Ribbon Application Server 14.0

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. The advent of fully virtualized software and simplified support for off-the-shelf hardware makes it cost effective to scale this same platform up or down to meet the deployment needs of enterprises of all sizes. That means enterprises gain cost effective access to a highly available, high-capacity, and feature-rich platform that natively supports SIP communications. Most importantly, it offers true enterprise-class communications features and services.

The Ribbon Application Server can replace legacy PBX systems, centralize or flatten enterprise 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 fully virtualized architecture is at home in local or centralized data center as well as private or public clouds. The platform and ecosystem is a security hardened solution that has been 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 27 million seats sold among Enterprise, Carrier, and Military customers.

Application Server 14.0 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 – Private/Public 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 10,000 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
- SOPI and REST API Integration with Open Programmability Suite
- Rich Feature Set for Residential, Analog or Guest Users

Ribbon Application Server

- Over 27 Million Seats Sold in 62 Countries
- Scales from 100 to 2 Million SIP Endpoints
- 62,500 Simultaneous Calls per Session Manager (500,000 Max)
- 55 TLS Device Registrations per second per Session Manager
- Cloud Ready Virtualization on RHEL KVM and VMWare
- Public Cloud Ready
- 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 14.0 Infrastructure Requirements

Requirements	Description
Intel-based Server Infrastructure	– Sandy Bridge or better processor – 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://www.vmware.com/resources/compatibility/search.php
Geographic Survivability / HA Networking Requirements	– <50ms one-way delay between components – Layer 2 networking between data centers – Bandwidth between data centers is configuration specific

Application Server 14.0 Engineering Capacities

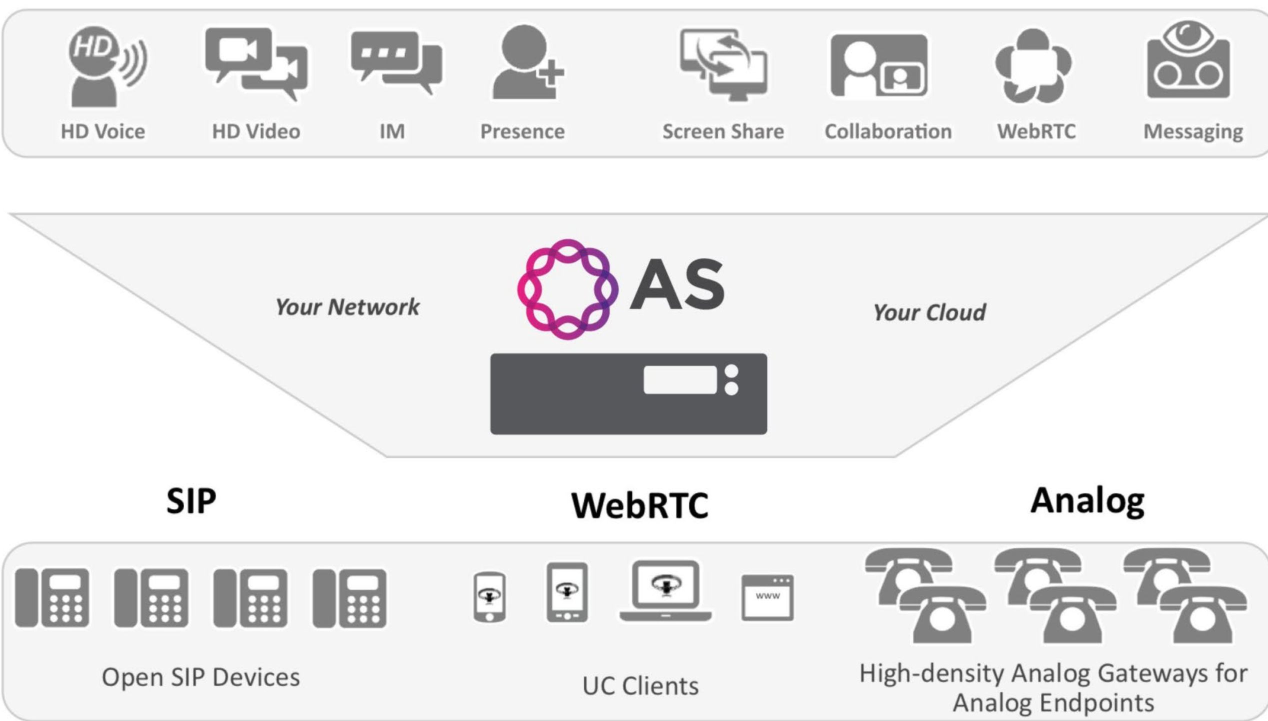
The Ribbon Application Server supports a variety of configurations to efficiently scale services from hundreds to millions of users. To minimize up front investments, modular server instances are deployed as the solution scales.

