SIP SERVER SDK v2.0

CONFIGURATION MANAGER
USER MANUAL v1.0

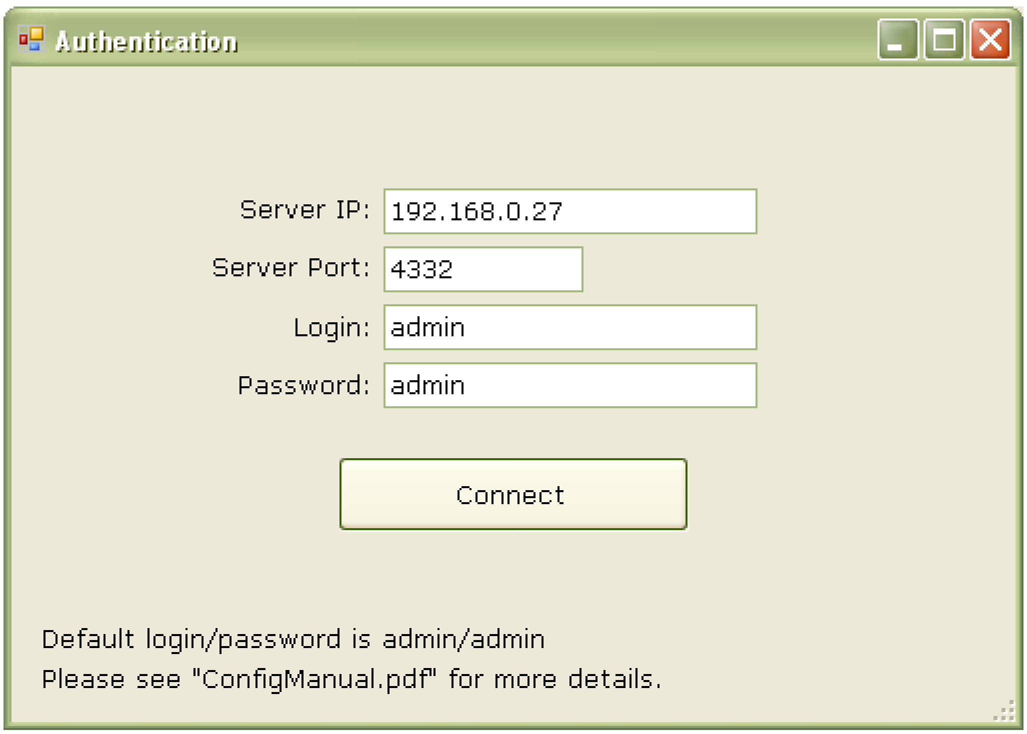
CONFIGURATION MANAGER

Configuration Manager is a graphical interface to remotely access and configure VaxTele integrated SIP server (sample code & demo application).

With Configuration Manager, you can start and stop SIP server, add or remove users, create queues, add or remove dialplans and many more.

1. Execute VaxTele SIP server demo or sample code.
2. Execute configuration manager.
3. Connect configuration manager to the SIP server.

