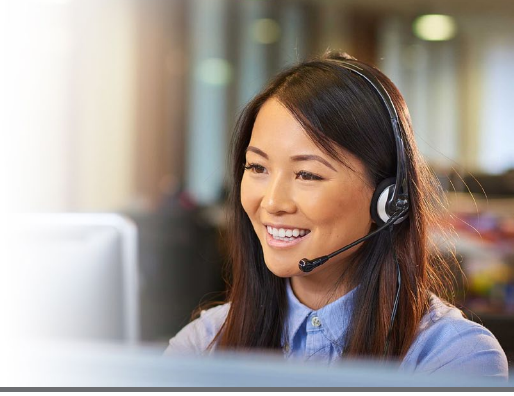


Ribbon Application Server for Enterprise & Government

The Ribbon Application Server (AS) is a high-performance communications platform, initially designed to meet the scalability and reliability requirements of the world's largest service providers. Modern virtualized software and versatile off-the-shelf hardware make it cost-effective to scale this same platform up or down to meet the deployment needs of enterprises and government agencies of all sizes. The platform's unique Service Anywhere architecture enables geographically dispersed deployments to act as a common platform. Nomadic users can register to a local instance and immediately inherit all of their capabilities and permissions. The design is so resilient that even if an entire instance were destroyed, users could register to another deployment instance and immediately regain their local services. Most importantly, it offers true enterprise-class communications features and services using industry-standard SIP-based clients and devices.

The Ribbon Application Server can replace legacy PBX systems, centralize or flatten campus communication networks, and elevate conversations to support Unified Communications (UC) services. The Application Server can be deployed as a stand-alone PBX, an Over-The-Top (OTT) services platform or centralized session manager. Its virtualized architecture is at home in local or centralized data centers as well as private or public clouds. The platform and ecosystem is security hardened solution, including being certified for deployment as a Local Session Controller (LSC) and Enterprise Session Controller (ESC) by the United States Department of Defense Joint Interoperability Test Command (JITC) and it meets the Center for Internet Security (CIS) security standards. The AS is also one of the most proven communications platforms in the world, with over 30 million seats sold among enterprise, carrier, and defense customers.

Application Server 15.2 Key Capabilities

- Rich Business Features on Open SIP Phones (digital phone emulation)
- Robust Unified Communications Clients and Services
- Integrated Mobility Services and Fixed-Mobile Convergence
- Fully Virtualized Design – Cloud-Ready
- Flexible Deployment Model as Standalone or C20 Adjunct
- High Availability Architecture with Geographic Survivability
- Stateful (Active) Call Failover
- Native E911 Support Extends Emergency Location Information to Nomadic Users
- Cost-effective Scalability from 100 to 2 Million Subscribers
- Multi-tenant Support with up to 100,000 Partitions
- Support for up to 6 layers of SIP Domains and Sub-Domains
- Military Grade Security and Certifications (JITC)
- Support for MLPP, including Shared Line Appearance
- Carrier Grade Reliability, Manageability, and Scalability
- REST and SOAP API Integration with Open Programmability Suite
- Rich Feature Set for Residential, Analog or Guest Users

Ribbon Application Server

- Over 30 Million Seats Sold in 62 Countries
- Scales from 100 to 2 Million SIP Endpoints
- 62,500 Simultaneous Calls per Session Manager (500,000 Max)
- 300 TLS Device Registrations per second per Session Manager (w/SBC)
- Virtualization on RHEL KVM and VMWare
- US DoD JITC Certified As Local Session Controller (LSC) and Enterprise Session Controller (ESC)

A Better Way to Deliver Communications Service

The Ribbon Application Server brings a simplified approach to Communications Services as compared to traditional PBX systems. Examples include:

- Route calls based on user assigned aliases instead of designing complex dial plan masks
- Assign users to a domain and sub-domain; use rules for each sub-domain to customize: call routing, local E911, Music on Hold, announcements, site/department billing data, and more
- Act as a centralized session manager to centrally provision services (including mobility and UC services) for end users on any PBX system connected to the Application Server via SIP trunks
- Deliver traditional "digital phone features" on standards-based SIP phones
- Leveraging the solutions' scale to reduce deployments costs by consolidating or centralizing users into a flat communications infrastructure

