Gartner

Market Guide for Enterprise SBC

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E-SBCs terminate SIP trunks from provider networks and provide enterprise communications security, interoperability, remote worker support, call control and resiliency capabilities. This report gives I&O leaders information on E-SBC vendors and the capabilities of their product lines.

Key Findings

- A software-defined-network-centric approach to E-SBC, including SBC as a VNF, vCPE support and alignment with NFV orchestration frameworks, is a priority to E-SBC vendors.
- The market for E-SBCs is expanding beyond North America and EMEA to Asia/Pacific as enterprises deploy SIP trunk service globally.
- Enterprises are expanding E-SBC use beyond SIP trunks to include user-side security and interoperability of endpoints with cloud services.
- Increased adoption of WebRTC-based voice and video services has made WebRTC gateway functionality a key investment area among E-SBC vendors.
- Leading E-SBC suppliers provide automation and deployment simplification of E-SBCs with configuration templates for the most prominent PBXs, and SIP trunk service providers.

Recommendations

When planning unified communications, I&O leaders should:

- Use E-SBCs to provide secure connectivity and quality of service optimization to remote voice and video devices that do not use VPN access. E-SBCs are tailored for voice and video services, whereas standard firewalls are not.
- Optimize call routing in global SIP trunking implementations by using E-SBCs with advanced dial plan and session management. Optimized routing can reduce cost and provide a better UX by selecting paths where media will have less delay, a higher class of service or both.
- Deploy virtual E-SBCs if virtual infrastructure already exists in your environment and your implementation requirements include TDM/PRI interworking or high-performance transcoding or

encryption. Virtual E-SBC deployments are more cost-effective and require less time and effort to implement than E-SBC using purpose-built hardware.

Market Definition

Enterprise session border controllers (E-SBCs) are situated at the edge of the enterprise network and provide secure voice and video connectivity to Session Initiation Protocol (SIP) trunking providers, users in remote branch offices, home workers/remote workers, and unified communications as a service (UCaaS) providers.

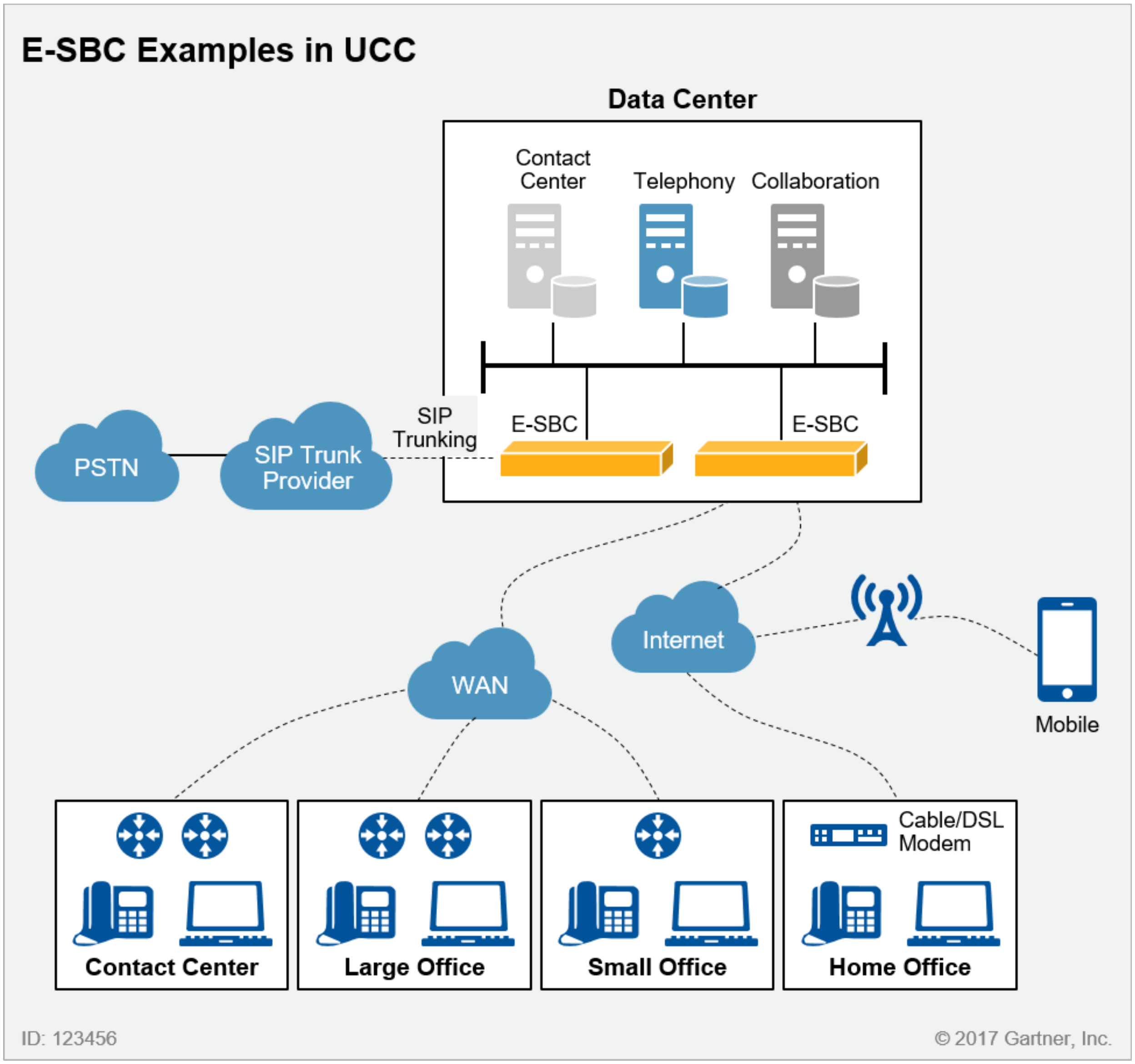
The name "session border controller" is derived from the following:

