HowTo

OpenStage@Asterisk Installation and Maintenance Guide

Issue 1.0

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Scope

This document provides a best practice guide on setting up, operating, servicing, and troubleshooting OpenStage phone in an Asterisk environment.

Contents

Preparation

This chapter contains all information that is necessary to connect an OpenStage phone to an Asterisk based communication system.
This includes the power supply options (PoE or external supply) for each OpenStage model and its possible sidecar combinations. To enable a secure environment, 802.1x support for OpenStage is specified.
For autoconfiguration, LLDP-MED and DHCP can be used. In addition to the standard DHCP options, SEN proprietary enhancements allow for assigning the address of a provisioning service at phone startup for easy Plug & Play installation. By means of the provisioning interface (WPI), mass deployment scenarios and remote administration during the phone's lifecycle are supported.
To facilitate configuration of a single phone by the administrator or user, OpenStage phones feature a web interface in addition to the phone's local interface.

Supplying Power for the Phones

The OpenStage phone family can be powered by

- External power supply
- Power over Ethernet (PoE)

<table>
<tr><th>Feature</th><th>OpenStage 15</th><th>OpenStage 20E
OpenStage 20
OpenStage 20 G</th><th>OpenStage 40
OpenStage 40 G</th><th>OpenStage 60
OpenStage 60 G</th><th>OpenStage 80
OpenStage 80 G</th></tr>
<tr><td>Legacy optiPoint external Power Supply (EU-, US- or UK-plug)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td></tr>
<tr><td rowspan="2">New OpenStage Power Supply (EU-, US- or UK-plug)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td><td>Yes (optional)</td></tr>
<tr><td colspan="5">In line with the further development of our portfolio we are now able to offer smaller and lighter power supplies for all OpenStage and optiPoint 410/ 420/ 500 variants.
They feature a higher degree of efficiency leading to 14 - 19% lower power consumption, depending on the connected devices.
Important: The new OpenStage power unit is a device of product category. The protection concept of the power installation needs to achieve the requirements of earth conductor (e.g. Schukostecker Typ F, country specific).</td></tr>
<tr><td>Power over Ethernet: IEEE 802.3af</td><td>Yes</td><td>Yes</td><td>Yes</td><td>Yes</td><td>Yes</td></tr>
<tr><td>Power over Ethernet, Cisco proprietary mode</td><td>No</td><td>No</td><td>No</td><td>No</td><td>No</td></tr>
<tr><td>802.3af Power class</td><td>OS 15: 1</td><td>OS 20E : 1
OS 20 : 1
OS 20 G: 2</td><td>OS 40: 2
OS 40 G: 3</td><td>OS 60 : 3
OS 60 G: 3</td><td>OS 80 : 3
OS 80 G: 3</td></tr>
</table>

Energy saving mode
To reduce the energy consumption to a minimum, OpenStage phones offer an energy saving mode. The display backlight (phone and Key Module, if attached) is switched off after a configurable timeout.
With OpenStage 40, the main display and key module backlight will be switched off after 90 seconds of inactivity (firmware version V2R0 onwards). Readability even without backlight is ensured by the transflective display.
With OpenStage 60 and 80, the timer is configurable by the administrator (**Local Functions** > **Energy saving**); the timeout ranges between 2 and 8 hours.

Power consumption [W] - Fast Ethernet variants

		Energy saving mode		Idle state		During call (handset)		Ringing (max. vol.)	
	# of Key Modules	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply
OpenStage 15	-	2,0	2,9	2,0	2,9	2,3	3,2	3,9	4,1
OpenStage 15 with 1 OpenStage Key Module 15 (9 LED's on)	1	2,2	3,1	2,2	3,1	2,5	3,4	4,1	4,3
OpenStage 20/20E	-	1,8	2,6	1,8	2,6	2,0	2,8	2,9	3,4
OpenStage 40	-	1,9	2,8	2,4	3,5	2,7	3,6	4,0	4,4
OpenStage 40 with 1 OpenStage Key Module 40	1	2,1	3,0	3,1	4,2	3,4	4,4	4,6	5,1
OpenStage 40 with OpenStage Key Module 40	2	2,3	3,2	3,8	4,9	4,1	5,2	5,2	5,8
OpenStage 40 with 1 OpenStage Key Module 15 (9 LED's on)	1	2,1	3,0	2,6	3,7	2,9	3,8	4,2	4,6
OpenStage 60	-	2,4	3,3	5,6	6,9	5,8	7,1	6,8	7,9
OpenStage 60 with 1 OpenStage Key Module 60	1	2,6	3,5	6,2	7,6	6,5	7,9	7,6	8,5
OpenStage 60 with 2 OpenStage Key Module 60	2	2,8	3,7	6,9	8,3	7,1	8,6	8,3	9,3
OpenStage 80	-	2,3	3,2	6,3	7,7	6,5	7,9	7,6	8,6
OpenStage 80 with 1 OpenStage Key Module 80	1	2,5	3,4	7,0	8,5	7,2	8,7	8,4	9,4
OpenStage 80 with 2 OpenStage Key Module 80	2	2,6	3,6	7,6	9,2	7,9	9,5	9,1	10,2

