Flyingvoice IP Phone Advanced Training&Troubleshooting

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Agenda

- Account Registration
- SIP Call
- Auto Provisioning
- BLF
- Network
- Phone become Slow/Stuck/ Reboot

Account Registration

Account Registration

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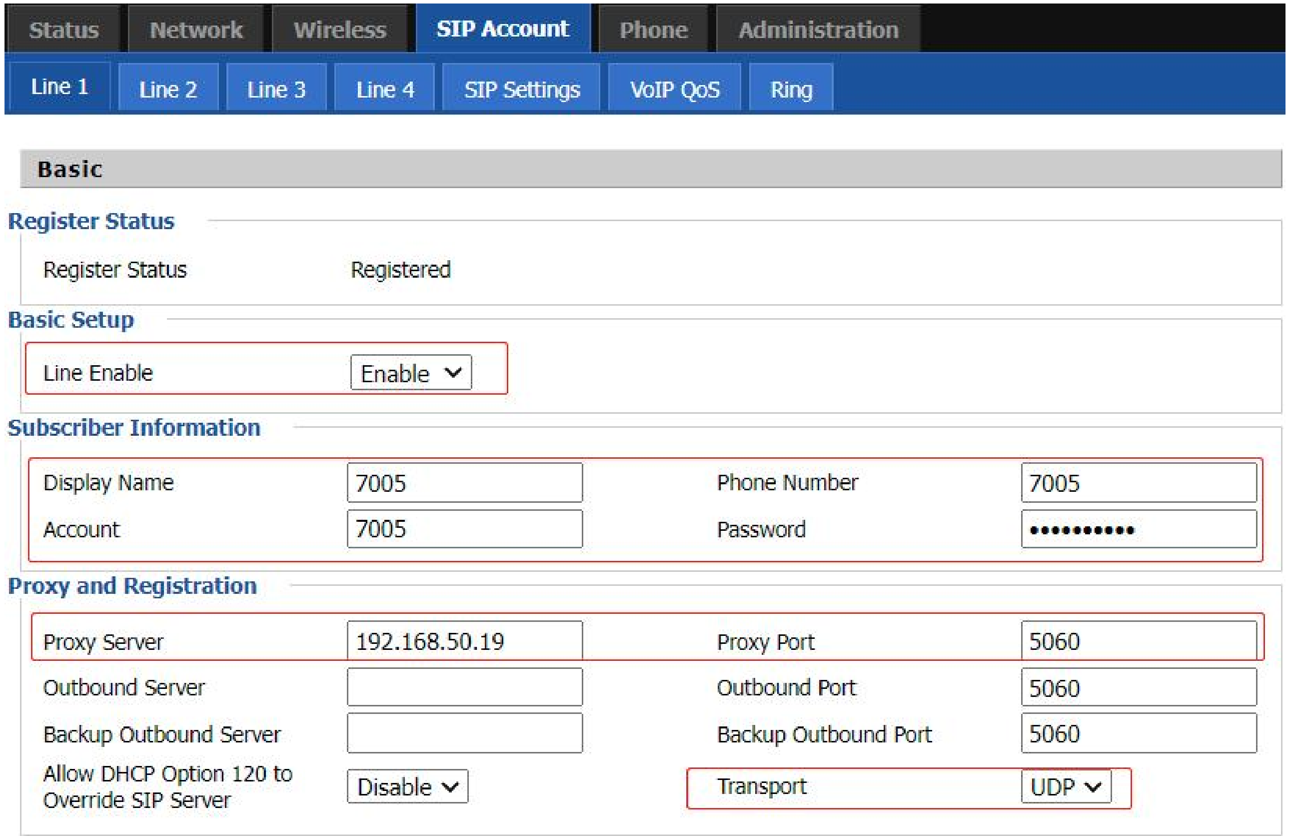
Parameter Configuration



Line Enable: Enable or Disable the account.

Display Name: The display name of account.

Phone Number: The register user name.

Account: The username for register authentication.

Password: The password for register authentication.

Proxy Server: The SIP server address.

