

Topic / Issue: MVPGSM-2 integration with IP PBX

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This document describes the configuration of the MVPGSM-2 GSM to SIP gateway for interfacing with an IP PBX system via the LAN. This will allow calls between the GSM network and the PBX system. Begin by logging in to the management interface of the MVPGSM-2 by browsing to the IP Address. The management PC will need to have the Java Runtime Environment installed.

Connect to the Web GUI of the MVPGSM-2 by browsing to the Ethernet (WAN) interface at the default IP Address of 192.168.3.143. Select the configuration pages shown in this document from the menu on the left of screen.

The screenshot displays the configuration interface for the MVPGSM-2. It is divided into several sections:

- Ethernet Parameters:** Includes a checkbox for "Packet Prioritization (802.1p)", a "Frame Type" dropdown set to "TYPE-II", and a sub-section for "802.1p Parameters" with dropdowns for "Priority" (Call Control: 6 - Voice, VoIP Media: 3 - Excellent Effort, Others: 0 - Best Effort) and a "VLAN ID" field set to 1. "OK" and "Cancel" buttons are visible.
- IP Parameters:** Includes a "Gateway Name" field set to "MVPGSM", a checkbox for "Enable DHCP", and fields for "IP Address" (192.168.15.195), "IP Mask" (255.255.255.0), and "Gateway" (192.168.15.208). This section is circled in red.
- Diff Serv Parameters:** Includes fields for "Call Control PHB" (34) and "VoIP Media PHB" (46).
- FTP Server:** Includes a checkbox for "Enable" which is checked.
- DNS:** Includes a checked checkbox for "Enable DNS", an unchecked checkbox for "Enable DNS SRV", and a "DNS Server IP Address" field set to 210.23.129.34.
- TDM Routing Option:** Includes an unchecked checkbox for "Use TDM Routing For Intra-Gateway calls".

Configure the IP Address of the gateway so it is on the same subnet as your IP PBX system. You can use static IP parameters or enable DHCP so a server can assign them.

The Gateway address is the IP Address of the IP PBX.

Voice/Fax Parameters

Select Channel : Channel 01

Voice Gain
 Input -3 dB Output 0 dB

Dtmf
 Gain High -6 dB Low -8 dB
 Duration 100 ms
 DTMF Out of Band - Fixed Duration
 Out Of Band Mode Rfc2833

Coder
 Manual Automatic
 Selected Coder G.711 U - law@64kbps
 Max bandwidth 10 kbps

Fax / Modem
 Fax Relay Enable
 Modem Relay Enable
 Max Baud Rate 14400 kbps
 Fax Volume 9.5 dB
 Jitter Value 400 ms
 Mode FRF 11

Advanced Features
 Silence Compression
 Echo Cancellation
 Forward Error Correction

Auto Call
 Auto Call Auto Call Generate Local Dial Tone
 Offhook Alert timer 10 sec
 Phone Number 55

Set the desired codec.

Select Auto Call and set the Phone Number as the extension number on the IP PBX that calls from the GSM network will be passed to. In the example the gateway is sending inbound GSM calls to ext 55 on the PBX.

Packetization Time Parameters

Select Channel : Channel 01

Packetization Rate (msec per packet)

G711 A law@64 Kbps:	30	G727@40/16 Kbps:	80
G711 U law@64 Kbps:	30	G727@40/24 Kbps:	85
G726 @16 Kbps:	80	G727@40/32 Kbps:	25
G726@24 Kbps:	80	G723.1@5.3 Kbps:	120
G726@32 Kbps:	80	G723.1@6.3 Kbps:	90
G726@40 Kbps:	80	G729@8 Kbps:	30
G727@16 Kbps:	80	NetCoder@6.4 Kbps:	80
G727@24/16 Kbps:	80	NetCoder@7.2 Kbps:	80
G727@24 Kbps:	80	NetCoder@8 Kbps:	80
G727@32/16 Kbps:	80	NetCoder@8.8 Kbps:	80
G727@32/24 Kbps:	80	NetCoder@9.6 Kbps:	80
G727@32 Kbps:	80		

OK
 Cancel
 Copy Channel
 Default

Default Ptimes are 80ms, too high for some systems. Suggest change to 20, 30, or 40 ms for the codecs that will be used.

Current Permission: Read/Write

Regional Parameters

Country/Region :

Standard Tones

Type	Frequency 1	Frequency 2	Cadence1 (ms)	Cad
Dial Tone	400	425	0	0
Ring Tone	440	550	400	200
Busy Tone	425	425	375	375

OK
Cancel
Default

Select local Country/Region for correct telephony parameters.

Inbound Phone Book Add Entry

Accept Any Number

Remove Prefix:

Add Prefix:

Channel Number:

Description:

OK
Cancel

The Inbound Phone Book is for routing any incoming calls from the IP PBX to the GSM network. (Call Routing configuration on IP PBX determines which calls go to the gateway).

Add an entry using the Any Number option. The gateway will accept any digits from the PBX and place the call to the GSM network.

Select a specific channel or use hunting which will cause the gateway to select an available channel.

Inbound Phone Book

Remove Prefix	Add Prefix	Forward Address
Any Number		Not Used

Number of Entries:

Details

Channel No:

Description:

Registration Options

SIP

Register With SIP Proxy

Any entries in the Inbound Phone Book can be viewed from the List Entries option

Add Outbound Phone Book

Phone Number Details

Accept Any Number

Destination Pattern

Total Digits

Remove Prefix

Add Prefix

OK

Cancel

Advanced

IP Address

Description

Protocol Type

SIP

SIP

Use Proxy

Transport Protocol

TCP

UDP

SIP Port Number

SIP URL

Outbound Phone Book

Destination Pattern	IP Address	Protocol
Any Number	192.168.15.208	SIP

Number of Entries

Details

Remove Prefix

Add Prefix

SIP Proxy Server

SIP Port

Transport Protocol

SIP URL

Round Trip Delay ms

Alternate Phone Number

The Outbound Phone Book is for routing any incoming calls from the GSM network to the IP PBX. The destination extension on the IP PBX is defined in the 'AutoCall' Settings shown in the 2nd picture (above).

Add an entry using the Any Number option. The gateway will accept any digits from the GSM network and place the call to the IP PBX.

Any entries in the Outbound Phone Book can be viewed from the List Entries option

Summary:

You should now be able to make calls between the IP PBX and the GSM network.

Inbound Phone Book Edit Entry

Accept Any Number

Remove Prefix

Any Number

Add Prefix

1831

Channel Number

Channel 01

Description

OK

Cancel

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In many cases the gateway is used for outbound calls to customers and the caller ID may be concealed to prevent customers calling back on the SIM card in the gateway. If this is the preferred behaviour, prefix the CLI blocking code (1831) to the entry in the Inbound Phone book.

The called party will see 'Private Number' as the caller ID.