

Configuration Guide for Yealink IP Phone & Datus PBX

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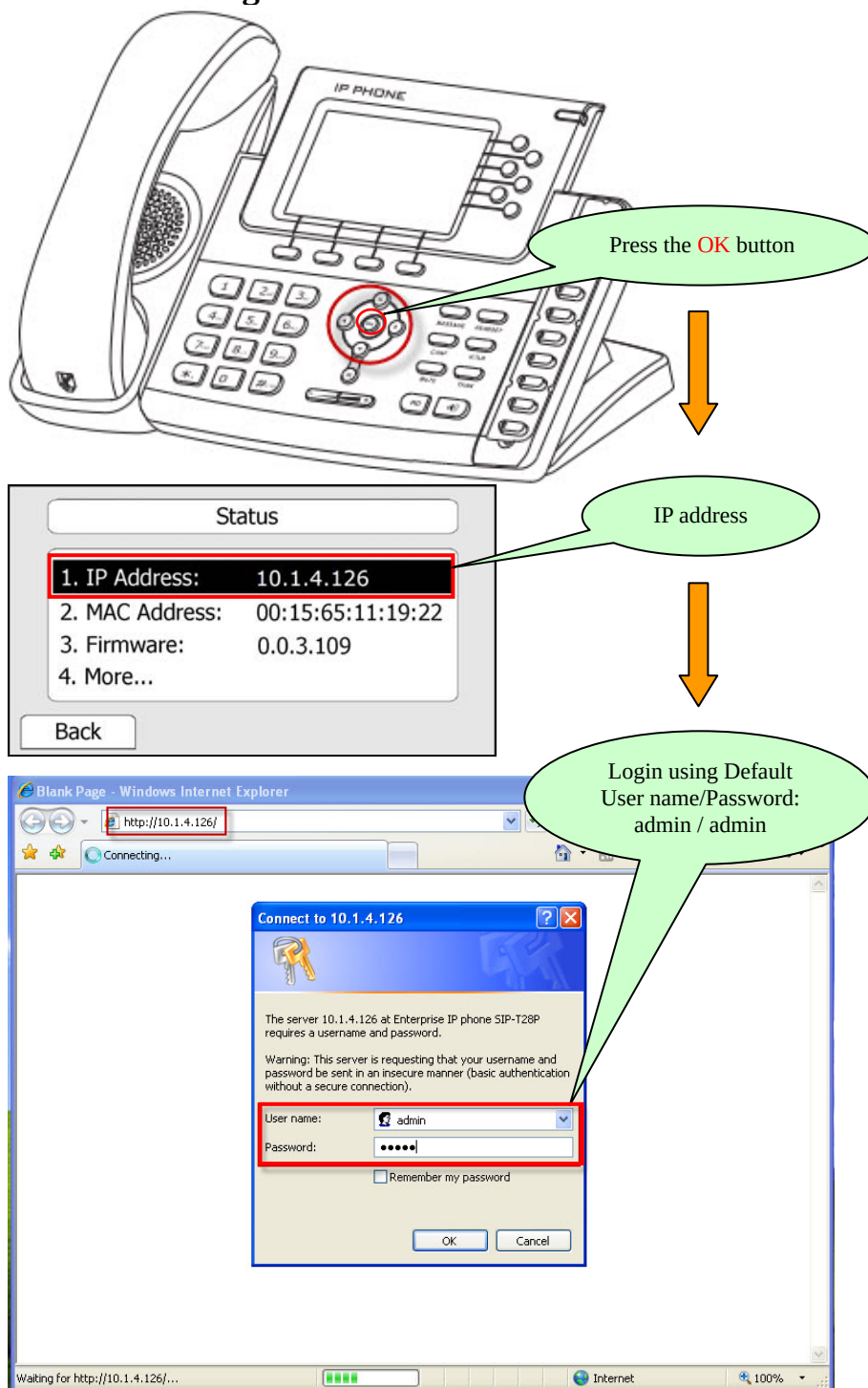
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1. Requirements and Preparation

- Software image v15-10-01-B07 or higher running on the epcx86.
- Software image x.42.x.x or higher running on the Yealink IP Phone.