- **Session,** from Session Initiation Protocol, refers to a real-time communication connection between endpoints or users. This is typically a voice and/or video call.
- **Border** refers to the interface between networks that do not have full trust of each other.
- **Controller** refers to the ability of the E-SBC to control (allow, deny, transform, end) each session that traverses the border.
- The **"E"** in E-SBC stands for "enterprise." It is intended to differentiate enterprise-focused SBCs from those aimed at carriers, which have higher capacity, support much higher call rates and support interfaces into carrier operations' support systems.

Gartner recommends using E-SBCs to terminate SIP trunks originating in network service provider networks. For enterprises that do not have the skill or desire to manage their own E-SBC, carriers can provide managed E-SBCs as part of their SIP trunking service. An emerging delivery model with wireline carriers is E-SBC as a service — both network-based (no customer premises equipment required) and virtual customer premises equipment (vCPE)-based. E-SBCs can also be used to integrate different UC platforms, such as Microsoft Skype for Business Server and Cisco Unified Communications Manager (CUCM).

Figure 1 shows an example of an enterprise utilizing one pair of E-SBCs to terminate SIP trunks, and a second pair of E-SBCs on the line side to support encryption, remote access, and interoperability between multiple platforms and the endpoints.

Figure 1. E-SBC Placement Within UCC Reference Architecture



UCC: Unified communications and collaboration

Source: Gartner (December 2017)

The primary roles of an E-SBC are to provide:

- **Security:**
  - Network topology hiding — E-SBCs use network address translation (NAT) at the Open Systems Interconnection (OSI) Layer 3 Internet Protocol (IP) level and the OSI Layer 5 SIP level to keep internal network details hidden.

- Voice application firewall — E-SBCs protect against telephony denial of service (TDoS) attacks, distributed denial of service (DDoS) attacks, toll fraud and theft of service, access control, inspection, and monitoring.
- Encryption — E-SBCs encrypt the signaling and media if the traffic traverses enterprise networks and the Internet using Transport Layer Security (TLS)/Secure Real-Time Transport Protocol (SRTP).

- **Interoperability:**
  - Protocol interworking — SIP⇄H.323, SIP⇄Primary Rate Interface (PRI) and SIP⇄SIP. Although SIP is a standard, there are variations among UC vendor and service provider implementations that require normalization for proper interoperability. SBCs rely on SIP header manipulation, Internet Protocol version 4 (IPv4) and Internet Protocol version 6 (IPv6) interworking, and an internal back-to-back user agent (B2BUA) to realize interoperability.
  - Media interworking — Provides transcoding between various codecs and transrating between different bitrates (for example, transcoding G.729 in enterprise network to G.711 on SIP service provider network).
  - Media services — Replicate the media for call recording, clean up the media (such as echo cancellation, silence suppression and so on) and inject media on certain call flows, such as music on hold or audible tones.
- **Session control:**
  - Call admission control — E-SBCs provide policies that prevent exceeding the bandwidth allocated for voice and video services over links to specific sites. This helps avoid the service quality degradation that occurs with oversubscription.
- **Resiliency:**
  - SIP trunk load balancing — Connects to the same destination over more than one SIP trunk group to balance call loads evenly.
  - Alternate routing — Routes to the same destination over more than one SIP trunk group to overcome overload, service unavailability or quality of service degradation on the primary SIP trunk group.
  - Local public switched telephone network (PSTN) access — In case of a WAN outage, connect to a local time division multiplexing (TDM) service for local outbound emergency service calls, inbound DID service, analog traffic such as fax (in normal operation) and/or tail-end hop-off routing of calls for toll bypass.
  - High availability — Redundancy strategies to increase the availability of a single E-SBC node (for example, dual power supply or multiple Ethernet interfaces), and multinode designs (1+1, N+1, N+M) that provide failover from the failing node to other nodes.
- **Serviceability:**
  - Reporting — Real-time and historical reporting on usage, performance and analytics, via web portal and also via call detail reporting (CDR) records.

