

# DWG2008 Using With Elastix



## *Dinstar Technologies Co., Ltd.*

**Address:** Floor 6, Guoxing Building, Changxing Road, Nanshan District, Shenzhen, China 518057

**Telephone:** +86 755 6130 2265

**Fax:** +86 755 2645 6659

**Emails:** sales@dinstar.com, support@dinstar.com

**Website:** [www.dinstar.com](http://www.dinstar.com)



1. Enter your elastix server's ip in the web browser, then input username and password,  
When you logging in, click the “PBX” label, then “Trunk”:

The screenshot shows the Elastix PBX Configuration interface. The top navigation bar includes tabs for System, Agenda, Email, Fax, **PBX** (which is highlighted with a red box), and IM. Below the navigation bar, a sidebar lists various configuration options: Option, Unembedded freePBX, Basic, Extensions, Feature Codes, General Settings, **Inbound Routes** (which is highlighted with a red box), Trunks, Inbound Call Control, Inbound Routes, Zap Channel DIDs, Announcements, Blacklist, CallerID Lookup Sources, and Day/Night Control. The main content area is titled "Add a Trunk" and lists several trunk types with plus signs: Add Zap Trunk (DAHDI compatibility mode), Add SIP Trunk, Add IAX2 Trunk, Add ENUM Trunk, Add DUNDI Trunk, and Add Custom Trunk.

2. Choose the “Add SIP Trunk” option, then input the following configurations in “Outgoing Settings”:

**Languages**

**Misc Applications**

**Misc Destinations**

**Music on Hold**

**PIN Sets**

**Paging and Intercom**

**Parking Lot**

**System Recordings**

**VoiceMail Blasting**

**Remote Access**

**Callback**

**DISA**

**Outgoing Settings**

Trunk Name:

PEER Details:

```
canreinvite=no
host=xx.xx.xx.xx (DWG's IP)
context=from-pstn
insecure=no
port=5060
type=friend
```

**Incoming Settings**

then click "Submit Changes" option in the bottom of this webpage.

- Click the "Outbound Routes", chose the sip trunk "sip\_DAG" in Trunk Sequence":

**Feature Codes**

**General Settings**

**Outbound Routes**

**Trunks**

**Inbound Call Control**

**Inbound Routes**

**Zap Channel DIDs**

**Announcements**

**Blacklist**

**CallerID Lookup Sources**

**Day/Night Control**

**Follow Me**

**IVR**

**Queue Priorities**

**Queues**

**Ring Groups**

**Time Conditions**

**Time Groups**

**Internal Options & Configuration**

**Conferences**

**Languages**

**Route Name:** 9\_outside

**Route CID:**   Override Extension CID

**Route Password:**

**PIN Set:**

**Emergency Dialing:**

**Intra Company Route:**

**Music On Hold?**

**Dial Patterns**

X.

**Dial patterns wizards:**

**Trunk Sequence**

0

**Submit Changes**

- Because the DWG2008's gsm ports will receive the incoming call which send to elastix with sip, so the DWG2008's gsm port should register to the elastix. Here i just config two gsm ports of DWG2008, if you have more gsm ports which insert the sim card, please add other sip trunks like the follow image. Chose the "Add SIP Trunk" option, then input the following configurations in "Outgoing Settings":

- [Languages](#)
- [Misc Applications](#)
- [Misc Destinations](#)
- [Music on Hold](#)
- [PIN Sets](#)
- [Paging and Intercom](#)
- [Parking Lot](#)
- [System Recordings](#)
- [VoiceMail Blasting](#)
- [Remote Access](#)
- [Callback](#)
- [DISA](#)

**Outgoing Settings**

Trunk Name:

PEER Details:

```
context=from-pstn
host=dynamic
qualify=yes
username=wport1
secret=wport1
type=peer
```

Incoming Settings

(the username and secret are important, we will use it in DWG2008)

5. When you finish to add the second sip trunk, you will find a red line “Apple Configuration Changes Here” on the top of the webpage, just click it.



