

Unlimited!

OpenScape Voice V3.1 R3 Enterprise-class Voice Application

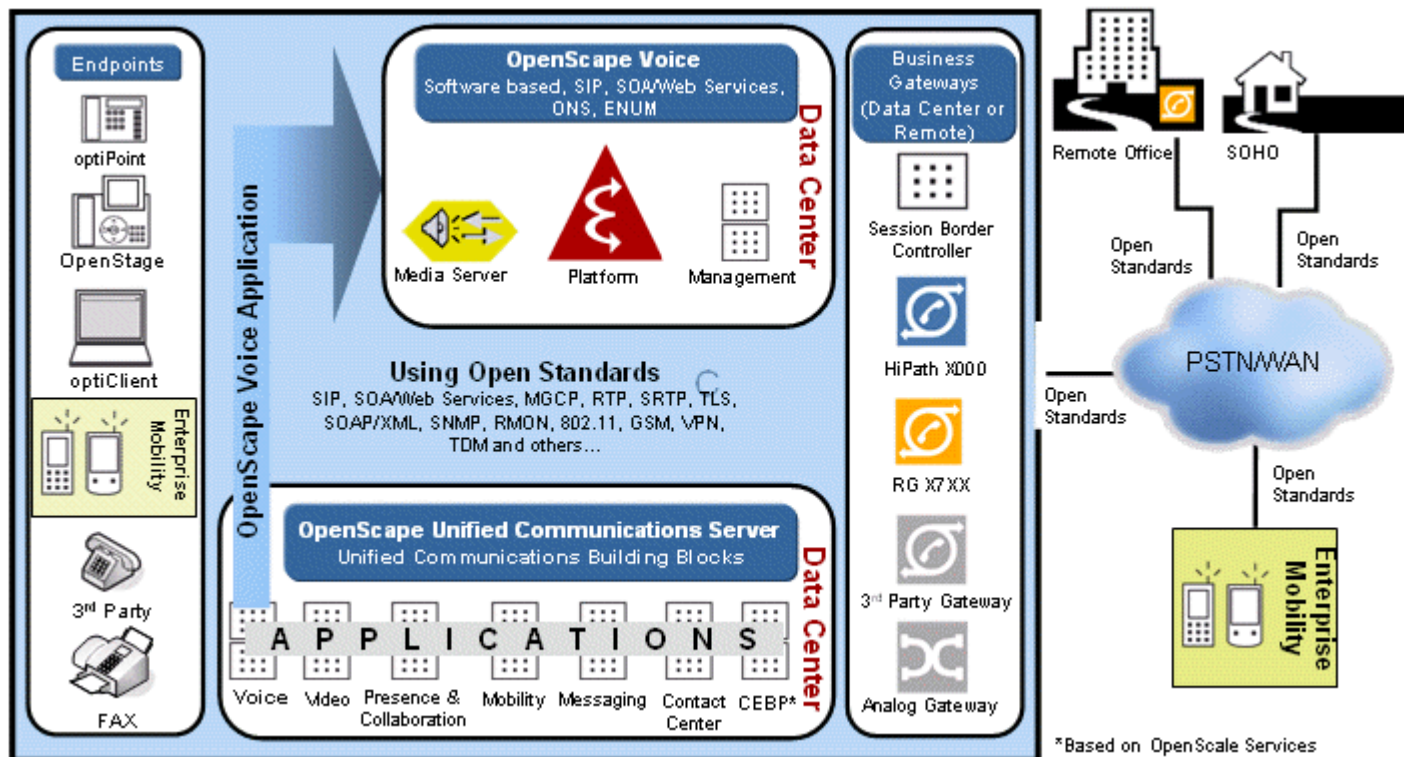
The essence of OpenScape Voice is its architecture, which serves as the foundation for a new multimedia business communications paradigm: effective, economical, efficient and environmentally friendly. It allows customers to build strong Communications-Enabled Business Processes (CEBP) using a unique blend of rich features, openness, scalability, resiliency, extensibility and manageability.

Communication for the open minded

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The OpenScope Voice Solution Landscape



The OpenScope Voice Solution Landscape

OpenScope Voice is an enterprise-class voice application that is fully integrated with, and offered as part of, a complete unified communications solution, the OpenScope Unified Communications (UC) Server. Supporting open standards, it is designed not only for centralized deployment within a distributed enterprise, but as a highly viable option for site-based deployments as well.

