Voice over IP (VoIP) Application Note

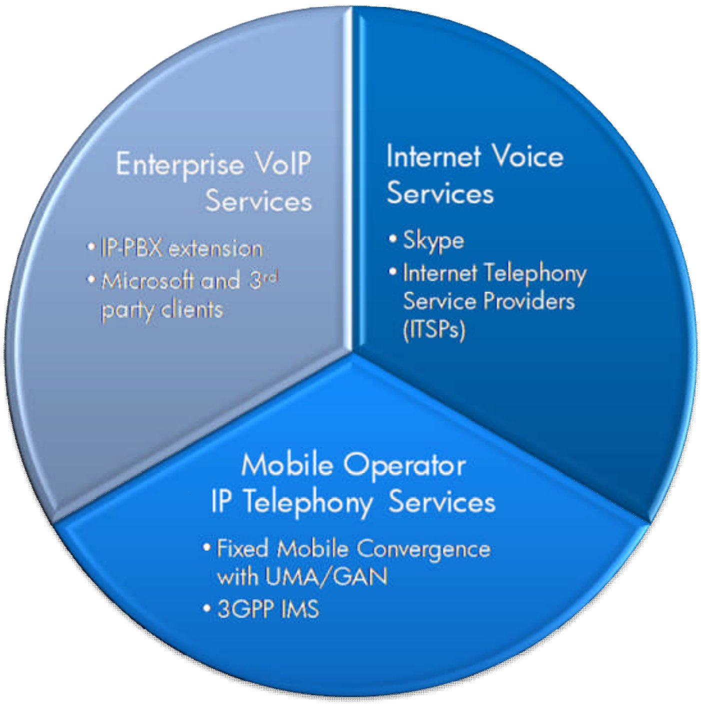
HP iPAQ 500 series Voice Messenger

The goal of this document is to clearly and concisely state what HP iPAQ 500 series Voice Messenger is and is not capable of supporting for mobile IP telephony, aka VoIP, or Voice over IP. All the capabilities and interoperability scenarios haven't been tested at this point, and until then this document will be a work-in-progress. As new information becomes available, it will be added to this document.

Contents

1 Overview

Mobile IP telephony services can be separated into three broad categories, each with its own specific target markets and technologies used. These categories are illustrated below:



Tested use cases for the built-in VoIP client on the HP iPAQ 500 series smartphone cover the enterprise VoIP segment exclusively and expect the enterprise has IT staff that manage their own IP-PBX systems and provision devices for enterprise users. Mobile enterprise VoIP services should be accessible anywhere the VoIP-enabled enterprise wireless local area network (WLAN) provides coverage, typically within company buildings and possibly around corporate campuses. Using the built-in VoIP client for mobile access to the company's IP-PBX from remote locations (homes, hotels, Internet hotspots, etc.) is not a supported capability.

HP uses the native Microsoft SIP (Session Initiation Protocol) client in Windows Mobile 6 to access VoIP services, and this SIP client's capabilities and limitations have been further split into the following categories for the purposes of this document:

- Media and Signaling Protocols
- Telephony Features

- Audio Quality
- Hardware and Firmware
- WLAN Infrastructure
- IP-PBX and SIP Server Support
- Accessory Support

While Internet voice services delivered by Internet Telephony Service Providers (ITSPs) may also be based on SIP, HP has done no testing or validation of interoperability with these services at this time.

2 Media and Signaling Protocols

This category is related to the industry protocols that exist today for carrying control messages and media messages.

Supported:

- **SIP –** Session Initiation Protocol for Signaling [RFC 3261]
- **G.711 –** Audio codec for compression/decompression of voice. Both A-law and u-law variants are included.

Not Supported:

- **H.323 –** signaling protocol used in some IP-PBXs
- **SCCP –** Cisco's proprietary call control protocol for Unified CallManager.
- **G.729 –** Compressed audio codec more suitable for voice over WLAN compared to G.711
- **Secure RTP [SRTP] –** secure version of the RTP protocol [RFC 3711]

A full list of the standards supported and not supported is included in Appendix A of this document.

3 Telephony Features

This category is related to the standard telephony features available in a VoIP implementation.

