SIP 2.0 Administrator's Guide
SoundPoint®/SoundStation® IP SIP

Version 2.0.3B Addendum

Version 2.1 Addendum

January 2007

Notices

1. Specifications subject to change without notice.

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1 Addendum

This addendum addresses changes to the SoundPoint IP / SoundStation IP SIP 2.0 Administrator's Guide made by the release of the SoundPoint IP 650 phone.

The SoundPoint IP 650 phone behaves in a similar manner to the SoundPoint IP 601 (supports the SoundPoint IP Expansion Module) unless otherwise specified.

For more information, refer to the Release Notes for the SIP Application, Version 2.0.3 B.

Note

The various .hd. parameters in **sip.cfg** (such as voice.aec.hd.enable, voice.ns.hd.enable, and voice.agc.hd.enable) are headset parameters. There are not connected to high definition or HD voice.

1.1   Added or Changed Features

1.1.1   Configurable Feature Keys

The SoundPoint IP 650 phone's default SIP key layouts is the same as the SoundPoint IP 600 and 601. Refer to 3.1.7 Configurable Feature keys on page 29.

1.1.2   Handset, Headset, and Speakerphone

The SoundPoint IP 650 phones are full-duplex speakerphones.

Changes can be found in the following parameters in the **sip.cfg** configuration file:
- Gains

Attribute	Default
voice.handset.rxag.adjust.IP_650	1
voice.handset.txag.adjust.IP_650	9
voice.handset.sidetone.adjust.IP_650	-3

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Attribute	Default
voice.headset.rxag.adjust.IP_650	1
voice.headset.txag.adjust.IP_650	18
voice.headset.sidetone.adjust.IP_650	-3

Important

Polycom recommends that you do not change these values.

1.1.3 LCD Backlight

Backlight intensity on the SoundPoint IP 650 phone has three modes:
- Backlight On
- Backlight Idle
- Dim

You can modify the Backlight On intensity and the Backlight Idle intensity separately. You can select high, medium, low, and off levels for both. Dim mode intensity is determined by the Backlight On intensity and the Backlight Idle intensity together.

Backlight settings can be found in the User Preferences <up/> parameter in the **sip.cfg** configuration file.

Attribute	Permitted Values	Default	Interpretation
up.backlight.onIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	3	This parameter controls the intensity of the LCD backlight when it turns on during normal use of the phone.
up.backlight.idleIntensity	0 (off), 1 (low), 2 (medium), 3 (high)	1	This parameter controls the intensity of the LCD backlight when the phone is idle. *Note: If idleIntensity is set higher than onIntensity, it will be replaced with the onIntensity value.*

1.1.4 Expanded Memory and Expanded Flash Memory

Changes can be found in the following parameters in the **sip.cfg** configuration file:
- Directory

Attribute	Permitted Values	Default	Interpretation
dir.local.volatile.maxSize	1 to 100	100	Maximum size in Kbytes of volatile storage that the directory will be permitted to consume.
dir.local.volatile.8meg	0, 1	0	Attribute applies only to platforms with 8 Mbytes of flash memory. If set to 1, use volatile storage for phone-resident copy of the directory to allow for larger size.
dir.local.nonVolatile.max-Size.8meg	1 to 100	100	Attribute applies only to platforms with 8 Mbytes of flash memory. This is the maximum size of non-volatile storage that the directory will be permitted to consume.

- Provisioning

Attribute	Permitted Values	Default	Interpretation
prov.fileSystem.rfs0.minFreeSpace	5-512	5	Minimum free space in Kbytes to reserve in the file system when downloading files from the boot server. *Note: Polycom recommends that you do not change these parameters.*
prov.fileSystem.ffs0.8meg.min-FreeSpace		512	

- RAM Disk

Attribute	Permitted Values	Default	Interpretation
ramdisk.bytesPerBlock	0, 32, 33, ..., 1024	0	These three parameters use internal defaults when value is set to 0.

- Finder

Attribute	Permitted Values	Default	Interpretation
res.finder.sizeLimit	positive integer	300	If a resource that is being downloaded to the phone is larger than this value * 1000 bytes (= the maximum size), the resource will be automatically truncated to the maximum size defined.
res.finder.minfree	1 to 2048	1200	A resource will not be downloaded to the phone if the amount of free memory is less than this value * 1000 bytes (= the minimum size). This parameter is used for 16MB SDRAM platforms and scaled up for platforms with more SDRAM.

1.1.5 **MicroBrowser**

The SoundPoint IP 650 phones support an XHTML microbrowser. This can be launched by pressing the Services key.

MicroBrowser parameter changes in the **sip.cfg** configuration file are as follows:

Attribute	Permitted Values	Default	Interpretation
mb.limits.nodes	positive integer	256	Limits the number of tags which the XML parser will handle. This limits the amount of memory used by complicated pages. A maximum total of 500 (256 each) is recommended. This value is used as referent values for 16MB of SDRAM. *Note: Increasing this value may have a detrimental effect on performance of the phone.*
mb.limits.cache	positive integer	200	Limits the total size of objects downloaded for each page (both XHTML and images). Once this limit is reached, no more images are downloaded until the next page is requested. Units = kBytes. This value is used as referent values for 16MB of SDRAM. *Note: Increasing this value may have a detrimental effect on performance of the phone.*

1.1.6 G.722 Audio Codec

The SoundPoint IP 650 supports the G.722 audio codec.

Changes can be found in the following parameters in the **sip.cfg** configuration file:
- Codec Preferences

Attribute	Permitted Values	Default	Interpretation
voice.codecPref.IP_650.G711Mu	Null, 1-3	2	Specifies the codec preferences for the SoundPoint IP 650 platform. 1 = highest 3 = lowest Null = do not use
voice.codecPref.IP_650.G711A		3	
voice.codecPref.IP_650.G729AB		4	
voice.codecPref.IP_650.G722		1	

- Audio Profiles

Attribute	Permitted Values	Default	Interpretation
voice.audioProfile.G722.pay-loadSize	10, 20, 30, ... 80	20	Preferred Tx payload size in milliseconds to be provided in SDP offers and used in the absence of ptime negotiations. This is also the range of supported Rx payload sizes.
voice.audioProfile.G722.jitter-BufferMin	20, 40, 50, 60, ... (multiple of 10)	40	The smallest jitter buffer depth (in milliseconds) that must be achieved before play out begins for the first time. Once this depth has been achieved initially, the depth may fall below this point and play out will still continue. This parameter should be set to the smallest possible value which is at least two packet payloads, and larger than the expected short term average jitter.

