

# MizuDroid

## User Guide

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## About

[MizuDroid](#) is a SIP softphone for Android devices including smartphones and tablets. The MizuDroid softphone allows you to connect to any SIP server or IP-PBX and make free or low-cost VoIP calls using a Wi-Fi internet connection or your mobile data plan (EDGE, 3G, 4G, LTE, 5G and others).

Its intuitive design ensures an easy setup and configuration also for beginner VoIP users.

## Editions

There are two versions of the MizuDroid SIP application:

### Public version

The public version of [MizuDroid](#) can be [downloaded](#) from Google Play named “MizuDroid SIP VOIP Softphone”.

This is an unlocked SIP softphone which can be used with any SIP server. You will need a valid SIP account to be able to use this softphone.

The softphone is available free for personal usage, however for extra features or commercial usage you will need to buy a license

### Customized version

The [custom Android SIP softphone](#) is provided for SIP service providers and other companies, customized with tailored settings, company name, brand, icon, server details, menu items, API integration, theme and [other settings](#).

## Features

MizuDroid is a feature rich professional SIP softphone.

### Standard features

All the usual VoIP client functionalities are supported:

- Incoming/outgoing calls
- Call details display
- Caller-ID
- Ringtones
- Speakerphone, Mute and Hold
- DTMF
- Call forward
- Call transfer
- Conference
- Contacts management
- Call history
- Voicemail
- Chat
- Presence

## Advanced Features

MizuDroid offers a wide range of possibilities for advanced users:

- Push notifications
- Native dialer integration
- Multiple accounts
- API integration for new user sign-up, balance request, payment and more
- Encryption (TLS/SRTP, tunneling and end to end encryption options)
- Group chat
- File transfer
- Video calls
- Automatic fine-tuning (including codec selection) depending on your peer capabilities, network speed and CPU power
- Presence
- BLF
- Call recording
- NAT/Firewall support, stable SIP and RTP ports, auto QoS, light STUN protocol and auto configuration
- Media enhancements
- Themes
- Server integration
- And many more

For technical background and full feature list see [JVoiP](#) and [AJVoIP](#) (The Android softphone is based on these libraries).

## Install

The MizuDroid can be installed from [Play Store](#) (just search for “MizuDroid” in the Play Store) or from an [APK file](#).

The installation process from Play Store is straightforward.

In case if you wish to install directly from MizuDroid.apk file you will need to enable “Unknown sources” on your android device. To ease the installation you might download an “APK Installer” app from the Google Play which is capable to install any apk from your phone flash memory or SD card. More details can be found [here](#).

## Screens

The softphone user interface is divided in the following main screens:

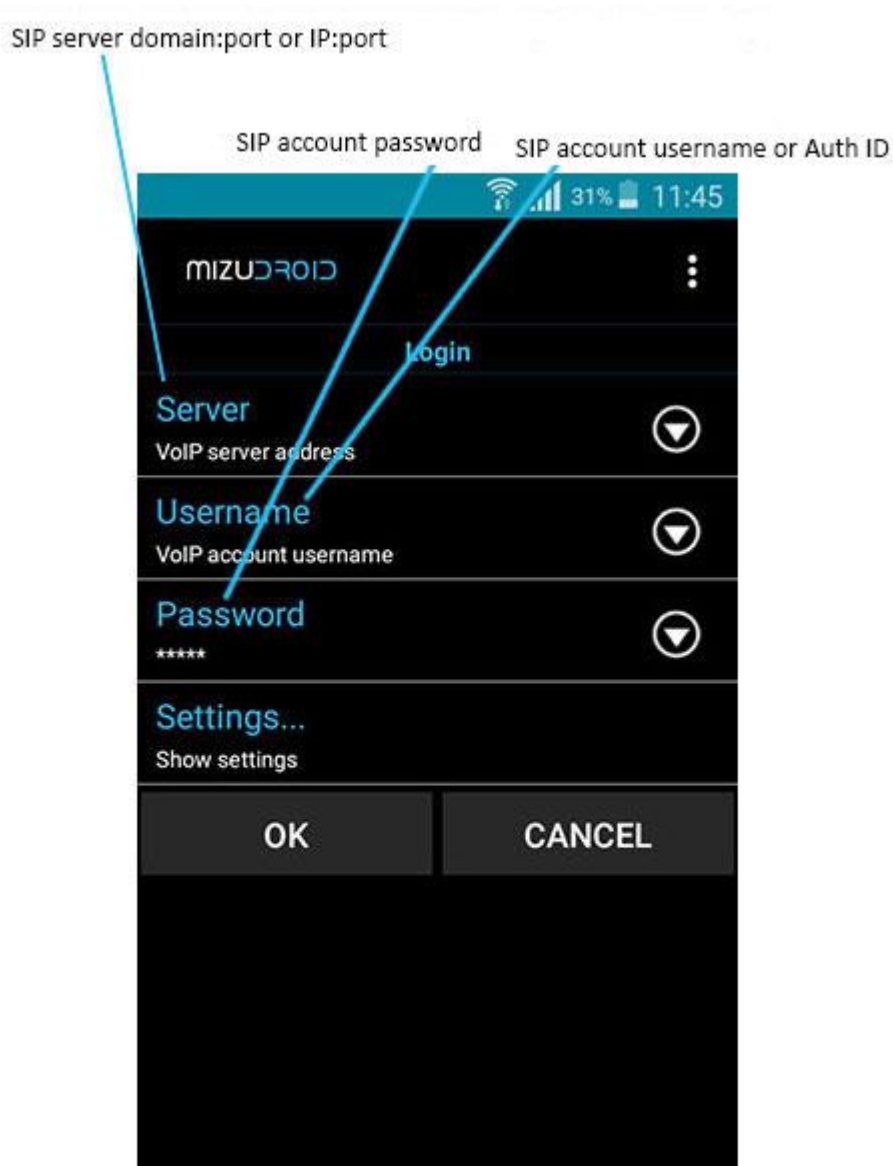
- Settings: login settings, basic settings, advanced settings
- Main page: will show a dial pad and/or your most important contacts
- Contacts: the full list of your address book (searchable)
- History: your call history, including missed calls
- Call page: shown when you are in call
- IM: used for chat
- Others: there are many other screens which might appear depending on your actions

## Softphone configuration

When you are starting the MizuDroid SIP softphone for the first time, you need to enter your SIP server address (domain name or IP address:port) and SIP credentials (username/password) in order to be able to register and initiate/receive SIP calls.

Some SIP servers might require using a proxy server. You can configure this from the Settings if needed.

Below is a screenshot of the login page:



Note: MizuDroid can be used also for peer to peer calls without the need of a server. In this case you don't need to enter any SIP server but you will need to enter the called party full URI ([sip:username@IP:port](#)) to be able to make a call. Then main reason of the SIP server is user location, call routing and NAT bypass, so you can just enter the destination user name to make a call if registered to a SIP server.

The full list of the configuration options are described [here](#).

After these login settings are entered, you need to press OK and you will be redirected to the main/dial page and if the SIP credentials were correct you will be able to register and receive calls from this moment.