Supported:

- Originate and Terminate Calls
- Caller ID
- Call Waiting
- Call Hold
- Call Mute
- Call Forwarding *only with a SIM card present
    - The number the call is forwarded to must be in a format [E.164] that can be validated on the GSM network.
- Configuring Caller ID *only with a SIM card present
- Configuring Call Waiting *only with a SIM card present
- Call Barring *only with a SIM card present
- Blind Transfer
- In-Band and out-of-band DTMF
    - Out-of-band DTMF is requested by default. If the other end-point declines, in-band DTMF is used.
- Emergency Calling (over GSM only)

Interoperability testing with SIP-enabled IP PBX systems is being done to confirm feature functionality with each vendor's SIP implementation. Refer to subsequent sections of this document for test results.

Not Supported:

- Conferencing a second line
- Consultative Transfer
- Call Park/Pick-up
- 'Do Not Disturb'
- Emergency Calling over IP

A full list of the RFCs supported and not supported is included in Appendix A of this document.

4 Audio Quality

This category is related to the components that need to be in the device to handle audio quality.

Supported:

- Automatic Gain Control

- Adaptive Jitter Buffer Management
- Voice Activity Detection
- Silence Suppression
- Comfort Noise Generation
- Acoustic Echo Cancellation (3rd party sourced, not provided by Microsoft)

5 Hardware and Firmware

This category is related to WLAN and other hardware/firmware components in the handset that are critical to the performance and usability of VoIP.

Supported:

- 802.11b/g
- 802.11i (PEAPv0 and EAP-TLS with certificates)
- Encryption suites: WEP64, WEP128, TKIP and AES CCMP

Not Supported:

- **Full 802.1X Supplicant –** Microsoft provides the limited set of EAP authentication noted above. Additional EAP methods may be required by some customers.

  *A 3rd party supplicant is currently under consideration for a future release

- **Fast Roaming –** A basic roaming agent has been implemented, and specific roaming performance results will be published when they become available. 802.11r "Fast Roaming" is currently unsupported.

  *Fast roaming enhancements are under consideration for a future release

- **Cisco Compatible Extensions (CCX) –** The WLAN module firmware and driver would have to be updated to support the CCX v4 ASD feature set.

  *A CCX v4 implementation is currently under consideration for a future release

- Certain features of **802.11e [WMM]** are not currently supported.
  - Automatic Power Save Delivery **(APSD)** and Unscheduled Automatic Power Save Delivery **(U-APSD)** power saving mechanisms are not currently supported.
  - Even though packet tagging [**802.1p** and **Diffserv**] is inherently supported by Windows Mobile 6, there may not be any benefit by tagging voice packets with a higher priority.

6 WLAN Infrastructure

This category is related to the wireless local area network (WLAN) infrastructure to which the handset connects to send and receive IP traffic. The key consideration from the mobile handset perspective is typically AP-AP roaming. Given the real time requirements of voice and the delays inherent in WLAN authentication (802.11i), the strongly recommended answer is a VoIP-enabled pervasive enterprise WLAN infrastructure. Note that HP Services has a well-developed practice for migrating customers' WLANs to this model. The baseline requirement here is a controller-based AP deployment; not standalone APs. Examples of these controller-based WLAN product offerings include:

- Cisco Aironet APs with Wireless LAN controller(s)
- HP ProCurve Radio Ports plus Wireless Edge Services xl module(s) in 5300xl switch chassis
- Extreme Networks Altitude APs and Summit switches
- Aruba Networks APs and Mobility Controller(s)
- Meru Networks APs and Controller(s)

Testing is underway with Cisco Aironet APs and wWireless LAN controllers. Other WLAN infrastructure testing and results may be available in the future.

For optimal performance, all 802.11b and 802.11g data rates should be enabled on the APs. Limiting the data rates on the APs may prevent iPAQ 500 series devices from connecting to the AP.

7 IP-PBX and SIP Server Support

This category is related to the SIP-based IP-PBX system or server with which the handheld device must interact for VoIP services. HP testing is underway with enterprise IP-PBX products from Cisco, Avaya, Alcatel, and Nortel. Detailed test criteria and results will be added to this section as they become available.

Results of other additional testing with operator-class SIP servers (Broadsoft, Sylantro, Huawei, Alcatel, etc.) will be included when they are made available to HP.

