Request for Information

for

User Agents for Pulver.com FWD services

Contents

1.1. Introduction

The editors of this document believe that most of the services features for SIP based communications and applications reside in the endpoints. As a consequence we would like to get information from developers of PC/laptop User Agent (UA) vendors, developers of SIP desktop phones and SIP mobile devices, such a PDAs and mobile phones about the implementation and availability of advanced SIP UAs.

Advanced SIP UAs have several key features that are at present not yet available in many implementations:

1. The ultimate user friendly experience for setting up the UA and using all features.

2. NAT traversal using ICE/STUN in the UA and deploying TURN servers when required. This enables communications in various NAT scenarios.

3. Use if the Internet Low Bite Rate Codec iLBC, echo and side tone control and automatic audio gain control for a superior user experience of Internet voice communications.

4. The use of cryptographically asserted identity and the SIP certificate service for prevention of spam and other attacks. Also, the use of SRTP for media encryption for confidentiality.

5. Dual functionality: Client-server (CS) SIP and peer-to-peer (P2P) SIP. P2P SIP can support the lowest cost SIP communications within an enterprise, without any servers and the without the associated cost for servers or PBX. P2P SIP can also support the lowest cost Internet consumer communications with good scalability. CS SIP will be used in business-business and consumer-business communications and will enable the use of SIP URLs, ENUM and gateway services to mobile networks and the PSTN. All SIP communication features envisaged by FWD can be supported by both CS SIP and P2P SIP.

Developers and implementers of SIP UAs are invited to submit information to mailto:UA@pulver.com.

1.2. Requirements for various service models

The table shows the summary of the communication features and applications for the FWD service using the PC/laptop, the PDA/wireless/mobile phone or the desktop SIP phone. The numbers in the left column show the major release versions:

Release Number	Features
1	Free UA for PC and PDA, free IP-IP service
2	Basic paid UA features and basic paid service
3	Advanced paid business services

Release number 1 is for free UAs for the PC/laptop and PDA. Releases number 2 and 3 are for more advanced paid UAs and advanced paid services.

1.3. Requirements for the FWD UAs[1]

Item		Summary	Rel.
1	Summary of Requirements	Best overall user experience	1
		100% standards based interoperability	1
		Several OS: Windows, Mac, PDA OS's	2
		High security	2
		Consumer and business communications	1
		Localization - easy switching of language	1
		Lowest cost	
2	UA Download	Single click download for PC/PDA	1
		Short download time/small footprint on HD	
3	Installation	Standard installation of UA and single click name and password choice.	1
		SIP phones must be configurable by user with the utmost ease. Simple procedures may be for example (1) going to a welcome account web page, (2) by pointing the PC browser to the displayed IP address of the phone or (3) by connecting the USB port of the SIP phone to the PC, so as to display the web page of the phone. The default SIP URI (sip:name@domain) is the same as for e-mail URI (mailto:user@domain).	
4	Upgrades	Automatic upgrades with user permission. Upgrades must keep previous user settings.	1
5	GUI/ Metaphor	Most intuitive for consumers	1
		Most functional for business	2

[1] CS: Client-Server, P2P: Peer-to-peer.

1.4. Communications and Applications

6	Presence	Presence with RPDIF attributes. XCAP data formats.	1/2
7	Media: IM	IM in page mode and MSRP mode. Use IM window for file sharing.	1/2
8	Media: Audio	iLBC as the default codec	1
		G.729 and G.711 for PSTN telephony gateways	1
		GIPS stereo wideband codec for conferencing	2
9	Media: Video	H.263 and optional H.264 CIF for 128, 256 and 512 kb/s	1
10	Media: ToIP	RFC 2793bis: RTP Payload for Text Conversation	2
11	Telephony Services CS and P2P	Presence, Redial, DND, Hold, Transfer, Conference, Mute, Call Waiting, Message Waiting. VM & e-mail integration. Emergency (911) and SOS support	2, 3 1
12	Conferencing/Collaboration CS and P2P	• IM/Audio/Video	1
		• Document sharing	2
		• Conference and Presentation modes	2
		• Floor control and floor display	2
		• Ad-hoc (w. presence) & scheduled	2
		• Roster support, call control and conference factory	
13	Desktop Sharing CS and P2P	• Web share (push)	2
		• White board sharing	2
		• Display of MS Office suite documents	3
14	File Transfer	Encrypted file transfer	2

