

Mizu VoIP Server Tutorial

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About

Mizu VoIP Server is a Class4/5 softswitch application running as a service on the Microsoft Windows operating systems.

Modules: SIP stack, H323 gateway/gatekeeper, SIP-H323 protocol converter, WebRTC, RTMP, access roles, routing (priority, weight, BRS or LCR), failover, load balancing, quality routing, e-payment, billing, PBX, accounting, CDRs, blacklist/whitelist filtering, callcenter, IVR, callback, calling cards, transcoding, call recording, conferencing, media server, alerting, statistics generation, watchdog, enduser web portal, client applications, API and others.

Install

The Mizu VoIP server is available in three editions:

- Free edition with unlimited ports but less performance due its embedded compact database (might not be suitable for VoIP service providers). Download from [here](#) (unlimited free) and follow the install wizard.
- High performance commercial edition with full database backend: this doesn't have an automated installer but it can be easily installed following the [install guide](#). Download from [here](#) (the downloadable version comes with 100 users, 10 concurrent calls and 2 sip trunks limit)
- SaaS VoIP service with server hosting by mizutech. Order [here](#).

Firewall/NAT consideration:

The Mizu Softswitch includes advanced NAT handling capabilities and it can be hosted also behind a NAT/router/firewall.

A VoIP server uses lots of different ports thus port based firewalls are not recommended (the built-in Windows firewall is enough since it can perform application level filtering).

We recommend to host your VoIP service on a server with public static IP unless you plan to use it only in your internal network (with no external peers).

If for some reason this is not possible, then you will need to allow (on your external firewall if any) and forward (on your router/NAT if any) all ports required for signaling and the RTP port range.

Paid license:

Contact our [support](#) if you are interested in a paid license as in this case the Mizutech support team can handle all the installation and configuration tasks for you for no extra costs.

MizuManage

All administration and monitoring tasks can be done from the **MizuManage** (MManage / MizuManagement) remote administration client. If you don't already have this application installed, then you can download it from here:

https://www.mizu-voip.com/Portals/0/Files/MizuManagement_Setup.exe

For the Compact Edition MManage is always preinstalled. If you are using the Compact Edition then don't use this download link! Also login is not required with the Compact Edition as this version will auto-login.

Quick Tip: MizuManage can be used from your VoIP server itself but also remotely, so you can manage your server(s) from your PC.

Login to MizuManage:

- Server: address of the server (and database port followed after a comma if not using the default port)
- Instance: database name ("mserver" by default)
- Node: node number (only if you have multiple app service instances)
- Username: database username ("sa" by default)
- Password: database password

Example:

Server: 127.0.0.1

Instance: mserver

Username: sa

Password: srEgtnkj34f

Quick Tip: if you have multiple servers then you can name it in the „Server“ input by entering any name and then the server address in parenthesis. For example myserver (11.22.33.44)

MManage is a modular MDI application: you can open the various forms from the left-side tree-view control or from the menu.

Nearly all forms can be filtered with the following filters:

- **Quick filter:** found in the top-left side in MizuManage. With the Quick filter you can search in almost any forms (Users, CDRs and others). For example type "44*" in the quick filter box then open the "CDR" form and click "Load". You should be able to see all calls to 44..... numbers. Or enter "test" and open the "Users and devices" form. Click on the load button to see accounts containing the "test" word (in name, username, address, etc)
- **Direction filtering:** accessible by double clicking on the space below the quick filter or from the Fields menu -> Filter -> Set direction filter. When you are doing operation which needs more precision (eg. billing), always use the Set Directions form and not the quick filter.
- **Date-Time filter:** found in the top-left side in the MizuManage. Useful to restrict statistics, reports and CDR listing intervals.
- To export data from the application, use File menu -> "Save As". A more advanced export tool can be accessed from File menu -> "Export/Import" or you can also use the SQL Management Studio if you have basic SQL skills.

- From the “Control” menu you can Start/Stop/Restart your service or reload its configuration.
- From the “Fields” menu you can manipulate the selected dataset (grids, etc).
- Use the Config menu to setup your server.

Quick Tip: you can filter almost any forms using the quick search, directions and date-time filter.