AUTHORIZATION WINDOW



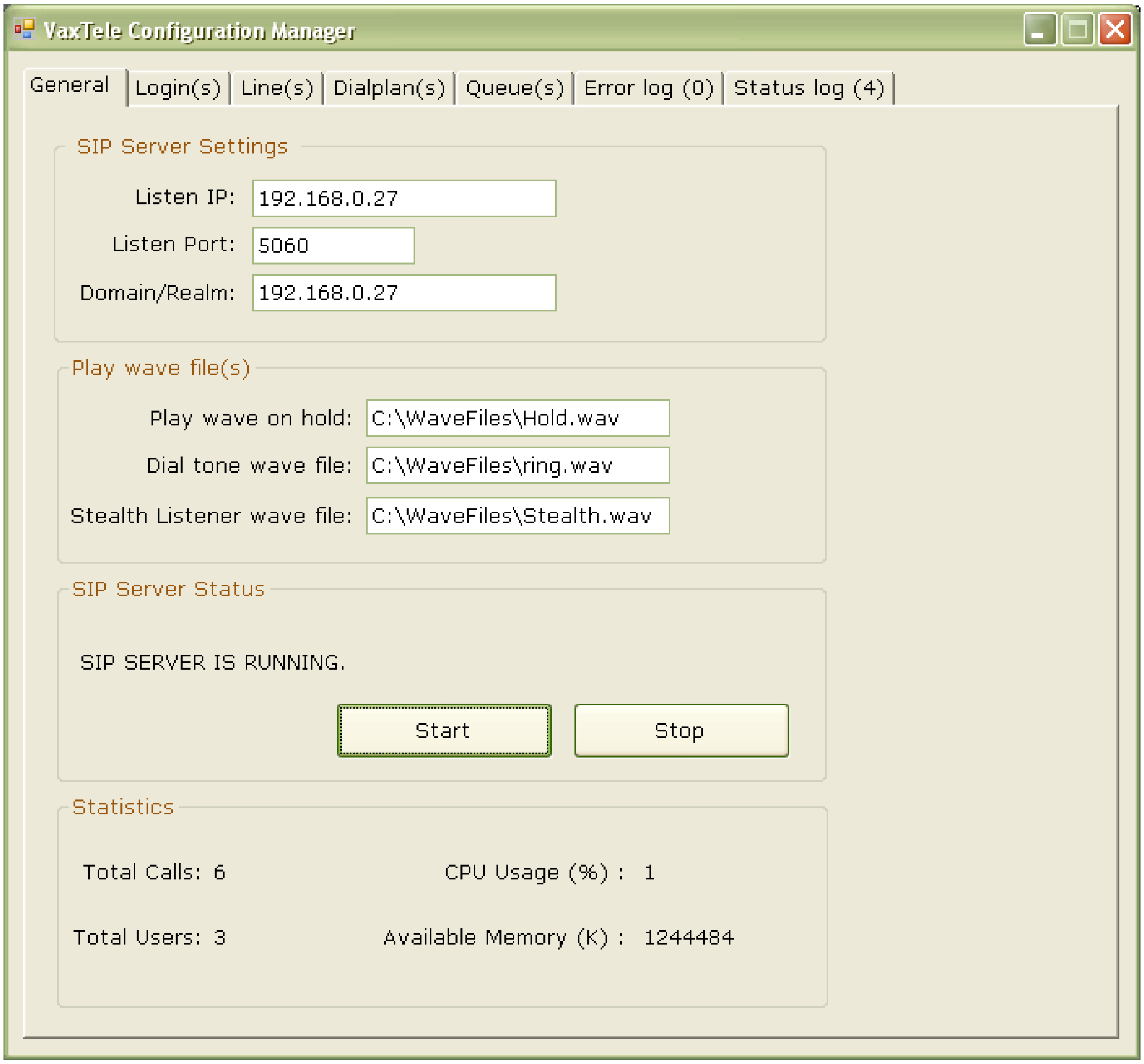
Enter the IP address of the computer on which SIP server is running, enter the port number, login and password and then click connect button.

Default login is 'admin' and password is 'admin'

Configuration manager connects to VaxTele SIP Server (sample code or demo application) on TCP/IP port (4332).

Connect configuration manager to SIP server on telnet port then you can easily start, stop and configure the SIP Server.

GENERAL TAB WINDOW



Listen IP is the IP assigned to the computer on which SIP Server is running. If that computer has multiple IP addresses then enter the single IP to which you want to bind the SIP Server.

Listen Port is the port on which SIP Server receives incoming SIP requests. Standard SIP port is 5060.

Domain/Realm can be any domain like: *sip.vaxtele.com or sip.abc.com* or it can be an IP address. Enter any text value and users use it as *Domain/Realm* to configure the SIP clients *(softphone, ATA, wifi phone, hardphone etc)*. Please see **LOGIN TAB WINDOW** for further details.

