Information

OpenScape Voice V5

OpenScape Voice is a Feature-rich Enterprise and Carrier-Class IP Voice Application that is fully integrated with, and offered as part of, a complete Unified Communications Solution.

Communication for the open minded

Siemens Enterprise Communications
www.siemens-enterprise.com/open

SIEMENS

The Leading Software-based Voice Communications System

OpenScape Voice is an enterprise-class voice application that is fully integrated with, and offered as part of, the OpenScape Unified Communications (UC) Suite. 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.

Comprehensive solution set

OpenScape Voice provides a comprehensive solution set for building or migrating a large enterprise, carrier or hosting service provider voice communications network:

- Data center-based communications
- Enterprise and carrier applications
- Dramatic operational savings
- Leading solution
- Simple migration
- Secure architecture
- Proven performance
- Open interfaces
- Comprehensive services
- Mature and stable company

The essence of OpenScape Voice is its architecture, which provides highly reliably and feature-rich communications across the campus or across the globe.

OpenScape 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.

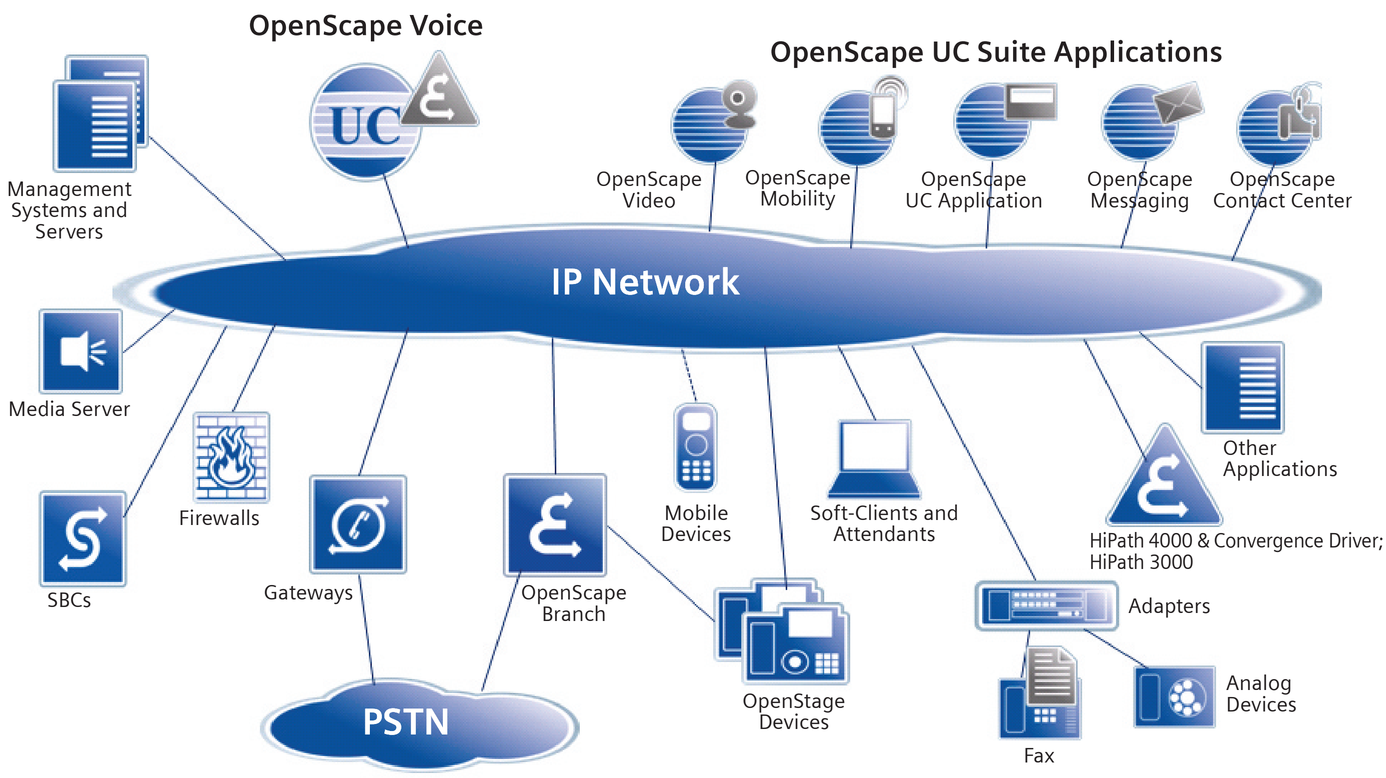
OpenScape Voice is fully integrated with the OpenScape Unified Communications (UC) Suite. The entire OpenScape UC Suite portfolio is optimized for the demands of businesses – easy to put into practice, reliable in performance, and easy to use. With it, you become even more efficient.