Quick Tip: always check the status bar (the bottom text display). MManage rarely displays popups and the success/error status of various operations is displayed on the status bar instead.

Basic configuration

For the basic server configuration you should walk through the **configuration wizard** accessible from the Config menu.

- Don’t change any setting that you don’t fully understand, just click on the “Next” button in this case. Most of the settings are self-explanatory with a short description near them and/or a hint if you hold the mouse over a control.
- The Mizu softswitch will automatically configure its NAT handling based on the circumstances and the settings you provide on the configuration wizard “Network” page.
- Take special attention for the Bind IP setting if your server has multiple IP’s assigned.
- Take special attention for the Public IP setting if your server has multiple IP’s assigned or your server is behind NAT.
- It is recommended to set the SIP signaling listening port to 5060 (this is the standard for SIP and set by default)
- It is recommended to set the main API port to 80 (Set to 8080 if some other app is already using port 80 on your box such as a web server)

If your server is behind NAT and you wish to access it from the internet, make sure to forward the required ports on you router (UDP 5060 for SIP signaling, UDP port range for RTP and TCP 80 for API, web access and other services). You can verify from Help menu -> Connectivity test.

- We recommend to always assign a (sub)domain (such as sip.yourdomain.com) to your SIP server IP so lately you can easily change your server or IP address without the need for your customers to reconfigure their device. If you already have a domain name (for example for your web site) then a sub-domain can be obtained for free from your domain name provider (usually domain name providers has a web interface where you can manage your domain and add as many sub-domains as you need).

After you have finished with the configuration wizard you might have to continue with the following tasks:

- add users: Access -> Users and devices -> Enduser
- add your outbound routes and traffic senders: Access -> Users and devices -> Sip Proxy and Traffic Senders, Routing
- fine-tune other settings: billing, blacklists, etc.

For more advanced options you can change global config options manually on the “Configurations” form. (Under the “Other” node in the tree view).

If you are not sure where to find a specific configuration option, search for your keyword in the “Configurations” form and also in the [admin guide](#).

Quick Tip: right click on the “Configurations” node in the main tree view to have some grouped options.

Server Start

Once you have gone through the configuration wizard, you can start the service (From Control menu -> Start server or from Windows Service manager locate the “mserver” entry and right click to start).

Once the service was started open the “Analyze” form to check for any potential issues.

The server will allow calls between endusers by default, so you can already test by registering with 2 softphone (such as X-Lite) and make calls between them. See the [first test calls](#) section for more details.

Listing users

Users can represent real people, extensions, devices or virtual endpoints.

Open the “**Users and Devices**” form (below the “Access” node) and (double) click on Endusers. You can apply various filtering using the already discussed quick filter (top-left) or use the “Type” dropdown-list (below the quick filter).

For registration details see the “**Registrar**” form below the “Monitoring” node.

Quick Tips:

- You can easily search for users using the “quick filter” box. For example to list all outbound routes whose username or name contains the “carrier” word, select the “SIP Server” and type “carrier” in the quick filter then hit Enter. The quick filter will also search in other fields such as IP, name, email and others.
- There are many fields in the user table which can be used for various settings. To quickly find a field, double click on a user node (such as “Endusers”) then click on the “Search fields” below the grid. Another way is to use the “Show Fields” from the “Fields” menu
- Try to right click on the “Users and devices” node in the main form tree view or the user type node from within the “Users and devices” form to see additional options.
- You can list registered users by selecting the “Endusers” node and then the “Active” filter (below the Quick Filter) or right click on the users node and select “Registered”

Creating users

The best way to create new users is to clone an existing working account with the same user type.

For this, launch the “**Users and Devices**” form, select a user type, and click on the “Load” button. Then select any user entry and click on the “New User” button.