Attribute	Permitted Values	Default	Interpretation
voice.audioProfile.G722.jitter-BufferShrink	10, 20, 30, ... (multiple of 10)	500	The absolute minimum duration time (in milliseconds) of RTP packet Rx with no packet loss between jitter buffer size shrinks. Use smaller values (1000 ms) to minimize the delay on known good networks. Use larger values to minimize packet loss on networks with large jitter (3000 ms).
voice.audioProfile.G722.jitter-BufferMax	> jitter-Buffer-Min, multiple of 10, <=500 for IP 430, 500, 501, and 600, <= 160 for IP 300 and 301	160	The largest jitter buffer depth to be supported (in milliseconds). Jitter above this size will always cause lost packets. This parameter should be set to the smallest possible value that will support the expected network jitter.

• Gains

Attribute	Default
voice.gain.rx.analog.chassis.IP_650	2
voice.gain.rx.analog.ringer.IP_650	0
voice.gain.rx.digital.chassis.IP_650	-9
voice.gain.rx.digital.ringer.IP_650	-21
voice.gain.tx.analog.chassis.IP_650	36
voice.gain.tx.digital.chassis.IP_650	0

Important

Polycom recommends that you do not change these values.

- Receive

Attribute	Default
voice.rxEq.hs.IP_650.preFilter.enable	1
voice.rxEq.hs.IP_650.postFilter.enable	0
voice.rxEq.hd.IP_650.preFilter.enable	1
voice.rxEq.hd.IP_650.postFilter.enable	0
voice.rxEq.hf.IP_650.preFilter.enable	1
voice.rxEq.hf.IP_650.postFilter.enable	0

Important

Polycom recommends that you do not change these values.

- Transmit <txEq/>.

Attribute	Default
voice.txEq.hs.IP_650.preFilter.enable	1
voice.txEq.hs.IP_650.postFilter.enable	1
voice.txEq.hd.IP_650.preFilter.enable	1
voice.txEq.hd.IP_650.postFilter.enable	0
voice.txEq.hf.IP_650.preFilter.enable	1
voice.txEq.hf.IP_650.postFilter.enable	1

Important

Polycom recommends that you do not change these values.

1.1.7 USB Diagnostics

The SoundPoint IP 650 phone has a USB port, which will be supported by future releases of the SIP application.

USB port parameters can be found in the USB <usb/> parameter in the **sip.cfg** configuration file.

Attribute	Permitted Values	Default	Interpretation
usb.enable	0, 1	0	This parameter enables or disables the USB port on the phone.
usb.bulkDrive.enable	0, 1	0	This parameter enables or disables support for a USB bulk drive connected to the USB port on the phone.
usb.bulkDrive.name	alphanumeric string	usb-Drive	This parameter is a string which specifies the name of the mounted USB drive.

Other changes to support a USB port can be found in the following parameter in the **sip.cfg** configuration file:
- Basic Logging

Attribute	Permitted Values	Default	Interpretation
log.level.change.usb	0-5	4	Control the logging detail level for the usb component. These are the input filters into the internal memory-based log system.

2 Addendum

This addendum addresses changes to the SoundPoint IP / SoundStation IP SIP 2.0 Administrator's Guide made by the release of the SIP 2.1 application.

For more information, refer to the Release Notes for the SIP Application, Version 2.1 .

Note
The various .hd. parameters in **sip.cfg** (such as voice.aec.hd.enable, voice.ns.hd.enable, and voice.agc.hd.enable) are headset parameters. They are not connected to high definition or HD voice.

2.1 Added or Changed Features

2.1.1 Digit Map

Enhancements have been made to the local digit maps that can eliminate the need for using the **Dial** or **Send** soft key when making outgoing calls. Refer to the "Technical Bulletin 11572: Changes to Local Digit Maps on SoundPoint® IP Phones " at www.polycom.com/support/voip/ .

2.1.2 Billing Code

Billing codes let administrators assign specific codes to all of their organization's outgoing calls. The prompt to signal employees to enter their billing codes has changed. Refer to the "Technical Bulletin 9268: Billing Code Entry on SoundPoint® IP phones with Sylantro " at www.polycom.com/support/voip/ .

2.1.3 Syslog

Syslog is a de facto standard for forwarding log messages in an IP network. The SIP application has been enhanced to support logging system level messages and error conditions with communications networks to a centralized location. Refer to the "Technical Bulletin 17124: Syslog on SoundPoint® IP Phones" at www.polycom.com/support/voip/ .

2.1.4 **Server Redundancy**

Server redundancy enhancements provides backup to other SIP server(s) by providing basic registration and redirection services. Refer to the "Technical Bulletin 5844: SIP Server Fallback Enhancements on SoundPoint® IP Phones" at www.polycom.com/support/voip/ .

2.1.5 **MicroBrowser**

An XHTML microBrowser is now supported on the SoundPoint IP 430 and 501 phones. The tables shows the platforms where the XHTML microBrowser is supported and where it is not.

:

Supported Platforms	Unsupported Platforms
IP 430	IP 300, 301
IP 501	IP 500
IP 600, 601, 650	IP 4000

This can be launched by pressing the **Services** key, or through the **Menu** key by selecting *Features*, and then *Services*, if there is no **Services** key on the phone.

The microBrowser auto-navigates to the first visible, selectable item on the web page (a hyperlink, for example):
- after initial page load
- after scrolling further down page (after the second or third down arrow key press)

Note
XHTML tables must be properly formatted (should include <tbody> and </tbody> tags). Improperly formatted tables could cause the phone to reboot.

