Information

OpenScape Voice V4R1
Simple yet Sophisticated

OpenScape Voice is the world's only Enterprise and Carrier-Class Voice Application that is fully integrated with, and offered as part of, a complete Unified Communications Solution. It provides a comprehensive solution for building or migrating a Large Enterprise, Carrier or Hosting Service Provider Voice Communications Network.
The essence of OpenScape Voice is its architecture, which forms the foundation for a new IT-integrated and business process-focused communications solution, providing the lowest Total Cost of Ownership (TCO) and Best Return on Investment (RoI).

Communication for the open minded

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The OpenScape Voice Solution Landscape

OpenScape Voice is an enterprise-class voice application that is fully integrated with, and offered as part of, a complete unified communications solution, 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.

OpenScape 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, OpenScape Voice creates technology choices, allowing customers to implement well thought out communication strategies at their own pace.

OpenScape 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.

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.

Best for Your Business

OpenScape 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 OpenScape 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 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.

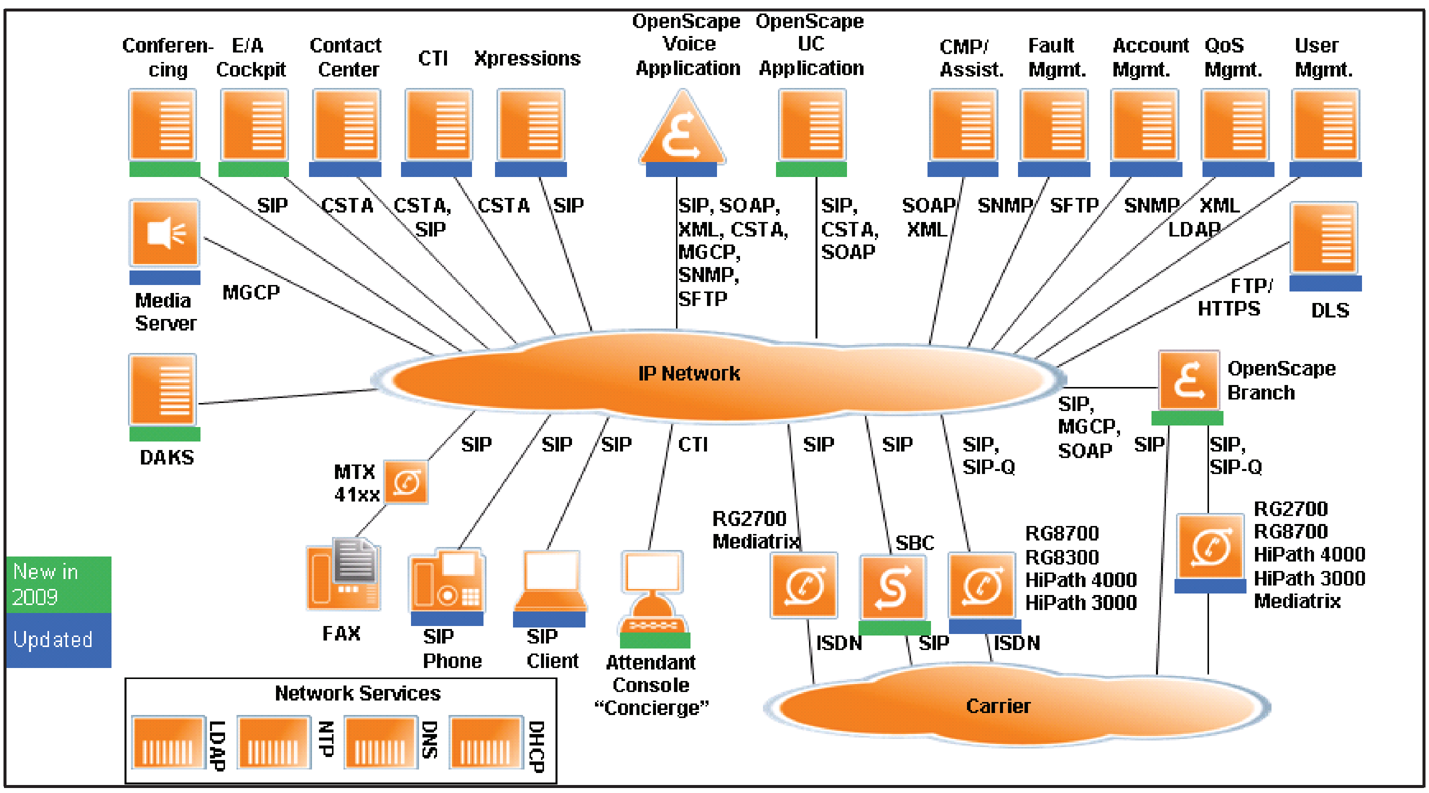
Key Attributes of OpenScape Voice

The key attributes are as follows:

Simple yet Sophisticated:
- Simple
  - To deploy and manage
  - To migrate and upgrade
  - Per-user licensing model
- Sophisticated
  - Architecture
  - Feature set
  - Networking options
  - Interworking and applications support

Mature yet Innovative:
- Mature
  - Proven, stable and reliable
  - In service worldwide
  - Large enterprise, carriers and MSPs
  - Comprehensive deployment options
- Innovative
  - Cloud and social media
  - Expanded networking architecture
  - Built on a sure and proven foundation

Standards yet Open options:
- Standards
  - Most open solution
  - Standards compliance
- Open options
  - Configuration options
  - Applications options
  - Networking and branch options
  - APIs for building new options



Economic yet Green
- Economic
  – Lowest TCO, best RoI
  – Simple but powerful management
  – IT and data-center integrated
- Green
  – With OpenStage devices provide lowest carbon footprint per user
  – Lowest power-cost per user

Value

With OpenScape Voice, businesses benefit from the investments already made in their customers, partners, employees, and communications infrastructure. OpenScape Voice V4 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, OpenScape Voice V4 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, OpenScape 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 OpenScape Personal Edition solution (soft-client). You set the pace in accordance with your demands and ideas.