6. Now we have finished the configuration of the elastix, 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	
0	GSM	460013611933436	Mobile Registered	No Limit	CHINA UNICOM GSM		
1	GSM	460013611933406	Mobile Registered	No Limit	CHINA UNICOM GSM		
2	GSM		No SIM Card	No Limit			
3	GSM		No SIM Card	No Limit			
4	GSM		No SIM Card	No Limit			
5	GSM		No SIM Card	No Limit			
6	GSM		No SIM Card	No Limit			
7	GSM		No SIM Card	No Limit			

SIP Information							
Port	SIP User ID	Register Status	Status	Port	SIP User ID	Register Status	Status
0	wport1	Registered	onhook	1	wport2	Registered	onhook
2		Unregistered	onhook	3		Unregistered	onhook
4		Unregistered	onhook	5		Unregistered	onhook
6		Unregistered	onhook	7		Unregistered	onhook

7. Choose the “SIP config” option under the “System Parameter”, then modify like below:

**SIP Configuration**

**SIP Proxy**

SIP Server Address	172.16.50.1
SIP Server Port(default: 5060)	5060

**Outbound Proxy**

Outbound Proxy Address	
Outbound Proxy Port	5060

**Use Random Port**

Local SIP Port	5060
----------------	------

**Is Register**

Register Interval(range: 1 - 3600s)	1800	s
-------------------------------------	------	---

T1: 500 ms  
T2: 4000 ms  
T4: 5000 ms  
TMAX: 32000 ms  
Keepalive Interval(range: 0 - 3600s, 0 means disable): 10 s  
Enable 100rel: No  
Refer to Use Target Contact: No

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

8. Choose 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**: Port 0

SIP User ID	wport1
Authenticate ID	wport1
Authenticate Password	*****

Tx Gain: 0dB  
Rx Gain: -2dB

To VOIP Hotline	s
To PSTN Hotline	
Auto-Dial Delay Time	3 s

**Note:** This the username and password which config in elastix's sip trunk.  
Offhook Auto-Dial is “s”, it means the DAG1000 will send the number “s” to elastix when the fxo receive the incoming call.

Save

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

Enable PSTN Incoming Configuration  No  Yes  
 Auto Outgoing Routing Type  Polling  
 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
RFC2833 Payload Type	101
DTMF Volume	0dB
DTMF Interval	200 ms

**Enable STUN**  No  Yes

Then save the webpage.

10. Choose “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	<input checked="" type="checkbox"/> Port 0 <input checked="" type="checkbox"/> Port 1 <input checked="" type="checkbox"/> Port 2 <input checked="" type="checkbox"/> Port 3 <input checked="" type="checkbox"/> Port 4 <input checked="" type="checkbox"/> Port 5 <input checked="" type="checkbox"/> Port 6 <input checked="" type="checkbox"/> Port 7

OK    Reset    Cancel

Then click “OK”.

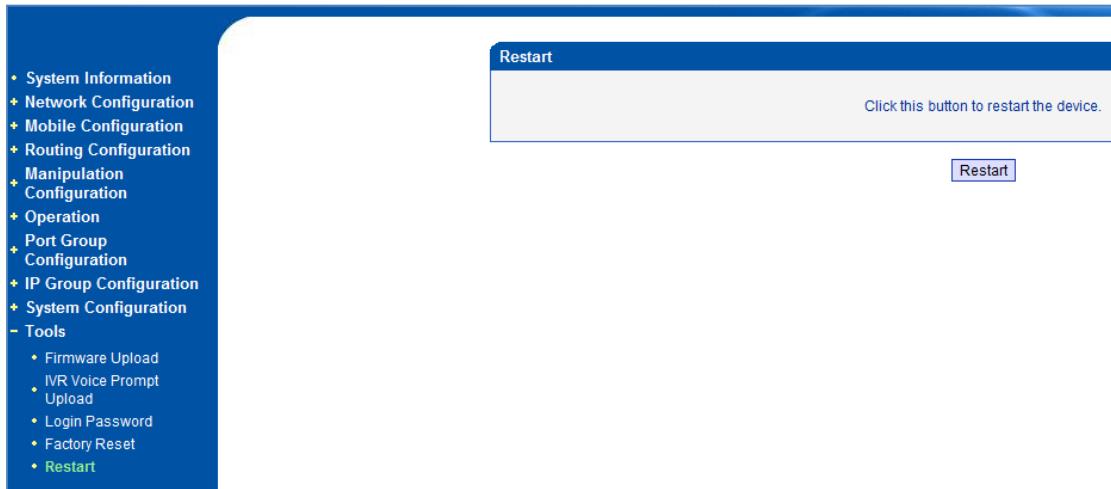
11. Choose “IP->Tel Routing” option under the “Routing Configuration”, add a routing group:

**IP->Tel Routing Modify**

Index	31
Description	all
Source Prefix	any
Source IP	<input checked="" type="radio"/> IP    Any <input type="radio"/> IP Group
Destination Prefix	any
Destination	<input type="radio"/> Port    0 <input checked="" type="radio"/> Port Group    0 <all>

OK    Reset    Cancel

12. The last step is to restart the DAG1000, chose the “Restart” option under “Tools”, click the Restart.



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