7.1 COMPATIBILITY WITH CISCO UNIFIED CALL MANAGER V5.1

Feature	Brief Description	Pass/Fail	Comments
Originate Calls	SIP End Point to End point, Extension, and PSTN	Pass	
Terminate Calls	From SIP End Point, Extension, and PSTN	Pass	
DTMF (in-band)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
DTMF (RFC 2833)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
Call Forward	Call Forward All (CFA), Busy (CFB) and Not Answered (CFNA)	Pass	Requires SIM card and accepts valid E.164 numbers only
Message Waiting Indicator	Voicemail notification passed from server to Endpoint	**Fail**	Cisco uses SIP Notify, Enhancement request submitted to Microsoft
Blind Transfer	Originator and Terminator transfer to Endpoint, Extension and PSTN	Pass	
Call Waiting	On Originator and Terminator, waiting, timeout, release before answer	Pass	Requires SIM card to configure
Caller ID	Calling Line ID Presentation (CLIP) Type I and II	Pass	Requires SIM card to configure
Call Barring	Calling Line ID Restriction (CLIR)	Pass	Requires SIM card to configure
Direct Inward Dial	DID from Endpoint to Local and Remote Extension and PSTN to Endpoint	Pass	
Call Hold (Music on Hold)	Originator and Terminator hold and resume with Endpoint, Extension, and PSTN	Pass	

7.2 COMPATIBILITY WITH AVAYA COMMUNICATION MANAGER V4.0 / SES V3.1.2

Feature	Brief Description	Pass/Fail	Comments
Originate Calls	SIP End Point to End point, Extension, and PSTN		Testing still in progress
Terminate Calls	From SIP End Point, Extension, and PSTN		
DTMF (in-band)	SIP End Point, Extension, and PSTN send and receive DTMF tones		
DTMF (RFC 2833)	SIP End Point, Extension, and PSTN send and receive DTMF tones		
Call Forward	Call Forward All (CFA), Busy (CFB) and Not Answered (CFNA)		
Message Waiting Indicator	Voicemail notification passed from server to Endpoint		
Blind Transfer	Originator and Terminator transfer to Endpoint, Extension and PSTN		
Call Waiting	On Originator and Terminator, waiting, timeout, release before answer		
Caller ID	Calling Line ID Presentation (CLIP) Type I and II		
Call Barring	Calling Line ID Restriction (CLIR)		
Direct Inward Dial	DID from Endpoint to Local and Remote Extension and PSTN to Endpoint		
Call Hold (Music on Hold)	Originator and Terminator hold and resume with Endpoint, Extension, and PSTN		

7.3 COMPATIBILITY WITH NORTEL MCS 5100 V4.5

Feature	Brief Description	Pass/Fail	Comments
Originate Calls	SIP End Point to End point, Extension, and PSTN	Pass	
Terminate Calls	From SIP End Point, Extension, and PSTN	Pass	
DTMF (in-band)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
DTMF (RFC 2833)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
Call Forward	Call Forward All (CFA), Busy (CFB) and Not Answered (CFNA)	Pass	Requires SIM card and accepts valid E.164 numbers only
Message Waiting Indicator	Voicemail notification passed from server to Endpoint	Pass	
Blind Transfer	Originator and Terminator transfer to Endpoint, Extension and PSTN	Pass	
Call Waiting	On Originator and Terminator, waiting, timeout, release before answer	Pass	Requires SIM card to configure
Caller ID	Calling Line ID Presentation (CLIP) Type I and II	Pass	Requires SIM card to configure
Call Barring	Calling Line ID Restriction (CLIR)	Pass	Requires SIM card to configure
Direct Inward Dial	DID from Endpoint to Local and Remote Extension and PSTN to Endpoint	Pass	
Call Hold (Music on Hold)	Originator and Terminator hold and resume with Endpoint, Extension, and PSTN	Pass	

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7.4 COMPATIBILITY WITH NORTEL CS1000 V4.5

