Instructions for using the EV8100 Voice Call function

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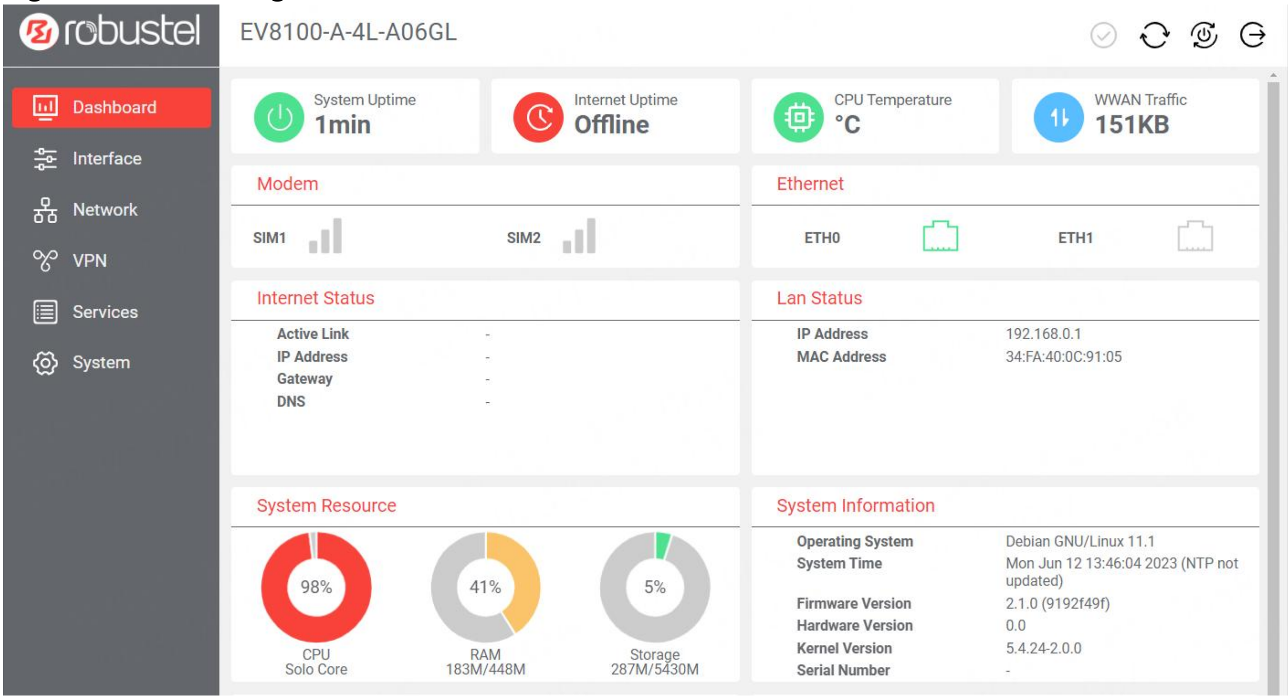
Versions	Modify time	Modifiers	Description
0.1	2023.6.12	Shen Can	First draft

