Marketing Info Sheet 24.3 - WTL Products & SIP

What is SIP?

The VoIP industry was founded on the H323 protocol. The limitations, inconsistencies and complexities of H323 meant that a replacement was needed. The successor to H323 as a VoIP signalling protocol was SIP. SIP is for peer-to-peer connectivity: Internet conferencing, telephony, presence, events notification and instant messaging. SIP was developed within the IETF MMUSIC (Multiparty Multimedia Session Control) working group, with work proceeding since September 1999 in the IETF SIP working group.

WTL products use our own SIP stack which has benefited from many years of development, debugging and improvement so that it can now be described as fully “battle hardened”.

WTL’s SIP Implementation

WTL has always positioned its products as ‘any protocol in, any protocol out’ and this includes SIP support. WTL offers SoftSwitch products (check our Soft IVR product). Also see SoIP Gateway for SS7 to SIP.
SIP support fits smoothly into the WTL architecture:

- SIP coexists with all other supported protocols (for example, SIP & H323 on the same PVX SoftSwitch).
- SIP calls interwork with other protocols (for example, call comes in via SIP and exits via H323).
- NOP bandwidth saving benefits apply (because NOP operates at the IP packet level not at the signalling protocol level)
- SIP inherits all existing WTL applications (Pre-Paid, Callback, CDR generation etc)
- Note that the implementation is of a SIP client / Back to Back User Agent
- SIP proxy support using Kamailio (formerly OpenSER)

SIP Support Across The Product Range

SIP is supported on all WTL products: SoIP Gateway, PVx and the IPNx.

SIP Standards Compliance

The equipment is compliant with the following IETF standards:

RFC Number	Description	Supported: (Yes, No, Partial)	Notes
2327	Session Description Protocol	yes	
2543	User Agent	yes	
2617	HTTP Authentication	yes	
2806	URLs for Telephone Calls	partial	only tel: scheme
2833	for DTMF transmission	yes	
2976	~~The SIP INFO Method~~	yes	
3204 **(updated by RFC 3459)**	~~MIME media types for ISUP and QSIG Objects~~	no	
3261 **(obsoletes RFC 2543/ updated by RFC 3853,RFC 4320)**	Session Initiation Protocol	yes	
3262 **(obsoletes RFC 2543)**	~~Reliability of Provisional Responses in SIP~~	partial	we send PRACK, we don't request them
3263 **(obsoletes RFC 2543)**	~~SIP: Locating SIP Servers~~	yes	
3264	An Offer/Answer Model with the Session Description Protocol (SDP)	yes	
3265 **(obsoletes RFC 2543)**	~~SIP-Specific Event Notification~~	partial	only in the context of Refer
3310	~~Hypertext Transfer Protocol (HTTP) Digest Authentication Using Authentication and Key Agreement (AKA)~~	no	
3311	~~The Session Initiation Protocol UPDATE Method~~	yes	
3312 **(updated by RFC 4032)**	~~Integration of Resource Management and SIP~~	no	

3313	~~Private Session Initiation Protocol (SIP)Extensions for Media Authorization~~	no	
3319	~~Dynamic Host Configuration Protocol (DHCPv6)Options for Session Initiation Protocol (SIP) Servers~~	no	
3323	~~A Privacy Mechanism for the Session Initiation Protocol (SIP)~~	yes	
3325	~~Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks~~	yes	
3326	~~The Reason Header Field for the Session Initiation Protocol (SIP)~~	yes	
3327	~~Session Initiation Protocol Extension for Registering Non-Adjacent Contacts~~	no	
3329	~~Security Mechanism Agreement for the Session Initiation Protocol (SIP) Sessions~~	no	
3361	~~DHCP Option for SIP Servers~~	no	
3389	Real-time Transport Protocol (RTP) Payload for Comfort Noise	yes	
3420	~~Internet Media Type message/sipfrag~~	yes	
3428	~~Session Initiation Protocol Extension for Instant Messaging~~	no	
3486	~~Compressing the Session Initiation Protocol~~	no	
3515	~~The Session Initiation Protocol (SIP) Refer Method~~	yes	
3581	~~An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing~~	yes	
3608	~~Session Initiation Protocol~~	no	

	Extension Header Field for Service Route Discovery During Registration		
3666	Session Initiation Protocol Extension Header Field for Service Route Discovery During Registration	yes	
3840	Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)	no	
3841	Caller Preferences for the Session Initiation Protocol (SIP)	no	
3853	S/MIME Advanced Encryption Standard (AES) Requirement for the Session Initiation Protocol (SIP)	no	
3891	The Session Initiation Protocol (SIP) "Replaces" Header	no	
3892	The Session Initiation Protocol (SIP) Referred-By Mechanism	no	
3893	Session Initiation Protocol (SIP) Authenticated Identity Body (AIB) Format	no	
3903	Session Initiation Protocol (SIP) Extension for Event State Publication	no	
3911	The Session Initiation Protocol (SIP) "Join" Header	no	
3959	The Early Session Disposition Type for the Session Initiation Protocol (SIP)	yes	
3960	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)	yes	
3968	The Internet Assigned Number Authority (IANA) Header Field Parameter Registry for the Session	yes	