Benefits of Installing OpenScape Voice V4

- 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 V4 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 V4's "Any-to-Any" IP payload switching ensures that you get the highest availability and quality.

Resiliency, Reliability and Recovery

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

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 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.

OpenScape Voice software runs on highly reliable, fault-tolerant industry-standard servers under the Linux SLES 10 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.

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 CO_2 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

Features in OpenScape Voice V4

The many new or enhanced features being introduced with OpenScape Voice V4 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
- New smaller server configuration for simplex configuration for up to 800 SIP users

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
- 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 Branch support
- OpenScape Contact Center integration
- OpenScape Media Server enhancements
  - CALEA/LI support
  - Support for additional languages/countries
- OpenScape UC Application integration
- OpenStage support
- Radisys Convedia CMS-3000: security, QoS and language enhancements
- RG 8700 enhancements: media encryption
- SIP trunking customization options
- Session border controller enhancements
- SIP signaling manager: internal audit mechanism

New Features in OpenScape Voice V4R1

There are several important new solution components and features being introduced in V4R1. These include:

- OpenScape Branch V1R2 (four size options - up to 50, 250, 1,000 and 6,000 users; see *OpenScape Branch*)
- OpenScape Voice Entry Edition (for enterprises of up to 800 users)
- New Attendant Solution (OpenScape Concierge) that does not need Contact Center (Automatic Call Distribution (ACD))
- New Executive / Assistant Application
- Digital alarm and communication server (DAKS) added as a tested element to the Solution Landscape
- HiPath 3000 V8 added as a tested element to the Solution Landscape (see *OpenScape Voice V4 Gateways*)
- Additional feature interworking with HiPath 4000, especially Network-Wide Call Pickup (see *OpenScape Voice V4 Gateways*)

OpenScape Voice Entry Edition

OpenScape Voice Entry is an attractive entry-level simplex deployment configuration of the OpenScape Voice Application which:

- is targeted at smaller enterprises and supports up to 800 SIP users and 400 trunks
- uses the IBM x3250 M2 server, and cannot be extended beyond it's maximum of about 800 users
- includes an integrated Session Border Controller (SBC) that can be used for SIP trunking