2.1.6 Disable Message Waiting Indicator by Registration

The SIP application has been enhanced to allow the message waiting indicator to be disabled by registration.

- Changes can be found in the following parameters in the **phone1.cfg** configuration file:

Attribute	Permitted Values	Default	Interpretation
msg.mwi.x.callBackMode	contact or registration or disabled	"registration"	Disables message retrieval and disables waiting message notification for the line. If set to "contact", a call will be placed to the contact specified in the callback attribute when the user invokes message retrieval. If set to "registration", a call will be placed using this registration to the contact registered (the phone will call itself). If set to "disabled", message retrieval is disabled.

2.1.7 Daylight Saving Time Changes for 2007

Daylight saving time dates will be changing in North America in 2007. Refer to the "Technical Bulletin 17803: Daylight Savings Time Changes for 2007 on SoundPoint® IP Phones" at www.polycom.com/support/voip/ .

Changes to default values and the Interpretations can be found in the following parameters in the **sip.cfg** configuration file:

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.daylightSavings.fixedDay-Enable	0, 1	0	If set to 0, month, date, and dayOfWeek are used in DST start date calculation. If set to 1, then only month and date are used.
tcpIpApp.sntp.daylightSavings.start.month	1-12	3 (March)	Month to start DST. Mapping: 1=Jan, 2=Feb, ..., 12=Dec
tcpIpApp.sntp.daylightSavings.start.date	1-31	8	Day of the month to start DST. Mapping (on or after): 1 = the first occurrence of a given day-of-the-week in a month, 8 = the second occurrence of a given day-of-the-week in a month, 15 = the third occurrence of a given day-of-the-week in a month, 22 = the fourth occurrence of a given day-of-the-week in a month
tcpIpApp.sntp.daylightSavings.start.time	0-23	2	Time of day to start DST, in 24 hour clock. Mapping: 2=2 am, 14=2 pm
tcpIpApp.sntp.daylightSavings.start.dayOfWeek	1-7	1	Day of week to apply DST. Mapping: 1=Sun, 2=Mon, ..., 7=Sat
tcpIpApp.sntp.daylightSavings.start.dayOfWeek.lastInMonth	0, 1	0	If set to 1 and fixedDay-Enable is set to 0, DST starts on the last day (specified by start.dayOfWeek) of the week in the month. The start.date is ignored.
tcpIpApp.sntp.daylightSavings.stop.month	1-12	11	Month to stop DST.

Attribute	Permitted Values	Default	Interpretation
tcpIpApp.sntp.daylightSavings.stop.date	1-31	1	Day of the month to stop DST.
tcpIpApp.sntp.daylightSavings.stop.time	0-23	2	Time of day to stop DST in 24 hour clock.
tcpIpApp.sntp.daylightSavings.stop.dayOf-Week	1-7	1	Day of week to stop DST.
tcpIpApp.sntp.daylightSavings.stop.dayOf-Week.lastInMonth	0, 1	0	If set to 1 and fixedDay-Enable=0, stop DST on the last day of the week (specified by stop.day-OfWeek) in the month. The stop.date is ignored.

2.1.8 Configurable Feature Keys

It has been determined that only some feature keys can be disabled. The exact feature keys that can be "null"-ified are platform-dependent.

:

Platform	Key IDs
IP 300, 301	5, 7, 16, 23, 29, 31, 32
IP 430	7, 8, 9, 10, 29, 31, 33, 34
IP 500, 501	7, 8, 9, 10, 29, 30, 31, 32, 36, 37
IP 600, 601, 650	7, 8, 9, 30, 32, 36, 37, 40
IP 4000	1, 2, 5, 7, 16, 29

2.1.9 Miscellaneous Configuration File Changes

2.1.9.1 sip.cfg

The following changes have also occurred in the **sip.cfg** configuration file:

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.useSendonlyHold	0, 1	1	If set to 1, the phone will send a reinvite with a stream mode attribute of "sendonly" when a call is put on hold. This is the same as the previous behavior. If set to 0, the phone will send a reinvite with a stream mode attribute of "inactive" when a call is put on hold. **NOTE:** The phone will ignore the value of this parameter if set to 1 when the parameter voIpProt.SIP.useRFC2543hold is also set to 1 (default is 0).

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Attribute	Permitted Values	Default	Interpretation
voIpProt.server.x.transport	DNSnaptr or TCPpre-ferred or UDPOnly or TLS or TCPOnly	DNSnaptr	If set to Null or DNSnaptr: If voIp-Prot.server.x.address is a hostname and voIp-Prot.server.x.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and servers, as per RFC 3263. If voIp-Prot.server.x.address is an IP address, or a port is given, then UDP is used. If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails. If set to UDPOnly: Only UDP will be used. If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. If set to TCPOnly: Only TCP will be used. ***NOTE: TLS is not sup-ported on SoundPoint IP 300 and 500 phones.***

Attribute	Permitted Values	Default	Interpretation
voIpProt.SIP.outboundProxy.transport	DNSnaptr or TCPpreferred or UDPOnly or TLS or TCPOnly	DNSnaptr	If set to Null or DNSnaptr: If voIpProt.SIP.outbound-Proxy.address is a host-name and voIpProt.SIP.outbound-Proxy.port is 0 or Null, do NAPTR then SRV look-ups to try to discover the transport, ports and serv-ers, as per RFC 3263. If voIpProt.SIP.outbound-Proxy.address is an IP address, or a port is given, then UDP is used. If set to TCPpreferred: TCP is the preferred transport, UDP is used if TCP fails. If set to UDPOnly: Only UDP will be used. If set to TLS: If TLS fails, transport fails. Leave port field empty (will default to 5061) or set to 5061. If set to TCPOnly: Only TCP will be used. *NOTE: TLS is not sup-ported on SoundPoint IP 300 and 500 phones.*
voice.gain.rx.analog.chassis.IP_650		0	Gain setting.
voice.handset.sidetone.adjust.IP_430		-13	Handset sidetone.
call.enableOnNotRegistered	0,1	1	If set to 1, calls will be allowed when the phone is not successfully regis-tered. If set to 0, calls will not be permitted without a valid registration. If a user picks up handset, presses the **New Call** soft key, or presses the speaker phone, speed dial or the line keys to get a dial tone, "Service unavailable" is displayed.