Feature	Brief Description	Pass/Fail	Comments
Originate Calls	SIP End Point to End point, Extension, and PSTN		Testing still in progress
Terminate Calls	From SIP End Point, Extension, and PSTN		
DTMF (in-band)	SIP End Point, Extension, and PSTN send and receive DTMF tones		
DTMF (RFC 2833)	SIP End Point, Extension, and PSTN send and receive DTMF tones		
Call Forward	Call Forward All (CFA), Busy (CFB) and Not Answered (CFNA)		
Message Waiting Indicator	Voicemail notification passed from server to Endpoint		
Blind Transfer	Originator and Terminator transfer to Endpoint, Extension and PSTN		
Call Waiting	On Originator and Terminator, waiting, timeout, release before answer		
Caller ID	Calling Line ID Presentation (CLIP) Type I and II		
Call Barring	Calling Line ID Restriction (CLIR)		
Direct Inward Dial	DID from Endpoint to Local and Remote Extension and PSTN to Endpoint		
Call Hold (Music on Hold)	Originator and Terminator hold and resume with Endpoint, Extension, and PSTN		

7.5 COMPATIBILITY WITH ALCATEL OMNIPCX ENTERPRISE V6.1

Feature	Brief Description	Pass/Fail	Comments
Originate Calls	SIP End Point to End point, Extension, and PSTN	Pass	
Terminate Calls	From SIP End Point, Extension, and PSTN	Pass	
DTMF (in-band)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
DTMF (RFC 2833)	SIP End Point, Extension, and PSTN send and receive DTMF tones	Pass	
Call Forward	Call Forward All (CFA), Busy (CFB) and Not Answered (CFNA)	Pass	Requires SIM card and accepts valid E.164 numbers only
Message Waiting Indicator	Voicemail notification passed from server to Endpoint	**Not tested**	No voicemail service on test system
Blind Transfer	Originator and Terminator transfer to Endpoint, Extension and PSTN	Pass	
Call Waiting	On Originator and Terminator, waiting, timeout, release before answer	Pass	Requires SIM card to configure
Caller ID	Calling Line ID Presentation (CLIP) Type I and II	Pass	Requires SIM card to configure
Call Barring	Calling Line ID Restriction (CLIR)	Pass	Requires SIM card to configure
Direct Inward Dial	DID from Endpoint to Local and Remote Extension and PSTN to Endpoint	Pass	
Call Hold (Music on Hold)	Originator and Terminator hold and resume with Endpoint, Extension, and PSTN	Pass	

8 Accessory Support

This category is related to headsets and other accessories that may be used with the handset in conjunction with VoIP.

Supported:

- Wired headsets
- Mono Bluetooth headsets

Not Supported:

- Stereo Bluetooth headsets

9 Configuring VoIP

The Phone Dialer on the HP iPAQ 500 series Voice Messenger can be used to make VoIP calls in addition to normal cellular calls. VoIP can be configured to work with SIP-enabled IP-PBX or other SIP servers. Refer to Section 7 above for specific interoperability test results.

9.1 ENABLING 'INTERNET CALLING'

Before VoIP can be configured, 'Internet Calling' needs to be turned on. 'Internet Calling' is turned off by default. The 'Internet Calling' plug-in on the Home screen displays 'Off' indicating that the feature is turned off. To turn the feature on, select 'Internet Calling' on the Home screen and press Enter. This displays the Internet Calling Settings page. Change the 'Use Internet Calling:' setting from 'Never' to any of the other available options. This changes the status of the plug-in on the Home screen from'**Off**' to '**Not Available'**. This indicates that even though VoIP is turned on, it hasn't been configured yet.

9.2 CHECKING WI-FI

Wi-Fi connectivity is required for VoIP to work. Before configuring VoIP, please set-up Wi-Fi and make sure the phone can connect to an Access Point [AP] and that Internet Explorer Mobile can be used to browse web pages.

9.3 USING HP IPAQ SETUP ASSISTANT

Once 'Internet Calling' is turned on and Wi-Fi is set-up, VoIP can be configured using the HP iPAQ Setup Assistant software included on the Companion CD. The VoIP tab under Setup Assistant provides three categories of settings – Account, Server and VoiceMail.

Account:

Please provide the username and password for the VoIP/SIP account. Also provide the Domain name for the account. The username and Domain are used to construct the SIP URI for the account as in the example below:

User name- johndoe
Domain- voipservice.sip.com
SIP URI- sip:johndoe@voipservice.sip.com

The name of the VoIP Service Provider can also be specified; this is optional.