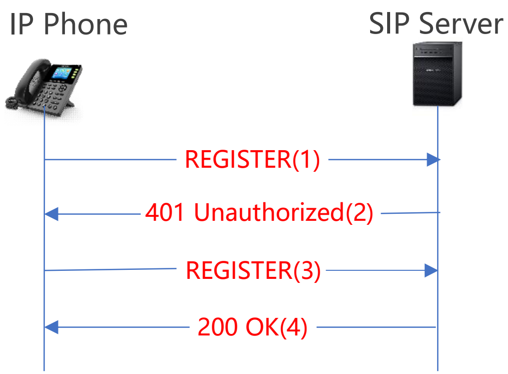
Proxy Port: The port for the SIP server.

Transport: The transport type.

Account Registration

Registration Process

The SIP registration process is shown as follows.



(1) IP Phone sends a Register request to the SIP server.

(2) SIP Server sends back a challenge to request authentication information.

(3) IP Phone sends a Register message with the authentication information.

(4) SIP Server reply "200 OK", the phone is registered successfully.

Below is the basic trace flow for registration.

2021-12-31 13:58:38.366121	22771	192.168.80.198	192.168.50.19	SIP	562 Request: REGISTER sip:192.168.50.19 (1 binding) |
2021-12-31 13:58:38.368498	22772	192.168.50.19	192.168.80.198	SIP	541 Status: 401 Unauthorized |
2021-12-31 13:58:38.646248	22781	192.168.80.198	192.168.50.19	SIP	823 Request: REGISTER sip:192.168.50.19 (1 binding) |
2021-12-31 13:58:38.649283	22782	192.168.50.19	192.168.80.198	SIP	488 Status: 200 OK (1 binding) |

Note:

Phone: 192.168.80.198

Server: 192.168.50.19

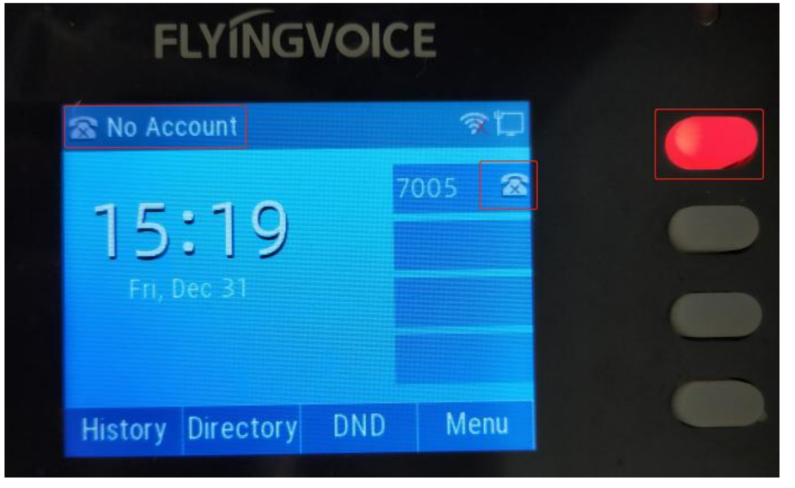
Account Registration

The second register message includes the authentication information.

13:58:38.366121	22771	192.168.80.198	192.168.50.19	SIP	562	Request: REGISTER sip:192.168.50.19 (1 binding) |
13:58:38.368498	22772	192.168.50.19	192.168.80.198	SIP	541	Status: 401 Unauthorized |
13:58:38.646248	22781	192.168.80.198	192.168.50.19	SIP	823	Request: REGISTER sip:192.168.50.19 (1 binding) |
13:58:38.649283	22782	192.168.50.19	192.168.80.198	SIP	488	Status: 200 OK (1 binding) |

```
Supported: replaces
User-Agent: FLYINGVOICE FIP13G SV0.6.59(202112251856) 202112251856
Contact: <sip:7005@192.168.80.198:5061>
Expires: 1800
[truncated]Authorization: Digest username="7005", realm="YSAsterisk", nonce="1640930319/8d1b4479dac24985db5cb5a46e58fff8", uri="sip:192.168.50.19", response="f
  Authentication Scheme: Digest
  Username: "7005"
  Realm: "YSAsterisk"
  Nonce Value: "1640930319/8d1b4479dac24985db5cb5a46e58fff8"
  Authentication URI: "sip:192.168.50.19"
  Digest Authentication Response: "f0cd1c9fcf774456419eb5563ccaff2a"
  Algorithm: MD5
  CNonce Value: "b4b"
  Opaque Value: "23278cf06c945dae"
  QOP: auth
  Nonce Count: 00000001
```

When the account fails to register, you will see "No Account" display on the top right corner of the screen, the Line Light is Red, and the Line Key icon is Gray.