  - Alarming — Real-time notification of network performance/availability, security vulnerabilities, platform resources (CPU, memory, storage, network interface, licenses) or other events that are affecting the availability or quality of voice and video sessions.
  - Troubleshooting — Tools to isolate and fix a problem with the signaling or media on either side of demarcation points between networks.
- **Endpoint security and application interoperability:**
  - Encryption — Provides an encrypted and secure path for voice/video sessions for remote users not using VPN on fixed devices, mobile devices or PC.
  - Third-party client support — Enables third-party devices to work with a UC platform, including Microsoft Skype for Business or Cisco Unified Communications Manager (CUCM).
- **WebRTC gateway:**
  - Gateway — Connects WebRTC endpoints to non-WebRTC devices, such as calling from a WebRTC client to a phone connected through the PSTN; provides transcoding resources from Opus to G.711; cleans up the media, such as echo cancellation.
  - QoE — Provides quality-of-experience (QoE) monitoring and reporting on WebRTC connections, and network performance.
  - Reverse HTTP proxy — Provides voice and video reverse HTTP proxy functions for enterprise users who are utilizing a cloud-based WebRTC platform.

Market Direction

Enterprises are moving to SIP trunking to reduce their telecom expenses by up to 50% in the U.S. (for more details, see "Slash U.S. Telecom Expenses With SIP Trunks"), and by 15% to 30% globally. While 80% of enterprises in North America have some SIP trunking in their production environments, a minority have fully completed their migration to SIP trunking. In Europe, fully completed SIP trunking deployments are less than 5% of enterprises, and the numbers are even lower in other regions.

Enterprises undertaking a TDM-to-IP migration initiative will focus on replacing legacy TDM PBXs with centralized UCC platforms. An important part of these initiatives is the transition from TDM to SIP trunking for PSTN connectivity. However, some enterprises are implementing SIP trunking prior to replacing the legacy PBXs in order to capture the cost savings sooner. Network service providers in North America and Europe have been enticing and encouraging customers to migrate to SIP trunking so they can retire their legacy TDM voice networks.

Adoption of SIP trunking in North America and Europe is ahead of the rest of the world by a few years, because the cost-saving opportunities are more significant in North America and Europe than in other regions. More than 40 countries allow SIP trunking service. Global SIP trunking is available from traditional service providers such as AT&T, BT, CenturyLink (Level 3), Orange, Tata Communications, Telefónica, Singtel and Verizon.

In addition to SIP trunking applications, E-SBCs can interconnect UC/voice over IP (VoIP) platforms in multivendor premises-based deployments. E-SBC suppliers have added WebRTC gateway functionality to their E-SBC platforms. As desktop web-based collaboration with native video adoption increases, one of the growing roles of E-SBCs includes providing interworking functions and video transcoding and transrating similar to what SBCs provide with voice services. This is expected to supplement and eventually replace transcoding currently performed by video multipoint control units (MCUs), and will complement cloud-based transcoding.

An emerging use of an E-SBC is to provide dynamic session (call) management between UC platforms and endpoints, or between external trunks and UC platforms. Dynamic session management is the ability to prioritize, limit and optimize communication sessions in real time, based on events such as network congestion, voice/video quality degradation, overload and so on. As UC applications that utilize voice and video consume more network bandwidth (recommended bandwidth for HD desktop video calls is 500 Kbps; for a three-way video call, it is 2 Mbps down and 500 Kbps up), network resources should be allocated according to business requirements. Enterprises should supplement their network quality of service (QoS) and static call admission control policies with dynamic session management.

Following are current industry price ranges based on the number of concurrent SIP sessions (see Table 1). Pricing includes the hardware platform, and assumes that transcoding or encryption is occurring. Industry list prices vary for site size and number of simultaneous sessions:

Table 1. Current Industry Price Ranges Based on the Number of Concurrent SIP Sessions

E-SBC Size	Capacity in Concurrent Sessions	Price Range (USD)
Small	Less than 100	$1,500 to $7,500
Midsize	100 to 999	$1,700 to $70,000
Large	1,000 to 5,000	$30,000 to $235,000

Source: Gartner (December 2017)

Market Analysis

There are 12 critical attributes that enterprises should use when evaluating E-SBCs:

1. **Perimeter security —** Secure voice and video applications above what a firewall can provide, including hiding Layer 3 information embedded in SIP headers through Layer 5 NAT, for both IPv4 and IPv6, DDoS/TDoS detection/prevention/notification, encryption of Real-time Transport Protocol (RTP) and RTP Control Protocol (RTCP), and SIP signaling. U.S. Federal Information Processing Standard (FIPS) publication 140-2, which defines the U.S. government computing security standard, is used to accredit cryptography and is required of E-SBCs in federal government environments.
2. **Scale —** Large enterprises that need thousands of concurrent sessions, or that require high call rates, and also require transcoding, such as G.711 to G.729 and/or Secure Real-time Protocol

(SRTP) encryption, will need E-SBCs running on purpose built-hardware platforms. Although the SIP signaling and security interworking can be easily supported on virtual machines, the voice and video media transcoding and encryption is CPU-intensive and requires specialized digital signal processors (DSPs) for deployments requiring large-scale or high call rates (typically above 100 calls per second).