Voice Call function configuration

Device connection

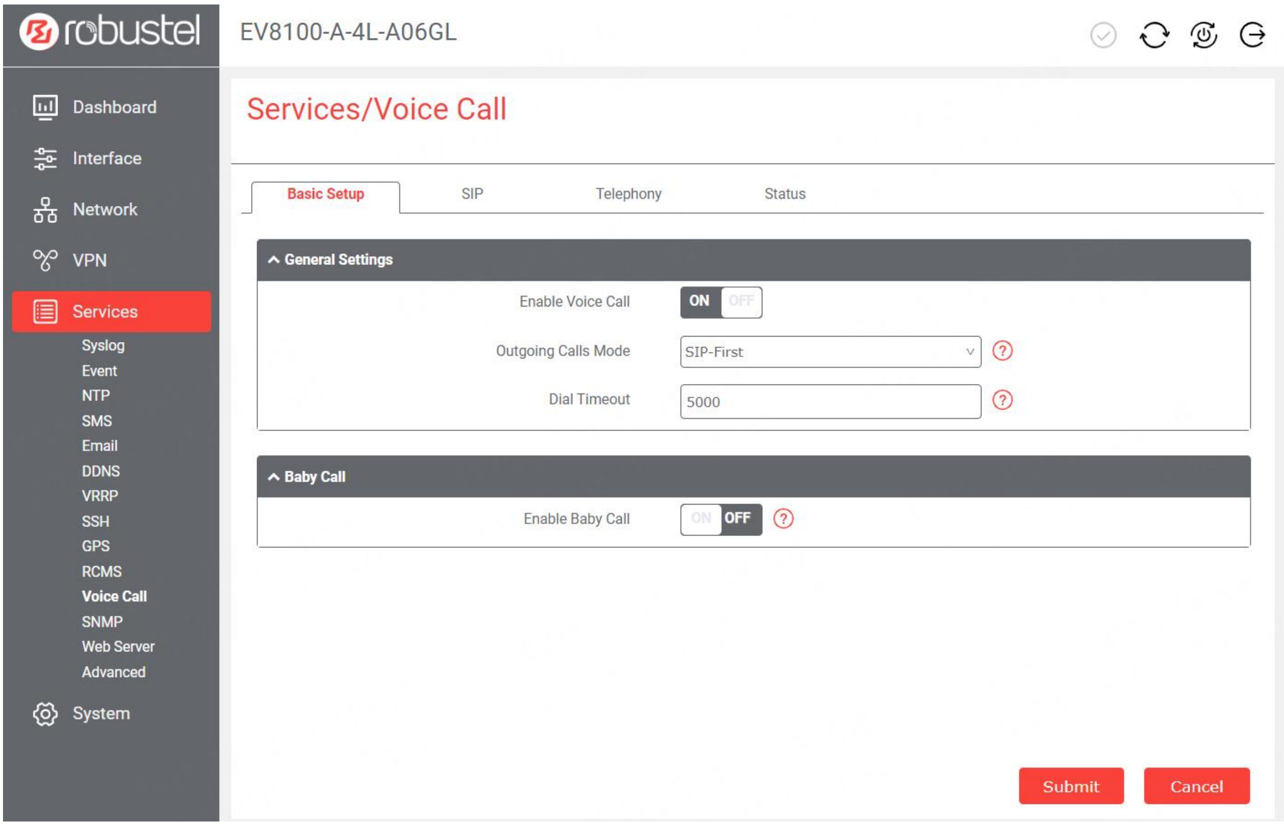
1. Set the computer ip address to 192.168.0.2, connect the device to the computer with a network cable and connect the landline to the FXO port of the device.
2. If you need to use the VoLTE function, you need to insert the SIM card into the SIM card and connect the antenna.
3. Login to the device WEB management page
Open a browser, enter the address of the device's WEB management page (default: http://192.168.0.1) and enter the user name and password (default is admin) to complete the login. A successful login will result in the screen shown below.



Voice Call basic configuration

Go to the configuration page

Click on the "Services" - "Voice Call" menu item in the left-hand navigation bar to access the Voice Call configuration page as shown below.



Enable/Disable Voice Call function (Enable Voice Call)

The Voice Call function can only be used if Voice Call is enabled; otherwise all voice calls will be disabled.

Outgoing Calls Mode

There are 4 modes to choose from for phone outbound calls, which are

- Block - prohibits call-out.
- SIP-First -- SIP first, i.e. the VoIP channel is selected first on outgoing calls, and the VoLTE channel is selected if the VoIP channel is not available.
- SIP-Only -- SIP only, i.e. only VoIP channels are selected for outbound calls.
- LTE-Only - LTE only, i.e. only the VoLTE channel is selected for outgoing calls.

Dial Timeout

The dial timeout time refers to the maximum timeout in milliseconds between the last dialed number and the official outgoing call when a user makes a call.

