

***TekSIP***  
**Installation & Configuration Guide**  
**Version 2.9**

## Document Revision 3.2

<http://www.teksip.com/>

TekSIP is built by Yasin KAPLAN

**Read “Readme.txt” for last minute changes and updates which can be found under application directory.**

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## Introduction

TekSIP is a SIP Registrar and Stateless 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 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 also provides single accounts proxy; if you have just one provider account and many internal clients, TekSIP proxies all external calls for the provider account. You can have different provider account for different destinations (*Prefixes*). TekSIP can also register itself to provider SIP server if needed. You can receive incoming calls with registration. Please see “**Routing**” section for details.

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.

## System Requirements

TekSIP requires Microsoft .NET Framework 2.0 installed with the latest patches. Pentium IV class CPU with 512 MB of RAM is ideal for most configurations.

## Installation

Unzip “TekSIP.zip” and click “Setup.exe” comes with the distribution. Follow the instruction of setup wizard. Setup will install TekSIP Manager and TekSIP Service, add a shortcut for TekSIP Manager to desktop and the start menu.

## Configuration

Run TekSIP Manager from Start Menu / Program Files / TekSIP. TekSIP automatically configures itself at first run. TekSIP selects first available IPv4 address to listen and make a reverse lookup of that IPv4 address to obtain SIP domain information. If TekSIP cannot resolve selected IP address to an alphanumeric FQDN address, selected IPv4 address is used as SIP domain.

TekSIP also checks if it is installed behind an UPnP supported NAT gateway. If so, TekSIP automatically detects external IP and display it on status bar. TekSIP also adds a reverse mapping for incoming UDP connections automatically (*Default UDP port 5060*).

## Settings Tab

Click Settings Tab to start configuration. Settings tab has four sub sections.

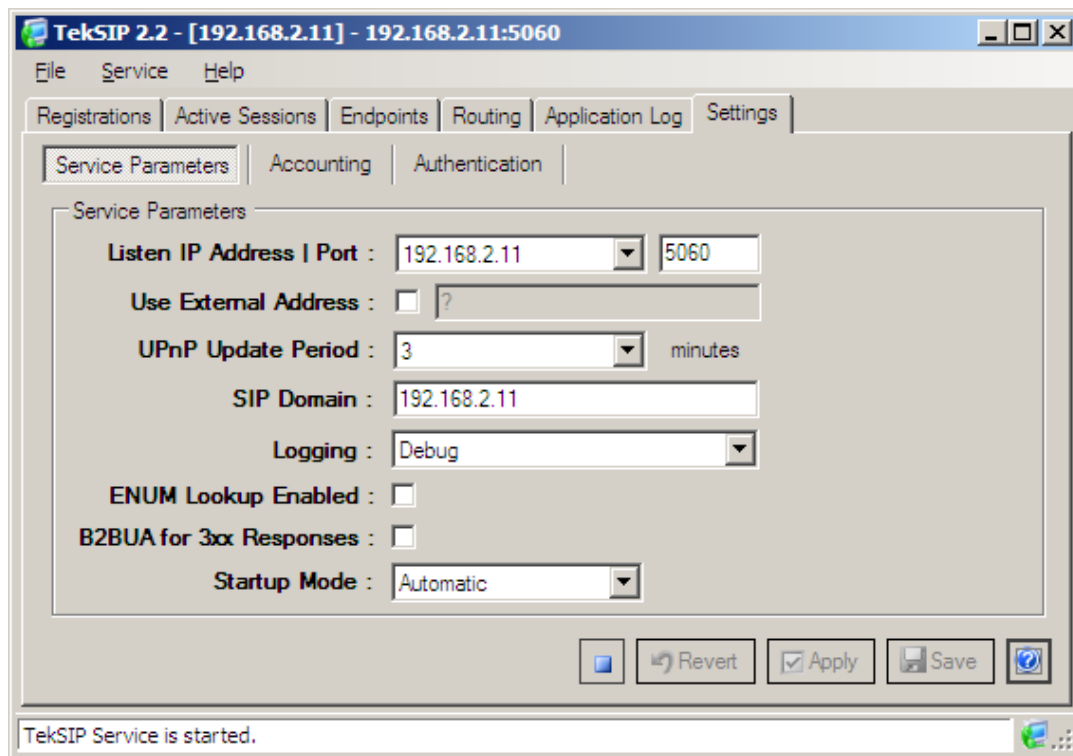


Figure - 1. TekSIP Settings / Service Parameters

Enter following information for Service Parameters:

- **Listen IP Address | Port:** You can select a detected IPv4 address configured on your system. When you change IP configuration of your system, the IP address list will automatically updated. You can define a port number to be listened (*Default 5060*).
- **Use External Address:** If TekSIP is installed behind a NAT gateway which does not UPnP you can set external IP address manually for NAT traversal. If your NAT gateway supports UPnP set UPnP Update Period to value greater than “0”.
- **UPnP Update Period:** You can specify period of querying UPnP Internet Access Gateway. Set “0” to disable UPnP support.
- **SIP Domain:** Enter a FQDN of your SIP domain. Please make sure that this address is resolvable by your SIP client and has a valid entry (*an A record*) in your DNS server. If you do not have an entry for your SIP domain in DNS you can simply use the IP address configured for listening incoming requests.