OpenScope Voice is designed to provide the architectural strength to such a framework through its rich feature set, scalability, resiliency, adherence to open standards, and manageability. As an enabler of information communication technology (ICT) convergence, OpenScope Voice creates technology choices, allowing customers to implement well thought out communication strategies at their own pace.

OpenScope Voice forms the foundation for a new multimedia business communications paradigm: pervasive, intuitive, interactive and effective. It allows customers to build strong pillars of multimedia communications functionality that provide business process integration (BPI) and business process enhancements through communications-enabled business processes (CEBP).

It boasts a very credible claim to being a core component of enterprise communications based on open standards, becoming a business tool to optimize and enhance enterprise communications and to enrich its processes – a tangible change from the traditional TDM, converged and IP PBXs.

OpenScope Voice serves enterprises of mid-to very large size and multi-tenant hosted services offered by Service Providers (SP). It serves as a core component of communications and is able to offer choices not only *in* unified communications, but *to* unify communications.

Best for Your Business

OpenScope Voice offers a key solution at the infrastructure level, interworking with a number of components to provide Voice over IP communications to the enterprise. The OpenScope Voice solution is in turn part of a broader solution set in the customer's environment, and as such, works with a variety of applications to enhance and support the customer's business practices.

The entire OpenScope UC Server portfolio is optimized for the demands of businesses – easy to put into practice, reliable in performance, and easy to use. With it, you become even more efficient.

Value

With OpenScope Voice, businesses benefit from the investments already made in their customers, partners, employees, and communications infrastructure. OpenScope Voice V3.1 R3 further demonstrates how SIP platforms can reduce communication costs. The separation of voice and data no longer exists. Only one infrastructure needs to be maintained. Processes and applications are more reliable and can be shared across the enterprise.

Evolution

As a product, OpenScope Voice V3.1 R3 has experienced a natural technological evolution far in advance of its competitors. It has been built from the ground up to ensure that leading-edge software technologies could be employed, rather than a patchwork of engineering. Therefore, OpenScope Voice is able to provide seamless, standards-based integration with multi-vendor systems and applications.

Choices

You decide when, where, how, and to what extent to invest in innovative technology. You can choose from a broad range of IP convergence platforms, optiPoint and OpenStage phones, and an optiClient solution (softclient). You set the pace in accordance with your demands and ideas.

Benefits of Installing OpenScape Voice V3.1 R3

- High availability and cost effective solution for enterprises in the medium to very large range
- Carrier-grade reliability and resiliency
- Scalability to tens of thousands of users
- Open unified communications platform
 - Any media, any time, anywhere
 - Support of open standards
- Excellent CAPEX and OPEX efficiencies of scale
- Seamless migration path from converged IP to SIP
- Web Services architecture
 - Access for end-user self-management
 - Integration with other Web-based applications and management systems
- Global licensing
- Batch command file and mass provisioning interface
- Communications as a Service (CaaS)

High-Performance IP Communication

OpenScape Voice V3.1 R3 offers a wide range of options for transforming your corporate communications solution into real-time IP communication. You can reduce your IP infrastructure costs even further by using high-performance gateways and standardized compression procedures. OpenScape Voice V3.1 R3's "Any-to-Any" IP payload switching ensures that you get the highest availability and quality.

Resiliency, Redundancy and Reliability

OpenScape Voice software runs on highly reliable, fault-tolerant industry-standard servers under the Linux SLES 9 operating system. Clustering software protects against hardware and software failures, and controls failover of redundant Ethernet links and cluster nodes (redundancy is optional for systems below 5000 lines). By ensuring that all functions and applications maintain unrestricted availability, OpenScape Voice provides a new level of quality in IP communications.

OpenScape Voice controls and supervises call setup; the actual voice traffic is carried over the LAN/WAN between endpoints. Administration/signaling and billing traffic is carried over a redundant pair of network interface cards through redundant, interconnected L2/L3 switches that provide redundant networking.

The two servers may be collocated or geographically separated. If geographically separated, the connections between the two nodes may be established at the layer 2 level, using a tunneling protocol such as L2TP, or at the layer 3 level using IP routing protocols.