Server:

This category is for the SIP settings associated with the SIP-enabled IP-PBX or the SIP Server. The name/s or IP Address/es of the SIP Proxy and the SIP Registrar must be specified here. By default, ‘Register with SIP Proxy’ is checked, indicating that the SIP Proxy is also used as the SIP Registrar. In this case, the name or IP Address of the SIP Proxy only needs to be provided and the same name/IP address is also used for the SIP Registrar. If the Proxy and the Registrar are different, please uncheck ‘Register with SIP Proxy’ and provide the name/IP address for the SIP Registrar in addition to the SIP Proxy.

Voice Mail:

If a Voice Mail number is provided with the account, please include it here. This is optional.

The ‘**Options**’ button can be used to access some advanced VoIP settings. The UDP signaling method can be toggled between symmetric and asymmetric. Symmetric UDP signaling is used by default. An option to enable early UDP packets on RTP and RTCP ports is available. This enables opening of ports on a NAT allowing NAT traversal of traffic. A prefix digit can be added to outgoing phone numbers. For example, if ‘9’ is specified as the prefix digit, it is automatically pre-pended to all 11-digit, 10-digit and 7-digit phone numbers. By default, no prefix digit is used. The DSCP field in the IP header of voice packets contains a QoS number that determines the priority voice packets get over data packets in the network. By default, 56 is used as the DSCP QoS number. This can be modified to increase or decrease the priority of voice traffic.

Once all the appropriate information is provided, apply the configuration to the phone. If Wi-Fi is on and connected to an AP, the phone will register with the specified SIP-enabled IP-PBX or the SIP Server. If Wi-Fi is off, SIP registration will occur when the next time Wi-Fi is turned on and connected to an AP. The status of the ‘Internet Calling’ plug-in will either be ‘**Available**’ or ‘**Selected**’. ‘**Available**’ means that the phone has Wi-Fi access and is registered with the IP-PBX, but the next outgoing call will be over cellular. ‘**Selected**’ means that the phone has Wi-Fi access, is registered with the IP-PBX and that VoIP will be used for the next outgoing call. The Phone Dialer can be used to make and receive VoIP calls. The Internet Calling status can be toggled between ‘**Available**’ and ‘**Selected**’ by pressing the Action [Enter] button on the Internet Calling plug-in on the home screen.

If the status of the plug-in is ‘**No Service**’, it indicates that the registration was unsuccessful. This could be due to lack of Wi-Fi access, the IP-PBX being unavailable or some other failure.

9.4 VOIP DIAL PLAN

The dial plan for VoIP consists of dialing rules. These dialing rules are constructed using regular expressions. Please refer to the Windows Mobile 6 documentation for information on creating dialing rules. The dial plan is an XML file based on the OMA provisioning standard. The Voice Messenger comes with a default dial plan which may not necessarily work for a specific set-up. The dial plan will have to be modified and to do this, a base XML template will be provided. This should be used to add new dialing rules specific to the infrastructure being targeted. Once the XML template is modified to add new rules, it should be applied to the phone/s to provision the phone/s with the updated VoIP dial plan. This can be done using any of the provisioning methods supported by Windows Mobile 6 –

1. OTA using OMA Client provisioning [WAP push].
2. OTA using OMA Device Management provisioning. [OMA DM server is required].
3. Using a CAB Provisioning Format [.cpf] file which can be delivered over HTTP, or by using a SD/MMC card or copied over to the phone directly using ActiveSync.
4. Using Remote API [RAPI] in ActiveSync to push the provisioning XML file to the phone.
5. Application Developers can use the DMProcessConfigXML API to provision.

Please refer to the Windows Mobile 6 documentation for additional details on the provisioning methods listed above.

HP iPAQ Setup Assistant cannot be used to make any significant changes to the dial plan. It can only be used to add a prefix digit to 11-digit, 10-digit and 7-digit outgoing phone numbers. It is HIGHLY RECOMMENDED that any modifications/additions/deletions to the dial plan should be done using any of the provisioning methods mentioned above. HP iPAQ Setup Assistant MUST only be used to configure/modify SIP settings.

