

# DWG2008/2016 Using With AskoziaPBX



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

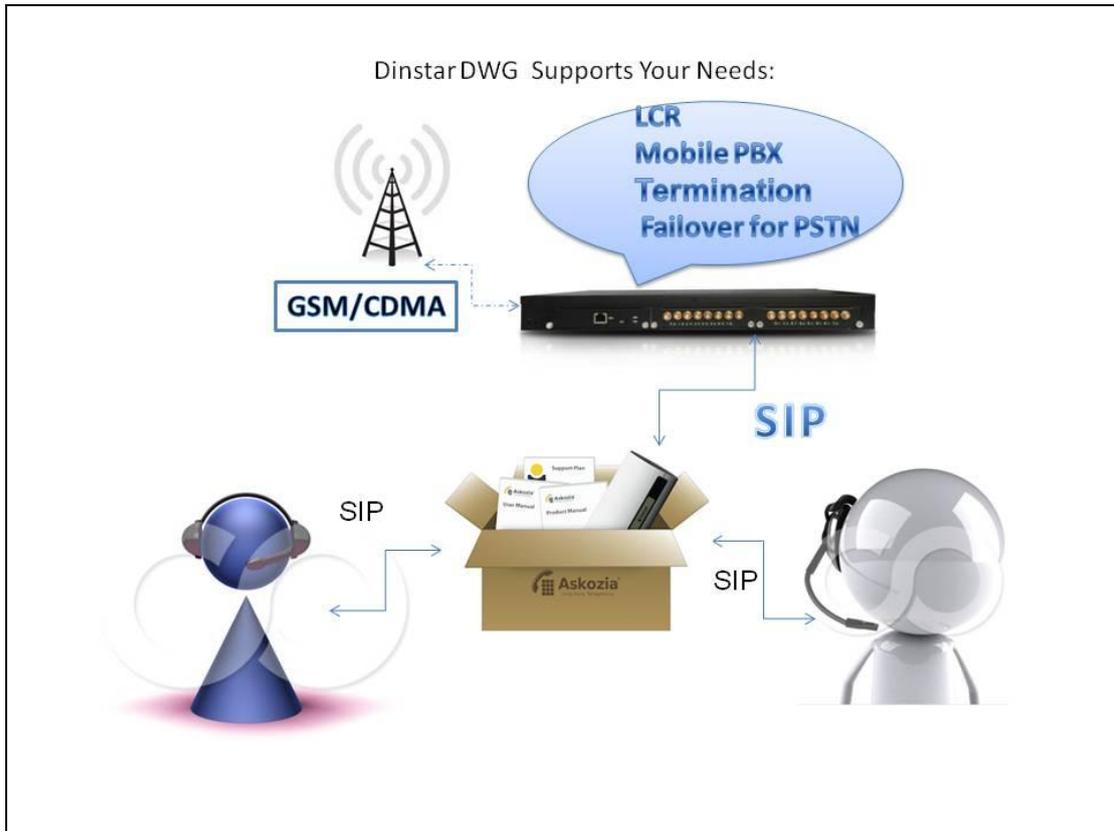
**Address:** Floor 6, Guoxing Building, Changxing Road, Nanshan District, Shenzhen, China 518057

**Telephone:** +86 755 6130 2265

**Fax:** +86 755 2645 6659

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

**Website:** www.dinstar.com



1. Enter your AskoziaPBX server's ip in the web browser, then input username and password, When you login in, click the "Phones" label, then "SIP":

**System**

- General Setup
- Storage
- Firmware
- Backup/Restore
- Factory Defaults
- Reboot

**Connectivity**

- Networking
- Telephony Ports
- Gateways

**Accounts**

- Phones**
- Providers
- Faxes

**Dialplan (print)**

- Applications
- Call Groups
- Transfers

**Accounts: Phones**

+ SIP    IAX ⌵    External ⌵

**SIP**

Extension	Caller ID	Description
💡 <span style="font-size: 0.8em;">+</span> 101	101	101 <span style="float: right;">⚙️ -</span>

Add a sip extension 101, password "123456".

<b>System</b> General Setup Storage Firmware Backup/Restore Factory Defaults Reboot <b>Connectivity</b> Networking Telephony Ports Gateways <b>Accounts</b> Phones Providers Faxes <b>Dialplan (print)</b> Applications Call Groups Transfers <b>Services</b> Notifications Voicemail Conferencing <b>Status</b> Summary Ports Network Traffic	<b>Accounts: Edit SIP Phone</b>	
	<b>General Settings</b>	
	<b>Number</b>	101
	The number used to dial this phone. Use this number as your username.	
	<b>Caller ID</b>	101
	Text to be displayed for Caller ID.	
	<b>Language</b>	English (US)
	Audio prompts will be played back in the selected language for this account.	
	<b>Ring Length</b>	indefinitely
	The number of seconds this phone will ring before giving up or going to voicemail.	
	<b>Description</b>	101
	You may enter a description here for your reference (not parsed).	
<b>Security</b>		
<b>Password</b>	123456	
This account's password.		
<b>Public Access</b>	<input type="checkbox"/> allow this number to be reachable over the Internet	
<b>Block Providers</b>	<input checked="" type="checkbox"/> dwg	
Block access to the providers selected above.		

