

How to config DWG with Asterisk



Dinstar Technologies Co., Ltd.

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

518057

Telephone: +86 755 2645 6664

Fax: +86 755 2645 6659

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

Website: www.dinstar.com

Contents

1. Config Asterisk.....	2
1.1 sip.conf.....	3
1.2 extensions.conf.....	3
2. Config DWG.....	3
2.1 System Configuration->SIP Configuration.....	3
2.2 System Configuration->Port Configuration.....	4
2.3 System Configuration->Service Configuration.....	4
2.4 Port Group Configuration--->Port Group.....	5
2.5 Operation.....	6
2.6 Routing Configuration.....	7

1. Config Asterisk

Dailplan: the calls from DWG will be sent to extension 101. Internal extensions can add "9" as the prefix to call out.

1.1 sip.conf

```
[general]
context=default
allowoverlap=no
bindport=5060
bindaddr=0.0.0.0
srvlookup=yes
```

```
[101]
type=friend
secret=101
host=dynamic
context=from-internal
username=101
```

```
[dwg]
type=friend
secret=dwg
host=dynamic
context=from-PSTN
username=dwg
```

1.2 extensions.conf

```
[default]
```

```
[from-internal]
exten => 101,1,Dial(sip/101,10)
exten => _9.,1,Dial(sip/dwg/${EXTEN:1},30)
```

```
[from-PSTN]
exten => s ,1,Goto(from-internal,101,1)
```

2. Config DWG

2.1 System Configuration->SIP Configuration

"SIP server Address" fill the IP of Asterisk.

"Is Register" select **Yes**

"From Mode when Caller ID Is Available" Select **Tel/Tel**

SIP Configuration	
SIP Proxy	<input type="text" value="172.16.33.52"/> <input type="text" value="5060"/>
Outbound Proxy	<input type="text"/> <input type="text" value="5060"/>
Use Random Port	<input checked="" type="radio"/> No <input type="radio"/> Yes <input type="text" value="5060"/>
Is Register	<input checked="" type="radio"/> No <input checked="" type="radio"/> Yes <input type="text" value="1800"/> s
T1	<input type="text" value="500"/> ms
T2	<input type="text" value="4000"/> ms
T4	<input type="text" value="5000"/> ms
TMAX	<input type="text" value="32000"/> ms
Keepalive Interval(range:0 - 3600s,0 means disable)	<input type="text" value="10"/> s
Enable 100rel	<input checked="" type="radio"/> No <input type="radio"/> Yes
Refer to Use Target Contact	<input checked="" type="radio"/> No <input type="radio"/> Yes
From Mode when Caller ID Is Available	<input type="text" value="Tel/Tel"/>
From Mode when Caller ID Is Unavailable	<input type="text" value="Anonymous"/>
Answer Mode	<input type="text" value="Answered"/>

2.2 System Configuration->Port Configuration

Fill the SIP Account, "To VOIP Hotline" Fill "**s**" , DWG will send "s" to Asterisk , Which will match the dailplan " exten => s ,1,Goto(from-internal,101,1) ".

Port Configuration	
All ports register used same user ID	<input checked="" type="radio"/> Yes
Current Port	<input type="text" value="Port 0"/>
SIP User ID	<input type="text" value="dwg"/>
Authenticate ID	<input type="text" value="dwg"/>
Authenticate Password	<input type="text" value="***"/>
Tx Gain	<input type="text" value="0dB"/>
Rx Gain	<input type="text" value="-2dB"/>
To VOIP Hotline	<input type="text" value="s"/>
To PSTN Hotline	<input type="text" value="0"/>
Auto-Dial Delay Time	<input type="text" value="s"/>

2.3 System Configuration->Service Configuration

"IP to PSTN One Stage Dialing" select to **yes** "DTMF Method "select to **RFC2833**

"User ID Is Phone Number" select to **Yes** "Only Accept Calls from SIP Server" select to **No**