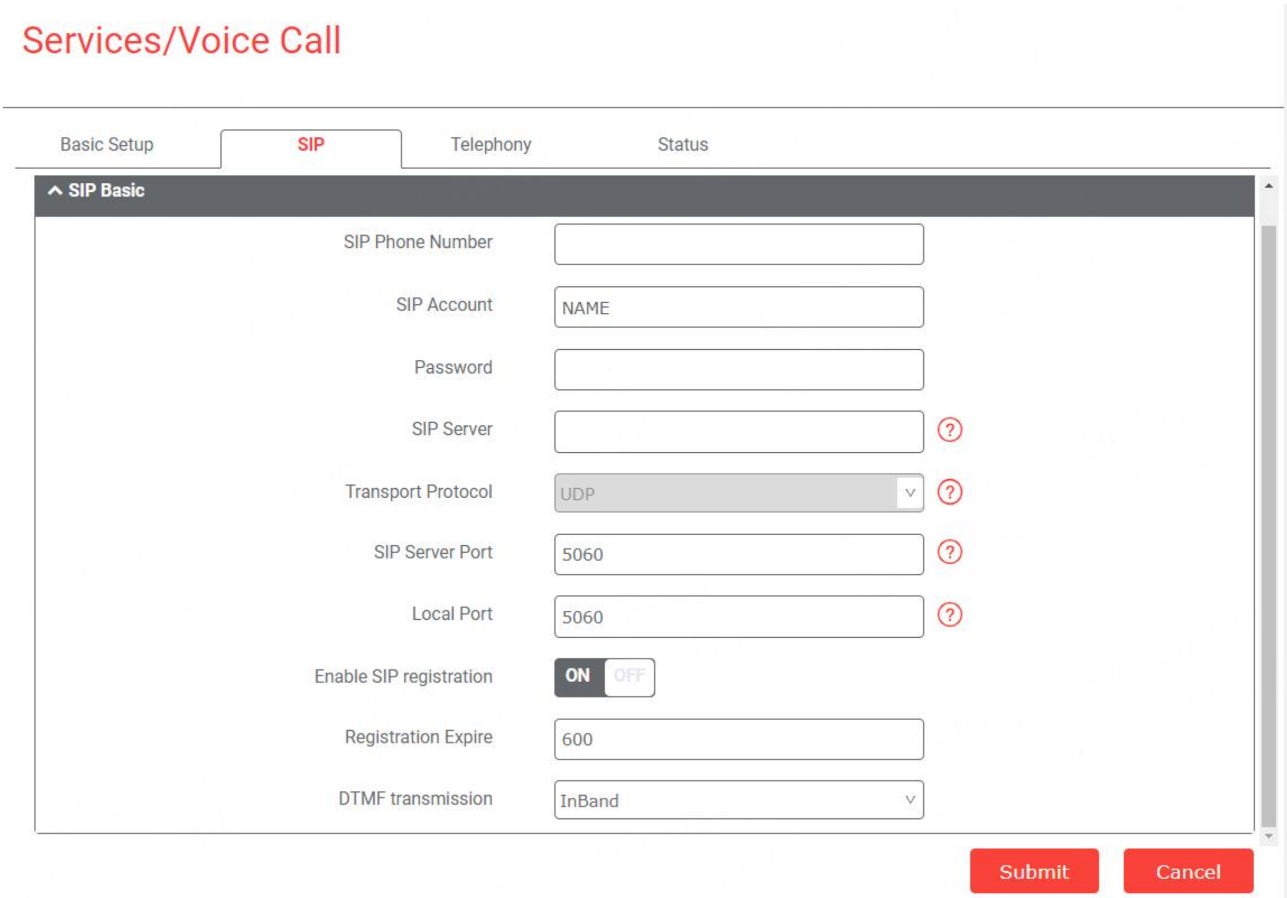
If the user never dials the number before the timeout or if the number is less than 3 digits long, the call will not be initiated and the phone will play a busy tone.
If the user does not want to wait for a timeout after the dialing is complete and wishes to call immediately, they can call out immediately by dialling # after the number.

Auto-dial (Baby Call)

Auto-dial means that when the user is off-hook, the system will automatically dial the preset number without the user having to do so. This function is suitable for emergency calls, service lines, etc. where no user dialling is required. The functional parameters involved are:

- Enable Baby Call
- Auto-dial number (Baby Call Number)
- Waiting time (Time) - within the waiting time, if the user does not make a valid dialing, then the automatic dialing will be carried out, otherwise the automatic dialing will be disabled for the current call out, set to 0 means no need to wait, automatic dialing directly after taking off the phone.

