About
This documentation is about API access for the Mizu SIP Softphone Classic Edition for Windows.

The Mizu Classic SIP Softphone can be easily controlled by external applications through it's configuration file(s) and commands sent by TCP connection.

Please ask MizuTech support for a modified build that is intended to run in the background with clear text configuration files. With the full version you can interact through its TCP interface, but the configuration files are encrypted.

Modifications in the special edition:
- the application is running in the background (no visible windows, no GUI tasks)
- only 2 file is required (Mizu.exe and zlib1.dll)
- no logfiles are created
- accepts application launch parameters: command port number, default profile name and ini file path (both are optional). The first parameter must be set to “console”
- automatically listen for external commands on the specified TCP port number (all network interfaces)
- automatically create the “mizuser” profile and automatic login in this profile
- only account number 1 is used (no multiple accounts) -optional
- the data path is the same with the application path by default (in the full version the data path is created in the user Application Data directory which can be found at C:\Documents and Settings\WINUSER\Application Data\MizuPhone\usrprofiles\MIZUUSER)

To “install”, just copy the Mizu.exe and zlib1.dll to the target directory. (You can integrate it in your install package)
If you need sound event notifications you have to copy the “sounds” directory near the executable.

Important: Make sure that the softphone has write access to its data directory!

TCP interface
The Mizuphone API passes commands in simple ASCII text messages terminated by \r\n between the softphone and client applications and devices. Clients can be applications which control the softphone or extend the softphone functionality.

The embedded TCP server is listening on port 58625 by default. This port number can be altered by application parameter (first parameter) or by configuration file setting:

```
[settings]
consoleport = 58625
```

Commands
The following commands are defined:

- `showmessage`, message
- `setaccount`, acc, address, username, password, proxy, regival, authusername, dtmf, immesagetype, autoanswer, signalingmode, mediamode
- `setnetwork`, localip, localport, rtpfrom, rtpto, usestun, useupnp, silencesupress, stunserveraddress
- `sethttpproxy`, address, username, password
- `setdevice`, acc, devicein, deviceout, volume_in, volume_out, agc, aec, denoise, soundnotifications
- `notify`, logs, status, cdrs, events, incomingcalls
- `dial`, acc, number, fromnumber
- `accept`, acc
- `senddtmf`, peer, dtmfstring
- `hangup`, peer
- `getstatus`
Parameters are separated by comma and messages are terminated by \n
Every command will be answered by an ERROR or OK message and in addition you can receive other notifications (defined by the “notify” command)

Most of the input parameters can be also empty strings (in this case the old setting will not be changed)

Configuration settings are stored in inifiles. This means that they values persist between restarts (there is no need to issue the “setaccount” command on every startup)

**showmessage**: can be used for testing

**message**: message to display

**setaccount**: configure sip server account. All parameters are optional

- **acc**: line. Default is 1
- **address**: sip server domain name or ip address. You can specify port number like address:port (the default port is 5060)
- **username**: sip server username
- **password**: sip server authentication password
- **proxy**: outbound proxy address (domain name, ip address or ip:port)
- **regival**: registration interval. Set to 0 to disable registrations. Default value is 120
- **authusername**: authentication username (if it is different from “username”)

**dtmf method**:  
0: In-Band  
1: INFO  
2: In-Band and INFO  
3: RFC 2833  
4: RFC 2833 and INFO  
5: In-Band+RFC 2833+INFO  
6: RFC 2833 or INFO

**immessagetype**: instant message (chat) type  
0: Autodetect  
1: HTML  
2: Plain Text

**autoanswer**: true or false (if set to true, all incoming calls will be accepted automatically)

**signalingmode**: compatibility option

**mediamode**: compatibility option  
0: Always force public address  
1: Use public address when possible  
2: Autodetect whether to use private or public address  
3: Use private address when behind NAT  
4: Always force private address

you may have to adjust this according to your SIP server NAT handling capability.