3. **High availability —** Stateful failover is a feature whereby, in a redundant (multinode) SBC implementation, when one SBC node fails due to loss of a power supply, software defect, network isolation or other technical failure, a redundant SBC node takes over within seconds without affecting active/stable calls. Georedundancy is a variant of high availability where the SBC nodes in an SBC cluster can be separated geographically — for instance, placing them in separate data centers.

4. **TDM interfaces —** ISDN PRI and analog interfaces are suitable for enterprises that want to adopt SIP trunking, but continue using legacy TDM PBX systems, or that deploy SIP trunking in a hybrid mode, where some connectivity to the PSTN is done over legacy TDM interfaces. The TDM interfaces reside on the E-SBC to minimize the hardware required at a site.

5. **Ease of configuration —** Preconfigured templates and a graphical user interface (GUI) enable an SBC to be configured to operate with a SIP service provider or with a PBX in minutes. Leading suppliers avoid the use of command line interfaces and parameter-by-parameter configuration of the E-SBC.

6. **Reporting —** Most enterprises require real-time and historical reports for usage, resource utilization, security events, performance and billing. E-SBCs typically come with a web-based element management application that includes standard/default reports, which can be customized. Other E-SBC products only record and store CDRs and rely on a third-party accounting and reporting applications to import the CDRs and generate the reports.

7. **Video transcoding —** Most E-SBCs support SIP signaling and security associated with video, but support for video transcoding and transrating is less common. Transcoding is the ability to allow interoperation of devices communicating through the E-SBC, but using different video codecs. Transrating or "speed matching" consists of supporting the same video codec at different characteristic bit rates. Some E-SBCs support temporal scaling of video between different frame rates. The most common video codec supported is H.264, with VP8 as an emerging requirement due to its prevalence in WebRTC implementations. H.265 and VP9 are the next generation of video codecs.

8. **Active Directory (AD) support —** AD support is required by enterprises using Microsoft Skype for Business Server as their UC platform. AD is used to store user profiles, phone numbers and permissions, such as the ability to call internationally or identify which devices to deliver incoming calls to.

9. **SIP Recording (SIPREC) —** SIPREC is an Internet Engineering Task Force (IETF) standard for call recording. Traditionally, call recording at the trunk-side level was accomplished by tapping or spanning the network port. SIPREC enables call recording to be forked/replicated at the IP layer, which offers a more robust way of balancing the load across multiple call-recording servers and failover options. Call recording can be used, for example, in environments that have

contact centers for quality assurance, or in financial services, healthcare or public-sector environments for e-discovery.

10. **Session management —** Session control for routing calls over different carriers or networks is based on cost, day/time, region, QoS, calling/called user, originating/terminating device IP address, codec type, domain name or other variables. For example, if the SBC detects errors on calls delivered to one network, the SBC can then deliver subsequent calls to alternate networks operating normally.
11. **Virtualization —** E-SBCs need to be deployed onto virtual infrastructure (especially including VMware vSphere, Linux Kernel-based Virtual Machine [KVM] and Microsoft Hyper-V), and support network function virtualization (NFV) on environments such as OpenStack. Virtualization support is required for E-SBCs to be deployed in cloud environments such as Amazon Web Services (AWS), Microsoft Azure and Google Cloud Platform.
12. **WebRTC gateway —** E-SBCs must integrate WebRTC endpoints, typically running on web browsers, with traditional UCC solutions. WebRTC gateways also provide Session Traversal Utilities for NAT (STUN) and Traversal Using Relay NAT (TURN) to allow WebRTC endpoints to traverse NAT boundaries. Support for the Opus audio codec, along with H.264 and VP8 video codecs, is standard with WebRTC gateway implementations.

These above capabilities are identified in the following tables for each vendor.

Representative Vendors

The vendors listed in this Market Guide do not imply an exhaustive list. This section is intended to provide more understanding of the market and its offerings.

AudioCodes

AudioCodes is headquartered in Airport City, Israel, with R&D facilities in Israel, China and New Jersey. AudioCodes' SBC product line is named Mediant. The Mediant 500L is the vendor's smallest SBC, and the Mediant 9000 is the largest model, with several models between these two. All SBCs share an identical codebase, and therefore provide consistency in features, configuration and management across the Mediant product line.

See Table 2 for details about the critical attributes of AudioCodes.