- **Logging:** Select logging level of TekSIP. Select “None” if you do not want logging, select “Errors” to log errors and select “Sessions” to log session information and errors. Log files are located under <Application Directory>\Logs directory.
- **ENUM Lookup Enabled:** TekSIP can resolve numbers in incoming SIP requests to a ENUM entry if exists. If TekSIP can not find a valid ENUM entry for the dialed number, SIP request will be forwarded to default route if it’s enabled. If a valid ENUM entry found for the dialed number, it is returned in a 302 response to originating endpoint by TekSIP. Call is forwarded to default route if ENUM lookup fails and default route is enabled. Visit [http://en.wikipedia.org/wiki/Electronic\\_Numbering](http://en.wikipedia.org/wiki/Electronic_Numbering) for detailed information on ENUM.
- **B2BUA for 3xx Responses:** If you wish TekSIP to handle 3xx responses select this option. When selected TekSIP processes 3xx responses and resends INVITE to the destination returned in 3xx response.

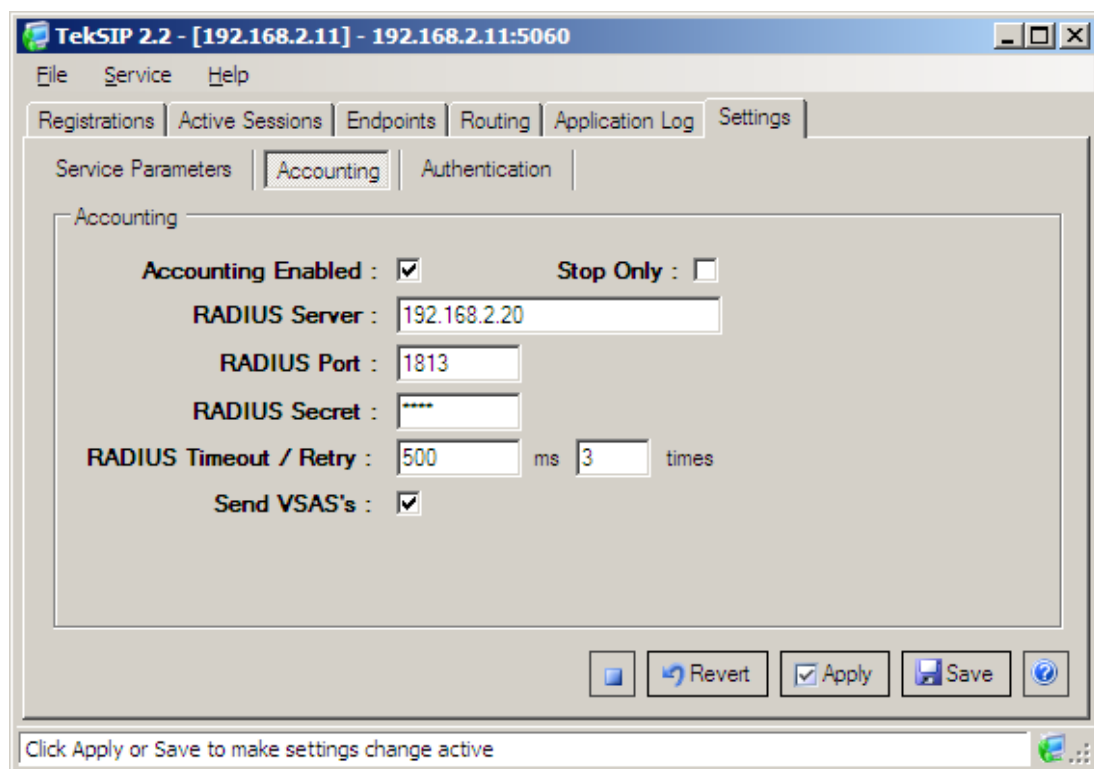


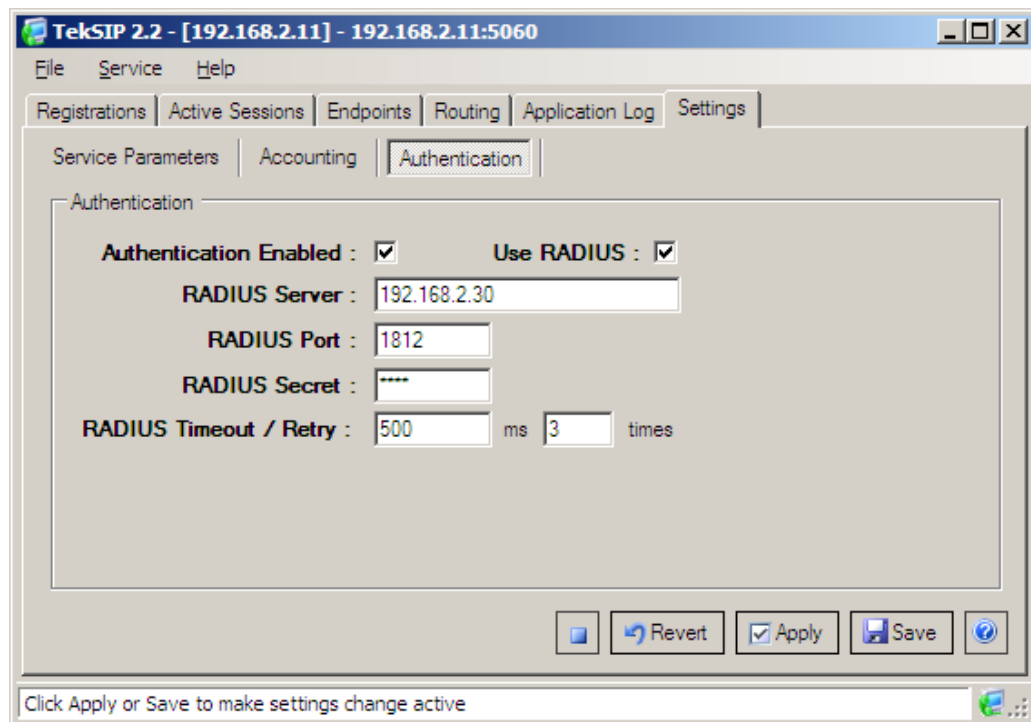
Figure - 2. TekSIP Settings / Accounting

Enter following information for Accounting:

- **Accounting Enabled:** TekSIP supports RADIUS accounting. RADIUS accounting is disabled by default. Click “Accounting Enabled” to enable RADIUS accounting.
- **Stop Only:** If you prefer to send only RADIUS Accounting stop messages to RADIUS server click check this option.
- **RADIUS Server:** Enter a valid IPv4 address for RADIUS server.
- **RADIUS Port:** Enter UDP port number of the RADIUS server. Default is UDP port 1813.
- **RADIUS Secret:** Enter RADIUS secret key for the RADIUS Server.
- **RADIUS Timeout / Retry:** You can set amount of time which TekSIP waits for a reply for the RADIUS accounting packets from the RADIUS Server. You can specify also how many

attempts will be made by TekSIP to deliver RADIUS accounting packets to the RADIUS server.

- **Send VSA's:** You can optionally send Cisco compatible VSA's for VoIP to RADIUS server in RADIUS accounting packets. Supported Cisco (*Vendor Id 9*) VSA's;
  - cisco-h323-conf-id (24)
  - cisco-h323-call-origin (26) [originate]
  - cisco-h323-call-type (27) [VoIP]
  - cisco-h323-disconnect-cause (30)
  - cisco-h323-gw-id (33)

