CORECASYS FXO-4

4-Line FXO SIP IP Gateway

Core FXO-4 front view

Core FXO-4 rear view

Dual IP Stack : IPv6 and IPv4 Simultaneously
Support up to 4 SIP Trunk Servers
Support different SIP Trunk to each FXO line
Auto HTTP Provision feature
Flexible Routes Plan, Dial Plan, Digit Manipulation
Redundant Firmware Image

Core FXO-4 is an 4-Line FXO gateway with SIP protocol IP device which allows to connect 4 Lines of analog PSTN telephone line or connect to analog extension of PABX to make or receive VoIP call over Internet or VPN network. This device is suitable for office IP-PBX application at office to office or office to branch office to call between PSTN Line and IP Call.

To select up to 4 SIP TRUNK Accounts

Core FXO-4 is appropriate to use four VoIP SIP Trunk or IP Centrex service or IP-PBX within offices and remote branch offices. One of four SIP Servers (or ITSP Service provider or alternative IP-PBX) can be configured freely at each line (FXO port) to make or receive IP Call. It provides 4 service platforms according to your dial number or routes plan.

IPv6 VoIP Gateway is ready to Market

IPv6 address was developed for years, however, it was not practical to our life up to date. More and more electronic devices are able to link to IP Network, this makes existing IPv4 address supply in shortage to global market. Meanwhile, the emerging countries are not able to increase IPv4 address supply due to strong market demand on broadband services. Core FXO-4 is an SIP based FXO gateway which built-in both IPv6 and IPv4 IP address. No matter when you are ready to deploy IPv6 network now, or reserve the future expansion to IPv6 from existing IPv4 address, Core FXO-4 is ready to grow up with you. Both IPv6 and IPv4 address are working simultaneously at Voice IP Call. Its flexibility of both IPv6 and IPv4 accept and interwork both addresses on today and tomorrow whenever you need

Flexible Dial plan and Route Plan Features

Core FXO-4 provides flexible Dial Plan between FXO and IP Trunk (SIP Soft Switch). Dial Plan is to configure in what condition the digits can be sent out to/from IP network. The dialing inter-digit time before dialing is configurable to meet local PSTN line or PBX's extension line. Dial Rule is able to detect the prefix code and maximum digits reached and then dial 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.

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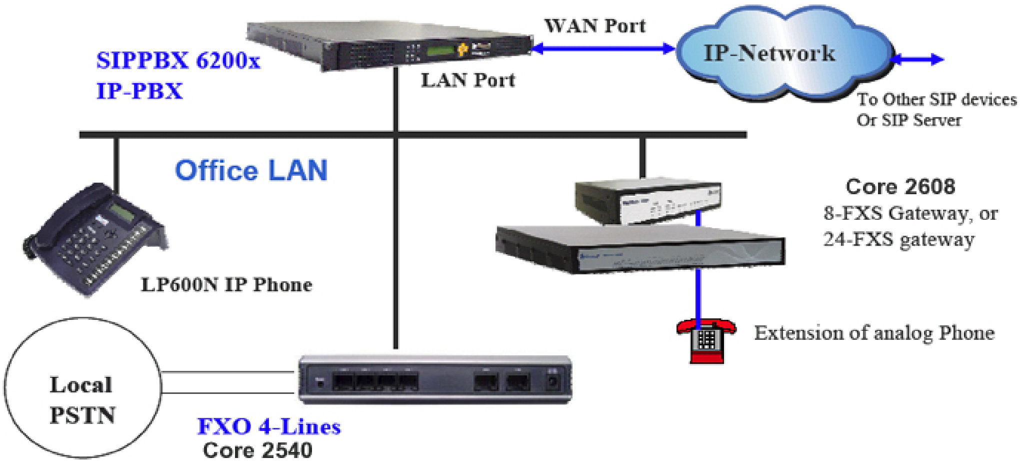
Route Plan is to configure the incoming and outgoing call routes which you desired this call to go out or allow to income. For instance, IP incoming call may Reach to one FXO port with Priority or Cyclic access. You can also configure IP incoming call by Matched prefix digits, Matched dialing number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

Suit to IP-PBX to access local PSTN Line

Core FXO-4 is a SIP IP device to connect with IP-PBX to access local PSTN network with FXO interface. Its telephony features, for instance, Caller ID detection and Releasing FXO port after call was dropped, are easy to integrate with Legend Telephony Line with IP-PBX in office and branch office IP call application. It is compatible with local Telecom network regulation and your office IP network to transmit analog voice between them



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 connect to local PSTN line (RJ-11 x 4 pcs)

DC +12V power input Jack

LED Indicator for System, SIP and FXO 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

MAC Address Filter

Application program Filter

Port Filter , Port Forwarding (TCP, UDP or both)

Bandwidth Control (Download and Upload)URL Filter

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

VoIP VLAN Support 802.1Q, 802.1P

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VLAN ID Range : 2 to 4094
VLAN Priority : 0 to 7 (Highest Priority)
QoS : DiffServ (RFC 2475), TOS (RFC791, 1394)

SIP Protocol :