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Attribute	Permitted Values	Default	Interpretation
call.stickyAutoLineSeize.onHookDialing	Null, 0, 1	Null	If call.stickyAutoLineSeize is set to 1, this parameter has no effect. The regular stickyAutoLineSeize behavior is followed. If call.stickyAutoLineSeize is set to 0 or Null and this parameter is set to 1, this overrides the stickyAuto-LineSeize behavior for hot dial only. (Any **New Call** scenario seizes the next available line.) If call.stickyAutoLineSeize is set to 0 or Null and this parameter is set to 0 or Null, there is no difference between hot dial and **New Call** scenarios.

2.1.9.2 phone1.cfg

The following changes has also occurred in the **phone1.cfg** configuration file:

Attribute	Permitted Values	Default	Interpretation
reg.x.server.y.transport	DNSnaptr or TCPpre-ferred or UDPOnly or TLS or TCPOnly	DNSnaptr	Refer to Interpretation of voIp-Prot.server.x.transport in 2.1.9.1 sip.cfg on page 6, the previous section. If specified, this attribute may override the value in **sip.cfg**.
reg.x.outboundProxy.transport	DNSnaptr or TCPpre-ferred or UDPOnly or TLS or TCPOnly	DNSnaptr	Refer to Interpretation of voIpProt.SIP.out-boundProxy.transport in 2.1.9.1 sip.cfg on page 6, the previous section. If specified, this attribute may override the value in **sip.cfg**.

2.1.9.3 device Parameter

The following changes has also occurred in the **device** parameter:

Attribute	Permitted Values	Default	Interpretation
device.prov.redunAttemptLimit	10, Null	10	Refer to the File Transmit Tries parameter in 2.2.1.3.3 Server Menu on page 11 of the SIP 2.0 Administrator's Guide.
device.prov.redunInterAttemptDelay	300, Null	300	Refer to the Retry Wait parameter in 2.2.1.3.3 Server Menu on page 11 of the SIP 2.0 Administrator's Guide.
device.em.power	Enabled, Disabled	Null	Refer to the EM Power parameter in 2.2.1.3.1 Main Menu on page 8 of the SIP 2.0 Administrator's Guide.

Technical Bulletin 11572
Changes to Local Digit Maps on SoundPoint® IP Phones

This technical bulletin provides detailed information on how to modify the configuration files to automate the setup phase of number-only calls.

This information applies to SoundPoint IP phones running SIP application version 2.1 or later.

Introduction

Enhancements have been made to this feature that can eliminate the need for using the **Dial** or **Send** soft key when making outgoing calls. For example, it can match the behavior of removing the 9 or 0 from a string of dialed digits or adding the area code before dialed digits when a switch to 10 digit phone numbers occurs.

As soon as a digit pattern matching the digit map is found, the call setup process will complete automatically. The configuration syntax is the same as that specified in 2.1.5 of RFC 3435. The phone's behavior when the user dials digits that do not match the digit map is configurable. It is also possible to strip a trailing '#' from the digits sent, prepend a '+' to digits, or to replace certain matched digits with the introduction of 'R' to the digit map.

Configuration File Changes

If a single dial plan is used for the entire company, the dial plan is best specified in the application configuration file (**sip.cfg**). You can also create multiple dial plans and specify which phones are to use which in the phone-specific configuration file (**phone1.cfg**).

Configuration changes can performed centrally at the boot server or locally:

Central (boot server)	Configuration file: **sip.cfg**	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value. • For more information, refer to Dial Plan in Application Configuration File on page 2.
	Configuration file: **phone1.cfg**	Specify per-registration impossible match behavior, trailing # behavior, digit map matching strings, and time out values that override those in **sip.cfg**. • For more information, refer to Dial Plan in Per-Phone Configuration File on page 4.

Local	Web Server (if enabled)	Specify impossible match behavior, trailing # behavior, digit map matching strings, and time out value.
		Navigate to: http://<phoneIPAddress>/appConf.htm#ls
		Changes are saved to local flash and backed up to ***<Ethernet address>*-phone.cfg** on the boot server. Changes will permanently override global settings unless deleted through the Reset Local Config menu selection.

Dial Plan in Application Configuration File

The `<dialplan/>` attribute is described below and also includes:

- Digit Map `<digitmap/>` on page 3.

Note	The dial plan is not applied against Placed Call List, VoiceMail, last call return, and remote control dialed numbers.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.applyToCallListDial	0, 1	0	This attribute covers dialing from Received Call List and Missed Call List including dialing from Edit or Info sub-menus. If set to 0, the dial plan is not applied against the dialed number. if set to 1, the dial plan is applied against the dialed number.
dialplan.applyToDirectoryDial	0, 1	0	This attribute covers dialing from Directory as well as Speed Dial List. Value interpretation is the same as for `dialplan.applyToCallListDial`. ***Note:** An Auto Call Contact number is considered a dial from directory.*
dialplan.applyToUserDial	0, 1	1	This attribute covers the case when the user presses the **Dial** soft key to send dialed number when in idle state display. Value interpretation is the same as for `dialplan.applyToCallListDial`.

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Attribute	Permitted Values	Default	Interpretation
dialplan.applyToUserSend	0, 1	1	This attribute covers the case when the user presses the **Send** soft key to send the dialed number. Value interpretation is the same as for `dialplan.applyToCallListDial.`
dialplan.impossibleMatchHandling	0, 1 or 2	0	If set to 0, the digits entered up to and including the point where an impossible match occurred are sent to the server immediately. If set to 1, give reorder tone. If set to 2, allow user to accumulate digits and dispatch call manually with the Send soft key.
dialplan.removeEndOfDial	0, 1	1	If set to 1, strip trailing # digit from digits sent out.

Digit Map

A digit map is defined either by a "string" or by a list of strings. Each string in the list is an alternative numbering scheme, specified either as a set of digits or timers, or as an expression over which the gateway will attempt to find a shortest possible match.

