

# DWG2008 Using With FreeSwitch



## ***Dinstar Technologies Co., Ltd.***

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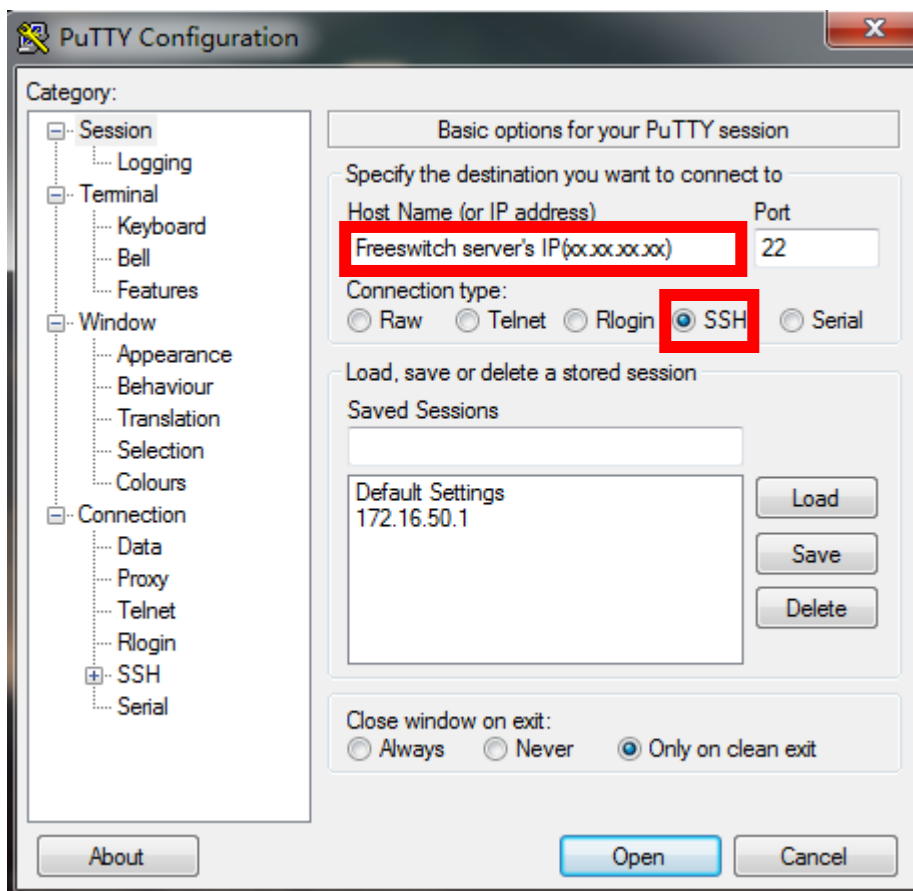
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1. User SSH to login to the FreeSwitch server, here we use putty, enter your server's IP address, chose SSH:



2. Enter the directory the freeswitch's default configuration directory, add the gateway's configuration in `/usr/local/freeswitch/conf/directory/default/`, Because the DWG2008's gsm ports will receive the incoming call which send to freeswitch with sip protocol, so the DWG2008's gsm port should register to the freeswitch. Here i just config one gsm ports of DWG2008, if you have more gsm ports which insert the sim card, please add other sip users.

*vi /usr/local/freeswitch/conf/directory/default/dag-fxo.xml*

add those in this file:

```

<include>
  <user id="dwg2008">
    <params>
      <param name="password" value="`${default_password}`"/>
      <param name="vm-password" value="9999"/><!--if vm-password is omitted
password param is used-->
    </params>
    <variables>
      <!--all variables here will be set on all inbound calls that originate from this user -->
      <variable name="user_context" value="public"/>
      <variable name="effective_caller_id_name" value=" dwg2008"/>
      <variable name="effective_caller_id_number" value="2008"/>
      <!-- Don't write a CDR if this is false valid values are: true, false, a_leg and b_leg -->
      <variable name="process_cdr" value="true"/>
    </variables>
  </user>
</include>

```

Note: the default\_password is "1234", default in freeswitch, of course you can change it to any number you want.

