

## IP Office Release 5: SIP phone details

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<b>IOC Document Definition:</b>		
	<b>Product/Platform Summary</b>	This document is an ongoing, evolving summary incorporating the latest updates on the entire product platform. (e.g., any version of IP Office). It is global and will note any regional distinctions within.
✓	<b>Detailed Product Description</b>	This document is the detailed version of the product/platform summary. It is the “everything you need to know” source for the respective product platform. It is global and will note any regional distinctions within.
	<b>Product Update</b>	(Formerly known as the Offer announcement.) This is a communication that summarizes “what’s new” within a given release regarding any changes or enhancements to an existing GA (Generally Available) product.
	<b>Sale Guide</b>	This document provides coaching on the key selling points to highlight with customers. It typically also includes information pertaining to implementation guidelines and “need to know” facts.



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## 2 Executive Summary

Avaya IP Office Release 5 is supporting standard SIP endpoints (telephones) for Voice and Fax transmission (T.38)

The SIP implementations allows to address special customer needs using standard market SIP devices e.g. for IP-based conference phones, analog terminal adapters in remote places or special purpose devices like Hotel phones.

It also gives customers a choice to e.g. reuse existing SIP-based telephones they already have or that better suite there specific needs.

Avaya has confirmed the IP Office SIP implementation against several manufacturers SIP devices to ensure greatest possible compatibility.

This document list

- a) the supported SIP functionality on IP Office including Application support.
- b) A list of endpoints that Avaya used in confirming the SIP implementation. It will list the phones and the used software version. Avaya confirms that at time of writing these phones were working as documented on IP Office. As these are third party phones, Avaya can however not gurantee that these phones – especially with different software versions – will flawlessly work with IP Office.
- c) Phones tested/planned for testing through Avaya DevConnect. These phones have/will be tested together with the vendor of the device in the Avaya DevConnect laboratory. A specific test reports and application notes will be published by Avaya DevConnect.

While great care has been taken to be compliant with SIP standards, no guarantee can be given that all devices claiming support of SIP will work flawlessly. The SIP standard is constantly evolving with new features and methods introduced. Also, while being standard compliant, not all devices implement all options of the standard, making it hard to almost impossible to predict if a device will work without further testing.

Note that all SIP endpoints require a “third party IP phone” license in order to work properly.

### Key messages:

SIP Device support	Resellers	Customers
IP Office supports standard SIP telephones <ul style="list-style-type: none"> <li>• Voice and T.38 FAX</li> <li>• Verified against several vendors products</li> <li>• DevConnect partners testing SIP devices</li> <li>• Uses existing “Third party IP phone license” at new, lower price</li> </ul>	<ul style="list-style-type: none"> <li>• Can close product Gaps, e.g. IP conference phones</li> <li>• Allows the reseller to propose a best in class solution e.g. with special Hotel phones</li> </ul>	<ul style="list-style-type: none"> <li>• Get solutions for specific needs, e.g. conference phones that Avaya does not offer.</li> <li>• Giving customers a choice to select the device that best fits there needs</li> <li>• Emphasizes the goal of Avaya to open standards- customer can make the decision now and in future that best fits there needs.</li> </ul>

## 3 Functional Description of IP Office SIP implementation

### 3.1 SIP Endpoint support

*Session Initiation Protocol (SIP)* is an open signaling protocol for establishing any kind of real-time communication session. The communication session can involve voice, video, or instant messaging, *and* can take place on one of many devices that people use for communicating: laptop computer, PDA, cell phone, IM client, IP phone, and so on. SIP has been developed in the *Internet Engineering Task Force (IETF)* by common participation from various vendors, including Avaya

Avaya IP Office supports SIP for telephony functions to enable the usage of standard based SIP endpoints for Voice and Fax communication. Different to pure SIP systems, IP Office expands the feature set beyond the SIP standard, offering a wealth of IP Office features also on SIP endpoints delivering a feature rich system that a pure-SIP server based on the SIP standard only can't deliver. With that, IP Office delivers the best of both worlds, supporting standard based IP telephones while delivering a wealth of features consistently between SIP, digital and Avaya IP endpoints.

IP Office SIP endpoint implementation is built on two major SIP components: SIP User agents, and SIP Server components.

### 3.2 SIP Components:

#### **SIP Endpoints (User Agents):**

User agents (UAs) are applications in SIP endpoints (such as a SIP phone, cell phone, PDA, or workstation) that interface between the user and the SIP network.