RFC3261 compliance
Support up-to 4 SIP Trunk to Register
SIP UDP Protocol
Support SIP compact Form
Support SIP HOLD Type: Send Only, 0.0.0.0 or inactive
SIP Session Timer (RFC 4028)
SIP Session Refresher: UAC or UAS
SIP Encryption
MD5 Digest Authentication (RFC2069/RFC2617)
Reliability of provision response PRACK (RFC3262)
Early/Delay Media support
Offer/Answer (RFC3264)
Message Waiting Indication (RFC3842)
Event Notification (RFC3265)
REFER (RFC3515)
Support Outbound Proxy
Support Primary and Backup SIP Server
Support STUN NAT Traversal
Support “rport” parameter (RFC 3581)
Configure SIP local Port
SIP QoS Type: DiffServe or QoS
Accept Proxy Only : YES or NO

Audio Codec :

G.711 A-law/μ-law, G.729A, G.723.1 (6.3K, 5.3K)
Select voice codec priority : Local or Remote
Voice Payload size (ms) configuration
Silence Suppression
VAD/CNG
LEC : Line Echo Canceller
Max Echo Tail Length (G.168): 32, 64 and 128ms
Packet Loss Compensation
Automatic Gain Control
In-band/out of band DTMF (RFC4733, RFC2833 / SIP INFO)
Adaptive/Configurable Jitter Buffer
G.168 Acoustic Echo Cancellation
Configure RTP basic Port
RTP QoS Type : DiffServ or TOS
Phone Book (50 records) for peer to peer calls
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 :

4-Line FXO connect to PSTN or PBX simultaneously
Caller ID recognition DTMF (before/after 1 ring) and FSK (before 1st ring), ETSI and Bellcore
DTMF Caller ID start and stop BIT configurable
Current Drop Detection to release FXO port
Disconnect tone recognition to release FXO port
Tone Generation: Ring Back, Dial, Busy, call waiting, ROH Warning, Holding, Stutter dial tone and disconnect tone
Configure Tone Frequency, Cadence, Level and Cycle
Select Tone specification by Country name List
Global Country Based Tone Specification
NAT Traversal support STUN, UPNP and Behind NAT
Out-Band DTMF : RFC2833 and SIP Info
RFC2833 Payload type : 101 or 96
DTMF send out ON and OFF Time configure
DTMF incoming recognition Minimum ON and OFF time
DTMF Relay Volume configuration
T.38 FAX Volume configuration
Flash Time transmit via SIP Info (Enable or Disable)
Message Waiting Indication (Stutter Tone Notice)
Block Anonymous Call
Call Hold
Call Transfer
FXO Line Configuration
Activate or deactivate
Line ID
FXO Line Phone number
Polarity Reversal detection for call establish and Billing
Current drop recognition to release port
Incoming call Handle: Hotline or 2 stage dialing
HOT Line to desired phone number
Play voice file to incoming call
Repeat playing voice file counts
Self-recorded voice files to upload
Generate FLASH TIME to PSTN network
T.38 or FAX Relay Type
Incoming and outgoing dB value configurable
Dialing Answer Delay time to establish call path
Answer PSTN incoming call after how many ring cycles
Caller ID detection mode by Country selection
VoIP dial to FXO/PSTN Line: 1 stage dialing and 2 stage dialing
Outgoing SIP Caller ID Selection
Support 4 SIP Trunk
Accept desired SIP Proxy incoming calls Only
Flexible Routing Plan
Prefix Match and Length
Priority Ring
Cyclic Ring
Simultaneous Ring
Programmable Hunting Cycle
Backup Routes with Digit Manipulation
Default Routes
Flexible Dial Plans
Retrieve transfer call from 3 rd party by dial Code (default: *#)
Inter digit time out setting
First digit dial out delay time setting
End of dial keypad number
Dial Rule : Match dial Prefix and Maximum digits length (1-15)
Phone Book can be Exported or Imported
Digit Manipulation (Drop and Replace Rule):
FXO DM Group
VoIP DM Group

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DM 1 Group
DM 2 Group
DM 3 Group
DM 4 Group
Matched Prefix
Matched digit length
Replace digit start position
Replace digit stop position
Replace number
Incoming Ring frequency recognition range: 10 to 70 Hz
Incoming Ring ON time recognition range: 0 to 8000ms
Incoming Ring OFF time recognition range: 0 to 8000ms
Incoming Ring Level recognition range: 10 to 95Vrms
Support Peer to Peer Dialing
Flash Time Detection: range from 80 to 800 ms
Configure Ring Cadence, Frequency and Voltage

Management :

Administrative Telnet CLI and HTTP, HTTPS
HTTP provision through MAC address
Multilingual Web User Interface
3 Levels of User Access Right with Password protection with different Web Language (Administrator, Supervisor and User)
HTTP/HTTPS Service Access limitation from WAN port
Configure Service ports at HTTP, HTTPS and telnet Services
Phone Debug Module: Device Control, Call Control, DB, Verbose
SIP Debug Module: Register, Call, SIP Message, Others
SNTP Debug Module
Device Debug Module
DSP Debug
Provide 8 Debug Levels :
Emergency
Alert
Critical
Error
Warning
Notice
Information
Debug
Provides System Status Logs
Connect to external SYSLOG Server
Status display: Network, Line, SIP Trunk status
Diagnostics (debug through Syslog Event Notice)
Debug in real time by Telnet
Auto Provision via HTTP Server
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 core SIP server or SIPPBX6200 IP-PBX

Environmental :

Actual Dimension: 17.5(W) ี 3.2(H) ี 12.6(D) CM
Weight: 0.5kg (One unit with packing)

Power Adaptor

INPUT: AC100V~240V, 50/60Hz
OUTPUT: DC 12V, 1.5A

Approvals:

CE, FCC (Part 15, Class B), LVD and RoHS

Country of origin:

Made in Taiwan

Warranty

One year

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