Table 2. AudioCodes

Attributes	Product Capability	Comments
Security	Yes	Complete set of security features. Dynamic Layer 3/4 wire-speed access control list, signaling and media integrity, malicious attack signature database and prevention, message policy enforcement.
Scale	Yes	Supports up to 60,000 concurrent SIP sessions at 600 calls per second. If encryption is required, 40,000 concurrent SRTP sessions are supported. Transcoding is supported internally or via an external media transcoding cluster, up to 25,000 transcoding sessions.
High Availability	Yes	Supported across all E-SBCs in the Mediant product line. Redundant power supplies, network connections and 1+1 active/standby system redundancy.
TDM Interfaces	Yes	Supports BRI and T1/E1/J1 ISDN PRI digital interfaces, and FXS/FXO analog interfaces on the following models: Mediant 500L, Mediant 500, Mediant 800, Mediant 1000 and Mediant 3000.
Ease of Configuration	Yes	The Mediant product line provides a configuration wizard that leverages an evolving database of SIP trunk providers and UC platforms that have been validation-tested.
Reporting	Yes	Centralized and consolidated reporting, including the ability to manage multiple SBCs reporting through a common reporting engine.
Video Transcoding	No	Video media streams are supported. Video transcoding and transrating are not supported.
AD Support	Yes	Including advanced migration topologies using drop-and-insert configurations.
SIPREC	Yes	SIPREC with encryption and transcoding is supported.
Session Management	Yes	Multitenancy dial-plan management is supported. Routing according to session attributes or external database query is supported. Centralized session management via ARM (AudioCodes Routing Manager) in environments with many E-SBCs.
Virtualization	Yes	Supported with AudioCodes Mediant Virtual Edition. Supported virtual platforms: VMware, KVM, Hyper-V, AWS, Azure and Alibaba Cloud. Supports VNF deployment of SBCs within OpenStack NFV frameworks.
WebRTC Gateway	Yes	Supports ICE lite, Opus audio codec, DTLS, SIP over web socket. Session forking for recording is supported. Video streams are supported without transcoding/transrating.
Foreign exchange station/foreign exchange office (FXS/FXO); Interactive Connectivity Establishment (ICE); virtual network function (VNF)		

Source: Gartner (December 2017)

The Mediant product line includes hardware and software-based SBCs suitable for deployment at small offices and midsize and large sites, which positions the company to be a sole-source vendor for centralized and distributed deployments within the same network. AudioCodes provides integrations of its E-SBC product line with Genesys contact center solutions and Microsoft's Skype for Business offer. AudioCodes has obtained certifications for both Microsoft Cloud Connector Edition (CCE) and Survivable Branch Appliance (SBA). Notable new features include VoIPerfect, which provides voice quality enhancement/remediation in lossy/adverse network conditions. AudioCodes has also updated its web-based management application with features to simplify E-SBC management.

Avaya

Avaya is headquartered in Santa Clara, California. Avaya's E-SBC offering is called Session Border Controller for Enterprise (SBCE). Avaya has continued to invest in SBCE, adding features to support cloud deployment environments and to leverage the Avaya customer base, especially those on legacy Nortel platforms that are migrating from TDM to VoIP/SIP. See Table 3 for details about the critical attributes of Avaya.

Table 3. Avaya

Attributes	Product Capability	Comments
Security	Yes	IPv6 support was recently introduced. Joint Interoperability Test Command (JITC)-certified.
Scale	Yes	Supports up to 20,000 concurrent SIP sessions at a call rate of 111 calls per second. If encryption is enabled, 10,000 concurrent SIP sessions are supported.
High Availability	Yes	Georedundancy is supported. No loss of feature functionality after failover.
TDM Interfaces	No	T1/E1 ISDN PRI are not supported.
Ease of Configuration	Partial	Web-based management tool is supported. Automated provisioning scheduled for future release.
Reporting	Partial	Quality-of-experience monitoring is limited. MOS/PESQ measurement and reporting are not supported.
Video Transcoding	No	Video transcoding and transrating are not supported.
AD Support	Yes	Supported.
SIPREC	Yes	Supported.
Session Management	Yes	Handled in the Aura Session Manager.
Virtualization	Yes	VMware vSphere, KVM, Nutanix and AWS support.
WebRTC Gateway	Partial	Port address translations and STUN/TURN services are supported. Conferencing, session forking, and transcoding are not supported.
Mean opinion score (MOS); perceptual evaluation of speech quality (PESQ)		

Source: Gartner (December 2017)

Avaya has delivered improvements in license management by introducing pooled licensing, where licenses can be dynamically shared by multiple Avaya SBCEs. Avaya SBCE also provides native voice transcoding support. Avaya SBCE is sold primarily in conjunction with Avaya deployments.

Cisco

Headquartered in San Jose, California, Cisco's primary SBC product is the Cisco Unified Border Element (CUBE). CUBE is well-positioned for SIP trunking, legacy PSTN interworking and line-side services for Cisco PBXs. CUBE is available as a virtual E-SBC embedded in Cisco's Integrated Services Router (ISR) and Aggregation Services Router (ASR) platforms, as well as a virtual machine in Cisco Unified Computing System (UCS) platforms. Businesses can choose the capacity of the router based on their SIP deployment architecture, scale, and need for TDM or analog interfaces. See Table 4 for details about the critical attributes of Cisco.