Application Server 15. 2 Infrastructure Requirements

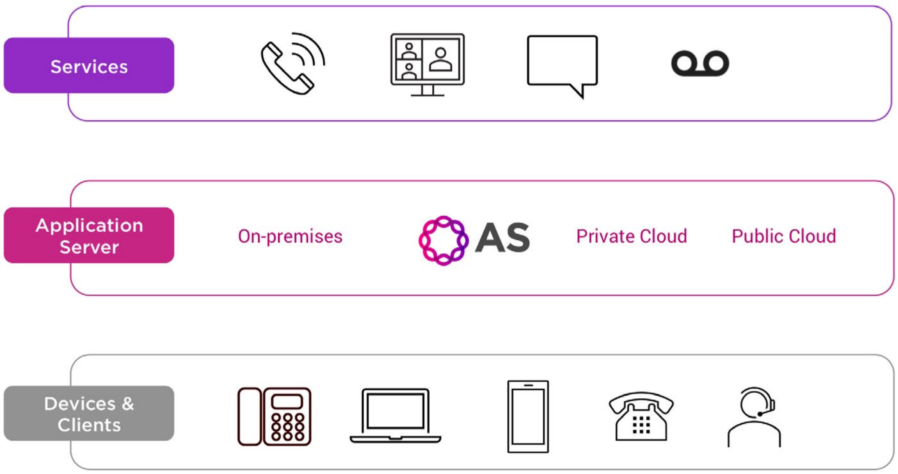
Requirements	Description
Intel-based Server Infrastructure	– 64 bit Intel (AMD not supported), 20 MB cache minimum – BIOS supports VT-x extensions/hyper-threading – Preferred - dual hot-swap power supplies – Quad port NIC, auto-negotiable – Hard drive storage replicated in RAID1 configuration at a minimum
Virtualization Software	– RHEL KVM or VMWare
VMware Certified Hardware	– https://compatibilityguide.broadcom.com/
Geographic Survivability / HA Networking Requirements	– <50ms one-way delay between components – Use Layer 2 networking between data centers or use Layer 3 for geographically resilient Service Anywhere deployments – Bandwidth between data centers is configuration specific

Application Server 15.2 Ribbon Endpoints

Manufacturer	Device Type	Family	Models
Poly	Phone	VVX	100, 101, 200, 201, 300, 400, 500, 600
Poly	Phone	VVX	150, 250, 350, 450, 501, 601
Poly	Phone	CCX	400, 500, 600, 700
Poly	Phone	D	230
Poly	Video Bar	X	30, 50
Poly	Phone	TRIO	8300, 8500, 8800, C60
Poly	Gateway	Obi	200, 300, 302, 202, 212, 508, 312, 504
Poly	Phone	SoundPoint	330, 331, 501, 550, 560, 650, 670
Poly	Phone	SoundStation IP	5000, 6000, 7000
Poly	Conferencing	Clariti	Clariti
Cisco	Phone	MPP	6800, 7800, 8800
Cisco	Phone	CUCM (K-)	6800, 7800, 8800
Cisco	Gateway	SPA	112, 122
Tone Commander	Phone	IP Phone	7810, 4104, 4101
Avaya	Phone	IP Phone	1120, 1140, 1210, 1220, 1240
Avaya	Phone	J	139, 159, 179
Avaya	Phone	oneX	9608, 9621, 9631, 9641
Yealink	Phone	T2	T29G, T27G, T23G
Yealink	Phone	T2	T29P, T27P, T23P
Yealink	Phone	T3	T33G, T31G
Yealink	Phone	T3	T33P, T31P
Yealink	Phone	T4	T40G,
Yealink	Phone	T4	T40P, T41S, T42S, T46S, T48S, T43U, T46U, T48U
Yealink	Phone	T5	T53, T53W, T54W, T56A, T57W, T58A, VP59
Yealink	Phone	CP	920, 930W, 960
Grandstream	Phone	GXP	2170, 2160, 2140, 213x,
Grandstream	Phone	GXP	178x, 1760, 1630, 162x, 161x,
Grandstream	Phone	GXV	3380, 3370, 3350
Grandstream	Gateway	GXW	4004
Grandstream	Gateway	GXA	4008, 4104
AudioCodes	Gateway	MediaPack	114, 118, 124, 202, 204, 1288

Please review Ribbon's product documentation for software release information and tested capabilities