OpenScape Concierge

The OpenScape Concierge Attendant Console as part of OpenScape Contact Center Extensions provides comprehensive Attendant function for OpenScape Voice (with and now also without the use of an external ACD system) and also provides a seamless migration from an IP/TDM platform to OpenScape Voice. It comprises the following features:

- User-friendly attendant console solution for OpenScape Voice and HiPath 4000
- Easy installation and configuration
- Character-Separated Value (CSV) importer for flexible customer data synchronization
- Standard Lightweight Directory Access Protocol (LDAP) synchronization tool for customer phone book integration
- Fast call pickup mode for pure attendant requirements
- Optimization of availability through integration in the OpenScape Contact Center routing strategy possible
- Operation of the solution on networked HiPath / OpenScape Voice systems
- Detailed display of customer data for incoming calls
- Extensive search function for efficient database research
- Presentation of additional information about destination stations
- Use of ACD and Computer Telephony Integration (CTI) functions in the intuitive user interface
- Snapshot function while switching, station status is already known before switching
- UC-based status information and status change for UC contacts
- Pictures for contacts can be shown
- Incorporation of a Web browser for using Web-based information sources
- 252 individual repertory keys (6 sections) with integrated presence function, also for stations in system network
- Up to 20 always visible "speed dial" buttons
- Administration tool for managing the data volume
- Adaptable layout of Concierge interface
- Extensive real-time and historic reporting, if you use Concierge with OpenScape Contact Center (OSCC)
- Standard real-time and historic reporting, if you use Concierge without OSCC
- Personal and group park queue
- Appending calls to busy stations

Executive / Assistant Application

The Executive / Assistant Application is an XML application developed to work only on the OpenStage Phones 60 and 80 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 Voice V4 Management

System management tools for OpenScape Voice V4 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 V4. 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 workpoints 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 workpoints 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 V4 Media Servers

OpenScape Voice V4 offers the following media server options:
- RadiSys Convedia CMS-3000 media server for up to 360 ports
- OpenScape Media Server
  up to 75 ports for the internal media server and up to 500 ports for the standalone media server
  for up to 500 channels with G.711 codec

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

RadiSys Convedia CMS-3000

The RadiSys Convedia CMS-3000 is a turnkey, 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 from 300 to 50,000 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, standalone server.

OpenScape Voice V4 Gateways

To access the Public Switched Telephone Network (PSTN), the OpenScape Voice V4 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
- RG 2700 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 through the 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 V4: 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.

RG 2700 Survivable Media Gateway

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.

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.

OpenScape Branch

The OpenScape Voice V4 solution provides the following branches:
- OpenScape Branch 50
- OpenScape Branch 250
- OpenScape Branch 1000
- OpenScape Branch 6000

OpenScape Branch 50

This new hardware is being released to support branch scenarios with less than 50 users.

OpenScape Branch 250 and 1000

New with OpenScape Voice V4, is the support of the new OpenScape Branch 250, and OpenScape Branch 1000 (for up to 250 and 1000 users respectively) where the remote branch office gets empowered with remote survivability, SIP Trunking, local tones, announcements and conference and Session Border Controller functionalities. During the total loss or partial service degradation between the remote branch and the HQ, the OpenScape Branch assures continued communication services with a feature rich set of survivable capabilities.

OpenScape Branch 6000

As there is a market demand for OpenScape Branch configurations larger than the current limit of 1000, OpenScape Branch 6000 has been introduced to support up to 6000 subscribers per branch using either of the following servers:
- IBM x3550 M2
- Fujitsu Primergy RX330 S1

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.

SIP Endpoints

The following Siemens SIP endpoints are supported:
- OpenStage 15/20/20E/40/60/80
- optiPoint 410 S and 420 S
- 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 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

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 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 Softphone 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.

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.

OpenScape Xpressions

OpenScape Xpressions combines voice, fax, e-mail and text (Short Message Service, SMS) services on a Windows 2003/2008 platform and transforms them into a Unified Messaging system for use together with OpenScape Voice.

Built using a modular, scalable client/server architecture, OpenScape Xpressions can be configured to meet users' individual communication needs. New functionality in OpenScape Xpressions V6 supports Unified Communication features like Audio- and Web-Conferencing and the OpenScape Web Client. OpenScape Xpressions can alternatively be used as central conferencing system.

