCP • Supporting OpenVPN, IEEE802.1X • Supporting VLAN
Management	• Configuration mode: browser/phone/auto configuration • Centralized configuration based on FTP/TFTP/HTTP/HTTPS protocols, supporting large-scale deployment • Supporting recovery mode • Supporting restoring the factory settings, and restart • Supporting exporting debug package, and system log
Physical specifications	• 10.1 Inch digital color LCD capacitance touch screen, resolution 1280×800, 16:10 wide screen, multiple-touch • 5 Million pixel camera, adjustable angle • CPU 4 cores 1GHz; Memory 1GB; Storage 16GB • 3 Touch buttons, 23 physical buttons • Supporting adapter power supply: input AC100 - 240V, output DC12V ± 10%/1A; • Supporting Power over Ethernet (POE): IEEE 802.3af • Two 10/100M network ports • One RJ11 interface (telephone line) • One TF card interface • One USB port (Master) • 2 Sets of 3.5mm audio interface • One MINI HDMI interface • WIFI, Bluetooth (optional) • Power consumption: 5-10W • Size: 350mm * 220mm * 230mm • Net weight: 2400g
Certification	• CE、FCC、ROHS、Broad soft CE. FCC, ROHS, Broad soft

6.2 IP Voice Phone

IP voice phone is designed based on enterprise application, for which the open SIP protocol standards are adopted along with innovative technical design, which can provide beautiful sound quality, rich functionality and high cost performance. GHT currently provides three SIP voice phone models, which are 2102, 2303, and 2706.

2102 2303 2706

Specifications

Model / Specification	2102	2303	2706
Phone features	Supporting two SIP accounts and hot line Supporting call waiting, call transfer, call forwarding Call hold, mute, auto answer, redial Three-party conference, no-disturb, speed dial, blacklist, call record, volume adjustment, and ring tone selection	Supporting three SIP accounts, hot line, emergency number call-out Supporting call waiting, call transfer, call forwarding, Call hold, mute, auto answer, redial Three-party call, no-disturb, speed dialing Phone book, blacklist, call record, volume adjustment, and ring tone selection	Supporting 6 lines, hot line, emergency number call-out Supporting call waiting, call transfer, and call forwarding Call hold, mute, auto answer, redial Three-party call, no-disturb, speed dial XML remote phone book search/export/import Blacklist, call record (100 entries) Volume adjustment, ring tone selection Country signal tone, system logs Multiple-language (more than 20 languages) Allowing connecting to six EXP38 key expansion

<table>
<tr><td></td><td></td><td></td><td>module, and each expansion module provides 38 custom hot keys, 228 keys in total.</td></tr>
<tr><td>Physical characteristics</td><td>Embedded Broadcom chips
Black and white LCD, 128x48 pixels
WAN port -10/100 BASE-T RJ-45 to LAN
LAN port -10/100 BASE-T RJ-45 to PC
Headset jack: RJ-9 (optional)</td><td>Embedded TI TITAN chip set
Black and white LCD, with back light 132x64 resolution graphics LCD screen
External 32 keys (including 4 soft keys)
5 LED indicators (1 power indicator, 3 account indicators and 1 information indicator)
1 RJ9 (4P4C) handle connector
1 RJ9 (4P4C) headset connector
2 RJ45 10/100M Ethernet ports
Supporting wall mount</td><td>Embedded TI TITAN chipset
Black and white LCD with back light 320 x 160 resolution graphics LCD screen
External 48 keys, including 16 programmable keys
9 LED indicators (1 power indicator, 6 account indicators and 1 information indicator, 1 headset indicator)
1 RJ9 (4P4C) handle connector
1 RJ9 (4P4C) headset connector
2 RJ45 10/100M Ethernet ports
1 RJ12 (6P6C) expansion module interface</td></tr>
<tr><td>Audio characteristics</td><td>G723.1, G726, G.729AB
Broadband coding: G.722 Narrow band coding: G.711, G723.1, G726, G.729AB
Supporting audio processing including VAD, CNG, AEC, PLC, AJB, and AGC
Full duplex hands-free, with automatic echo cancellation</td><td colspan="2">Broadband coding: G.722
Narrow band coding: G.711, G723.1, G726, and G.729AB
Supporting audio processing including VAD, CNG, AEC, PLC, AJB, and AGC
Full duplex hands-free, with automatic echo cancellation</td></tr>
<tr><td>Working power</td><td>AC100 - 240V input, DC5V/1A output</td><td colspan="2">AC100 - 240V input, DC5V/1.2A output</td></tr>
</table>

supply			
Overall dimensions	155mm*185mm*130mm	185mm*200mm*90mm	273mm*204mm*42mm