

## Using the GS8 Modular Gateway with Asterisk

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This document is an application note that briefly describes how to connect the GS8 modular gateway to an Asterisk software PBX. The GS8 modular gateway is a gateway that can accommodate interfaces for VoIP (standard SIP), GSM, CDMA, analog FXO, and analog FXS. It can connect any interface to any other interface. In this application note it is used to connect VoIP to GSM.

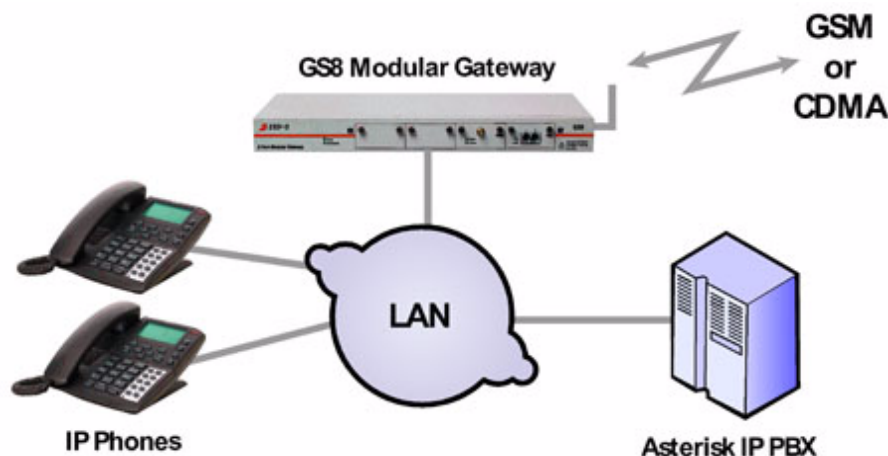
Please refer to the Data Sheets, Quick Start Guides, and User's Manuals for both products for further details.

### 1 Assumptions

We have an IP network to which an Asterisk IP PBX, several SIP telephones, and a GS8 modular gateway are connected. The connection instructions assume the following information

- the IP address of the Asterisk PBX is 192.168.1.250
- the IP address of the GSM modular gateway is in same subnet and has the address 192.168.1.240
- you have installed a SIM card in the GSM module

The typical connection diagram shown in the diagram:



## 2 Configuring the GS8

### 2.1 Log in to the GS8

2.1.1 Ensure the GS8 is properly connected to the network.

2.1.2 Launch a web browser on your PC and enter the default IP address of GS8 **http://192.168.0.100**

2.1.3 Enter user name and password in the log in window. The default user name and password are **admin**.

### 2.2 Configuring The IP Address of the GS8

2.2.1 Select [Administration menu | Network settings](#)

- Enter the IP address **192.168.1.240**
- Subnet Mask **255.255.255.0**
- Gateway Depends on your network
- DNS depends on your network.

2.2.2 After entering all parameters click on SET

The screenshot displays the 'Administration' tab of the GS8 web interface. The 'Network Setting' section is active, showing the following configuration fields:

Field	Value
Work Mode	Static
IP Address	192.168.1.240
Net Mask	255.255.255.0
Default Gateway	192.168.1.1
Primary DNS	125.22.47.125
Secondary DNS	
Host Name	
VLAN	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled

A 'Set' button is located at the bottom of the configuration area.

### 2.3 Line Settings

You now configure the parameters that instruct the GS8 where to send the invite packet during a GSM to VoIP incoming call. You also configure the parameters to instruct the GS8 from which IP address the GS8 may receive SIP packets.

2.3.1 Select [Line settings | IP Line](#) and enter the following:

- Description: **To Asterisk**
- Register: **Disable**
- Proxy (the IP address of the Asterisk system): **192.168.1.250:0**
- Phone Number: Enter any virtual number
- User name: Enter any name
- Password: Select some digits
- Outbound: Keep it blank
- Home Domain: Keep it blank or you can enter Asterisk sever IP address
- MAX Connections: **4**
- Enable AA: **Disable**

2.3.2 Click on [Add](#)

The screenshot shows the 'Line Settings' page in the ZED-3 interface. The top navigation bar includes 'Monitor', 'Line Settings', 'Call Settings', 'Administration', and 'Maintenance'. The user is logged in as 'admin'. The page displays a table of existing line settings and a form for adding a new one.

