

Yealink SIP IP Phones Release Note of Version 70

Firmware Version : From x.61.0.80 To x.70.0.90

Applicable Models : SIP- T20/T22/T26/T28

Release Date : November 29th, 2012

1. New features

1. Added new mechanism of Auto Provisioning feature.
2. Added automatic BLF list configuration feature on DSS key.
3. Added one sharedline account supports multiple calls feature.
4. Added Call Pickup soft key feature.
5. Added “Visual Alert for BLF Pickup” and “Audio Alert for BLF Pickup” configuration.
6. Added the prompt interface for “Action URI” feature.
7. Supported to show the hardware version and firmware version of the EHS and EXP device in the Status page.
8. Added “Play local DTMF Tone” configuration.
9. Added “DHCP Active” configuration.
10. Added “HeadSet Key In Talk” configuration
11. Added “Phone Unlock PIN” on LCD.
12. Added Mac address info to User-Agent header on SIP message.

2. Optimization

1. Supported to enter the Voicemail box by pressing the Message button when then Menu key is locked.
2. Added the “Transfer Mode via DSS Key” configuration.
3. Classified the configurations in the Phone→Features web setting page.
4. Supported DND & FWD separated feature.
5. Optimized keypad lock feature.
6. Supported TAP mode of OpenVPN.
7. Added the OpenVPN feature to T20/T22 models.
8. Modified ATS conference feature to be Optional.
9. Modified Draft BLA feature to be Optional.
10. Added PEAP-MSCHAPV2 authentication for 802.1X feature.
11. Added EAP-TLS authentication for 802.1X feature.
12. Added domain name supported for PushXML server address.
13. Optimized “Logo + character HD” icon display in the LCD of T28/T26 when calling with codec G722.
14. Did optimization of dealing with BLF Notify message which is out of dialog.
15. Did optimization of ringtone.
16. Default Input Method of the Broadsoft network phonebook.
17. Added some fields to the snmp section.
18. Added the feature that the phone can use the SNMP to reboot the phone.
19. Added the dual-headset function.
20. Added new parameters to Action URL feature.
21. Added “Voice Mail Tone “configuration.
22. Optimized XML browse.
23. Added “Allow IP Call “configuration.
24. Did optimization that the phone can disable the local DTMF Tone.
25. Did optimization that the phone can support 99 entries for Dial-now Rule.
26. Did optimization of the LDAP search interface, show the default IME when enter this interface and the IME can be configured.
27. Did optimization of DHCP, supported to get NTP server address through Option 42.
28. Did optimization that the Direct IP Call feature can be configured through Auto Provisioning.

3. Bug Fixes

1. Fixed the issue that after the other party answers the incoming call, the phone still rings the ringback tone.
2. Fixed the issue that the phone can't make or answer calls normally because of the phone can't obtain the right IP address from the SIP INVITE header.
3. Fixed the issue that the phone will prompt "No data to import "when import the *.csv file which exported from the Outlook.
4. Fixed the issue that the phone will have noise /short interruption every 60s during the voice talking.
5. Fixed the issue when Attended Transfer, the phone cancel the call immediately after press OK, the third party will ring, but it can't create the connection after answer the call.
6. Fixed the issue that the account registration will time out when T28/T26 connected to the open VPN.
7. Fixed the issue T28/T26/T22 can't accomplish the Push XML feature when use the TextScreen.php file.
8. Fixed the issue about the Asterisk server: When redial, the phone can't make calls to the account which including the server IP.
9. Fixed the issue when use the STUN feature, the phone always registers with its private IP instead of public WAN IP of Router, which causes no-audio issue.
10. Fixed the issue that VOP (soft phone base on sip presence) can't be compatible with RFC4235.
11. Fixed the issue that when A press Transfer +C number +Transfer, on the LCD of the A, it will show Resume.
12. Fixed the issue that can't hear the local ring back tone when change from speaker to handset.
13. Fixed the issue that the configure file which the phone received from the auto provision server will be damaged when SSL is activated
14. Fixed the issue that when you do some operation, the phone will have slow response to it.
15. Fixed the issue that the number can't be included the underline symbol "_"when added the contacts from the web UI.
16. Fixed the issue that the LDAP can't search the right result.
17. Fixed the issue that the Broadsoft network phonebook can't be compatible with the enterprise contacts phonebook.
18. Fixed the issue that the phone will exit from the dial interface if the phone exits from the pool.
19. Fixed the issue that the phone can't send the URL which contains the username and password.
20. Fixed the issue that BLF can't work normally after reboot.
21. Fixed the issue that the LED of PLANTRONIC CS60 wireless headset can't turn off when the phone hangs up the call.
22. Fixed the issue when the caller cancels the calling, the called party still ringing.
23. Fixed the issue that the phone will not auto answer the call even though it receives the Answer-After=0 message.

24. Fixed no voice issue when using Multicast IP paging feature.
25. Fixed the issue that the phone can't support multicast in VLAN.
26. Fixed the issue that attended transfer failed in a special server.
27. Fixed the issue that the BLF light doesn't turn on after reboot.
28. Fixed the issue that it can't create a new call in the Broadsoft call manager.
29. Fixed the issue that when change SIP port, it is ok to make a call but failed in sending a message.
30. Fixed the issue that it will answer automatically when the second call is an Intercom call when use TR069 feature.
31. Fixed the issue that turn error when change the network type of PC port from 'as bridge' to 'as router'.
32. Fixed the register failed issue when register to a public server.
33. Fixed the issue that when locked all the keys, it can't be unlock when there 's new voice mail .
34. Fixed the issue that after set several BLF DSS Keys, the phone will response very slowly.
35. Fixed the issue that when the language of the phone web interface is Russian, the DSS Key interface is garbled.
36. Fixed the issue that when the language of the phone web interface is English, the Phone-Features interface is garbled
37. Fixed the issue that when disable the Button Sound, there is no dial tone when pick up the handset.
38. Fixed the issue that incoming call can be seen from LCD but without ring tone, also no audio when answered the call.
39. Fixed the issue that in the Broadsoft PBX, when making a call with speaker, there is no more ring back tone after the first ring back tone.
40. Fixed the issue that when button sound set to disable, A call B with speaker, but there's no ring back tone when change from speaker to handset.
41. Fixed the issue that when search in remote phonebook, phone didn't deal with input characters with urlencode before sending URL.
42. Fixed the issue that can't use the X button to hang up a call.
43. Fixed the issue that can't reply correctly to the refer message.
44. Fixed the issue that DNS_SRV can't register with TLS but with UDP instead.
45. Fixed the issue that can't register with TLS after 2~3 minutes of reboot.
46. Fixed the issue that register failed with TLS.
47. Fixed the issue that after registered and first subscribe successfully, after subscribe expire time, the phone will not refresh the subscribe message.
48. Fixed the issue that the BLF light status is not correct when the call is transferred or held.
49. Fixed the issue that the phone didn't use default ringtone when there's no custom ringtone or selected ringtone.
50. Fixed the issue that it will subscribe repeated when using BLF list function.
51. Fixed the issue that the phone can't use the authentication information in refer message.
52. Fixed the issue that can't cancel the subscribe message for BLF.
53. Fixed the issue that there is no subscribe message reply after received the 481 message send from the server.