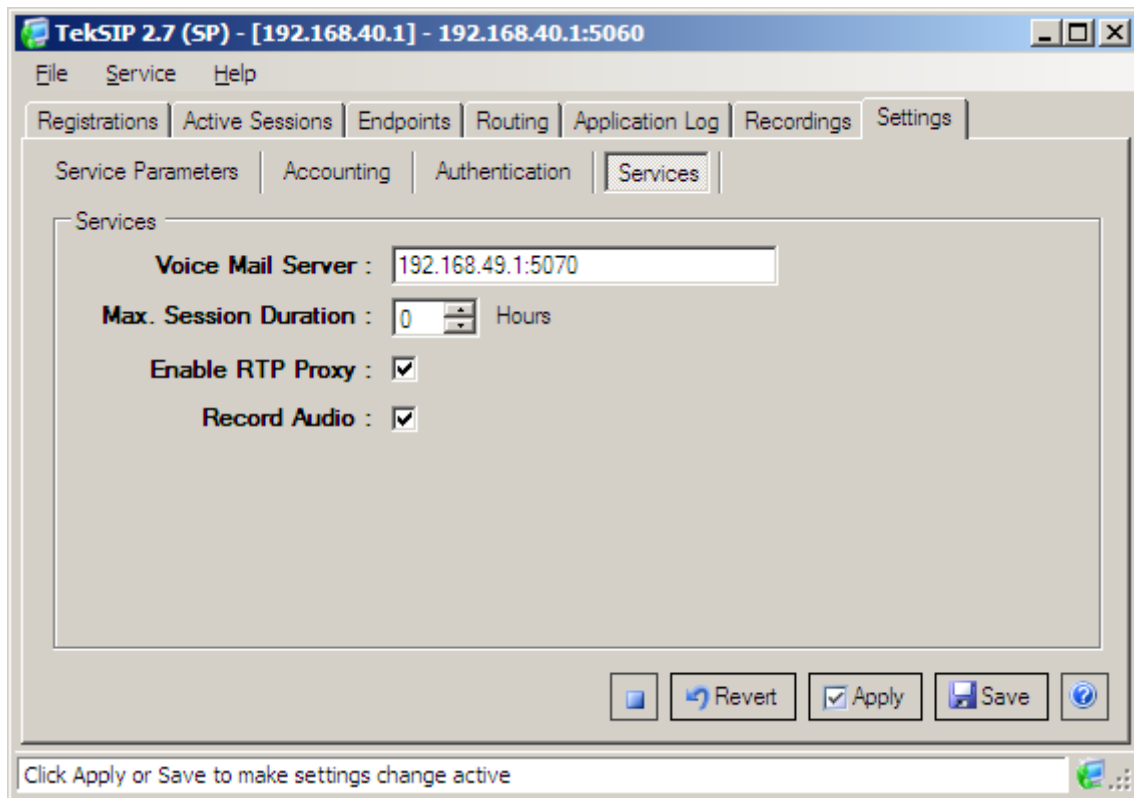


**Figure - 3.** TekSIP Settings / Authentication

Enter following information for Authentication:

- **Authentication Enabled:** SIP endpoint authentication is enabled by default. If you do not want to authenticate SIP registration and SIP requests uncheck this option.
- **Use RADIUS:** If you prefer to direct authentication requests to a RADIUS Server check this option. If you do not check this option TekSIP will use local endpoint database to authenticate the endpoints.
- **RADIUS Server:** Enter a valid IPv4 address for RADIUS server.
- **RADIUS Port:** Enter UDP port number of the RADIUS server. Default is UDP port 1813.
- **RADIUS Secret:** Enter RADIUS secret key for the RADIUS Server.
- **RADIUS Timeout / Retry:** You can set amount of time which TekSIP waits for a reply for the RADIUS accounting packets from the RADIUS Server. You can specify also how many attempts will be made by TekSIP to deliver RADIUS accounting packets to the RADIUS server.

You can re-direct calls to a voice mail server if user is unavailable to answer (*Busy or off-line*). Enter Voice Mail Server information to Voice Mail Server parameter at Settings / Services tab:



**Figure - 4.** TekSIP Settings / Services tab.

You can set which calls for a specific endpoint should be re-directed to voice mail server at Endpoints tab.

You can undo all settings changes by clicking [Revert] button. If you click [Apply] button setting changes will be applied and TekSIP will be re-started. If you click [Save] button setting will be saved to TekSIP.ini. You can start and stop TekSIP any time clicking service control button which is located left hand side of [Revert] button.

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 (*Recordings tab*). RTP recording can be performed only for G.711 A and mu law calls. If audio recording is enabled TekSIP will reject calls which does not use G.711 A and mu law codecs.

## Endpoints

You can define SIP endpoints using “Endpoints” tab. Enter SIP username to bottom leftmost textbox, enter password to the textbox right hand side of username entry. If you wish TekSIP to route incoming requests destined to this endpoint to another endpoint when it’s unavailable (*Off-line, busy...*), select endpoint to be used as alternative endpoint, select voice mail or leave “None”. You can set voice mail information at settings / services tab. Click “Add/Update” button to add new entry. If a valid entry is found with the same SIP username, that entry will be replaced or updated with the new entry. Click [Edit] button to edit existing entry or double click on the entry. Click

[Remove] button to remove a SIP endpoint. All SIP endpoint data is stored in TekSIP.mdb which is located under application directory. TekSIP clears expired registrations automatically.

