

INTERRA

INTERRA SIP SERVER INSTRUCTION MANUAL



INTERRA

OVERVIEW

This guide includes required information for installation and making the settings of Sip Server product. Please read this manual carefully before using the program.

In case those information are ignored, please do not forget that Yonnet Bilisim shall not accept any reliability, and the device shall not be covered by the warranty.

Please keep your User Manual since it is an crucial source for the reliably and securely use of your device.

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1 USER LOGIN

It is enough to write server ip in order to login Interra Sip Server. When Interra Sip Server interface is appeared, a user name and password is requested from you.

Default values:

Username : admin

Password : yonnet

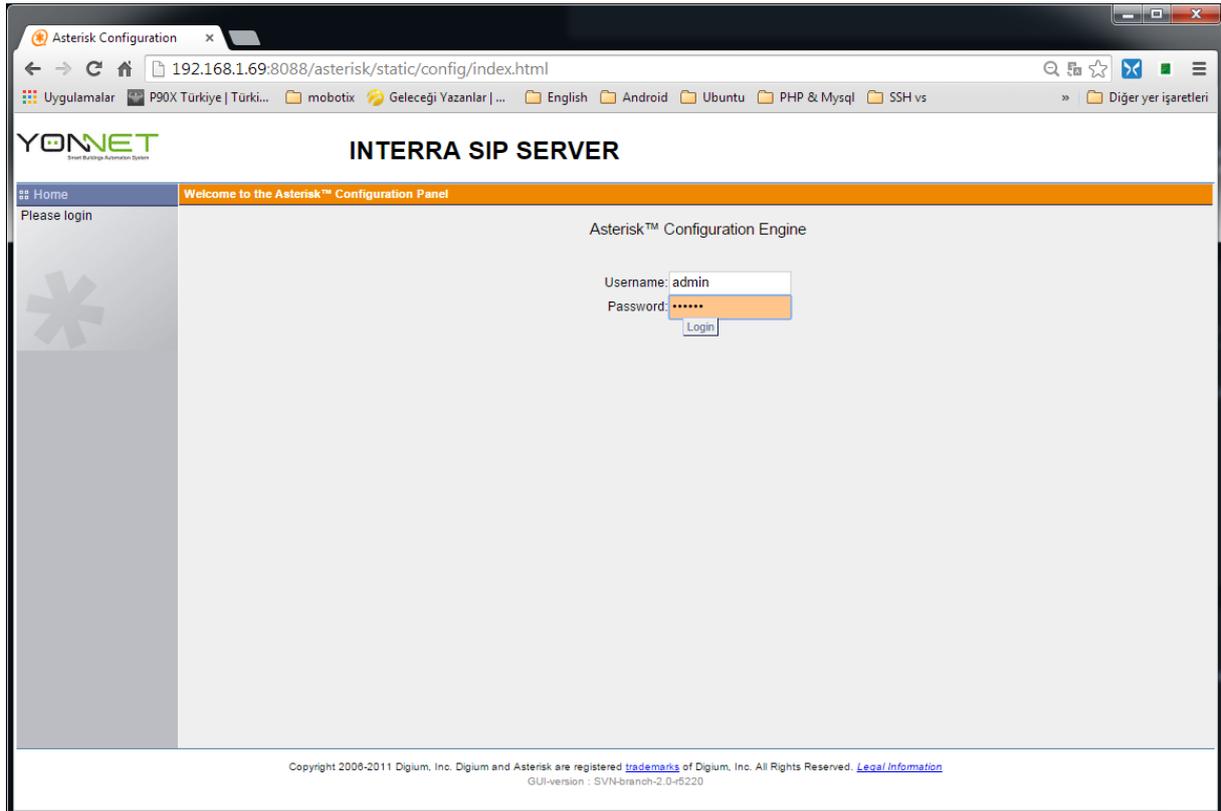


Figure 1 Interra Sip Server User Login

2 HOME PAGE– WELCOME SCREEN

You will be directed to System Status screen after login with username and password. You can see logged in users in sip server and their status in this screen.