- **Endusers** are the most commonly used type (retail customers/company employees, known as “extensions” in PBX systems). Usually you will select “Username/password” authorization for this type of users and enter a valid username and password. The username can be also used as a real phone number; will be used also as the caller-id if not specified otherwise. Endusers can make voice or video calls between them for free of charge and also IM and presence are enabled by default. By default the server will route the RTP if needed (if users are behind NAT) or allow it to bypass your server saving your bandwidth. **Sub-Endusers** can represent VoIP devices, extension, child accounts or callshop cabins. Calls from sub-endusers account are billed for the parent Enduser.
- **Traffic sender** users are used for receiving traffic from other SIP servers and carriers. The authorization type is usually set to “Auth ip must match” and you have to enter a correct “Auth Ip”

(IP address based authentication). If you don't have special requirements, the only thing that you have to communicate to your peers to be able to send calls to your server is only your IP address. (Your server needs a public IP for this or you have to setup proper port forwarding).

- For outbound traffic you need one (or more) **SIP Server** user. These are your outbound trunks. The most important parameter here is the "IP" where the VoIP calls will be sent. Then you will have to configure these also on the Routing form. To be able to send and receive traffic to/from another SIP server or carrier you will have to add it as both a "traffic sender" and "sip server" user.

To **import endusers** from other data sources, go to Config menu -> Users.

You can also generate **users in bulk** from the "Generate users and PIN's" menu item.

Users can also sign-up to your service with the following methods:

- From the enduser web portal ("New user" link on the login page)
- From customized softphones (all of them have a user interface where new users can sign-up to your service)
- Integrate the sign up capability into your web site or any application using the Mizu VoIP server newuser/adduser API

By default the new users can already call each other for free (unless you set otherwise). If you use the server for your own company, then you can create new user record as postpaid so they will be able to call outside by default. Otherwise the user records are created as prepaid and must have positive credit balance to be able to initiate outside calls.

Quick Tip: Right click on the "New" button for other options to add users. Multiple endusers can be added more comfortably from "Add Enduser" (or "Bulk generate" if you need many with random credentials)

Accounts, DID numbers and extensions [optional]

Unless in other traditional PBX's, in the MizuVoIP server you can just add a new user with the "username" and "password" settings to be reachable for incoming calls also. Then the "**username**" field (which can be a phone number) will act as a SIP username for authentication but also as an extension number or a DID number. You can also use the same username/password to login on the enduser web-portal and in any other operations requiring authentication. If you wish to use a separate username and CallerID for the users, then you can enter the Caller ID to the "**Other numbers**" edit box. If one user has multiple DID number assigned, then you can add them using the "..." button near the "Other numbers" edit box (Users and devices form, Edit tab). More than one users can share the same DID number: simply add it to the required users as "other number" with type 0. The call will be routed to the "best" device (based on the user status whether it is registered or in-call). You can also setup ring-groups to allow forked calls to ring on multiple devices. If you need a more specific routing, use the "Routing" form.

Quick Tip: you can buy real phone numbers for your endusers from providers like [DIDx](#). Some carriers or VoIP trunk service providers might also provide real DID numbers for you for no extra cost if you terminate your traffic at them.

Setup outbound routing

This is needed if you would like to route the calls from the users to another servers, gateways or carriers (for example calls to mobile and landline numbers).

Add a “**SIP server**” user from the “Users and devices” form with any meaningful username and make sure to set the IP field correctly on the Edit tab. (This field can also contain a domain name instead of IP address). Some carriers require the usage of a techprefix which can be entered in the “Tech Prefix” field. Others settings are rarely used.

You must also add this server(s) to your routing:

Open the “**Routing**” form. In the left side you have to define your pattern which will restrict the condition when the actual route entries can be used. If all fields are empty and the time definition is set to “All times” then all patterns will match. You can make restriction if you make specification here (caller, called prefix, time restriction, etc) . Make sure that you increase the priority for the pattern (to be higher than the your “general” pattern where you have not made any restrictions)

On the right side you will have to add one or more sip proxy user. If you set more than one route with equal priority, then you have load balancing, LCR or BRS (depending on the “brs_lcr” global config option); otherwise the traffic will be routed after the prioritizations (will flow to the lower priority servers only if you have reached the maximum port limitations or because automatic failovering). For more details please read the [routing guide](#).

Quick Tip: Make sure to add all your outbound SIP servers to the routing, otherwise they will not be used at all.