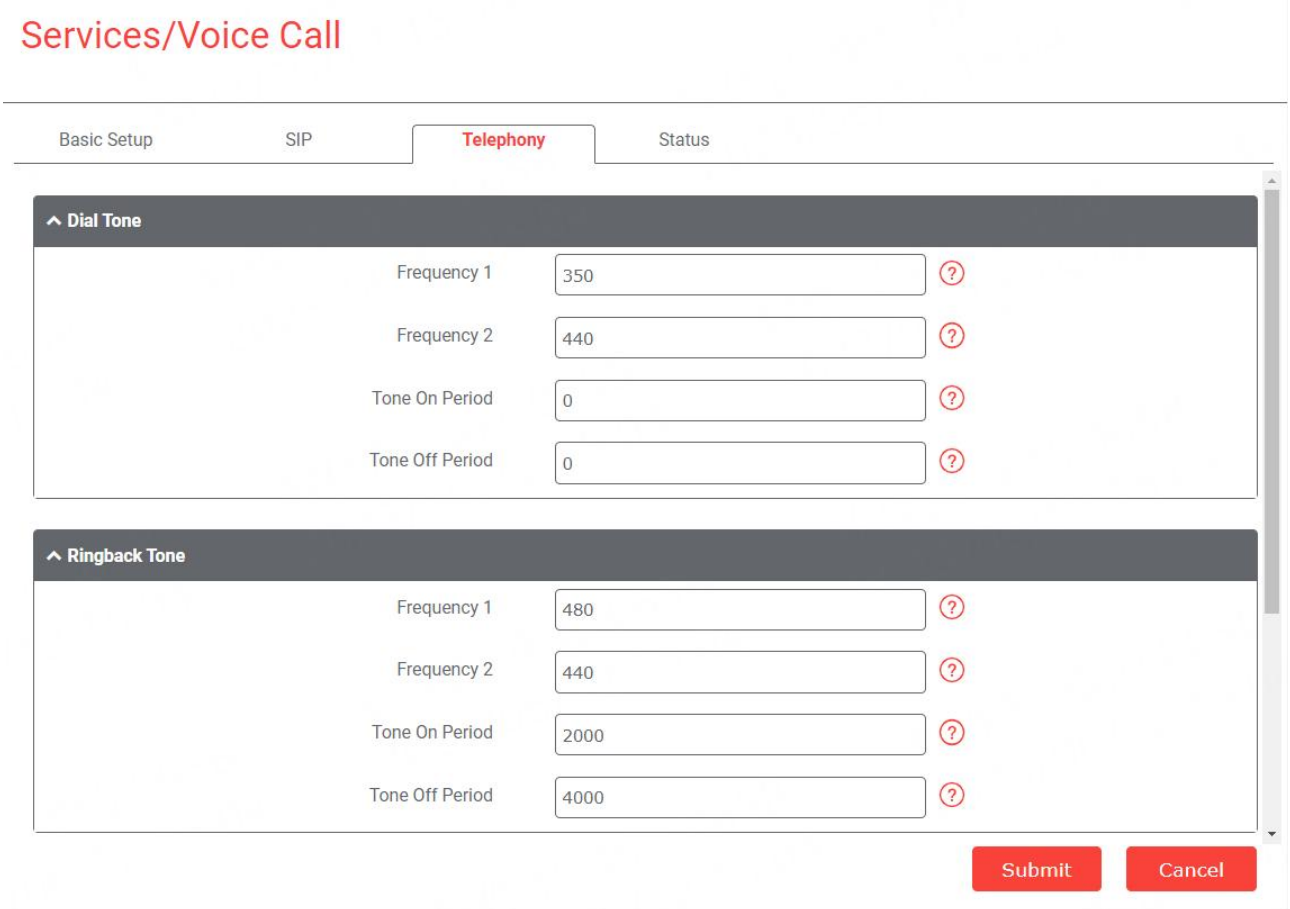
SIP function configuration



SIP Phone Number - supports a combination of letters, numbers, -, * and #
SIP Account Name (SIP Account)
SIP account password (Password)
SIP Server address (SIP Server) - supports IP or domain name
Transport Protocol - currently only supports UDP, can be extended with TCP, TLS

SIP Server Port - the port number the SIP server is listening on, default is 5060
SIP Local Port - the port number for listening to SIP signalling locally, default is 5060
Enable SIP registration - the SIP registration process will only be initiated if SIP registration is enabled, otherwise no registration will take place
Registration Expire - set in seconds
DTMF transmission - three DTMF transmission modes are currently supported: InBand, RTP (RFC2833), SIP INFO

Handset features



Dial tone setting

Dial tone frequency 1 (Fequency 1) - in Hertz (Hz)
Dial tone frequency 2 (Fequency 2) - in Hertz (Hz)
Tone On Period - in milliseconds
Tone Off Period - measured in milliseconds
When both of these durations are set to 0, it means continuous playback. Otherwise, the dial tone will be played cyclically for the configured duration.