OpenScape Voice utilizes Fujitsu Siemens Computers (FSC) PRIMECLUSTER clustering software and Resilient Telco Platform (RTP) middleware to provide a highly reliable platform which can operate in both active-active and active-standby mode, and can switchover automatically without loss of active calls or billing records.

OpenScape Voice also provides a Survival Authority (SA), a separate component which normally resides on the OpenScape Voice Assistant administration server. The SA can assist in determining the proper cluster response in the event that communication between the two nodes is severed due to a network failure. Activation of the Survival Authority is optional in the case of collocated cluster nodes, but required in the case of geographic separation of the nodes.

Environmentally Friendly Architecture

Two servers versus many – OpenScape Voice uses only two servers for fully redundant call control. The environmental costs, measured in terms of the power consumption and CO₂ output associated with the manufacture, acquisition, operation, maintenance and disposal of two servers, are significantly lower than for a site-based communications system or a system that requires more servers.

The unique scalability up to 100,000 users with only two servers is achieved through software-based growth, not by adding more hardware. This leads to:

- Less overhead power usage in the Data Center
- Lower heating, ventilating and air conditioning (HVAC) requirements in the Data Center
- Less rack space use in the Data Center

New Features in OpenScape Voice V3.1 R3

The many new or enhanced features being introduced with OpenScape Voice V3.1 R3 can effectively be grouped into the following functional categories:

- Enhancements to the OpenScape Voice softswitch
- Support for Integration or Interworking with other products in the OpenScape UC Solution Landscape

Enhancements to the OpenScape Voice Softswitch

- Call diversion for invalid destinations
- Connected outgoing line presentation (COLP) enhancements
- Continuous trace tool
- Deployment Service (DLS) V2.0 R4
- Emergency call handling enhancements
- HiPath MetaManagement enhancements
- Media encryption enhancements
- Remote patching with HiSPA (HiPath Serviceability Platform for Applications)

Integration or Interworking with Other Products in the OpenScape UC Solution Landscape

- Application-provided billing party via SIP
- Application-provided call correlation via SIP
- Call admission control for video telephony
- Display enhancements for CCBS/NR
- Geographic node separation: low-bandwidth layer-3 cluster interconnect links
- Interworking with HiPath 3000
- Mediatrix 4102 analog adapter support
- Mediatrix gateway enhancements: media encryption
- OpenScape Contact Center V7.0 R3 integration
- OpenScape Media Server enhancements
 - CALEA/LI support
 - Support for additional languages/countries
- OpenScape UC Application V3.1 R3 integration
- OpenStage V1.0 R5 support
- OpenStage V2.0 support
- Radisys Convedia CMS-3000: security, QoS and language enhancements
- RG 8700 enhancements: media encryption
- SIP trunking customization options
- Session border controller enhancements
 - Covergence Border Manager
 - Covergence Trunk Manager
- SIP signaling manager: internal audit mechanism

OpenScape Voice V3.1 R3 Management

System management tools for OpenScape Voice V3.1 R3 include the following:

- OpenScape Voice Assistant
- RTP Command Line Interface (CLI)
- Deployment Service (DLS)

OpenScape Voice Assistant

For all user configurations, OpenScape Voice Assistant is the strategic Web-based tool for administering OpenScape Voice V3.1 R3. For installations with fewer than

5000 users, the Assistant can be installed on the same server as the OpenScape Voice software and the integrated OpenScape Media Server. In installations with more than 5000 users, it is necessary to install OpenScape Voice Assistant as well as the OpenScape Media Server on a separate, external server.

RTP Command Line Interface

OpenScape Voice system provisioning and administration can be performed using a traditional command line interface (CLI). Features/functions which must be activated or provisioned only once are still managed using the RTP CLI, e.g., tracing and other maintenance functions. The RTP CLI is always accessible via a secure shell, and can be accessed via Assistant or directly from the maintenance port of OpenScape Voice.

Deployment Service

The Deployment Service (DLS) is a management tool used for administering work-points in the OpenScape Voice network. DLS is a Java-based application with a Web-based user interface, and is functionally integrated into OpenScape Voice Assistant.

The DLS is required to support the mobility feature on Siemens SIP endpoints. It provides options to migrate existing work-points and to implement mobile user standards. Other important functions of the DLS include software deployment, inventory data management, configuration management, and Plug and Play support.