Register to outbound server [optional]

Usually for a B2B usage, uppers servers (your carrier or VoIP service provider) will setup IP based authentication. This is the favorite method for a trunk interconnection. If your outbound server (where you are sending traffic and receive incoming calls) needs username/password based digest authentication instead of IP based authentication, you can set it from the “SIP server” user configuration. Create a **SIP server** user, then switch to the “Edit” page. On the bottom of the page you can find a grid named “**Proxy authentication**”. Here you can add the login details (multiple username/passwords can be used for your convenience). Then select “Username/password must match” from the “Authorization” drop-down list.

These are the basic and most commonly used authentication settings. There are many other combinations, for example you can forward the username/password as received from your users. For more details please consult the Admin Guide.

Setup inbound routing [optional]

If you would like to accept traffic from other servers (for example you are doing a wholesale business or to offer VoIP trunk services for other companies), then you have to create **“Traffic sender”** user. Usually you can use IP based authentication. For this, add the peer IP to the **“Auth IP”** field. Then the traffic sender software or service have to be configured with your server address to send traffic to you (IP:port or domain:port where port is usually 5060).

For each incoming calls, the server will first check if the called party is a local user. If not, than the call are routed regarding the rules which is set by the **“Routing”** form.

Actually you could also use **“Enduser”** users for the same thing, but for a bigger traffic volume it is always to differentiate normal endusers from **“traffic sender”** so your statistics will be easier to understood.

Quick Tips:

- You can easily create a transit server by using only **“Traffic sender”** and **“SIP server”** users.
- One **“Traffic sender”** user entry can handle incoming traffic with IP authentication from multiple IP addresses . Use the **“...”** button near the **“Auth IP List”** to add more IP address.

First test calls

For a test call create 2 enduser accounts with username/password authentication.

To find out how to connect to your server see the **“Client configuration”** from the Help menu.

Register with two softphones and call from the first account to the second account.

Softphone configuration:

- domain: your server IP or domain name (and the server port if your server is not using the standard 5060 UDP port)
- proxy: you can leave it empty
- username: the **“username”** field from the newly created user (tb_users.username)
- password: the **“password”** field from the newly created user (tb_users.password)

No any other special settings are required (such as NAT, STUN, etc).

The network setting should be automatically handled by the server. If you don't hear any voice you might change the RTP routing for the user(s) to **“always route RTP”** from MizuManage -> Users and devices -> Edit tab.

During the call, you can open the **“Current calls”** form in the MizuManage to see the details.

After the call you can see the CDR by opening the **“CDR”** form in the MizuManage.

If there are no CDRs, it means that the call has not reached the server (wrong network settings on server or client side).

Troubleshooting:

If you can't register or make calls, check the followings:

- Have a look at the **“Dashboard”**
- Have a look at the **“Analyze”** for to detect any potential issues
- Check any important errors or warnings on the **“Logs”** form
- If you are making call to a local user, make sure that the called user is registered when you call it
- In case of call failure you can check the disconnect reason from the **“CDR”** form

- Verify the log files. You can find the logs in the server app folder (near the mserver.exe) -> “logs” subfolder. You can also open the folder from MManage -> Files Menu -> Folders -> Server logs directory.
- Open the last logfile (“log_XXX.dat”) with a fast text viewer (For example F3 from TotalCommander). To find application errors, open the last log file in the server app directory and search for “ERROR” and “WARNING”. To find a call, search for “INVITE sip:callednumber”.
- You can modify the trace details with the “loglevel” configuration option: from 1 to 5.

Setup billing [on commercial servers]

User to user calls are not charged by default (this can be changed with the “internal_endusercost” global configuration option).

All other calls are checked against the user credit and prepaid/postpaid setting which can be set from Users and devices form (select a user and switch to the “**Billing**” tab)

From here you can assign a billing packet for the user explicitly but the better way is to setup one or more packets to be valid for all your users, for a group of users or on special circumstances (caller, called, techprefix, time, etc)

These packets (call rating) can be set on the “**Price setup**” form (below the “Billing” section)

- On the left column add a billing group with any name (“default” is ok). This is used only to logically group your tariffs but not used by the server.
- On the middle column specify your conditions. You should have at least one Enduser cost type without any further restriction on the traffic direction (so it will be applied for all endusers/directions/time)
- On the right column enter or import your pricelist applied in the conditions defined by the middle column.

