SIP Endpoint Support

Session Initiation Protocol (*SIP*, pronounced just like *sip*, as in sipping from a fire hose on a hot day) 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.

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 functionaliy 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 the 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 is involved only in the setup and teardown of a communication session. After user agents establish a session, communications occur 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.

Supported functionality for SIP endpoints in IP Office

Starting with IP Office R5, 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 manufacterers 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 other 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 a flexibility to support Fax machines and Audio/T.38

SIP extensions function like any other IP Office extension: 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). See chapter "VCM modules" for details
- 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 efficent. Succesfull registration of a endpoint will consume one thrid 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 refered to as "Sipping-19". This includes:

- **Calling line identification**
- **Hold/Consultation 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 not that this does not include "bridged appearances" or " (outside)-line appearances)

A large number of additional features are supported on IP Office using Featue activation keys. These feature include but 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**

SIP 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)

For codecs support please refer to chapter to VoIP Standards Supported.

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 will only confirm functionality of devices that we have tested and will publish a list of devices that have been tested including – if required – implementation details and software version used on that device.

As of time of writing, the following devices have been tested successfully with IP Office Release 5 for Audio and/or Fax transmission.

SIP Telephones:	• Polycom Soundpoint IP 320, IP 601 • Grandstream GXP 2000, GXP 2020
SIP clients on mobile cell phones:	• Nokia S60 v3 SIP client (e.g. Nokia E61i mobile cell phone)
SIP Analog Teminal Adapters	• Quick Edition A10 ATA • Patton single line M-ATA • Innovaphone IP22, IP24, IP28
SIP PC-based softphones:	• CounterPath eyebeam/xlite

A updated list will be provided in the IP Office knowledge base and on http://support.avaya.com.

15-601041 Issue 20.e.-
15:01, 22 February 2010
(sip_endpoint_support.htm)

http://marketingtools.avaya.com/knowledgebase/businesspartner/ipoffice/mergedProjects/productdescription/sip_endpoint_support.htm?zoom_highlight=sip+phones

Last Modified: 25/01/2010