1.5. Protocols and Security

16	DNS/ENUM	• DNS resolver	1
		• ENUM resolver	1
		• Registration service	2
17	SIP	Compliance w. list of applicable RFCs (Attached)	1
18	STUN and ICE	STUN client	1
		ICE client	1
19	P2P	Chord for P2P mode	2
20	Media Encryption	SRTP with MIKEY key distribution	1
21	Identity	Validation of Identity signatures	2
		SIP Certificate Service	
		Connecting SIP/VoIP islands using the registrar and identity	

1.6. References

The references chosen here represent the required subset of Internet standard documents that will be supported.

Sinnreich, H., Lass S., Stredicke C: " IP Telephony Device Requirements and Configuration" http://www.ietf.rg/internet-drafts/draft-sinnreich-sipdev-req-06.txt

Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M. and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

Sparks, R., "The Session Initiation Protocol (SIP) Refer Method", RFC 3515, April 2003.

Roach, A., "Session Initiation Protocol (SIP)-Specific Event Notification", RFC 3265, June 2002.

Sparks, R., "Internet Media Type message/sipfrag", RFC 3420, November 2002.

Johnston, A., Donovan, S., Sparks, R., Cunningham, C. and K. Summers, "Session Initiation Protocol (SIP) Basic Call Flow Examples", BCP 75, RFC 3665, December 2003.

Johnston, A., Donovan, S., Sparks, R., Cunningham, C. and K. Summers, "Session Initiation Protocol (SIP) Public Switched Telephone Network (PSTN) Call Flows", BCP 76, RFC 3666, December 2003.

Rosenberg, J. and H. Schulzrinne, "Reliability of Provisional Responses in Session Initiation Protocol (SIP)", RFC 3262, June 2002.

Rosenberg, J. and H. Schulzrinne, "Session Initiation Protocol (SIP): Locating SIP Servers", RFC 3263, June 2002.

Rosenberg, J. and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)", RFC 3264, June 2002.

Niemi, A., "Session Initiation Protocol (SIP) Extension for Event State Publication", RFC 3903, October 2004.

Campbell, B., Rosenberg, J., Schulzrinne, H., Huitema, C. and D. Gurle, "Session Initiation Protocol (SIP) Extension for Instant Messaging", RFC 3428, December 2002.

Rosenberg, J., "The Session Initiation Protocol (SIP) UPDATE Method", RFC 3311, October 2002.

Faltstrom, P., "E.164 number and DNS", RFC 2916, September 2000.

Peterson, J., "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323, November 2002.

Jennings, C., Peterson, J. and M. Watson, "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", RFC 3325, November 2002.

Mahy, R., Biggs, B. and R. Dean, "The Session Initiation Protocol (SIP) "Replaces" Header", RFC 3891, September 2004.

Arkko, J., Torvinen, V., Camarillo, G., Niemi, A. and T. Haukka, "Security Mechanism Agreement for the Session Initiation Protocol (SIP)", RFC 3329, January 2003.

Rosenberg, J. and H. Schulzrinne, "An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing", RFC 3581, August 2003.

Rosenberg, J., Schulzrinne, H. and P. Kyzivat, "Caller Preferences for the Session Initiation Protocol (SIP)", RFC 3841, August 2004.

Sparks, R., "The Session Initiation Protocol (SIP) Referred-By Mechanism", RFC 3892, September 2004.

Peterson, J., Liu, H., Yu, J. and B. Campbell, "Using E.164 numbers with the Session Initiation Protocol (SIP)", RFC 3824, June 2004.

Mahy, R., "A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)", RFC 3842, August 2004.

Mahy, R., "A Call Control and Multi-party usage framework for the Session Initiation Protocol (SIP)", draft-ietf-sipping-cc-framework-03 (work in progress), October 2003.

Rosenberg, J., "A Session Initiation Protocol (SIP) Event Package for Conference State", draft-ietf-sipping-conference-package-08 (work in progress), December 2004.

Rosenberg, J., "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)", draft-ietf-sip-callee-caps-03 (work in progress), January 2004.