Business communications applications

OpenScape Voice offers a rich eco-system of UC applications that meet the business needs of the enterprise:

- Voice messaging
- Unified messaging
- Voice conferencing
- Attendant Console
- Executive/assistant/teamworking solution
- Contact Center
  - Campaign management (only released in USA)
  - Voice portal
- Mobility*
- Video*

* project-specific release



OpenScape Voice Solution Landscape

Green Enterprise Solutions

The unique scalability up to 100,000 users with only two servers is achieved by software-based growth, not by adding more hardware. This results in green enterprise solutions – smart for the environment and smart for business:

- Lowest carbon footprint of any similar system
- Lowest power cost per user of any modern IP-based communications system
- Lower cost, size, and power consumption of HVAC equipment (Heating, Ventilation, Air Condition) in the data center
- Less rack space and floor space use in the data center
- Software-based growth and applications

High-Performance IP Communication

OpenScape Voice V5 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's "Any-to-Any" IP payload switching ensures that you get the highest availability and quality.

Resiliency

Resiliency is about how well systems behave under stress conditions (e.g. overload). OpenScape Voice has an Active – Active software architecture and behaves far better with overloads and fault conditions than traditional Worker-Standby systems.

Reliability

Reliability is about how often things fail and how quickly they are restored to normal operation. It is a key feature of OpenScape Voice that it does not lose a single call-in-progress or a single billing record on any single failure.

Recovery

Recovery is about how fast the system recovers after faults or overloads. Again OpenScape Voice, because of its hardware and software architecture, recovers extremely fast, and is superior to all its competitors.

Servers

OpenScape Voice software runs on highly reliable, fault-tolerant industry-standard servers under the Linux SLES 10 SP2 64-bit 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 3 level using IP routing protocols.

OpenScape Voice utilizes Fujitsu's 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.

OpenScape Voice Features

Overview

- Simple deployment as an application in HQ or data centers
- Users anywhere the IP network goes, across the campus or across the globe
- Comprehensive Enterprise feature set
- Up to 100,000 users on a system, virtually unlimited in a network
- Supports up to 3,000 branches, 6,000 business groups
- Independent numbering plans across all groups and branches
- Proven interworking with multiple competitor's systems and in multiple networks, worldwide
- Simple per-user licensing, not tied to sites or branches
- Simple migration – one user, group, branch, or network at a time
- Comprehensive management, including servers, users, branches, licenses, including cross-platform user licenses portability, etc.
- No loss of call control for any calls in progress or any billing data on any single failure anywhere
- Security independently certified
- Feature interworking with HiPath 4000, especially network-wide call pickup (see *OpenScape Voice Gateways*)
- Attendant Solution (OpenScape Concierge) that does not need Contact Center (Automatic Call Distribution (ACD))
- Executive/Assistant Application

OpenScape Voice Entry

OpenScape Voice Entry is an attractive simplex deployment configuration of the OpenScape Voice Application.

- OpenScape Voice Entry is targeted at enterprises requiring up to 800 SIP users and 400 trunks.
- OpenScape Voice Entry uses the IBM x3250 M2/M3 server and cannot be extended beyond its maximum of about 800 users.
- OpenScape Voice Entry Edition includes an integrated Session Border Controller (SBC) that can be used for SIP trunking.