The screenshot displays the Asterisk Configuration web interface for the Interra SIP Server. The browser address bar shows the URL: 192.168.1.69:8088/asterisk/static/config/index.html. The page title is "INTERRA SIP SERVER".

System Status

Please click on a panel to manage related features

Extensions

Extension	Name/Label	Status	Type
1000	1000	Messages: 0/0	SIP User
1001	1001	Messages: 0/0	SIP User
1002	1002	Messages: 0/0	SIP User
1003	1003	Messages: 0/0	SIP User
1004	1004	Messages: 0/0	SIP User
1005	1005	Messages: 0/0	SIP User
1006	1006	Messages: 0/0	SIP User
1007	1007	Messages: 0/0	SIP User
1008	1008	Messages: 0/0	SIP User
1009	1009	Messages: 0/0	SIP User
1010	1010	Messages: 0/0	SIP User
1011	1011	Messages: 0/0	SIP User
1012	1012	Messages: 0/0	SIP User
1013	1013	Messages: 0/0	SIP User
1014	1014	Messages: 0/0	SIP User
10800	10800	Messages: 0/0	SIP User
--	200		Ring Group
--	201		Ring Group
--	*No Extension assigned	Check Voicemails	VoiceMailMain
--	*No Extension assigned	Dial by Names	Directory

System Info

General Network Memory Disk

Hostname:
13-c753

OS Version:
Linux 13-c753 3.4.79-ectrac+ #3 SMP PREEMPT Sun May 4 23:45:43 EEST 2014 armv7l GNU/Linux

Asterisk Build:
Asterisk/1.8.13.1-dfsg1-3+deb7u3
Asterisk GUI-version : SVN-branch-2.0-r5220

Server Date & Timezone
Fri Dec 19 14:00:55 EET 2014

Uptime:
14:00:55 up 24 days, 21:12, 2 users,
Load Average: 1.09, 1.09, 1.08

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GUI-version : SVN-branch-2.0-r5220

Figure 2 Home Page

3 NETWORK

You will see NETWORK button enabling you to make network settings, on upper right corner of Home Page. You can change ip address, net mask, gateway and dns settings of sip server from network screen. Device should be restarted after modifications are performed.