For a default price enter prefix “*” (this will be applied to all destinations that is not specified explicitly) Make sure you have set the proper currency (in the global configuration, in the price setup and also for your users)

Read the [Billing guide](#) for more details.

The users can recharge their credits with various built-in methods:

- by pin code (recharge cards), calling cards (“Pincodes” form)
- PayPal (can be set from the “Configuration” form)
- http and database API
- credit transfer between users (from/to)
- postpaid/invoice (invoices form)
- ePayments, credit card payments by integration with a payment gateway. Any third party payment method can be added (see the [database interface](#) and [http interface](#) documentations or by extra customization/integration work by the Mizutech development team)

These payment methods are accessible for the endusers from:

- the built-in enduser web interface
- client applications (built-in the softphone)
- your own custom interface (using the softswitch http and/or database API)
- IVR
- SMS
- desktop, mobile or web applications

Monitoring

The Mizu server provides endless possibilities for monitoring both real-time and statistics. Some of the most important tasks are the followings:

Dashboard: a summary of the most important parameters and a start point for management

List the active sessions: Monitoring -> “Current Calls” form

Call detail record: Monitoring -> CDR form

Statistics by users: Monitoring -> Statistics -> Group By: caller

Statistics by day: Monitoring -> Statistics -> Group By: day

By using the “**Analyze**” form you can have a quick overview about the system.

Other more **advanced statistics** can be generated by using the Statistics form and using different fields/options/grouping/directions.

All statistics can be filtered by the “set direction” form or the “quick filter” edit-box and by a time interval selection.

Statistics can be exported as csv or html from File menu -> Save as. For other data formats you can use the Export/Import wizard.

Real time monitoring

Start the MSupervisor application to get notified about errors and malfunctions. (This application should be available in the start menu if you have installed the MizuManage). You can also set various alerts from the “Scheduled tasks” form.

Automatic reports

The server can send daily reports for administrators or email/sms alerts on malfunctions. For this you have to setup an “Admin” user with a valid email address. Then set the following fields to 1 (after your needs): “sendemailalert”, “senddailyemail”, “sendmonthlyemail”, “sendsmsreport”, “sendsmsalert”
The server will be able to send SMS messages only if an SMS provider is configured (see the Admin Guide)

Quick Tip: You can access various statistics by just clicking on the Dashboard items. For example click to “CCalls” will show the current calls.

CDR

CDRs (call detail records) can be listed by using the “**CDR**” form. By default only the most important fields are listed (date-time, connect time, call duration, etc). You can see more details if you check the “All fields” checkbox.

To quickly list the CDRs that belongs to a user, open the “Users and devices” form. Find the user record, then right-click on it and select “Set Direction”. Then go back to the CDR record form and click on the “Load/Reload” button.

If you have enabled **voice recording** for some users, than you can play the recorded audio by filtering for “Recorded Conversation” (select the desired record and click on the Play button)

Quick Tip: You can easily filter calls with a specific prefix by typing the prefix in the quick search box following with an asterisk and hit enter. For example searching for 44* will list all CDRs where the called number begins with 44.

You can find more details about CDR record listings from [this wiki](#).

Dial plans

You can manipulate number format, SIP headers or the Caller-ID from the following settings:

- Users and devices form: caller-id, username, other numbers (DID match), tech prefix
- Routing form: you can only specify routing direction here without number changes
- **Rules** form: this is a powerful module which you can use to change almost anything (including caller-id, called number format and many others)
- Global configuration: a few global configuration options might also affect the dial plan
- Prefix rules and the dial plan form: use the “Rules” form instead when possible.

You can find more details in the [VoIP Admin Guide](#) below the Routing section.

PBX features [optional]

A number of PBX features are enabled by default. More PBX modules can be activated if you select the “PBX Extra” on the configuration wizard or set the “fs_pbx” global configuration options to “1”.

Some of the features are implemented in multiple ways so you can select which better fits your needs.