3. Edit the outband dialplan in /usr/local/freeswitch/conf/dialplan/default.xml

```
vi /usr/local/freeswitch/conf/dialplan/default.xml
```

here we set number's prefix which you want to send to the gateway is "0", add one extension:

```

<extension name="dwg2008_gateway">
  <condition field="destination_number" expression="^0(\d+)$">
    <action application="answer"/>
    <action application="set" data="ringback=${us-ring}"/>
    <action application="bridge" data="sofia/internal/$1@Gateway's IP"/>
  </condition>
</extension>

```

Please note the red line, the IP address is your gateway's IP.

4. Edit the outband dialplan in /usr/local/freeswitch/conf/dialplan/public/00\_inbound\_did.xml. Modify this file:

```

<include>
  <extension name="public_did">
    <condition field="destination_number" expression="^(900)$">
      <action application="set" data="domain_name=${domain}"/>
    </condition>
  </extension>

```

```

        <action application="transfer" data="1000 XML default"/>
    </condition>
</extension>
</include>

```

This configuration match the DID "900", when there is a incoming call, ring the extension 1000.

- Press "F6" in freeswitch's console to refresh configuration:

```

freeswitch@debian> 2011-05-31 19:17:56.635307 [INFO] mod_enum.c:808 ENUM Reloaded
2011-05-31 19:17:56.638435 [INFO] switch_time.c:915 Timezone reloaded 530 definitions

```

- Now we have finished the configuration of the freeswitch, let's config the DWG2008, first enter the IP address of the DWG2008 in the browser, the default username and password is the same: "admin".

Mobile Information										
Port	Type	IMSI	Status	Remaining Call Duration(min)	Carrier	Signal Quality	ASR(%)	ACD(s)	PDD(s)	Call Status
0	GSM	460029197588834	Mobile Registered	No Limit	CHINA MOBILE	↓	0	0	0	Active
1	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
2	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
3	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
4	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
5	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
6	GSM		No SIM Card	No Limit		↓	0	0	0	Idle
7	GSM		No SIM Card	No Limit		↓	0	0	0	Idle

SIP Information							
Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status
0		Registered	onhook	1		Unregistered	onhook
2	dwg2008	Unregistered	onhook	3		Unregistered	onhook
4		Unregistered	onhook	5		Unregistered	onhook
6		Unregistered	onhook	7		Unregistered	onhook

- Chose the "SIP config" option under the "System Parameter", then modify like below:

### SIP Configuration

#### SIP Proxy

SIP Server Address:

SIP Server Port(default: 5060):

#### Outbound Proxy

Outbound Proxy Address: 
Outbound Proxy Port:

#### Use Random Port

Local SIP Port: 
 No  Yes

#### Is Register

 No  Yes

Register Interval(range: 1 - 3600s):  s
  
T1:  ms
  
T2:  ms
  
T4:  ms
  
TMAX:  ms
  
Keepalive Interval(range:0 - 3600s,0 means disable):  s
  
Enable 100rel:  No  Yes
  
Refer to Use Target Contact:  No  Yes

Then click the “save” option to save this page.

8. Chose the “Port Config” option to config the fxo port, I just have one analog pstn line which insert to the port0, so the configuration should like this:

**Port Configuration**

All ports register used same user ID  No  Yes

Current Port

SIP User ID

Authenticate ID

Authenticate Password

Tx Gain

Rx Gain

To VOIP Hotline

To PSTN Hotline

Auto-Dial Delay Time  s

Save

This the username and password which config in FreeSwitch

Offhook Auto-Dial “900” is DID, it means the DWG208 will send the number “900” to freeswitch when the fxo receive the incoming call.

9. Chose the “Service Config” option to modify the last step:

Enable PSTN Incoming Configuration  No  Yes

Auto Outgoing Routing Type

IP to PSTN One Stage Dialing  No  Yes

Play Voice Prompt for PSTN Incoming Calls  No  Yes

Send Original Caller ID for PSTN Incoming Calls  No  Yes

**DTMF Parameter**

DTMF Method

RFC2833 Payload Type

DTMF Volume

DTMF Interval  ms

Enable STUN  No  Yes

Then save the webpage.