OpenScape Voice V3.1 R3 Media Servers

OpenScape Voice V3.1 R3 offers the following media server options:

- **RadiSys Convedia CMS-3000 media server** for up to 360 ports
- **OpenScape Media Server** for up to 100 ports (on the OpenScape Voice server) or 150 ports (on an external server)

Multiple media servers can be employed for large installations or for added reliability.

RadiSys Convedia CMS-3000

The RadiSys Convedia CMS-3000 is a turn-key, high performance media server for enterprise-sized OpenScape Voice network deployments.

OpenScape Media Server

The OpenScape Media Server is an integral part of OpenScape Voice systems for medium-size enterprises supporting 300-5000 subscribers. This software-only server solution provides tones, announcements and user prompts to support the functionality of

OpenScape Voice features. Announcements are generated in the language requested by OpenScape Voice, or in a configurable default language. The OpenScape Media Server also supports redundancy, station controlled conferencing, and media encryption using Secure RTP (SRTP) and the MIKEY key management protocol.

The OpenScape Media Server can be installed on the same server as the OpenScape Voice application for systems with fewer than 5000 subscribers, requiring no additional hardware; on an external server (the same server as OpenScape Voice Assistant); on a separate, standalone server; on any combination of these options.

OpenScape Voice V3.1 R3 Gateways

To access the public switched telephone network (PSTN), the OpenScape Voice V3.1 R3 solution provides the following gateway options for media and signalling:

- HiPath 4000 Survivable Media Gateway
- HiPath 3000 Media Gateway
- RG 8700 Survivable Media Gateway
- RG 2700 Survivable Media Gateway
- Mediatrix gateways
- Coverage Session Border Controller

Survivable Media Gateways (SMG)



In branch offices with a **HiPath 4000**, survivability is made possible through the use of a Comdasys Convergence SIP proxy. This proxy takes the registrations from the phones and the HiPath 4000 gateway and passes them to OpenScape Voice via the WAN. If OpenScape Voice drops out or does not respond in a timely manner, the local SIP proxy takes over and tries to mediate the calls, including routing PSTN calls through the HiPath 4000 gateway. When connectivity to OpenScape Voice is re-established, the proxy resumes forwarding the requests to OpenScape Voice as usual. Branch offices with one or more **HiPath 3000** systems (survivability option not available with OpenScape Voice V3.1 R3) can serve as media gateways when connected to the OpenScape Voice IP network infrastructure.



The **RG 8700** provides a complete Siemens solution for OpenScape Voice, as well as basic survivability for branch offices in the event of network failure. Survivability, a standard feature of the RG 8700 gateway, is accomplished through the use of SIP phones that are dual-registered with OpenScape Voice and the RG 8700. If the RG 8700 can no longer communicate with OpenScape Voice, it switch-

es to survivable mode, allowing the dual-registered SIP phones access to the PSTN trunks and, conversely, allowing incoming calls from the PSTN to be distributed directly to the SIP phones.

The RG 8700 family of Survivable Media Gateways comprises 3 models that interwork with OpenScape Voice V3.1 R3: RG 8716 with up to 16 T1/E1 spans, RG 8708 with up to 8 T1/E1 spans, and RG 8702 with up to 2 T1/E1 spans. No license is required.

The RG 8700 V1.2R1 software adds SIP-Q functionality for connectivity to HiPath 4000 and third party products which support QSIG.



The **RG 2700** gateway, designed for organizations with a head office and small- to medium-size branch offices, is used for cross-site networking. This SMG includes a built-in SIP proxy that provides continued inbound and outbound calling service for up to 30 subscribers when the connection to the central OpenScape Voice system is temporarily lost.

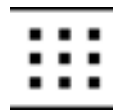
Small Branch Office (SBO) Gateways



These gateways connect endpoints in a small office scenario. Gateways from **Mediatrix**, with optional survivability provided by **Comdasys**, connect these locations to OpenScape Voice.

Customers can also continue to use their previously installed third-party SIP gateways with OpenScape Voice. The supported functionality depends on how these gateways adhere to the relevant SIP standards. Interoperability testing may be required to confirm feature behavior. The HiPath Ready Lab is available to vendors seeking to certify their products with OpenScape Voice.