#### **SIP Servers:**

IP Office has implemented the required functionality of the SIP servers mentioned below not only to provide SIP endpoint support but also to allow full interoperability between SIP endpoints, other IP telephones based on H.323, Digital and Analog telephones as well as IP Office trunks (Analog, digital or SIP based)

SIP servers provide centralized information and enablement services in a SIP ecosystem. The core SIP servers and their functions are summarized here. IP Office provides the required features of the following two servers for Voice and FAX communication.

**Registrar Server:** When SIP IP phones come online, they need to make sure that others are aware that they're available to take and make calls. The Registrar authenticates and registers the IP phone (often directly related to a specific user) when it comes online, and then stores information on the phones logical identities.

**Proxy Server:** A proxy server takes SIP requests, processes them, and passes them downstream while sending responses upstream to other SIP servers or devices. A proxy server may act as both a server and a client, and can modify a SIP request before passing it along. A proxy can be involved only in the setup and teardown of a communication session or it can stay in the communication loop for the complete session. After user agents establish a session, communications occurs directly between the parties.

Functionality of the following two SIP servers are generally available by IP Office using existing IP-Office functionality. Therefore, while functionality is provided, e.g. allowing hotdesking (also for

users using a SIP-endpoint) in a small community network, a consistent methodology between SIP and non SIP endpoints is used to deliver those features

**Location Service:** As users roam, the network needs to be continually aware of their locations. The location service is a database that keeps track of users and their locations. The location service gets its input from the registrar server and provides key information to the proxy and redirect servers. IP Office provides hotdesking support, delivering a similar functionality but working consistently between SIP and non SIP endpoints.

**Redirect Server:** If users are not in their home domains, sessions bound for them needs to be redirected to them.

The redirect server maps a SIP request destined for a user to the device “closest” to the user. In IP Office, call forwarding and Follow me functionality is used to provide again consistent functionality between all type of endpoints.

### 3.3 Supported functionality for SIP endpoints in IP Office

Starting with IP Office Release 5, SIP endpoints are supported on IP-Office for Voice (Audio) and Fax (T.38) communication.

This allows the usage of standard compliant IP telephones using the open SIP standard, giving customers a choice of endpoints of different manufactureres including special purpose devices like conference phones, hotel phones or terminal adapters.

In order to use a SIP endpoint with IP Office, a “Third party IP endpoint license” is needed. This license will continue to support endpoints based on the H.323 standard but will also be required for generic SIP endpoints on IP Office

SIP Endpoint support is fully integrated into IP Office core. No SIP specific components are needed. SIP endpoints will need VCM module capacity in IP Office like any other IP phone. Next to SIP telephones, SIP terminal adapters are supported to connect analog phones and fax machines. This offers to e.g support these devices in remote sites were otherwise only IP-phones are used.

SIP extensions function like any other IP Office extension with the supported features below. This means they:

- Can make and receive calls to any other extension, independet of type of extension
- Delivers end to end media just like any other IP telephone on IP Office. For calls between two SIP extensions of a SIP extension and a Avaya IP telephone, the audio is transmitted end to end for basic telephone calls. (Conferences etc. however require a VCM resource).
- Can use short codes and authorization codes like any other phones
- Transmit In band call progress tones are delivered from IP Office
- A SIP phones needs to register with IP Office like any other IP telephone, Authentification with Username and password is possible
- SIP extensions support “auto create” in IP Office to make installation fast and efficient. Succesfull registration of a endpoint will consume one third party license
- On one IP address, several extension can register with IP Office, each consuming a license. This enables the connection of SIP terminal adapters with more then one analogue port, giving a different extension number to each of the ports.

Advanced features:

SIP endpoints support a number of extended features according to the “SIP service samples-draft”, also referred to as “Sipping-19”. This includes:

- Calling line identification
- Hold/Consulation Hold
- Attended/Unattended Transfer
- Message Waiting
- Do not disturb
- Conference Add

Some phones support several call appearances making it easy to switch between calls. Please note that this does not include “bridged appearances” or “(outside)-line appearances”

A large number of additional features are supported on IP Office using Feature activation keys. These features include but are not limited to:

- Call forward: Unconditional/Busy/no Answer
- Follow me
- Park/Unpark
- Music on Hold
- Meet me conferencing
- Conference join
- Ring back when free
- .....