From there you can make calls, send chat messages/SMS, check your contact list, call history and perform various other tasks.

Below is a screenshot about it, with some explanations:



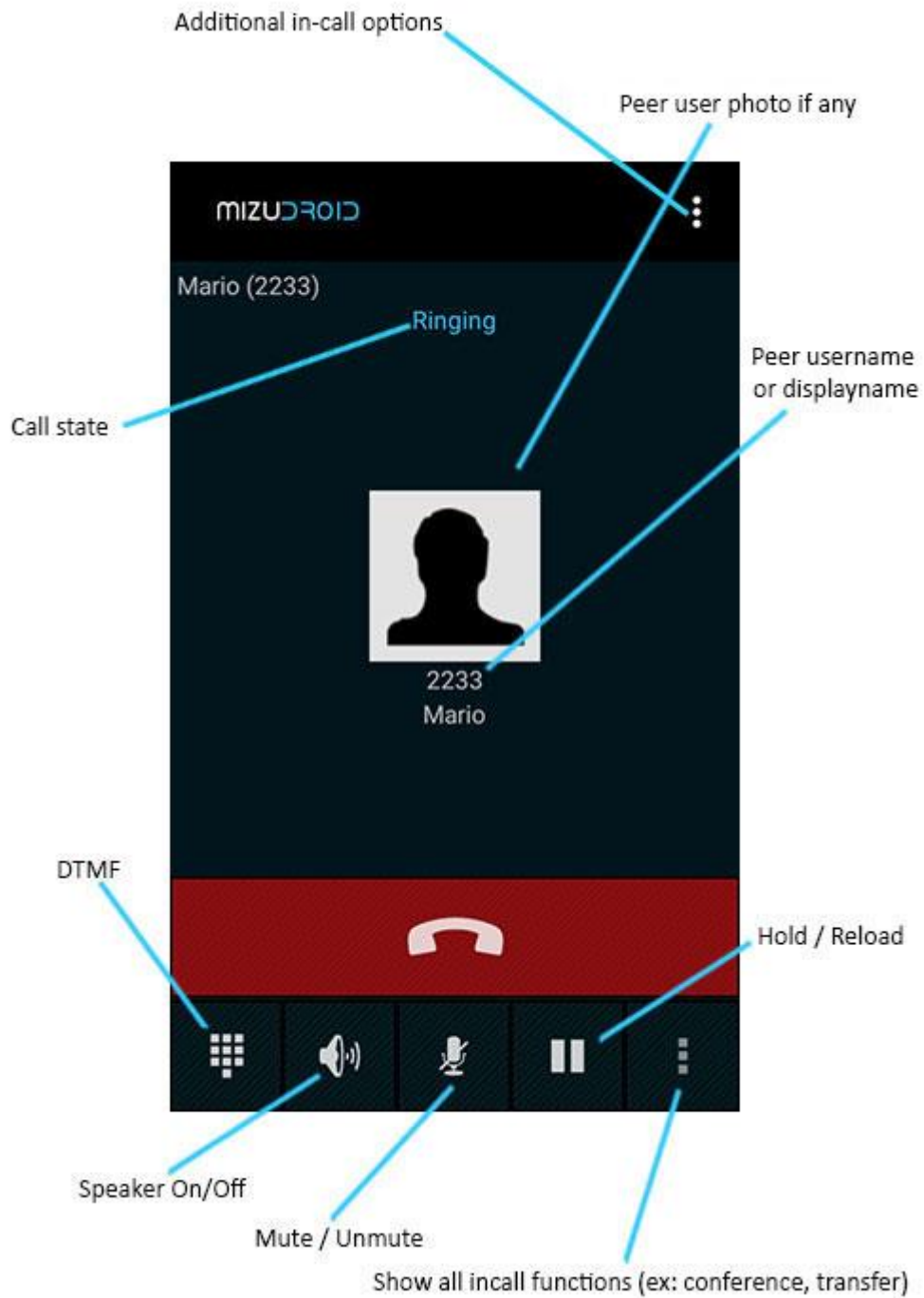
## Placing a call

There are many ways to initiate SIP calls from MizuDroid:

- Your most used contacts will appear on the main page after some time so you can initiate call with a simple touch on the desired contact
- Enter the destination number/username/extension or SIP URI on the main page and touch the Enter/Go/Call button
- Bring up the dial pad and enter the destination number
- Go to the Contacts page, search for Contact and call it
- Go to the History page to call a recent contact
- The big call button on the main page will redial the last number if no destination is entered
- You can also call from a contact card
- You can also make SIP calls from your native phone dialer if you enabled the native call integration

## Handling an established call

During your call, you are able to perform a few actions like: mute/unmute, speaker mode on/off, hold/reload, send DTMFs, create a conference and others:



## Incoming calls

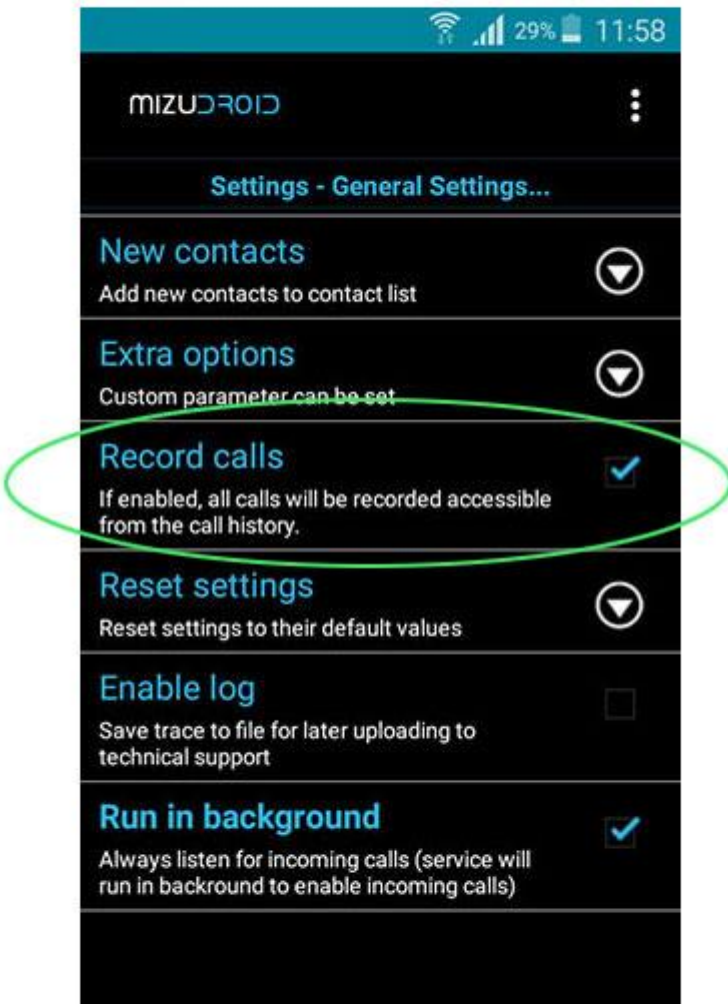
You can pick up an incoming call by pressing the “Answer button” or you can refuse it by pressing the “Reject button”. From the incoming call page menu you have additional options such as ignoring the call.