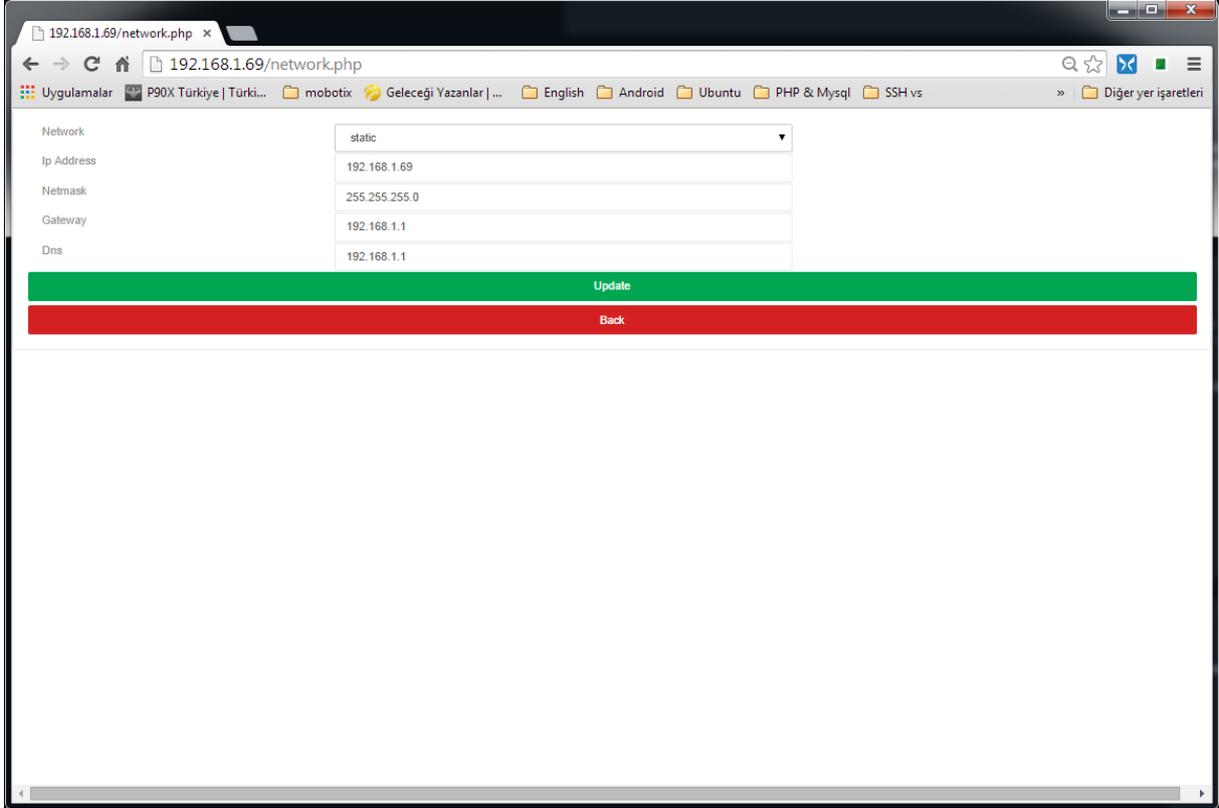


Figure 3 Network Settings

4 USERS

When you enter Users from left menu, you will see previously created users. You can add new users, edit or delete existing users in this screen.