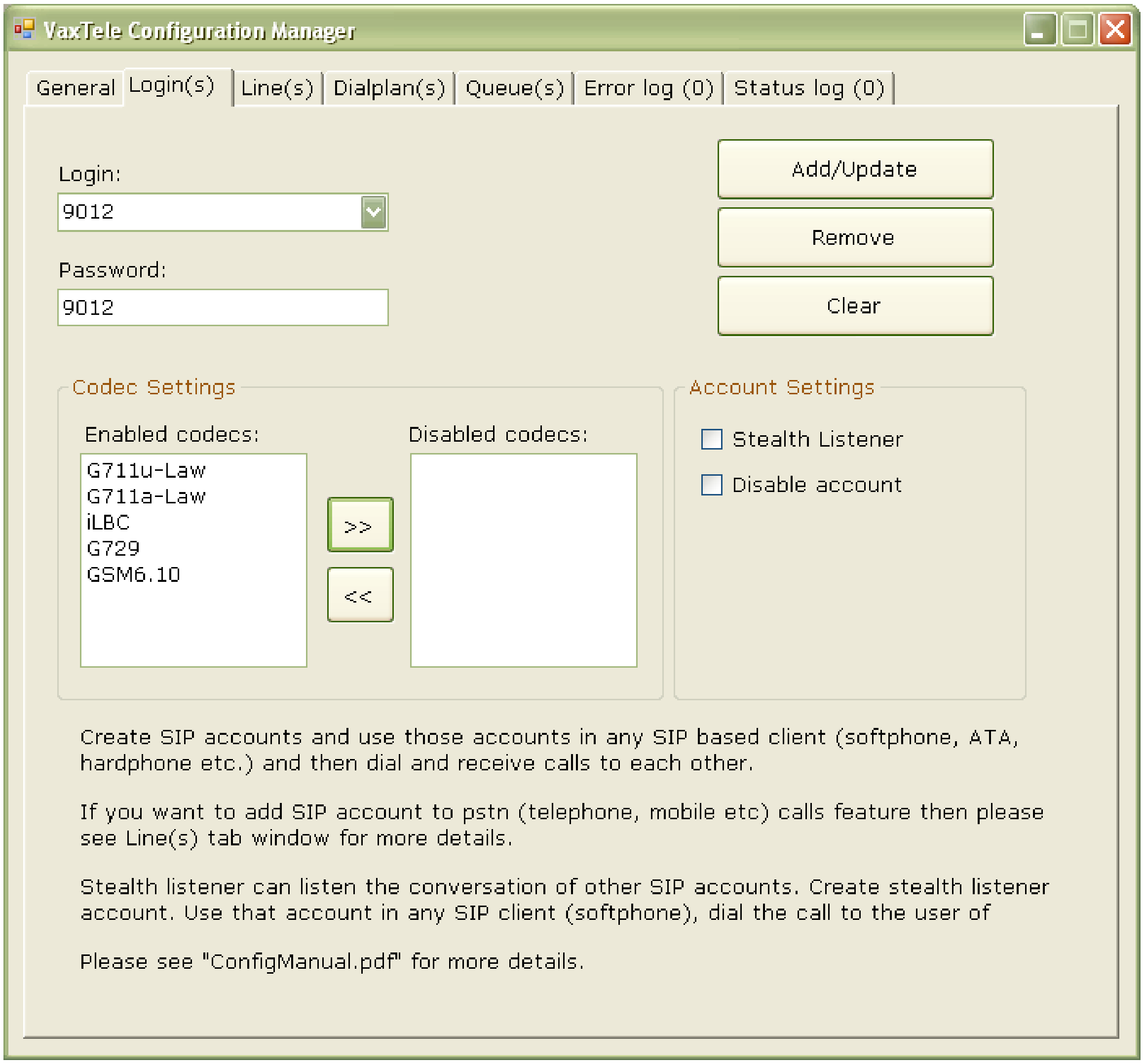
General tab window shows SIP server status and other settings. If you make any change on **GENERAL TAB WINDOW** then stop and start the SIP server again.

VaxTele support only uncompressed wave (.wav) files of sound quality (8000Hz, 16bit, Mono) such wave files can easily be created or converted from other formats by using Microsoft's sound recorder.

VaxTele does not support CD-Quality media files (MP3, wmv etc.) These files require CD-Quality to Telephony-Quality media conversation and such conversion change the voice quality and put a high load on the CPU.

VaxTele only works and play uncompressed (8000Hz, 16bit, Mono) wave file (.wav) to save CPU cycles and put minimum load on the CPU processing and increases the server's efficiency.

LOGIN TAB WINDOW



LOGIN tab window is used to add, remove, update and block SIP accounts.

Stealth listener is used to listen the conversation of any SIP account. In call-centers, administrators/managers use this feature for training and quality assurance purposes.

SIP softphones and other SIP client devices require the following SIP account settings to register/connect to the SIP servers.

- DisplayName
- Authorization UserName
- Domain/Realm
- UserName
- Authorization Password
- SIP Proxy

HOW TO CREATE AND USE SIP ACCOUNT

1. Enter Login (e-g: 9013)
2. Enter password 123
3. Select all codecs.
4. Click Add/update button.

Once the account is created then that SIP account can be used in any SIP softphone, hardphone, ATA and SIP client devices.

Execute any softphone and configure it by using the following SIP account settings.

- 9013 *(Displayname)*
- 9013 *(UserName)*
- 9013 *(Authorization UserName)*
- 123 *(Authorization Password)*
- 192.168.0.27 *(Domain/Realm value on GENERAL tab window)*
- 192.168.0.27 *(ListenIP value on GENERAL tab window)*

Use the settings in softphone and softphone will register to the SIP server without any problem.

HOW TO CREATE AND USE STEALTH LISTENER SIP ACCOUNT

1. Enter Login (e-g: 9090)
2. Enter password 123
3. Select all codecs.
4. Click “Stealth listner” check box.
5. Click Add/update button.