54. Fixed the issue that in UCAP platform, the phone doesn't reply the corresponding subscribe message after receiving the subscribe authentication information from the UCAP.
55. Fixed the issue that when using the distinctive ringtone, it will crash and reboot automatically after download the custom ringtone failed.
56. Fixed the issue that the refer message is the same between semi-attended transfer and the busy forward.
57. Fixed the issue that the conference function is not available when using a soft phone in PC.
58. Fixed the issue that when using IPv6 address, the phone will not process the 183 Session description message when call mobile phone, and when a call comes in (IPv6 call IPv6), the phone will always send 486 a message to the server.
59. Fixed the issue that phone will sometimes register failed.
60. Fixed the issue that the phone will reboot after search the WINDOW AD LDAP phonebook for several times.
61. Fixed the issue that the phone will response very slowly and even crashed when using LDAP phonebook.
62. Fixed the issue that when A is talking with B with speaker and then C calls A, at this time, a serious echo will be heard on B side.
63. Fixed the issue that when answer to the second call, the phone will answer second call first and then hold the previous call.
64. Fixed the issue that the mechanism changed when received a reboot message from Broadsoft.
65. Fixed the issue that it will dial out name but not the correct number when press the BLF list button.
66. Fixed the issue that when registered with TLS and set the DSS Key as BLF list, after that the DSS Key can't dial out.
67. Fixed the issue that it becomes mute automatically when answer a call, and then shows busy when dialing out a second call.
68. Fixed the transfer failed issue on Broadsoft Call Manager.
69. Fixed the issue that it cannot import the contact groups when import contacts with CSV format.
70. Fixed the issue that the phone can't compatible with "88.lb4.keyyo.net" format in Outbound server address.
71. Fixed the issue that when use DNS SRV in the Avaya SCS server, it will sometimes shows "NO SERVICE" on the LCD.
72. Fixed the issue that when using 3CX 10 server, the local extension can work normally, but the remote extension can't work normally.
73. Fixed the issue that it will reply a 200 ok message when REINVITE failed.
74. Fixed the issue that it will auto provision failed when there is a sipAccount.cfg exists.
75. Fixed the resume failed issue in the Mitel serve.
76. Fixed the unable to get IP from the network issue.
77. Fixed the issue that the BLF works not stable, it works fine sometimes but sometimes doesn't work.
78. Fixed the issue that the phone will stop sending Keep Alive message when using DNS-SRV feature.
79. Fixed the LDAP search failed issue.

80. Fixed the issue that the phone received the wrong via header of invite message from some server, which cause replying to the wrong address.
81. Fixed the issue that the BLF light sometimes can't turn on after phone reboot.
82. Fixed the issue that the phone sends the incorrect ACK (sdp) message which cause the one-way audio call.
83. Fixed the issue that it can't show the caller ID after pick up.
84. Fixed the issue that the BLF light can't turn on in EP39.
85. Fixed the issue that BLF LIST can't show the name and the light state can't change.
86. Fixed the transfer fail issue.
87. Fixed the issue that the auto provision can only success at the first time.
88. Fixed the issue that the phone response slowly after connected with an EXP38.
89. Fixed the LDAP display unusually issue.
90. Fixed the upload certification failed issue.
91. Fixed the issue that the default ring tone is common ringtone, and it can't be changed to others.
92. Fixed the issue that it will hang up automatically after hold a call for 2 minutes.
93. Fixed the issue that it will response very slowly after a moment and
94. Fixed the issue that it will register fail after reboot on the HUAWEI SX3000 server.
95. Fixed the issue that make a call in the soft phone CTI, and there is no music ringtone and not available notify message.
96. Fixed the delay issue of LCD display when dialing.
97. Fixed the issue that it can't work when Action URL is using HTTPS.
98. Fixed the issue that A call B, before B answer, B forward the call to C, B can't hear the ringback of C.
99. Fixed the issue that when doing attended transfer, press ok and then hang up very fast, the third party can ring but when it pick up the phone ,it can't set up the connection.
100. Fixed the issue that the Broadsoft server can't recognize the auto provision parameter '+'.
101. Fixed the issue that there will be mixed noise/short interruption every 60 second during the voice call.
102. Fixed the ICMP issue.
103. Fixed the issue that when login in ACD and input the wrong user name and password, it can't popup the error message.
104. Fixed the issue that under the UC system, it will hang up automatically after the first ring.
105. Fixed the issue that when IP PBX SIP-INVITE contained the message of AUTOANSWER, the SPEAKER and HEADSET can't ring at the same time.
106. Fixed the issue that it will not send the Invite message if DND on code and off code are set as the same.
107. Fixed the issue that when the host of the conference hang up, then the other two parties can't go on the conference.
108. Fixed the issue that when press redial button to redial a number which needed prefixed digit, it will not add prefixed digit automatically.
109. Fixed the issue that DND/Forward can't be synchronization with Huawei server.

110. Fixed the issue that A call B, B transfer the call to C, at this time, there are two incoming calls on C.
111. Fixed the issue that Chile daylight-saving time is incorrect.
112. Fixed the issue that it can't hang up an incoming call with the handset.
113. Fixed the issue that SCA is set on line key, the phone will not display the icon of line key when synchronization.
114. Fixed the issue that transplant the last RPS firmware to the customer specific version.
115. Fixed the issue that it can't answer the call when pick up the handset very fast after the first ring.
116. Fixed the issue that it can't answer the calls successfully using the Active URI feature.
117. Fixed the issue that it can't register the account which contains space through auto provision.
118. Fixed the issue that the supported password digits are too short.
119. Fixed the issue that when using BLF, if several calls come to the monitored number, the BLF light status is not correct.
120. Fixed the issue that it can't return to the idle interface after the semi-attended transfer.
121. Fixed the issue that when press Reject, the phone will send the BYE message first and then the 486 message.
122. Fixed the issue that the phone gets the incorrect value of option43 when refresh DHCP.
123. Fixed the issue that it will response a 486 message to hang up a call when received an invite message with replace information.
124. Fixed the no audio issue after pick up the call.
125. Fixed the issue that it can't adjust the volume to mute.
126. Fixed the issue that it can't support the value of outbound server address which started with a digit.
127. Fixed the issue that there is no audio during a call in the Aastra MX-ONE Telephony Server
128. Fixed the issue that the password and AES key can be seen in the syslog when doing auto provision.
129. Fixed the issue that if the phone filled AES key, the phone still download file without encryption from auto provision server.
130. Fixed the issue that the phone can't support the 'alert-info: info=alert-autoanswer' header.
131. Fixed the issue that the phone can't refresh the LCD display according to the SIP INFO.
132. Fixed the issue that when enable call waiting tone, there is a second call during a call, the volume will increase instantly.
133. Fixed the issue that when the Intercom Barge is set to disable, during a normal call, when there is an intercom call coming, the normal call will be held, and response to the intercom call.
134. Fixed the issue that when the phone started for a while, it will stop to send the UDP keep alive message to the server which will make the incoming call unavailable.
135. Fixed the issue that when delete all the accounts in the BLF list, and phone received the message from the server, the BLF list in the webpage will not disappear.
136. Fixed the issue that when reset the auto provision password, the new password will not take effect, but the admin password is OK.
137. Fixed the issue that the when ALLKEY is locked, if there is a voicemail, there is no MENU key so that it can't unlock the keys.

