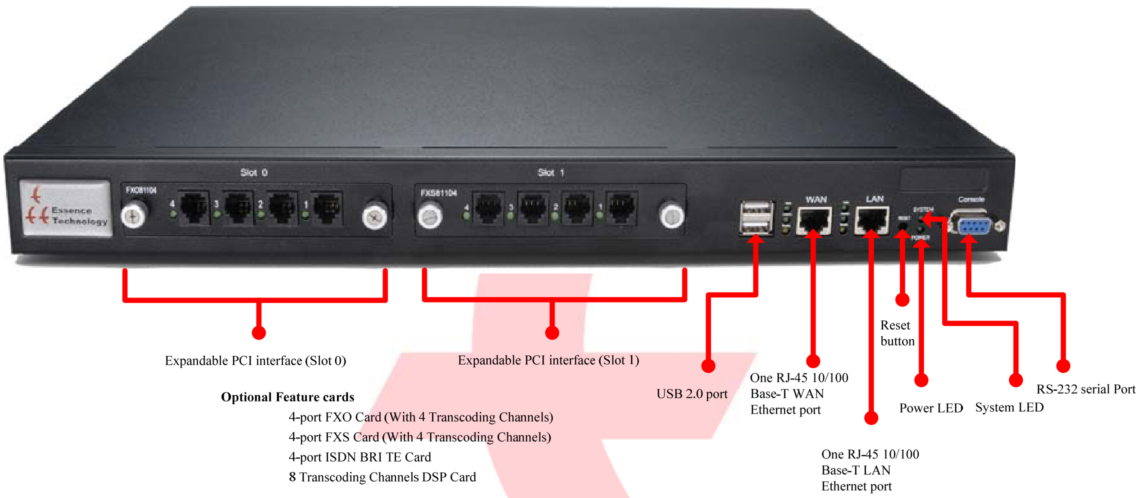
Essence®
Technology

What you need today is Essence IPBX for tomorrow's productivity

Energize your business with effective Essence IPBX system which comes with easy setup and administrator-friendly designs. Essence IPBX system delivers you the best---for its high technology integration that can maximize the work efficiency of *any business* and is worth of every dollar of your investment.



Front View of IPBX-1825

System Highlight

- Highly integrated, embedded system for more stability and worth of your investment
- Immediately provides VoIP connection to ITSP and to any remote site
- Analog interface to PSTN
- Seamless integration with legacy PBX
- Flexible dialing plan settings for applying to a virtual IP-PBX system
- Off-ramp and on-ramp call for VoIP and PSTN
- Remote management capability
- Quick-batch configuration by Command Line Interface (CLI)

Features Highlight

- Multi-lingual voice prompts for international business
- Customizable 3-layer IVR for creating your own IVR scenarios
- ACD (Automatic Call Distribution) to form basic Call Centers for facilitating small-medium enterprise's needs
- Various automatic schemes including auto provision and auto firmware upgrade
- Stackable design for scalability and for preserving your pervious investment
- UMS (Unified Messaging feature) supported
- Meet-me conference service to expend you meeting room virtually
- Newly released wizard function for easier configuration
- Function-rich Voice Mail System to manage your voice message efficiently
- Voice Mail System with E-mail Notification
- IP Intercom/ IP Paging
- D-Auth (Dialer Authentication)
- Time-based Memo Call
- Scheduled Broadcast Event
- Group Call

- Caller ID/DISA
- CAC (Call Admission Control)
- Call keep alive scheme
- Most legacy PBX features supported
- CDR (Call Detail Record) with .CSV file format
- FAX relay and pass through (T.38 and T.30)

SIP Compliance

Supported Standards
RFC 3261, RFC 3311, RFC 3515, RFC 3265, RFC 3892, RFC 3361 RFC 3842, RFC 3389, RFC 3489 RFC 3428, RFC 2327, RFC 2833 RFC 2976, RFC 3263, RFC 3264 RFC 3362, RFC 4612,

SIP Registrar
- Static/Dynamic registration
- Configurable Expiry Time
- MD5 authentication
- Handle loose RFC-compliant SIP devices
- Resilient message retry mechanism
- Cache client registrations

SIP Proxy
- Stateful proxy server
- NAT traversal for clients
- Inter-proxy call hand-off
- Outbound Proxy behind NAT Device

PBX System

Call Features
- 200 users and extensions with Voice Mail account
- 50 concurrent sessions[2]
- Codec G.711 (μ /A-law), G.723.1 (6.3k/5.3k bit/s), G.729A, and G.726 (16k/24k/32k/40k bit/s) supported
- Transcoding channel 0~16, subject to add-on card
- In-band/RFC2833/SIP-INFO DTMF translation
- Two expandable slots for telephony interfaces
- 50 SIP trunks for ITSP account or private trunking shared by extensions
- 200 DID SIP trunks to extensions
- Support gateway trunk mode per SIP trunk
- Enable/Disable NAT Traversal per SIP trunk
- Call admission control of call count or bandwidth per SIP trunk
- Long call audit
- Support Call keep alive
- Support Registered keep alive
- NAT session keep alive
- Configurable RFC 2833 payload type per SIP trunk
- FXS/FXO analog trunking
- FXO disconnection tone detection
- FXO disconnection tone parameter setting
- FXS hot line
- FXS warm line
- Caller ID detection
- Trunk hunting
- Digits manipulation during hunting
- Life-line priority call
- Support SIP Call Hold, Call Waiting
- Support SIP phone[5] 3-way conference
- Support Blind/ Attended Transfer
- In-line Call Transfer
- Unconditional, Unavailable, Busy Call forward
- Call Back on Busy between extensions
- Per calling number forward and rejection
- Blacklist of number patterns
- 32 call pick-up groups
- Call Park and Retrieve
- Recording on demand with Essence IP phones
- Remote extension registration via Internet
- Direct line to extension (DID to Extension)
- Direct line by called number (DID by Number)
- Direct line by privilege (DID by Privilege)
- Echo Cancellation (G.168)
- Flexible numbering plan
- Call privilege grouping
- Configurable Music on Hold
- Memo Call for extension
- Schedule-based Broadcast
- Support T.38 FAX over IP
- Support T.30, T.38 FAX pass through
- ENUM resolution
- Auth. Dial passcode
- Group Call
- Support H.261, H.263, H.264, MPEG-4 and MJPEG Video codec pass through
- Peer to Peer (Invite/Update)
- Fast Bridging for expressing media forwarding