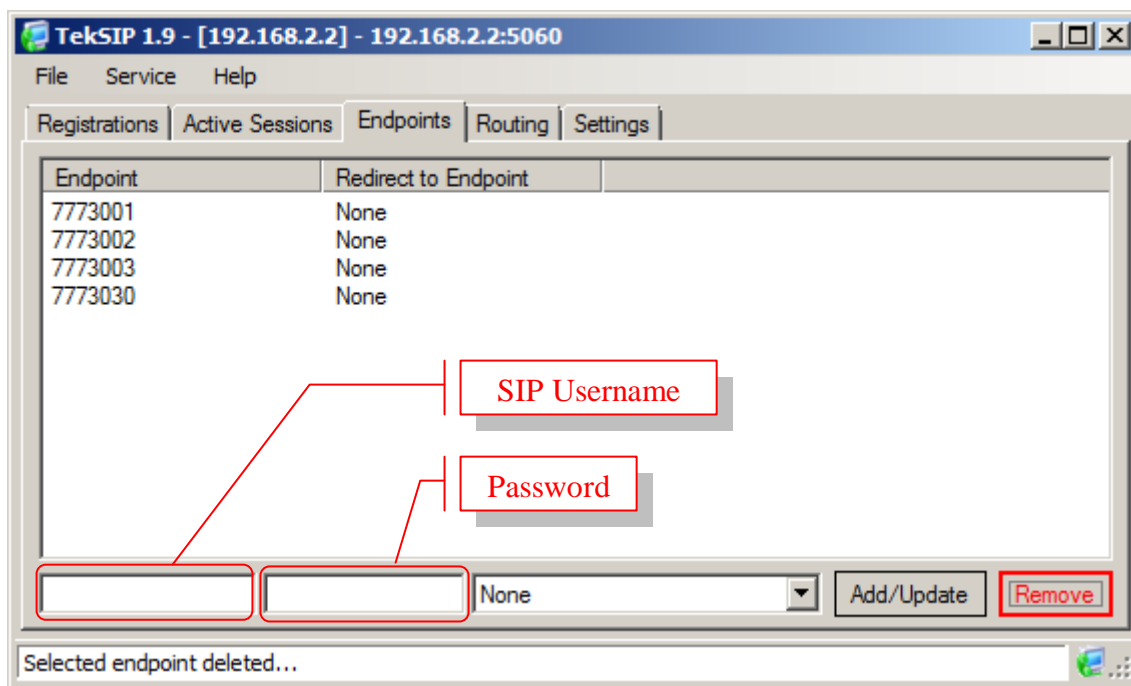


Figure - 5. Endpoints Tab

## Routing

You can define static routes to SIP endpoints through “Routing” tab. Enter a phone number prefix to bottom leftmost textbox, enter endpoint IP address to the textbox right hand side of prefix entry. You can also enter SIP port (*Default 5060*) used by the SIP Endpoint and Endpoint type (*Default SIP UA*).

You can have also a default route entry as shown the figure below. TekSIP chooses longest match prefix route. If any match can not be found default route is chosen if exists. ENUM lookup has precedence over static routes. If ENUM lookup fails, TekSIP consults to static routing table.

If the next hop configured for a phone prefix requires authentication, you can specify a username and password for the particular routing entry. If authentication is not required you can leave username and password fields blank.

TekSIP requests Proxy authentication for the incoming SIP requests from unregistered endpoints. But SIP request from the endpoints defined in routing table are not authenticated if the incoming SIP request destined to one of the defined endpoints in TekSIP’s endpoint database.

Enter a prefix and click “Add Route” button to add a new routing entry. You must edit at least Gateway entry to commit changes. You can specify a separate domain name if needed.

