MizuPhone Classic API (v2.2)

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 trough 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 trough 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, defaultprofilename 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 ASCI 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]
cconsoleport = 58625

Commands

The following commands are defined:

-**showmessage**,message
-**setaccount**,acc,address,username,password,proxy,regival,authusername,dtmf,immessagetype,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**

-**reload**
-**quit**

Parameters are separated by comma and messages are terminated by \r\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
rtpto: 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:

```
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 successfull 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

- Unknown
-Ep status (see below)

statusex,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

"staus" 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()
{
```

```
 //write default configuration in inifile (optional because the most important settings can also be done by tcp commands)
  String appdir = "C:\\dialer\\";
  if(!DirectoryExists(appdir+"usrprofiles")) CreateDirectory(appdir+"usrprofiles");
  if(!DirectoryExists(appdir+"usrprofiles\\mizuser")) CreateDirectory(appdir+"usrprofiles\\mizuser");
  String inifile = appdir+"usrprofiles\\mizuser\\mizuphone.ini";
  WritePrivateProfileString("epsettings_1","settings","serverip",SIPSERVERIP,inifile);
  WritePrivateProfileString("epsettings_1","settings","serverport",SIPSERVERPORT,inifile);
  WritePrivateProfileString("epsettings_1","settings","cc_username",USERNAME,inifile);
  WritePrivateProfileString("epsettings_1","settings","cc_password",PASSWORD,inifile);
 //start the softphone
 if(ShellExecute(NULL, "open", appdir + "mizuphone.exe","console 58625", NULL,SW_HIDE) <= 32)
 {
  Show("Cannot launch the dialer");
  exit();
 }

 //wait a little for startup
 Sleep(1000);

 //try to connect (use a timeout here instead of infinite loop)
 tcpclientsocket->ReadEvent = HandleReceived;
 while(true)
 {
  if(tcpclientsocket->Connect(58625))
  {
   break; //successfully connected
  }
 }

 //configure account (if not already done by config file)
 tcpclientsocket->Send("setaccount,,mysipdomain.com,username,password\r\n");
 String read = ReadFromSoftphone();
 if(read != "OK")
 {
   Show("error or timeout");
   exit();
 }

 //initiate outgoing call
 tcpclientsocket->Send("dial,,callto\r\n");
 String read = ReadFromSoftphone();
 if(read != "OK")
 {
   Show("error or timeout");
   exit();
 }

 //process messages by "HandleReceived"

 exit();
}

void HandleReceived(String received)
{
 if(received.Pos("OK") == 1) ; //ack for command
 else if(received.Pos("ERROR") == 1) ; //invalid command or error
 else if(received.Pos("statusex") == 1) ;//process main phone status  -optional
 else if(received.Pos("status") == 1) ;//process endpoint status  -recommended
 else if(received.Pos("cdr") == 1) ;//process call details -recommended
 else if(received.Pos("log") == 1) ;//process received logs (check for errors) -optional
 else ; //unknown command
}
```

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 lover 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)

3. Company specific default settings (embedded in the software based on your requirements sent to MizuTech)
4. Company configuration file
5. Default config file (can be downloaded from your web server)
6. User preferences (Customer settings). The location of this file can be specified by command line. By default it is stored in APPDIR\usrprofiles\mizuser\mizuphone.ini