The following points may cause registration failure.

Phone: Incorrect registration information(wrong username/password/server address/port/transport type)

Network: Unavailable network(no IP address/wrong physical connection/can't ping through the server)

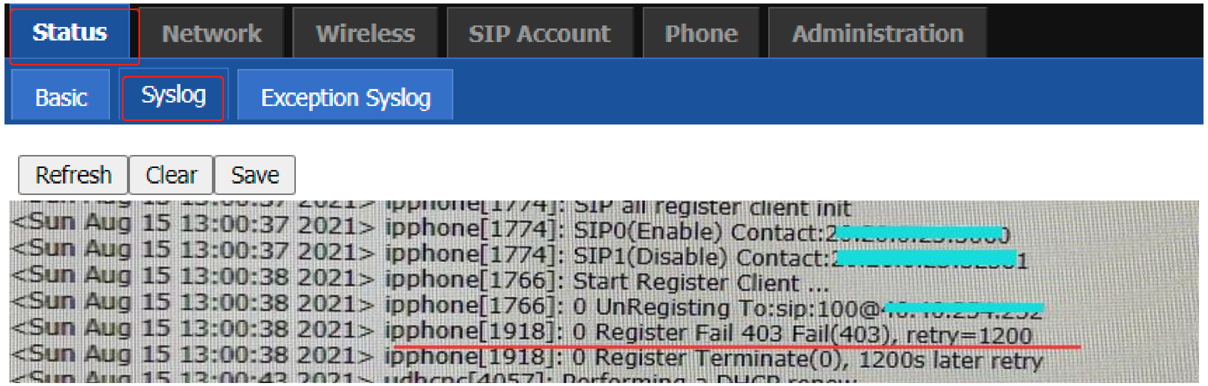
SIP server: Account availability(account enabled/disabled, multiple registrations), firewall settings(Block third-party device/block the phone/block the network/block the area)