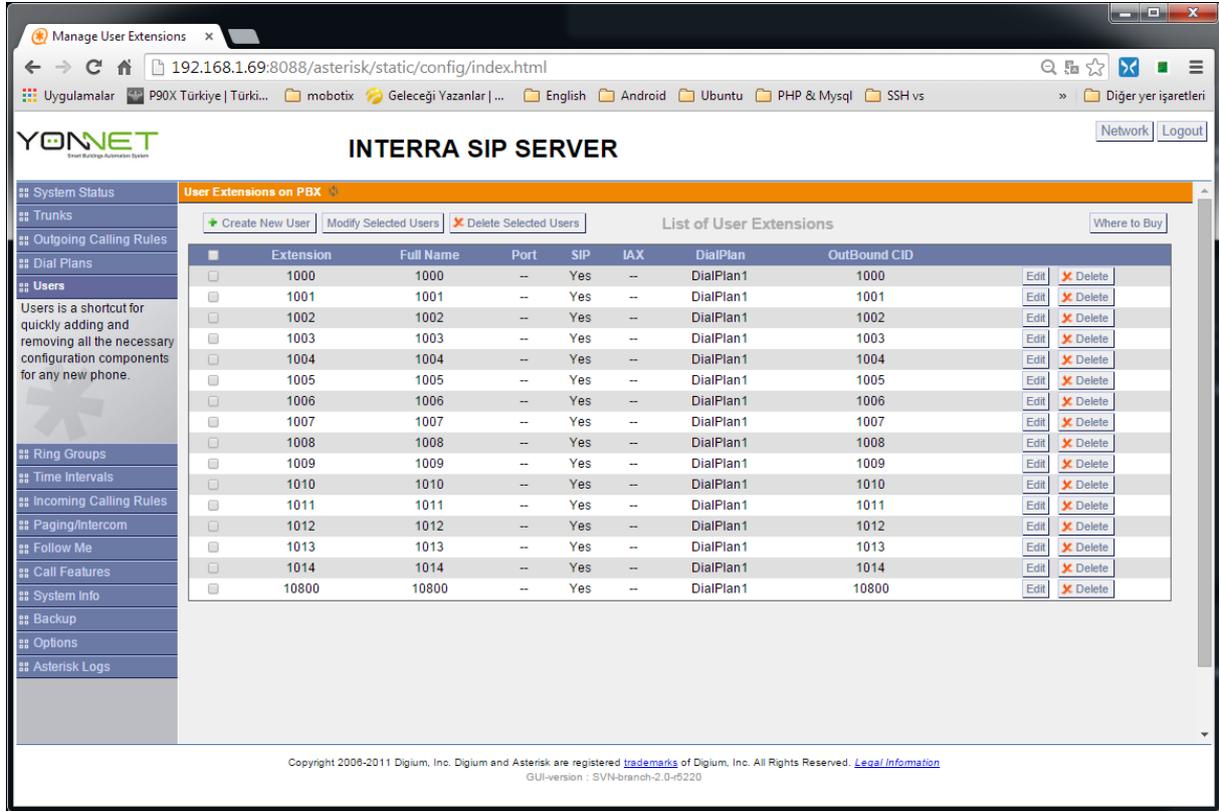


Figure 4 User Screen

Add New User page is displayed with Create New User button. You will have to fill the fields specified below.

Extension: User ID

CallerId Name : User name

CallerId No : User no

SIP : Active/Inactive status of SIP Protocol Make it active.

IAX : Active/Inactive status of IAX Protocol Make it inactive.

Codec Preference : Displays active codecs.

MAC Address : Mac adress of user. User no could be entered.

SIP/IAX Password : User's password.