NAT
- Auto NAT discovery and traversal
- Built-in STUN client
- RTP proxy
- RTP port range designation

IVR
- 50 configurations of 3-layer IVR
- Worktime/Holiday setting for different IVR
- Configurable greeting prompts
- Music on Ringing extensions
- Forward to Voice Mail on No-answer
- Support 3 languages in IVR tree
- Hot key to operator

Voice Mail
- User Authentication by PIN
- Multilingual, 3 languages
- Multi-folder Archive
- Fast-forward /Rewind /Undelete
- MWI notification
- VMWI notification
- E-mail notification and attachment (Unified messaging)
- Personal greeting on unavailability and busy
- Record personal greeting through phone
- Voicemail Forwarding
- Reply call or new call after logged in Voicemail menu
- Built-in 40GB hard disk drive for Voicemail
- Support USB 2.0 interface for Voicemail, CDR, and system configuration backup
- Support NFS remote backup for Voicemail, CDR, and system configuration

Meet-me Conference
- 24 conference rooms with configurable number and PIN
- Up to 24 parties[4,1] among all conference rooms
- Lock/Mute/Join/Drop control for administrator
- Music on First Dial-in Party
- Hot key to leave the conference
- Hot key for administrator to manage the conference

Automatic Call Distribution
- 32 queues with 32 agents among all queues.
- 32 inbound call among all queues
- Configurable waiting length for

individual queue
- Support five distribution policies including round robin, ring all, least recent, fewest call, and random
- Configurable waiting time for each queue
- Allow agent remotely log-in
- Agent can participate multiple queues
- Agent phones also allow extension calls

Stackable

- Support LAN stacking up to 4 units in the same model
- Automatic intra-trunking creation among stacking units
- Automatic configuration publishing from Master to Slaves
- Automatic load balancing in hosting feature phones[3]

Administration

System Management

- Web-based configuration with session control
- User and administrator configuration mode
- Automatic expiring the idle sessions
- Support firmware upgrade through the Internet
- Configuration Wizard for mass extensions and users creation
- Step-by-Step Wizard for adding users, extensions and trunks
- Built in online help in wizard
- Command Line Interface (CLI) for configuration
- System event Syslog
- Downloadable Call Detail Record (CDR)
- Extension registration status
- Active call status
- TFTP server and TFTP repository maintenance
- NTP synchronization
- Real Time Clock setting
- DHCP Server with multiple partitions, Per-MAC IP binding, list of options
- Configurable Time Zone
- Firmware Upgrade through web interface and console

Network Management

- DHCP/PPPoE/Static IP on WAN
- Support MAC Clone on WAN
- Allow WAN to Respond PING
- Allow LAN use only
- Static LAN routing
- Firewall on predefined services
- Virtual Server for client device
- NAT for outbound traffic from LAN
- WAN QoS queuing mechanism for VoIP and data traffic
- Support TOS setting
- DNS forwarder and dynamic DNS
- SNMPv2 with standard MIB format
- Adaptive WAN bandwidth and DSP channel saving

Optional Feature cards

- FXO81104, 4-port FXO card (with 4 transcoding channels)
- FXS81104, 4-port FXS card (with 4 transcoding channels)
- BRI 81004, 4-port ISDN BRI TE card
- DSP81000, DSP card with 8 transcoding channels
- IPSec 81003

Hardware Specification

Hardware Interfaces

- One RJ-45 10/100 base-T WAN Ethernet port
- One RJ-45 10/100 base-T LAN Ethernet port
- Two USB 2.0 ports
- One RS-232 serial port
- Two expandable PCI interface slots

System Dimension

- 443 x 315 x 44 (mm), 1U rack mount

System Power Requirement

- Power input 100~240V AC, 50~60 Hz
- 40 W (max)

Environment

- Operating temperature 0~50℃
- Storage temperature -10~70℃
- Humidity (RH) 10~80% non-condensing

Regulatory and Safety

- FCC Class A certified, FCC part 68, CE/EMC/LVD/TBR21, VCCI, JATE, ROHS

1. Transcoding channel: If the caller and the callee use different codecs in a session, IPBX will use a transcoding channel to translate the codecs. Some IPBX services consume the transcoding channel to service the end devices which only support one codec such as IVR, Voice Mail, and Meet-me Conference. The number of transcoding channel in 1825 series is subject to the add-on card.
2. A current session is a communication session with two call parties, the caller and callee. For example, extension 101 calls to extension 102 occupies one concurrent session.
3. Feature phone means Essence EIP7012 IP Phone series
4. The number of transcoding parties is subject to the transcoding channels of IPBX. Maxmum number of transcoding party is 8.
5. Essence EIP 7012 IP Phone series are fully compliance with this feature.

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