Account Registration--Troubleshooting

1. Double check the account info.

2. Check the System log on the phone.

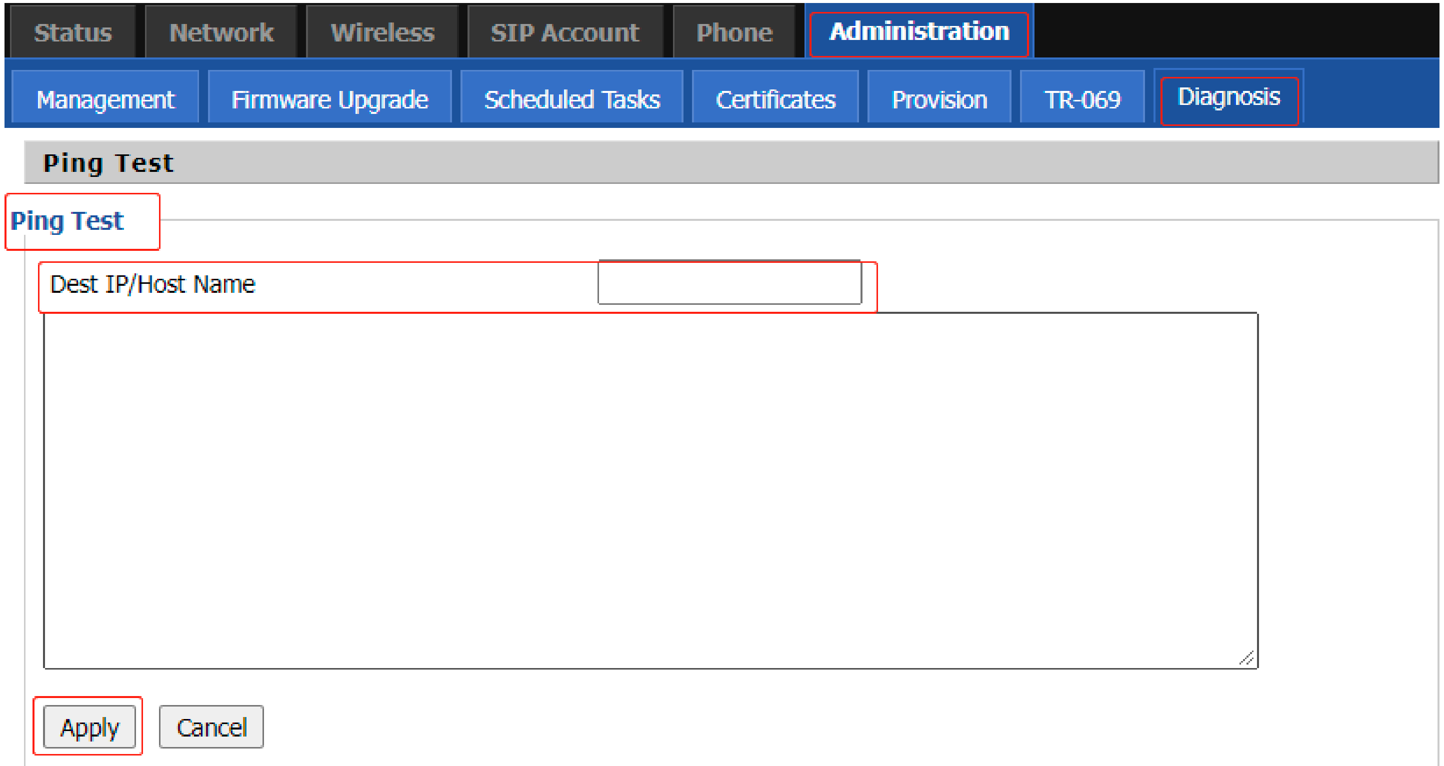
403 Fail: wrong account info, or multiple devices registered to the same account, or blocked by the server.



Register Fail, Timeout: network is not avaiable or wrong SIP server address/port

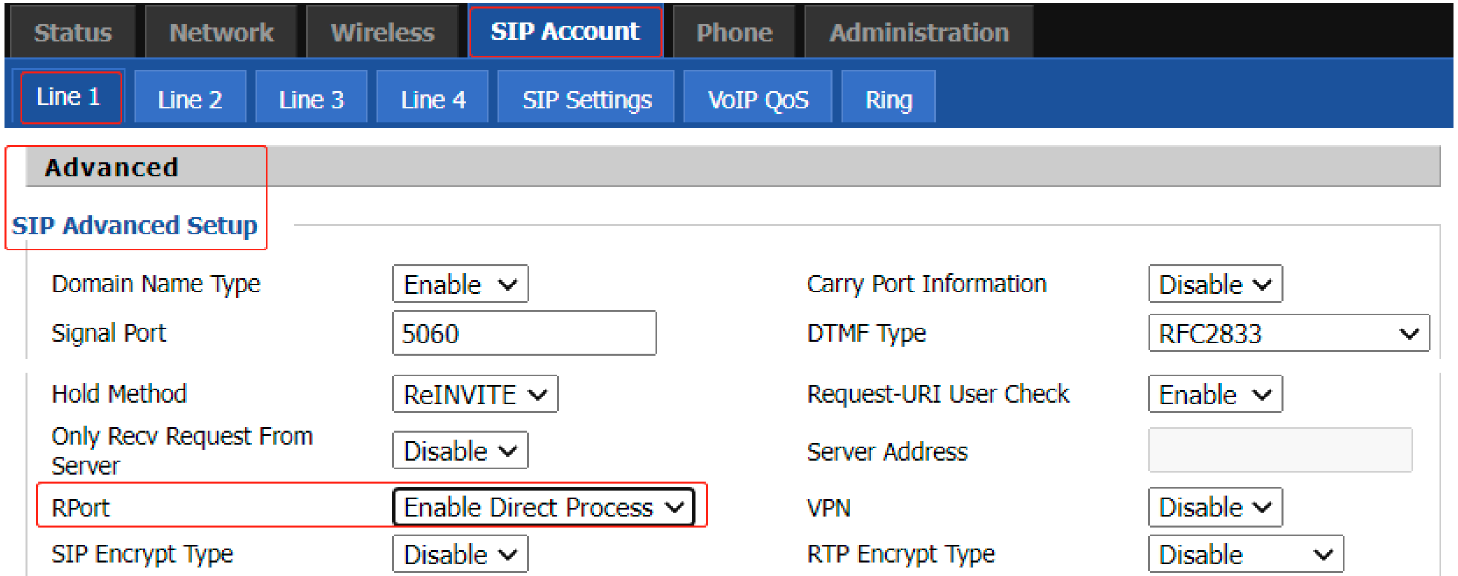
```
<Tue Nov  2 15:28:24 2021> ipphone[30593]: 0 Register Terminate(0), 2s later retry
<Tue Nov  2 15:28:26 2021> ipphone[30593]: 0 Registing To:<sip:6618@[redacted]
<Tue Nov  2 15:28:58 2021> ipphone[30593]: 0 Register Fail, Timeout
<Tue Nov  2 15:28:58 2021> ipphone[30593]: Reg terminated eReason=0
<Tue Nov  2 15:28:58 2021> ipphone[30593]: 0 Register Terminate(0), 30s later retry
```