## Recording a call

If you wish to record all your calls, you just need to enable the Record calls setting (General settings) and the recorded calls will be accessible from the call history. Please note that laws in some countries prohibit call recording except if accepted by the peer. The usage of this functionality is up to your responsibility.





## Conference calls

Using the SIP softphone you will be able to create conference calls.

First of all you will need to call a desired person and after the call is established, you need to press the “Show all incall functions” button (check the screenshot from the “Handling established calls” section above) and after it you will need to press the Conference button and enter the desired phone number or SIP username to add more people to the conference.

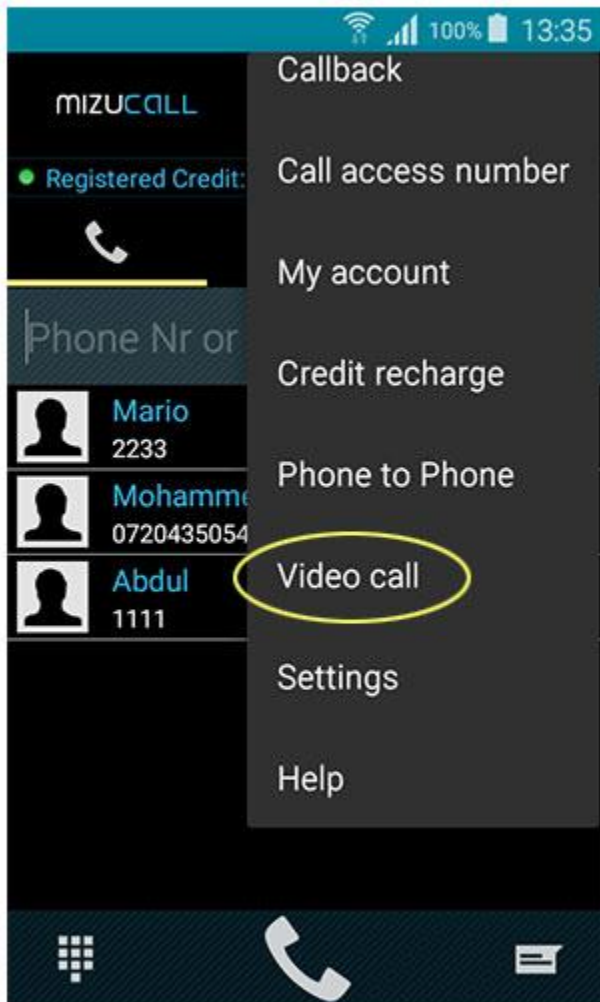
## Video call

Video calls can be initiated in multiple way with the MizuDroid softphone.

One method is from contact list, when you will select the desired contact, there you will see a Video call option:



The second method is from the Main menu, where you will find a Video call button. Once you will press it, you will get a pop-up, where you will need to enter the called party phone number or SIP URI:

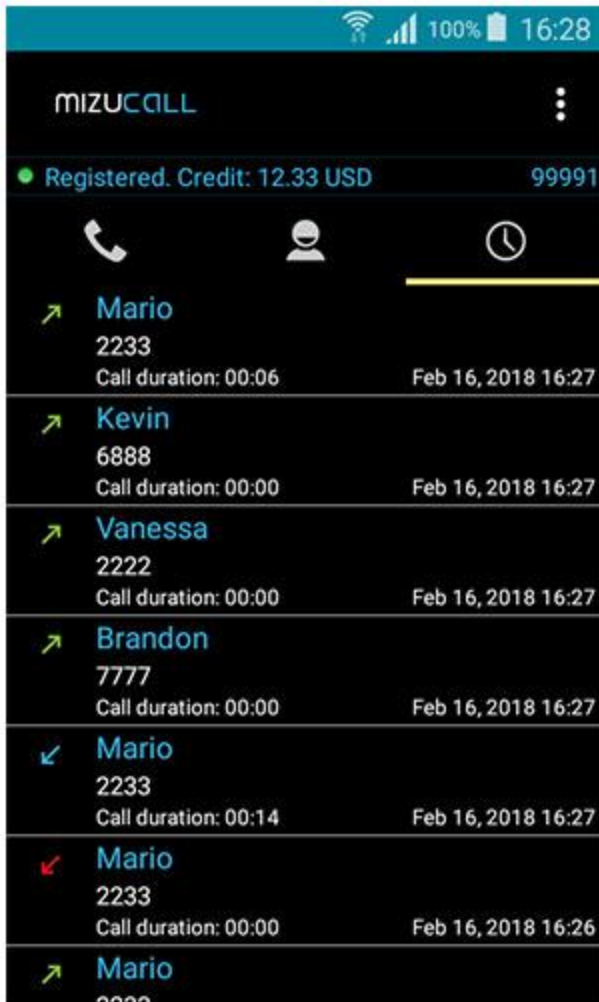


*Note: by default the video call option will appear only if your SIP service is reachable over the public internet with RTC video call capabilities.*

## Call history

On the call history page you can find the list of your outgoing, incoming and missed calls.

From here can also playback the recorded calls.



## DTMF

The SIP softphone allow you to send instant messages or SMS to your friends.

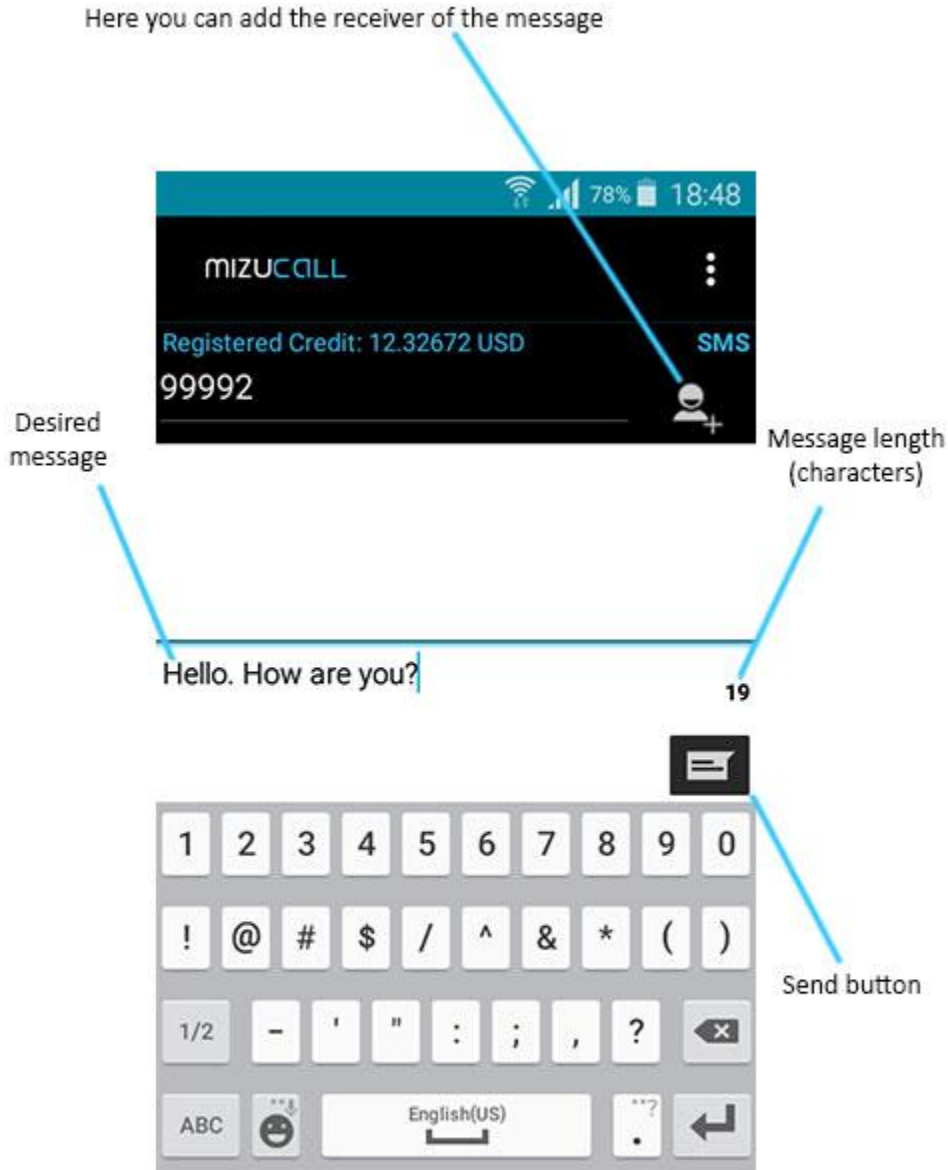
Press the keypad icon during call to be able to send DTMF tones (for interaction with IVR's).