	Initiation Protocol (SIP)		
3969	The Internet Assigned Number Authority (IANA) Universal Resource Identifier (URI) Parameter Registry for the Session Initiation Protocol (SIP)	yes	
4028	Session Timers in the Session Initiation Protocol (SIP)	yes	
4040	RTP Payload Format for a 64 kbit/s Transparent Call	yes	

Features Supported

- Voice calls
- DTMF supported
  - RFC 2833 method
  - Cisco proprietary 'INFO' method
- Fax calls
  - G711 fallback
  - T38

SIP methods currently supported:

INVITE
REINVITE with SDP redirect
Codec support: G723.1, G726, G728, G729A, G711 Alaw/uLaw, GSM
RPORT (response)
SDP in provisional responses for early media
DTMF over RTP (RFC 2833)
REFER
REGISTER (client side)
proxy (client side)
UPDATE (for keepalive, no SDP redirect)
DTMF over INFO (Cisco method)
fax over G711 (fallback)

REGISTER (server side)
HTTP digest authentication (server side)
fax over T38 (media line in SDP)
SIP-I (SIP to SS7 mapping)
UPDATE (with SDP redirect)

SIP Proxy & Registrar Support
- Kamailio (formerly OpenSER) can be run within SoIP, PVx or IPNx
- Independent (not integrated) service
- Most SIP Proxy functions already handled by switch software
- Proxy main benefit is routing calls ~~to~~ SIP direct connect customers

SIP methods not planned but may be supported later:

reliable provisional responses (PRACK)

SIP FAQs

What Layer 4 protocols are currently supported and are they mandatory or optional? (UDP, TCP, TLS, SCTP)?
UDP only

Is UDP fragmentation for both send and receive supported?
Yes.

What Layer 4 protocols are planned for future releases and when are they likely to be rolled out?
TCP planned for Q4 07

What Uri's types are supported? (SIP Uri, Tel Uri without phone context, Tel Uri with phone context)
SIP Uri (assuming that user name is a phone number). Tel Uri planned to be added later.

Maximum length of SIP-URI supported?
26 characters

Is RFC 3261 SIP signalling supported?
Yes.

Are 7, 10, or 11 digit numbers sent as dialled (i.e. not converted to e.164)?
Yes.
Does the service require the support of P-RACK
No (will be added as part of SIP-T support).

What voice codecs are supported?
G723.1, GSM-FR, G726-32, G729, G711 A-Law, G711 u-Law

Can Silence Suppression be disabled?
Yes.

Does network provide ring back or does it require this to be locally provided
We can generate ring back (pass transparently from the SS7 network or generate locally within the SoIP gateway)

Info sheet

Changing the Way the World Communicates

Do you require early media capability (180/183 with SDP) in the CPE SIP endpoints?
We do not ~~require~~ early media capability but is preferred.

What type of Authentication used i.e. digest, with or without "qop=auth-int", etc.
CLI, IP, prefix or Digest without qop

What services/features are provided by the network, that requires feature support on the CPE in order to interoperate e.g. Message waiting indication, hold, transfer…
Only basic SIP features are required

Can a particular contact be removed through unregistration leaving all other registrations intact?
Unregister is supported

Is RFC 3323 Privacy header supported?
Yes.

Do you envisage operation behind a NAT? If so, do you provide SBC/ALG functionality for endpoints to traverse through NAT?
Yes, operation behind NAT is supported, can redirect RTP stream based on 'real' IP address. However, we recommend use of a STUN server. SBC functionality provided: we can have multiple IP connections (one private, one public) and switch RTP streams between.

How many numbers can you allocate per trunks? Can you describe the trunk account that we will be given? Is there for each number an equivalent PSTN number that we can dial to perform loop-back testing?
No limit. Up to 60 types of trunks are possible with different characteristics (local address binding, authentication etc). The loop-back question is a function of the number translation in the routing plan.

Do you use refresh on an existing call (RFC 4028, support for session timers)? Do they support UPDATE (RFC 3311)?
Session timer supported but not according to RFC. UPDATE supported for purpose of Session Timer.

Do you support Fax? If so, can Fax be offered at the beginning of the session? Or does fax transmission occur only in the middle of a call?
Fax supported by fallback to G711. Fax transmission occurs only in the middle of a call.

Do you support Invite with no SDP?
Yes.

Do you support ReInvite?
Yes.

If you support G729 do you support Annex B or Annex A or both? Do you send and accept attribute for Annex B or Annex A in SDP?
Both supported for send & accept.

Do you support RFC 2833 for DTMF tones?
Yes. DTMF can be sent inband with G.711 or as named events with G.711 and G729a

Can you receive full and compact SIP headers (ie. "From:" & "f:")
No

Support of rfc2543 call hold (ie. a=sendonly or c=0.0.0.0)
No

Can you receive RFC 3262 call hold (ie. a=sendonly)
No.

Supports RFC 3325 P-Asserted-Id header
No

Supports SIP Early Offer/Answer w/SDP (i.e. Request-URI w/SDP and 183 response w/SDP)
No

Supports SIP Diversion Header (draft-levy-sip-diversion-08)
No

Supports SIP Re-Invites for Transfers
No

Sends INVITEs with SIP URLs (not tel URLs)
No