138. Fixed the issue that when a phone start a conference, and the host hang up, then another party will also hang up, just the last party still on the call.
139. Fixed the issue that it can't lock the Headset button during a call.
140. Fixed the issue that if enable the RFC2543 Hold, when held, the call will send a reconvl which contain the IP 0.0.0.0, and this will cause the server can't send the held music to the phone.
141. Fixed the issue that when mute during a call, it will also hear the DTMF Tone behind the caller on the other side of the call.
142. Fixed the issue that the phone import the VPN certificate which needed a password, then the phone will stay in the 'Initializing, Please Wait' interface when rebooted.
143. Fixed the issue that the phone incorrect analysis the cancel message sent by the server in TCP, which will cause the phone always ringing.
144. Fixed the issue that when dial out the third party with the DSS Key set as speed dial, if the third party answer the call and then hang up, the phone can't get back the original call by press the resume key.
145. Fixed the issue that in some server, when transferring, the phone will hold the call first and then transfer, which will make another side of the call to hear a short music on hold tone.
146. Fixed the issue that the phone in the announce group can't answer automatically when received an announce call.
147. Fixed the issue that the light status is not correct when the Music on hold and BLF function enabled at the same time.
148. Fixed the issue that the phone will stop sending RTP package sometimes when there is an incoming call.
149. Fixed the issue that when download the Broadsoft Phonebook error, it will prompt to change password message.
150. Fixed the issue that it can't support the character “#” in the auto provision URL.
151. Fixed the issue that there is no ring back tone when call out and received the 183 and 180 message.
152. Fixed the issue that it is not available to configure the DNS SRV type.
153. Fixed the issue that after the configuration of SCA, when press the CONF button during a call, it will show “shared line not available” on LCD.
154. Fixed the issue that when the phone subscribe the BLF list, log out and then subscribe again, the server will response a 403 message which will cause the BLF light status not correct.
155. Fixed the issue that there is no audio during the call when open the SonicWall fire wall.
156. Fixed the issue that the SMS can't send correctly because of the MESSAGE sending port is not correct.
157. Fixed the issue that the call hang up because of the ACK sending port is not correct.
158. Fixed the MWI subscribe failed issue.
159. Fixed the issue that paging function can't compatible with the Cisco PBX.
160. Fixed the issue that the after Transfer, the phone will send the 302 moved temporarily message continually.
161. Fixed the issue that when using BLF, the phone reply a 481 message to the notify message from other caller ID, which cause the server never reply a notify message to the phone.

162. Fixed the issue that the phone will not send the subscribe message to subscribe the BLF list after the phone launched.
163. Fixed the issue that when set a DSS Key as BLF, add and delete the BLF again and again, it will not subscribe successfully sometimes, and the BLF light will not turn on.
164. Fixed the issue that it will not subscribe again when there is no reply after the first subscribe message of BLF LIST.
165. Fixed the issue that it can't pick up the incoming anonymous call.
166. Fixed the issue that if the phone received the replace message which has changed call ID in the ring back status, it will not finish the call and there is no audio at the same time.
167. Fixed the issue that after the configuration of BLF LIST, when you reregister or log out an account, it will cause the memory leak.
168. Fixed the issue that when connect an EXP39 and EHS (without wireless headset), the label of some parameters like speed dial can't edit.
169. Fixed the issue that in the Broadsoft Platform, when using the HUNT GROUP function and blind transferred to that same GROUP, the phone will crash.
170. Fixed the issue that it can't transfer successfully in the ASTERISK server.
171. Fixed the issue that Option reply the 200OK message to the wrong IP address.
172. Fixed the issue that the phone will create a new call when received the same parameters for the 183 music ring server.
173. Fixed the issue that when auto provision with RPS, the phone can't access to FTP server with defined user name and password, but with anonymous user name.
174. Fixed the issue that there is no isfocus in the 200OK message received by the phone, which will cause send refer message failed.
175. Fixed the memory leak issue when auto provision with https server.
176. Fixed the issue that it can't set the T1 time as decimal point when doing the configuration of SIP Session Timer(seconds).
177. Fixed the issue that when BLF need to subscribe again, the Server will reply 401 or 407 message which will make the BLF light status not correct.
178. Fixed the issue that it can't hide the information which is related to the Black list.
179. Fixed the issue that there is no ringtone before the phone answered automatically when received the 'sip:answer-after' message.
180. Fixed the issue that the phone will not reply to the refer message correctly.
181. Fixed the issue that the BLF light status will not correct because the phone received the notify message that contain several status but phone can't recognize the correct priority.
182. Fixed the issue that when two phones set the same BLF number, the status of BLF light will not be the same at the same time.
183. Fixed the issue that there is no audio during the call after pick up successfully.
184. Fixed the issue that the phone can't reboot after received the reboot message from Broadsoft Server.
185. Fixed the issue that when using the 3CX server, it will still show busy status of the number in the Server when it has finished the attended transfer and hang up.
186. Fixed call back failed issue when the number including server IP.
187. Fixed the issue that the phone doesn't verify the character format in the webpage.

188. Fixed the issue that when use DNS SRV, it can't call out because the phone received the OPTION message but analysis failed.
189. Fixed the issue that it can't support the character “#” in the remote phonebook URL.
190. Fixed the issue that it can't login Hot Desking account successfully if the password length is longer than 3 digits.
191. Fixed the issue that in a 3-way conference, the other two parts can't hear the voice of the people who start the conference.
192. Fixed the issue that when the phone is ringing, it can't answer the call through the answer key and Speaker button.
193. Fixed the issue that when auto provision with PNP, the phone will send the HTTP message twice.
194. Fixed the issue that when hang up a call with the EHS36, but it is still connected in headset.
195. Fixed the issue that when dialing, there is no ringtone because the codec difference and no analysis for the DSP.
196. Fixed the issue that it needs to configure the line key as a SCA after you have done the SCA configuration on Account-Advanced interface.
197. Fixed the issue that when picked up the BLF monitored number, it can't blind transfer to the same monitored number again.
198. Fixed the issue that when searching LDAP for several times, it will cause the memory leak.
199. Fixed the issue that A (open SRTP) call a GRANDSTREAM phone (SRTP closed), GRANDSTREAM phone answer and then hang up the call, at this time, A will crashed.
200. Fixed the issue that when the phone is using the external IP but the PBX is using the internal IP, then make a call, the call will lose one-way audio.
201. Fixed the issue that it will not notice the network change and hang up the call when unplug the network label.
202. Fixed the issue that the LDAP search delay time is not effect.
203. Fixed the issue that the parameters which contain the character '@' will not take effect when doing auto provisioning.
204. Fixed the issue that the current call will randomly lose audio when there is a second call comes in and goes to call waiting.
205. Fixed the issue that the audio will delay if action-URL is used to off hook the phone before answering the call.
206. Fixed the issue that the exact password of HTTPS server can be seen in the syslog exported from the Phone.
207. Fixed the issue that when using the wireless headset to answer the first call and hang up with handset, the answer key of the headset needs to be pressed twice in order to answer the second call.
208. Fixed the issue that when T26 and T28 are close to each other, the DTMF sound will have interference on each other.
209. Fixed the issue that it will not do the validity inspection of the input, in the DSSKYE/Dial Plan/ Local Directory/ Blacklist/Broadsoft/Calllog input field.
210. Fixed the issue that the first call will randomly lose audio when there is a second call comes in.
211. Fixed the issue that the phone will repeatedly reboot when using PoE.