ID	Description	Register	Proxy	AA
1	IP2_Server	Disable	10.0.0.250	Disable

**IP Setting**

Description:

Register:

Proxy:

Phone Number:

Username:

Password:

Outbound:

Home Domain:

Max Connections:

Enable AA:  Enable  Disable

2.3.3 Click on [GSM Line](#) on the left

- Enter Phone Number - Same SIM card number
- Enable AA – Disable

### 2.4 Routing Settings

#### 2.4.1 Select [Call settings](#) | [Route](#)

2.4.2 Enter the routing table. The following parameters have to be configured to route GSM call on IP (Asterisk), and IP (Asterisk) call on GSM:

- Description – Asterisk To GSM
- Source Line – check IP 1 Box
- Caller pattern – leave it blank (blank means all)
- Callee pattern – leave it blank (blank means all)
- Schedule – Null (you can program if required)
- Destination Line – check the slot number on which you have inserted GSM module

#### 2.4.3 Click on [Add](#)

The screenshot shows the Asterisk Call Settings interface. At the top, there are navigation tabs: Monitor, Line Settings, Call Settings (selected), Administration, and Maintenance. On the right, there are links for 'admin' and 'Logout'. The main content area displays a routing table with the following data:

Description	Source	Destination	Caller Pattern	Callee Pattern	Schedule
SE500toFXO	IP/1	FXO/ALL	ALL	ALL	NULL
FXOtoSE500	FXO/ALL	IP/1	ALL	ALL	NULL
GSMtoSE500	GSM/ALL	IP/1	ALL	ALL	NULL

Below the table is a configuration form for a new route. The fields are as follows:

- Description: GSMtoSE500
- Source Line: FXO  ALL  slot1-1  slot1-2; GSM  ALL  slot3; IP  ALL  1
- Caller Pattern: (empty)
- Callee Pattern: (empty)
- Schedule: NULL (dropdown menu)
- Destination Line: FXO  ALL  slot1-1  slot1-2; GSM  ALL  slot3; IP  ALL  1

At the bottom of the form, there are buttons for 'Add', 'Modify', 'Remove', 'Move Down', and 'Move Up'.

2.4.4 Now add another route GSM to Asterisk

- Description – GSM To Asterisk
- Source Line – Check GSM slot on which you have inserted GSM module
- Caller pattern – leave it blank (blank means all)
- Callee pattern – leave it blank (blank means all)
- Schedule – Null (You can program if required)
- Destination Line – Check IP one Box

2.4.5 Click on [Add](#).

## 3 Configuring the Asterisk IP PBX

Now add a few lines in the IP PBX configuration for both proper routing of outgoing calls to the GS8 modular gateway and receiving calls coming from the GSM gateway to Asterisk.

### 3.1 Outgoing Calls

The core of Asterisk connection lies in the file:

```
/etc/asterisk/extensions.conf
```

Open this file in your favorite editor and add the following lines:

```
exten => _Z.,1,Dial(sip/${EXTEN:0}@192.168.1.240)
```

When you want direct dialling without any prefix, add the country code before Z.

```
exten => _5Z.,1,Dial(SIP/${EXTEN:1}@192.168.1.240)
```

```
exten => _6Z.,1,Dial(SIP/${EXTEN:1}@192.168.1.240)
```

```
exten => _7Z.,1,Dial(SIP/${EXTEN:1}@192.168.1.240)
```

Once you have saved and closed the file, restart Asterisk and from now on all calls starting with country code and number and 5,6,7 should be routed to the GS8 modular gateway.

You can use any pattern.

### 3.2 Incoming Calls

The GS8 modular gateway works with SIP, so modify the `/etc/asterisk/sip.conf` file where the GS8 modular gateway section could look as follows, for example:

```
type=peer
insecure=very
disallow=all
allow=ulaw
allow=alaw
allow=gsm
context=-----?
host=192.168.1240
username=GS8
permit=192.168.1.240/255.255.255.255
qualify=yes
canreinvite=no
call-limit=4
```

The context should be the same as you have written in `extensions.conf` for the incoming call.

Again, restart Asterisk after saving the file. After that, Asterisk will be ready to receive calls coming from the GS8 modular gateway.