Session Border Controllers



A session border controller (SBC) enables VoIP networks to extend SIP-based applications beyond an enterprise's network boundaries, such as for example, when the SIP clients of an OpenScape Voice system reside in different IP networks. The **Covergence Border Manager (CBM)** solution performs the necessary interoperability, security, management, and control functions needed to support SIP trunking applications, as well as SIP endpoint registration services for remote user and branch office applications.

SIP Endpoints

The following Siemens SIP endpoints are supported:

- OpenStage 20/40/60/80
- optiPoint 410 S and 420 S
- optiPoint 150 S
- optiPoint WL2 professional S (wireless)
- optiClient 130 S
- OpenScope Video VHD100/400/600

Selected third-party phones may also be certified through the Siemens Ready Lab.

OpenStage 20/40/60/80

OpenStage™ is the name for Siemens' new generation of IP phones, setting the benchmark for open, unified communications in a productivity-enhancing business tool. OpenStage phones have an intuitive and innovative interface that is available in a wide variety of languages; all models are fully compliant with IEEE 802.3af Power Over Ethernet (PoE) standard.

The OpenStage family of SIP telephones comprises four models:

OpenStage 20, the economy model, is a full-featured speakerphone and a universal solution for efficient and professional telephony. The new **OpenStage 20 E** variant offers open listening only.

OpenStage 20



OpenStage 40, the flexible office phone, is customizable for various workplace environments – desk sharers, work teams, call center staff, and so on.

OpenStage 40



OpenStage 60 incorporates an open application platform and personalization options, and is especially well-suited for executive-assistant environments and users who interact with mobile devices.

OpenStage 60



OpenStage 80, the high end model, incorporates premium features, materials and components, and a productivity-enhancing open platform for applications. It is designed with the needs of the top-level manager and executive in mind.

OpenStage 80



Eco-Friendly Endpoints

OpenStage phones have been designed with the environment in mind. Environmental protection standards have been fully adhered to in regard to materials and the manufacturing process, power usage during operation, and disposal when the time comes. This new family of devices is designed to reduce power consumption by as much as 35%.

optiPoint Phone Family optiPoint 410 S and 420 S

The feature that distinguishes the optiPoint 410 S / 420 S family of SIP phones, in particular, is the wide range of customizable models, from the optiPoint 410 entry S for basic telephony to the optiPoint 420 advance S for high-volume callers with sophisticated needs. A total of five different telephone models are available to suit all workstation requirements. A choice of expansion options and accessories provide the ability to accommodate future needs.

optiPoint 150 S

The optiPoint 150 S phone is a cost-optimized entry level device for voice over IP telephony, suited for environments where basic business features are required, such as reception areas, corridors, and production facilities.

optiPoint WL2 professional S

The SIP-compliant optiPoint WL2 professional S is a single-line WLAN handset that supports converged mobile voice and data applications on a single wireless infrastructure. It is interoperable with all standards compliant WLAN infrastructure products for seamless wireless connectivity and mobility.

optiClient 130 S Softclient

The optiClient 130 S is a PC-based multimedia client that mirrors the functionality of optiPoint telephones. It supports a variety of media, such as data, video, e-mail and Internet, and offers convenient user features, including data access during calls, application sharing and three-way videoconferencing. Media encryption is achieved via Secure RTP (SRTP) and the MIKEY key management protocol.

OpenScope Video VHD100, VHD400, VHD600

The OpenScope Video SIP endpoints provide the capability to locally host high-definition videoconferences of varying sizes, depending on the model:

OpenScope Video VHD100

The VHD100 provides entry-level video communication capabilities. It supports two-way video calling and is designed to be used as an executive desktop video system.

OpenScope Video VHD400

The VHD400 provides audio and video capabilities suited for small groups and organizations. It includes an embedded multipoint bridge that supports up to four conference participants. In addition, the VHD400 supports **continuous presence**, allowing a conference participant to view all other participants simultaneously.

OpenScope Video VHD600

The VHD600 provides audio, video and data capabilities suited for large conference rooms. It includes an embedded multipoint bridge that supports up to six conference participants. Like the VHD400 model, the VHD600 supports continuous presence.

Need More Capacity?

For larger videoconferences than the OpenScope Video family of SIP endpoints supports, the **Codian MCU 4500** series of voice and video bridges permits additional parties to participate, either in full videoconference mode or in the voice portion only.