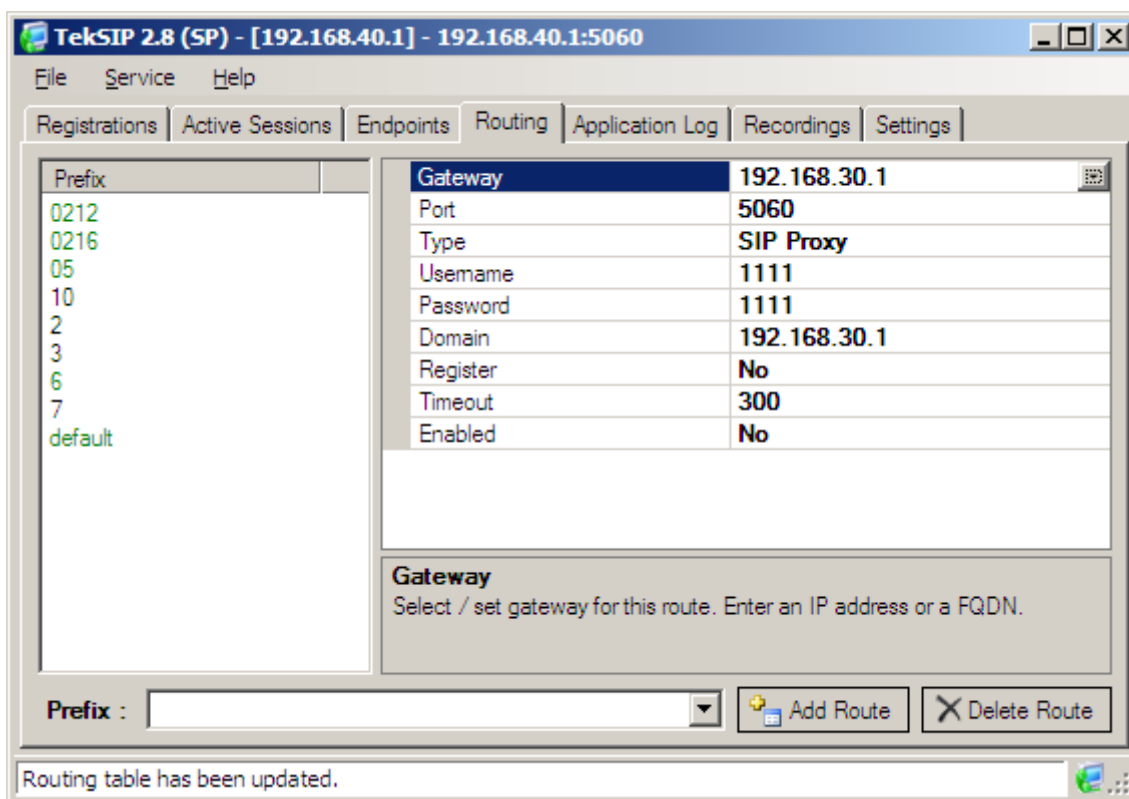


Figure - 6. Routing Tab

## Registrations

You can monitor active registrations through “Registrations” tab. You can unregister one entry clicking [Clear] button or all entries clicking [Clear all] button. If you unregister an entry, client must re-register itself. If you stop TekSIP service, all clients must re-register after re-starting TekSIP service.

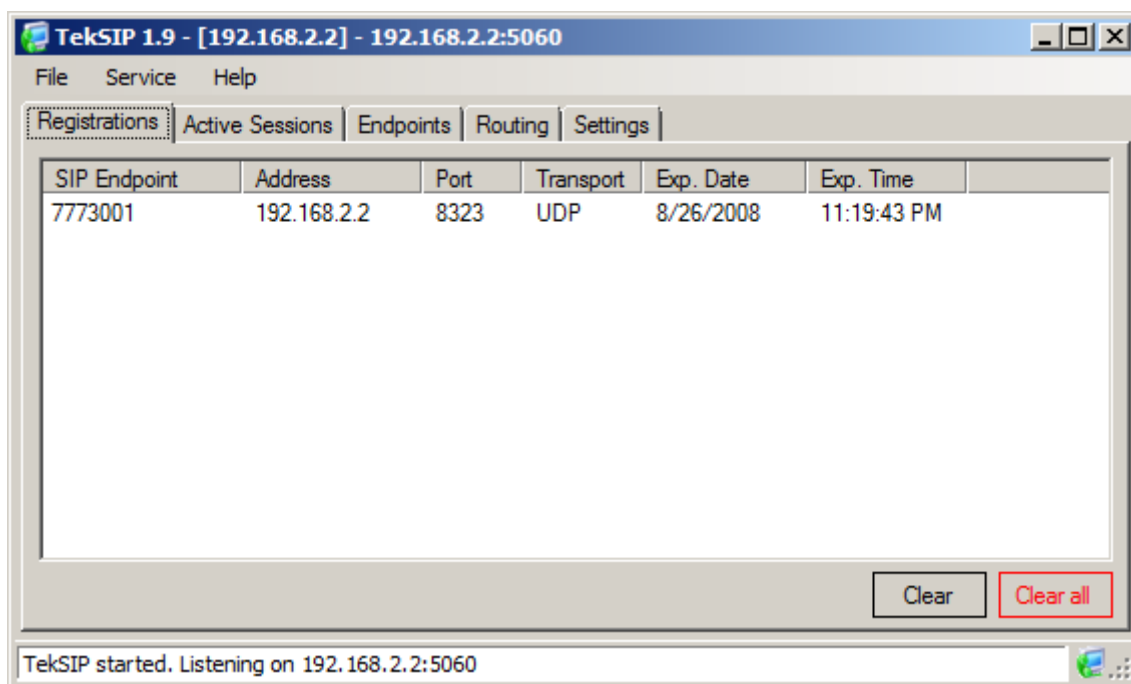


Figure - 7. Registrations Tab

## Active Sessions

You can monitor Active SIP Sessions through Active Sessions tab. Sessions can be terminated by clicking **Clear** or **Clear all** buttons.