Ping 8.8.8.8 to check the network connectivity, ping the server address to check the connectivity.



Account Registration--Troubleshooting

3. If the network environment involves NAT(Network Address Translation), you can set the "RPort" to "Enable Direct Process" and try again.



4. Check whether there are any other Models (brand) or softphones that work fine in the same scenario with the same account.

Account Registration--Troubleshooting

5. Check package trace.

403 Authentication Failure: Wrong password.

403 Forbidden: Wrong username or password, or multiple devices registered to the same account, or blocked by the server.

192.168.80.198	192.168.50.246	SIP	823 Request: REGISTER sip:192.168.50.246 (1 binding) |
192.168.50.246	192.168.80.198	SIP	384 Status: 403 Forbidden |

404 Not Found: Wrong account, check the account information on the SIP server.

Source	Destination	Protocol	Length	Info
192.168.80.105	192.168.50.166	SIP	580	Request: REGISTER sip:192.168.50.166 (1 binding) |
192.168.50.166	192.168.80.105	SIP	323	Status: 404 Not Found |

423 Interval Too Brief: The phone's register refresh interval is too short, change it to a longer time.

Source	Destination	Protocol	Length	Info
192.168.80.105	192.168.50.166	SIP	574	Request: REGISTER sip:192.168.50.166 (1 binding) |
192.168.50.166	192.168.80.105	SIP	345	Status: 423 Interval Too Brief |

No respond from server: Check server connection and network environment

Source	Destination	Protocol	Length	Info
192.168.8.120	165[illegible]	SIP	925	Request: REGISTER sip:ppr[illegible]5060 (1 binding) |
192.168.8.120	165[illegible]	SIP	925	Request: REGISTER sip:ppr[illegible]5060 (1 binding) |