Carrier grade virtualized architecture - deploy in your network or in the cloud



Application Server 15.2 Features

Account Codes	Dual CLI	Presence-based Routing
Active Directory Integration	Emergency Call Handling	Private And Public Name And Number Display
Ad Hoc Conferencing	Enhanced 911 Service (native)	Program Key Support
Advanced Screening and Routing services	Equal Access Subnet	Redirect
Alias Routing Service	Eternal Element – Status Audit	Reject Reasons
Alpha Tagging	Fixed Mobile Extension	Reminder Ring
Announcements	Group Chat	Response Code Mapping
AS Intercom	Global Address Book	Roaming
Assistant Console	Group Intercom	Screen Notifications (Polycom)
Assistant Support	Group Intercom - All Calls	Secure Warning Announcement
Assured Services Access Control (ASAC)	Hold	Service Nodes
Assured Services – SIP (AS-SIP)	Hotline	Sequential Ringing
Authorization Codes	Hunting and Hunt groups	Shared Line Appearances (MADN) - Multi Call
Call Admission Control (CAC)	Ignore	Shared Line Appearances (MADN) - Single Call
Call Forward	Instant Messaging	Simultaneous Ring
Call Grabber	JITC Certification	SIP Call Control - Native
Caller ID (ANI)	Language Selection	SIP PBX
Caller ID Blocking	LDAP Integration	SIP PBX Accounting
Caller ID Name And Number	Location Selection	SIP PBX Authentication Service
Caller ID Update on Transfer	Malicious Call Trace Support	SIP PBX Call Admission Control – System-wide
Calling Line ID Restriction (Privacy)	MeetMe Audio Conferencing	SIP PBX Emergency Call Options Service
Call Park and Retrieve	Message Waiting Indicator (MWI) Support	SIP PBX Emergency Default Location Routing
Call Park Notification	Missed Call Email Notification Service	SIP PBX Feature Interactions
Call Pickup	Mobile Extension	SIP PBX Link Transmission Mode
Call Origination	Mobile VPN Services	SIP PBX Processing Multiple Q-valued Contacts
Call Redirections through 3xx and 603	Multiple Call Restriction	SIP PBX International Dial Plan Support
Call Screening and Routing	Multiple Device/Client Registration Support	SIP PBX History-Info Header Information
Call Termination	Multiple Level Priority and Pre-emption (MLPP)	SIP Trunk Group Routing to Gateways
Call Transfer - Blind	Music On Hold (multiple variants)	SIP Trunk Group Routing to PBXs
Call Transfer - Consultative	Mutual Authentication	SRTP-based Media Support
Call Transfer Blocking	Network-based Call Logs	Transfer and Ad-Hoc Conferencing
Call Transfer Connected Party Display	Network Call Waiting Disable	TR-87 based Phone Control
Call Subject Presentation	Network Speed Dial	Transport Layer Security (TLS) Support
Call Waiting	Network Transformation Overlay Services	Transport Link Scenarios
Click to Call	PA Carrier	Trunk Group-based Translations
Closed User Groups	PBX Router	Telephony Routes
CODEC Selection	P-Charging-Vector AMA Correlation	Tone Selection
Common Access Card Support	Picture ID	Uniform Call Distribution
Customizable Audio Services	Point to Point Video	Voice Mail Integration
Country Tones	Portal	VSC-based Services
Device Restrictions	Pre-paid Support via Ro Interface	Wake-up Call
Distinctive Alerting	Presence	Web Collaboration Support
Do Not Disturb	Presence Federation (SIP & XMPP)	Whisper Mode Conferencing