IP endpoints also support Computer Telephony Integration “CTI” and therefore applications like One-X portal for Small business: the following features are supported with One-X portal and via the TAPI interface:

- Outgoing call (*without remote activation of speakerphone/headset*)
- Hang up
- Hold
- Attended/Unattended transfer
- Conference (IP Office based)
- Voicemail collect
- Set forwarding/DND (IP Office based)
- Park/Ride (IP Office based)

The features work similar like “CTI” features in combination with an analog telephone, e.g. a outgoing call will first ring the SIP phone and after connect the outgoing call will be placed. Avaya Phone Manager/ Phone Manager Pro and Soft console are currently not supported in combination with SIP-endpoints.

The SIP endpoint support implementation is compliant with the following standards or “RFCs”.

- RFC 3261 - SIP session Initiation Protocol
- RFC 1889 – RTP
- RFC 1890 - RTP Audio

- RFC 4566 – SDP
- RFC 2833 - RTP /DTMF
- RFC 3264 - SDP Negotiation
- RFC 3265 - Event Notification
- RFC 3515 - SIP Refer
- RFC 3842 - Message Waiting
- RFC 3310 – Authentication
- RFC 2976 – INFO
- RFC 3323 - Privacy for SIP (PAI) and draft-ietf-sip-privacy-04 (RPID)

### 3.4 Support Statement

While great care has been taken to be compliant with SIP standards, no guarantee can be given that all devices claiming support of SIP will work flawlessly. The SIP standard is constantly evolving with new features and methods introduced. Also, while being standard compliant, not all devices implement all options of the standard, making it hard to almost impossible to predict if a device will work-

Avaya has therefore developed a support policy that should be followed and understood when using phones that Avaya has not fully tested:

- a) Avaya fully supports telephones that have been tested in the DEV-connect program. Please refer to the DEV connect testing document for the details of these phones and the used software version. We will work – if required with the vendor of these devices – to fix any problems with these devices . Phones in this category can be found in chapter 5 of this document
- b) Phones that Avaya used to verify the SIP implementation. The list in Chapter 4 shows devices that have been used to verify the SIP implementation of IP Office. If the devices on this list have not been certified by the DevConnect program, Avaya will address any issues in the IP Office software when using these devices provided that the devices are using the documented software versions and features. However, Avaya can not provide support if the problems are caused by the device itself. Avaya will periodically update this list and will only provide support against devices and software versions specified in the most recent list.
- c) The list in chapter 6 of this document shows devices that have been used and verified outside of Avaya. This list is provided solely for informational purposes. Avaya can not guarantee that these devices will work flawlessly. Avaya can not verify that these devices will work in future releases of IP Office. Avaya Back Bone Engineering team will not be obliged to support these phones, and in many cases they are not available to Avaya.

Avaya therefore recommends to use phones from category a). If a specific device is requested for a customer scenario, we suggest that the manufacturer of these phones contacts Avaya DevConnect for full verification.

## 4 Phones used to Verify the Avaya IP Office SIP implementation.

Avaya confirms that at time of writing the phones listed in this were working as documented on IP Office. As these are third party phones, Avaya can however not guarantee that these phones – especially with different software versions – will flawlessly work with IP Office.

Note that this list is specific to IP Office. Other Avaya Call servers might support a different line of phones:

### 4.1 Conference Phones:

ClearOne ([www.clearone.com](http://www.clearone.com))

*MAX IP*

Polycom ([www.polycom.com](http://www.polycom.com))

*Soundstation 7000*

### 4.2 Analog Terminal Adapters:

Avaya

*Quick Edition A10 ATA (not longer available new)*

Innovaphone ([www.innovaphone.com](http://www.innovaphone.com))

*IP22*

*IP24*

*IP28*

Patton ([www.patton.com](http://www.patton.com))

*Single line ATA*

### 4.3 WIFI phones:

Ascom ([www.ascom.com](http://www.ascom.com))

*I75*

### 4.4 Desktop Telephones:

DORO ([www.doro.com](http://www.doro.com))

*IP 810C*

Grandstream ([www.grandstream.com](http://www.grandstream.com))

*GXP 2000*

*GXP2020*

Polycom ([www.polycom.com](http://www.polycom.com))

*Soundpoint 7000*

SNOM ([www.snom.com](http://www.snom.com))

*Snom 320*

### 4.5 Softphones phones:

Counterpath ([www.counterpath.com](http://www.counterpath.com))

*X-Lite*

*Eyebeam*

## 4.6 Test Results

The list shows the features that were successfully tested with the phones/devices listed. If no remark is given, this functionality was not tested and therefore it might work. Only in case a comment is made that the feature is not working/not supported on a specific device then testing of this feature has failed.