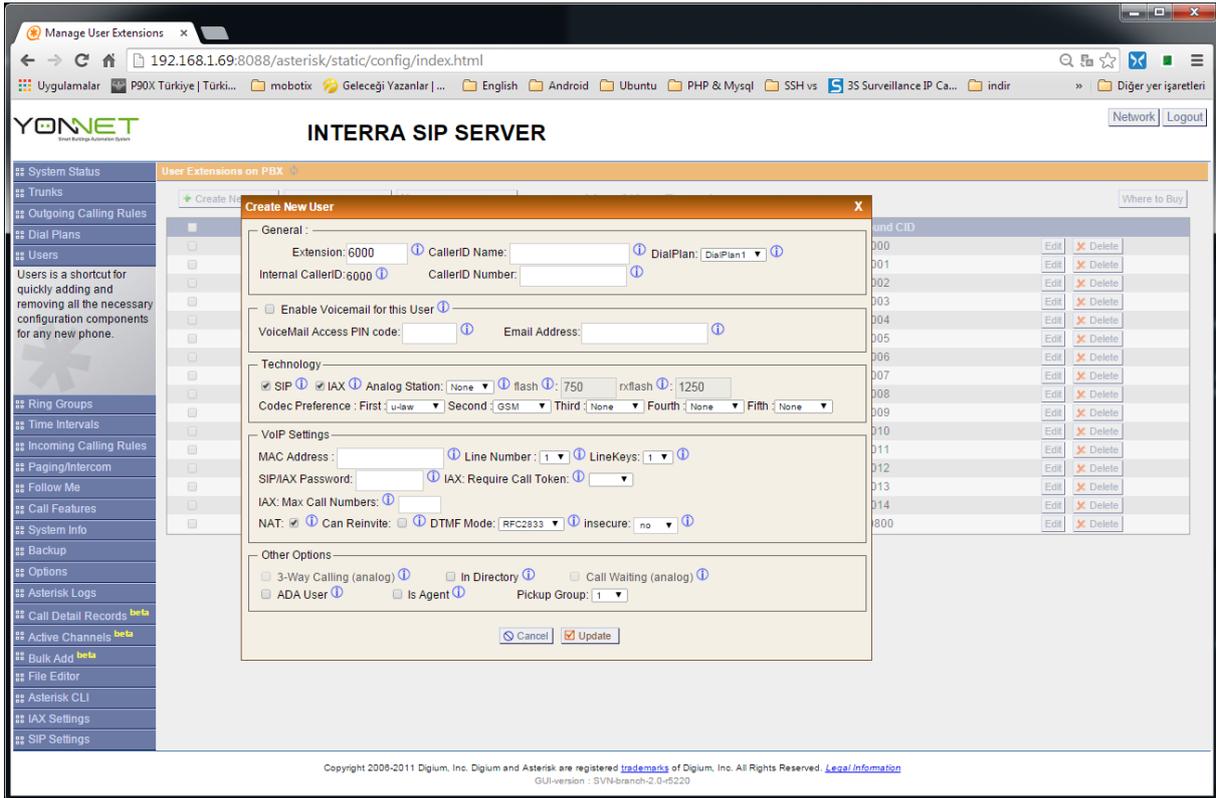


Figure 5 New User

An Example of User Settings:

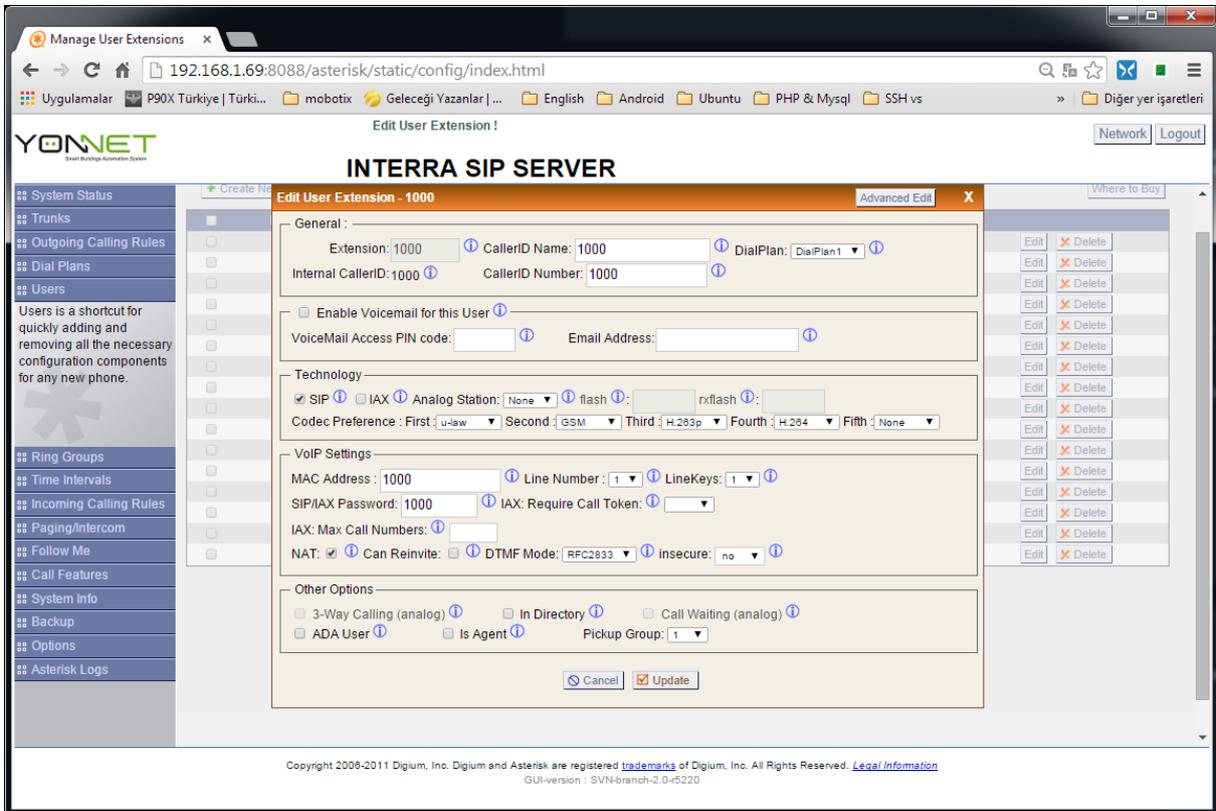


Figure 6 Sample User

When extra settings should be needed about users, those settings shall be performed by entering Advanced Edit in user edit screen.