503: Server unavailable

Account Registration--Troubleshooting

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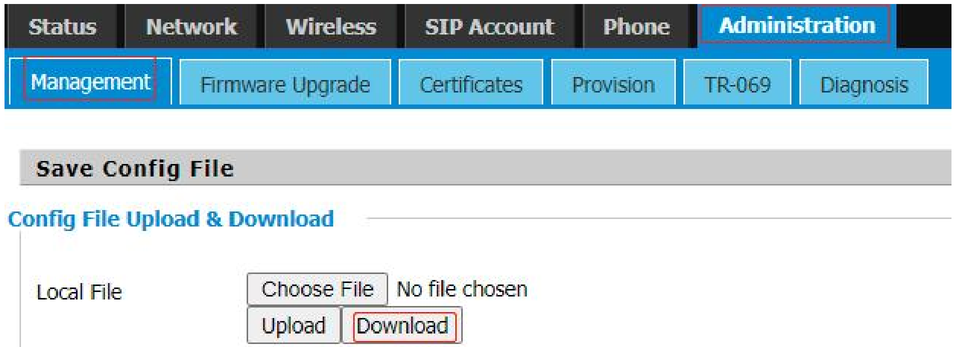
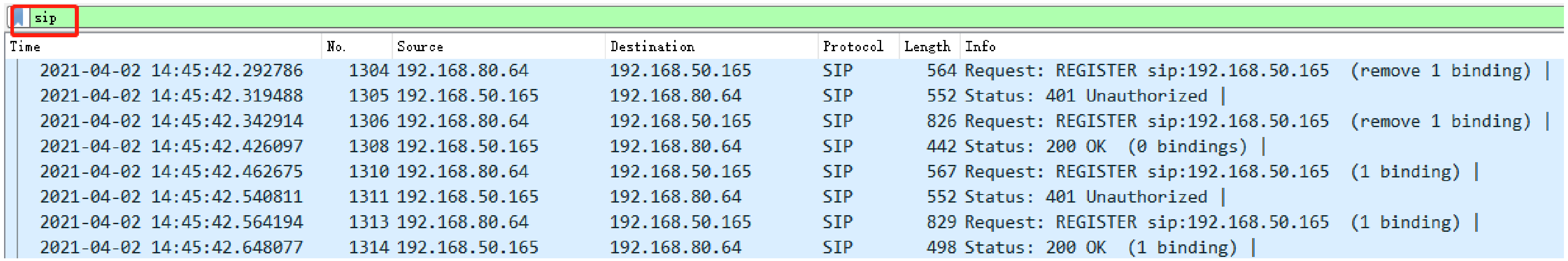
If the above troubleshooting still can't resolve the problem. Please send us the following debug files to check.

1. The phone's configuration file.



2. The phone's packet trace. (If another phone can register successfully, get a pcap trace on it to compare, send us the OK/NOK trace.)

Please confirm the file contain effective information, after filtering sip, you can see trace like below.


sip

Time	No.	Source	Destination	Protocol	Length	Info
2021-04-02 14:45:42.292786	1304	192.168.80.64	192.168.50.165	SIP	564	Request: REGISTER sip:192.168.50.165 (remove 1 binding) |
2021-04-02 14:45:42.319488	1305	192.168.50.165	192.168.80.64	SIP	552	Status: 401 Unauthorized |
2021-04-02 14:45:42.342914	1306	192.168.80.64	192.168.50.165	SIP	826	Request: REGISTER sip:192.168.50.165 (remove 1 binding) |
2021-04-02 14:45:42.426097	1308	192.168.50.165	192.168.80.64	SIP	442	Status: 200 OK (0 bindings) |
2021-04-02 14:45:42.462675	1310	192.168.80.64	192.168.50.165	SIP	567	Request: REGISTER sip:192.168.50.165 (1 binding) |
2021-04-02 14:45:42.540811	1311	192.168.50.165	192.168.80.64	SIP	552	Status: 401 Unauthorized |
2021-04-02 14:45:42.564194	1313	192.168.80.64	192.168.50.165	SIP	829	Request: REGISTER sip:192.168.50.165 (1 binding) |
2021-04-02 14:45:42.648077	1314	192.168.50.165	192.168.80.64	SIP	498	Status: 200 OK (1 binding) |

SIP Call

Basic Call

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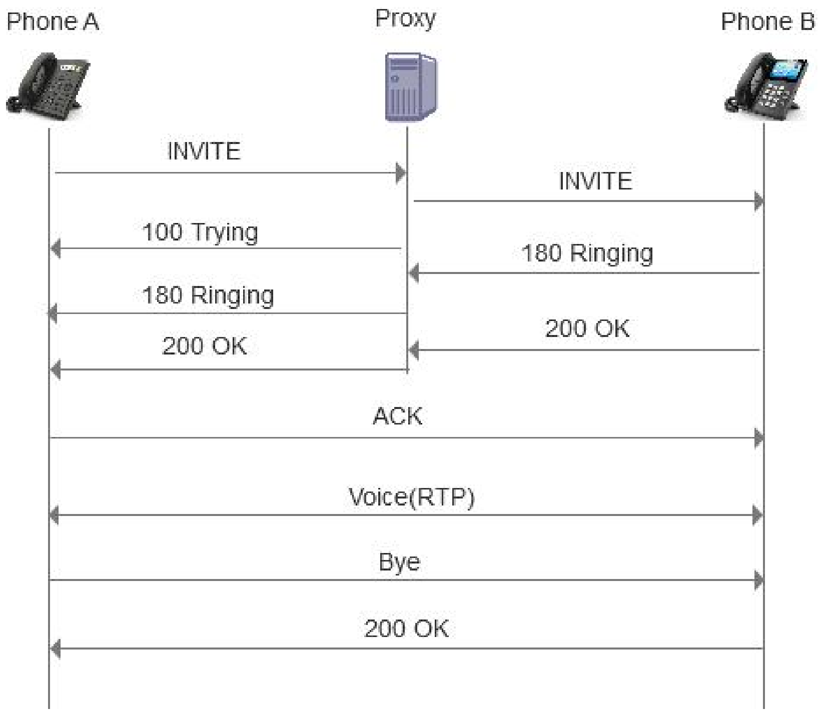
Basic Call Process