This includes:

- **call forward:** can be enabled from users and devices form -> functions tab. [More details...](#)
- **call transfer:** available as specified in SIP standards (by all devices with support for transfer)
- **voicemail:** enable from users and devices form -> functions tab. Default access number are 5000 (with pin) and 5001 (with auto authentication). Auto-email forward is enabled by default. [More details...](#)
- **conference:** via SIP standard, via dtmf *1*number# or via conference rooms using extensions between 5100-5199 for narrowband and 5200-5299 for wideband (these can be changed)
- **call recording:** just check the “Voice Record” on the users and devices form -> functions tab. Playback the recorded voices from the “CDR” form. [More details...](#)
- **special numbers** and IVR’s such as music, record/playback, vide record/playback and others

- **missed call notifications** by email
- **barge-in:** via the “Voice here” form, or right click on current call or 5009 access number
- **many others** such as caller-id, ring groups, call hold, call waiting/park/pickup and others

Service access numbers [optional]

You can setup your **calling card** or **callback** business by using access numbers and assigning them to one of the existing or newly created IVR's. You should be able to request **DID numbers** from your existing VoIP carrier or by contacting other companies e.g. www.didx.net. In this case you will have to add it as a Traffic Sender user usually with IP based authentication (fill the AuthIP box with the provider IP or domain name)

After you have terminated with the traffic sender configuration, you can add the access numbers like usual endusers. Type the phone number in the “username” field or you can also use the “SIP number” field for the same reason. Then switch to the “Functions” tab and set the “Campaign ID” and the “Callback access” (if the DID number will be used as a callback access number); optionally you can enable A number authentication (PIN less dialing). The campaign id means the ID field from the tb_ccampaigns table (You can see them by opening the “Campaigns” form).

For more complex authentication and billing options please consult the admin guide.

IVR setup [optional]

The IVR module is used for various tasks like access numbers, calling-card operation, customer support etc.

You can assign different IVR's to different access numbers by using the “**Campaigns**” form. To create a new campaign, just click on the + sign and enter a “name” for the new record. The most important configuration for an IVR campaign is the script. Switch to the “details” tab to select a “Script”.

Scripts can be created by using the “IVR” form with lots of possibilities such as dtmf handling, voice announcements, text to speech, call forward and many more.

The server is shipped with several preconfigured script examples, but you should easily add new scripts or modify the existing ones by following the [admin guide](#) or the [IVR documentation](#).

WebPortal [on commercial servers]

The server has a built-in web interface (enduser control panel) usually listening on HTTPS 443 and/or HTTP port 80 or 8080 (as you have configured the main API port for your service), accessible as:

<http://serverdomainorip/webvoip>

or <http://serverdomainorip:port/webvoip>

The portal can be used by endusers, resellers, traffic senders and callshops to manage their account. Login with any valid user account. For the customization options login as an admin user (load or create any admin user from MizuManage Users and devices form).

Note: this portal is not a full website. The webportal is an optional module which you might deploy only if you are working with retail customers. You can integrate it with your existing [website](#) if needed

(simple way: just put in an iFrame and if users are already logged in on your website, you can automate the login also to this control panel, so users will not have to type their credentials again; There are several options to customize the colors to match you design). The control panel will display different content depending on the logged in users and can be also used:

- as a [callshop](#) interface
- by your business customers (traffic senders)
- by administrators for basic tasks

Optionally you can **rewrite our webportal** to match your needs (The portal was written in C++. Request source code from support) or **write your own portal** and use the http and/or database API. You will find the documentation [here](#).

If you already have a website, then you can easily integrate it with VoIP capabilities using the above mentioned methods. You can also use MizuTech [website template](#) if you don't already have a website and you don't have any knowledge or web developer to build your own.

Note: The webportal is not available for the Compact Edition.

Reseller

Resellers typically will use the web frontend for all their activity. First you should login on the web interface with as an admin user (you can create admin user from MManage "Users and devices" form). First edit the portal settings after your needs then create one or more "top" resellers. Then these resellers can login and create its own sub-resellers after they have created their tariff list(s).

Callshop

To create a "callshop owner" open the "Users and devices" form in the MizuManage and create a new Enduser then from the "Functions" tab tick the "Is Callshop" checkbox. From now the user can login on the web user interface, create its cabins (which are actually represented as sub-endusers) and monitor it's cabins activity.