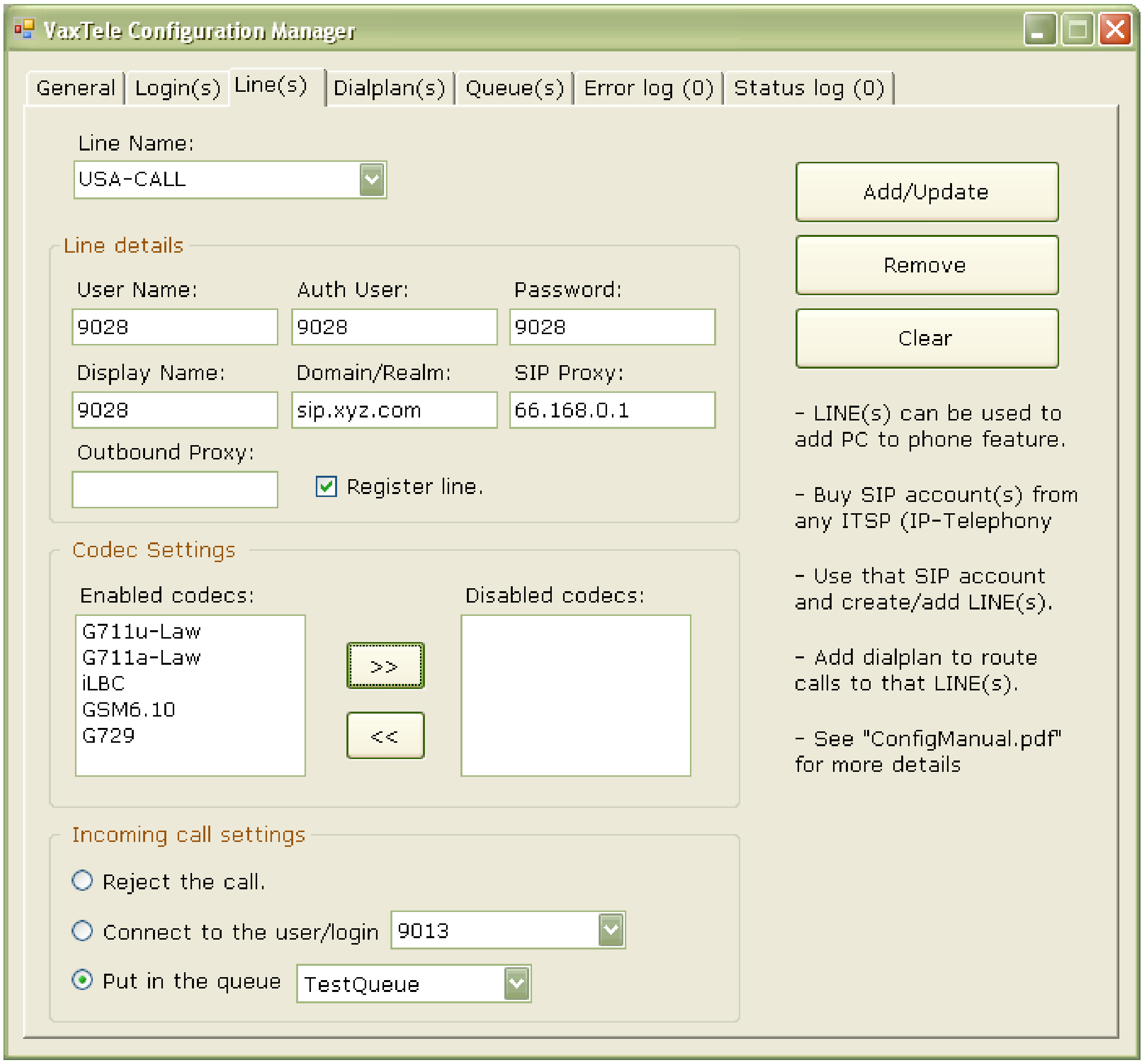
Once the account is created then use those SIP account settings in any softphone and register it to the SIP server. Once then strealth listener account 9090 registers successfully then dial the call to the user of which you want to listen the conversation.

For Example

Add stealth listener SIP account 9090 settings in any softphone and register it to the SIP server. Dial call to login 9013 to capture and listen all the outgoing and incoming call of 9013.

So, SIP Server (demo or sample code) waits for SIP account 9013 to dial/receive any call. When 9013 dials a call then SIP Server adds stealth listener 9090 to 9013 & dialed party conversation and in that way stealth listener 9090 joins the conversation and starts listening it.

LINE TAB WINDOW



LINE tab window is used if you want to provide softphone to PSTN *(telephone, mobile)* calls feature.

There are many SIP based IP-Telephony Service Providers (ITSP) are available on the internet, some of them are;

- www.broadvoice.com
- www.voipvoip.com
- www.inphonex.com
- www.verizon.com
- www.voxbone.com

Buy IP-Telephony service from them and they provide SIP account settings, simply use those SIP account settings in **LINE TAB WINDOW** and calls can be routed to those ITSP by adding proper dialplans. Please see **DIALPLAN TAB WINDOW** for further details.

NOTE: ITSP provides SIP account settings, first test those settings directly by using any softphone. Dial and receives phone calls with softphone, just to make sure that settings are working properly and then use those settings in SIP server.

So, here is the simple procedure:

1. Buy SIP account settings from ITSP *(IP-Telephony service provider)*.
2. Add those settings in **LINE TAB WINDOW** and create a line.
3. Add a dialplan to route incoming calls from any SIP client (user) or softhone to that line. e-g: if user dials 00 as prefix to the dial number then dial that number by using any specific line.

You can also put the incoming call from a specific LINE to the queue. Please have a look at **QUEUE TAB WINDOW** for more details.