**setnetwork**: network configuration

- **localip**: local interface (default is all)
- **localport**: local signaling port (default is random)
- **rtpfrom**: rtp port interval
- **rtpo**: rtp port interval
- **usestun**: enable STUN requests
- **useupnp**: enable UPNP
- **silencesupress**: enable silence suppression
- **stunserveraddress**: ip or domain of the STUN server in case when usestun is set to true (otherwise random server will be used)

**sethttpproxy**: http proxy configuration for tunneling clients

- **http_proxy_address**: address of the http server. For example 192.168.1.10:8080
- **http_proxy_username**: proxy authentication
- **http_proxy_password**: proxy authentication
**setcodec**: set preferred voice codec  
**acc**: line. Default is 1  
**codec**: codec name  

The following codecs are supported:  
- G.729  
- G.723  
- PCMU  
- PCMA  
- G722  
- G728  
- ILBC  
- SPEEXUWB  
- SPEEXWB  
- SPEEX  
- GSM  

**disableothers**: set to true if you wish to use only this codec (otherwise this codec will have the highest priority but other codecs will be also allowed)

**setdevice**: audio device configuration  
**acc**: : line. Default is 1  
**devicein**: microphone audio device name or number  
**deviceout**: speaker audio device name or number  
**volume_in**: microphone volume level from 0 to 100  
**volume_out**: speaker volume level from 0 to 100  
**agc**: enable auto gain (true or false)  
**aec**: enable automatic echo cancellation (true or false)  
**denoise**: enable noise suppression (true or false)  
**soundnotifications**: enable sound (true or false). Set to false to disable all audio events

C/C++ code to list audio devices:

```c
WAVEINCAPS incaps;
UINT innr = waveInGetNumDevs(); // recording devices
//UINT innr = waveOutGetNumDevs(); // speakers
for(unsigned int i=0; i<innr; i++)
{
    MMRESULT ret = waveInGetDevCaps(i,&incaps,sizeof(incaps));
    //MMRESULT ret = waveOutGetDevCaps(i,&incaps,sizeof(incaps));
    if(ret == MMSYSERR_NOERROR) mydevicelist->Add(incaps.szPname);
}
```

**notify**: define events you wish to receive on the tcp interface  
**logs**: loglevel from 0 to 5. Default value is 1 (only important events)  
**status**: status notifications  
0: off  
1: receive “status” messages on best status change  
2: receive “statusex” messages about all endpoints separately (default)  
3: receive “statusex” also from other endpoints (like register)  
**cdrs**: set to 1 if you wish to receive “cdr” notifications. Otherwise set to 0 (Default is 1)  
**events**: notification about events otherwise displayed in the history. 0 or 1 (Default is 0 that means no events)  
**incomingcalls**: separate notification about incoming calls (incoming,... -see below). You can also catch incoming calls from statusex messages. (Default is 1)

**dial**: initiate outgoing calls  
**acc**: : line. Default is 1
number: destination phone number, username or SIP URI
fromnumber: caller number (optional. Otherwise the account username will be used)
On successful call initiation the OK message will be returned with the sip call-id (unique identifier for the session)
Otherwise and ERROR message will be returned.

accept: accept incoming call
acc: line. Default is 1

senddtmf: send dtmf to the connected peer
peer: remote party name or number (optional parameter useful if there is more than one call in progress)
message: dtmf in clear text (can be multiple characters)

hangup, callid
peer: remote party name or number (optional parameter useful if there is more than one call in progress)

getstatus: will return the current phone status (you can also receive status changes automatically)

reload: read the configuration file (can be useful if you changed it runtime)
quit: stop the program

Notifications
The following strings can be sent by the softphone: ERROR, OK, log, status, statusex, cdr, event.
The messages sent can be controlled by the “notify” command.

ERROR, details: can be sent as answer to commands

OK, details: sent as an answer to successfully completed processed commands. The details parameter is optional.

log, type, message: type can be ERROR, WARNING or EVENT and the message is the actual text

incoming, acc, callid, callernumber, callerip, callednumber
acc: used line
callid: unique session ID
callernumber: caller username or phone number (A number)
callerip: originating IP
callednumber: local number

status, statustext: sent when the softphone main status is changed.
The statustext can have the following values
  - Not connected
  - Offline
  - Not configured
  - Initializing...
  - Starting...
  - No network
  - Not Used
  - Not Connected
  - Connected but call failed
  - Not connected but can call
  - Connecting...
  - Active
  - Connected
  - Connected with successful calls
status, acc, callid, callernumber, callednumber, status: endpoint status sent on every status change (if requested)

acc: line (usually 1)
callid: sip call-id (random string, unique for every session)
callernumber: caller number (A number)
callednumber: called number (B number)
status: endpoint status. The followings are defined:
Unknown
Init
Ready
Outband
SignIn
Subscribe
Chat
Setup
CallProgress
Routing
Routed
Ringing
CallInitiated
CallStarted
Midcall
CallFinishing
CallFinished
Deletable
Error