- 212. Fixed the issue that there is no ringback when the phone received the 180 message from Broadsoft server.
- 213. Fixed the issue that when the login expire is set to 3600, the phone will not re-register after 3600 seconds.
- 214. Fixed the issue that when use OpenVPN, the IP of DNS will change, at this time, phone will not send the request to the correct DNS.
- 215. Fixed the pickup fail issue when fill in the SIP server as domain name.
- 216. Fixed the issue that it is unavailable to fill in the value of label for BLF in the web page of EXP39, when connected EXP39 and EHS36 at the same time.
- 217. Fixed the issue that it needs to press Hold key twice in order to resume the held call.
- 218. Fixed the issue that the port in the contact domain of SIP message is not correct when using TLS.
- 219. Fixed the issue that the auto provisioning failed because the incorrect way to deal with the character '% '.
- 220. Fixed the issue that the AES Key, trusted certificates and server certificates are all in the syslog exported.
- 221. Fixed the issue that the Send Pound Key doesn't make effect when it is set to enable.
- 222. Fixed the issue that the PhoneLock password set on LCD doesn't take effect.
- 223. Fixed the issue that the volume is not stable when adjust it with Speaker.
- 224. Fixed the issue that when set the dial plan, it will add the prefix twice when dialing from the call log.
- 225. Fixed nine spelling mistakes of Time zone.

4. Description to the new added features

1. Added new mechanism of Auto Provisioning feature.

Detail : For more details please refer to Yealink Auto Provision User Guide.pdf

2. Added automatic BLF list configuration feature on DSS key.

Description : After configuring BLF list URI on one account, corresponding BLF list will be configured automatically on available DSS key in order.

Detail: For more details please refer to Yealink SIP-T2xP User Guide.pdf.

3. Added one shared line account supports multiple calls feature.

Path: Account -> Advanced page.

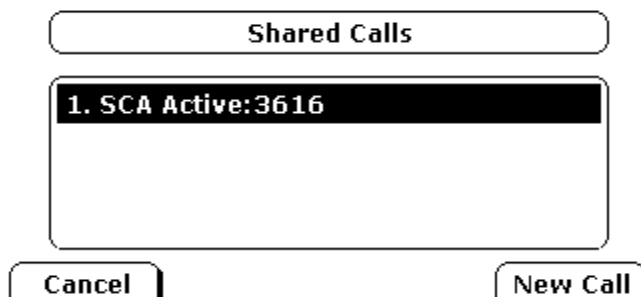
BLF List Code	<input type="text"/>	?
BLFListBargeInCode	<input type="text"/>	?
Shared Line	Broadsoft SCA	?
Dialog-Info Call Pickup	Disabled	?
BLA Number	<input type="text"/>	?

Description : There is no need to do the configuration in the DSS Key interface, it just need to select the Broadsoft SCA option in the drop-down menu of shared line parameter.

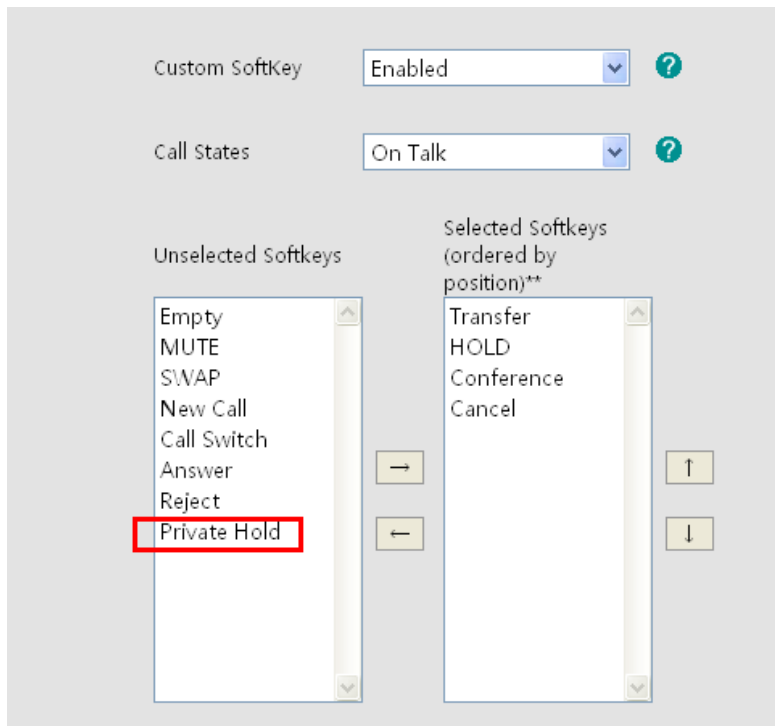
There is difference between the symbol of SCA account and normal account on the idle interface.



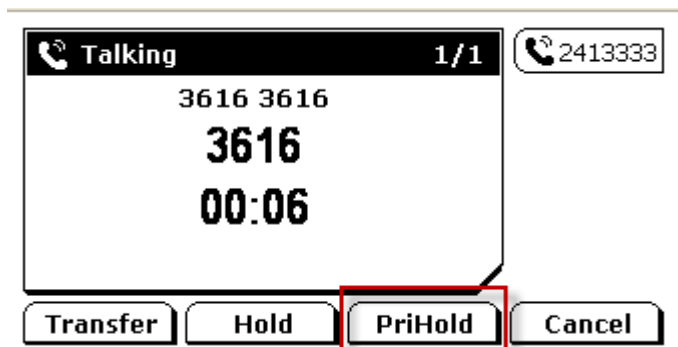
Long press the LineKey/DssKey for 1s to show the call list of the SCA account.



Add the PriHold soft key on the talking interface of the SCA account.



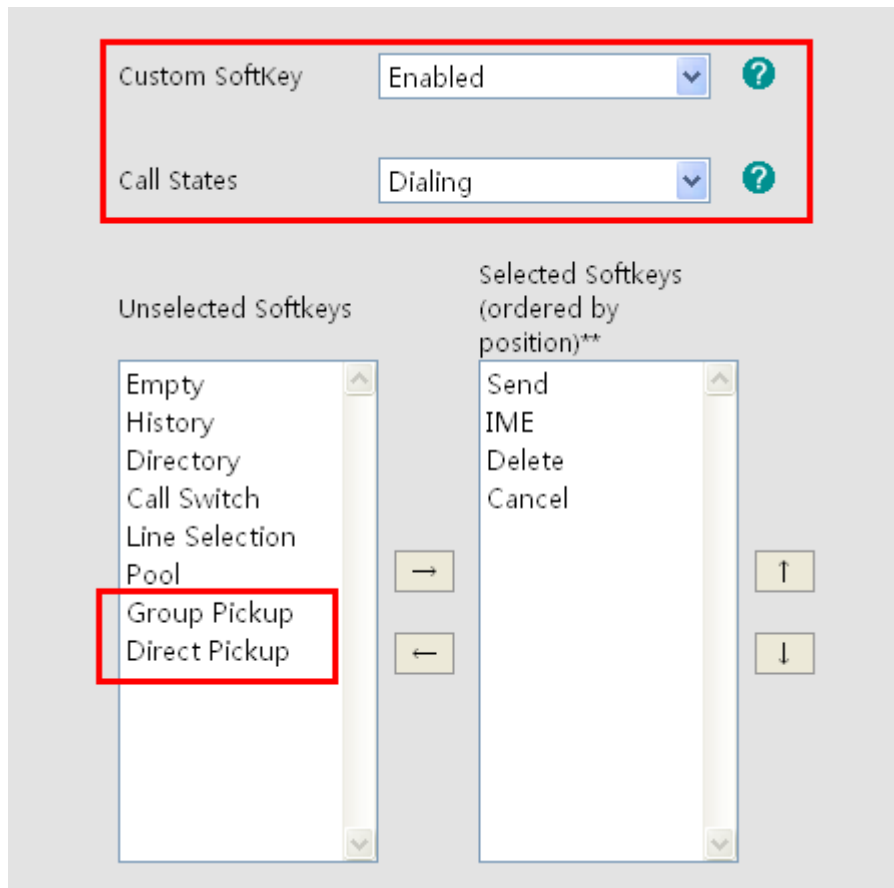
The display of the PriHold soft key on the talking interface of SCA account :



Detail : For more details please refer to the introduction of SCA in Yealink SIP-T2xP User Guide.pdf.