Power consumption [W] - Gigabit Ethernet variants

			Energy saving mode		Idle state		During call (handset)		Ringing (max. vol.)	
	# of Key Modules	Busy Lamp Field	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply	Power over LAN (802.3af)	Power via OpenStage switched-mode power supply
OpenStage 20 G	-	-	3,6	4,7	3,6	4,7	4,0	5,0	4,5	5,3
OpenStage 40 G	-	-	3,8	4,9	4,4	5,6	4,7	5,9	5,2	6,3
	1	-	4,0	5,1	5,1	6,4	5,5	6,8	6,2	7,4
	2	-	4,1	5,3	5,9	7,3	6,3	7,7	6,9	7,9
	-	1	4,6	5,8	5,3	6,6	5,6	7,0	6,4	7,4
OpenStage 60 G	-	-	3,8	4,9	7,0	8,7	7,4	9,0	7,8	9,4
	1	-	4,0	5,1	7,7	9,3	8,3	10,0	9,3	10,3
*Please see note *)*	2	-	*4,2 *)*	5,3	*8,4 *)*	10,0	*9,0 *)*	10,7	*10,1 *)*	11,0
OpenStage 80 G	-	-	3,6	4,9	7,4	9,2	7,9	9,6	8,4	10,0
	1	-	4,4	5,6	8,1	9,8	8,4	10,1	9,4	10,6
*Please see note *)*	2	-	*4,5 *)*	5,7	*8,8 *)*	10,6	*9,2 *)*	10,9	*10,3 *)*	11,5

*) These values are still within the 802.3af PD class 3, which allows up to 12,95 W, but they are averaged, not maximum values. As soon as the USB interface is used, PD class 3 is exceeded. Therefore, an external power supply has to be used for an OpenStage 60G/80G with 2 Key Modules.

Connecting OpenStage Phones to the IP Network

802.1x

OpenStage phones support 802.1x EAP-TLS. Certificates for authentication can be downloaded via the WPI.

LLDP-MED Configuration Options

OpenStage SIP phones support the layer 2 protocol LLDP-MED (Link Layer Discovery Protocol-Media Endpoint Discovery). It is used for simplification of auto-configuration and network management. The auto-configurable parameters of OpenStage phones are mainly the VLAN ID, power consumption (power class) and quality of service (QoS) parameters.

LLDP-MED is able to replace various other established mechanisms like DHCP options, manual configuration, or proprietary solutions like Cisco CDP. Example parameters are LAN speed and duplex discovery, network policy discovery (VLAN and QoS capabilities) or extended power via MDI discovery (PoE).
When an OpenStage phone is connected to a switch with LLDP-MED capabilities, the phone is able to

a) advertise and receive a VLAN ID,

b) advertise and receive QoS parameters,
c) advertise the power requirements to the LAN access switch by means of an "Extended Power via MDI" TLV.

Note: LLDP-MED should only be used with LLDP enabled network access switches. Older network access switches that don't adhere to the 802.1D-1998 MAC bridging specification might appear to be propagating the LLDP multicasts through the subnet. In this case, LLDP-MED should be deactivated on the phone. For further information, please refer to the OpenStage Asterisk Admin Guide [3].

DHCP Options

The following parameters can be obtained by DHCP:

Basic Configuration

- IP Address
- Subnet Mask (option 1)

Optional Configuration

- Default route (option 3)
- Static IP routing (option 33)
- SNTP server (option 42)
- Timezone offset (option 2)
- Primary/secondary DNS server (option 6)
- DNS domain name (option 15)
- SIP Addresses / SIP Server & Registrar (SIP Server option 120)
- Vendor unique (option 43)

The vendor specific option (code 43), or alternatively, a vendor class, is used to provide the phone with the location of an optional configuration/ provisioning service. By this means, full Plug & Play is possible (see the following section). For further information, including an example configuration for dhcp, please refer to the OpenStage Asterisk Admin Guide [3].