Maximum Size	"East" Server Requirements	"West" Server Requirements
100-50,000 Endpoints (High Call Volume)	20 Cores 30 GB RAM 370 GB Storage	20 Cores 30 GB RAM 370 GB Storage
100-125,000 Endpoints (High Call Volume)	36 Cores 62 GB RAM 600 GB Storage	36 Cores 62 GB RAM 600 GB Storage
125,000 - 1,000,000 Endpoints (High Call Volume)	90 Cores 132 GB RAM 1650 GB Storage	82 Cores 124 GB RAM 1570 GB Storage

Application Server 14.0 Ribbon Endpoints

Verified Supports Shared Line Appearance (MADN)

Poly	Yealink	Nortel/Avaya	Ribbon EdgeMarc	Cisco
Polycom VVX 1500	Yealink SIP-T57W	Avaya 1140	EdgeMarc 6000	Cisco 88xx MPP Series -
Polycom VVX 450	Yealink SIP-T54W	Avaya 1120	EdgeMarc 2900 Series	(8865, 8861, 8851, 8845, 8841, 8811)
Polycom VVX 350	Yealink SIP-T53W	Avaya 1230	EdgeMarc 7000 Series	Cisco 78xx MPP Series -
Polycom VVX 250	Yealink SIP-T53	Avaya 1220	EdgeMarc 4808	(7861, 7841, 7821, 7811)
Polycom VVX 150	Yealink SIP-T48S	Avaya J series	EdgeMarc 4806	Cisco 68xx MPP Series -
Polycom VVX 601	Yealink SIP-T46S	(J129, J139, J169, J179)	EdgeMarc 4601/4603	(6871, 6861, 6851, 6841, 6821)
Polycom VVX 501	Yealink SIP-T42S	Avaya 9640	EdgeMarc 4570/4571	Cisco SPA504G, SPA303, SPA302D
Polycom VVX 401/411	Yealink SIP-T41S	Avaya 9621G	EdgeMarc 4550/4552	**Grandstream**
Polycom VVX 301/311	Yealink VP-T49G	Avaya 9611G	**Mediatrix**	Grandstream GXW4004
Polycom VVX 201	Yealink SIP-T48G	Avaya 9608	Mediatrix Sentinel	Grandstream GXW4008
Polycom VVX 101	Yealink SIP-T46G	**AudioCodes Phones**	Mediatrix C710	Grandstream GXW4104
Polycom D60	Yealink SIP-T42G	AudioCodes C450 HD	Mediatrix C711	AudioCodes Gateways
Polycom Trio Series 8800/8500	Yealink SIP-T40G	AudioCodes 450 HD	Mediatrix C730	AudioCodes M800
Polycom Soundstation 7000	Yealink SIP-T4xP and T2xG/P Series -	AudioCodes 445 HD	Mediatrix C733	AudioCodes MP-114
Polycom Soundstation 6000	T41P,T40P,T29G,T23G,T23P,T21P,T19)	AudioCodes 440 HD	Mediatrix C731	AudioCodes MP-118
Polycom Soundstation 5000	Yealink W60P/W53P/CP930W	AudioCodes 430 HD	**SpectraLink**	AudioCodes MP-124D
Polycom SoundPoint Family	Yealink CP960/CP920/CP860	AudioCodes 420 HD	SpectraLink 8440	AudioCodes Mediant 1000/3000
(650,560,550,335,331,321)		AudioCodes 405 HD		**Adtran**
				Adtran 924



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

Application Server 14.0 Features

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

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

About Ribbon

Ribbon Communications (Nasdaq: RBBN) delivers communications software, IP and optical networking solutions to service providers, enterprises and critical infrastructure sectors globally. We engage deeply with our customers, helping them modernize their networks for improved competitive positioning and business outcomes in today's smart, always-on and data-hungry world. Our innovative, end-to-end solutions portfolio delivers unparalleled scale, performance, and agility, including core to edge software-centric solutions, cloud-native offers, leading-edge security and analytics tools, along with IP and optical networking solutions for 5G. We maintain a keen focus on our commitments to Environmental, Social and Governance (ESG) matters, offering an annual Sustainability Report to our stakeholders. To learn more about Ribbon, please visit rbbn.com.

Contact Us We are here to help. Let us know if you are interested in a quote or if you have any questions.