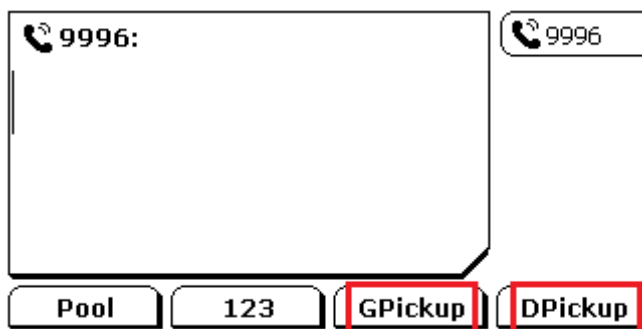
4. Added Call Pickup soft key feature

Path : Phone→Softkey Layout



Description : Support the customer set soft key as all Pickup in the soft key layout of dialing interface.

The display of Call Pickup soft key on LCD :



Detail :For more details please refer to the introduction of Call Pick in Yealink SIP-T2xP User Guide.pdf

5. Added “Visual Alert for BLF Pickup” and “Audio Alert for BLF Pickup” configuration.

Pate: Phone→Feature→Call Pickup

Call Pickup >>

Direct Call Pickup	Enabled
Direct Call Pickup Code	
Group Call Pickup	Enabled
Group Call Pickup Code	
Visual Alert for BLF Pickup	Disabled
Audio Alert for BLF Pickup	Disabled

Description :

Visual for “BLF +Pick code “: when set the “BLF +Pick code “, when a call comes to the monitored number, a Visual will show on LCD as below

Calls for Pickup

1.2536 <- 2534

DPickup Dial New Call Cancel

BLF pickup tone: when set the ‘BLF + pickup code’, it will have an audio like ‘du’ when a call comes to the monitored number. The default value of this parameter is ‘disable’.

Detail: For more details, please refer to the introduction of Busy Lamp Field (BLF) in Yealink SIP-T2xP User Guide.pdf.

6. Added the prompt interface for “Action URI” feature

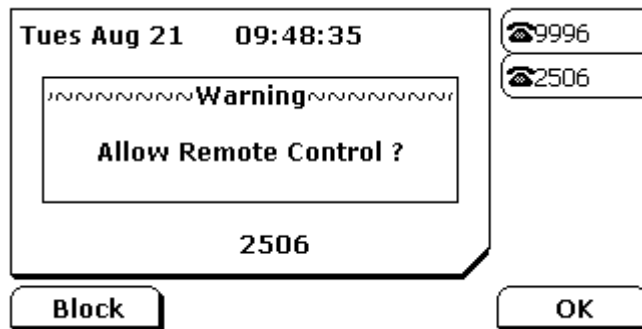
Path : Phone→Feature→API Security→Trusted Action URI Server List

API Security >>

Push XML Server IP		?
XML SIP Notify	Disabled	
Trusted Action URI Server List	10.3.5.47	

Description : Trusted Action URI Server List, When you active the Action URI function(fill in a IP address), if the phone received a remote control command of Action URI , it will pop up a prompt interface first, and customers can choose to accept or reject it(the similar scenario like remote assistance).

The prompt interface on LCD:



The format of the IP address :

- 1) Support one or more Discontinuous IP address, separated with a comma.
For example, when you fill in the IP like “10.1.3.66,192.168.1.20”, at this time, only the person using the IP address of 10.1.3.66 or 192.168.1.20 can remote control the phone with Active URI.
- 2) Support a continuous IP section, use the format like ‘*. *.*.*.*’ to design the IP section that can access to the phone. The range of ‘*’ is 0~255
For example
10.1.3.* means the IP section of 10.1.3.0 ~ 10.1.3.255

10.1.*.* means the IP section of 10.1.0.0 ~ 10.1.255.255
- 3) The default value of Trusted Action URI Server List is empty, that means prohibit all the remote control of the phone.

7. Supported to show the hardware version and firmware version of the EHS and EXP device in the Status page.

Path: Status →EXP (it will display only when the EHS or EXP device is connected)

EXP ?	
EXP39 1 Hard Version	32.0.0.0
EXP39 1 Soft Version	5.16.0.0

Description :

When EXP38/39 or EHS36 is connected, phone will show the hardware version and firmware version of them in the status interface of the phone web GUI.

When EXP38/39 or EHS36 is disconnected, phone will refresh the phone web GUI, the information of the EXP and EHS will disappear.

Firmware information :

Type	Hardware Version	Old Firmware Version	Newest Firmware Version
EHS36	48.0.0.0	0.16.0.0	1.16.0.0

EXP39	32.0.0.0	4.16.0.0	5.16.0.0
EXP38	16.0.0.0	4.17.0.0	5.17.0.0

8. Added “Play local DTMF Tone” configuration

Path : Phone→Feature→Audio Settings

Audio Settings >>

Call Waiting Tone	Enabled	?
Play Hold Tone	Enabled	
Play Hold Tone Delay	30	
Key Tone	Enabled	?
Send Tone	Enabled	?
Redial Tone		?
Headset Send Volume (1~53)	30	?
Handset Send Volume (1~53)	25	
Handfree Send Volume (1~53)	35	
Ringer Device for Headset	Use Speaker	?
Dual Headset	Disabled	
Headset Prior	Disabled	
Send Pound Key	Disabled	
Play Local DTMF Tone	Enabled	

Description :

When set to Enabled, the user can hear the DTMF Tone when pressing the DTMF during a call, when set to Disabled, the user can't hear the DTMF Tone when pressing the DTMF during a call.

9. Added “DHCP Active” configuration.

Path : Upgrade→Advanced

The screenshot shows the 'Network' configuration page in the Yealink web interface. The 'DHCP Active' dropdown menu is highlighted with a red box and is set to 'Enabled'. Other visible settings include 'Custom Option(128 ~ 254)', 'Custom Option Type' (String), 'DHCP Option 60' (yealink), and 'Provisioning Server' (http://10.3.5.47:8080/remote). A 'NOTE' section on the right indicates that custom options can be specified for provisioning.

Description :

If this option is enabled, it means phone can get the Auto Provisioning URL by DHCP Option value .If it is disabled, phone can't get the Auto Provisioning URL by DHCP Option value.

10. Added “HeadSet Key In Talk” configuration

Path : Phone->Features->General

The screenshot shows the 'HeadSet Key In Talk' configuration page. The 'HeadSet Key In Talk' dropdown menu is highlighted with a red box and is set to 'Enabled'. Other visible settings include 'Login Timeout (minutes)' (1000), 'BLF Notify Type' (Disabled), 'Allow Trans Exist Call' (Enabled), 'Voice Mail Tone' (Enabled), 'Allow IP Call' (Enabled), 'Call Number Filter' (,-), 'Use Logo' (Custom Logo), and 'Upload Logo (The pixel < 236*82)' (with a '浏览...' button). A 'Delete' button is also present.