OpenScape Voice V5

There are several important new solution components and features being introduced in V5. These include:

OpenScape Branch enhancements

- Four size options - up to 50, 250, 1,000 and 6,000 users; see *OpenScape Branch*
- New OpenScape Branch 50i with integrated Analog Adapter and PSTN Gateway

Expanded solution landscape

OpenScape Voice is integrated in an expanded solution landscape as shown in the graphic on page 2.

HiPath Cordless IP

HiPath Cordless IP* updates IP communications systems by a campus-wide mobility solution.

The base stations establish a network from cells and communicate with the handsets. The multi-cell technology enables the user to move between the cells with their handsets during a call.

The software of the base station contains complete DECT and IP functionality. The software does not need to be configured and administered locally for each base station but instead can be conveniently operated centrally via the HiPath Cordless IP server software.

The HiPath Cordless IP handsets allow calls to be made in the area covered by the network.

Detailed informationen on HiPath Cordless IP can be found in the corresponding data sheet.

* planned for release in Q2 2011

OpenScape Convergence Driver

- OpenScape Convergence Driver is the standard server deployment of the new HiPath 4000 V6 software architecture and is an important step in the OpenPath Transformation strategy for new and existing OpenScape Voice solutions. The Convergence Driver strategy is to combine the OpenScape Voice SIP capabilities with the HiPath 4000 capabilities, offering our customer the best of both worlds in hybrid deployments.
- With OpenScape Convergence Driver, our solution bridges the OpenScape Voice and HiPath 4000 worlds seamlessly and cost-effectively on standard servers, allowing customers to move to OpenScape Voice when they are ready.

OpenScape Alarm Response

- Current demands on modern telecommunications go far beyond simply making telephone calls. In particular, they include the automation and optimization of critical communication in emergency and crisis situations. With its high flexibility and its multi-faceted communication options, **OpenScape Alarm Response** Pro caters for many of these demands.
- **OpenScape Alarm Response** Eco is the ideal mini-server for alarms for low-end customer needs, suitable for nursing homes, small branch offices and limited use in larger enterprises.
- Detailed informationen on **OpenScape Alarm Response** Pro and **OpenScape Alarm Response** Eco can be found in the corresponding data sheets.

Virtualization

- Each application and its required operating system runs on top of a thin "Virtualization Layer".
- The "Virtualization Layer" enables multiple applications to run simultaneously on the same server – typically saving 30% to 60% of servers.
- Virtualization of Voice and SBC with VMWare
- Support for high availability dynamic relocation with VMotion

Executive/Assistant Application

The Executive/Assistant Application is an XML application developed to work only on the OpenStage 60 and 80 phones and interacts with OpenScape Voice to provide an intuitive implementation of the Executive/Assistant function that up to now has only been available on TDM systems. It comprises the following functions:

- Streamline the executive's calling processes with the support of one or more assistants
- Assistants control and manage calls for executives, providing support with a great degree of flexibility
- One or more assistants can answer all incoming calls for the executive, handling them exactly as the executive wants
- Incoming calls for the executive are directly forwarded to the assistant, or both the executive and assistant are signaled simultaneously
- Assistant can always monitor all incoming calls for the executive and react accordingly

OpenScape Concierge

OpenScape Concierge is a Unified Communications (UC) Attendant Console application for OpenScape Voice. It extends the benefits of UC beyond desktop users to attendants and receptionists with real-time UC-based presence status information. It also supports call queue and corporate directory integration, allowing telephone attendants to more easily direct incoming calls to anyone in the organization.

Detailed information on OpenScape Concierge can be found in the corresponding data sheet.