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.

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.

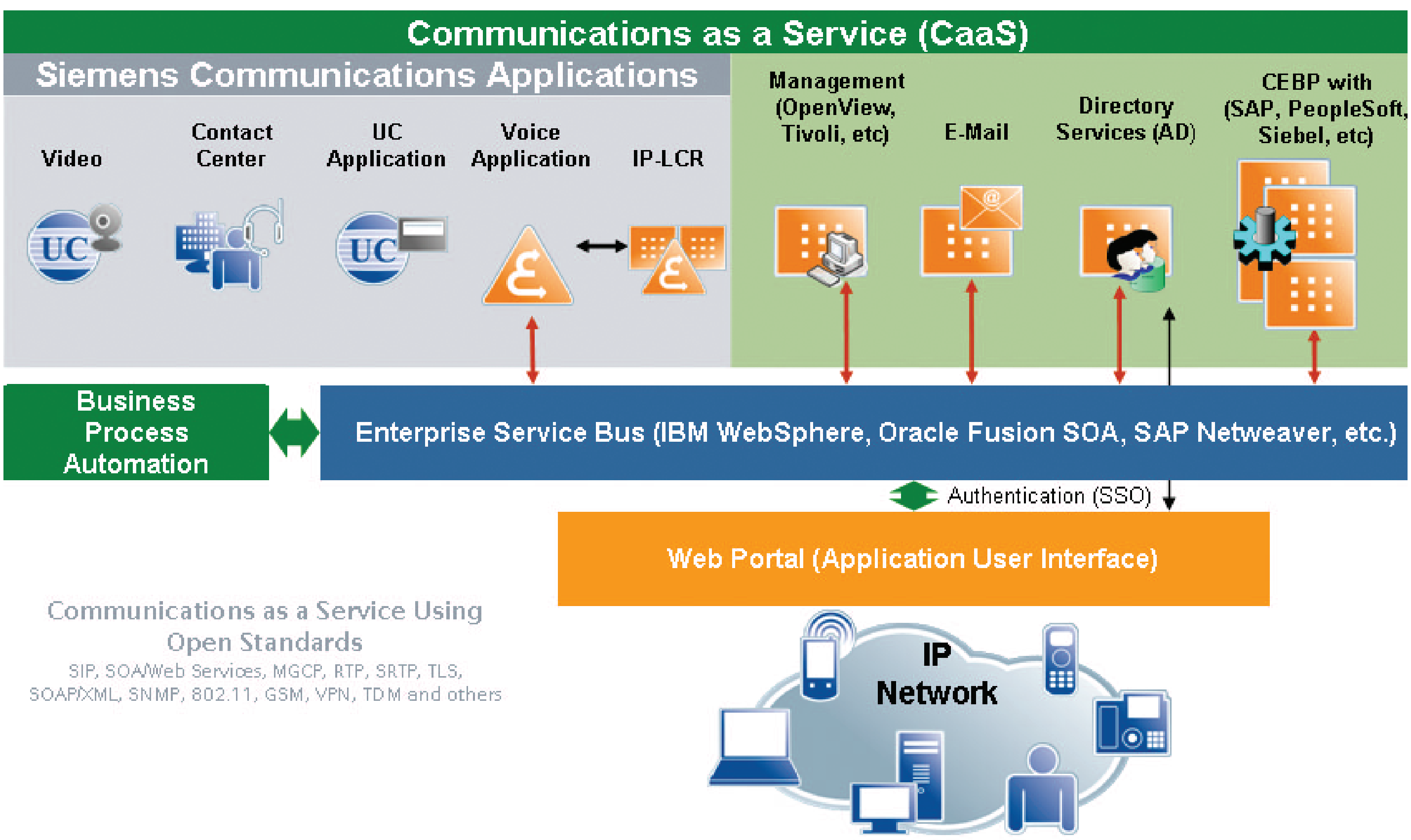
OpenScape Voice Server Technical Data

The OpenScape Voice V4 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 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 V4 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 Wideband (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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