Overview of SIP conference phones, Analog Terminal Adapters and WIFI phones verified:

			SIP Conference Phones		Analog Terminal Adapters					WIFI
Feature	Detail	Note	Clear One	Polycom	Avaya Quick Edition	Patton	Innovaphone			Ascom
			MAX IP	7000	A10 ATA	SmartLink™ M-ATA	IP22	IP24	IP28	175 Messenger
Software version used on the phone/Device			03-05-07	3.1.2.0392	R3.21		7,00	7,00	7,00	1.6.12
Basic Call			x	x	x	x	x	x	x	x
	Calling Line Identification		x	x	no	no	no	no	no	x
Multiple Call appearances		No bridged or line appearances!	x	x	no	no	no	no	no	x
Message Waiting				x						x
Hold/Consultation Hold			x	x	x	x	x	x	x	x
Transfer	Attended		x	x	no		no	no	no	x
Unattended Transfer	Unattended		x	x	no		no	no	no	x
Do not disturb			x	x	x	x	x	x	x	x
Forward	Unconditional		x	x	x	x	x	x	x	x
	Busy				x	x	x	x	x	
	No Answer		x	x	x	x	x	x	x	x
Conference	Add		no	x						
	Join		x	x	x	x	x	x	x	x
Park		Feature Access code	x	x	x	x	x	x	x	
Unpark		Feature Access code	x	x	x	x	x	x	x	
MoH		IP-Office based	x	x	x	x	x	x	x	x
Audio/Payload										
	G.711		x	x	x	x	x	x	x	x
	G.729a		x	x	x	x	x	x	x	x
	DTMF Inband				no		no	no	no	no
	DTMF Out of band				x		x	x	x	x
	T.38		n.a.	n.a.	x	x	x	x	x	n.a.
Use shortcodes			x	x	x		x	x	x	x
Use Authorization codes			no	x	x		x	x	x	
CTI/Tapi Support										
	Outgoing call	(No auto hook of)	x	x	x	x	x	x	x	x
	Hang up	(Terminate call but will not hang-up telephone)	x	x	x	x	x	x	x	x
	Hold		x	x	x	x	x	x	x	x
	Attended/unattend transfer		x	x	unattended	unattended	unatt.	unatt.	unatt.	attend.
	Conference (IPO based)		x	x	x		x	x	x	x
	Voicemail collect		x	x	x		x	x	x	x
	Set Forwarding/DND (IP Office based)		x	x	x		x	x	x	x
	Park/Ride (IPO based)		x	x	x		x	x	x	x
Manufacturers WEB-site:			<a href="http://www.clearone.com">www.clearone.com</a>	<a href="http://www.polycom.com">www.polycom.com</a>		<a href="http://www.patton.com">www.patton.com</a>	<a href="http://www.innovaphone.com">www.innovaphone.com</a>			



Overview of SIP Clients (Softphone) and Desktop phones verified:

			PC and mobile phone client			Desktop Telephones					
Feature	Detail	Note	Counterpath		Nokia SIP client	Doro	Grandstream		Snom	Polycom Soundpoint	
			Eyebeam	X-lite	E61i		IP 8x0C	GXP 2000	GXP 2020	320	320
Software version used on the phone/Device			1.5.19.2	3.0	RM-227 1.0633.22.1	7.1.31	1.1.6.16	1.1.6.16	6.5.18	2.1.2.0049	1.6.7.0130
Basic Call			x	x	x	x	x	x	x	x	x
	Calling Line Identification		x	x	x	x	x	x	x	x	x
Multiple Call appearances		No bridged or line appearances!	x	x	x	x	x	x	x	x	x
Message Waiting			x	x	no		x	x	x	x	x
Hold/Consultation Hold			x	x	x	x	x	x	x	x	x
Transfer	Attended		x	x	no	x	x	x	x	x	x
Unattended Transfer	Unattended		x	x	no	x	x	x	x	x	x
Do not disturb			x	x	x	x	x	x	x	x	x
Forward	Unconditional		x	x	x	x	x	x	x	x	x
	Busy										x
Conference	No Answer		x	x	x	x	x	x	x	x	x
	Add		x	no		x					
	Join		x	x	x					x	x
Park		Feature Access code	x	x	x	x	x	x	x	x	x
Unpark		Feature Access code	x	x	x	x	x	x	x	x	x
MoH		IP-Office based	x	x	x	x	x	x	x	x	x
Audio/Payload											
	G.711		x	x	x	x	x	x	x	x	x
	G.729a		x		x	x	x	x	x	x	x
	DTMF Inband				no	no	x	x			
	DTMF Out of band		x	x	x	x	x	x	x	x	x
	T.38				n.a.	n.a.	n.a.	n.a.	n.a.	n.a.	n.a.
Use shortcodes			x	x	x	x	x	x	x	x	x
Use Authorization codes			x	x	x	x	x	x	x	x	x
CTI/Tapi Support											
	Outgoing call	(No auto hook of)	x	x	x	x	x	x	x	x	x
	Hang up	(Terminate call but will not hang-up telephone)	x	x	x	x	x	x	x	x	x
	Hold		x	x	x	x	x	x	x	x	x
	Attended/unattend transfer		x	x	x	x	x	x	x	x	x
	Conference (IPO based)		x	x	x	x	x	x	x	x	x
	Voicemail collect		x	x	x	x	x	x	x	x	x
	Set Forwarding/DND (IP Office)		x	x	x	x	x	x	x	x	x
	Park/Ride (IPO based)		x	x	x	x	x	x	x	x	x
Manufacturers WEB-site:			<a href="http://www.counterpath.com">www.counterpath.com</a>	<a href="http://www.nokia.com">www.nokia.com</a>		<a href="http://www.doro.com">www.doro.com</a>	<a href="http://www.grandstream.com">www.grandstream.com</a>	<a href="http://www.snom.com">www.snom.com</a>	<a href="http://www.polycom.com">www.polycom.com</a>		