Description :

When set HeadSet Key In Talk to Disable,
If phone is in HeadSet mode, at this time there will be no response when the user press the Speaker or pick up the handset.
If phone is not in HeadSet mode, then the user press the Headset button to enter the HeadSet mode, at this time, there will be no response when the user press the Speaker or pick up the handset.
When set HeadSet Key in Talk to enabled, the phone will work normally.

11. Added “Phone Unlock PIN” on LCD.

Path : Menu->Setting->Basic Setting->Phone Unlock PIN

Basic Settings

1. Language

2. Time & Date

3. Ring Tones

4. Phone Unlock PIN

Back

Enter

Change PIN

1. Current PIN:

2. New PIN:

3. Confirm PIN:

Back

123

Del

Save

Description : it can be used to configure the unlock PIN on LCD. And the default value of the PIN is '123'.

Detail : For more details, please refer to the introduction of the Phone unlock PIN in Yealink SIP-T2xP User Guide.pdf

12. Added Mac address info to User-Agent header on SIP message.

Description :

This feature is configurable only by Auto Provisioning, it is defaulted to be disable.

Enable this Feature:

Set 'network.sip.tag_mac_to_ua.enable = 1' in your configure files.

And the MAC address will show in the User-Agent header.

Capturing from Atheros L1C PCI-E Ethernet Controller (Device\NPF_{A3F963AF-C134-46EB-95CC-C575F8E05B0}) [Wireshark 1.8.2 (SVN Rev 44520 from /trunk-1.8)]

Filter: sip

No.	Time	Source	Destination	Protocol	Info
1843	10:57:38.350	10.3.4.161	10.3.12.78	SIP	Status: 500 Server Internal Error
1844	10:57:38.352	10.3.2.152	10.3.12.78	SIP	Status: 500 Server Internal Error
1918	10:57:39.520	10.3.12.78	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
1919	10:57:39.526	10.2.1.199	10.3.12.78	SIP	Status: 200 OK (1 bindings)
26336	11:03:28.676	10.3.12.74	192.168.1.199	SIP	Request: REGISTER sip:192.168.1.199
26343	11:03:28.757	192.168.1.199	10.3.12.74	SIP	Status: 407 Proxy Authentication Required (0 bindings)
26347	11:03:28.852	10.3.12.74	192.168.1.199	SIP	Request: REGISTER sip:192.168.1.199
26349	11:03:28.864	192.168.1.199	10.3.12.74	SIP	Status: 200 OK (1 bindings)
37196	11:05:22.369	10.3.12.80	224.0.1.75	SIP	Request: SUBSCRIBE sip:MAC0015652443140224.0.1.75
37206	11:05:22.471	10.3.4.161	10.3.12.80	SIP	Status: 500 Server Internal Error
37207	11:05:22.472	10.3.2.152	10.3.12.80	SIP	Status: 500 Server Internal Error
43868	11:06:24.682	10.3.12.80	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
43869	11:06:24.688	10.2.1.199	10.3.12.80	SIP	Status: 200 OK (1 bindings)
54657	11:08:49.535	10.3.12.80	224.0.1.75	SIP	Request: SUBSCRIBE sip:MAC0015652443140224.0.1.75
54663	11:08:49.637	10.3.4.161	10.3.12.80	SIP	Status: 500 Server Internal Error
54664	11:08:49.637	10.3.2.152	10.3.12.80	SIP	Status: 500 Server Internal Error
54833	11:08:52.632	10.3.12.80	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
54834	11:08:52.638	10.2.1.199	10.3.12.80	SIP	Status: 200 OK (1 bindings)

Frame 54833: 533 bytes on wire (4264 bits), 533 bytes captured (4264 bits) on interface 0
 Ethernet II, Src: Xiamenve_24:43:14 (00:15:65:24:43:14), Dst: cisco_1e:47:46 (24:b6:57:1e:47:46)
 Internet Protocol Version 4, Src: 10.3.12.80 (10.3.12.80), Dst: 10.2.1.199 (10.2.1.199)
 User Datagram Protocol, Src Port: na-localise (5062), Dst Port: sip (5060)
 Session Initiation Protocol
 Session Initiation Protocol (SIP as raw text)
 REGISTER sip:10.2.1.199 SIP/2.0\r\n
 Via: SIP/2.0/UDP 10.3.12.80:5062;branch=z9hG4bK432775651\r\n
 From: "2541" <sip:2541@10.2.1.199>;tag=627541106\r\n
 To: "2541" <sip:2541@10.2.1.199>\r\n
 Call-ID: 1080536491010.3.12.80\r\n
 CSeq: 1 REGISTER\r\n
 Contact: <sip:2541@10.3.12.80:5062>\r\n
 Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH, UPDATE, MESSAGE\r\n
 Max-Forwards: 70\r\n
 User-Agent: Yealink SIP-T22P 7.70.0.80 00:15:65:24:43:14\r\n
 Expires: 3600\r\n
 Content-Length: 0\r\n
 \r\n

Disable this Feature :

Set 'network.sip.tag_mac_to_ua.enable = 0' in your configure files.

And the MAC address won't show in the User-Agent header.

Capturing from Atheros L1C PCI-E Ethernet Controller (Device\NPF_{A3F963AF-C134-46EB-95CC-C575F8E05B0}) [Wireshark 1.8.2 (SVN Rev 44520 from /trunk-1.8)]

Filter: sip

No.	Time	Source	Destination	Protocol	Info
1843	10:57:38.350	10.3.4.161	10.3.12.78	SIP	Status: 500 Server Internal Error
1844	10:57:38.352	10.3.2.152	10.3.12.78	SIP	Status: 500 Server Internal Error
1918	10:57:39.520	10.3.12.78	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
1919	10:57:39.526	10.2.1.199	10.3.12.78	SIP	Status: 200 OK (1 bindings)
26336	11:03:28.676	10.3.12.74	192.168.1.199	SIP	Request: REGISTER sip:192.168.1.199
26343	11:03:28.757	192.168.1.199	10.3.12.74	SIP	Status: 407 Proxy Authentication Required (0 bindings)
26347	11:03:28.852	10.3.12.74	192.168.1.199	SIP	Request: REGISTER sip:192.168.1.199
26349	11:03:28.864	192.168.1.199	10.3.12.74	SIP	Status: 200 OK (1 bindings)
37196	11:05:22.369	10.3.12.80	224.0.1.75	SIP	Request: SUBSCRIBE sip:MAC0015652443140224.0.1.75
37206	11:05:22.471	10.3.4.161	10.3.12.80	SIP	Status: 500 Server Internal Error
37207	11:05:22.472	10.3.2.152	10.3.12.80	SIP	Status: 500 Server Internal Error
43868	11:06:24.682	10.3.12.80	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
43869	11:06:24.688	10.2.1.199	10.3.12.80	SIP	Status: 200 OK (1 bindings)
54657	11:08:49.535	10.3.12.80	224.0.1.75	SIP	Request: SUBSCRIBE sip:MAC0015652443140224.0.1.75
54663	11:08:49.637	10.3.4.161	10.3.12.80	SIP	Status: 500 Server Internal Error
54664	11:08:49.637	10.3.2.152	10.3.12.80	SIP	Status: 500 Server Internal Error
54833	11:08:52.632	10.3.12.80	10.2.1.199	SIP	Request: REGISTER sip:10.2.1.199
54834	11:08:52.638	10.2.1.199	10.3.12.80	SIP	Status: 200 OK (1 bindings)