“status” notification means overall program status, which will represent the “best” endpoint status if you have multiple accounts.
“statusex” notification is about one endpoint status changes defined by call-id.
in your program you can choose to check only for statusex messages and not to parse status messages because they are redundant.

cdr, in/out, party, connecttime, duration, caller, called, disconnectreason: call detail record sent after each call

in/out: incoming or outgoing call
party: other party number (caller or called)
connecttime: call between setup and connect in seconds
duration: call length in seconds
caller: A number
called: B number
disconnectreason: Terminated by local UA/peer Code: SIPCODE message
where SIPCODE can be “CANCEL”, “BYE” or SIP reason codes Please check RFC 3261 for reason codes.

event, type, party, flag, comment: events that are displayed in history otherwise

Example
To test the tcp command interface just start the softphone and connect to the tcp port with any tcp client utility.
telnet 127.0.0.1 58625 //connect to the tcp server
showmessage,test //test if working
setaccount,,mysipdomain.com,username,password //set sip account
setdevice,,,,,,false //disable automatic echo cancellation
dial,,number //initiate call
quit //terminate the softphone process

Code Example
void main()
{
Configuration files

There are several “levels” where the default settings can be stored. But the most important configuration file is the user preferences (6). Levels with higher priority always overwrite the default settings with lower priority. There are a few settings that cannot be overwritten by higher levels (for example the list of provider ip address which ensures proper licensing)

The following levels are defined (in priority order):

1. Default values (usually empty strings and 0 values)
2. Hardcoded basic settings (basic settings embedded with the executable)
This means that user preferences will always overwrite default company settings (but in the default company settings you can define values that cannot be overwritten. For example you can specify 60 sec for the registration interval and set it as read-only, thus it cannot be overwritten by higher level settings)

### Configuration file hierarchy

There are more than 200 settings that can be controlled with configuration files. There can be changes in new version and MizuTech doesn't offer support for working with configuration files. However you should not have any issue changing the configuration settings directly.

Configuration files are checked in the following order:

1. in the install (bin) path (usually C:\Program Files\Mizu) – usually empty
2. in the data path (usually C:\Documents and Settings\WINUSER\Application Data\MizuPhone)
3. in the user profile path (usually C:\Documents and Settings\WINUSER\Application Data\MizuPhone\usrprofiles\MIZUUSER)
4. defconfig.ini – this can be downloaded from your server and overwrite any user config. In this way you can easily force new configuration settings for your users.

In the special edition usually all configuration files are placed in the application path and its subdirectories.

The config value stored in the user profile will overwrite the data path config, which will also overwrite the install path config values. This means that you can ship some default configuration with the install package (stored in the bin path), but some of them can be overwritten explicitly by the user (if you allow it on the GUI).

### Configuration settings

*This is listed only for informational purposes. Usually you will not need to change the default configuration options.*