OpenScape Unified Communications Suite

Unified Communications

OpenScape UC Application

The OpenScape Unified Communications (UC) Application is a high-functionality collaboration application that fits into an enterprise'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:

- Tools for productivity, agility and mobility, such as Presence, one-number service, UM and IM
- Collaboration tools, such as group presence and voice, web, and video collaboration, and chat
- Business Process Integration tools

OpenScape Fusion accelerators

A range of open integration capabilities and offerings that include:

- Pre-built UC component plug-ins, such as for Microsoft Outlook and Lotus Notes
- Integration for social media and groupware
- Open integration programs, such as SDKs and a developer's portal

OpenSOA

OpenSOA provides exclusive "from-bottom-to-top", SOA-based software integration hooks.
Integration into the Microsoft and IBM ecosystems, as well as the ability to integrate into other line of business applications through Communication-Enabled Business Processes (CEBP), leverages our OpenSOA capabilities.

OpenScape Contact Center

OpenScape Contact Center is a comprehensive set of integrated software applications that suit every type and size of organization, offering a range of customer experiences. From customer service to collections, agent-assisted to self-service, OpenScape Contact Center provides the customer care solutions you need to retain and grow your customer base.

The OpenScape Contact Center solution set covers multi-channel inbound, outbound and self-service/IVR that improve the effectiveness and efficiency of your contact center operations.

OpenScape Mobility

A robust collection of tools to tailor mobility across your enterprise, including:

HiPath Wireless

HiPath Wireless is our elegant and powerful WLAN solution.

HiPath MobileConnect

HiPath MobileConnect is an appliance-based fixed-mobile convenience approach to enable seamless mobility between your WLAN and the public network, thereby increasing flexible mobility while significantly reducing mobile carrier costs.

OpenScape Mobile UC

OpenScape Mobile UC client brings the power of integrated Open Communications to your mobile device.

OpenScape Video

A suite of tools to integrate advanced video applications in UCC.

- Videoconferencing solution integrated with voice & UC applications
- Unifying video from boardroom, meeting room, and desktop with voice on one network
- Hi-Def videoconferencing democratized
- Providing the same scalability when controlling video sessions

OpenScape Xpressions

OpenScape Xpressions V6 enables built-in voice messaging, unified messaging, voice conferencing, web conferencing, instant messaging, text messaging, presence, and fax. These unified communications options coupled with CTI services help embed powerful communications capability directly into business processes creating an efficient and effective workplace.

OpenScape OpenExchange

OpenScape OpenExchange provides IP-Least Cost Routing between existing PBX systems (any vendor) and optional conferencing.

FastViewer

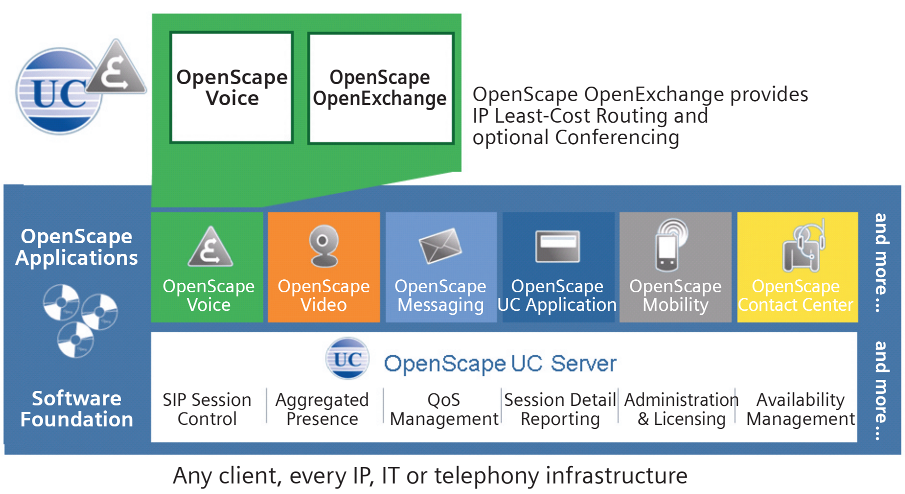
The FastViewer desktop sharing software is an extremely secure and simultaneously simple solution for communication of a large number of participants from different locations at a virtual table.

The software enables conferences, presentations and training throughout the world without any installation.

The solutions suit for enterprises of any size, from smallest companies to largest organizations.