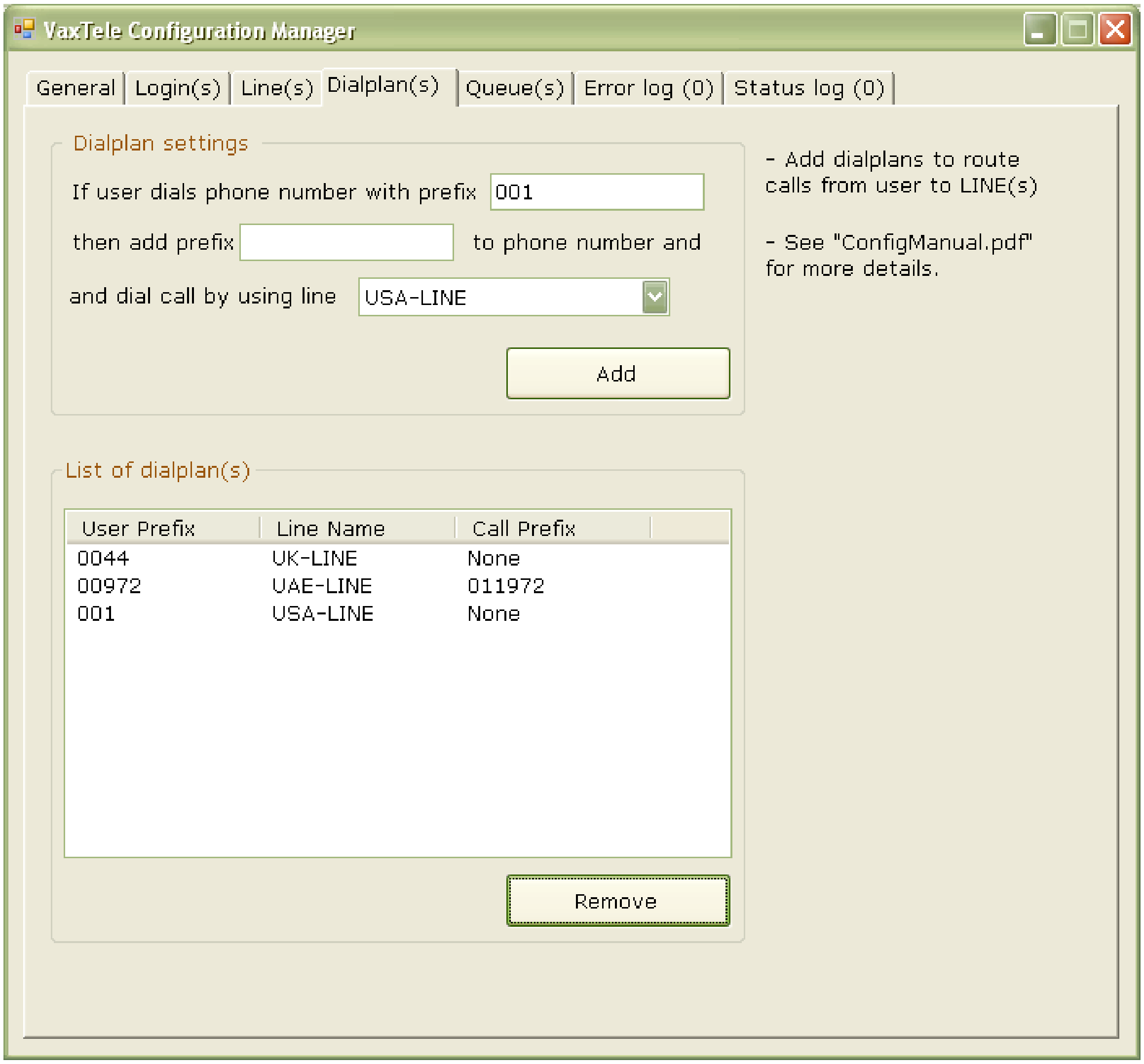
You can add multiple LINEs and route calls to those lines by adding multiple dialplans. Suppose you can add two LINEs one for USA and other for UK. In the **DIALPLAN TAB WINDOW** simply add dialplans;

- If user using softphone or SIP device dials phone number with prefix 001 then dial the phone number by using USA-LINE.

- If user using softphone or SIP device dials phone number with prefix 0044 then dial the call by using UK-LINE

00 prefix is used for international call dialing and 1 is USA country code, 44 is UK country code.

DIALPLAN TAB WINDOW



DIALPLAN tab window is important to route the incoming calls from any SIP account/user to the specific LINE. Please see **LINE TAB WINDOW** for more details about lines.

You can add dialplan, if user dials phone number with prefix 001 then use line or service provider 'USA-CALL' to route calls to USA.

Suppose, any user using softphone, register/connect to the SIP server *(sample code or demo application)* by using SIP account *(please see* **LOGIN TAB WINDOW** *for more details about SIP accounts)* and dials phone number 001614xxxxxxx.