<table>
<thead>
<tr>
<th>Category</th>
<th>Setting</th>
<th>Default Value</th>
<th>Comment</th>
</tr>
</thead>
<tbody>
<tr>
<td>setting</td>
<td>Alertonlowdiskspace</td>
<td>5</td>
<td>alert on low disc space</td>
</tr>
<tr>
<td>setting</td>
<td>allownumbersendback</td>
<td>false</td>
<td>allow to route back the call to the caller</td>
</tr>
<tr>
<td>setting</td>
<td>autodetectlocalip</td>
<td>true</td>
<td>automatically overwrite the localip value if set to true</td>
</tr>
<tr>
<td>setting</td>
<td>bindip</td>
<td>false</td>
<td>bind to this ip (for multithomed servers or if we run multiple servers on the same machine)</td>
</tr>
<tr>
<td>setting</td>
<td>boostonfirstcall</td>
<td>true</td>
<td>if we can save unsaved items</td>
</tr>
<tr>
<td>setting</td>
<td>canautomatoussavefiles</td>
<td>false</td>
<td>if to start with low priority and boost it when the first call arrives</td>
</tr>
<tr>
<td>setting</td>
<td>cfg_block711</td>
<td>0</td>
<td>if g?11 (PCMU,PCMA) calls are not allowed</td>
</tr>
<tr>
<td>setting</td>
<td>checkqueryrelease</td>
<td>false</td>
<td>if cpumt ime is constantly high, will restart</td>
</tr>
<tr>
<td>setting</td>
<td>checkqueryrelease</td>
<td>99</td>
<td>if to check the incoming number len //14</td>
</tr>
<tr>
<td>setting</td>
<td>country</td>
<td>1</td>
<td>if to connect the sipcall before to play the disconnect reason. 0=not connect, 1=connect only local users, 2=connect all users</td>
</tr>
<tr>
<td>setting</td>
<td>countryprefix</td>
<td>0</td>
<td>used in number normalizations</td>
</tr>
<tr>
<td>setting</td>
<td>cpuaaffinity</td>
<td>0</td>
<td>used on routing</td>
</tr>
<tr>
<td>setting</td>
<td>currency</td>
<td>1</td>
<td>local currency</td>
</tr>
<tr>
<td>setting</td>
<td>dailymainttaskhour</td>
<td>1</td>
<td>when to perform daily maintenance tasks</td>
</tr>
<tr>
<td>setting</td>
<td>dbloglevel</td>
<td>0</td>
<td>db server loglevel 0=only errors to monitor, 1=only all to monitor, 2=only errors to db, 3=no protocol and events, 4=no protocol and filtering, 5=no duplicates, 6=log all</td>
</tr>
<tr>
<td>setting</td>
<td>dbmaint_backuplevel</td>
<td>1</td>
<td>0=no backups, 1=daily, 2=daily, monthly, 3=hourly, daily.monthly, weekly, 4=full, 5=keep lots of files</td>
</tr>
<tr>
<td>setting</td>
<td>dbtimeout</td>
<td>1</td>
<td>database query timeout</td>
</tr>
<tr>
<td>setting</td>
<td>deletemaintlogfiles</td>
<td>-1</td>
<td>delete old backup files after this day elapsed</td>
</tr>
<tr>
<td>setting</td>
<td>deletemaintlogfiles</td>
<td>14</td>
<td>delete older logfiles than the specified day (set to 0 to disable)</td>
</tr>
<tr>
<td>setting</td>
<td>emailfromaddr</td>
<td><a href="mailto:smtpserver@your.com">smtpserver@your.com</a></td>
<td>default email config</td>
</tr>
<tr>
<td>setting</td>
<td>emailfromname</td>
<td>smtpserver</td>
<td>default email config</td>
</tr>
<tr>
<td>setting</td>
<td>emailhost</td>
<td>127.0.0.1</td>
<td>smtp server used for alerting</td>
</tr>
<tr>
<td>setting</td>
<td>emailsubject</td>
<td>SIP Notification</td>
<td>default email config</td>
</tr>
<tr>
<td>setting</td>
<td>emailuser</td>
<td>smtpserver</td>
<td>smtp username used for alerting</td>
</tr>
<tr>
<td>setting</td>
<td>emergencydir</td>
<td>unknown</td>
<td>route emergency calls to this gateway (user id)</td>
</tr>
<tr>
<td>setting</td>
<td>enablefirewall</td>
<td>true</td>
<td>enable/disable builtin firewall and dos attack filtering</td>
</tr>
<tr>
<td>setting</td>
<td>enforcestrongauth</td>
<td>true</td>
<td>enforce authorization and strong passwords</td>
</tr>
<tr>
<td>setting</td>
<td>faxfromaddr</td>
<td><a href="mailto:smtpserver@your.com">smtpserver@your.com</a></td>
<td>fax sender configuration (email to fax server)</td>
</tr>
<tr>
<td>setting</td>
<td>faxfromname</td>
<td>smtpserver</td>
<td>fax sender configuration (email to fax server)</td>
</tr>
<tr>
<td>settings</td>
<td>value</td>
<td>description</td>
<td></td>
</tr>
<tr>
<td>------------------</td>
<td>----------------</td>
<td>-----------------------------------------------------------------------------</td>
<td></td>
</tr>
<tr>
<td>faxhost</td>
<td>127.0.0.1</td>
<td>fax sender configuration (email to fax server)</td>
<td></td>
</tr>
<tr>
<td>faxnormalize</td>
<td>0</td>
<td>fax sender configuration (email to fax server)</td>
<td></td>
</tr>
<tr>
<td>SIPServer</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>fxsubject</td>
<td>Notification</td>
<td>fax sender configuration (email to fax server)</td>
<td></td>
</tr>
<tr>
<td>fxsuffix</td>
<td>sipserver</td>
<td>fax sender configuration (email to fax server)</td>
<td></td>
</tr>
<tr>
<td>settings</td>
<td>fileloglevel</td>
<td>3</td>
<td>file server loglevel (0=only errors to monitor, 1=all only to monitor, 2=only errors to db, 3=no protocols and events, 4=no protocol and filtering, 5=no duplicates, 6=log all)</td>
</tr>
<tr>
<td>settings</td>
<td>filetransferbufflen</td>
<td>10000</td>
<td>fileserver buffer length</td>
</tr>
<tr>
<td>settings</td>
<td>filetransferfict</td>
<td>500</td>
<td>fileserver speed</td>
</tr>
<tr>
<td>settings</td>
<td>gmtdiff</td>