Table 4. Cisco

Attributes	Product Capability	Comments
Security	Yes	Full set of security services, and FIPS-140-2 and DISA-certified. TLS 1.2 and SRTP with SHA384 are supported.
Scale	Yes	On the UCS platform, CUBE supports 42,000 concurrent sessions at 260 calls per second. If encryption is required, CUBE supports 21,000 concurrent sessions.
High Availability	Yes	Stateful failover for both signaling and media is supported.
TDM Interfaces	Yes	Available when deployed simultaneously with Cisco TDM gateways on the 800, ASR and ISR router platforms. Not available for UCS platforms.
Ease of Configuration	Yes	Templates integrated with Cisco Prime management are supported. Multitenant template-based automated provisioning is also supported.
Reporting	Yes	Quality-of-experience monitoring is supported.
Video Transcoding	No	Video transcoding and transrating not currently supported.
AD Support	Partial	Limited ability to use AD groups to enable/restrict features such as toll restrictions.
SIPREC	Yes	SIP-REC draft revision 18 is supported.
Session Management	Yes	CUBE offers session control.
Virtualization	Yes	VMware is supported.
WebRTC Gateway	No	Opus and WebRTC are not currently supported.

Source: Gartner (December 2017)

Cisco CUBE can be easily deployed onto existing 800, ISR and ASR series router platforms for small to midsize E-SBCs. Cisco is also well-positioned for environments using Cisco Unified Communications Manager (CUCM) or Hosted Collaboration Solution (HCS). Cisco CUBE has been tested for SIPconnect 1.1 certification— a standard that defines a reference architecture and protocol procedures for E-SBCs connecting to service provider environments. Cisco CUBE is integrated with other router-based functions, including zone-based firewall security and software-defined WAN (SD-WAN).

Huawei

Huawei is based in Shenzhen, China. Huawei includes in its SBC lineup the SE1000-E600 and SE1000-E300 SBC products aimed at the enterprise market. Huawei continues to invest in expanding support of enterprise-focused features and functions on the SE1000 series. The Huawei SE1000 has compatibility with Huawei's Cloud EC solution, and has completed certification testing with Microsoft Skype for Business.

See Table 5 for details about the critical attributes of Huawei.

Table 5. Huawei

Attributes	Product Capability	Comments
Security	Yes	Huawei has tested the SE1000 at DDoS rates of up to 500 kpps.
Scale	Yes	SE1000-E600 can support 40,000 concurrent sessions at 300 calls per second.
High Availability	Yes	Georedundancy is supported by the SE1000 SBC.
TDM Interfaces	No	T1/E1 ISDN PRI not currently supported.
Ease of Configuration	Partial	Support for third-party SDN is on the roadmap for future release. WebGUI for HTTPS access is supported.
Reporting	Yes	MOS reporting is supported.
Video Transcoding	No	Video transcoding and transrating are not supported.
AD Support	Yes	
SIPREC	No	
Session Management	Yes	Partial support for external routing interface (for example, ENUM).
Virtualization	Yes	Linux KVM, VMware vSphere and Huawei FusionSphere Cloud OS are supported.
WebRTC Gateway	Partial	Media clean-up and QoS monitoring are supported. Opus and co-browsing are not yet supported.
E.164 number to URI mapping (ENUM); Kilo packets per second (kpps)		

Source: Gartner (December 2017)

Huawei's primary enterprise markets are Asia/Pacific and Latin America; Europe is an emerging market for Huawei. Huawei E-SBCs have been validation-tested and will support enterprise environments with Cisco CUCM, Mitel MiVoice, BroadSoft BroadWorks, Avaya Aura and Avaya IP Office.

Ingate Systems

Ingate Systems is headquartered in Stockholm, Sweden, and designs and develops E-SBCs and firewalls. Ingate's SBC product line is SIParator, which is sold primarily in the U.S. and Europe, with 90% of its sales through indirect channel partners.

See Table 6 for details about the critical attributes of Ingate Systems.

Table 6. Ingate Systems

Attributes	Product Capability	Comments
Security	Yes	IPv4 and IPv6 NAT are supported for both signaling and media.
Scale	Yes	20,000 concurrent SIP sessions are supported. If encryption or transcoding are required, 8,000 concurrent SIP sessions are supported.
High Availability	Yes	
TDM Interfaces	No	T1/E1 ISDN PRI and analog interfaces are not supported.
Ease of Configuration	Yes	Templates provided for 57 IP-PBXs and 29 SIP service providers.
Reporting	Yes	Quality of experience/MOS score reporting are supported.
Video Transcoding	No	Voice and video transcoding/transrating are not supported.
AD Support	No	
SIPREC	No	SIP-REC is not currently supported.
Session Management	Yes	
Virtualization	Yes	VMware vSphere and Microsoft Hyper-V are supported.
WebRTC Gateway	Yes	Supports Opus and other WebRTC gateway features.

Source: Gartner (December 2017)

Ingate has three main tiers of E-SBCs: the S21 is for small businesses or sites; the S52 is the midsize platform; and the S95, S97 and S98 are Ingate's large enterprise platforms. The S95 and S97 are for large enterprises, but are price-optimized for enterprises that don't need scalability up to 20,000 sessions. The SIParator negotiates media features between endpoints, but does not support transcoding between voice codecs. Ingate's SIParator E-SBC has been certified for SIPconnect 1.1.

Oracle

Oracle is headquartered in Redwood City, California. Oracle offers dual-tone multifrequency (DTMF) suppression capabilities, which are required in PCI DSS compliant contact center deployments.