2. Configure Yealink IP Phone

2.1 Login to WEB management



2.2 Configure the Account Settings

① Select "Account"

② Select one Account

③ Active Account

④ Fill in these fields

⑤ Voicemail code

check Note

Account Account 1

Basic >>

Register Status	Registered
Account Active	☒ On ☐ Off
Label	100
Display Name	100
Register Name	100
User Name	100
Password	***
SIP Server	192.168.1.199 Port:5060
Enable Outbound Proxy Server	Disabled
Outbound Proxy Server	Port:5060
Transport	UDP
Backup Outbound Proxy Server	Port:5060
NAT Traversal	Disabled
STUN Server	Port:3478
Voice Mail	*97
Proxy Require	
Anonymous Call	Off
On Code	
Off Code	
Anonymous Call Rejection	Off
On Code	
Off Code	
Missed call log	Enabled
Auto Answer	Disabled

Register Name
SIP service subscriber's ID used for authentication.

User Name
User account, provided by VoIP service provider.

Proxy Require
A special parameter just for Nortel server. If you log into Nortel server, the value should be: cisco.nortelnetworks.firewall

Codes
Select the codecs you want to use.

Codecs >>

Advanced >>

Confirm Cancel

After entering the above settings, Account1 will be available to make calls.

Note : If the SIP server is behind a NAT, you should enable "NAT Traversal" as "STUN" and then specify a STUN Server. For more details about STUN, please refer to

<http://www.voip-info.org/wiki/view/STUN>. To learn about NAT, you should refer to

<http://www.voip-info.org/wiki/view/NAT+and+VOIP>

2.3 Configure the DSS Key as BLF

① Select "Phone"

② Select "DSS Key"

③ Select "BLF"

④ Select the right line

⑤ Extension number

⑥ Pickup code

Key	Type	Mode	Line	Expansion	Pickup Number
DSS Key 1	BLF	Conference	Line 1	101	*66
DSS Key 2	N/A	Call Park	Line		
DSS Key 3	N/A	Conference			
DSS Key 4	N/A	Conference			
DSS Key 5	N/A	Conference			
DSS Key 6	N/A	Conference	Line 1		
DSS Key 7	N/A	Conference	Line 1		
DSS Key 8	N/A	Conference	Line 1		
DSS Key 9	N/A	Conference	Line 1		
DSS Key 10	N/A	Conference	Line 1	*9315	

NOTE

Key Type
The free function key 'Types' Speed Dial, BLF, Key Event, Intercom, URL.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

URL
This key function allows you to send HTTP requests to a web server.

Confirm Cancel

After entering the above settings, DSS Key1 is ready as BLF for Line 1, monitoring Extension 101. When there's an incoming call on 101, DSS Key1 will flash red and you can directly press it to pick up the call.

2.4 Configure the DSS Key as Call Park

② Select "DSS Key"

① Select "Phone"

③ Select "KeyEvent"

④ Select "Call Park"

⑤ Select the right line

⑥ Park code + parking orbit

Key	Type	Mode	Line	Expansion	Pickup Number
DSS Key 1	BLF	Conference	Line 1	101	*66
DSS Key 2	KeyEvent	Call Park	Line 1	8300	
DSS Key 3	N/A	Conference	Line 1		
DSS Key 4	N/A	Conference	Line 1		
DSS Key 5	N/A	Conference	Line 1		
DSS Key 6	N/A	Conference	Line 1		
DSS Key 7	N/A	Conference	Line 1		
DSS Key 8	N/A	Conference	Line 1		
DSS Key 9	N/A	Conference	Line 1		
DSS Key 10	N/A	Conference	Line 1	*9315	

NOTE

Key Type
The free function key 'Types' Speed Dial, BLF, Key Event, Intercom, URL.