<td>the difference to gmt (useful for sip date header)</td>
<td></td>
</tr>
<tr>
<td>settings</td>
<td>InternalIP</td>
<td>sipserver internal ip (interface to clients)</td>
<td></td>
</tr>
<tr>
<td>settings</td>
<td>keepbackprecord</td>
<td>310</td>
<td>days to keep voice records in the backup directory</td>
</tr>
<tr>
<td>settings</td>
<td>keeprecorded</td>
<td>92</td>
<td>days to keep voice records</td>
</tr>
<tr>
<td>settings</td>
<td>LocalIP</td>
<td>sipserver external ip</td>
<td></td>
</tr>
<tr>
<td>settings</td>
<td>loglevel</td>
<td>3</td>
<td>other server loglevel (0=only errors to monitor, 1=all only to monitor, 2=only errors to db, 3=no protocols and events, 4=no protocol and filtering, 5=no duplicates, 6=log all)</td>
</tr>
<tr>
<td>settings</td>
<td>lognofreecardc</td>
<td>0</td>
<td>list free card data when no route found</td>
</tr>
<tr>
<td>settings</td>
<td>logtodb</td>
<td>true</td>
<td>trace to database (log)</td>
</tr>
<tr>
<td>settings</td>
<td>logtofile</td>
<td>true</td>
<td>trace to file (log)</td>
</tr>
<tr>
<td>settings</td>
<td>maxloglisten</td>
<td>600</td>
<td>max log message queue length</td>
</tr>
<tr>
<td>settings</td>
<td>maxmemoryutilization</td>
<td>590000</td>
<td>max memory utilization in KB (restart if exceed)</td>
</tr>
<tr>
<td>settings</td>
<td>maxudpsselect</td>
<td>-2</td>
<td>max socket on select (set to -2 to autoconfigure. -1 means no limits)</td>
</tr>
<tr>
<td>settings</td>
<td>minlogdelay</td>
<td>15</td>
<td>minimum delay between writing two log messages in msec</td>
</tr>
<tr>
<td>settings</td>
<td>minmemoryutilization</td>
<td>250000</td>
<td>will restart on offpeak if exceed</td>
</tr>
<tr>
<td>settings</td>
<td>normalizenumbers</td>
<td>3</td>
<td>0=not at all, 1=medium, 2=all but no endusers, 3=check endusers too, 4=full</td>
</tr>
<tr>
<td>settings</td>
<td>priority</td>
<td>2</td>
<td>0=low, 1=below, 2=normal, 3=abowe, 4=high, 5=realtime</td>
</tr>
<tr>
<td>settings</td>
<td>removeretaininghash</td>
<td>1</td>
<td>remove # when routing</td>
</tr>
<tr>
<td>settings</td>
<td>rotatelogfile</td>
<td>true</td>
<td>create separate logfiles for every day</td>
</tr>
<tr>
<td>settings</td>
<td>sendmailalert</td>
<td>if to send alerts on critical errors (please configure the emailalertX settings)</td>
<td></td>
</tr>
<tr>
<td>settings</td>
<td>servername</td>
<td>SIPServer</td>
<td>server name (will appear in reports, alerts, etc)</td>
</tr>
<tr>
<td>settings</td>
<td>usedefaultdiscodes</td>
<td>false</td>
<td>don't use customized disconnect codes</td>
</tr>
<tr>
<td>settings</td>
<td>usedelayedupdate</td>
<td>true</td>
<td>sql updates in separate thread</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>ABSOLUTETIMEOUT</td>
<td>11100</td>
<td>max session time (call duration setup time clearing time) in seconds</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>addcontentdisposition</td>
<td>2</td>
<td>0=no</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>allowcallunregistered</td>
<td>true</td>
<td>allow to call before registered (terminals)</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>allowdisconnectmessage</td>
<td>true</td>
<td>allow disconnect reason voice playback</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>allowlist</td>
<td>E,CANC</td>
<td>sip methods</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>CanAcceptLocalip</td>
<td>false</td>
<td>Can call from 127.0.0.1</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>cancutsipnumbers</td>
<td>true</td>
<td>packet dialplan for sipnumbers</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>canmove</td>
<td>1</td>
<td>0=not allowed, 1=callednumber change allowed, 2=domain change allowed</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>checknomedia</td>
<td>true</td>
<td>disconnect calls on rtp disconnect</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>COMPANYNAME</td>
<td>this will appear in the sip signaling</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>def_max_sessiontimer</td>
<td>3600</td>
<td>sip session-timer config</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>def_mid_sessiontimer</td>
<td>1800</td>
<td>sip session-timer config</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>def_min_sessiontimer</td>
<td>90</td>
<td>sip session-timer config</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>domainnames</td>
<td>registrar domainnames (used for inter-domain rerouting)</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>eventlist</td>
<td>refer, telephone-event, keep-alive</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>forwardretrytimer</td>
<td>8</td>
<td>ivr forward retry interval</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>fwdtootherdomains</td>
<td>0</td>
<td>0=no, 1=check, 2=notelnnumbers, 3=all, 4=reroute all, 5=unconditional</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>fwdunknownheaders</td>
<td>true</td>
<td>forward unknown sip headers</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>HasinternalAccess</td>
<td>true</td>
<td>accept from 192.165... or 10.0.... etc</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>identityrwmode</td>