The screenshot shows the 'Edit User 1000 - Advanced' dialog box in the YONET INTERRA SIP SERVER interface. The dialog contains the following configuration parameters:

```

fullname=1000
registersip=no
hostedynamic
calgroup=1
qualify=yes
mailbox=1000
call-limit=100
type=peer
usertime=1000
transfer=yes
callcounter=yes
context=DLPN_DialPlan1
cid_number=1000
hasvoicemail=no
vmsecret=
    
```

Below the dialog, a table lists several users with their respective configurations:

Extension	CID	Other	Plan	Other	Other	Other	Other	Other	Other
1012	1012	--	Yes	--	DialPlan1	1012	Edit	Delete	
1013	1013	--	Yes	--	DialPlan1	1013	Edit	Delete	
1014	1014	--	Yes	--	DialPlan1	1014	Edit	Delete	
10800	10800	--	Yes	--	DialPlan1	10800	Edit	Delete	

At the bottom of the interface, the following text is visible:

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Figure 7 User Details Screen

5 RING GROUP

It is used to make a certain ring group or a ringing order between the users created by you.

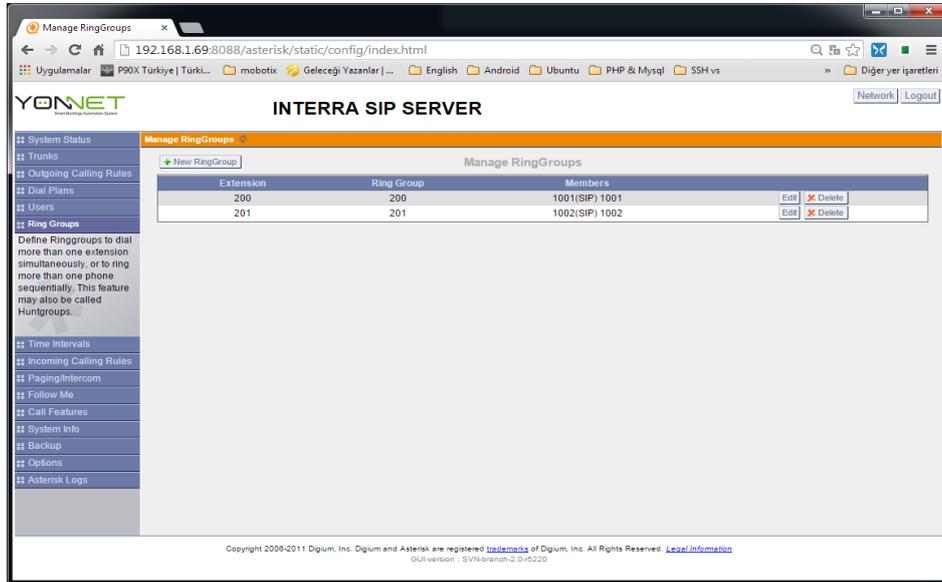


Figure 8 Ring Group Creation

You will be requested to enter Ring Group Name and user id of this group when you create a new group here. Users assigned to Ring Group Members by you shall be included in the group.

Strategy: You can determine the users to be rung in turn or simultaneously.

Seconds to ring each member: You can determine the duration to ring users in seconds.

If not answered Goto: You can determine the next action if not answered

It will be enough to ring group user ID that you gave from the user (IP Phone, Intercom) which will ring later.