WebRTC [optional]

The Mizu VoIP server can also accept WebRTC connections such as [webphone](#), [sipml5](#) or [SIP.JS](#). which you can use to initiate plugin-less calls from browsers to implement services such as [click to call](#).

The WebRTC includes all components for full WebRTC SIP protocol conversion such as Websocket, ICE, STUN ,DTLS/SRTP, auto TLS and built-in TURN service.

To enable WebRTC just select the "WebRTC" module on the Server configuration wizard -> Roles and features page. Once this is enabled, go to the Help -> "Client configuration" menu to find out the exact configuration details to be used in WebRTC clients.

Push notifications [optional]

The Mizutech SIP server has support also for VoIP push notifications. This is a useful feature to improve the availability of mobile SIP clients by sending a push notification on incoming call or text message

which will wake-up the client app, thus the call/chat can be delivered even if the app is closed or the device is sleeping.

Follow [this documentation](#) to enable push notification and integrate push support into your Android/iOS/Web app.

SMS [optional]

Provide SMS services to your users by interconnecting with an SMS provider capable to provide a HTTP API. For this just set the smsurl global config option or create SMS GW endpoints if you wish to use more than one provider. A guide can be found [here](#).

To setup an **outbound sms** routing, you have to contact a company providing SMS services. (For example [Clickatel](#))

Then open the “Configurations” form and search for “smsurl”. Enter the details in this format:
`http://api.clickatell.com/http/sendmsg?api_id=APID&user=USERNAME&password=PASSWORD&to=[to num]&text=[message]`

Pricing is done after the “smsprice” global config options or you can setup detailed pricing by using the “price setup” form.

Users will be able to send sms messages by using a softphone, the webportal or there is a possibility to create an SMS sender application yourself by using the [http or database api](#).

For incoming SMS applications (**SMS callback**, **balance request**, etc) you will have to request a two way SMS service (to get a DID number)

Wholesale/Transit VoIP business [optional]

You can quickly start a whole-sale, transit or sip trunk business using the Mizu VoIP server.

If just need to route traffic from A to B, then:

1. [Create](#) a Traffic Sender account for A with IP based authentication (“Users and devices” form). This is from where you will receive the traffic.
2. Create a SIP server account for B (“Users and devices” form). This is where you will send the traffic (another SIP trunk, carrier or gateway)
3. Add the SIP server account to your routing (“Routing” form)
4. Define pricing for A and B so you can generate billing reports and monitor your revenue (“Price Setup” form)

Done, you are ready to accept traffic now from A.

Add more traffic senders and/or SIP servers after your needs.

For example you might find good prices to destination X at a carrier, but for destination Y another carrier would be more optimal.

In this case just add both of them to your [routing](#) and define routing rules to decide which traffic to send in which direction.

SIP clients [optional]

The softswitch is compatible with any SIP, WebRTC and H.323 endpoints such as softphones and IP phones offering rich media and collaboration capabilities such as voice, video, chat, file-transfer, fax, presence and others.

Optionally you can use the Mizutech webphone, softphones and mobile clients. Mizutech provides **customized VoIP clients for web, desktop and mobile** as part of the [all in one softphone](#) package or separately, so you can offer preconfigured dialers for your customers with your name, logo, icon, branding, links and many options.

- [Browser webphone](#)
- [Windows softphone](#)
- [Android softphone](#)
- [iOS softphone](#)
- [Symbian dialer](#)
- [Other softphones](#)

You can also use your IP phone, third part softphone (such as X-Lite) or any SIP compatible device. Compatibility with browser VoIP clients is assured by the built-in WebRTC and RTMP modules.

Integration [optional]

The VoIP service is based on open protocols so you can [integrate](#) it in any environment:

- use any third party SIP, WebRTC or H323 softphone or device (in softphone settings, just type your server domain or IP and a valid enduser username/password)
- use it with any third-party SIP proxy, [SBC](#) or [Load Balancer](#) (these are required only if you have some specific needs, otherwise the softswitch doesn't require any SBC or proxy)
- [webportal integration](#) with your [website](#)
- its standard [SQL database](#) access and [API](#) can be used to integrate with existing website or software
- export/import any data (such as VoIP users) from the SQL Management studio or from MizuManage Export/Import wizard

Backup

All data is stored in the database, so you have to make sure that you always have a working backup for it. A nightly backup to some other PC on the LAN is an affordable solution for this (depending on your business requirements).