Integration/interworking with other products in the UC solution landscape

- Application-provided billing party via SIP
- Application-provided call correlation via SIP
- 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: media encryption
- OpenScape Branch support
- OpenScape Contact Center integration
- OpenScape Media Server:
  - CALEA/LI support
  - Support for additional languages/countries
- OpenScape UC Application integration
- OpenStage support
- RG 8700: media encryption
- SIP trunking customization options
- Session border controller
- SIP signaling manager: internal audit mechanism



OpenScape UC Suite: OpenScape UC Server and OpenScape Voice

OpenScape Voice Management

System management tools for OpenScape Voice V5 include the following:

- OpenScape Voice Assistant
- Deployment Service (DLS)

OpenScape Voice Assistant

For all user configurations, OpenScape Voice Assistant is the strategic web-based tool for administering OpenScape Voice V5. 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.

Deployment Service

The Deployment Service (DLS) provides an integrated solution for customers and service personnel to administer IP devices (IP phones and clients) in HFA/H.323 and SIP-based networks, including OpenScape Voice. DLS is the central system where device- and QoS-related parameters of HiPath IP devices are administered for the customer's entire network. Additionally, DLS takes over the distribution of certificates for deploying TLS (Transport Layer Security) and is also able to create certificates where there is no existing customer PKI (Public Key Infrastructure) framework.

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 Accounting Management (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 Fault Management (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

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 Management

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 Voice Media Servers

OpenScape Voice V5 offers the following Media Server options:

- OpenScape Media Server can support up to 75 ports for the internal media server and up to 1000 G.711 ports for the stand-alone media server.

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

OpenScape Media Server

The OpenScape Media Server is an integral part of OpenScape Voice systems for medium-size enterprises supporting from 300 to 100,000 subscribers, per single server. 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, stand-alone server.

OpenScape Voice Gateways

To access the Public Switched Telephone Network (PSTN), the OpenScape Voice V5 solution provides the following gateway options for media and signalling:

- HiPath 4000 Survivable Media Gateway
- HiPath 3000 Media Gateway including HiPath 3000 V8
- RG 8700 Survivable Media Gateway
- Mediatrix Gateways

HiPath 4000 and HiPath 3000 Survivable Media Gateways (SMG)

In branch offices with a HiPath 4000 or HiPath 3000, survivability is made possible by use of the OpenScape Branch SIP proxy functionality. 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 OpenScape Branch resumes forwarding the requests to OpenScape Voice as usual.
Interworking of HiPath 4000 with network-wide call pickup is now supported.

RG 8700 Survivable Media Gateway

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 switches 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 V5:
- 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 V1.3 software adds SIP-Q functionality for connectivity to HiPath 4000 and third-party products which support QSIG.

Mediatrix Gateways for Small Branch Offices (SBO)

The OpenScape Branch connects these locations to the OpenScape Voice and provides the survivability for the small office scenario. It also supports SIP Trunking functionality and can also interwork with the gateways from Mediatrix.

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.

SIP Trunking

SIP Trunking directly interconnects VoIP communications between enterprise IP-PBXs and the PSTN via a SIP-based carrier. It uses the SIP standard for call control and can combine all forms of traffic (voice, UC, video) on a single connection to the service provider. This is significantly lower cost than separate data and voice connections and provides far greater flexibility.

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. For the branch office location, the OpenScape Branch's Session Border Controller (SBC) functionality is also a very efficient and cost-effective solution.

OpenScape Branch

OpenScape Branch assures continued communication services – while providing a feature-rich set of survivability capabilities at a remote branch location – during the loss or degradation of service between the remote branch and the main office.

OpenScape Branch is offered on several hardware platforms, allowing a wide range of maximum user capacity:
up to 50, 250, 1000, 6000 users.