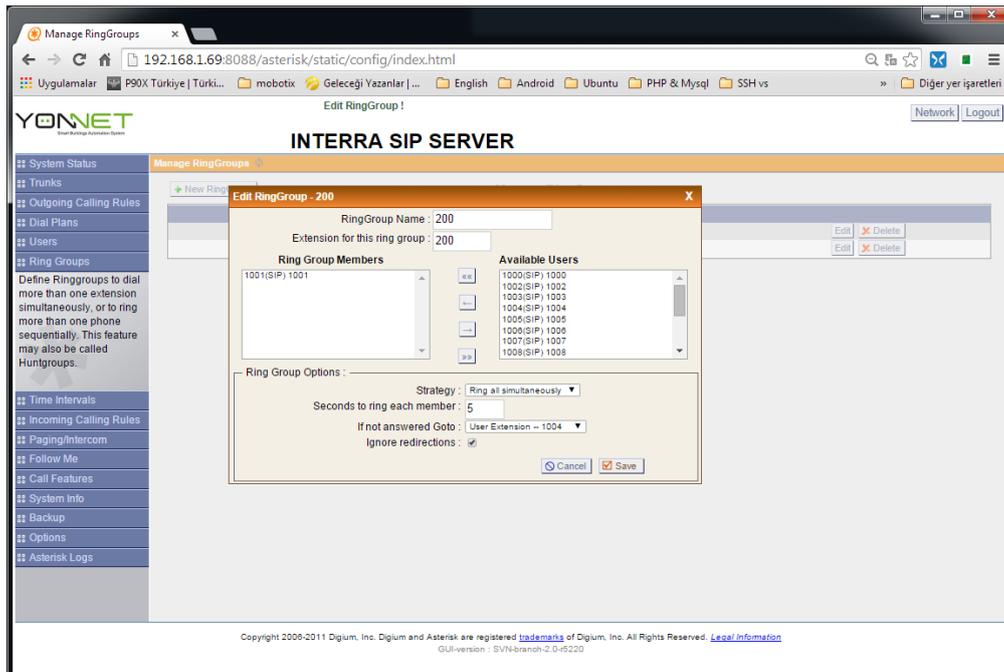


Figure 9 Ring Group Settings

6 DIAL PLAN

Dial Plan allows you to perform permission and authorization transactions among users.

The screenshot displays the 'Manage DialPlans' interface of the YONNET Interra SIP Server. The browser address bar shows the URL '192.168.1.69:8088/asterisk/static/config/index.html'. The page title is 'INTERRA SIP SERVER'. A sidebar on the left contains navigation links for various system components. The main content area features a 'Manage DialPlans' section with a 'New DialPlan' button and a descriptive text box. Below this is a table with columns for 'Default', 'Dial Plan', 'Calling Rules', and 'Options'.

Default	Dial Plan	Calling Rules	Options
<input checked="" type="checkbox"/>	DialPlan1	default, parkedcalls, conferences, ringgroups, voicemenus, queues, voicemailgroups, directory, pagegroups, page_an_extension	Edit Delete
<input type="checkbox"/>	DialPlan2	default, parkedcalls, conferences, ringgroups, voicemenus, queues, voicemailgroups, directory, pagegroups, page_an_extension	Edit Delete

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Figure 10 Dial Plan

7 COMMAND SCREEN

You can see user IDs, IP addresses and their registration status of users which have been registered on sip server, by giving commands on command screen. You should follow **Options -> Advanced Options -> Show Advanced Options** order in order to be able to open command screen. Afterwards, **Asterisk CLI** menu shall be appeared.

