



How to integrate the Analog Gateway and the GSM Gateway in One OpenVox Box.

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OpenVox VoxStack GSM Gateway and Analog Gateway are Feature-rich, high availability, flexible modular gateway products. This Application Note introduces some methods about “how to integrate Analog Gateway and GSM Gateway in one OpenVox VoxStack Box”.

Chapter 1: Hardware Setup



You should have hardware as below:

Some VS-GWM400G modules, Some GWM800S modules, and One OpenVox Gateway Box (2 Socket BOX or 5 Socket VoxStack BOX, as shown above is our 2 Socket VoxStack BOX).

Then connect your power supply.

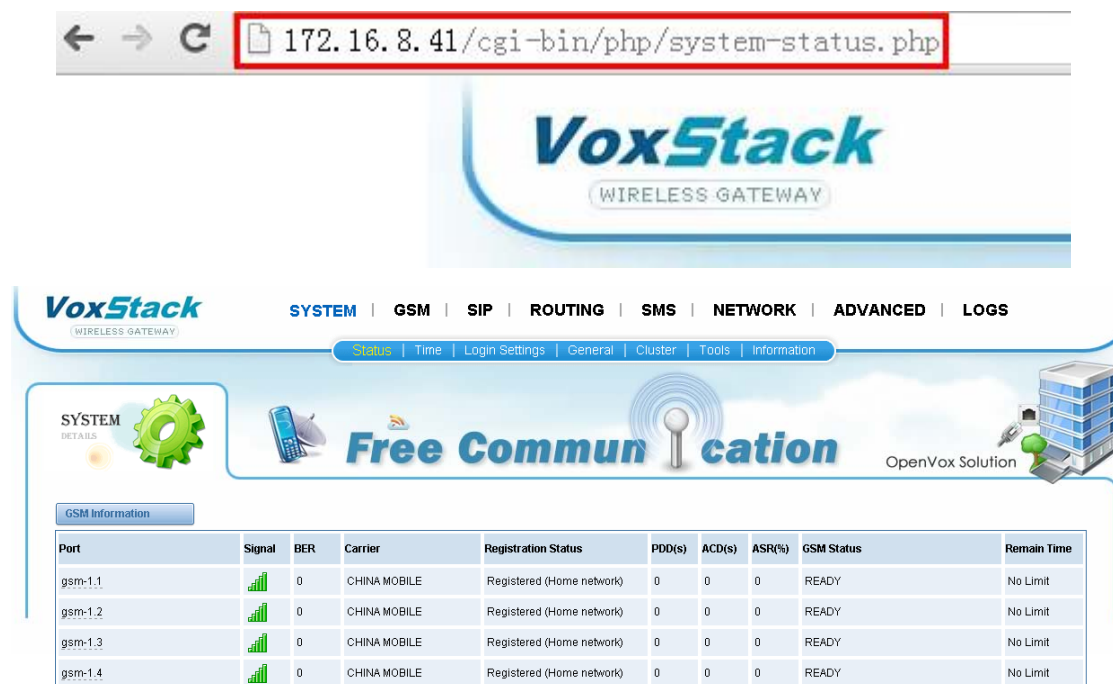
Chapter 2: Configure the Integrated Gateway

The GSM gateway modules (VS-GWM400G) and the Analog gateway modules (GWM800S) have their own independent operating systems.

The communication between GSM gateway modules and Analog Gateway modules is via the built-in switches of the OpenVox Box.

2.1: Login GSM Gateway Modules

For example:



The screenshot shows a web browser address bar with the URL `172.16.8.41/cgi-bin/php/system-status.php` highlighted in red. Below the browser is the VoxStack Wireless Gateway web interface. The page features a navigation menu with options: SYSTEM, GSM, SIP, ROUTING, SMS, NETWORK, ADVANCED, and LOGS. A sub-menu for 'Status' is active, showing options for Time, Login Settings, General, Cluster, Tools, and Information. The main content area displays 'Free Communication' and 'OpenVox Solution' branding. A 'SYSTEM DETAILS' section is visible, and a 'GSM Information' table is shown below.

| Port | Signal | BER | Carrier | Registration Status | PDD(s) | ACD(s) | ASR(%) | GSM Status | Remain Time |
|---------|--------|-----|--------------|---------------------------|--------|--------|--------|------------|-------------|
| gsm-1.1 | | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |
| gsm-1.2 | | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |
| gsm-1.3 | | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |
| gsm-1.4 | | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |

You could see the GSM Gateway GUI.