BLF
The button can be configured Busy Line function with and account. This must be supported by the sip server.

Key Event
Key events are predefined shortcuts to phone and call functions.

Intercom
Enable the 'Intercom' mode and it is useful in an office environment as a quick access to connect to the operator or the secretary.

URL
This key function allows you to send HTTP requests to a web server.

Confirm Cancel

After entering the above settings, DSS Key2 is ready as Call Park for Line1.

An active call can be parked by pressing DSS Key2 during your conversation.

3. Auto-configure Yealink Phone

Please refer to Yealink Document [Auto Provision Manual](http://www.yealink.com/fae/Auto Provision Manual.rar):
<http://www.yealink.com/fae/Auto Provision Manual.rar>

4. Configure Datus

4.1 Login to Datus

DATUS indali web-portal

Login at 'Yealink&indali-obx'

Extension number: admin

Password: ...

Select language: English

Secure login via SSL

Default Extension number(admin)
Default password(pbx)

4.2 Add a Subscriber

DATUS indali web-portal

Startpage Generate AstConf Logout

Subscriber

Number	Name on phone display	Protocol	Dialtable	PUG	Forw.
205	test-205	SIP (Voice over IP)	Only internal		
206	test-206	SIP (Voice over IP)	Only internal		
207	test-207	SIP (Voice over IP)	Only internal	Pickupgroup 2	
208	test-208	SIP (Voice over IP)	Only internal	Pickupgroup 2	
209	test-209	SIP (Voice over IP)	Only internal	Pickupgroup 2	
210	test-210	SIP (Voice over IP)	Only internal	Pickupgroup 2	
230	wwt	SIP (Voice over IP)	Only internal	Pickupgroup 1	
231	wwt	SIP (Voice over IP)	Only internal	Pickupgroup 1	
232	wwt	SIP (Voice over IP)	Only internal	Pickupgroup 1	

Subscriber Form:

Number*: 201
 Name on phone display*: John
 Description: 201
 Activate Subscriber's WEB-access: ☒
 Display in presence monitor: ☐
 Refresh time for presence monitor: deactivate
 Protocol: SIP (Voice over IP)
 DTE Service class: Voice
 SIP-Password:
 SIP DTMF: RFC 2833
 Outbound Proxy:
 Dialtable: Only internal
 Pickupgroup: Pickupgroup 1
 Email address:
 Voicemailbox PIN:
 Voicemail notification: Email + Att. + Delete
 Call recording: ☒
 CI IR: ☐
 Busy on busy: ☐
 NAT to subscriber: ☐
 Check SIP availability: ☒
 Automatic call-back: ☐
 Forwarding approved: ☒
 Mobile Callnumber:
 CTI username:
 CTI password:
 (* - Value is imperative)

Callouts:

- Select Subscriber
- Click Insert
- Enter Number Display Name
- Enter SIP-Password
- Select a Pickupgroup
- Click Submit

4.3 Active VoiceMail

The screenshots show the DATUS web portal interface for configuring Active VoiceMail. The interface includes a sidebar with navigation links and a main content area with various settings.

Step 1: Select a Subscriber

The first screenshot shows the 'Settings for F' page. The 'Subscriber' dropdown menu is set to '201 (test-201)'. Below it is a table of forwarding rules.

Type of forwarding	External number	Internal number	VM	Announcement	Active
Call					☒
identification					☒
restricted					☒
Always					☒
Not connected					☒
Busy		202 (test-202)			☒
Not reachable		202 (test-202)			☒

Step 2: Click to Edit

The second screenshot shows the 'Edit' page for the selected subscriber. The 'Destination Type' is set to 'Voicemailbox'. The 'Active' checkbox is checked.

Step 3: Select Destination Type as Voicemailbox

Step 4: Tick Active

Step 5: Click Submit

Appendix

1. Default Basic Dial Code on Datus System

Voice Mail (VMail)	*97
Call Park	8300
Callpark Retrieve	8301

Reference

◆ www.Datus.com
◆ www.yealink.com