TekSIP - SIP Registrar and SIP Proxy for Windows

TekSIP complies with RFC 3261, RFC 3263, RFC 3311, RFC 3581 and RFC 3891. It supports NAT traversal and ENUM. You can select IP address to be listened and default SIP endpoint for outgoing calls. You can also log session details into a log file and monitor active registrations and sessions in real-time. TekSIP has a built-in Presence Server *(SIP/SIMPLE based)*. TekSIP can be deployed as a signaling server for WebRTC based SIP phones.

TekSIP also supports UPnP IGD specification. If installed behind UPnP supported Internet gateway device *(ADSL router e.g.)*, TekSIP automatically detects if it is behind a new NAT gateway and the external IP address. All outgoing requests manipulated for NAT traversal. You do not need to add manual reverse mappings for SIP and RTP protocols.

TekSIP can optionally act as a B2BUA for incoming 3xx SIP responses. TekSIP Supports RADIUS Authentication *(RFC 2865)* and RADIUS Accounting *(RFC 2866)* with the method described in **draft-sterman-aaa-sip-00.txt** and **draft-schulzrinne-sipping-radius-accounting-00.txt** respectively. TekSIP runs as a Windows service.

TekSIP Settings / Service Parameters

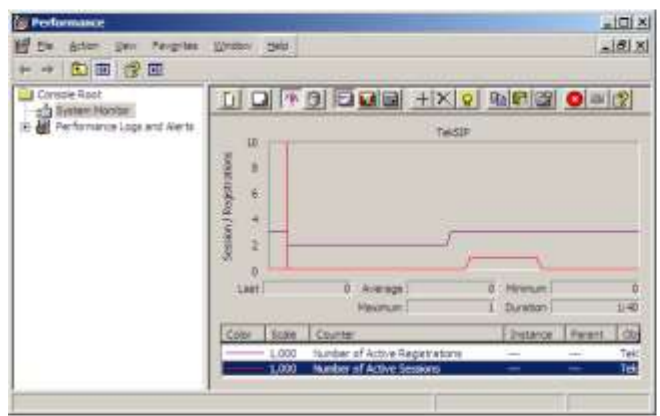
You can define dialed prefix-based routes n TekSIP; TekSIP can proxy all external calls for the provider accounts. TekSIP can register to upstream SIP servers to receive incoming calls.

You can re-direct calls to a voice mail server if user is unavailable to answer *(Busy or off-line)*. You can pick up incoming calls to registered extensions by dialing user defined call pickup prefix and extension number.

You can deploy TekSIP as a proxy for standard SIP phones to connect to a Microsoft Lync system. Lync supports IP phones only if they support SRTP with SIP TLS transport. TekSIP register standard SIP phones on behalf of them to a Lync system and maintain their presence status.

Active Sessions Tab

TekSIP provides many diagnostic messages when problems occur. You can see error messages on TekSIP Status bar or in the log file of TekSIP service. You can enable logging in Settings Tab. There are four levels of logging; None, Errors, Sessions and Debug. If you select Errors TekSIP logs just error messages. If you select Sessions both Session and Error messages will be logged. You have to save or apply settings changes if you change logging level setting. TekSIP log files can be found in the application directory.



TekSIP counters on
Windows Performance Monitor

TekSIP also utilizes Windows Performance Monitor, providing numerous counters. You can add and monitor them using Windows Performance Monitor *(Perfmon.exe)*.

TekSIP can act as a RTP Proxy and record audio streams if RTP proxy enabled. Recorded audio streams are saved in wave format can be played using TekSIP Manager. TekSIP can act as a WebRTC media proxy for SIP based WebRTC softphones. This enables WebRTC softphones to make calls to and accept calls from legacy SIP systems. TekSIP SP edition provides SRTP <-> RTP interworking with RTP proxy.

TekSIP monitors failed registration and call attempts from suspicious endpoints and blacklists them. You can monitor black listed endpoints through TekSIP Manager and you can remove black listed endpoints from quarantine list if required. You can also ban specific user agents.

TekSIP supports auto provisioning of IP phones based on SUBSCRIBE / NOTIFY PnP mechanism.

TekSIP can act as SMPP Gateway. Instant messages sent by registered SIP endpoints can be sent as SMS through an SMPP gateway and received SMS' can be routed to registered SIP endpoints as SIP messages.

System Requirements

1. A Windows system with at least 2048 MB of RAM.
2. Microsoft.NET Framework 4.6.1 *(Min.)*
3. 10 MB of disk space for installation.
4. Administrative privileges.

Supported SIP Standards

RFC 2617 HTTP Authentication: Basic and Digest Access Authentication
RFC 2782 A DNS RR for specifying the location of services (DNS SRV)
RFC 2976 SIP INFO Method
RFC 3261 SIP: Session Initiation Protocol
RFC 3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
RFC 3265 Session Initiation Protocol (SIP)-Specific Event Notification
RFC 3428 Session Initiation Protocol (SIP) Extension for Instant Messaging
RFC 3515 The Session Initiation Protocol (SIP) Refer Method
RFC 3581 An Extension to the Session Initiation Protocol (SIP) for Symmetric Response Routing
RFC 3711 The Secure Real-time Transport Protocol (SRTP)
RFC 3842 A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)
RFC 3856 A Presence Event Package for the Session Initiation Protocol (SIP)
RFC 3857 A Watcher Information Event Template-Package for the Session Initiation Protocol (SIP)
RFC 3863 Presence Information Data Format (PIDF)
RFC 3891 The Session Initiation Protocol (SIP) "Replaces" Header
RFC 3892 The Session Initiation Protocol (SIP) Referred-By Mechanism
RFC 3903 Session Initiation Protocol (SIP) Extension for Event State Publication
RFC 4568 Session Description Protocol (SDP) Security Descriptions for Media Streams
RFC 4480 RPID: Rich Presence Extensions to the Presence Information Data Format (PIDF)
RFC 4660 Functional Description of Event Notification Filtering
RFC 5262 Presence Information Data Format (PIDF) Extension for Partial Presence
RFC 5263 Session Initiation Protocol (SIP) Extension for Partial Notification of Presence Information
RFC 7118 The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP)

Kaplan Bilişim Teknolojileri Yazılım ve Ticaret Ltd.

Büyükdere Cd. 67-71 K:9 Mecidiyeköy, Şişli, 34387, İstanbul

Phone: +90 (212) 356 12 12, E-mail: info@kaplansoft.com / sales@kaplansoft.com