<td>2</td>
<td>0=no rewrite, 1= basic, 2= conform sip specification (identity)</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>IDLETIMEOUT</td>
<td>120</td>
<td>used for various session timers</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>im_parentid</td>
<td>-1</td>
<td>used for instant messaging</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>lastlocalssupport</td>
<td>-1</td>
<td>used in terminals</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>loadcallednumberfromfron</td>
<td>true</td>
<td>load the called number from sip to instead from the sip first line (URI)</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>localclientport</td>
<td>useful for 2 port configurations</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>LocalDomain</td>
<td>sipserver domainname</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>localinternaldomain</td>
<td>sipserver internal domainname</td>
<td></td>
</tr>
<tr>
<td>SIPSettings</td>
<td>LocalPort</td>
<td>5060</td>
<td>sipserver listen port</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>logsipmsgexchange</td>
<td>true</td>
<td>store the sip message headers in the cdr comment</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>MAINTIMERIVAL</td>
<td>2000</td>
<td>sip background process timer. used for garbage collections mainly</td>
</tr>
<tr>
<td>SIPSettings</td>
<td>MAXEXP_COUNTTRESHOL</td>
<td>1000</td>
<td>maximum number of registered endpoint (it may be limited by license too)</td>
</tr>
</tbody>
</table>
SIPSettings
MAXH323GKCDRCACHE 300 this must at least the maximum h323-h323 simultaneous call number
SIPSettings maxreroute 3 max number rute retry
SIPSettings MaxRTP 44000 rtp port range begin for sip
SIPSettings MAXSPEACHLEN 10800 max allowed call duration in sec
SipSettings maxstatchangepermin 80 max allowed enduser status changes/60 sec (slower if exceed)
SIPSettings MAXSUBSMSGCOUNT 12000 max subsequent messages before block
SIPSettings MAXSUBSMSGPERIOD 180 max subsequent messages before block are checked for this interval (sec)
SIPSettings WED 6000 dos attack protection
SIPSettings MEDIATIMEOUT 300 will disconnect if the media disappears
SIPSettings MEDIATIMEOUTSTART 150 will disconnect if no media detected
SIPSettings MINRESENDIVAL 500 sip udp resend timer (T1) in msec
SipSettings MinRTP 24001 rtp port range begin for sip
SIPSettings MINUSRNAMELEN 3 minimum accepted username length
SIPSettings MAXWRONGMSGALLO WED 6000 dos attack protection
SIPSettings MEDIATIMEOUTSTART 150 will disconnect if no media detected
SIPSettings MINRESENDIVAL 500 sip udp resend timer (T1) in msec
SipSettings MinRTP 24001 rtp port range begin for sip
SIPSettings MINUSERNAMELEN 3 minimum accepted username length
SIPSettings MINUSRCREDITONROUTE 1 minimum credit for the caller when the router will still route the call
SIPSettings MINUSERCREDITONROUTE 1 minimum credit for the caller when the router will still route the call
SIPSettings PRODUCTNAME the name of the product. this will appear in the sip signaling
SipSettings propersipcomments false set to true if you want personalized sip headers
SIPSettings REBUILDREGCLIENTS 10800 usually the same as maxspeachlen
SIPSettings REENABLEDOSBLOCKED 43200 reenable blocked endpoints after this interval. defaults to 12 hour
SIPSettings REGISTERIVAL 40 upper registration interval in msec. defaults to 40 min
SIPSettings RELOADPROXYLISTIVAL 7200 reload proxies from the config
SIPSettings repeatdisc false if to send sip disconnect more than once
SIPSettings AL 13 used for rtp routing
SipSettings AL_MAIN 1300 used for main routing
SIPSettings resoludns true resolve uri domain names
SIPSettings ringtimeout 120 sip calls ring timeout
SIPSettings routepresence false route subscribes
SIPSettings rtpsendonlytorec 1 send the rtp only to the rec address
SIPSettings dpalshandling true send a rtp packet after connect (to open NAT)
SIPSettings sendfakesms 0 if to send fake sms
0: don’t use, 1: load from global config, 2: autodetect (and using the sesskeepalive interval from the global configuration), Other: use with the specified timeout (minutes)
SIPSettings sessiontimer 2 resend sip message count
SIPSettings sipsmsresendcount 3 resend sip message count
SIPSettings sipsmsresendival 1500 sip message resend timer
SIPSettings sipsendtodefport true try to send to port 5060 too
SIPSettings statussaveival 2 minutes. used when predective is active
SIPSettings supportlist places sip supported
SIPSettings trace1 user id, callednumber or callerip. all messages related to this ep will be written in logfile. set to negative to disable (log)
SIPSettings trace2 user id, callednumber or callerip. user id. all messages related to this ep will be written in logfile. set to negative to disable (log)
SIPSettings udpkeepalive 25 send keepalive messages
SIPSettings udppriority 3 rtp thread priority: 1=normal, 2 = higher, 3 = highest
SipSettings upperexpire 31 register expire
SIPSettings usedateheader true send date to user agents
SIPSettings userofflinemin 360 enduser will be considered offline if no register or invite for this period