Application Server IETF Compliance

RFC	Description	Support
RFC 3581 (draft-ietf-sip- symmetric-response	An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing	Supported
RFC 3605 (draft-ietf- mmusic-sdp4nat)	Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP)	Supported
RFC 3665 (draft-ietf- sipping-basic-call-flows)	Session Initiation Protocol (SIP) Basic Call Flow Examples	Supported
RFC 3666 (draft-ietf- sipping-pstn-call-flows)	Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows	Supported
RFC 3680 (draft-ietf- sipping-reg-event)	A Session Initiation Protocol (SIP) Event Package for Registrations	Partially Supported
RFC 3711 (draft-ietf-avt- srtp)	The Secure Real-time Transport Protocol (SRTP)	Supported
RFC 3725 (draft-ietf- sipping-3pcc)	Best Current Practices for Third Party Call Control (3pcc) in the Session Initiation Protocol (SIP)	Partially Supported
RFC 3842 (draft-ietf- sipping-mwi)	A Message Summary & Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)	Partially Supported
RFC 3856 (draft-ietf- simple-presence)	A Presence Event Package for the Session Initiation Protocol (SIP)	Supported
RFC 3857 (draft-ietf- simple-winfo-package)	A Watcher Information Event Template- Package for the Session Initiation Protocol (SIP)	Supported
RFC 3858 (draft-ietf- simple-winfo-format)	An Extensible Markup Language (XML) Based Format for Watcher Information	Supported
RFC 3863 (draft-ietf-impp- cpim-pidf)	Presence Information Data Format (PIDF)	Supported
RFC 3891 (draft-ietf-sip- replaces)	The Session Initiation Protocol (SIP) “Replaces” Header	Supported
RFC 3892 (draft-ietf-sip- referredby)	The Session Initiation Protocol (SIP) Referred-By Mechanism	Supported
RFC 3903 (draft-ietf-sip- publish)	Session Initiation Protocol (SIP) Extension for Event State Publication	Supported
RFC 3911 (draft-ietf-sip- join)	The Session Initiation Protocol (SIP) “Join” Header	Supported
RFC 3966 (draft-ietf-iptel- rfc2806bis)	The tel URI for Telephone Numbers Partially	Supported
RFC 3994 (draft-ietf- simple-iscomposing)	Indication of Message Composition for Instant Messaging	Supported
RFC 4028 (draft-ietf-sip- session-timer)	Session Timers in the Session Initiation Protocol (SIP)	Supported
RFC 4040 (draft-ietf-avt-rtp- clearmode)	RTP Payload Format for a 64 kbit/s Transparent Call Partially	Supported
RFC 4119 (draft-ietf- geopriv-pidf-lo)	A Presence-based GEOPRIV Location Object Format	Supported
RFC 4235 (draft-ietf- sipping-dialog-package)	An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)	Supported for draft 06
RFC 4244 (draft-ietf-sip- history-info)	An Extension to the Session Initiation Protocol (SIP) for Request History Information	Supported for draft 06
RFC 4245 (draft-ietf- sipping-conferencing- requirements)	High-Level Requirements for Tightly Coupled SIP Conferencing	Supported for draft 00
RFC 4353 (draft-ietf- sipping-conferencing- framework)	A Framework for Conferencing with the Session Initiation Protocol (SIP)	Supported for draft 05
RFC 4538 (draft-ietf-sip- target-dialog)	Request Authorization through Dialog Identification in the Session Initiation Protocol (SIP)	Supported
RFC 4566 (draft-ietf- mmusic-sdp-new)	SDP. Session Description Protocol	Supported
RFC 4567 (draft-ietf- mmusic-kmgmt-ext)	Key Management Extensions for Session Description Protocol (SDP) and Real Time Streaming Protocol (RTSP)	Supported
RFC 4568 (draft-ietf- mmusic-sdescriptions)	Session Description Protocol (SDP) Security Descriptions for Media Streams	Supported
RFC 4579 (draft-ietf- sipping-cc-conferencing)	Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents	Supported for draft 07
RFC 4629 (draft-ietf-avt- rfc2429-bis)	RTP Payload Format for ITU-T Rec. H.263 Video	Supported for draft 06
RFC 4694 (draft-ietf-iptel- tel-np)	Number Portability Parameters for the “tel” URI Partially	Supported
RFC 4733 (draft-ietf-avt- rfc2833bis)	RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals	Supported
RFC 5031 (draft-ietf-ecrit- service-urn)	A Uniform Resource Name (URN) for Emergency and Other Well-Known Services	Partially Supported
RFC 5359 (draft-ietf- sipping-service-examples)	Session Initiation Protocol Service Examples	Supported for draft 09
RFC 5373 (draft-ietf-sip- answermode)	Requesting Answering Modes for the Session Initiation Protocol (SIP)	Partially Supported
RFC 5589 (draft-ietf- sipping-cc-transfer)	Session Initiation Protocol (SIP) Call Control - Transfer	Supported for draft 05
RFC 5626 (draft-ietf-sip- outbound)	Managing Client-Initiated Connections in the Session Initiation Protocol (SIP)	Partially Supported
RFC 5727 (draft-peterson- rai-rfc3427bis)	Change Process for the Session Initiation Protocol (SIP) and the Real-time Applications & Infrastructure Area	N/A
RFC 5806 (draft-levy-sip- diversion)	Diversion Indication in SIP	Supported for draft 08
RFC 5923 (draft-ietf-sip- connect-reuse)	Connection Reuse in the Session Initiation Protocol (SIP)	Supported for draft 08