Digit map extension letter "R" indicates that certain matched strings are replaced. The following examples shows the semantics of the syntax:

- `R9RRxxxxxxx` — remove 9 at the beginning of the dialed number

    - For example, if a customer dials 914539400, the first 9 is removed when the call is placed.

- `RR604Rxxxxxxx` — prepend 604 to all 7 digit numbers

    - For example, if a customer dials 4539400, 604 is added to the front of the number, so a call 6044539400 is placed.

- `R9R604Rxxxxxxx` — replaces 9 with 604

- `R999R911R` — convert 999 to 911

- `xxR601R600Rxx` — when applied on 1160122 gives 1160022

- `xR60xR600Rxxxxxxx` — any 60x will be replaced with 600 in the middle of the dialed number that matches

    - For example, if a customer dials 16092345678, a call is placed to 16002345678.

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The following guidelines should be noted:

- You must use only *, #, or 0-9 between second and third R

- If a digit map does not comply, it is not included in the digit plan as a valid one. That is, no matching is done against it.

- There is no limitation on the number of R triplet sets in a digit map. However, a digitmap that contains less than full number of triplet sets (for example, a total of 2Rs or 5Rs) is considered an invalid digit map.

- Using T in the left part of RRR syntax is not recommended. For example, R0TR322R should be avoided.

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.digitmap	string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435. String is limited to 768 characters and 30 segments; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on;'+' is allowed as a valid digit; extension letter 'R' is used as defined above.	[2-9]11|0T| +011xxx.T| 0[2-9]xxxxxxxxx| +1[2-9]xxxxxxxx| [2-9]xxxxxxxxx| [2-9]xxxT	When this attribute is present, number-only dialing during the setup phase of new calls will be compared against the patterns therein and if a match is found, the call will be initiated automatically eliminating the need to press Send. Attributes `dialplan.applyToCallListDial`, `dialplan.applyToDirectoryDial`, `dialplan.applyToUserDial`, and `dialplan.applyToUserSend` control the use of match and replace in the dialed number in the different scenarios. Refer to page 2.
dialplan.digitmap.timeOut	string of positive integers separated by '|'	3 | 3 | 3 | 3 | 3 | 3 | 3	Timeout in seconds for each segment of digitmap. *Note: If there are more digit maps than timeout values, the default value of 3 will be used. If there are more timeout values than digit maps, the extra timeout values are ignored.*

Dial Plan in Per-Phone Configuration File

Per-registration dial plan configuration is supported.

The `<dialplan/>` attribute is described below and also includes:

- Digit Map `<digitmap/>` on page 3.

In the following tables, *x* is the registration number. IP 300, 301, and 430: *x*=1-2; IP 500 and 501: *x*=1-3; IP 600: *x*=1-6; IP 601: *x*=1-12; IP 4000: *x*=1

Attribute	Permitted Values	Default	Interpretation
dialplan.x.applyToCallListDial	0, 1	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.
dialplan.x.applyToDirectoryDial	0, 1	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.
dialplan.x.applyToUserDial	0, 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.
dialplan.x.applyToUserSend	0, 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.impossibleMatchHandling	0, 1 or 2	0	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.
dialplan.x.removeEndOfDial	0, 1	1	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For interpretation, refer to Dial Plan in Application Configuration File on page 2.

Digit Map

The digit map syntax is the same as for the application configuration file (refer to Digit Map on page 3).

This configuration attribute is defined as follows:

Attribute	Permitted Values	Default	Interpretation
dialplan.x.digitmap	A string compatible with the digit map feature of MGCP described in 2.1.5 of RFC 3435; string is limited to 768 characaters and 30 segments; a comma is also allowed; when reached in the digit map, a comma will turn dial tone back on;'+' is allowed as a valid digit; extension letter 'R' is used as defined above.	Null	When present, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For more information, refer to Digit Map <digitmap/> on page 3.

Attribute	Permitted Values	Default	Interpretation
dialplan.x.digitmap.timeOut	string of positive integers separated by '|'	Null	When present, and if `dialplan.x.digitmap` is not Null, this attribute overrides the global dial plan defined in the **sip.cfg** configuration file. For more information, refer to Digit Map <digitmap/> on page 3.

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Technical Bulletin 9268

Billing Code Entry on SoundPoint® IP Phones with Sylantro

POLYCOM®

This technical bulletin provides detailed information on how the SIP application has been modified for billing code entry when managed by a Sylantro call server.

This information applies to SoundPoint IP phones running SIP application version 2.1 or later.

Introduction

Note   This feature is only supported on Sylantro call servers.

Billing codes let administrators assign specific codes to all of their organization's outgoing calls.

When a SoundPoint IP phone managed by a Sylantro call server is configured to require billing codes, calls are not connected until the a valid billing code is entered.

The modified user interface on a SoundPoint IP phone running SIP 2.1 is described in the following section, Billing Code Entry.

Billing Code Entry

This section describes the steps the user must perform to enter a billing code.

To enter a billing code when placing a call:

1. Do one of the following to a place a call:

   a With the handset on-hook, enter the long-distance number (including prefix).

   You may need to press the **Dial** soft key to indicate you are finished entering the number.

   b Pick up the handset and enter the long-distance number (including prefix).

You may need to press the **Send** soft key to indicate you are finished entering the number.

The cursor pauses after the last digit has been entered. The call is not placed at this time.

A secondary dial tone is played and the text "Enter more digits" appears on the display just above the soft keys.

2. Enter the billing code.

If the billing code is accepted, the call is placed at this time.

If the billing code is not accepted, you will hear a fast busy tone and the call is not placed.

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Technical Bulletin 17124

Syslog on SoundPoint® IP Phones

POLYCOM®

This technical bulletin provides detailed information on how the SIP application has been modified to support logging system level messages and error conditions with communications networks to a centralized location.

This information applies to SoundPoint IP phones running SIP application version 2.1 or later.

Introduction

Syslog is a de facto standard for forwarding log messages in an IP network. The term "syslog" is often used for both the actual syslog protocol, as well as the application or library sending syslog messages.