Plug & Play – One Step Provisioning and Configuration

Provisioning via the WPI (WorkPoint Interface)

The WPI allows for controlling and provisioning the phone by a remote service using XML messages which are transmitted over HTTPS. Unlike many other VoIP phones, which are limited to prefabricated configuration files for download at startup time, OpenStage phones exchange information with the provisioning service continuously. When local changes have been executed on the phone, it will inform the provisioning service automatically.
Any kind of administration task is supported, for instance, updating the firmware on a selection of phones, or setting SIP codes for server-based telephony features.
For further information, please refer to the WPI Developer's Guide [8].

Plug & Play / Autoprovisioning

A fully automated mass rollout of OpenStage phones can be realized by combining a DHCP server and the provisioning service. On startup, the phone receives the IP address of the provisioning server from the DHCP server. After that, it contacts the provisioning service. The provisioning service may then request all settings from the phone in order to decide which

parameters must be set or updated. When all these parameters have been sent to the phone, it is ready for operation.
For further information, please refer to the WPI Developer's Guide [8].

If a Firewall or NAT get in the Way

In case the phones and the provisioning service reside in different networks or subnets that are separated by a firewall and/or NAT, it may be impossible for the provisioning service to contact the phones.
To enable a solution for this problem, the phone can be configured to periodically poll the provisioning service, or a special proxy, for new messages. Thus, provisioning service driven interactions are possible even when the provisioning service is located behind a firewall, or in a DMZ. For further information, please refer to the WPI Developer's Guide [8], Section 1.4.4.3, "Polling Request To Bridge A Firewall" and Section 3.1.2.2, "Provisioning Service Located Behind A Firewall".

Single Phone Configuration (Local Menu, WBM)

Generally, it is recommended to administrate and configure an OpenStage phone installation remotely using a provisioning service via the phone's WPI (WorkPoint Interface). However, there are two further configuration interfaces; these can be used to administrate individual phones:

- Local menu: The user interface of the device itself.
- Web Based Management (WBM): The phone's web interface.

Note: The default password for OpenStage administration is <123456>.

Local Menu

The phone's application key () is used to access the user and administration menu at the phone.

Web Based Management (WBM)

The phone's web interface can be accessed by any common web browser, like Firefox or Internet Explorer. As HTTPS is used, the URL must be entered as follows:

https://<phone-ip-address>

If the browser displays a certificate notification, accept it.

Alternatively, the DNS name of the phone can be entered, if it has been configured and DNS is available in the network.

Using OpenStage@Asterisk

This chapter contains some tips and tricks for operating OpenStage phones with Asterisk. It is not intended to be exhaustive, but provides some major topics and solutions from different successful customer projects.

Busy Lamp Function (BLF)

The "Busy Lamp Field" feature (available for OpenStage 15/40/60/80) allows users to monitor the dialog state of another phone via the LED associated to an FPK. Please note that, sometimes, the term 'Direct Station Selection' is used for the same functionality.
Depending on which function is configured for the FPK, the user can also pick up a call for another user, which is of much use in working teams. The feature is described in detail here:
http://wiki.siemens-enterprise.com/index.php/
Asterisk_Feature_Busy_Lamp_Field_%28BLF%29#For_Users

XML Applications

OpenStage 60/80 phones feature a graphical user interface and an XML application platform, which allows for developing custom applications based on HTTP/HTTPS. The phone acts as a front-end for a server-side program. The interaction can be initiated by the phone or by the server-side program. Besides displaying, modifying, and submitting data, XML applications have the capability of starting calls on the phone. Possible uses for OpenStage XML Applications might be

- Integration with groupware (e.g. Microsoft Exchange Server) or Unified Messaging systems (e.g. Siemens Enterprise Communications OpenScape)
- phonebooks with access to address databases
- call recording
- presence applications
- collecting information provided by web services (e.g. news, weather, traffic, stocks)
- attendance clock

For detailed information, please refer to:
http://www.siemens-
enterprise.com/developerportal/Resource%20Center/OpenStage%20XML.aspx

Send URL / Remote Server Control

The FPKs and FFKs (firmware V2R1) can be utilized for communication with a server-side program.
To enable feedback from the server, the LED associated with the key can be controlled remotely. This can be done via SIP notify messages, or, with firmware version V2R2 onwards, via a combination of HTTP push requests and XML documents. For detailed information, please see the

OpenStage XML Applications Developer's Guide [9],

and

http://www.siemens-
enterprise.com/developerportal/Resource%20Center/OpenStage%20XML.aspx

Call Completion (CCBS/CCNR)

Call completion is a telephony feature which takes action on a failure to complete a call. It allows for notifying the calling user when the called user is available again.