Notice: About the IP address, please refer to the GSM Gateway user manual the Chapter 7.

2.2: Login the Analog Gateway Modules

As follows:

The screenshot shows the VoxStack Analog Gateway GUI. The browser address bar contains the URL `172.16.8.68/cgi-bin/php/system-status.php`. The page header includes the VoxStack logo and the text "ANALOG GATEWAY". The navigation menu consists of **SYSTEM**, **ANALOG**, **SIP**, **NETWORK**, **ADVANCED**, and **LOGS**. Below the navigation menu, there are links for **Status**, **Time**, **Login Settings**, **General**, **Cluster**, **Tools**, and **Information**. The main content area features the text "Free Communication" and "OpenVox Solution". A table titled "Port Information" is displayed, showing the status of 8 ports.

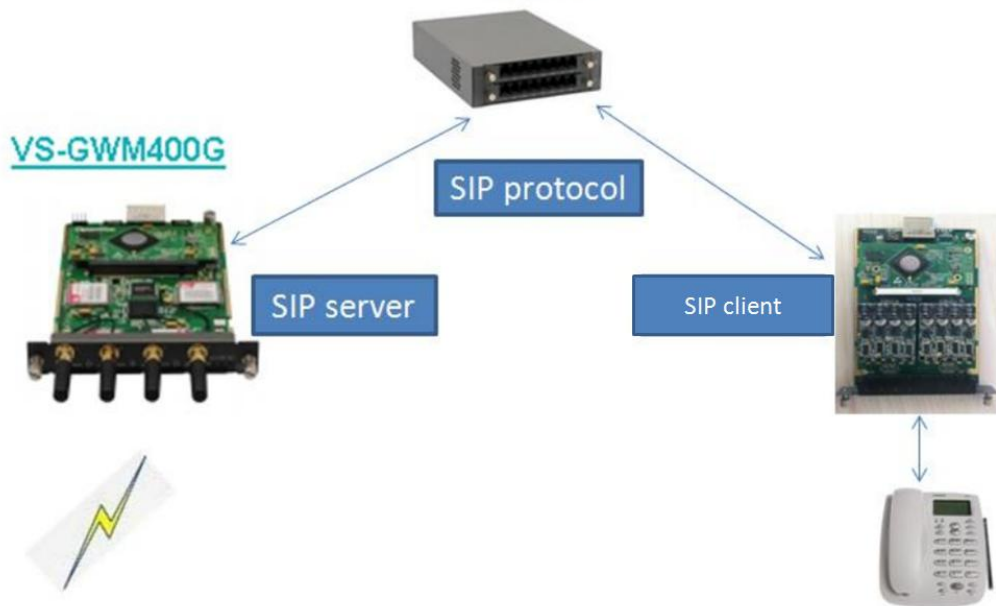
| Port | Name | Type | Caller ID | Sip Account | Port Status |
|------|--------------|------|-----------|-------------|-------------|
| 1 | board1-port1 | FXS | 8001 | | OnHook |
| 2 | board1-port2 | FXS | 8002 | | OnHook |
| 3 | board1-port3 | FXS | 8003 | | OnHook |
| 4 | board1-port4 | FXS | 8004 | | OnHook |
| 5 | board1-port5 | FXS | 8005 | | OnHook |
| 6 | board1-port6 | FXS | 8006 | | OnHook |
| 7 | board1-port7 | FXS | 8007 | | OnHook |
| 8 | board1-port8 | FXS | 8008 | | OnHook |

You will see the Analog Gateway GUI, about how to login Analog Gateways, please refer to the user manual of Analog Gateway.

2.2: Configure the communication

The GSM modules with the Analog Gateway modules communicate via the SIP protocols. So we should create SIP connection between the GSM gateway and the Analog Gateway. The topology is shown below:

Switches integration in BOX



Notice: You can use other ways (for example: IP2IP) to connect the GSM modules and the Analogy modules, about more details, please refer to the manual of GSM Gateway user manual.

2.3: Application 1: Use Analog Phones to Send Calls via GSM Modules

2.3.1: Step 1: Setup a SIP Server on the GSM Gateway Modules.