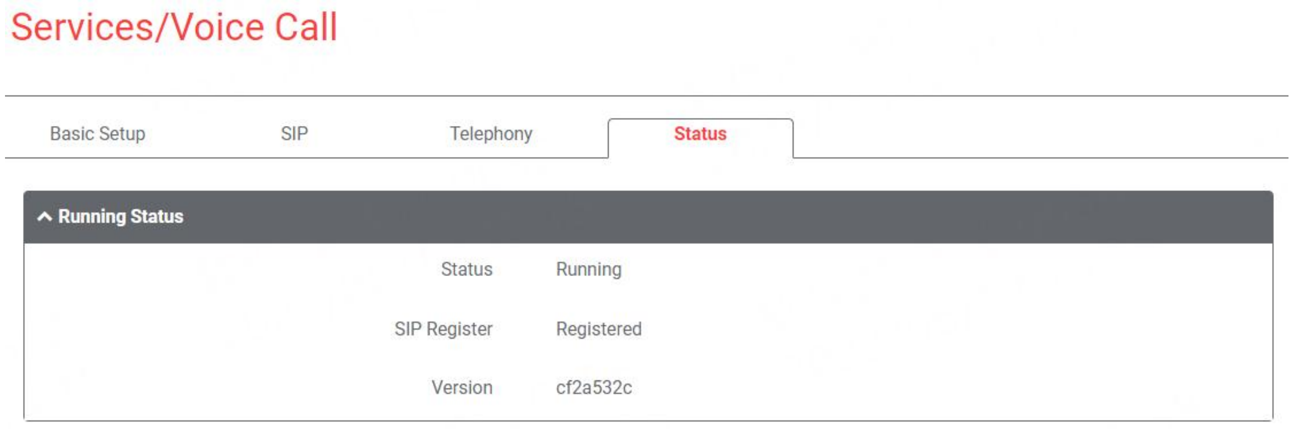
Ringback tone setting

Ibid

Busy tone setting

Ibid

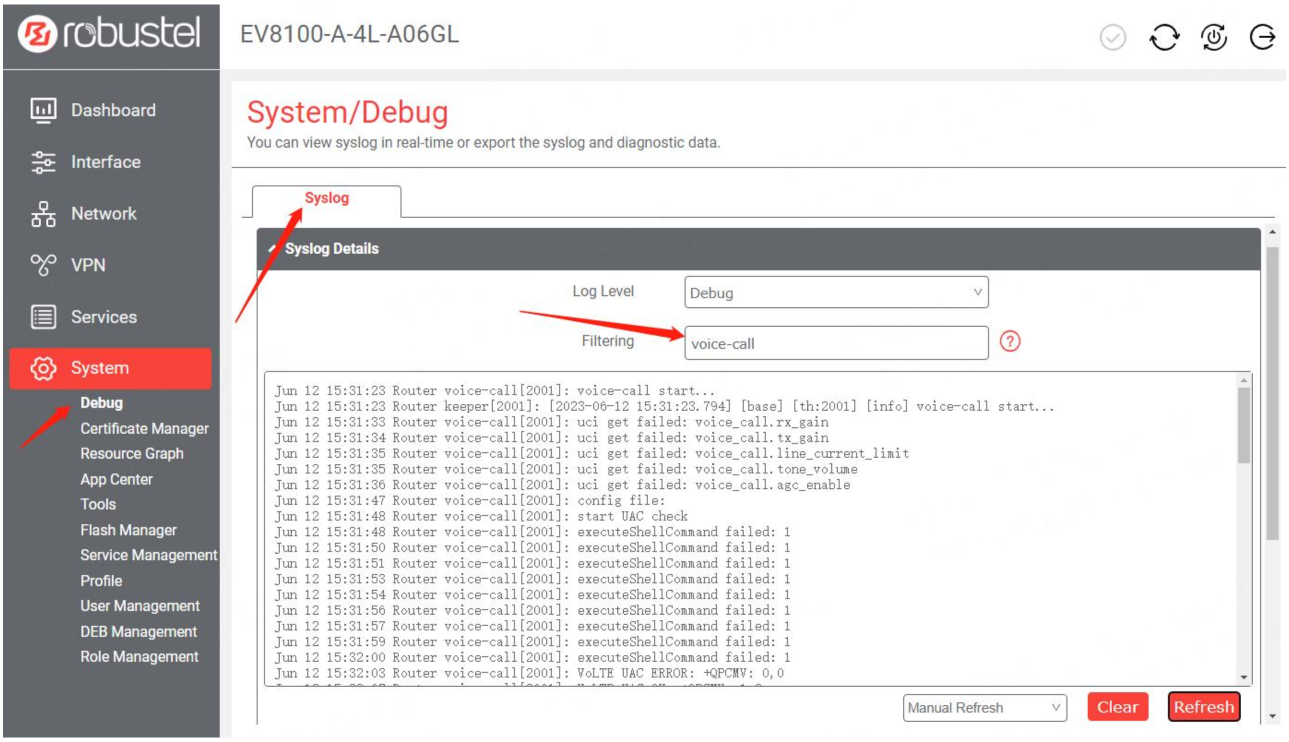
Status Search



Diagnostic tools

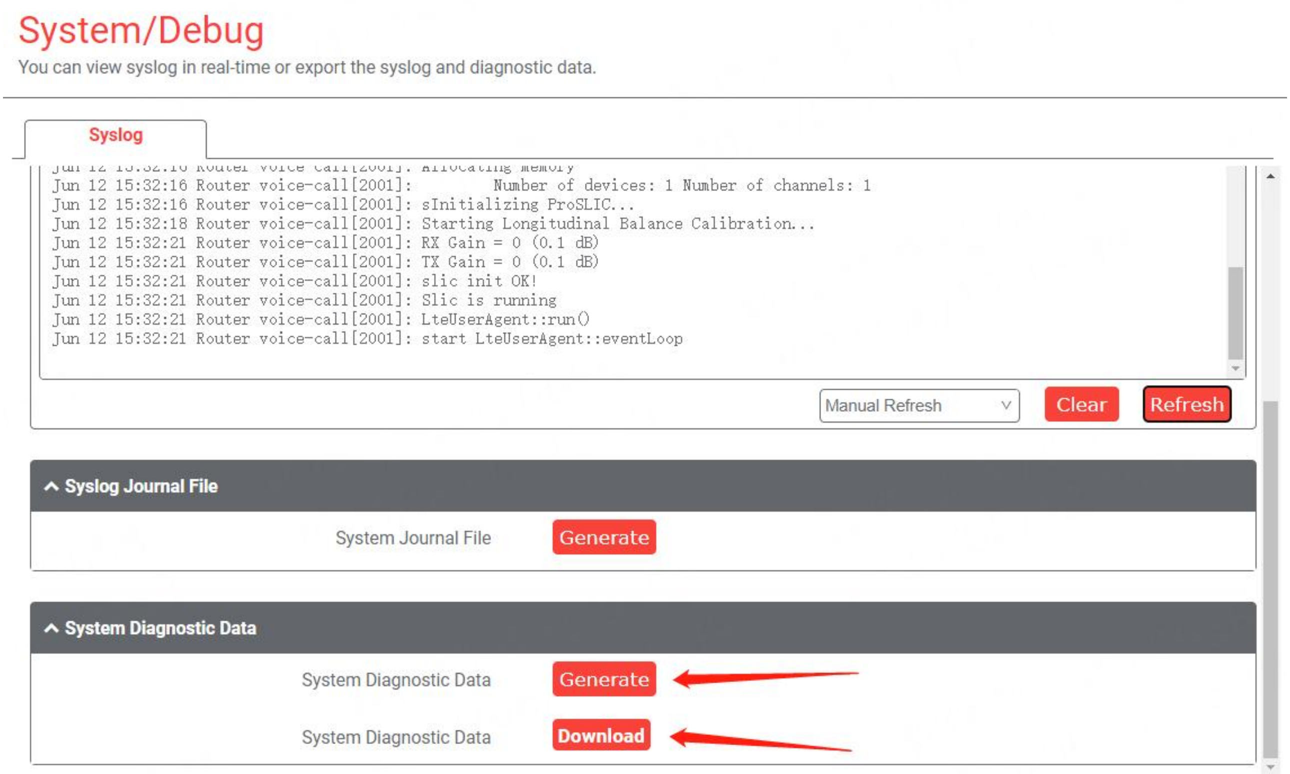
Query of logs