The dtmf send method is automatically negotiated, however you can also force a specific method to be used if required from the settings.

## IM

The SIP softphone allow you to send instant messages (chat) or SMS to your friends.

Once you will press the Chat/SMS button you will get a pop-up, where you need to choose the message type (instant message or SMS) and then you will arrive to the Chat page:



## Presence

If your SIP server or service provider has support for presence, then presence information can be exchanged automatically with your important contacts.

You can set your presence state from the presence icon on the main screen.

For contacts with presence capability, a small icon will appear near their name with the following meaning:

- Green: online/available

- Red: do not disturb
- Orange/Yellow: pending/forwarding/busy/speaking
- Yellow: away/idle
- White: invisible
- Empty icon: unreachable/offline/unregistered
- No icon: unknown/not set/not exists

BLF (Busy lamp field) is also available but this must be explicitly activated if needed. The server and the remote peer must also support this feature to work. Presence is for general availability, while BLF is to be used to track the call state of a peer.

## Voicemail

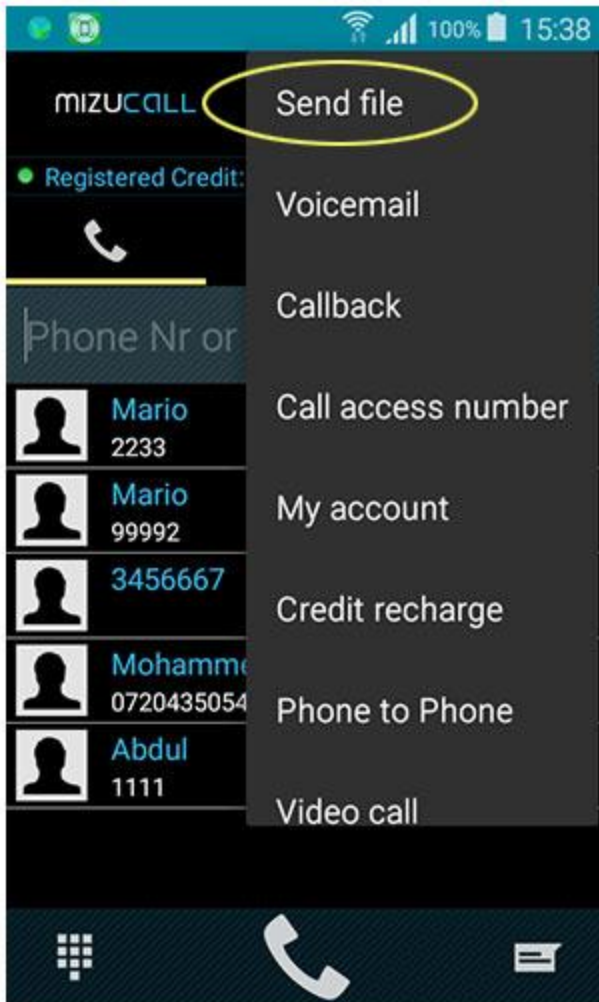
Some servers or SIP service providers offers voicemail functionality. Usually the voicemail access number can be automatically discovered by the SIP client. If the voicemail number was not loaded automatically then you can set it at the first usage or from the settings.

## File transfer

MizuDroid offers the possibility to send files (like photos, documents, audio or video files) to your friends.

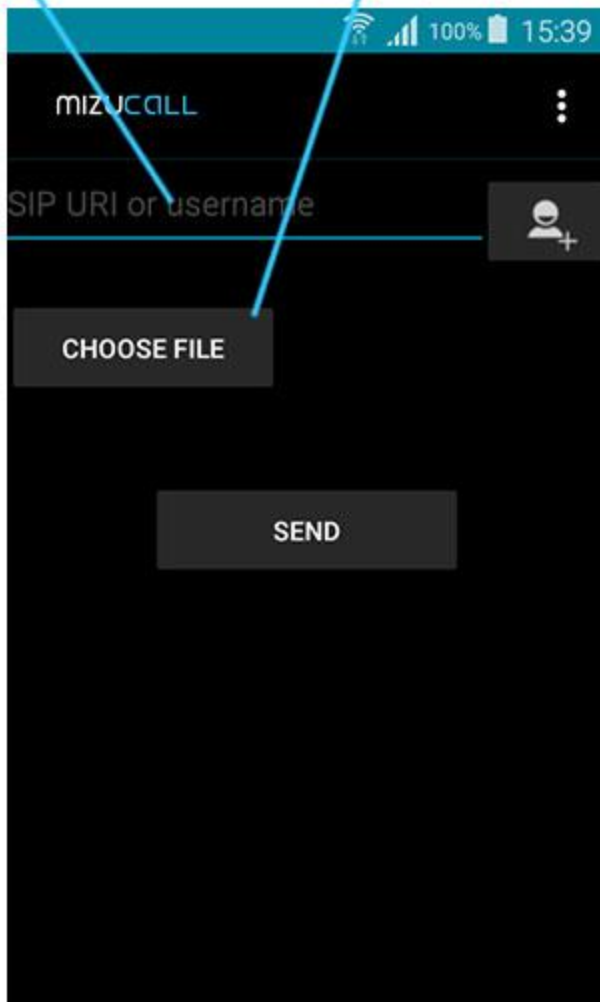
On the main menu there is a button for this feature, called Send file.