Oracle also supports comfort noise generation (CNG) for environments with Microsoft Skype for Business Cloud Connector Edition (CCE).

See Table 7 for details about the critical attributes of Oracle.

Table 7. Oracle

Attributes	Product Capability	Comments
Security	Yes	FIPS-140-2 and JITC-features.
Scale	Yes	The Oracle 6300 supports up to 80,000 concurrent SIP sessions at 1,000 calls per second, or 60,000 sessions if transcoding and encryption are enabled.
High Availability	Yes	Redundant power supplies, network connections and 1+1 active/standby system redundancy.
TDM Interfaces	Yes	One- or four-port T1/E1 ISDN PRI support on the Oracle 1100 platform is offered.
Ease of Configuration	Yes	Some support of templates for PBXs. No templates that map to providers.
Reporting	Yes	Full support of usage, performance and quality of experience reporting.
Video Transcoding	No	Video transcoding and transrating are not currently supported.
AD Support	Yes	
SIPREC	Yes	Increased capacity in Oracle Virtual Machine Edition from 1,000 to 3,500 concurrent SIP-REC sessions.
Session Management	Yes	REST APIs are not supported. ENUM routing is supported.
Virtualization	Yes	VMware vSphere and Microsoft Hyper-V are supported. Increased capacity for Oracle Virtual Machine edition from 1,000 to 7,200 sessions (3,000 with transcoding enabled).
WebRTC Gateway	No	No longer implemented in E-SBC; however, the Oracle Communications WebRTC Session Controller supports WebRTC.

Source: Gartner (December 2017)

Oracle supports FIPS and JITC security-related capabilities, higher scalability on Virtual Machine Edition SBCs, and features to enable tighter integration to Microsoft Skype for Business. Additionally, Oracle has invested in features aimed at contact center environments, which have specialized requirements.

Patton

Patton is headquartered in Gaithersburg, Maryland. Patton's E-SBC product line is SmartNode, and it is focused on the small and midsize business (SMB) market. The Patton SN5300 is a small-size E-SBC and supports 60 concurrent sessions. The SN5500 is the midrange E-SBC (200 concurrent SIP sessions), and the SN10500 is the E-SBC with highest capacity (5,000 concurrent SIP sessions). All the SmartNode E-SBCs offer optional integrated access device functionality for fiber or DSL/ADSL WAN access interfaces. Patton's primary enterprise market is Europe, via channel partners.

See Table 8 for details about the critical attributes of Patton.

Table 8. Patton

Attributes	Product Capability	Comments
Security	Yes	Firewall, TLS/SRTP, and VPN with IPsec AES/DES encryption supported.
Scale	Yes	Patton SmartNode E-SBCs scale up to 60 concurrent sessions on the smaller SN5300, up to 200 on the SN5500 and up to 5,000 concurrent sessions at 100 calls per second on the SN10500.
High Availability	Yes	Redundancy is currently supported on the SmartNode 10K Series (e.g., SN10500). Branch-office survivability, including E911 services in survivability mode, is supported.
TDM Interfaces	Yes	T1/E1 ISDN PRI, analog interfaces and ISDN BRI are all supported.
Ease of Configuration	Yes	Automated HTTPs and TR-069 provisioning are supported, as well as customized application-oriented configuration wizards.
Reporting	Yes	SNMP- and RADIUS-based data acquisition. CDR and Syslog are supported. Report generation relies on third-party server and tools in addition to the Patton cloud.
Video Transcoding	No	Video transcoding and transrating are not currently supported, but are scheduled for future release.
AD Support	No	
SIPREC	No	SIP-REC is not currently supported.
Session Management	Yes	Supports call routing based on many variables, including prefix, suffix, source, destination, priority (E911) and failover.
Virtualization	Yes	Patton E-SBCs supports VMware vSphere, Linux KVM, and Oracle VM VirtualBox virtualization platforms.
WebRTC Gateway	Yes	WebRTC gateway is available in the SmartNode SN5000 series.
Advanced Encryption Standard (AES); Basic Rate Interface (BRI); Data Encryption Standard (DES); enhanced 911 (E911); Simple Network Management Protocol (SNMP)		

Source: Gartner (December 2017)

Patton specializes in cost-optimized E-SBC solutions for SMBs. SmartNode also integrates other services into the E-SBC platform, such as routers and Long Term Evolution (LTE) support for diversity and backup connectivity. Patton SmartNode also supports branch office survivability. Patton has certification-tested their E-SBCs with Microsoft, Avaya, BroadSoft, Metaswitch and Unify.

Sonus

On 30 October 2017, Sonus completed its merger with Genband, another OEM of E-SBCs. The combined company will continue to operate as Sonus until the end of 2017, at which time it will change to a new corporate name: Ribbon Communications.

Sonus is headquartered in Westford, Massachusetts. The Sonus SBC 1000 and SBC 2000 are aimed at SMBs. Sonus also offers the SBC SWe Lite (Software Edition), a virtual E-SBC that supports feature parity with the SBC 1000 and SBC 2000. For midsize to large enterprises, Sonus offers the SBC 5000 series and SBC 7000.