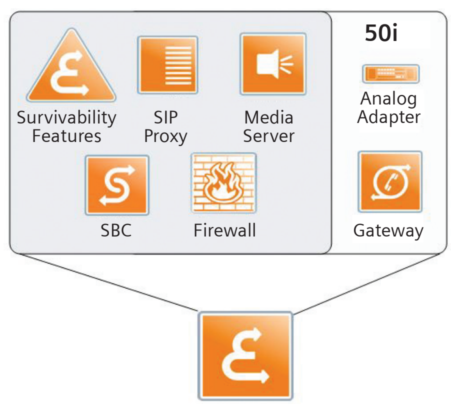
All OpenScape Branch models include survivability features, Proxy, Media Server, Session Border Controller (SBC), and Branch SIP trunking functionalities.

The new OpenScape Branch 50i provides the additional functionalities of integrated Analog Adapter and PSTN Gateway.

The local Media Server supports tones, announcements and conferencing reducing the bandwidth needed to provide the same resources from a central location. This yields direct operational cost savings.

The Session Border Controller (SBC) provides security functions like Firewall and Virtual Private Network (VPN).

The OpenScape Branch is fully manageable via the Common Management Portal (CMP) as a single network element, making it an exceptional value.



OpenScape Branch

SIP Endpoints

The following Siemens SIP endpoints are supported:

- OpenStage 80/80G/60/60G OpenStage 40/40G/20/20E/20G/15
- optiPoint WL2 professional S (wireless)
- OpenScape Personal Edition

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

OpenStage

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 following models:

OpenStage 80

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

OpenStage 60

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 40

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 20

OpenStage 20, the economy model, is a full-featured speakerphone and a universal solution for efficient and professional telephony.

OpenStage 20

OpenStage 15

OpenStage 15 is a full-featured speakerphone with display and illuminated feature keys that could be used for up to 8 line appearances, for example.

OpenStage 15

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 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.

OpenScape Personal Edition

OpenScape Personal Edition is an IP Soft-phone for installation on portable laptop and desktop PCs.

The Personal Edition serves as an entry point into OpenScape UC Application and can be used as a stepping stone for the subsequent deployment of OpenScape Enterprise Edition.

The new user interface has the look and feel of Windows® Office 2007 and offers the user a wide range of technical and graphical features that can effectively replace desktop phones entirely. The IP Softphone thus provides the ideal solution for normal users who want to eliminate their desktop phones or mobile users who are not tied down to a specific workplace and view their notebooks as their office.

Analog Adapters

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

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 V5 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, 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 10 Service Pack 2 (SLES 10 SP2). A SolidTech database runs on each server.

Each RX330 S1 server has two (2) Dual-Core or Quad-Core 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.)

Virtualization

Virtualization is one of the key architectural cornerstones in the continuing development of the complete OpenScape Unified Communications Suite as well as a key competence focus in our professional services organization.