Once you fill tap this button, you will arrive to a new page, where you need to enter the receivers SIP URI or username and you will choose the desired file:



Here you can browse the desired file

Here you need to enter the SIP URI or username of the file receiver



## Other features

The SIP client has a long list of extra features built-in.

You can activate these features from the settings (discussed below) and/or from the menu items under various screens.

For example when you are in call a list of options will appear in the menu including mute, hold, Bluetooth and more.

## Configuration options

MizuDroid is a powerful SIP client with a long list of features and it has numerous configuration options to enable maximum flexibility. By default the softphone is already optimized thus you need to set only the basic mandatory



parameters to begin the usage: server, username and password. Some SIP service might require to set also the proxy address and the caller-id. Other than these all the other parameters can be left with the default values.

## Basic settings

The most important and mandatory settings are presented on the login screen:

**Server address** – The domain or IP:port of the SIP service. For example: sip.myserver.com, myserver.com:5075, 11.22.33.44 (default port is 5060, otherwise it must be specified)

**Username** - SIP account auth user name

**Password** - SIP account password

*Note: Some SIP server requires a different username/auth username. In this case go to advanced settings and set the caller-id/auth user name accordingly.*

The following additional parameters can be set from the basic settings:

**Dialer integration** – To be able to initiate VoIP calls from the native phone dialer

**Incoming calls** –The softphone will use push notification or a background service to allow incoming calls even when closed or idle

**Caller ID** – You might set a different username here used for A number display. Some servers has a different auth user name (Auth ID) and user name (Extension). In this case set the Extension as the Caller ID and set the Auth-ID as the Username on the login page.

**Display name** – Your full name to be displayed on the called side

**Theme** – Change the softphone's skin color

**Email** – Your email address

**Tunneling and encryption** – Use alternative encrypted transport

**Transport protocol** - Select transport layer (by default is UDP)

**Proxy address** – Outbound SIP proxy address required by some VoIP service providers

**Number rewrite** – Add destination number rewrite rules (add/remove/rewrite prefix)

**Call forward number** – Set call forward/transfer number used when call is rejected on busy or for auto forward/transfer

**Auto answer, forward or ignore** – Automatically handle all incoming calls

**On VoIP busy** – Set behavior if there is already a VoIP call in progress

**On phone busy** – Set behavior if there is already a native phone call in progress

**Access number** – IVR access number to call from mobile operator network

**Record calls** – If enabled, all calls will be recorded and accessible from the call history

**Enable logs** – Save trace to file for later uploading to technical support

## Advanced settings

We recommend to change these only if you know what you are doing or strictly necessary for your use-case:

### *General settings*

*Accounts* – Manage your SIP accounts/profiles. Use it only if you have multiple SIP accounts.

*Dialer integration* – When dialing with native dialer you can be asked if you want to use SIP or mobile

*Select theme* – Change the softphone's colors

*Email* – Your email address

*Ringtone* – Change your ringtone for incoming calls

*Vibrate* – Vibrate on incoming calls

*Keypad feedback* – Vibrate and sound feedback on keypad touch

*CPU always partial lock* – Keep a partial lock on the CPU to prevent deep sleep

*Force WiFi* – Keep the Wifi awake to handle incoming calls

*WiFi only* – Allow calls only on Wifi network

*Keep device awake incall* – Keep device awake during calls

*Unlock keyguard* – Automatically unlock phone keyguard on incoming calls

*Proximity sensor* – Turn off the screen while near ear

*Display notification* – Show notifications in phone notification bar

*Call history storage* – Specify where to store the call history

*Keep recordings for* – Keep call recordings for last specified days

*Screen rotation* – Control screen orientation

*New contacts* – Add new contacts to contact list

*Extra options* – Custom parameter can be set

*Record calls* – If enabled, all calls will be recorded and accessible from the call history

*Reset settings* – Reset settings to their default values

*Enable logs* – Save trace to file for later uploading to technical support

*Run in background* – Always listen for incoming calls (service will run in background to enable incoming calls)

*Enable push notification* – Use push notifications to receive incoming calls

*Kill on exit* – Kill application on exit

### *SIP settings*

The SIP softphone has a lot of basic and advanced SIP related settings.

By default all of them has optimal values, please don't change their values, just if you are sure what are you doing.

*Caller ID* – You might set a different username here used for A number display

*Display name* – Your full name to be displayed on the called side

*Hide my identity* – If you will enable this feature, then your identity will not be shown

*Transport protocol* - Select transport layer (by default is UDP)

*Realm* – Optional parameter to set the SIP realm if not the same with the server address or domain

*Use ICE* – Fast ICE negotiations (for P2P RTP routing)

*Use STUN* – Enable external address discovery (public IP discover)

*Use rport* – Rport usage in Via header

*Register* – Enable / Disable the registration to SIP domain

*Register interval* – Register time interval in seconds

*Keep-alive* – Keeps you NAT opened

*NAT open packets* – UDP packets to send to open the NAT devices (this should be changed just if you have NAT issues)

*Ring timeout* - Maximum ringing time allowed in seconds (default is 90 sec)

*Call timeout* – Maximum speech time allowed in seconds (default is 10800sec = 3hours)

*DTMF* – Choose DTMF (tone dialing) send method

*Chat/SMS* – Set messaging method

*Send early media* – Start to send media when session progress is received (by default it is disabled)

*Send media on muted* – Send RTP even if muted. Set to true only if your server is malfunctioning when RTP is suppressed. (by default is disabled)

*Mute direction* – Specify how to mute the calls when the mute button is pressed (by default only the microphone is muted)

*Ring on session progress* – Treat Session Progress (183) responses as ringing (180)

*Local signaling port* – Specify local SIP signaling port

*Local RTP port* – Specify local RTP port base

*Capability request* – If set, then will send a capability request (OPTIONS) message to the SIP server on startup

*Custom SIP header* – Set a custom SIP header (a line in the SIP signaling)

*DNS SRV* – If the SRV lookup returns multiple records, than the softphone will failover to the next server on connection failures

### *Media settings*

The android softphone with the default media settings should work smoothly on almost all android devices, but here you are able to fine-tune your calls quality, change codecs, perform volume related and hardware related modifications.

*Ringtone* – Change your ringtone for incoming calls

*Ringback for* – Generate ringtone for incoming and outgoing calls

*CPU speed* – Features are selected automatically based on your CPU speed

*Network speed* – Helps to choose the best codec and media enhancements for your calls

*Video* – Video related settings

*Audio codec* – Choose preferred codec

*Set final codec* – Workaround for multiple codec negotiation issues

*Disable wideband for PSTN* – Disable wideband audio codecs for calls to mobile/landline numbers

*AEC* – Acoustic echo cancellation. Turn on if your peers hears their echo and you have enough CPU power (at least 800 MHz ARMv6)

*Denoise* – Remove noise from the conversation (such as background noise or noise generated by the microphone). Enable only if your CPU is strong enough for extra processing (at least 800 MHz ARMv6)

*AGC* – Automatic gain control. Will control the volume automatically. Enable only if your CPU is strong enough for extra processing (at least 800 MHz ARMv6)

*PLC* – Packet loss concealment. Helps to improve the call quality on packet loss which occurs on low quality networks

*Silence suppression* – Do not transmit media when you don't speak. Enable only if your bandwidth is really poor or very expensive and CPU is strong enough for extra processing (at least 800 MHz ARMv6)

*Hardware media* – Use hardware media enhancements

*Volume in* – Set default microphone volume between 0 and 100%. 50 means no changes

*Volume out* – Set default speaker volume between 0 and 100%. 50 means no changes

*Media encryption* – Set media encryption method

*QoS* – Sets traffic class or type-of-service octet in the IP header for UDP packets

*RTP frame/packet* – Number of codec payloads in one UDP packet

*Double send RTP* – Double send RTP packets

*Jitter size* – The jitter buffer is used for smooth playback under variable bandwidth. The size is calculated dynamically, but its behavior can be modified with this setting.