2. Chose the "Providers" option, then click "SIP" label:

<b>System</b> General Setup Storage Firmware Backup/Restore Factory Defaults Reboot <b>Connectivity</b> Networking Telephony Ports Gateways <b>Accounts</b> Phones Providers Faxes <b>Dialplan (print)</b> Applications Call Groups Transfers <b>Services</b> Notifications Voicemail Conferencing <b>Status</b> Summary Ports Network Traffic CPU Load Logs ▶ <b>Advanced</b>	<b>Accounts: Edit SIP Provider</b>	
	<b>General Settings</b>	
	<b>Name</b>	dwg
	Descriptive name for this provider.	
	<b>Host</b>	172.16.90.90
	Provider host OR IP address and optional port.	
	<b>Username</b>	100
	The DWG gateway's use "100" as sip user id	
	<b>Password</b>	
	<b>Language</b>	English (US)
	Audio prompts will be played back in the selected language for this account.	
	<b>Public Number</b>	
This 'external' number will be read back to the caller when reaching voicemail; defaults to account's username if it is numeric. If it is not, the internal extension this call was routed to will be read back.		
<b>Call Routing</b>		
<b>Outgoing Patterns</b>	9 .	
Enter patterns, one per line, to define this provider's outgoing calls. To access the "X!" to use this provider for all outgoing calls. To access the "9" prefix, enter "9 .". If no patterns are entered, only incoming calls will be accepted.		
<ul style="list-style-type: none"> <li>• + - adds a prefix (i.e. "1+555" matches "555" and passes "1555" to the provider)</li> <li>•   - removes a prefix (i.e. "1 555" matches "1555" but only passes "555" to the provider)</li> <li>• X - matches digits 0-9</li> <li>• Z - matches digits 1-9</li> <li>• N - matches digits 2-9</li> <li>• [13-5] - matches any digit in the brackets (here, 1,3,4,5)</li> <li>• . - matches one or more characters (not allowed before   or +)</li> <li>• ! - matches zero or more characters (not allowed before   or +)</li> </ul>		
<b>Incoming Extensions</b>	101 <101>	
When the pbx get a incoming call from dwg, the extension 101 will ring		

Failover Provider	<input type="text" value="none"/> <p>If an outgoing call to this Provider fails, try routing it over the account selected above. Dialpatterns must be compatible, no extra care is yet taken to match them.</p>
Caller ID Options	
Codecs	
Advanced Options	
Registration	<input checked="" type="checkbox"/> Do not register with this provider. <input type="text"/> <p>Completely override the generated registration string with a manually defined one.</p>
Qualify	<input type="text" value="2"/> seconds The provider will be contacted this often to check its availability.
NAT	<input type="text" value="always use NAT mode."/> Is there a NATing device between the PBX and this account?
DTMF Mode	<input type="text" value="rfc2833"/>
Authorization User	<input type="text"/> Some providers require a separate username for authorization. Defaults to username entered above.
From User	<input type="text"/> Some providers require a separate 'from' user. Defaults to username entered above.

- Now we have finished the configuration of the AskoziaPBX, 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	[Signal Quality Icon]	
1	GSM	460013611933406	Mobile Registered	No Limit	CHINA UNICOM GSM	[Signal Quality Icon]	
2	GSM		No SIM Card	No Limit		[Signal Quality Icon]	
3	GSM		No SIM Card	No Limit		[Signal Quality Icon]	
4	GSM		No SIM Card	No Limit		[Signal Quality Icon]	
5	GSM		No SIM Card	No Limit		[Signal Quality Icon]	
6	GSM		No SIM Card	No Limit		[Signal Quality Icon]	
7	GSM		No SIM Card	No Limit		[Signal Quality Icon]	

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

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

**SIP Configuration**

**SIP Proxy**

SIP Server Address: 172.16.0.147

SIP Server Port(default: 5060): 5060

**Outbound Proxy**

Outbound Proxy Address: [ ]

Outbound Proxy Port: 5060

**Use Random Port**

Local SIP Port: 5060

**Is Register**  No  Yes

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  Yes

Refer to Use Target Contact:  No  Yes

From Mode when Caller ID Is Available: Tel/User

From Mode when Caller ID Is Unavailable: Anonymouse

Answer Mode: Answered

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

5. 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: Port 0

SIP User ID: 100

Authenticate ID: [ ]

Authenticate Password: [ ]

Tx Gain: 0dB

Rx Gain: -2dB

To VOIP Hotline: 900

To PSTN Hotline: [ ]

Auto-Dial Delay Time: 0

Save

Offhook Auto-Dial is “900”, it means the DWG2008 will send the number “900” to AskoziaPBX when the fxo receive the incoming call.

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

Enable PSTN Incoming Configuration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Auto Outgoing Routing Type	Polling
IP to PSTN One Stage Dialing	<input type="radio"/> No <input checked="" type="radio"/> Yes
Redirect Call When All Ports Busy	<input checked="" type="radio"/> No <input type="radio"/> Yes
Play Voice Prompt for PSTN Incoming Calls	<input type="radio"/> No <input checked="" type="radio"/> Yes
<b>DTMF Parameter</b>	
DTMF Method	RFC2833
RFC2833 Payload Type	101
DTMF Volume	0dB
DTMF Interval	200 ms

Then save the webpage.

7. 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	<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”.

8. 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	<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	

9. Chose “Tel->IP Operation” under “Operation”, Add one:

**Tel->IP Operation Modify**

Index: 9

Source Prefix: any

Source Port:  Port Any  Port Group 0 <I>

Destination Prefix: any

Operation:  Forbid Call  Allow Call

Auto Call  Password Authentication

Description: auto

OK Reset Cancel

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

**Restart**

Click this button to restart the device.

Restart

- System Information
- + Network Configuration
- + Mobile Configuration
- + Routing Configuration
- + Manipulation Configuration
- + Operation
- + Port Group Configuration
- + IP Group Configuration
- + System Configuration
- Tools
  - Firmware Upload
  - IVR Voice Prompt Upload
  - Login Password
  - Factory Reset
  - Restart

Wait just one minute. Then use the sip phone 101 which registers with AskoziaPBX to call out and call in.