Analog Adapters

Analog adapters from Mediatrix allow users with existing analog phones, analog fax machines and modems to connect to the OpenScope Voice SIP environment, thereby preserving their investments.

Other OpenScape UC Server Applications

OpenScape UC Application

The OpenScape Unified Communications (UC) Application is a high-functionality collaboration application that fits into an enterprises's existing voice and data infrastructure, tying together phones, voice mail, e-mail, text messaging, directories, calendaring, instant messaging and conferencing services.

The tight integration between the OpenScape UC Application and OpenScape Voice allows users to take advantage of market-leading collaboration and mobility features, and provides the ability to leverage advanced user and group presence features.

HiPath Xpressions

HiPath Xpressions combines voice, fax, e-mail and text (Short Message Service, SMS) services on a Windows 2003 platform and transforms them into a Unified Messaging system for use together with OpenScape Voice.

Built using a modular, scalable client/server architecture, HiPath Xpressions can be configured to meet users' individual communication needs. New functionality in HiPath Xpressions V5.0 supports voice and message waiting indication (MWI) over SIP.

HiPath ComAssistant

HiPath ComAssistant is a Web-based call control and communication filtering application that enables users to manage incoming voice and e-mail communications from their desktop.

HiPath ComAssistant offers computer telephony integration (CTI) features such as "click & dial", call logging, LDAP address book search, and One Number Service (ONS). With a choice of two easy-to-use graphical user interfaces (GUI), HiPath ComAssistant provides home and business users with rules-based communication filters and routing capabilities to optimize accessibility and increase efficiency.

HiPath MetaManagement

The **HiPath MetaManagement Suite** provides a comprehensive and all-embracing management solution for the standardized administration of all HiPath platforms and applications.

HiPath Accounting Management (HiPath AM) is the accounting application for processing and analyzing call data for incoming and outgoing voice and VoIP calls over different network operators (carriers) as well as internal connections in HiPath standalone systems and networks.

HiPath Fault Management (HiPath FM) supports and simplifies network management by graphically displaying the complete communications network, showing the status of each element. Special plug-ins optimize the detection, diagnosis and removal of failures. HiPath FM also monitors hardware and software from other manufacturers, interfacing via SNMP (using the manufacturer-specific enterprise MIB).

HiPath User Management provides a simplified "umbrella solution" for the creation, deletion and modification of user data and communication resources across all HiPath platforms and applications in a HiPath network. All relevant user data are stored in a Directory Service and are available for all HiPath applications with an LDAP interface.

HiPath Quality of Service (QoS) Management provides comprehensive, easy to use functions for configuring, monitoring and analyzing all HiPath VoIP components in a HiPath network with respect to the relevant QoS parameters.