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.*

Category	Setting	Default Value	Comment
Setting	Alertonlowdiskspace	5	alert on low disc space
settings	allownumbersendback	false	allow to route back the call to the caller
settings	autodetectlocalip	true	automatically overwrite the localip value if set to true
settings	bindip		bind to this ip (for multihomed servers or if we run multiple serers on the same maschine)
settings	boostonfirstcall	false	if to start with low priority and boost it when the first call arrives
settings	canaoutosaveinifiles	false	if we can save unsaved items
settings	cfg_block711	0	if g711 (PCMU,PCMA) calls are not allowed
settings	checkcputime	false	if cputime is constantly high, will restart
settings	checknumlen	99	if to check the incoming number len //14
settings	checkqueryrelease	false	check query object consistence on release
settings	connectondisccode	1	if to connect the sipcall before to play the disconnect reason. 0=not connect,1=connect only local users,2=connect all
settings	country		used in number normalizations
settings	countryprefix		used on routing
settings	cpuaffinity	0	0=dont change,1=cpu1,2=cpu2,3=cpu 1,other=number of cpu cores
settings	currency		local currency
settings	dailymainttaskhour	1	when to perform daily maintenance tasks
settings	dbloglevel	1	db server loglevel (0=only errors to monitor, 1=only all to monitor, 2=only errors to db, 3=no protocoll and events, 4= no protocoll and filtering, 5=no duplicates,6=log all)
settings	dbmaint_backuplevel	1	0=no backups,1=daily,2=daily,monthly,weekly,3=hourly,daily,monthly,weekly,4=full,5=keep lots of files
settings	dbtimeout	40	database query timeout
settings	deldbbackup	-1	delete old backup files after this day elapsed
settings	deleteoldlogfiles	14	delete older logfiles than the specified day (set to 0 to disable)
settings	emailfromaddr	sipserver@yourcompa	default email config
settings	emailfromname	sipserver	default email config
settings	emailhost	127.0.0.1	smtp server used for alerting
settings	emailsubject	SIP Notification	default email config
settings	emailuser	sipserver	smtp username used for alerting
settings	emergencydir	unknown	route emergency calls to this gateway (user id)
settings	enablefirewall	true	enable/disable builtin firewall and dos attack filtering
settings	enforcestrongauth	true	enforce authorization and strong passwords
settings	faxfromaddr	sipserver@yourcompa	fax sender configuration (email to fax server)
settings	faxfromname	sipserver	fax sender configuration (email to fax server)

settings	faxhost	127.0.0.1	fax sender configuration (email to fax server)
settings	faxnormalize	0	fax sender configuration (email to fax server)
settings	faxsubject	SIPServer Notification	fax sender configuration (email to fax server)
settings	faxsuffix		fax sender configuration (email to fax server)
settings	faxuser	sipserver	fax sender configuration (email to fax server)
settings	fileloglevel	3	file server loglevel (0=only errors to monitor, 1=only all to monitor, 2=only errors to db, 3=no protocoll and events, 4= no protocoll and filtering, 5=no duplicates,6=log all)
settings	filetransferbufflen	10000	fileserver buffer length
settings	filetransfertick	500	fileserver speed
settings	gmtdiff		the difference to gmt (useful for sip date header)
settings	InternalIP		sipserver internal ip (interface to clinets)
settings	keepbackuprecorded	310	days to keep voice records in the backup directory
settings	keeprecorded	92	days to keep voice records
settings	LocalIP		sipserver external ip
settings	loglevel	3	other server loglevel (0=only errors to monitor, 1=only all to monitor, 2=only errors to db, 3=no protocoll and events, 4= no protocoll and filtering, 5=no duplicates,6=log all)
settings	lognofreecardc	0	list free card data when no route found
settings	logtodb	true	trace to database (log)
settings	logtofile	true	trace to file (log)
settings	maxloglisten	600	max log message quee length
settings	maxmemoryutilization	590000	max memory utilization in KB (restart if exceed)
settings	maxudpselect	-2	max socket on select (set to -2 to autoconfigure. -1 means no limits)
settings	minlogdelay	15	minimum delay between writing two log messages in msec
settings	minmemoryutilization	250000	will restart on offpeak if exceed
settings	normalizenumbers	3	0=not at all,1=medium,2=all but no endusers,3=check endusers too, 4=full
settings	ppriority	2	0=low,1=below,2=normal,3=abowe,4=high,5=realtime
settings	removetrailinghash	1	remove # when routing
settings	rotatelogfile	true	create separate logfiles for every day
settings	sendmailalert		if to send alerts on critial errors (pease configure the emailalertX settings)
settings	servername	SIPServer	server name (will appear in reports, alerts, etc)
settings	usedefaultdisccodes	false	don't use customized disconnect codes
settings	usedelayedupdate	true	sql upfates in separate thread
SIPSettings	ABSOLUTETIMEOUT	11100	max session time (call duration setup time clearing time) in seconds
SipSettings	addcontentdispozition	2	0=no
SIPSettings	allowcallunregistered		allow to call before registered (terminals)
SIPSettings	allowdiscmessage	true	allow disconect reason voice playback
SipSettings	allowlist	REGISTER,INVITE,CANC	sip methods
SIPSettings	CanAcceptLocalIp	false	Can call from 127.0.0.1
SipSettings	cancutsipnumbers	true	packet dialplan for sipnumbers
SipSettings	canmove	1	0=not allowed,1=callednumber change allowed,2=domain change allowed
SipSettings	checknomedia	true	disconnect calls on rtp disconnect
SIPSettings	COMPANYNAME		this will apear in the sip signaling
SipSettings	def_max_sessiontimer	3600	sip session-timer config
SipSettings	def_mid_sessiontimer	1800	sip session-timer config
SipSettings	def_min_sessiontimer	90	sip session-timer config
SipSettings	domainnames		registrar domainnames (used for inter-domain rerouting)
SipSettings	eventlist		refer,telephone-event,keep-alive
SipSettings	forwardretrytimer	8	ivr forward retry interval
SipSettings	fwdtootherdomains	0	0=no,1=check,2=notelnumbers,3=all,4=reroute all,5 unconditioanal
SIPSettings	fwdunknownheaders	true	forward unknown sip headers
SIPSettings	HasInternalAccess	true	accept from 192.168... or 10.0...., etc
SIPSettings	identityrwmode	2	0=no rewrite, 1=basic, 2=conform sip specification (identity)
SIPSettings	IDLETIMEOUT	120	used for various session timers
SIPSettings	im_parentid	-1	used for instant messaging
SIPSettings	lastlocalsdpport	-1	used in terminals
SIPSettings	loadcallednumberfromto	true	load the called number from sip to instead from the sip first line (URI)
SIPSettings	localclientport		useful for 2 port configurations
SIPSettings	LocalDomain		sipserver domainname
SIPSettings	localinternaldomain		sipserver internal domainname
SIPSettings	LocalPort	5060	sipserver listen port
SipSettings	logsipmsgexchange	true	store the sip message headers in the cdr comment
SIPSettings	MAINTIMERIVAL	2000	sip background process timer. used for garbage collections mainly
SIPSettings	MAXEPCOUNTTRESHOLD	1000	maximum number of registered endpoint (it may be limited by license too)