The syslog protocol is a very simplistic protocol: the syslog sender sends a small textual message (less than 1024 bytes) to the syslog receiver. The receiver is commonly called "syslogd", "syslog daemon" or "syslog server". Syslog messages can be sent through UDP or TCP. The data is sent in cleartext.

Syslog is supported by a wide variety of devices and receivers. Because of this, syslog can be used to integrate log data from many different types of systems into a central repository.

The syslog protocol is defined in RFC 3164. For more information on syslog, go to http://www.ietf.org/rfc.html .

The `log.render.level` maps to syslog severity as follows:

0 -> SeverityDebug (7)
1 -> SeverityDebug (7)
2 -> SeverityInformational (6)
3 -> SeverityInformational (6)
4 -> SeverityError (3)
5 -> SeverityCritical (2)
6 -> SeverityEmergency (0)
7 -> SeverityNotice (5)

For more information on `log.render.level`, refer to Basic Logging <level/><change/> and <render/> on page 138 of the *SIP 2.1 Administrator's Guide*.

Network configuration changes required to support this feature are described in the following section, Network Configuration Changes.

Network Configuration Changes

The Network Configuration menu on the SoundPoint IP phone running SIP 2.1 has been modified to include:

• Syslog Menu

To access the Syslog menu:

1. From the idle display on a SoundPoint IP phone, press the **Menu** key.

2. Using the **Down** Arrow key and the **Select** soft key, select *Settings > Advanced > Admin Settings > Network Configuration*.

   You must enter the administrative password to access this menu. The default value is "456".

3. Using the **Down** Arrow key and the **Select** soft key, scroll down to *Syslog Menu*.

Syslog Menu

The following syslog configuration parameters can be modified on the Syslog menu:

Name	Possible Values	Description
Server Address	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.
Server Type	None=0, UDP=1, TCP=2	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
Facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3165. The default value is 16, which maps to "local 0".
Render Level	1 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on `log.render.level` and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page 138 of the *SIP 2.0 Administrator's Guide*.
Prepend MAC Address	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.

Flash Parameter Configuration

The global `device` parameter has been modified to include the following:

Name	Possible Values	Description
device.syslog.serverName	dotted-decimal IP address OR domain name string	The syslog server IP address or host name. The default value is NULL.
device.syslog.transport	None=0, UDP=1, TCP=2	The protocol that the phone will use to write to the syslog server. If set to "None", transmission is turned off, but the server address is preserved.
device.syslog.facility	0 to 23	A description of what generated the log message. For more information, refer to section 4.1.1 of RFC 3165. The default value is 16, which maps to "local 0".
device.syslog.renderLevel	1 to 6	Specifies the lowest class of event that will be rendered to syslog. It is based on `log.render.level` and can be a lower value. Refer to Basic Logging <level/><change/> and <render/> on page 138 of the *SIP 2.0 Administrator's Guide*.
device.syslog.prependMac	Enabled, Disabled	If enabled, the phone's MAC address is prepended to the log message sent to the syslog server.

Note	The parameters for this feature should be put in separate configuration files to simplify maintenance. Do not add them to existing configuration files (such as **sip.cfg**). Create a new configuration file for parameters that should apply to all phones. Polycom recommends that you test the new configuration files on two phones before initializing all phones.

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Technical Bulletin 5844

SIP Server Fallback Enhancements on SoundPoint® IP Phones

This technical bulletin provides detailed information on how the SIP application has been enhanced to support SIP server fallback.

This information applies to SoundPoint IP phones running SIP application version 2.1 or later.

Introduction

Server redundancy is often required in VoIP deployments to ensure continuity of phone service for events where the call server needs to be taken offline for maintenance, the server fails, or the connection from the phone to the server fails.

Two types of redundancy are possible:

- **Fail-over**: In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using DNS mechanisms or "IP Address Moving" from the primary to the back-up server.

- **Fallback**: In this mode, a second less featured call server (router or gateway device) with SIP capability takes over call control to provide basic calling capability, but without some of the richer features offered by the primary call server (for example, shared lines, presence, and Message Waiting Indicator). Polycom phones support configuration of multiple servers per SIP registration for this purpose.

In some cases, a combination of the two may be deployed.

In SIP 2.1, the fallback behavior has been enhanced and this behavior is described in this document.

Note   Your SIP server provider should be consulted for recommended methods of configuring phones and servers for fail-over configuration.

Warning   The server redundancy behavior in SIP2.1 has changed from that implemented in prior releases. Prior to SIP 2.1, the `reg.x.server.y` parameters (see section 4.6.2.1 of the *SIP 2.0 Administrator's Guide*) could be used for fail-over configuration. The older behavior is no longer supported. Customers that are using the `reg.x.server.y` configuration parameters where y>=2 should take care to ensure that their current deployments are not adversely affected. For example the phone will only support advanced SIP features such as shared lines, missed calls, presence with the primary server (y=1).

Terminology

Before you read this document, it is important to understand certain terminology and become familiar with the server/registration configuration as described in the references listed in the References on page 8. The behavior described in this document supersedes that described in section 3.6.5 of the *SIP 2.0 Administrator's Guide*.

SIP Registrations: SoundPoint IP phones support the ability to have multiple SIP Registrations per phone. This is often used to support multiple "Lines" on a single phone and normally the SIP server(s) used for each Registration are the same. However, they could be different.

Primary and Fallback Servers: Each of these SIP Registrations may be configured for concurrent registration with multiple servers for fallback purposes. For example, a phone may be configured to have two SIP Registrations and each SIP Registration may be configured with two separate servers (a primary server and a fallback server). DNS mechanisms (as described in RFC3263) may be used such that the servers are capable of resolving to multiple physical SIP servers for fail-over purposes.

Note

The primary server is the only one that will be used for advanced SIP features such as shared lines, message waiting indicators, and presence. This is a change in behavior from software releases before SIP 2.1 All other configured servers are referred to as fallback servers.