VaxTele SIP Server *(sample code or demo application)* simply checks that dialed phone number and if the prefix is 001 then it dials the same phone number 001614xxxxxxx by using LINE or service provider 'USA-CALL'. Call gets connected and voice conversation starts between user *(using softphone)* and the dialed party *(using telephone or mobile)*.

You can also add multiple lines and add multiple dialplans to route calls to those lines.

Suppose, you find a service provider ABC for UK, who provides cheap calls to UK and service provider XYZ who offers cheap calls to USA then you can route calls to those service providers by adding two different dialplans.

- If user using softphone or SIP device dials phone number with prefix 001 then dial the call by using USA-LINE.

- If user using softphone or SIP device dials phone number with prefix 0044 then dial the call by using UK-LINE

00 prefix is used for international call dialing and 1 is USA country code, 44 is UK country code.

Some of the service providers require that add different prefix to dial calls to a specific country.

Suppose, there is another service provider DEF, who offers low rate calls to UAE/Dubai and you buy the SIP account from that service provider and add it as LINE. DEF service provider requires that add prefix 011 to dial calls to Dubai/UAE.

Then you can add the following dialplan in **DIALPLAN TAB WINDOW**.

- If user using softphone or SIP device dials phone number with prefix 00971 then add prefix 011971 to dialed phone number and dial the call by using UAE-LINE

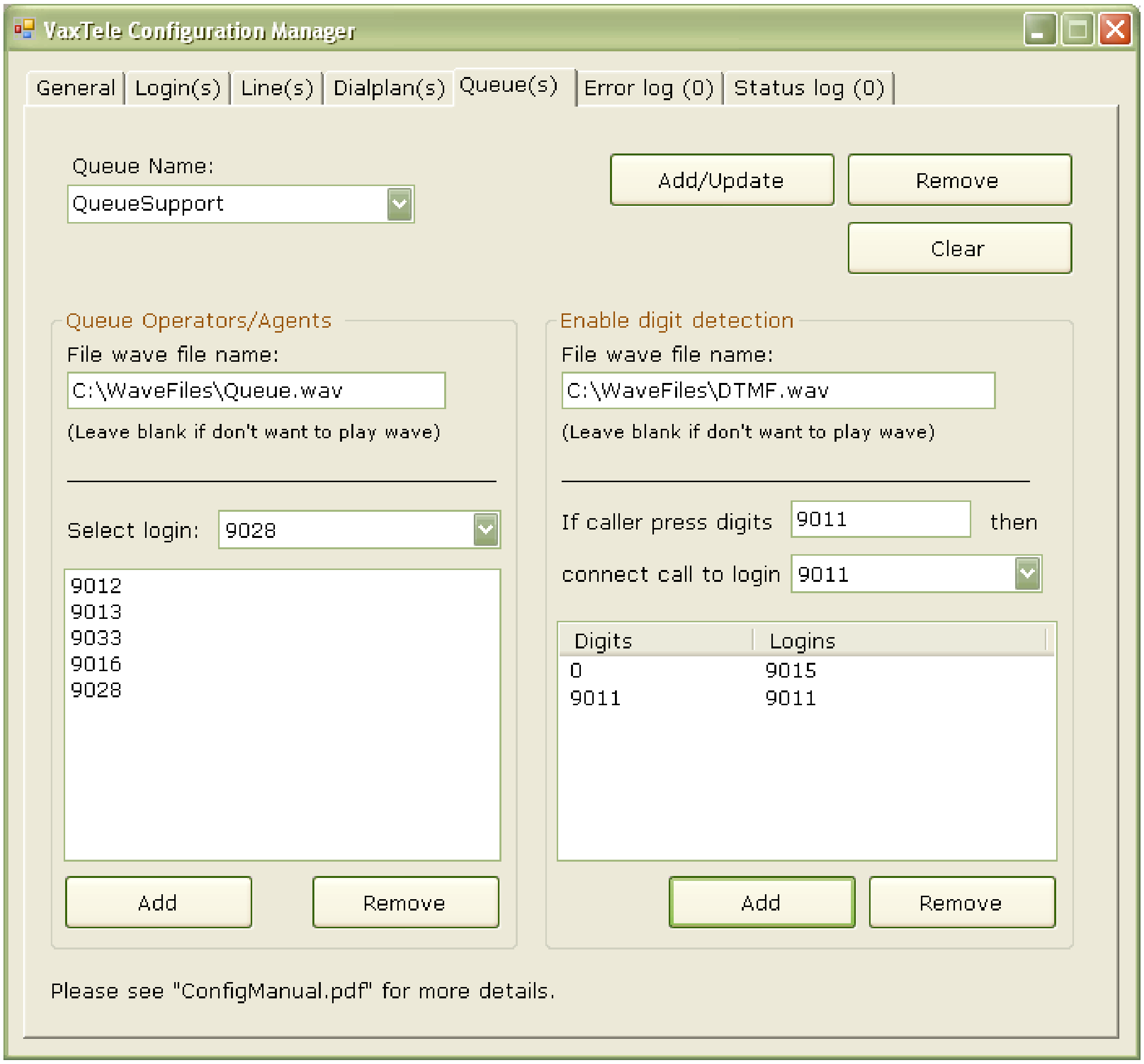
00 prefix is used for international call dialing and 971 is Dubai country code.