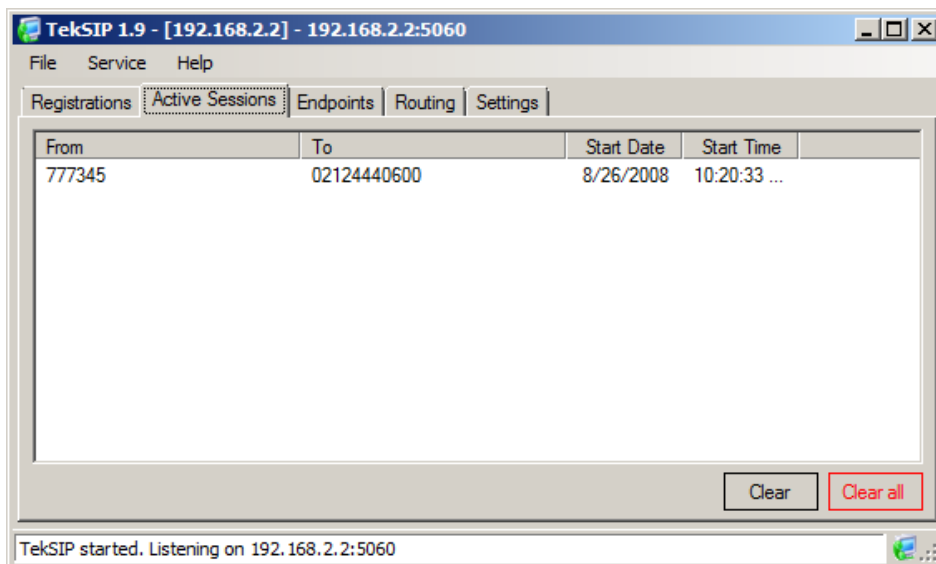


Figure - 8. Active Sessions Tab

## Application Log

You can monitor TekSIP service events from Application Log tab. Active SIP Sessions through Active Sessions tab. Session clearing functions just clear entries in the list box. When you clear a session you just remove the entry in the list box for that particular SIP session; if there is an active session between listed endpoints, the session stays active.

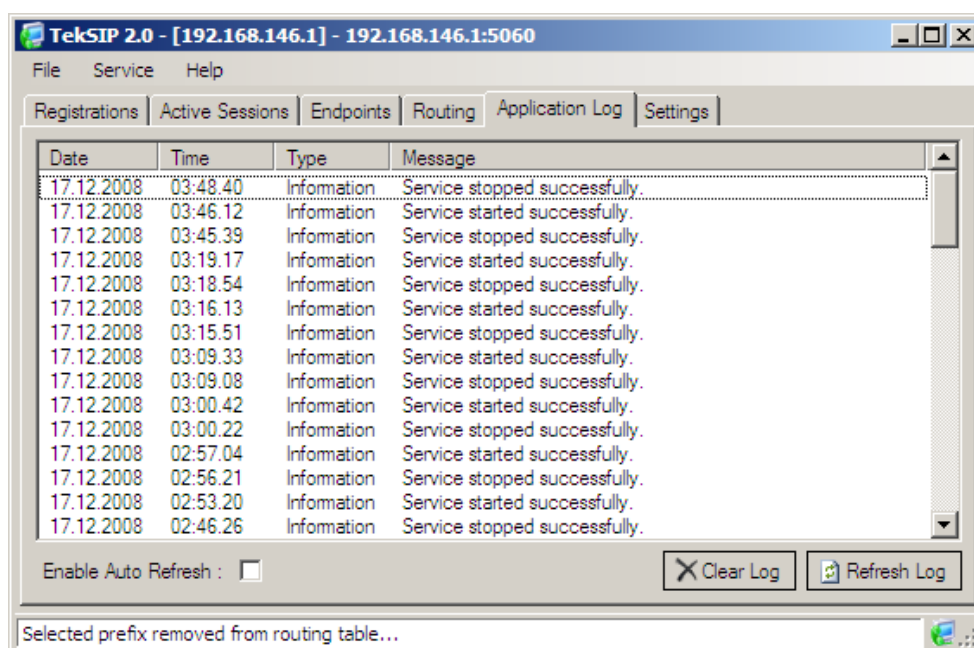


Figure - 9. Application Log Tab

## Starting TekSIP

Click “Service” menu and select “Start” to run TekSIP after making necessary configuration and saving configuration. If service starts successfully you will see “TekSIP Service is Started” message at bottom left message section of TekSIP Manager. Optionally you can start/stop TekSIP using the button on Settings tab. When you make any change(s) in configuration, TekSIP will ask you if you wish to restart TekSIP to make settings changes active if TekSIP service is running.

If TekSIP service can not start please examine Application Log tab as well as TekSIP log file under <Application Directory>\Logs if you were enabled logging in “Settings/Service Parameters”.