9.4.1 IMPORTANT NOTE ABOUT EMERGENCY CALLING

Emergency Calling over IP is not supported by the Voice Messenger. All emergency calls must be placed over cellular. To ensure this, a dialing rule specifically for emergency calling MUST be added to the VoIP dial plan. This rule will indicate what the emergency number is and will also indicate that this number should be placed on the cellular network only. The emergency dialing rule for the U.S is shown below:

```
<rule pattern='911' display='911' restrict='VoIP' />
```

The base template for the VoIP Dial Plan is specified below. It is HIGHLY RECOMMENDED that the rules included in the base template should not be removed.

```
<wap-provisioningdoc>

<characteristic type="VoIP">
<parm name="DialPlan" value="<dialplan xmlns='http://schemas.microsoft.com/embedded/VoIP'>

<dialplan-header>
        <host>#use_sipsrv_host_name#</host>
</dialplan-header>

<!-- Dial Plan rules -->

<!-- IP address rules -->
<!-- EQUIVALENT OF '\d{1,3}\.\d{1,3}\.\d{1,3}\.\d{1,3}' -->
<rule pattern='(\d|\d\d|\d\d\d)\.(\d|\d\d|\d\d\d)\.(\d|\d\d|\d\d\d)\.(\d|\d\d|\d\d\d)' restrict='Cell,SMS' />
<!-- EQUIVALENT OF '(\d{1,3})\*(\d{1,3})\*(\d{1,3})\*(\d{1,3})' -->
<rule pattern='(\d|\d\d|\d\d\d)\*(\d|\d\d|\d\d\d)\*(\d|\d\d|\d\d\d)\*(\d|\d\d|\d\d\d)' dial='\1.\2.\3.\4'
display='\1.\2.\3.\4' restrict='Cell,SMS' />

<!—Add Emergency Dialing Rules here  -->

<!—Add Other Dialing Rules here  -->

<!-- SIP URI rules -->
<!-- EQUIVALENT OF '[Ss][Ii][Pp][Ss]?:\w*(\d{3})(\d{3})(\d{4})@(.+)' -->
<rule pattern='[Ss][Ii][Pp][Ss]?:\w*(\d\d\d)(\d\d\d)(\d\d\d\d)@(.+)' display='(\1) \2-\3'
restrict='Cell,SMS' />
<rule pattern='([Ss][Ii][Pp][Ss]?:)?\w*([a-zA-Z0-9_-]+)@(.+)' display='\2' restrict='Cell,SMS' />
<rule pattern='[Ss][Ii][Pp][Ss]?:\w*([^@]+)' display='\1' restrict='Cell,SMS' />

<!—Catch All -->
<rule pattern='(\d+)' dial='sip:\1@$host$' display='\1' transfer='sip:\1@$host$' />
<rule pattern='([a-zA-Z0-9_-]+)' dial='sip:\1@$host$' display='\1' transfer='sip:\1@$host$' />

</dialplan>" />
</characteristic>

</wap-provisioningdoc>
```

A sample dialing rule for 10-digit phone numbers is shown below. This rule pre-pends a ‘9’ to 10-digit dialed phone numbers.

```
<rule pattern='(\d{3})\s*(\d{3})\s*-?\s*(\d{4})(\s*[Xx]\s*\d+)?' dial='sip:9\1\2\3@$host$'
display='(\1) \2-\3'
```

```
transfer='sip:\1\2\3@$host$' />
```

Please refer to the Windows Mobile 6 documentation for additional details about the VoIP Dial Plan and some examples.

9.5 ADDITIONAL CONFIGURATION

Windows Mobile 6 provides a set of registry keys to customize the WM6 SIP Client. Some of the key customization features are provided by HP iPAQ Setup Assistant as explained in section 9.3. The other customization options can be set by modifying the registry. Below are some examples of the configuration options provided using the registry –

- Enable/disable redirection of SIP calls.
- Allow/disallow listening for incoming SIP traffic on port 5060.
- Modify the registration expiry time (default is 19 seconds).
- Modify Session Timers, specifically ‘Session-Expires’ and ‘Min-SE’ (default is 90 seconds).