*Audio buffer length* – Advanced setting for the audio playback

*Audio mode* – Select the output device when you switch to speakerphone

*Audio recorder* – Select audio recorder stream for calls. Should be changed only on special devices or if you have special requirements or the automatic setting is not the preferred

*Audio player* – Select audio player stream for calls. Should be changed only on special devices or if you have special requirements or the automatic setting is not the preferred

*Speakerphone player* – Select audio speakerphone player stream. Should be changed only on special devices or if you have special requirements or the automatic setting is not the preferred

*Ringer player* – Select audio player stream for ring. Should be changed only on special devices or if you have special requirements or the automatic setting is not the preferred

*Auto Bluetooth* – Auto use Bluetooth when available

*Speaker mode* – Select speaker mode

*Focus audio* – Focus on VoIP audio

*User routing API* – Use routing API for compatibility reasons. Some (usually old) devices might need this for correct audio handling

### *Call divert settings*

The call forward, transfer and prefix rules related configurations can be done here.

*Number rewrite* – Add destination number rewrite rules (add/remove/rewrite prefix)

*Tech prefix* – Add any prefix for all called numbers

*Call forward number* – Set call forward/transfer number used when call is rejected on busy or for auto forward/transfer

*Auto answer, forward or ignore* – Automatically handle all incoming calls

*On VoIP busy* – Set behavior if there is already a VoIP call in progress

*On phone busy* – Set behavior if there is already a native phone call in progress

*Native call on VoIP busy* – Set behavior of native call if there is already a VoIP call in progress

*Voicemail* – Set voicemail access number

*Callback number* – Callback access number

*Callback mode* – Choose when to use callback if any

*Access number* – IVR access number to call from mobile operator network

*Access number mode* – Choose when to use access number if any

*Transfer type* – Choose call transfer method

*Transfer with replace* – Specify if replace should be used with transfer so the old call is not disconnected but just replaced

*Mute on new call* – Specify whether existing calls should be muted on a new call

*Hold on new call* – Specify whether existing calls should be held on a new call

*Enable direct calls* – Specify whether to enable direct call to SIP URI

## Troubleshooting

In case if the softphone can't connect or register, double check that you typed your credentials correctly (server address, username, password).

Some servers might require a different SIP **user name** (sometimes called as extension) and **auth user name** (sometimes called as Authentication ID). In this case you must set the auth user name as the MizuDroid username on the login screen and set the user name as the Caller-ID in the Settings.

Some SIP services requires an **outbound proxy**. You can set it in the Settings as Proxy address.

If you still have any issue with connect, register or call, **try with a third-party softphone** first to make sure that the problem is not with your server or configuration. You can use Zoiper or X-Lite for example.

If the problem persists, you can use the [forum](#) to ask for help.

If you have or plan to purchase a paid corporate, customized or bulk license, then **contact Mizutech support** by [email](#). You should also [upload a log](#) to support.

See the FAQ below about specific issues and their resolutions.

## FAQ

### How to use ?

To be able to use the softphone, you need a SIP account (at any SIP service provider or your own IP-PBX).

In the settings login section, enter the followings:

- SIP server address (domain or IP and port if port if not the default 5060. Example: sip.myserver.com:5060)
- SIP auth username (Auth ID)
- SIP password

Optional settings:

- If your SIP service requires an outbound proxy then go to the Settings and enter set the Proxy address accordingly.
- If the user name (extension) is different from the auth user name (Auth ID), then go to the Settings and enter it as Caller ID.
- If your server requires TCP instead of UDP then go to the Settings and set the Transport Protocol accordingly.

Save the settings (tap on OK) and the softphone should connect/register.

Now you are ready to initiate and accept phone calls.

During calls you have some additional options such as call transfer, hold, loudspeaker and others (icons are displayed or press the menu for more).

### I don't have a SIP account

MizuDroid can be used with any SIP server or service provider, however it doesn't come with it's own SIP account. You must have a valid SIP account to be able to use this application (on your own IP-PBX or at any SIP service provider).

You can create a SIP account at any SIP service provider: <https://www.google.com/search?q=sip+service+provider>  
Creating a SIP account is usually free at any service provider. Calls and IM between users are also usually free, however you might need to pay for calls to mobile/landline.

Mizutech also offer [SIP service](#).

You can create free SIP account [here](#).

### What is MizuDroid?

MizuDroid is a VoIP softphone for Android using the open standard SIP protocol. MizuDroid is not tied with any VoIP provider. You are free to use it with any VoIP server.

The application is trying to use as much extra features as possible depending on your device/server/network capabilities.

You can find more on MizuDroid homepage: <https://www.mizu-voip.com/Software/Softphones/AndroidSoftphone.aspx>

With MizuDroid you can do the followings:

- connect to any SIP server
- free calls to other SIP softphones
- make (cheap) calls to VoIP/landline/mobile/international (depending on your VoIP service provider)
- exploit SIP features such as presence/chat/CallerID/forward/transfer/hold/file-transfer/conference and others
- extra features such as file transfer or BLF

You can also upgrade to the Pro version which offers more:

- commercial usage allowed
- more features
- broad codec support
- enhanced VoIP tunneling and encryption support
- enhanced conference support with local RTP mixer
- and more

Choose the Pro version for business or professional usage.

## What is SIP/VoIP?

VoIP means "Voice over Internet Protocol", a few protocols to allow voice calls to be made over IP networks. SIP is the newest VoIP standard adopted by many organizations for their telecom infrastructure. By using the open standard SIP protocols you can keep yourself free of locked-in solutions such as Skype and you are free to use any standard based network for your telecom needs.

More details:

[https://en.wikipedia.org/wiki/Voice\\_over\\_IP](https://en.wikipedia.org/wiki/Voice_over_IP)

[https://en.wikipedia.org/wiki/Session\\_Initiation\\_Protocol](https://en.wikipedia.org/wiki/Session_Initiation_Protocol)

<https://www.voip-info.org/>

You can make VoIP calls in the following ways:

- direct IP to IP calls (this can be done by MizuDroid by typing the peer full URI: sip:user@IP:port)
- using your own VoIP server such as Asterisk on Linux or Mizutech VoIP server on Windows servers
- using a VoIP service provider

## Settings/Features

All the settings except serveraddress/username/password is optional. Change them only if you have some special needs. There are some easy to use settings such as ringtone, call forward or number rewrite which you might wish to modify to enhance your VoIP experience.