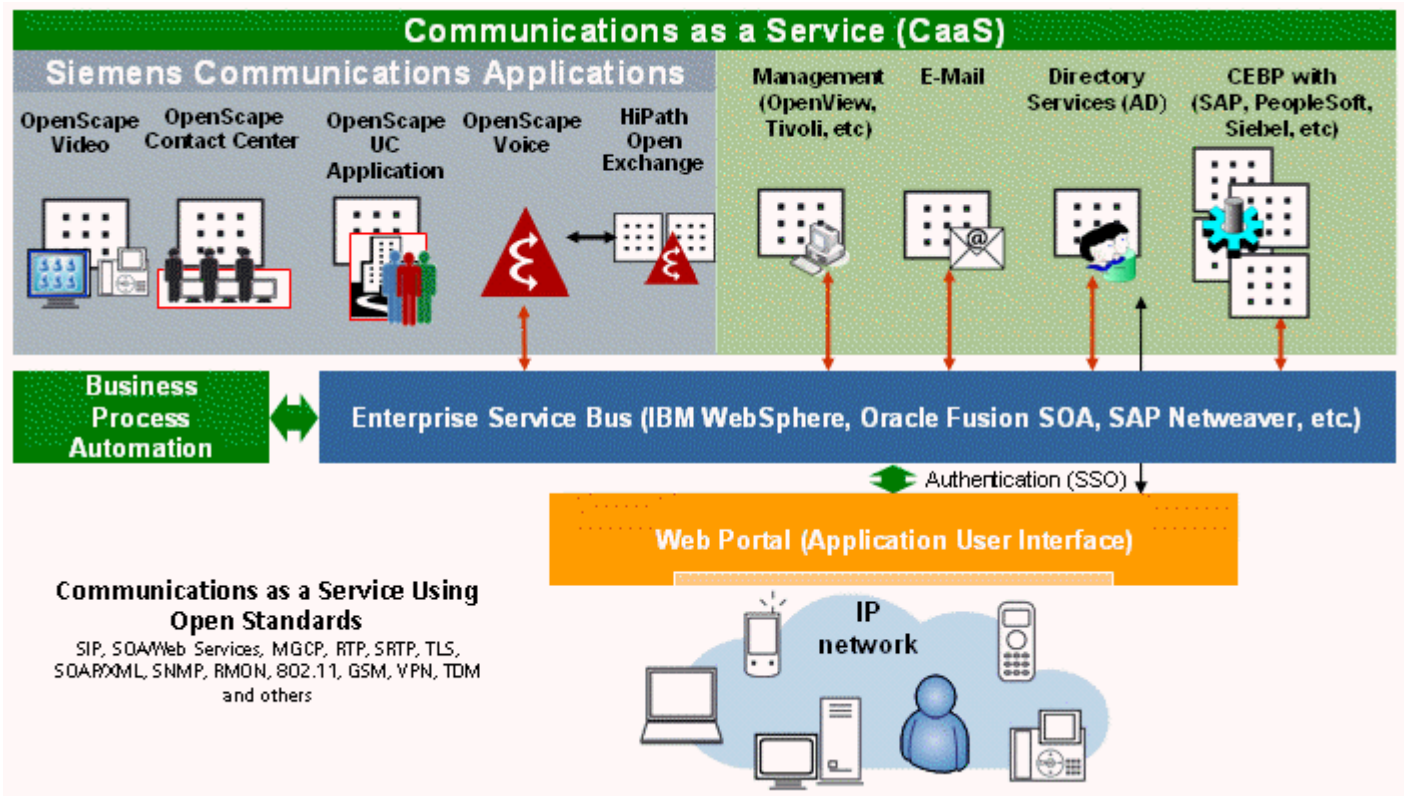
OpenScape Contact Center

OpenScape Contact Center is the Siemens **contact center** application for the OpenScape Voice and HiPath switching platforms. It provides an intuitive agent interface with powerful visual management tools.

CallTicket Attendant Application

CallTicket (Open Communications Solutions for CRM attendant application) is a feature-rich and flexible PC-based attendant application. When integrated as a core component in your telephony solution, it monitors the status of every station, receives incoming calls, and allows designated attendants to transfer calls to the right employees quickly and efficiently. As a result, it improves employee accessibility, call handling and customer service. It also helps your business provide intelligent and personalized interactions, as well as faster and more effective communication.

OpenScape Voice: Communications as a Service (CaaS)



Communications as a Service

Siemens' **Communications as a Service (CaaS)** is much richer than mere hosted telephony. CaaS offers a modular approach to building applications, allowing enterprises to select the feature sets they need today, with the flexibility to change them or add to them in the future.

The flexibility inherent in CaaS allows customers to not only grow, but to do so at their own pace. CaaS provides growth choices ranging from basic telephony to business process-embedded, presence-based rich communications environments; from contact centers with remote agents optimized through group- or skill-based call routing to multimedia-based and presence-enhanced contact center solutions.

Whether your goal is interoperability with an existing communications infrastructure in order to optimize existing investments, or inexpensive migration to a survivable remote office, these choices and many other data center deployment options are made possible through Siemens' commitment to open IT-based communications.

OpenScape Voice Server Technical Data

The OpenScape Voice V3.1 R3 software runs on highly reliable, fault-tolerant industry-standard servers, providing carrier-grade reliability. A typical hardware configuration consists of a two-node cluster of PRIMERGY RX330 S1 servers from Fujitsu Siemens Computers (FSC), running in a fully redundant load-sharing operation. For installations with up to 5000 users, redundancy is optional, so the second server is not required. The operating system is SUSE Linux Enterprise Server 9 Service Pack 3 (SLES 9 SP3). A SolidTech database runs on each server.