Please refer to the Windows Mobile 6 documentation for complete coverage of all available configuration options and the associated registry entries.

10 Appendix A – Standards Support

10.1 SIGNALING STANDARDS IMPLEMENTED

Standard or Reference	Title/ Decription
RFC 3261	SIP: Session Initiation Protocol
RFC 3261	Call Waiting
RFC 2327	SDP : Session Description Protocol
RFC 2246	TLS Protocol
RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers
RFC 3263 (obsoletes 2543)	SIP: Locating servers
RFC 2782 (obsoletes 2052)	DNS SRV
RFC 3264	An Offer/Answer Model with the Session Description Protocol (SDP)
RFC 3262	Reliability of Provisional Responses : Provisional Response ACKnowledgement (PRACK)
RFC 3361	DHCP Option for location the outbound SIP Proxy server
RFC 3265	Session Initiation Protocol (SIP)-Specific Event Notification (SUBSCRIBE/NOTIFY)
RFC 3842	A Message Summary and Message Waiting Indication
RFC 3863	Presence Information Data Format (PIDF)
RFC 2976	The SIP INFO Method
RFC 3428	Session Initiation Protocol (SIP) Extension for Instant Messaging
RFC 3711	Secure RTP (SRTP)
RFC 3856	Presence Event Package for the Session Initiation Protocol (SIP)
RFC 3515	The Session Initiation Protocol (SIP) Refer Method
RFC 3515	SIP: Refer Method/ Call Transfer
RFC 3323	Privacy Mechanism for SIP
RFC 3324	Short term requirements for Network asserted identity
RFC 3325	Private extensions to SIP of asserted identify within trusted networks
RFC 1321	MD5
RFC 2387	MIME content type
RFC 2605	RTCP in SDP
RFC 3841	Call Preferences
RFC 1321	MD5
RFC 2474	Diffserv
RFC 2617 (obsoletes 2069)	An Extension to HTTP: Digest Access Authentication

10.2 MEDIA/OTHER STANDARDS

Standard or Reference	Description
RFC 3550 (obsoletes RFC 1889)	RTP: A Transport Protocol for Real-Time Applications
RFC 3551 (obsoletes RFC 1889)	RTP Profile for Audio and Video Conferences with Minimal Control
draft-wing-behave-symmetric-rtprtcp-01.txt	Symmetric RTP/RTCP ports over UDP
RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals (in-band and out of band)
G711 u/A	Voice codec
RFC 2198	RTP Payload for Redundant Audio Data

10.3 STANDARDS NOT IMPLEMENTED

Standard or Reference	Decription
Draft-ietf-core-23	*XMPP core implementation*
Draft-ietf-xmpp-cpim-04	*Mapping of XMPP to CPIM*
RFC 3326	*The Reason Header Fields of the SIP*
RFC 4028	*Session Timers in the SIP*
RFC 3611	*RTP Control Protocol Extended Reports (RTCP XR)*
RFC 3389	*(RTP) Payload for Comfort Noise (CN)*
RFC 3311	*SIP Update Method*
Draft-ietf-sip-join-xx	*SIP Join header*
Draft-ietf-sipping-cc-conferencing-xx	*SIP Call Control - Conferencing for User Agents*
RFC 3891	*SIP 'Replaces' Header*
RFC 3266	*Support for IPv6 in Session Description Protocol (SDP)*
RFC 3581	*Symmetric Routing*
Call Park/Call Pickup	*Call Park/Call Pickup (through 'Replaces' header)*
RFC 3903 (Draft-ietf-sup-publish-04)	*Extension to SIP event notification framewirking for aggregation of notification under the same AOR?*
UPnP IDG	*Internet Gateway Device (IGD) Standardized Device Control Protocol V 1.0*
RFC 3047	*RTP Payload Format for ITU-T Recommendation G.722.1*
G.722.1	*Voice codec*
G.723	*Supported if a 3rd party codec is used*
G.723.1	*Supported if a 3rd party codec is used*

iLBC	*Supported if a 3rd party codec is used*
G.726	*Supported if a 3rd party codec is used*
G.729	*Supported if a 3rd party codec is used*
SIP Auth -AKA	*AKA - Auth based on USIM*
3GPP Specific RFCs/Headers	