However the advanced settings (Setting page -> Menu -> Show advanced settings) should be modified only by experts or if you have some issues using the softphone or some specific needs.

You can find a feature list on the MizuDroid [Android SIP softphone homepage](#).

A few useful settings:

- Registration to a server: by default the MizuDroid will try to register to the server what you have set to allow incoming calls, but this is optional (if you need only outbound calls) and can be disabled from the "SIP Settings"
- Transport protocol: UDP (the default) usually works with all servers. You might try TCP to save some battery
- Proxy address: set this only if you are instructed so by your VoIP provider

- Tunneling and encryption: VoIP is blocked in some network such as behind corporate firewalls, several countries. This is when the tunneling will help you. The tunneling can also circumventing NAT issues. With encryption both your media and signaling will be encrypted. By default the "tunneling and encryption" setting is set to "automatic" which means that it will be used only if otherwise you could not connect. Set to "Never" if you encounter any issues or to "Always" to encrypt all your communication (only in the Pro version)
- CalledID display: by default it is your username, but you can also set a "CallerID" and/or "Display name" from the "SIP settings" section. Note: not all SIP server accept a different user name/auth user name.
- call forward always: from the "Call Divert" setting, set a "Call forward number" and then set the "Auto answer" setting to "Forward"
- call forward on busy: from the "Call Divert" setting, set a "Call forward number" and then set the "On phone busy" and/or "On VoIP busy" setting to forward calls
- Dialer integration: when you try to make a call from the native dialer, it can offer SIP call option. This can be set from under "General settings". Use the "Allow native calls"/"Don't allow native calls" to control which numbers should be handled by the softphone.
- Incoming calls: When set to "Yes always", it means that the softphone will use push notifications or a background service to remain always available for incoming calls (Service is used only if push notifications are not available for some reason such as if your SIP server is on a local LAN with private IP). When set to "Yes while running" then you might not receive incoming calls if the softphone is closed or the device is sleeping. When set to "No" then incoming calls are always rejected.

### How much bandwidth is required?

When you are not in a call, the bandwidth usage is not significant (around 15 MB for an entire month if you are constantly registered to a SIP server)

For voice calls: between 10 and 80 kBit/s depending on the codec (around 0.8 MB per minute while you are in call)

For video calls: around 180 kBit/s

### Does it work in all mobile networks?

Mobile phone operators are often blocking VoIP to force the users to use their (expensive) call facility.

MizuDroid comes with built-in tunneling and encryption to bypass this kind of blockades.

By default the tunneling is activated only if otherwise clear SIP/RTP doesn't work.

This will also help to bypass all firewalls, NAT's or HTTP proxies.

### System requirements?

MizuDroid is optimized for both the latest Android OS and old Android OS versions.

Any android since 1.4 ARM or x86 (99.99 of the total android devices). It runs well also on ancient low-end devices.

Some features are working properly only if your device CPU speed is high enough (at least 800 MHz ARMv6). MizuDroid will automatically guess your device speed and will fine-tune the default settings regarding your device and CPU capabilities. For example advanced AEC and G.729 requires at least a mediocre CPU and opus wideband requires a modern CPU like (mid-range phone on better).

### VoIP server compatibility

MizuDroid is compatible with all VoIP servers using the open standard SIP protocol. You are free to create a VoIP account at any service provider or use it with your own softswitch/IP-PBX such as Asterisk.



How can I get a customized android softphone for my company

Contact Mizutech by email at [info@mizu-voip.com](mailto:info@mizu-voip.com)

Where can I find more details about MizuDroid

Visit the [SIP Android softphone software homepage](#).

Fine-tune performance/quality

MizuDroid is capable to auto-fine tune itself depending on your device hardware and network capabilities.

- If you are using Wifi or 4G/5G and your CPU is fast enough (better than 1000 MHz ARM7 or multi core) then you can fine-tune the quality by the following settings:
  - codec: leave on auto or try to force Speex wideband
  - RTP Frame/Packet: set to 1
  - Jitter size: set to small
  - enabled aec, denoise and agc
- On a low-end device you can try the followings:
  - enable PCMU only
  - disable aec, denoise and agc (these are found under "Media settings" if you enabled "Advanced settings")
- On slow networks (2G/3G):
  - RTP Frame/Packet: set to 6
  - Jitter size: set to big
  - use a low bandwidth codec: G.729 or iLBC if your CPU is fast or GSM if your CPU is low-end

In new versions you can do all these optimizations and more with a single setting: Go to the Advanced settings -> General settings -> Call quality and change it after your needs.

Fine-tune battery usage

MizuDroid is optimized by default for minimal energy usage which can be further fine-tuned with the following settings.

- if you use the softphone only for outgoing calls (not to accept incoming calls) then you can stop the listener from the "Incoming calls" setting option
- try to use TCP transport if supported by your server
- if TCP is supported, disable keepalive by setting the interval to 0 and also increase the Register interval (3600 should be fine)
- use push notifications instead of background running service

In new versions you can do all these optimizations and much more more with a single setting: Go to the Advanced settings -> General settings -> Availability and change it after your needs.

## Cannot start

Contact us with your hardware/software version.

## Cannot register

- make sure that your sever address is typed correctly
- make sure that the "Register" option is actually selected
- make sure that your server is listening on UDP default SIP port (5060). Otherwise change the settings accordingly
- make sure that you use a valid username/password
- make sure that you have an internet connection (your browser is functional)
- check the [troubleshooting](#) section

## Username vs Auth user name

Some softswitch (such as 3CX) or VoIP service provider require to set a different username and auth user name. In this case go to advanced settings and

- Set the Username as Caller-ID (This is usually the extension number)
- Set the Auth user name (username used for authentication or authentication ID)

## Cannot initiate calls

- enable only one codec (for example PCMU) to circumvent codec negotiation issues
- make sure that your account has enough balance if you try mobile/landline/international calls
- check the [troubleshooting](#) section

## Doesn't receive incoming calls

To be able to receive calls, SIP service must support push notifications (directly or via a gateway) or the softphone must be connected/registered to the SIP server.

Make sure that the "Incoming calls" option is set to "Always"

You might set the "Run in background" (from Advanced settings -> General settings) to Yes and make sure that the "Register" option is checked.

For the register interval you might set a lower value such as 60. The keepalive interval might be also adjusted to 25. Some devices might close the internet connection when the phone is idle. To prevent it, you can set the Force wifi to always and maybe also the "CPU always partial lock"

Another reason can be if your server doesn't handle NAT properly.

For this you might need change the STUN and rport settings (set them to no or try with always yes).

Also make sure that the keepalive interval is set below 100 seconds.

If the softphone still doesn't receive calls for some reasons and always availability is very important for you then you might force "Always running" from Advanced settings -> General settings -> Availability.