The OpenStage callback feature covers two conditions for call completion:

CCBS (Call Completion Busy Subscriber) : The called party is busy.
CCNR (Call Completion No Reply) : The called party does not respond.

Call Completion features can be implemented on a PBX, a dedicated server (e.g. a voicemail Server) or directly on the client device (e.g. messaging applications).

There are several commercial companies which provide call completion features, as well as IETF documents specifying call completion features for open standards, such as SIP.

The RFC 5359 gives a best practice example for call completion. However, the SEN OSCAR group has evaluated this RFC with the result that it is not useful for a B2BUA architecture.

The IETF BLISS working group currently provides a draft paper (http://www.ietf.org/id/draft-ietf-bliss-call-completion-06.txt) on how call completion can be implemented. However, no RFC is available for this topic yet.

Although no standard has been released until now, OpenStage phones already support server based call completion.
An implementation according to the IETF BLISS draft is planned, but currently not available.

The OpenStage call completion implementation is purely stimulus based and can be found at:

http://wiki.siemens-enterprise.com/images/6/65/White_Paper_CC_10090.pdf

CTI for OpenStage - UACSTA

There are several use cases where remote control of a VoIP phone is required. Among these are server based features like 'Call Forwarding' or 'Do not Disturb', or agent desktop applications requiring a seamless desktop integration of the phone.

There is one ECMA standard in place which covers all those requirements: uaCSTA.

By means of the uaCSTA interface, the OpenStage SIP user agent can use call and device control services at the SIP Server and vice versa. A complete set of CSTA services are defined in ECMA-269 [6], which should be referenced for additional information.

The following subset of CSTA services and events are supported by OpenStage:

➔ **Services on the SIP Server:**
- Set Forwarding
- Get Forwarding
- Set Do Not Disturb
- Get Do not Disturb

➔ **Events Generated by the SIP Server:**
- Forwarding Event

- Do Not Disturb Event
- Diverted Event

➔ **Services on the OpenStage device:**

- Make Call
- Answer Call
- Hold Call
- Retrieve Call
- Clear Connection
- Consultation Call
- Generate Digits
- Get Volume
- Set Volume
- Get Mute
- Set Mute

➔ **Events Generated by OpenStage:**

- OpenStage does not generate CSTA Events.

With these services a SIP server can easily control basic OpenStage functions. For futher information please have a look at:

http://wiki.siemens-enterprise.com/images/e/e7/white_paper_uaCSTA_Public_version_2010803.pdf

Changing the Caller Information – PAI Header

SIP is a great protocol for call processing. However, in some use cases, additional and up-to-date information about a caller might prove to be very useful. Among the possibilities are:

- Add caller information from an external database
- Update caller information during call transfer
- Add hunt group information for incoming calls
- Enhance Executive/Assistant features with additional information

OpenStage supports RFC 3325 [7]. A SIP server can change the OpenStage display information using SIP INVITE or UPDATE requests or any SIP response code.
Especially the P-Asserted-Identity Header can be used to carry additional information to the phone user.

Multi Address Appearance (MAA)

A telephone is normally associated with a directory number, or, generally, with a SIP AoR. This number is used for placing calls to the associated telephone and for displaying the telephone's (user's) identity when placing calls to another party. The number is also used when more than one call appearance is supported due to additional features like call waiting.

The term ‘keyset’ denotes a telephone which is associated with more than one number. This allows a given telephone to act on behalf of other phone numbers, resp. users. Just like with traditional telephony systems, people sometimes refer to lines instead of numbers, hence keyset phones are also referred to as ‘multiline phones’. The main line, i.e. the line/directory number associated with a given physical telephone, is called primary line. All other lines, which can also be handled on other phones, are denoted as secondary lines. Please note that call log and MWI (Message Waiting Indication) are supported only for the primary line, not for secondary lines.

The programmable feature keys are used for handling the lines and their respective call appearances, supported by the associated LEDs reflecting the line/call status. The number of lines that can be configured is depending on the phone model. For OpenStage 60/80, up to 30 lines can be configured. OpenStage 15/20/40 are limited to 18 lines.