Other config values

[contacts]
contact0=we can set default contacts here
dns

[epsettings_1]
accountname=bt_basic
aec=false
agc=false
audiodevicein=default
audiodeviceout=default
audiodeviceering=default
autoaccept_all=false
autoaccept_fax=true
cc_password=encrypted
cc_username=9991234570
delayedack=0
denoise=true
dtmftype=1
enablephototransfer=true
faxmode=1
forkallowed=true
hidecli=false
isfax=1
keepalivenotifynotsupported=
last_rtpresendin=0
maxjittersize=300
mediaaddressmode=2
minjittersize=31
missingnotify_calls=false
missingnotify_emails=false
noanswer_timeout=30
onlyencryptedsessions=false
plc=true
presence=1200000
proxyport=-1
registerival=120
retrywithallcode=true
rtpsendonlytorec=0
sendcomposingnotification=true
sendmail_missedcalls=false
sendmail_missedmsg=false
serverip=
serverport=-1
sessiontimer=0
signalingaddressmode=2
smsinsertname=true
startjittersize=41
startmediawithsessionp=0
transportprotocol=0
udpkeepalive=25
udpkeepalivetype=0
useaccount=false
useencryption=false
usefaxecm=false
usesrtp=false
voicemail_address2=
voicemail_always=false
voicemail_busystatus=false
voicemail_dontdisturb=false
voicemail_forward=false
voicemail_incall=false
voicemail_noanswer=false
voicemail_reject=false
voicemail_subscribe=4
volume_in=50
volume_out=50
volume_ring=50
webcamdevide=default