Working Server: The phone maintains a list of all possible servers gained from both DNS and configuration. The highest priority server which has an active registration is treated as the working server and will be the first server tried for call initiation purposes. At any time, there is only one working server recognized by the phone.

Registrar Server: Servers (both primary and fallback) may be configured with registration enabled or disabled using the `reg.x.server.y.register` configuration parameter. Servers that have this parameter enabled will attempt registrations and are termed a registrar server. If a server is not a registrar server, calls will be attempted on that server if appropriate, but registration will not be attempted. Only a registrar server can become the working server.

For the purposes of this document, we will use examples where the phone has only one SIP Registration.

The sections Server <server/> on page 95 and Registration <reg/> on page 149 of the *SIP 2.0 Administrator's Guide* describe the parameters that are relevant to the configuration of the phones for server redundancy and fallback behavior.

Configuration file changes for SIP 2.1 are described in Configuration File Changes on page 7.

2

SIP 2.1 Server Fallback Implementation

In the SIP 2.1 release, the redundancy behavior of Polycom SoundPoint IP and SoundStation IP phones has been changed and improved by adding the ability for a single SIP Registration (Line) to be registered to more than one server concurrently. In previous releases, the phone would only maintain one active server registration per SIP Registration (Line). The concurrent server registration capability adds an ability to do a faster and more efficient hand-over to an independent call server both for incoming as well as outgoing calls.

To assist in explaining the redundancy behavior, an illustrative example of how a system may be deployed is defined in the following section.

Example Deployment

A small medium business (SMB) customer uses a hosted IP-Centrex service from a Service provider. The Service provider has two redundant call servers at their network operations center (NOC) and uses a DNS server to resolve the IP addresses of these servers. The SMB customer also has an on-premise router which has the ability to handle SIP call traffic and has a connection to an on-site PSTN gateway. This gateway is intended to be used in conditions in which the Internet connection to the service provider is not working.

3
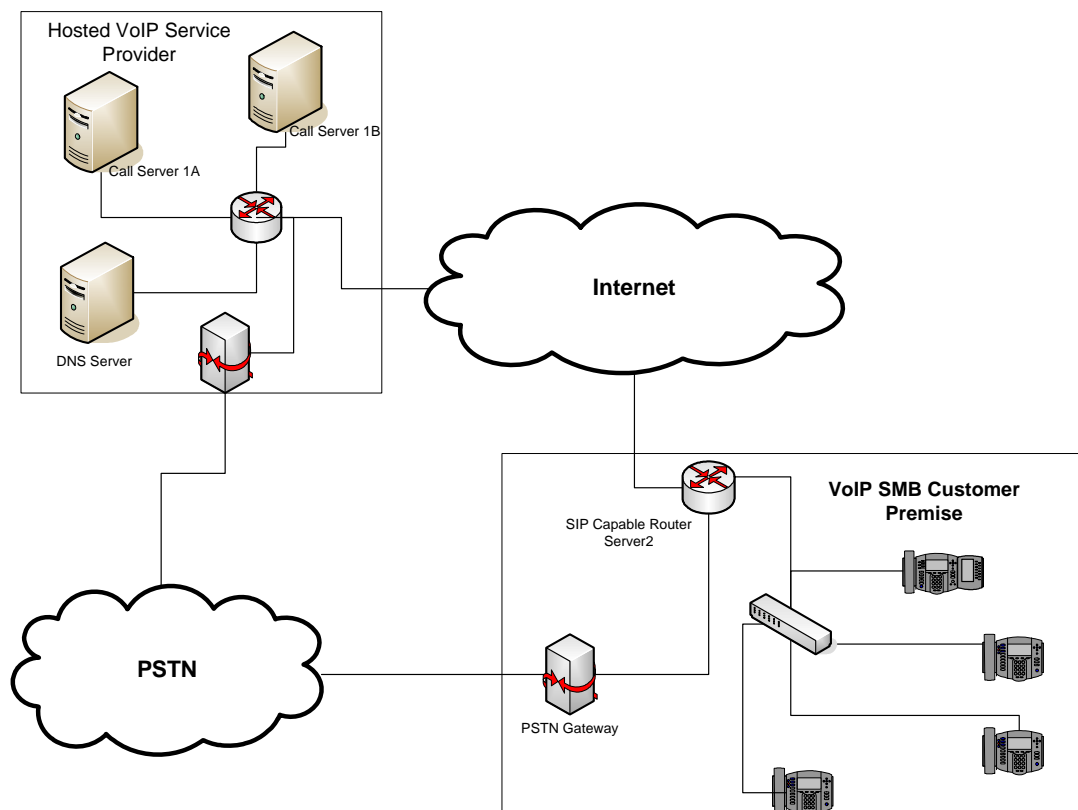
Phone Configuration

The phones at the customer site are configured as follows:

- Server 1 (the primary server) will be configured with the address of the service provider call server. The IP address of the server(s) to be used will be provided by the DNS server. For example:

  ```
  reg.1.server.1.address="voipserver.serviceprovider.com"
  ```

- Server 2 (the fallback server) will be configured to the address of the router/gateway that provides the fallback telephony support and is on-site. For example:

  ```
  reg.1.server.2.address=172.23.0.1
  ```

Note

It is possible to configure the phone for more than two servers per registration, but you need to exercise caution when doing this to ensure that the phone and network load generated by registration refresh of multiple registrations do not become excessive. This would be of particularly concern if a phone had multiple registrations with multiple servers per registration and it is expected that some of these servers will be unavailable.

Phone Operation for Registration

After the phone has booted up, it will register to all the servers that are configured.

Server 1 is the primary server and supports greater SIP functionality than any of servers. For example, SUBSCRIBE/NOTIFY services (used for features such as shared lines, presence, and BLF) will only be established with Server 1.

Upon registration timer expiry of each server registration, the phone will attempt to re-register. If this is unsuccessful, normal SIP re-registration behavior (typically at intervals of 30 to 60 seconds) will proceed and continue until the registration is successful (for example, when the Internet link is once again operational). While the primary server registration is unavailable, the next highest priority server in the list will serve as the working server. As soon as the primary server registration succeeds, it will return to being the working server.