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	MAXWRONGMSGALLOWED	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	MINSPEACHLENONROUTE	40	minimum remained speechlength 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	MINUSERNAMELEN	3	minimum accepted username length
SIPSettings	PRODUCTNAME		the name of the product. this will apear 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	REPOPUPULATEFDSETIVAL	13	used for rtp routing
SipSettings	REPOPUPULATEFDSETIVAL_MAIN	1300	used for main routing
SipSettings	resolvedns	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	rtpwritefirst	true	send a rtp packet after connect (to open NAT)
SipSettings	sdpalthandling	1	0=not handled,1=check,2=delete,3=replace
SIPSettings	sendfakesms	0	if to send fake sms
SipSettings	sessiontimer	2	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	sipmsgresendcount	3	resend sip message count
SipSettings	sipmsgresendival	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	timer,privacy,replaces	sip supported
SIPSettings	traceep1		user id, callednumber or callerip. all messages related to this ep will be written in logfile. set to negative to disable (log)
SIPSettings	traceep2		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
[dnscache]
c0=predefined dns cache values here
c1=
[epsettings_1]
accountname=bt_basic
aec=false
agc=false
audiodevicein=default

```
audiodeviceout=default
audiodevicering=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
retrywithallcodec=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
usepublish=false
usesrtp=false
videobitrate=2
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]
inifilesavedt=2008/10/20 15:06:48
inifilesavetick=16696437
[lastcallist]
num0=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
loglevelex=0
logsipmsgexchange=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 -P 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_ulogin.exe -U username
scriptuserlogoff=mscript_ulogoff.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,
lastlocalsdpport=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:

https://www.mizu-voip.com/Software/Softphones/Windowssoftphone/WindowsSoftphoneClassic.aspx
For help, contact:
info@mizu-voip.com

MizuTech