In the above scenario, user *(using softphone, ATA, hardphone or SIP client devices)* dials phone number 00971342xxxxxx then VaxTele SIP Server *(sample code or demo application)* removes the prefix and adds prefix 011971 to the phone number and then dials the phone number 011971342xxxxxx by using 'UAE-LINE' line.

NOTE: ITSP provides SIP account settings, first test those settings directly by using any softphone. Dial and receives phone calls with softphone, just to make sure that settings are working properly and then use those settings in SIP server.

So, it is very simple to route calls according to the dialplans. Sample source code is also available you can change or add more advance dialplans according to your requirements by changing the sample source code.

QUEUE TAB WINDOW



QUEUE tab window is used to create, remove and update queues.

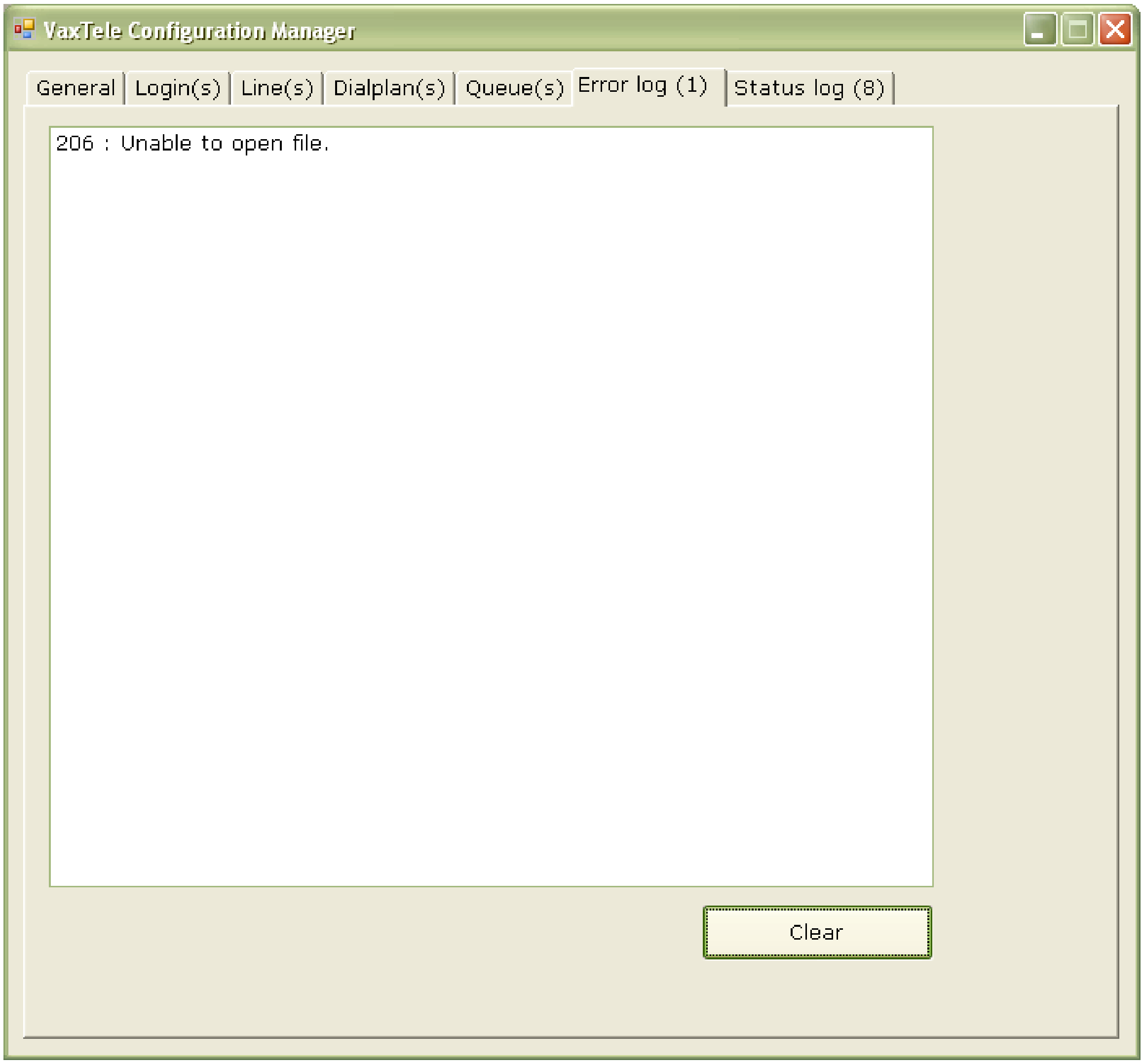
1. Create queues by using **QUEUE TAB WINDOW**.
2. Put incoming call into those queues by using **LINE TAB WINDOW**.