See Table 9 for details about the critical attributes of Sonus.

Table 9. Sonus

Attribute	Product Capability	Comments
Security	Yes	FIPS-140-2 and JITC-certified. No loss of performance or capacity during high-packet-rate network DDoS attack.
Scale	Yes	The SBC Sonus portfolio scales from 25 to 150,000 sessions. The Sonus SBC 5400 supports up to 75,000 concurrent sessions at 700 calls per second. The Sonus SBC 7000 supports 150,000 concurrent sessions at 1,350 calls per second.
High Availability	Yes	Redundant power supplies, network connections and 1+1 active/standby system redundancy (clustering).
TDM Interfaces	Yes	T1/E1 ISDN PRI and analog interfaces are supported on the SBC 1000 and SBC 2000.
Ease of Configuration	Yes	Rapid configuration of E-SBC via templates for SIP service providers and UC platforms.
Reporting	Yes	Full support of usage, performance and quality of experience reporting.
Video Transcoding	No	Video transcoding and transrating are on the Sonus roadmap.
AD Support	Yes	Active Directory services are supported. The Sonus PSX centralized routing and policy service interfaces with SBC 5000 series and SBC 7000.
SIPREC	Yes	Supports recording to multiple recorders simultaneously.
Session Management	Yes	Full support of session/call management and routing.
Virtualization	Yes	VMware vSphere, Linux KVM and Microsoft Hyper-V virtual infrastructures supported. Support for AWS and OpenStack cloud deployment tools.
WebRTC Gateway	Yes	WebRTC SDK supports all mobile device operating systems and browsers.
Software development kit (SDK)		

Source: Gartner (December 2017)

Sonus supports Microsoft Skype for Business Cloud Connector Edition (CCE). CCE functionality can be deployed on same hardware platforms hosting the E-SBCs. Sonus also supports multitenancy and multiple methods for user authentication (OAuth 2.0, Lightweight Directory Access Protocol [LDAP], operator/subscriber DB and SIP registrants). Sonus has enhanced the SBC 5400 to allow 2 Gbps to 10 Gbps Ethernet connectivity without additional hardware. Gartner anticipates that as the Sonus and Genband operations are integrated, the new combined company will rationalize overlaps in the current Sonus and Genband E-SBC portfolios.

Market Recommendations

Enterprises should deploy E-SBCs, whether their UC platforms are on-premises, in the cloud or a hybrid of the two. E-SBCs provide security, interoperability, resiliency and serviceability between different networks and/or UC telephony platforms. Enterprises that do not utilize an E-SBC will experience higher mean time to repair (MTTR) for user-reported problems, such as dropped calls, one-way audio or poor call quality. The higher MTTR is due to the increased time required to isolate the reported problem.

In addition to evaluating the technical capabilities and costs of E-SBCs, enterprises should ensure that they develop a partnership with their E-SBC vendors for planning investments in enterprise UC telephony platforms. A tight customer/vendor partnership will ensure that the E-SBC will be able to manage and support new features and functionality emerging in UC solutions. All the E-SBC platforms in this Market Guide support the prominent UC technology providers and SIP service providers. Enterprises should use the same technology provider for E-SBCs as their on-premises UCC platform if performance, feature and pricing requirements are met, because there is a higher level of integration, and element management is more efficient. Enterprises should not consider E-SBC vendors that have not validation-tested their UC platform or SIP service provider.

E-SBCs can be purchased and managed by the enterprise, or they can be delivered to enterprises as a managed service by SIP service providers or managed service providers. When enterprises choose to manage their E-SBCs, the UC/telephony team typically has management responsibility, as opposed to network or security teams, since the primary role of the E-SBC is as a voice and video application firewall.

Enterprises should ensure their E-SBC hardware platforms have the capacity for growth for three to four years. Estimate the number of sessions that will be needed to support additional video sessions, along with overall capacity growth, as the enterprise and/or usage of real-time collaboration tools grow. If encryption, call recording with SIP-REC or transcoding are possible future requirements, be aware that capacity can be significantly reduced when these features are enabled. Select platforms that have enough spare capacity when these features are enabled.

Gartner Recommended Reading

Some documents may not be available as part of your current Gartner subscription.

"Slash U.S. Telecom Expenses With SIP Trunks"

"How to Use SIP Trunks to Attain High-Availability Voice"

"How to Choose the Best SIP Trunk Architecture"

"Leverage These Six Best Practices When Choosing U.S. SIP Trunk Providers"

"Toolkit: RFP Template for SIP Trunking Services"

"Hype Cycle for Unified Communications and Collaboration, 2017"

"Hype Cycle for Enterprise Networking and Communications, 2017"

"Microsoft's New Vision for Communications and Collaboration in Office 365 Impacts Microsoft Teams and Skype for Business Customers"

Evidence

This research draws from more than 100 inquiries with Gartner clients, vendor responses to detailed questionnaires specific to this Market Guide and vendor briefings specific to this Market Guide.

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