## Troubleshooting

TekSIP provides many 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 three levels of logging; None, Errors, Sessions. 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. Log files are located under <Application Directory>\Logs directory.

TekSIP also utilizes Windows Performance Monitor providing two counters; one for number of active registrations and other for number active sessions. You can add and monitor them using Windows Performance Monitor (*Perfmon.exe*).

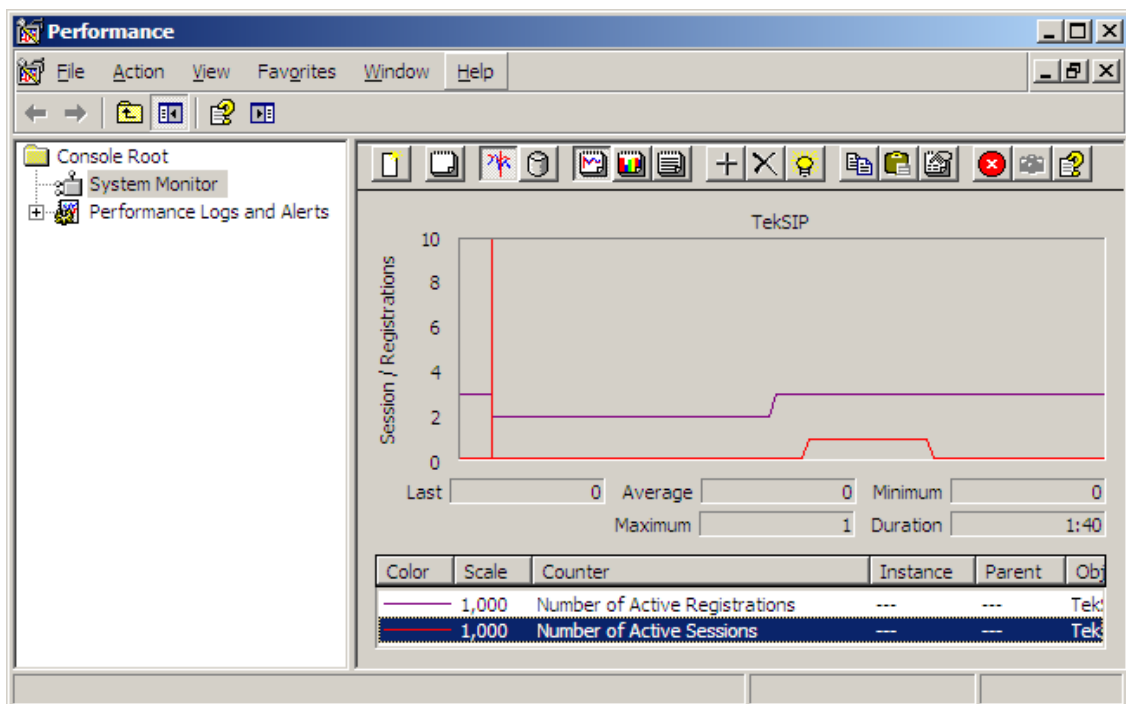


Figure - 10. TekSIP counters on Windows Performance Monitor

## TekSIP Messages

**TekSIP started. Listening on x.x.x.x.**

This message notifies that TekSIP service is started.

**Listened IP Address is being changed from x.x.x.x to y.y.y.y.**

TekSIP has detected a change in system's IP configuration and automatically changed listened IP address. You might change or remove the IP address configured for listening.

**Endpoints could not be loaded.**

TekSIP can not find or read "TekSIP.mdb" which is located under application directory. Please make sure that this file exists, not corrupted and not exclusively opened by another application.

**Settings could not be loaded. Initializing with default values.  
TekSIP Service is being started with default values on : x.x.x.x**

You get this message at first run of TekSIP. If TekSIP can not find or read TekSIP.ini initialize itself with default settings.

**Unable to initialize UDP/TCP thread [x.x.x.x:5060]**

If another application is configured to use same UDP/TCP port (5060) with TekSIP, TekSIP can not initialize respective thread.

**Default route points to this host**

You can not specify listened IP address as default route.

**New setting(s) applied and activated. Check default route.**

There is a problem with the IP address or FQDN of the default route.

**Can not apply changes; enter minimum configuration**

There is missing configuration data.

**Endpoint 'abc' added, but could not saved.**

There is a problem with the TekSIP.mdb file. It may be opened by another application or required database tables are missing.

**You can not redirect an endpoint to itself.**

You can not re-direct an endpoint to itself.

**Invalid endpoint information or illegal character detected in entries.**

Invalid characters found in SIP username or entry. You can just use numeric characters in SIP username entries. You can not use “;” (*Semicolon*) character in password entries.