10. Chose “Port Group” option under the “Port Group Configuration”, add a gsm port group:

**Port Group Modify**

Index: 0

Description: all

Select Mode: cyclic ascending

Port:

- Port 0
- Port 1
- Port 2
- Port 3
- Port 4
- Port 5
- Port 6
- Port 7

OK    Reset    Cancel

Then click "OK".

11. Chose "IP->Tel Routing" option under the "Routing Configuration", add a routing group:

**IP->Tel Routing Modify**

Index: 31

Description: all

Source Prefix: any

Source IP:

- IP: Any
- IP Group:

Destination Prefix: any

Destination:

- Port: 0
- Port Group: 0 <all>

OK    Reset    Cancel

12. Chose "Tel->IP Operation" under "Operation", Add one:

**Tel->IP Operation Modify**

Index: 9

Source Prefix: any

Source Port:

- Port: Any
- Port Group: 0 <|>

Destination Prefix: any

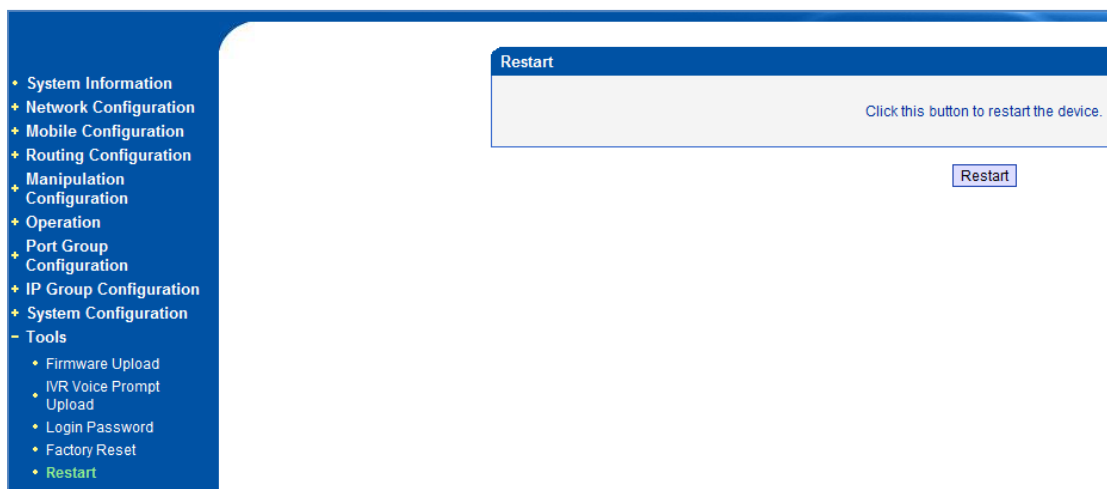
Operation:

- Forbid Call
- Allow Call
- Auto Call     Password Authentication

Description: auto

OK    Reset    Cancel

13. The last step is to restart the DWG2008, chose the “Restart” option under “Tools”, click the Restart.



Wait just one minute. Then use the sip phone which register with freeswitch to call out.

14. Use command “sofia status profile internal” in freeswitch console to check whether the gateway register to freeswitch:

```
Call-ID: 07129518405fe410cbfb86aba03b263f@172.16.50.3
User: dwg2008@172.16.50.19
Contact: "user" <sip:dwg2008@172.16.50.3>
Agent: dunno
Status: Registered(UDP)(unknown) EXP(2011-05-31 23:08:34)
Host: debian
IP: 172.16.50.3
Port: 5060
Auth-User: dwg2008
Auth-Realm: 172.16.50.19
MWI-Account: dwg2008@172.16.50.19
```

15. Use command “show calls” to show the call status:

```
freeswitch@debian> show calls
call_created,call_created_epoch,function,caller_cid_name,caller_cid_num,caller_dest_num,caller_chan_name,caller_uuid,callee_cid_name,callee_cid_num,callee_dest_num,callee_chan_name,callee_uuid,hostname
2011-05-31 22:15:27,1306851227,switch_ivr_multi_threaded_bridge,1234,1234,010010,sofia/internal/1234@172.16.50.19,6cd5c04a-8b90-11e0-b6fa-6f81408a8e6c,Extension 1234,1234,10010,sofia/internal/10010@172.16.50.3,6ce99c96-8b90-11e0-b6fb-6f81408a8e6c,debian
1 total.
```