### Calls are disconnecting

- enable only one codec (for example PCMU) to circumvent codec negotiation issues
- make sure that your account has enough balance if you try mobile/landline/international calls
- enable/disable stun
- check the [troubleshooting](#) section

### Crashing at call setup or after call

- Disable G.729, aec, denoise and agc
- Don't use loudspeaker and Bluetooth

### No audio or one way audio

(The other party can hear you but you can't hear the other end or reverse)

This is commonly caused by two reasons:

NAT/SIP issues:

1. Set the "setfinalcodec" parameter to 0 (especially if you are using Asterisk or OpenSIPS)
2. Set usestun and userport to No (especially if you are using SIP aware routers). If these don't help, set them to "Always yes"
3. Make a test call with only one codec enabled such as PCMU (this will solve codec negotiation issues if any)

Audio device issues:

Try to experiment with the following settings:

-audio mode

-speaker mode

-use routing api

If you have succeeded to make it work, please share your results with us so we can improve further versions.

### Cropping sound

- disable aec, denoise and agc
- your cpu usage might be high: enable only PCMU
- set the jitter buffer to big

- turn off permanent location updates from your phone settings

### The remote party hear itself back (echo)

By default the softphone will guess the optimal AEC settings depending on the device capabilities, speaker mode and other factors.

You can further fine-tune AEC (acoustic echo cancellation) if needed, with the following settings:

- From the media settings enable AEC: software, native and fast
- You might also enable Denoise

Echo is generated by some devices/circumstances when the output audio signal (from the speaker) arrives back to the input signal (microphone). This usually happens only if you use loudspeaker.

Please note that acoustic echo cancellation is not an exact science and its success rate depends on many factors such as device, OS version, audio chip, audio driver, environment, network delay, codec, loudspeaker usage and others.

Our app is capable to remove more than 95% of the echo for more than 95% of the calls. There are some circumstances (depending on the factors listed above) when echo cannot be detected correctly and it is left in the voice signal. This happens for less than 5% of the devices.

### You hear yourself back (echo)

In this case the echo is generated by the other end and there is little to do with it on the android softphone side.

Make sure that the other party uses an AEC capable SIP client or device.

### Bad voice quality

Call quality is influenced primarily by the followings:

- Codec used to carry the media: fine-tune with the Audio codec setting
- Network conditions (check also your upload packet loss/delay/jitter): make sure that you have enough bandwidth/QoS for the codec that you are using. adjust the jitter (big/small) and the frame/packet settings
- Device capabilities (CPU power):
- Hardware: enough CPU power and device microphone/speaker quality: disable agc, aec, denoise, G.729, wideband
- (Missing) AEC and denoise: enable/disable after your needs
- Audio device issues (many device vendors are handling the audio differently): fine-tune from "Media settings"
- Issues at your ISP or VoIP service provider

See the "Fine-tune performance/quality" and "Cropping sound" FAQ point for more settings.

### What is the best codec?

There is no such thing as the "best codec". All commonly used codec's present in the softphone are well tested and suitable for IP calls.

This depends mainly on the circumstances.

For internal IP to IP calls we recommend OPUS.

For outbound calls we usually we recommend G.729 since this provides both good quality and good compression ratio. If G.729 is not available in your license plan, then the other codecs are also fine (GSM, speex, iLBC) Otherwise the G711 codec is the best quality narrowband codec. So if bandwidth is not an issue in your network, than you might prefer PCMU or PCMA (both have the same quality) Between softphone users (or other IP to IP calls) you should prefer wideband codec's (this is why you just always leave the OPUS and speex wideband with the highest priority if you have calls between your VoIP users. These will be picked for IP to IP calls and simply omitted for IP to PSTN calls) To calculate the bandwidth needed, you can use this tool. You might also check this blog entry: [Codec misunderstandings](#)

## How to switch between accounts?

You can switch between accounts by going to Settings -> Accounts OR Settings -> Advanced Settings -> General Settings -> Accounts and tap on the desired account

## IEC

IEC means international escape code such as + or 00) before the numbers.

Create rule(s) from below Call Divert settings/Number rewrite.

Examples (rewrite prefix A to-> B):

In Hungary: 06 to 0036

In US: +1 -> 1 or + -> 011

Alternatively you can install the [Contacts Cleanup](#) to normalize all the numbers in your phone-book:

When integrated to android, i want some numbers to keep on native while others on VoIP

Use the "Allowed prefixes" / "Not allowed prefixes" filters (prefixes separated by comma)

## No ringback tone

Depending on your server configuration, you might not have ringback tone or early media on call connect.

To fix this, set the "sendearlymedia" parameter to true or set the "natopenpackets" parameter to 10.

## High cpu usage

- enable G.711 (PCMU/PCMA) codec only
- disable aec, denoise and agc (these are found under "Media settings" if you enabled "Advanced settings")
- follow the [Fine-tune performance](#) advices

## My phone is hot during a call

Your phone might become hot during calls due to high CPU usage. This is usually not a problem to be addressed as the battery usage is much more affected by the idle behavior, except if you make a lot of VoIP calls every day. Usually this is caused because by default the SIP softphone will tend to select the best possible codec which is still supported on your phone (often OPUS wideband if also supported by your SIP server and peers) which might result in high CPU usage. This can be resolved by just forcing a codec with low computational requirements such as PCMU. Follow the above "High cpu usage" instructions described above for more details.

## Known issues

- The handling of the audio devices are not consistent across Android devices due to the differences introduced by the device vendors. We are trying our best to cover most of the devices, but there can be some (mostly exotic) phones where the default settings might not be optimal. In this case you can adjust the configuration from the "Advanced settings" -> "Media settings".
- The Video feature requires RTC and might not work if your server is on Local LAN with private IP (handled automatically for SIP servers available over the public internet)
- We offer Bluetooth support only "as-is" due to high device fragmentation (no email support for this feature).

## Got worst results after setting change

Use the "Reset settings" option to revert to default settings.

## I can't solve the problem

If you have any issue with connect, register or call, try with a third-party softphone first to make sure that the problem is not with your server or configuration. You can use Zoiper or X-Lite for example.

If the problem persists, you can use the [forum](#) to ask for help.

If you have or plan to purchase a paid corporate, customized or bulk license, then contact Mizutech support by [email](#).

## How to send log?

Follow these steps to get the detailed logs:

1. Enable logs from the settings ("Enable log" checkbox in the basic settings) help menu (Help/Logs/Start collecting logs)
2. Restart the application (via "Exit" menu or back, back)
3. Reproduce the problem (if it is related to server connection and registration than just wait a few seconds. If it is related to call, then make a test call)
4. Send the logs: Help menu -> Send logs to support with a short description of the problem.
5. Write an email to [info@mizu-voip.com](mailto:info@mizu-voip.com) with the description of the problem and include also your SIP username (this is necessary so we can find your log file)

Also described [here](#).

Technical details

MizuDroid is based on the Mizutech [Android SIP SDK](#) which is a high performance SIP library for the Android platform.

How to test?

In case if Mizutech send you a new trial/demo/release build, you can either [upload it to the Google Play](#) app store or [install from file](#).

Submit bugs or suggestions

Please use the [forum](#).

Contact us

Contact us if you are interested on a custom build or corporate bulk license

<https://www.mizu-voip.com>  
[info@mizu-voip.com](mailto:info@mizu-voip.com)

Home page

[Android SIP client](#)

## Resources

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