"Allow Call from PSTN to IP without Registration" select to **Yes**

"Allow Call from IP to PSTN without Registration" select to **Yes**

"Reject Anonymous Call from IP to PSTN" select to **No**

Service Configuration	
Local Start RTP Port	8000
Enable Silence Suppression	<input type="radio"/> No <input checked="" type="radio"/> Yes
Call Progress Tone	USA
Preferred Coders(in listed order)	
1st	PCMA
2nd	PCMU
3rd	G.729AB
Voice Frames per Tx	2
Notice: The device will restart automatically when 'preferred coders' is changed between G.723.1 and G.729AB.	
Enable PSTN Incoming Configuration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Auto Outgoing Routing Type	Calling Line
IP to PSTN One Stage Dialing	<input type="radio"/> No <input checked="" type="radio"/> Yes
Answer Delay	5 s
Redirect Call When All Ports Busy	<input type="radio"/> No <input checked="" type="radio"/> Yes
Play Voice Prompt for PSTN Incoming Calls	<input type="radio"/> No <input checked="" type="radio"/> Yes
DTMF Parameter	
DTMF Method	RFC2833
RFC2833 Payload Type	101
DTMF Volume	+2dB
DTMF Interval	200 ms
Enable STUN	<input type="radio"/> No <input checked="" type="radio"/> Yes
Other Configuration	
Enable Private Service	<input type="radio"/> No <input checked="" type="radio"/> Yes
User ID Is Phone Number	<input type="radio"/> No <input checked="" type="radio"/> Yes
Only Accept Calls from SIP Server	<input type="radio"/> No <input checked="" type="radio"/> Yes
Allow Call from PSTN to IP without Registration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Allow Call from IP to PSTN without Registration	<input type="radio"/> No <input checked="" type="radio"/> Yes
Reject Anonymous Call from IP to PSTN	<input type="radio"/> No <input checked="" type="radio"/> Yes
Use # as End Key	<input type="radio"/> No <input checked="" type="radio"/> Yes
Interdigit Timeout	4 s

2.4 Port Group Configuration--->Port Group

Click "add" to add a port group, Select the port including in this group.

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 type="checkbox"/> Port 4 <input type="checkbox"/> Port 5 <input type="checkbox"/> Port 6 <input type="checkbox"/> Port 7

2.5 Operation

Go to the menu "Operation" ---> "IP->Tel Operation", Click "add" .
"any" will match all numbers.

Index	31
Source Prefix	any
Source IP	<input type="radio"/> IP 31 <x-lite> <input type="radio"/> IP Group
Destination Prefix	<input checked="" type="radio"/> SIP Server any
Operation	<input type="radio"/> Forbid Call <input checked="" type="radio"/> Allow Call <input type="checkbox"/> Auto Call <input type="checkbox"/> Password Authentication
Description	any

Go to the menu "Operation" ---> "Tel-> IP Operation", Click "add" .

"any" will match all numbers.

"Auto Call" selected. When call make from GSM network to DWG , DWG will send auto send call to Asterisk .

Index	31
Source Prefix	any
Source Port	<input type="radio"/> Port 0 <input checked="" type="radio"/> Port Group 0 <all>
Destination Prefix	any
Operation	<input type="radio"/> Forbid Call <input checked="" type="radio"/> Allow Call <input checked="" type="checkbox"/> Auto Call <input type="checkbox"/> Password Authentication
Description	test

2.6 Routing Configuration

Go to the menu "Routing Configuration" ---> "IP->Tel Routing", Click "add" .

Index	0
Description	default
Source Prefix	any
Source IP	<input type="radio"/> IP 31 <x-lite> <input type="radio"/> IP Group
Destination Prefix	<input checked="" type="radio"/> SIP Server any
Destination	<input type="radio"/> Port 0 <input checked="" type="radio"/> Port Group 0 <all>

Go to the menu "Routing Configuration" ---> "Tel->IP Routing", Click "add" .

Index	0
Description	default
Source Prefix	any
Source	<input type="radio"/> Port 0 <input checked="" type="radio"/> Port Group 0 <all>
Destination Prefix	any
Destination	<input type="radio"/> Port 0 <input type="radio"/> Port Group 0 <all> <input type="radio"/> IP 31 <x-lite> <input type="radio"/> IP Group <input checked="" type="radio"/> SIP Server