For example:



Edit SIP Endpoint "1001"

▼ Main Endpoint Settings

| | |
|-------------------------|-----------------------------------------|
| Name: | 1001 |
| User Name: | 1001 <input type="checkbox"/> Anonymous |
| Password: | |
| Registration: | Endpoint registers with this gateway ▼ |
| Hostname or IP Address: | dynamic |
| Transport: | UDP ▼ |
| NAT Traversal: | Yes ▼ |

▶ Advanced:Registration Options

▶ Call Settings

Save Cancel

2.3.2: Setup a SIP Client Register to the GSM Gateway SIP Server

As follows:

VoxStack ANALOG GATEWAY | SYSTEM | ANALOG | SIP | NETWORK | ADVANCED | LOGS

SIP Endpoints | Batch SIP Endpoints | Advanced SIP Settings

Edit SIP Endpoint "1001"

▼ Main Endpoint Settings

| | |
|-------------------------|--------------------------------------------|
| Name: | 1001 |
| User Name: | 1001 <input type="checkbox"/> Anonymous |
| Password: | 1001 |
| Registration: | This gateway registers with the endpoint ▼ |
| Hostname or IP Address: | 172.16.8.41 |
| Transport: | UDP ▼ |
| NAT Traversal: | Yes ▼ |
| SUBSCRIBE for MWI: | No ▼ |

Now you can see the SIP register statues in the GSM Gateway and the Analog Gateway.

GSM Gateway

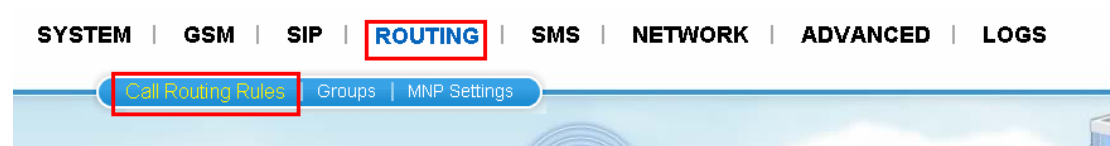
| SIP Information | | | | |
|-----------------|-----------|-------------|--------------|------------|
| Endpoint Name | User Name | Host | Registration | SIP Status |
| 1001 | 1001 | 172.16.8.68 | server | OK (3 ms) |

Analogy Gateway

| SIP Information | | | | |
|-----------------|-----------|-------------|--------------|------------|
| Endpoint Name | User Name | Host | Registration | SIP Status |
| 1001 | 1001 | 172.16.8.41 | client | Registered |

2.3.3: Setup the Router in the GSM Gateway

For example:



Modify a Call Routing Rule

▼ Call Routing Rule

| | |
|---------------------|---------|
| Routing Name: | CallOut |
| Call Comes in From: | 1001 |
| Send Call Through: | gsm-1.1 |

▶ Advance Routing Rule

Save Cancel

2.3.4: SIP and the Analogy Port in the Analogy Gateway

VoxStack ANALOG GATEWAY

SYSTEM | **ANALOG** | SIP | NETWORK | ADVANCED | LOGS

Channel Settings | Dial Matching Table | Global Settings

ANALOG

| Port | Type | Name | Caller ID | Sip Account | CID signalling | Actions |
|------|------|--------------|-----------|-------------|----------------|---------|
| 1 | FXS | board1-port1 | 1001 | 1001 | bell | |
| 2 | FXS | board1-port2 | 8002 | None | bell | |
| 3 | FXS | board1-port3 | 8003 | None | bell | |
| 4 | FXS | board1-port4 | 8004 | None | bell | |

2.3.5: Test Calls Out via the GSM Gateway





Now we connect an Analogy phone to the Analogy Gateway port 1, and try to call the number 10086.

You will see the call have been established in the GSM Gateway and the Analogy Gateway.

Analog Gateway

| Port | Name | Type | Caller ID | Sip Account | Port Status |
|------|--------------|------|-----------|-------------|--------------------------------------|
| 1 | board1-port1 | FXS | 1001 | 1001 | Call Active Called to 10086 00:00:18 |
| 2 | board1-port2 | FXS | 8002 | | OnHook |

GSM Gateway

| Port | Signal | BER | Carrier | Registration Status | PDD(s) | ACD(s) | ASR(%) | GSM Status | Remain Time |
|---------|-----------------------------------------------------------------------------------|-----|--------------|---------------------------|--------|--------|--------|-------------------------------|-------------|
| gsm-1.1 |  | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |
| gsm-1.2 |  | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |
| gsm-1.3 |  | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | CALL PROGRESS Called to 10086 | No Limit |
| gsm-1.4 |  | 0 | CHINA MOBILE | Registered (Home network) | 0 | 0 | 0 | READY | No Limit |

Notice: This application can be used to any OpenVox Gateway BOX, 2 sockets or 5 sockets, and you can combine your BOX freely, only according to your needs.

Of course, you could pull network cables when you finished the configuration. In the other words, you can use it without any PBX.