You can access the log query page via "System" - "Debug". To troubleshoot other log messages, enter voice-call in Filtering and click Refresh. Filtering log messages.



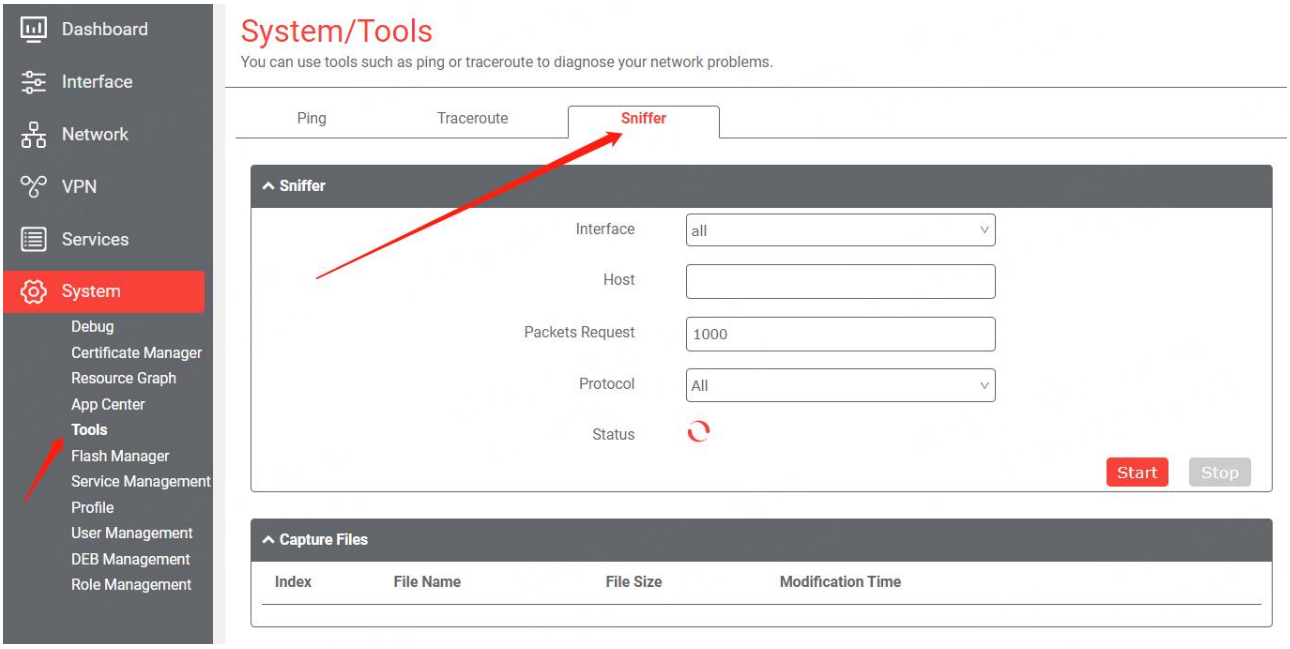
Generation and export of diagnostic data