You can setup **scheduled backup** or (nearly) real time log shipping from [SQL Server Management Studio](#) or alternatively the Mizu Server can schedule your backups (see the detailed documentation).

Optionally you can use a dual server setup. This will increase the performance and in this way, you can always have a hot backup server in case if the active server fails.

To **clone** a VOIP server, just backup its database and restore it on your new server. Also install the VoIP server software (or copy the old directory) and make sure that your vserver.ini points to the new database.

The only setting that must be changed is the local IP global config option. For more details check the [cloning guide](#).

If you migrate the application to another server a new license file might be needed from Mizutech. For the compact edition all data is stored in a file named "mserver.sdf". You just have to copy that file to backup your server.

Usage

Other ideas you can do with the VoIP server:

- Retail business by offering VoIP services for people
- Call transit and call termination businesses
- Easily and transparently convert between various VoIP protocols by just enabling them on the config wizard: SIP/H.323/WebRTC/RTMP
- Setup [VoIP encryption](#) and tunneling to be able to bypass any NAT, firewall, proxy by using VoIP over encrypted UDP, TCP or HTTP. Just enable the "Encryption" module from the config wizard for this and use our softphones (all has built-in support for encryption or tunneling) or third party software/hardware via the [mtunnelclient](#)
- Setup number portability as described in the documentation
- Provide call termination / SIP trunk services for other companies. Just set them as Traffic Senders (specify auth IP), send your server address (IP or domain) to these companies, set proper pricing and you are ready to go
- Setup a PayPal account to let users to pay for your service (add prepaid user credit). See the "PayPal" section in the documentation.
- Codec transcoding by setting the "needcodeconversion" and "choosecodecs" fields for the user records
- Offer a [branded softphone](#) to your endusers covering all the popular platforms: Web, Windows, iOS, Android and others
- And [many others](#)

Common terms

SIP: The Session Initiation Protocol (SIP) the most important signaling protocol used for VoIP

H323: is an ITU recommended standard which is currently mostly replaced by SIP

WebRTC: is a media streaming protocol found in modern browsers

RTMP: is a media streaming protocol used by Flash

RTP: media channel protocol (used for audio/video/fax routing)

Enduser: Can represent your endusers (customers) or VoIP client devices (same as "Extension" in other systems)

SIP Trunk: in-bound or out-bound links. In MizuManage this can be set as "SIP Server" and "Traffic sender" users

Global Configuration: configurations applied to all users stored in tb_settings. Changes can be made from the "Configurations" form

User Configuration: configurations specific to a user. Changes can be made from the "Users and Devices" form

Dial Plan: describe the format of the phone number. In MizuManage this can be changed by global configuration options, rules, prefix rules and from the "dial plan" forms.

Routing: define where to send/forward outbound calls ("Routing" form)

PBX: means “Private Branch Exchange”, which is a phone network used by companies. Users can call and chat with each other and share a number of outside lines (DID) for external calls.

Callback: DID or toll free number configured as enduser with iscallback set to the required IVR

Click-to-call: a html button placed on websites to request an immediate connection with another person by phone call.

Can be implemented using the [webphone](#)

ASR: Answer Seizure Ratio. The percent of the connected calls compared to all calls (How many calls are connected from 100 calls in average)

ACD or ACL: Average Call Duration (measured in seconds for connected calls only)

IVR: Interactive Voice Recognition (used for calling-card for example)

ANI / CLI: Automatic Number Identification or Caller Line Identification (important on IVR systems for user authentication)

ANI callback: same as callback with User-ID based authorization (A number)

LCR: least cost routing (routing the traffic to the least cost outbound server)

BRS: best route selection (intelligent routing based on price + quality + other statistics and settings)

Resources

For more details, please consult the [Admin Guide](#) and other server related documentations on our [website](#).

If you are a paid customer or plan to purchase a license, contact serversupport@mizu-voip.com for any help. Install, configuration, training and support services are included in our pricing so no extra payment is required for these.

If you are using one of our free products, please post your questions to the [forum](#).