Frame 43868: 515 bytes on wire (4120 bits), 515 bytes captured (4120 bits) on interface 0
 Ethernet II, Src: Xiamenve_24:43:14 (00:15:65:24:43:14), Dst: cisco_1e:47:46 (24:b6:57:1e:47:46)
 Internet Protocol Version 4, Src: 10.3.12.80 (10.3.12.80), Dst: 10.2.1.199 (10.2.1.199)
 User Datagram Protocol, Src Port: na-localise (5062), Dst Port: sip (5060)
 Session Initiation Protocol
 Session Initiation Protocol (SIP as raw text)
 REGISTER sip:10.2.1.199 SIP/2.0\r\n
 Via: SIP/2.0/UDP 10.3.12.80:5062;branch=z9hG4bK1932278011\r\n
 From: "2541" <sip:2541@10.2.1.199>;tag=947987127\r\n
 To: "2541" <sip:2541@10.2.1.199>\r\n
 Call-ID: 883399637@10.3.12.80\r\n
 CSeq: 1 REGISTER\r\n
 Contact: <sip:2541@10.3.12.80:5062>\r\n
 Allow: INVITE, INFO, PRACK, ACK, BYE, CANCEL, OPTIONS, NOTIFY, REGISTER, SUBSCRIBE, REFER, PUBLISH, UPDATE, MESSAGE\r\n
 Max-Forwards: 70\r\n
 User-Agent: yealink SIP-T22P 7.70.0.80\r\n
 Expires: 3600\r\n
 Content-Length: 0\r\n
 \r\n

Detail : For more details please refer to Yealink Auto Provision User Guide.pdf

5. Optimization description

1. Supported to enter the Voicemail box by pressing the Message button when then Menu key is locked.
2. Added the “Transfer Mode via DSS key” configuration.

Path : Phone-> Features->Transfer Settings→Transfer Mode via DSS Key

The screenshot shows the 'Transfer Settings' configuration page. On the left, there are links for 'Call Pickup >>', 'API Security >>', and 'Auto Call Distribution Settings >>'. The main area contains five settings, each with a dropdown menu and a help icon (question mark):

- Semi-Attended Transfer: Enabled
- Blind Transfer on Hook: Enabled
- Attended Transfer on Hook: Enabled
- Transfer on Conference Hang Up: Disabled
- Transfer Mode via DSSkey: Blind Transfer (highlighted with a red box, showing a dropdown menu with options: New Call, Attended Transfer, Blind Transfer)

Description :

- (1) When DSS Key is set to BLF , BLF List , Speed Dial , Transfer, if “Transfer Mode via DSS Key” is set to “Blind Transfer” ,phone will make blind transfer when pressing this DSS Key during an active call
- (2) When DSS Key is set to BLF , BLF List , Speed Dial , Transfer, if “Transfer Mode via DSS Key” is set to “Attended Transfer” ,phone will make attended transfer when pressing this DSS Key during an active call
- (3) When DSS Key is set to BLF , BLF List , Speed Dial , Transfer, if “Transfer Mode via DSS Key” is set to “New Call” ,phone will make a new call when pressing this DSS Key during an active call

3. Classified the configurations in the Phone→Features web setting page.

Path : Phone-> Features

Forward >> ?

DND >>

General >>

Audio Settings >>

Intercom Settings >>

Transfer Settings >>

Semi-Attended Transfer: Enabled ?

Blind Transfer on Hook: Enabled ?

Attended Transfer on Hook: Enabled ?

Transfer on Conference Hang Up: Disabled ?

Transfer Mode via DSSkey: New Call ?

Call Pickup >>

API Security >>

Auto Call Distribution Settings >>

Confirm Cancel

NOTE

Forward
This feature allows you to forward an incoming call to another phone number.

Target
The number to which the incoming calls will be forwarded.

On Code
The code that will be sent to PBX when it is switched On.

Off Code
The code that will be sent to PBX when it is switched Off.

Call Waiting
This call feature allows your phone to accept other incoming calls during the conversation.

Key As Send
Select * or # as the send key.

The Web frame structure of V61 version:

Forward: ?

Always ☐ On ☐ Off

Target: ?

On Code: ?

Off Code: ?

Busy ☐ On ☐ Off

Target: ?

On Code: ?

Off Code: ?

No Answer ☐ On ☐ Off

After Ring Time(seconds): 20 ?

Target: ?

On Code: ?

Off Code: ?

General Information:

Call Waiting: Enabled ?

Call Waiting Tone: Enabled ?

Auto redial: Disabled ?

Key As Send: # ?

Reserve # in User Name: Enabled ?

Button Sound: Enabled ?

Send Sound: Enabled ?

Hotline Number: ?

Hotline Delay: 4 ?

ReDialTone: ?

NOTE

Forward
This feature allows you to forward an incoming call to another phone number.

Target
The number to which the incoming calls will be forwarded.

On Code
The code that will be sent to PBX when it is switched On.

Off Code
The code that will be sent to PBX when it is switched Off.

Call Waiting
This call feature allows your phone to accept other incoming calls during the conversation.

Key As Send
Select * or # as the send key.

Hotline Number
When you pick up the phone, it will dial out the hotline number automatically.

Upload Logo
The picture must be format of jpg, it can be black and white, or 2 gray scale.

4. Supported DND & FWD separated feature.

Path: Phone-> Features->Forward/DND

Forward >> ?

Call Forward Key Mode: ☒ Phone ☐ Custom

Account	Call Forward	Status	Target	Time Out	On Code	Off Code
0.Phone	Always	<input type="checkbox"/>	<input type="text"/>		<input type="text"/>	<input type="text"/>
	Busy	<input type="checkbox"/>	<input type="text"/>		<input type="text"/>	<input type="text"/>
	No Answer	<input type="checkbox"/>	<input type="text"/>	12 <input type="button" value="v"/>	<input type="text"/>	<input type="text"/>

DND >>
General >>

Forward >> ?

DND >>

DND Key Mode: ☒ Phone ☐ Custom

Account	DND	On Code	Off Code
0.Phone	<input type="checkbox"/>	<input type="text"/>	<input type="text"/>

General >>
Audio Settings >>

Description :

When the “mode” is set to “Phone”, DND&FWD configuration will effect on all accounts
When the “mode” is set to “Custom”, DND&FWD can be controlled separately by account.

Detail : For more details, please refer to the introduction of the DND and FORWARD feature in Yealink SIP-T2xP User Guide.pdf

5. Optimized keypad lock feature

Path : Phone-> Preference

Keypad Lock

Phone-Unlock PIN(0~15 digital)

Phone Lock Time Out(0~3600s)

Emergency

Detail : For more details, please refer to the introduction of Keypad Lock feature in Yealink SIP-T2xP User Guide.pdf

6. Supported TAP mode of OpenVPN.

- 1) dev-TUN mode, works in network layer , already supported in T2X V61.
- 2) dev-TAP mode, works in data link layer, support serveralOpen VPN connection.

7. Added the OpenVPN feature toT20/T22 models.

8. Modified ATS conference feature to be Optional.

Description: Deleted ATS conference feature, not support in the neutral version, but set as a customer special feature.

9. Modified Draft BLA feature to be Optional.

Description: Deleted Draft BLA feature, not support in the neutral version, but set as a customer special feature.