Each RX 330 S1 server has two (2) Dual-Core or Quad-Core (Q1 2009) AMD Opteron™ processors, up to 32 GB of DDR2-667 direct addressable memory, two (2) L2/L3 Ethernet switches and eight (8) 10/100/1000 base-T Ethernet links, set up as pairs connected to the Ethernet switches (two external L2/L3 Ethernet switches are required for a redundant configuration.)

Note: OpenScape Voice V3.1 R3 continues to support IBM's **System x3650 T** servers; However, customers using the x3650 T platform must expand the DDR2 memory to 8 GB.

Supported Standards

The OpenScape Voice platform and its standard solution components (phones and application servers) support the relevant aspects of the following standards specific to Voice over IP (VoIP):

IETF Standards

- RFC 1213: Management Information Base for Network Management of TCP/IP-based internets: MIB-II
- RFC 1442: Structure of Management Information for Version 2 of the Simple Network Management Protocol (SNMPv2)
- RFC 1443: Textual Conventions for Version 2 of the Simple Network Management Protocol (SNMPv2)
- RFC 1889 & RFC 1890: RTP - Real-Time Transport
- RFC 2131: Dynamic Host Configuration Protocol
- RFC 2234: Augmented BNF for Syntax Specifications: ABNF
- RFC 2246: The TLS Protocol
- RFC 2327: Session Description Protocol (SDP)
- RFC 2474: Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers
- RFC 2475: An Architecture for Differentiated Services
- RFC 2597: Assured Forwarding PHB Group
- RFC 2705: Media Gateway Control Protocol (MGCP)
- RFC 2780: IANA Allocation Guidelines For Values In the Internet Protocol and Related Headers
- RFC 2806: URLs for Telephone Calls
- RFC 2833: RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 2848: The PINT Service Protocol: Extensions to SIP and SDP for IP Access to Telephone Call Services

- RFC 2865: Remote Authentication Dial In User Service (RADIUS)
- RFC 2976: SIP INFO Method
- RFC 3016: RTP Payload Format for MPEG-4 Audio/Visual Streams
- RFC 3047: RTP Payload Format for ITU-T Recommendation G.722.1
- RFC 3168: The Addition of Explicit Congestion Notification (ECN) to IP
- RFC 3204: MIME Type for ISUP and QSIG
- RFC 3260: New Terminology and Clarifications for Diffserv
- RFC 3261: SIP: Session Initiation Protocol
- RFC 3262: Reliability of Provisional Responses in SIP
- RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers
- RFC 3264: SDP Offer/Answer Model
- RFC 3265: SIP-specific Event Notification
- RFC 3267: Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wide-band (AMR-WB) Audio Codecs
- RFC 3272: Overview and Principles of Internet Traffic Engineering
- RFC 3288: Using the Simple Object Access Protocol (SOAP) in Blocks Extensible Exchange Protocol (BEEP)
- RFC 3311: SIP UPDATE Method
- RFC 3323: SIP Privacy Mechanism
- RFC 3515: SIP REFER Method
- RFC 3605: Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP)
- RFC 3711: The Secure Real-time Transport Protocol (SRTP)
- RFC 3725: SIP Third Party Call Control
- RFC 3761: The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)
- RFC 3824: Using E.164 Numbers with SIP
- RFC 3830: MIKEY: Multimedia Internet Keying
- RFC 3842: SIP Message Waiting
- RFC 3852: Cryptographic Message Syntax (CMS)
- RFC 3892: The Session Initiation Protocol (SIP) Referred-By Mechanism
- RFC 3952: Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech
- RFC 3959: The Early Session Disposition Type for the Session Initiation Protocol (SIP)
- RFC 3960: Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- RFC 4028: Session Timers in SIP
- RFC 4049: BinaryTime: An Alternate Format for Representing Date and Time in ASN.1
- RFC 4235: An INVITE-Initiated Dialog Event Package for the Session Initiation Protocol (SIP)
- RFC 4353: Framework for Conferencing with the Session Initiation Protocol (SIP)
- RFC 4568: Session Description Protocol (SDP) Security Descriptions for Media Streams
- RFC 4575: A Session Initiation Protocol (SIP) Event Package for Conference State

CSTA Standards (Ecma)

- ECMA-269: Services for Computer Supported Telecommunications Applications (CSTA) Phase III
- ECMA-323: XML Protocol for CSTA Phase III
- ECMA-354: Application Session Services
- ECMA TR/82: Scenarios for CSTA Phase III

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