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Topic / Issue: Multitech GSM Gateway with Asterisk (elastix 1.3-2 but should be the same on any Asterisk Build)

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Calling from GSM network to Asterisk:

Multitech Gateway side:

- 1) Ethernet
Best to define a Static IP.
- 2) Voice/Fax
 - a) Coder (Codec) best to static define the Codec. G729 or G711 (Suggest G711 initially)
 - b) Auto Call - enable, Number should be the extension to send inbound GSM to asterisk calls. i.e. "101"
- 3) Call Signalling - SIP
 - a) Use SIP Proxy - enable
 - b) Primary Proxy = Asterisk IP.

SIP Parameters

Signaling Port

Use SIP Proxy

Allow Incoming Calls Through SIP Proxy Only

SIP Proxy Parameters

	Proxy Domain Name / IP Address	Port Number
Primary Proxy	<input type="text" value="172.30.0.12"/>	<input type="text" value="5060"/>
Alternate Proxy 1	<input type="text"/>	<input type="text" value="5060"/>
Alternate Proxy 2	<input type="text"/>	<input type="text" value="5060"/>

Append SIP Proxy Domain Name in User ID

Default Subscriber

Default Username

Default Password

Re-RegistrationTime

secs

Proxy Polling Interval

secs

TTL Value

secs

OK

Cancel

4) Phonebook - Outbound Phone Book

Configure simple receive anything on GSM - pass to Asterisk

a) Pattern - accept any number

b) Type = SIP, UDP, 5060

(The Account Details from the proxy settings above will be applied)

Edit Outbound Phone Book

Phone Number Details

Accept Any Number

Destination Pattern

Total Digits

Remove Prefix

Add Prefix

IP Address

Description

Protocol Type

SIP

SIP

Use Proxy

Transport Protocol

TCP UDP

SIP Port Number

OK

Cancel

Advanced

>>

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Double check that each screen is saved correctly, (click OK on each screen)
Then perform the "Save and Reboot" function to activate.

The above would result in any inbound GSM Call presenting like so.

Invite To:101@172.30.0.12

Authentication user will be MVP, password MVP.

NOW for the Asterisk Side:

It's assumed that Ext 101 exists and is registered.

Add a Trunk to Asterisk.

Trunk Type = SIP Trunk

Option

Unembedded freePBX

Basic

Extensions

Feature Codes

General Settings

Outbound Routes

Trunks

Inbound Call Control

Inbound Routes

Announcements

Follow Me

IVR

Misc Destinations

Queues

Ring Groups

Time Conditions

Internal Options & Configuration

Edit SIP Trunk

Delete Trunk MVP1

WARNING: This trunk is not used by any routes!

General Settings

Outbound Caller ID:

Never Override CallerID:

Maximum Channels:

Disable Trunk: Disable

Monitor Trunk Failures: Enable

Outgoing Dial Rules

Dial Rules:

Clean & Remove duplicates

- Conferences
- Misc Applications
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- Remote Access
- Callback
- DISA

Dial Rules Wizards:

Outbound Dial Prefix:

Outgoing Settings

Trunk Name:

PEER Details:

```
host=172.30.0.195
username=MVP
secret=MVP
type=peer
```

Incoming Settings

USER Context:

USER Details:

```
secret=MVP
type=user
context=from-trunk
```

Registration

Register String:

Once these Settings are saved to Asterisk Confirm successful Proxy registration from MVP to Asterisk Review Statistics - Servers - SIP Proxies on MVP. Registration should show OK.

Calling From Asterisk to GSM network.

On the Multitech side:

- 1) Phone Book - Inbound Phone Book
Configure the unit to "Accept Any Number"
Select GSM Channel - Hunting (or Specific)

On Asterisk:

- 1) Create an Outbound Route, and define the Dial Pattern.
(in this case all "." is selected)
- 2) Select the earlier configured Trunk. (MVP1)

Edit Route

Delete Route GSM_GW

Route Name: GSM_GW

Route Password:

PIN Set:

Emergency Dialing:

Intra Company Route:

Music On Hold?

Dial Patterns

Dial patterns wizards:

Trunk Sequence

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That's it, basic VoIP to GSM in and out from Asterisk.