Peterson, J., "Enhancements for Authenticated Identity Management in the Session Initiation Protocol (SIP)", draft-ietf-sip-identity-05 (work in progress), September 2004.

Mahy, R., "Connection Reuse in the Session Initiation Protocol (SIP)", draft-ietf-sip-connect-reuse-03 (work in progress), October 2004.

Rosenberg, J., "Obtaining and Using Globally Routable User Agent (UA) URIs (GRUU) in the Session Initiation Protocol (SIP)", draft-ietf-sip-gruu-02 (work in progress), July 2004.

Levin, O., "Suppression of REFER Implicit Subscription", draft-ietf-sip-refer-with-norefersub-00 (work in progress), January 2005.

Johnston, A. and R. Sparks, "Session Initiation Protocol Service Examples", draft-ietf-sipping-service-examples-07 (work in progress), July 2004.

Rosenberg, J., "An INVITE Inititiated Dialog Event Package for the Session Initiation Protocol (SIP)", draft-ietf-sipping-dialog-package-05 (work in progress), November 2004.

Sparks, R., "Session Initiation Protocol Torture Test Messages", draft-ietf-sipping-torture-tests-04 (work in progress), July 2004.

Sparks, R. and A. Johnston, "Session Initiation Protocol Call Control - Transfer", draft-ietf-sipping-cc-transfer-03 (work in progress), October 2004.

Petrie, D., "A Framework for Session Initiation Protocol User Agent Profile Delivery", draft-ietf-sipping-config-framework-05 (work in progress), November 2004..

Camarillo, G. and A. Roach, "Subscriptions to Request-Contained Resource Lists in the Session Initiation Protocol (SIP)", draft-ietf-sipping-uri-list-subscribe-02 (work in progress), January 2005.

Jennings, C. and J. Peterson, "Certificate Management Service for SIP", draft-ietf-sipping-certs-01 (work in progress), October 2004.

Wijk, A., "Framework of requirements for real-time text conversation using SIP.", draft-ietf-sipping-toip-00 (work in progress), October 2004.

RFC 2327: "SDP: Session Description Protocol" by M. Handley, V. Jacobson, IETF, April 1998

RFC 3999: "The TEL URI for Telephone Numbers" by H. Schulzrinne, IETF, June 2004.

A. Johnston, H. Sinnreich, A. Clark, and A. Pendleton, " SIP Service Quality Reporting Event" IETF Internet-Draft, October 2004, Work in progress.

RFC 3550: "RTP: A Transport Protocol for Real-Time Applications" by H. Schulzrinne, S. Casner, R. Frederick, V. Jacobson, IETF, July 2003.

RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control " by H. Schulzrinne and S. Casner, IETF, July 2003.

RFC 3761: "The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)" by P. Falstrom and M. Mealing, IETF, April 2004.

RFC 3952: "Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech" by A. Duric, S. Andersen, IETF, December 2004.

RFC 3611: "RTP Control Protocol Extended Reports (RTCP XR)" by T. Friedman, Ed., R. Caceres, Ed., A. Clark, November 2003.

RFC 3711: " The Secure Real-time Transport Protocol (SRTP)," by M. Baugher et al, March 2004

RFC 3830: "MIKEY: Multimedia Internet KEYing" by J. Arkko et al, August 2004

"Key Management Extensions for Session Description Protocol (SDP) and Real Time Streaming Protocol (RTSP)" by J. Arkko et al, draft-ietf-mmusic-kmgmt-ext-12.txt, November 2004.

RFC 2246: "The TLS Protocol Version 1.0" by T. Dierks and C. Allen, IETF, January 1999

RFC 3489: "STUN-Simple Traversal of UDP Through NAT" by J. Rosenberg et al. IETF, March 2003.

"Traversal Using Relay NAT" by J. Rosenberg, R. Mahy, and C. Huitema, IETF Internet-Draft, October 2004.

"Interactive Connectivity Establishment (ICE): A Methodology for Network Address Translator (NAT) Traversal for Multimedia Session Establishment Protocols" by J. Rosenberg, IETF Internet Draft, October 2004.

"Symmetric RTP and RTCP Considered Helpful" by D. Wing, IETF Internet Draft, October 2004.

RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals" by H. Schuzrinne and S. Petrack, IETF, May 2000.