Exporting diagnostic data is an effective tool when the developer is required to assist in the analysis and location of faults in the equipment. The user can export the diagnostic data of the current device with a simple operation on the page.



Network Packet Capture

When using VoIP, especially when connecting to a new SIP service platform, there are times when SIP accounts cannot be registered, calls cannot be established, calls cannot be opened and closed, there is no voice or only a single call. This can be done by capturing network packets to help engineers locate the problem. The packet capture tool can be found in the "System" - "Tools" menu item, open Sniffer to open the packet capture control page.



Frequently Asked Questions

What to do if the SIP server port is not 5060

If the SIP *service port is* not 5060, please change the *SIP service port* to the appropriate port number.

SIP registration status always unsuccessful

Please check that the device is properly connected to the SIP server;

Please check that the firewall configuration of the device is blocking SIP packets on the corresponding port;

If the network and SIP server configuration is confirmed to be OK, diagnostic tools such as packet capture can be used to assist in locating the problem.

Automatic calls are not possible

Check that the automatic call function is enabled;

Check that the automatic call timeout is not too long;

In the auto call please see if a dialing operation has been carried out.

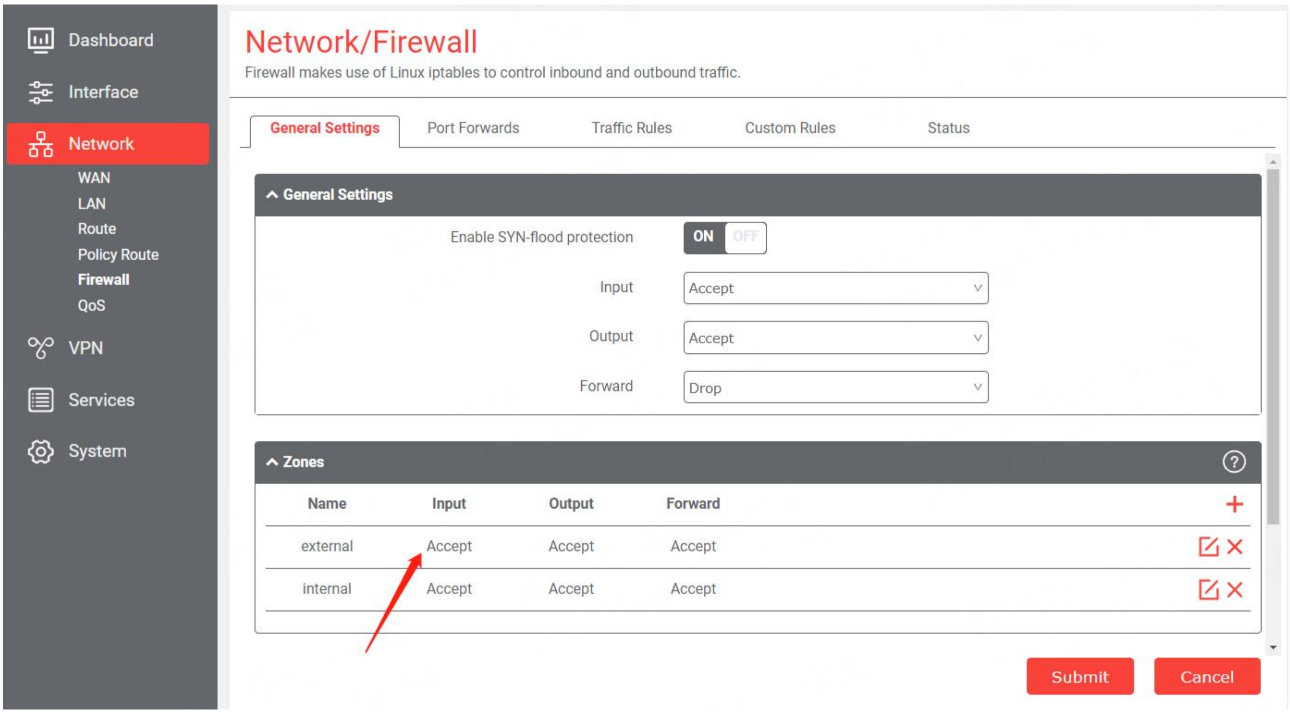
Instead of a dial tone, a busy tone appears when the phone is taken offline in an idle state

Confirmation that the last hook-up was done;

Check the log to confirm if you have just had an incoming call or are in a call status.