You can also play music on queue calls. VaxTele only support uncompress wave (.wav) files of sound quality format (8000Hz, 16bit, Mono) such wave file can easily be created or converted from other formats by using Microsoft's sound recorder.

VaxTele does not support MP3, wmv etc media files. Because these are CD-quality media files and require conversion to telephony quality sound, such conversion makes the media quality low and put a high load on the CPU.

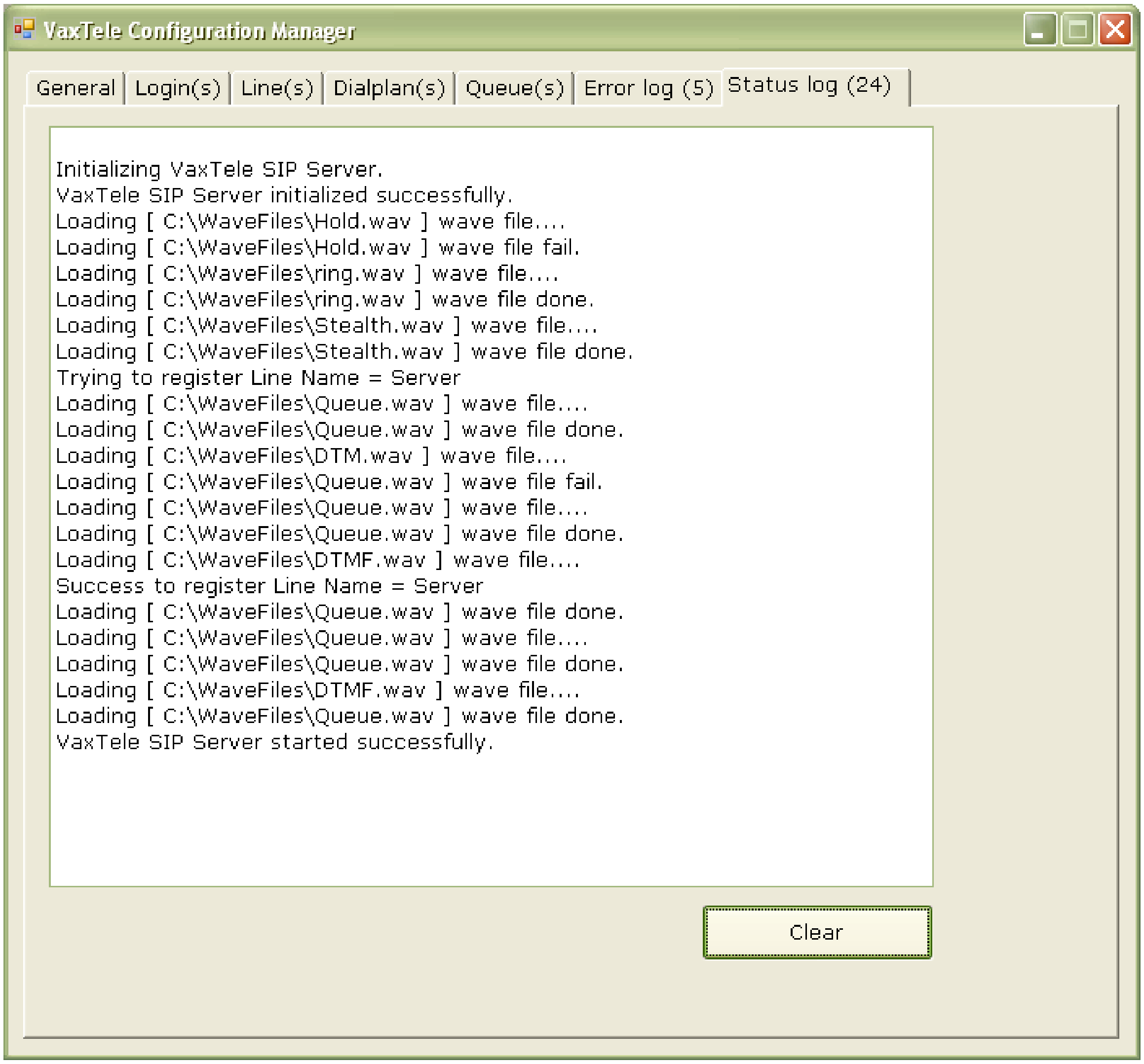
VaxTele only works and play uncompressed (8000Hz, 16bit, Mono) wave file (.wav) to save CPU cycles and put minimum load on the CPU processing and increases the server's efficiency.

ERROR LOG TAB WINDOW



ERROR LOG tab window receives error log/messages from SIP server. Each error message contains error code. For more details about error codes, please read technical manual TechManual.pdf or see the sample code.

STATUS LOG TAB WINDOW



STATUS LOG tab window receives status log/messages from VaxTele SIP server *(sample code & demo application)*. Please have a look at sample code for more details and coding.