#### 4.7 Known issues with above devices:

##### Counterpath Eye-Beam/X-Lite

Counterpath EyeBeam doesn't adhere to the codec negotiation when it is a called party. It uses G.711A for sending although it wasn't first in the offer sent from IP Office or in the SDP response sent by the phone. There are 2 workarounds for this behavior:

- a) disable all Codecs but one on the Counterpath Softphone
- b) from the phone dial \*\*\*7469 (SEND)
  - filter for honor
  - double click on the honor line and change the value to 1
  - save and restart

## 5 Phones fully tested in DevConnect

DevConnect testing has just started. As of time of writing, only one phone has passed the test but a line of phones are scheduled for testing:

**For Partners interested in testing SIP devices with IP Office please contact DevConnect at:**

<https://devconnect.avaya.com>

For DevConnect tested phones, a full test report with applications notes will be provided. This document will be updated on a regular basis in order to include information about successful test:

The following phone has passed test. A test-report is or will be published shortly

### ClearOne: Conference phones

- *MAX IP*
- *MAXAttach IP*

Testreport: <http://marketingtools.avaya.com/knowledgebase/ipoffice/mergedProjects/manuals/appnotes/ClearOne-IPO.pdf>

Company web-page: <http://www.clearone.com/>

### Polycom: Conference Phones

- *SoundStation IP 6000*
- *SoundStation IP 7000*

Testreport: <http://marketingtools.avaya.com/knowledgebase/ipoffice/mergedProjects/manuals/appnotes/Polycom-SS-IPO.pdf>

Company web-page: <http://www.polycom.com/>

### Snom: Desktop telephones

- *3x0: Desktop telephones (Tested sample 370)*
- *820: Desktop phone*
- *M3 wireless phone (DECT)*
- *MeetingPoint: Conference phone*

Testreport: in preparation

Company web-page: <http://www.snom.com>

### Teledex: Hotel phones:

- *LD4200: Large screen Hotel phone (Tested sample LD4210S)*
- *ND2000: Entry level Hotel phone (Tested sample ND2210S)*
- *NDC2000: Entry Level Hotel phone (Tested sample NDC2210S)*

Testreport: in preparation

Company web-page: <http://www.teledex.com/>

The following phone is scheduled for testing:

Ascom I75 WIFI phone <http://www.ascom.com/>

## 6 Field proven phones

In this list we provide details about phones that have been reported to work.

Avaya recommends performing all functional tests as mentioned in the included verification list. If a fully filled out test-report is send to Avaya showing the device type, software version and test results and test-person/site we will add these phones to that list.

Avaya has not confirmed this and information is therefore given without any liability. Even if a device has been verified against a single IP Office release, Avaya can not guarantee that different IP Office releases will support these phones in the same way, e.g. after new functionality has been added to IP Office or field-reported issues have been fixed.

Yealink T28P VoIP Phone (Software 2.3.0.3)

Company web-page: global: <http://www.yealink.com/>: UK: [www.yealink.co.uk](http://www.yealink.co.uk)