This feature can be used for different use cases, for example:

- Address multiple users at one phone
- Enhanced call hold scenarios
- Allow more than two incoming calls at one phone

For some use cases, however, this feature can not be used, for example:

- MLA. If the line is configured at more than one phone, incoming calls are sent to the last registered device
- Line status observation. If the same line is configured at more than one phone, the line status is not presented at these phones.

MAA is automatically activated if line keys are configured at the phone. Line key administration is done by the administrator; the user has no influence on these settings. The phone operates as an MAA phone ‘out of the box’. Depending on the administrator settings, the phone will react slightly different in basic user interactions.
Even if only one line key is configured, the phone changes into line presentation mode. The line presentation mode helps the user to keep track of the different line statuses.

Example: OpenStage operates in MAA mode.

Further information can be found at:

http://wiki.siemens-enterprise.com/images/a/a3/White_Paper_MAA.pdf

Automatic Call Answering Using Alert-Info Header

Besides using uaCSTA, the phone can be set to automatically answer a call by adding an alert info header to the call. Thus, the SIP server is enabled to control whether a call is to be answered immediately and without user interaction. However, the user has the possibility to allow or disallow this feature by setting **User Pages** > **Configuration** > **Incoming calls** > **CTI calls** > **Auto answer**.

The Alert-Info header must be set as follows:

```
Alert-Info: http://www.example.com;info=alert-autoanswer
```

Logging and Tracing

OpenStage phones are perfect. But if something should go wrong anyhow, the customer service needs tools to focus on the problem. Service effort is needed, but should be minimized. Therefore, OpenStage phones provide plenty of tools and options to find the cause of a problem quickly, even if it is not located at the phone.

LAN Port Mirroring

Every OpenStage phone has a built-in Ethernet switch. One of the ports is used to connect the phone to the local network. The other port is intended for connecting a PC, thus allowing network connectivity for both devices with only one wire from the desk.
In addition, the PC port enables network monitoring which might be useful for development and error tracing. For this purpose, the PC port must be configured as a mirror for the LAN port by setting **PC port mode** to "mirror" (see [3]). If configured this way, the complete traffic of the LAN port will be passed through to the PC port, just like with a simple network hub. Now, a network tracing tool on the PC can trace all IP traffic, like SIP over UDP, or XML over HTTP, for instance.

Tracing Capabilities within the Phone

OpenStage phones provide strong support for system integration, testing, and troubleshooting. Besides the Administration Guide [3], the tracing capabilities of the phone are described in [5].

Basic Troubleshooting

For tracking network issues, the phone can execute ping and traceroute tests; these can be controlled and viewed online using the WBM.

For elementary troubleshooting, the phone provides an overview about basic issues in the user menu. The admin can ask the user to read that basic information to get a first hint about the possible causes of an issue:

Problem Description	Error code
Network Problem No network connection	LI1
Not Initialised Waiting for data	I1
Unable to use LAN 802.1x error	LX1
Unable to use LAN Physical connection missing	LP1
Unable to Register Server timeout	RT2
Unable to Register Server failed	RF2
Unable to Register Authentication failed	RA2
Unable to Register No number configured	RN2
Unable to Register No server configured	RS2
Unable to Register No registrar configured	RG2
Unable to Register No DNS domain configured	RD2
Unable to Register Rejected by server	RR2
Unable to Register No phone IP address set	RI2
Survivability Backup route active	B8
Survivability Backup not configured	RS8
Survivability Backup timeout	RT8
Survivability Backup authentication failed	RA8

Local and Remote Tracing

The phone is able to write internal trace files, and to send the trace data to a remote syslog server. The tracing can be configured in a differentiated way by setting discrete trace levels for each service.
Please note that it is not recommended to enable all traces to the deepest level. The generated trace file will exhaust the phone's memory shortly, and the overall functionality will slow down.