10. Added PEAP-MSCHAPV2 authentication for 802.1X feature.

- 1) Add PEAP-MSCHAPV2 option on LCD, path: Advanced->Network->802.1X Setting



- 2) Add PEAP-MSCHAPV2 authentication option and upload certificate option in Network->Advanced->802.1X webpage (certificate should be uploaded through webpage).
- 3) network.802_1x.root_cert_url = (upload the root certificate of 802.1x)
- 4) network.802_1x.root_cert_file_name = (save the name of 802.1X CA certificate)

11. Added EAP-TLS authentication for 802.1X feature.

- 1) Add EAP-TLS option on LCD, path: Advanced->Network->802.1X Setting



- 2) Add EAP-TLS authentication option and upload certificate option in Network->Advanced->802.1X webpage (certificate should be uploaded through webpage).
- 3) network.802_1x.client_cert_url = (upload the client certificate of 802.1x)
- 4) network.802_1x.client_cert_file_name = (save the device certificate of 802.1x)

12. Added domain name supported for PushXML server address.

Description :

When fill in the domain name to the push XML server IP, www.xmyealink.com for example, the phone will analysis correctly to the domain and after get the result, it will receive the correct XML information from the corresponding IP address.



13. Optimized “Logo + character HD ”icon display in the LCD of T28/T26 when calling with codec G722.

When T28 call T26 with the codec G722, both of T28 and T26 will display “Logo + character HD ” in the lower right corner of LCD.



14. Did optimization of dealing with BLF Notify message which is out of dialog.

Description :

Normally, the phone only deal with BLF Notify message which matched with subscribe message. But the BLF Notify message from some server doesn't match with subscribe message.

When you enable "Out Dialog BLF" option, the phone can deal with such BLF Notify message which is out of Dialog.

Path : Account → Advanced

Conference URI	<input type="text"/>	?
ACD Subscription Period(120~3600)	<input type="text" value="3600"/>	?
Caller ID Header	<input type="text" value="FROM"/>	?
Early Media	<input type="text" value="Disabled"/>	?
SIP Server Type	<input type="text" value="Default"/>	
Music on Hold Server	<input type="text"/>	
Distinctive Ring Tones	<input type="text" value="Enabled"/>	
Unregister after Reboot	<input type="text" value="Disabled"/>	
Out Dialog BLF	<input type="text" value="Disabled"/>	

15. Did optimization of ringtone.

Description :

Old ringtone is not standard,

Optimized five standard ringtones

- 1) Ring2S then stop for 4S then repeat.
- 2) Ring400ms then stop for 200ms then ring 400ms then stop2000ms then repeat.(For incoming call from external)
- 3) Ring 400ms then stop for 200ms then ring 400ms then stop2000ms then repeat. (For incoming call from internal)
- 4) Ring 500ms then stop for7500ms then repeat.

16. Default Input Method of the Broadsoft network phonebook.

Description: Default Input Method of the Broadsoft network phonebook to ABC

17. Added some fields to the snmp section.

Description: Added some fields to the snmp section ,for example, community , sysContact .

18. Added the feature that the phone can use the SNMP to reboot the phone.

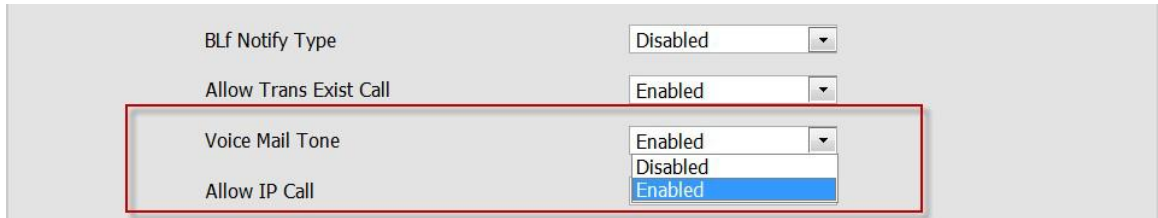
19. Added the dual-headset function.

20. Added new parameters to Action URL feature.

Description: New parameters added to Action URL feature are showing below
AnswerNewIncomingCall, RejectIncomingCall, TransferFinished ,TransferFailed,
ForwardIncomingCall

21. Added “Voice Mail Tone “configuration

Path: Phone -> Feature ->General



BLf Notify Type	Disabled
Allow Trans Exist Call	Enabled
Voice Mail Tone	Enabled
Allow IP Call	Enabled

Description:

When the “Voice Mail Tone” is set to enable, phone will play the voice mail tone when received a voice mail.

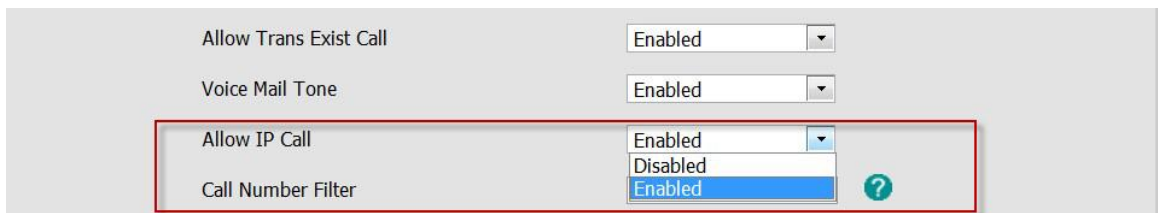
When the “Voice Mail Tone” is set to disable, phone will not play the voice mail tone when received a voice mail.

22. Optimized XML browse.

Description: Optimized XML browse that it can support hot desking feature.

23. Added “Allow IP Call “configuration.

Path: Phone -> Feature -> General



Allow Trans Exist Call	Enabled
Voice Mail Tone	Enabled
Allow IP Call	Enabled
Call Number Filter	Enabled

Description:

When “Allow IP Call” is set to enable, the user can make an IP call.

When “Allow IP Call” is set to disable, the user can’t make an IP call.

24. Did optimization that the phone can disable the local DTMF Tone.

25. Did optimization that the phone can support 99 entries for Dial-now Rule.

26. Did optimization of the LDAP search interface, show the default IME when enter this interface and the IME can be configured.

Description:

When enter the LDAP interface through the DSSKEY, it will show the IME immediately, and this IME is the one you used last time.

When doing LDAP search in the Dial/PreDia interface, the default IME can be configured by the configure file.

Feature effect.

1. When enter the LDAP interface through the DSSKEY, it will show the IME immediately, and this IME is the one you used last time.
2. When doing LDAP search in the Dial/PreDia interface, the default IME only can be configured by Auto Provision, and it is decided by the parameter 'IMEType=X'
X= 0 means 2ab
X= 1 means 123
X= 2 means abc
X= 3 means ABC

27. Did optimization of DHCP, supported to get NTP server address through Option 42.**Description:**

When this feature is enabled, once the phone boot up and get IP address through DHCP, the phone will get the NTP server address at the same time from the DHCP message that send from DHCP server. And fill in the NTP server info to the NTP server address bar, and get the time info through NTP server.

Note: Before you use this feature, please enable the getting time through DHCP feature and the NTP serve address must be an IP address.

Detail: For more details please refer to Yealink SIP-T2xP User Guide.pdf.

28. Did optimization that the Direct IP Call feature can be configured through Auto Provisioning.**Description:**

The Direct IP Call feature can be enabled or disabled through Auto Provisioning.

Enable Direct IP Call:

Set 'sip_ip_call.enable=1' in your configure files. Then you can make a Direct IP Call.

Disable Direct IP Call:

Set 'sip_ip_call.enable=0' in your configure files. Then you can't make a Direct IP Call.

Detail : For more details please refer to Yealink Auto Provision User Guide.pdf