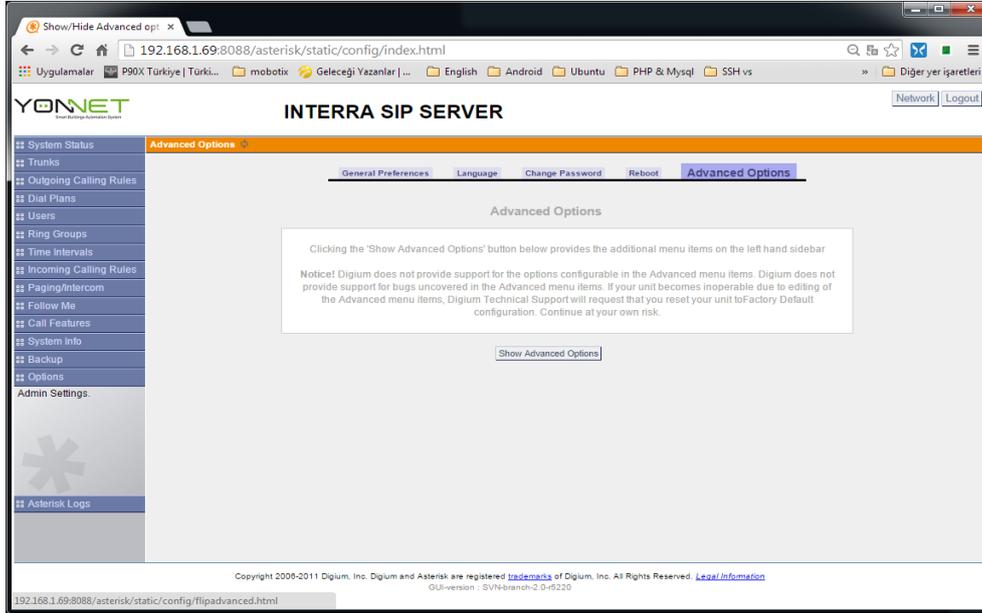


Figure 11 Command Screen

Command screen shall request a command from you.

Sip show pers is the command you will need most. You can follow the status of registered users by this command.

YONET **INTERRA SIP SERVER** Network Logout

Asterisk CLI Emulator x
192.168.1.69:8088/asterisk/static/config/index.html

Uygulamalar P90X Türkiye | Türki... mobotix Geleceği Yazanlar | ... English Android Ubuntu PHP & Mysql SSH vs 3S Surveillance IP Ca... indir » Diğer yer işaretleri

Asterisk CLI sip show peers

Command-sip show peers

Name/username	Host	Dyn	Forcerport	ACL	Port	Status
1000/1000	192.168.1.202	D	N		5060	OK (7 ms)
1001/1001	192.168.1.65	D	N		32945	OK (1 ms)
1002/1002	192.168.1.119	D	N		47652	OK (1 ms)
1003/1003	(Unspecified)	D	N		0	UNKNOWN
1004/1004	(Unspecified)	D	N		0	UNKNOWN
1005/1005	192.168.1.207	D	N		45599	OK (17 ms)
1006/1006	192.168.1.138	D	N		43505	OK (321 ms)
1007/1007	192.168.1.140	D	N		50664	OK (2 ms)
1008/1008	192.168.1.138	D	N		41363	OK (14 ms)
1009/1009	(Unspecified)	D	N		0	UNKNOWN
1010/1010	192.168.1.67	D	N		41512	OK (34 ms)
1011/1011	(Unspecified)	D	N		0	UNKNOWN
1012/1012	192.168.1.171	D	N		43458	OK (67 ms)
1013/1013	(Unspecified)	D	N		0	UNKNOWN
1014/1014	(Unspecified)	D	N		0	UNKNOWN
10800/10800	(Unspecified)	D	N		0	UNKNOWN

16 sip peers [Monitored: 9 online, 7 offline Unmonitored: 0 online, 0 offline]

Call Detail Records [beta](#)

Active Channels [beta](#)

Bulk Add [beta](#)

File Editor

Asterisk CLI

Asterisk Command Line Interface

IAX Settings

SIP Settings

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Figure 12 Command Entry