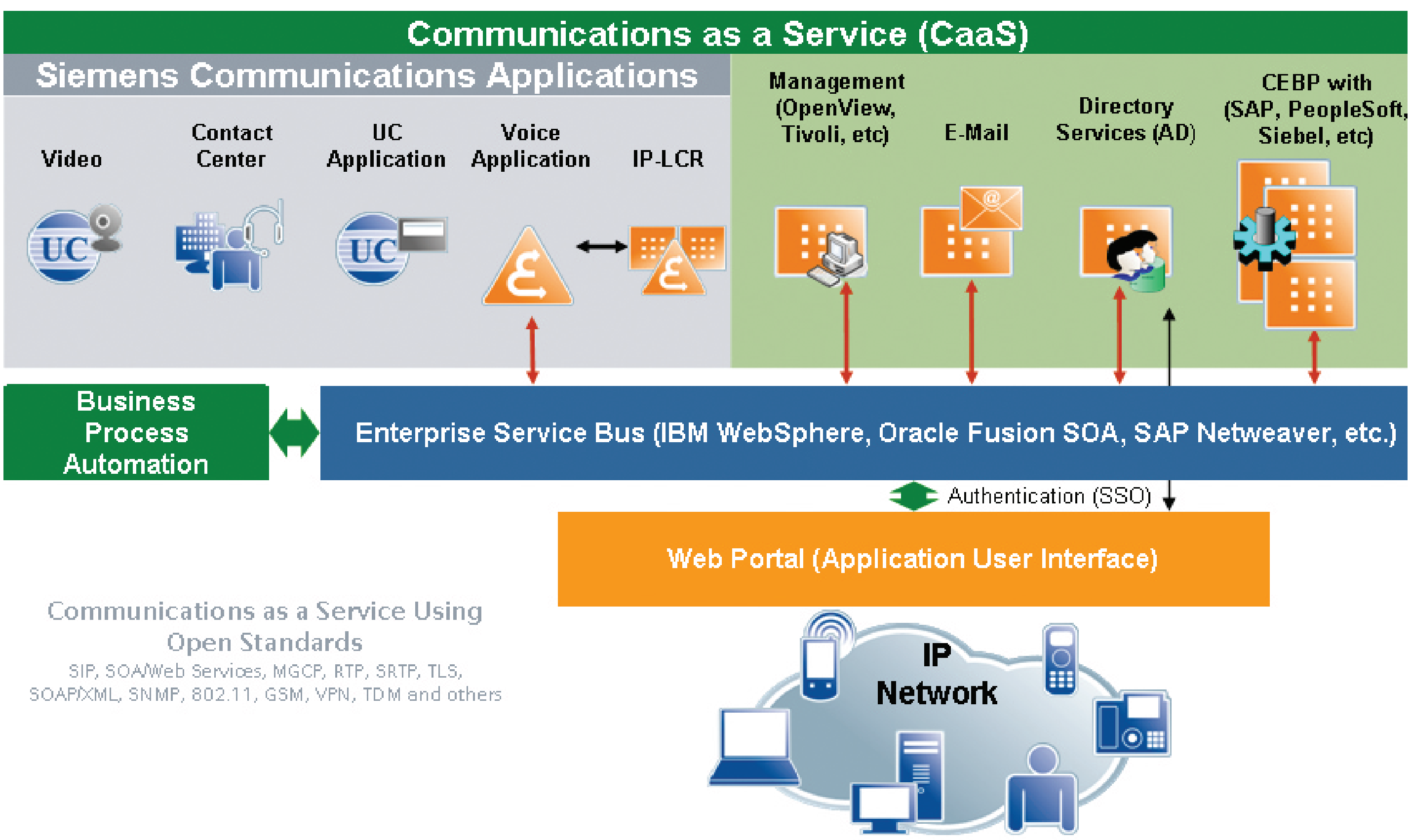
Virtualization packages each Application plus its required Operating System (OS) as a Virtual Machine, and provides a thin virtualization layer that runs on top of the server hardware to support these virtual machines. Multiple applications (each with its own OS layer) can be installed on top of the virtualization layer.

The virtualization layer also enables applications with different operating systems (e.g. Linux and Windows) to run on the same physical server at the same time. Management tools are also provided to load, monitor, update and move virtualized applications. Virtualization can also be used to provide some levels of redundancy.

The following applications work in a virtual environment using VMware ESX/ESXi 4.0 technology:

- OpenScape Voice
- OpenScape UC
- CMP
- OpenScape Media Server
- DLS
- HiPath Management tools (Accounting, User, QoS and Fault Management)

OpenScape Voice V5 R0 allows virtualized hardware-independent deployment of all its related software components (HQ and branch) using VMware ESX/ESXi 4.0 technology.



Hardware Platforms

With OpenScape Voice V5, the following hardware platforms can be ordered:

- IBM x3550 M2*
- IBM x3550 M3*
- IBM x3250 M3 (for OpenScape Entry)
- Fujitsu RX200 S6**
- Fujitsu RX330 S1**

* The IBM x3550 M2 can be ordered until February 2011. Afterwards, it is replaced by the IBM x3550 M3.
** The Fujitsu RX330 S1 can be ordered until February 2011. Afterwards, it is replaced by theFujitsu RX200 S6.

Note: The IBM x3650T is supported for upgrades only.

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):

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

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 Wideband (AMR-WB) Audio Codecs
- RFC 3272: Overview and Principles of Internet Traffic Engineering
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Reference No.: A31002-H8050-D100-1-7629