QoS Data Collection

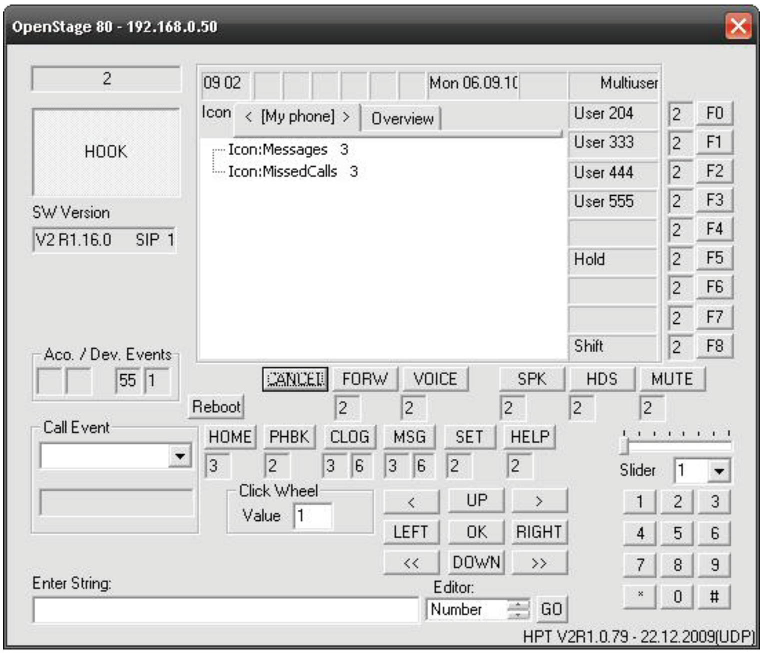
OpenStage phones generate QoS reports using a HiPath specific format, QDC (QoS Data Collection).
The reports created for the last 6 sessions, i. e. conversations, can be viewed on the WBM or are reported to the QCU (QoS data Collection Unit).
SEN provides a server application to collect the data. The collected data is sent via SNMP. If an SNMP server is available, the QDC MIBS can be downloaded from our software supply server (SWS).
Meanwhile, third party solutions are available which can also deal with the OpenStage QDC data.

Remote Control - the HUSIM Phone Tester

Sometimes everything goes wrong: tracing is of no help, issues are sporadic and the customer's problem cannot be understood. It seems that the last resort is a visit to the customer to get the needed information.

In such situations, the HUSIM phone simulator can produce relief. It enables the service staff to access a defined group of phones remotely. The tool works similar to well known remote control tools for PCs like VNC.
For each phone, a PC application window shows the current status. Every OpenStage phone model is represented with its complete key layout and display content. The remote visitor can see all user interactions on the phone. Moreover, he can access the phone keys actively and in this way operate the phone by remote control. Please note that, for privacy protection, the user is always informed about the remote interaction.

To get the phone tester up and running, a special dongle key must be uploaded to the phone. The dongle key and the HUSIM software can be downloaded without additional charge from SWS/SEBA. The key can be distributed to the phone using the SEN DLS (Deployment Service) or the phone's WPI (WorkPoint Interface).



Example Screen: OpenStage 80 represented in the HUSIM Phone Tester

Limitations

Not known yet :-)

References

[1] **TIA-811-A**: Performance and Interoperability Requirements for Voice-over-IP (VoIP) Feature Telephones (http://www.tiaonline.org/standards/technology/voip/documents/TIA-811-A-final-for-global.pdf)

[2] **Session Initiation Protocol (SIP)-Specific Event Notification**: (RFC 3261)

[3] **OpenStage Asterisk Admin Guide**: (http://wiki.siemens-enterprise.com/images/e/e1/Administration_Manual_OpenStage_Asterisk.pdf)

[4] **WPI Guide**: (http://wiki.siemens-enterprise.com/images/c/c7/OpenStage_Provisioning_Interface_Developer%27s_Guide.pdf)

[5] **Trace Guide Openstage SIP**: http://wiki.siemens-enterprise.com/images/1/1b/Service_Info_How_to_trace_OST_SIP.pdf

[6] **Services for Computer Supported Telecommunications Applications** (ECMA-269). http://www.ecma-international.org/publications/standards/Ecma-269.htm

[7] **Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks**: http://www.ietf.org/rfc/rfc3325.txt

[8] **Provisioning Interface Developer's Guide:** http://wiki.siemens-enterprise.com/images/c/c7/OpenStage_Provisioning_Interface_Developer%27s_Guide.pdf

[9] **OpenStage XML Applications Developer's Guide**: https://app-enterprise.siemens-enterprise.com/gdmsproxy/retrieve?id=40776497

Abbreviations

AoR	Address of Record
BLA	Bridged Line Appearance
CCBS	Call Completion Busy Subscriber
CCNR	Call Completion No Reply
CTi	Computer Telephony Integration
FFK	Free Function Key
FPK	Free Programmable Key
HTTP	Hypertext Transfer Protocol Overview
HTTPS	HyperText Transfer Protocol Secure
MAA	Multiple Address Appearance
MLA	Multiple Line Appearance
MSA	Multiple Station Appearance
MWI	Message Waiting Indication
SCA	Shared Call Appearance
SIP	Session Initiation Protocol
UA	User Agent
uaCSTA	User Agent Computer Supported Telephony Application

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