[events]
ev1=we can put some default event in the history
ev2=
firsteventid=1
lasteventid=2
[inifilehandling]
infilesavedt=2008/10/20 15:06:48
infilesavetick=16696437
[lastcallist]
um0=some predefine phone numbers here
num1=
[privacy]
blockcalls=0
blockchats=0
blockvideo=0
newpeople=1
transfer=0
[settings]
advancedsettingclicked=true
allcallcount=55
allowmultipleinstance=false
allphonecallcount=28
allspeechlength=885
allsucccallcount=29
alluseminutes=993
alwaysontop=false
autodetectlocalip=true
autogain_out_default=0.773807942867279
autogain_out_sb live! audio [ac00]=0.356729373335838
autoqos=true
autostart=true
cc_password=
cc_recording2=1
cc_username=fenesiistvan
checkfornewversions=true
configtest=0
cpu_optimizations=true
defaultpage=0
dialidentity=false
dialvideo=false
displayfriendlydt=true
dnscache=3000
enableice=true
enablescripts=false
-enablestun=false
enableupnp=false
faxdirectory=C:\Documents and Settings\root\Application Data\Mizu\usrprofiles\u\incomingfax
fileloglevel=0
formstate=normal
history_chat=3
history_events=3
history_video=0
history_voice=3
keeprecorded=31
language=English
lastlocalip=
lastlocaliplist=10.0.0.1
lastpage=Dial
lastsavedepacc=1
lastselaccount=2
lastusedaccount=2
loglevelx=0
logipmsgexchange=1
logsqlcommands=1
logtofile=false
msgtype=0
mydetails=user details here
nearpeople_allow=1
needbigbuttons=true
needtoolbar=false
ondoubleclick=2
periodicbackup=3
personid=pi2064526267
profilestorage=0
profilestoragepwd=$ppassword
profilestorageurl=http://$pdomain/webdav/$pusername/
profilestorageusername=$pusername
scriptcallconnect=mscript_call_connect.exe -D inout -A caller -B called -S account
scriptcalldisc=mscript_call_disc.exe -D inout -A caller -B called -S account -D duration -R reason
scriptcallring=mscript_call_ring.exe -D inout -A caller -B called -S account
scriptcallstart=mscript_call_start.exe -D inout -A caller -B called -S account
scriptcontactpresence=mscript_presence.exe -A contact -S account -F status
scriptdtmfrec=mscript_dtmf.exe -A from -S account -T dtmf
scriptfaxrec=mscript_fax.exe -A from -S account -F file
scriptim=mscript_im.exe -D inout -A from -B to -S account
scriptmystatus=mscript_status.exe -S status
scriptuserlogin=mscript_uologin.exe -U username
scriptuserlogoff=mscript_uologoff.exe -U username
sendemailvia=0
sendmessageaction=0
separatechatwindow=false
showadvancedsettings=true
showadverisments=true
showcontactgroups=true
showcontacts=true
showdialpad=true
showhistory=true
showofflinecontacts=true
sielncesupress=false
silentmode=false
sortby_name=true
sortby_online=false
sortby_provider=false
sortby_ussage=false
stundomain=
use_rport=true
useridletime=10
[sipsettings]
blockselfcall=true
canmove=2
lastlocaladdresslist=
lastlocaliplist=10.0.0.100,
lastlocalsdportport=10100
localport=10001
maxrtp=10200
minrtp=10100
stunmap0=19616:19616
stunmap1=23974:23974
stunmap2=23976:23976
stunmap3=23978:23978
stunmap4=23980:23980
usetcp=false
usetls=false
useudp=true
[statuslist]
0=New custom status message
1=-
2=Call Me
3=Available
4=Do Not Disturb
5=Invisible
6=Away
7=Offline

Resources
Mizu Classic Softphone homepage: