24-Line FXS SIP IP Gateway



WellGate 2424S front view



WellGate 2424 rear view

Dual IP Stack: IPv6 and IPv4 Simultaneously

Support up to 16 SIP proxy Servers
Support 24 analog Phone sets at one IP address
Support different SIP Trunk to each FXS line
Auto HTTP Provision feature
Flexible Routes Plan and Dial Plan
Redundant Firmware Image

WellGate 2424S is an 24-Line FXS gateway with SIP protocol IP device which allows to connect 24 sets of analog telephone to make or receive VoIP call over Internet or VPN network through Internet Telephony service provider. This device is suitable for office user through ITSP service provider to install at office or branch office to call between different offices. It can be installed at basement of apartment or dweller building to provide analog phone set to each house to make/receive telephony internet call via Telco or ITSP's broadband device and network.

To select up to 16 SIP service Accounts

WellGate 2424S is appropriate to use up to 16 VoIP Service Providers, IP Centrex service and IP-PBX within offices and remote branch offices. One of 16 SIP Servers (or ITSP Service provider or alternative IP-PBX) can be configured freely at each line (FXS port) to make or receive IP Call. It provides 16 service platforms to select lowest rate or different purpose according to your dial number or country or application.

19-inch, 1U chassis easy to install

WellGate 2424S is an 19-inch, 1U chassis and suitable to install at Relay Rack for vast lines installation at office, Telco and ITSP service provider. 1-WAN and 1-LAN with NAT feature together with RS-232 DB-9 local console port allows engineer to configure and maintain this device locally and remotely. It can be installed at outdoor electric pole and sidewalk. There are 24 ports RJ-11 line connectors to connect to telephone cable via existing Telecom MDF. No more



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new cable is needed.

Flexible Dial plan and Route Plan Features

WellGate 2424S provides flexible Dial Plan from FXS to IP Trunk (SIP Soft Switch). Dial Plan is to configure in what condition the digits can be sent out to IP network. The dial inter digit time before dialing is configurable to meet busy users or home user. Dial Rule is able to detect the prefix code and maximum digits reached and then dial out automatically. The Digit Manipulation (DM) allows you to configure matched prefix code, digits length, start and stop digit position to be replaced digits as well.

Routes Plan is to configure the incoming and outgoing call routes which you desire this call to go out or allow to income. For instance, IP incoming call may Ring to FXS port with Priority, Cyclic or Simultaneously ring. You can also configure IP incoming call by Matched prefix digits, Matched incoming FXS line number and Matched digit length. For FXS outgoing call routes, the hunting type supports Priority, Cyclic or Simultaneously ring and select which SIP trunk (SIP Proxy Server) to go. FXS outgoing call routes also support by Matched prefix digits, Matched incoming FXS line number and Matched digit length. Both direction supports No Answer time out and Backup Routes.

SPECIFICATION

Interface:

Ethernet port (RJ-45, 10/100 base-T)

1-WAN port, connect to IP Network

1-LAN port connect to PC with NAT Support Bridge, NAT and Gateway mode

Telephony port (RJ-11 x 24 pcs)

RS-232 Console port, DB9 Male

AC power input Jack

AC Power ON/OFF Switch

LED Indicator for System, SIP and FXS status

IP Network connection

IPv4 (RFC 791) and IPv6 Simultaneously

IPv6 Auto Configuration (RFC 4862)

IPv6 Only, IPv4 Only or dual stack

MAC Address (IEEE 802.3)

MAC Clone Setting

Vendor Class ID

IP/ICMP/ARP/RARP/SNTP

Static IP

DHCP Client (RFC 2131), WAN port

DHCP Server, LAN port

NAT Server (RFC 1631)

PPPoE Client

DDNS (DynDNS)

DNS Client

Firewall

URL Filter

IP Filter

IP Filter

MAC Address Filter

Application program Filter

Port Filter

Port Forwarding

Bandwidth Control (Download and Upload), Maximum Bandwidth

and reserved bandwidth

UPnP Server at LAN port

Behind NAT, use DMZ for NAT traversal SNTP with time zone and Daylight Saving

TCP/UDP (RFC 793/768)

RTP/RTCP (RFC 1889/1890)

IPV4 ICMP (RFC 792),

TFTP Client

VLAN Support 802.1Q, 802.1P

VLAN ID Range: 2 to 4094

VLAN Priority: 0 to 7

QoS: DiffServ (RFC 2475), TOS (RFC791, 1394)

SIP Protocol:

RFC3261 compliance

Support up-to 16 SIP Server Register Accounts

SIP UDP Protocol

Support SIP compact Form

Support SIP HOLD Type

SIP Session Timer (RFC 4028)

Configure SIP port and SIP QoS Type

MD5 Digest Authentication (RFC2069/RFC2617)

SIP PRACK (RFC3262)

Early/Delay Media support

Offer/Answer (RFC3265)

Message Waiting Indication (RFC3842)

Event Notification (RFC3265)

REFER (RFC3515)

Support Outbound Proxy

SIP Proxy Keep Alive time setup

Support Primary and Secondary SIP Server

Support STUN NAT Traversal

Support "rport" parameter (RFC 3581)

Audio Codec :

G.711 A-law/ μ -law, G.729A, G.723.1 (6.3K, 5.3K), GSM-FR Full

Rate (13kbps)

Select voice codec priority: Local or Remote



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Configure RTP port and RTP QoS Type

Silence Suppression

VAD/CNG

LEC: Line Echo Canceller

Max Echo Tail Length (G.168): 32, 64 and 128ms

Packet Loss Compensation Input (Encode) Gain setup

Output (Decode) Gain setup

In-band/out of band DTMF (RFC4733, RFC2833 / SIP INFO)

Adaptive/Configurable Jitter Buffer G.168 Acoustic Echo Cancellation

Dialing Plan with drop, replace, Insert dialing digits

Select First digit and Inter digit timeout duration (Sec)

Selectable Call Progress Tone Support Specified Line Calling

Call Features:

Caller ID display DTMF (before/after 1st ring) and FSK (before 1st ring), ETSI and Bellcore

DTMF Caller ID start and stop BIT(A to F) configurable

Polarity Reversal before Caller ID or not

Tone Generation: Ring, Ring Back, Dial, Busy, call waiting, ROH,

Warning, Holding, Stutter dial tone and disconnect tone

Configure Tone Frequency, Cadence, Level and Cycle

Global Country Based Tone Specification

NAT Traversal support STUN, UPnP and Behind NAT

Out-Band DTMF: RFC2833 and SIP Info Configure DTMF send ON and OFF time

Configure DTMF detect Min. ON and OFF time

DTMF Relay Volume

Flash Time transmit via SIP Info (Enable or Disable)

Message Waiting Indication (Stutter Tone Notice)

Speed Dialing

Call Waiting/Switching between Calls

Call Forward (Busy, Unconditional, No Answer)

No Answer Time out

Block Anonymous Call

Hot Line

Call Hold

Call Transfer

Flexible Dial Plan

First digit and Inter digit time out timer setup

Manual SEND digits selection key

Retrieve transfer call from 3rd party by dial Code (default: *#)

Dial Rule: Prefix and Maximum digits, digit position to replace

Digit Manipulation (Drop and Replace Rule)

Extension Hunting

Support SIP Trunk up to 16 SIP Servers

Import and Export SIP Trunk configuration

Outgoing SIP Caller ID Selection

Accept desired SIP Proxy incoming calls Only

Flexible Routing Plan

Prefix Match and Length

Matched FXS Line (port) number

Incoming call type: FXS or VoIP

Priority Ring

Cyclic Ring

Simultaneous Ring

Programmable Hunting Cycle

Backup Routing with Digit Manipulation

Default Routing

T.38 FAX: ECM, Redundant depth, Volume

FAX Relay: T.38 or Disable Retrieve Voice Mail from IP-PBX

FXS Caller ID Mode: Transparent or Inhibit SIP Caller ID Mode: Transparent or Inhibit

Support Peer to Peer Dialing

Flash Time Detection: range from 80 to 800 ms

ON-HOOK Voltage -48Vdc

Ring Sine Wave frequency: 10 to 70 HZ

Ring Level: 10 to 95 Vrms

Ring REN: 3

Configure Ring ON time: 0 to 8000 ms
Configure Ring OFF time: 0 to 8000 ms
Configure Ring Codones, Fraguency and Volta

Configure Ring Cadence, Frequency and Voltage

Support Polarity reversal for Billing

Service Up to 1 Kilo-meter distance to analog telephone set Generate Current Drop Time (Open Loop Disconnect time)

MANAGEMENT:

Administrative Telnet CLI and HTTP, HTTPS with desired Port number

Enable/Disable HTTPS and Telnet Service

http provision through MAC address

RS-232 console cable to configure WAN/LAN IP Address

Multilingual Web User Interface

3 Levels of User Access Right with Password protection

(Administrator, Supervisor and User)

HTTP/HTTPS Service Access limitation from WAN port

Provides System Status Logs

Status display: Network, Line, SIP Trunk status Diagnostics (debug through Syslog Event Notice)

Debug in real time by Telnet

8 Debug Level: Emergency, Alert, Critical. Error, Warning, Notice,

Information, Debug

Auto Provision via HTTP Server

Support SNMP V2/Trap

Configuration Backup/Restore

Dual Firmware Image Backup

Reset to factory Default

** Support Welltech proprietary encryption protocol at SIP Signal and Voice codec during transmitting to IP network in order to Anti-ISP block of VoIP call. This feature only be available with Welltech SIP server or SIPPBX6200 IP-PBX

Environmental:

Actual Dimension: 44(W)×4.4(H)×26.2(D) CM

19-inch, 1U chassis with Relay Rack Mount Bracket Weight: 4.3kg (One unit with packing)

Operating Temp. & Humidity

- Temp.: 0°C~45°C (32°F~113°F)

- Humidity: 10%~90% relative humidity, non-condensing

Power Input: AC100V to 240V, 50/60Hz



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24-Line FXS SIP IP Gateway

Country of origin:

Made in Taiwan

Packing Accessories

WellGate 2424S gateway

x 1 pcs

Relay Rack Mount Bracket

AC Power cable

CD User Manual

Warranty One year

x 2 pcs

x 1 pcs

x 1 pcs