Note

If `reg.x.server.y.register` is set to 0, then phone will not register to that server. However, the INVITE will fail over to that server if all higher priority servers are down.

Behavior When the Primary Server Connection Fails

For Outgoing Calls (INVITE Fallback)

When the user initiates a call, the phone will go through the following steps to connect the call:

1. Try to make the call using the working server.

2. If the working server does not respond correctly to the INVITE, then try and make a call using the next server in the list (even if there is no current registration with these servers). This could be the case if the Internet connection has gone down, but the registration to the working server has not yet expired.

3. If the second server is also unavailable, the phone will try all possible servers (even those not currently registered) until it either succeeds in making a call or exhausts the list at which point the call will fail.

At the start of a call, server availability is determined by SIP signaling failure. SIP signaling failure depends on the SIP protocol being used as described below.

- If TCP is used, then the signaling fails if the connection fails or the Send fails.

- If UDP is used, then the signaling fails if ICMP is detected or if the signal times out. If the signaling has been attempted with all servers in the list and this is the last server then the signaling fails after the complete UDP timeout defined in RFC 3261. If it is not the last server in the list, the maximum number of retries using the configurable retry timeout is used. For more information, refer to Server <server/> on page 95 and Registration <reg/> on page 149 of the *SIP 2.0 Administrator's Guide*.

Warning   If DNS is used to resolve the address for Servers, the DNS server is unavailable, and the TTL for the DNS records has expired, the phone will attempt to contact the DNS server to resolve the address of all servers in its list *before* initiating a call. These attempts will timeout, but the timeout mechanism can cause long delays (for example, two minutes) before the phone call proceeds "using the working server". To mitigate this issue, long TTLs should be used. It is strongly recommended that an on-site DNS server is deployed as part of the redundancy solution.

For Incoming Calls (Incoming Call Fallback)

The primary call server can use mechanisms for detecting that the Internet connection is down and route incoming calls through the PSTN link to the back-up gateway/router on-site. Since the phone is simultaneously registered to both servers, it will receive calls through the gateway even if the primary registration has not expired. This is a key advantage of the new behavior introduced in SIP 2.1.

Changes From Previous Phone Behavior (Releases Before SIP 2.1)

Before SIP 2.1	In SIP 2.1
A Line is only capable of maintaining one server registration.	A Line will maintain registrations with all servers that are configured as registrar servers.
If two servers are configured (for example, `reg.1.server.1.address = "server1"` and `reg.1.server.2.address = "server2"`, the phone will initially register with Server1 as the working server. Phone calls will be placed and received through Server1 only. If the registration to Server1 fails or expires, then the phone will attempt to register with Server2. If this registration succeeds, then incoming calls will be received using this server. At this point, Server2 takes over as the working server.	If two servers are configured (for example, `reg.1.server.1.address = "server1"` and `reg.1.server.2.address = "server2"`, the phone will register with both Server1 as the working and Server2. Phone calls will be placed through Server1, but may be received through either Server1 or Server2. If the registration to Server1 fails or expires, then the Server2 will become the working server.
The phone will continually attempt registration using SIP registration protocols with Server1. At the point that this succeeds, the registration with Server2 will expire and Server1 will resume as the working server.	The phone will continually attempt to register with Server1 and, when this is successful, will switch back to using Server1 as the working server. The Server2 registration will be maintained.
The phone attempts to maintain full SIP functionality with each server, but it is questionable how effective this is.	Only basic SIP registration for INVITE functions is maintained with servers other than the primary server.

Recommended Practices for Server Fallback Deployments

The best method for ensuring optimum server redundancy is to deploy two identical call servers and use either DNS methods or "IP Address Moving" together with call server recommended practices for maintaining synchronization of records between the redundant servers. This is termed fail-over (refer to). Deployment varies dependent on the SIP call server being used. Consult your SIP call server supplier for recommended practices for fail-over configuration.

In situations where server redundancy for fall-back purpose is used, the following measures should be taken to optimize the effectiveness of the solution:

1. Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.

2. Do not use OutBoundProxy configurations on the phone if the OutBoundProxy could be unreachable when the fallback occurs. SoundPoint IP phones can only be configured with one OutBoundProxy per registration and all traffic for that registration will be routed through this proxy for all servers attached to that registration. If Server 2 is not accessible through the configured proxy, call signaling with Server 2 will fail.

3. Avoid using too many servers as part of the redundancy configuration as each registration will generate more traffic.

4. Educate users as to the features that will not be available when in "fallback" operating mode.

Configuration File Changes

Configuration changes can performed centrally at the boot server:

Central (boot server)	Configuration file: **sip.cfg**	Specify global primary and fallback server configuration parameters. • For more information, refer to Protocol in Application Configuration File on page 7.
	Configuration file: **phone1.cfg**	Specify per registration primary and fallback server configuration parameters values that override those in **sip.cfg**. • For more information, refer to Registration in Per-Phone Configuration File on page 8.

Protocol in Application Configuration File

The `<voIpProt/>` attribute includes:

• Server <server/> on page 7.

Server

This configuration attribute now includes:

Attribute	Permitted Values	Default	Interpretation
voIpProt.server.x.lcs	0, 1	0	This attribute overrides the `voIpProt.SIP.lcs`. If set to 1, the proprietary "epid" parameter is added to the From field of all requests to support Microsoft Live Communications Server.

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Registration in Per-Phone Configuration File

Per-registration configuration is supported.

The `<registration/>` attribute now includes:

Attribute	Permitted Values	Default	Interpretation
reg.x.server.y.lcs	0, 1	0	This attribute overrides the `reg.x.lcs`. If set to 1, the Microsoft Live Communications Server is supported for registration x.

References

1. SIP 2.0 Administrator's Guide for the SoundPoint IP and SoundStation IP Phones, August 2006. Go to http://www.polycom.com/common/pw_cmp_updateDocKeywords/0, 1687,6314,00.pdf

2. RFC3263 - Locating SIP Servers. Go to http://www.ietf.org/rfc/rfc3263.txt?number=3263

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