VoIP voice single pass

To check whether the device's firewall is blocking actively connected packets, refer to the configuration of the firewall in the following diagram: external zone accepts Input input packets.



Unable to make calls via VoLTE

Confirmation that the SIM card and network support VoLTE voice call functionality;

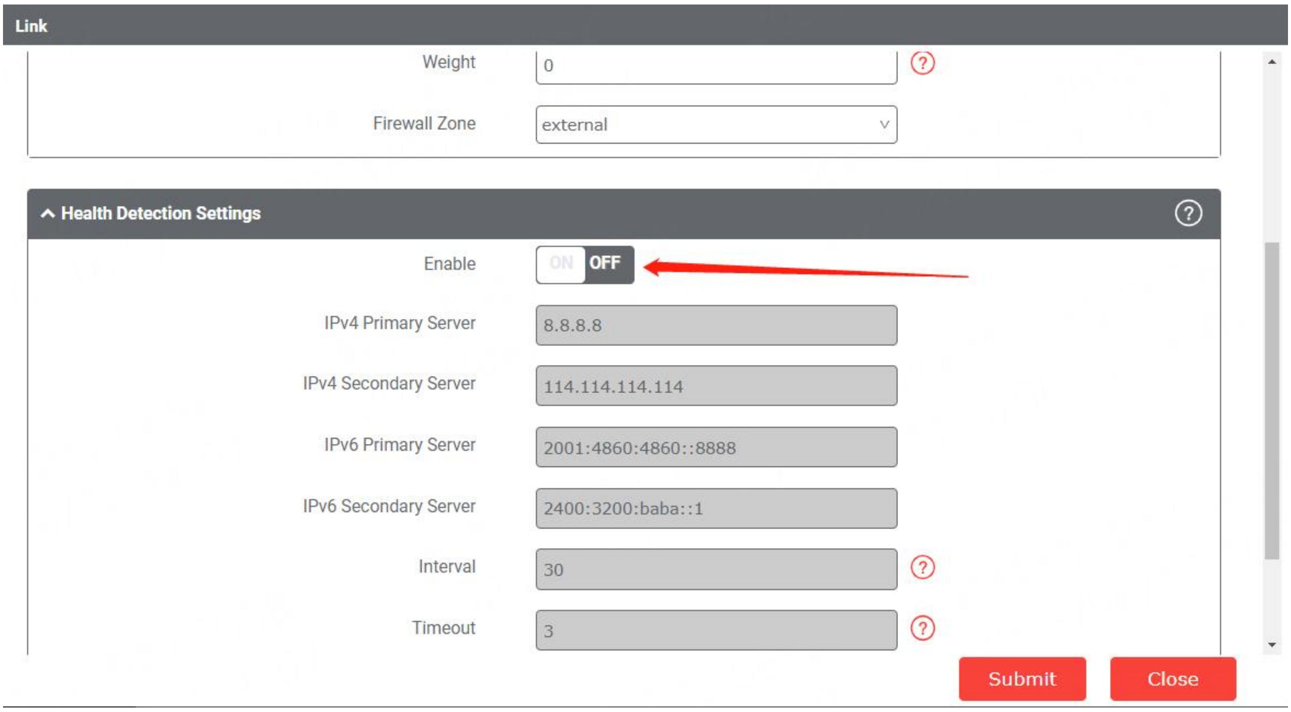
Check the status of the cellular network to ensure that it is registered successfully;

check that the outbound mode in the configuration is set to VoIP First or VoLTE Only mode;

Automatic hang-up during VoLTE call

Confirmation that the SIM card and network support VoLTE voice call functionality;

Turn off network health status detection on the wwan port, as shown in the following figure:


Link
Weight
0
Firewall Zone
external
Health Detection Settings
Enable
ON
OFF
IPv4 Primary Server
8.8.8.8
IPv4 Secondary Server
114.114.114.114
IPv6 Primary Server
2001:4860:4860::8888
IPv6 Secondary Server
2400:3200:baba::1
Interval
30
Timeout
3
Submit
Close