Pcap Trace

Source	Destination	Protocol	Length	Info
192.168.50.19	192.168.80.198	SIP/SDP	1189	Request: INVITE sip:7005@192.168.80.198:5060 |
192.168.80.198	192.168.50.19	SIP	511	Status: 100 Trying |
192.168.80.198	192.168.50.19	SIP	517	Status: 180 Ringing |
192.168.80.198	192.168.50.19	SIP/SDP	1162	Status: 200 OK |
192.168.50.19	192.168.80.198	SIP	433	Request: ACK sip:7005@192.168.80.198:5060 |
192.168.50.19	192.168.80.198	SIP	457	Request: BYE sip:7005@192.168.80.198:5060 |
192.168.80.198	192.168.50.19	SIP	536	Status: 200 OK |

Note:

Phone A or Server: 192.168.50.19

Phone B: 192.168.80.198

One-way audio issue/Two-way no voice issue

1. Check the audio codec, make sure that both sides negotiate the same audio codec. Filter **rtp,** you can only see one type of codec.

rtp

Time	No.	Source	Destination	Protocol	Length	Info
2022-01-13 17:09:53.668313	4387	192.168.80.23	192.168.50.19	RTP	214	PT=ITU-T G.711 PCMA, SSRC=0x20948B74, Seq=61214, Time=132000
2022-01-13 17:09:53.679423	4389	192.168.50.19	192.168.80.23	RTP	214	PT=ITU-T G.711 PCMA, SSRC=0x3A436E38, Seq=28332, Time=261760, Mark
2022-01-13 17:09:53.688377	4391	192.168.80.23	192.168.50.19	RTP	214	PT=ITU-T G.711 PCMA, SSRC=0x20948B74, Seq=61215, Time=132160

2. In the network environment involving NAT(Network Address Translation), set the "**RPort**" to "**Enable Direct Process**" and try again.

3. Check whether there are any other Models (brand) or softphones that work fine in the same scenario with the same account.

4. Contact Flyingvoice FAE for help. Provide the details below.

(1) Describe the detailed steps to reproduce the issue. E.g. A calls B, B answers, the LCD of A and B are in talking status, A can hear B but B can't hear A.

(2) Provide more case details.

- The network topology. E.g. the connection of the router/PBX/phones, the model of A and B, the advanced settings(VLAN/SIP ALG).
- Does the issue happen on internal calls or external calls?
- Does it work fine before? When does the issue happen?
- How many phones does the customer have and how many of them have the issue.

(3) The phone's configuration file.

(4) If the phone register to the cloud PBX, provide us with some test accounts to test and reproduce the issue.

(5) The phone's packet trace. (If another phone works fine, get a pcap trace on it to compare, send us the OK/NOK trace.)

Please confirm the file contain effective information, after filtering sip, you can see trace like on the previous page.

sip

	No.	Source	Destination	Protocol	Length	Info
17:06:32.278900	6215	192.168.50.19	192.168.80.23	SIP/SDP	1190	Request: INVITE sip:7002@192.168.80.23:5060 |
17:06:32.281600	6216	192.168.80.23	192.168.50.19	SIP	551	Status: 200 OK |
17:06:32.330380	6222	192.168.80.23	192.168.50.19	SIP	510	Status: 100 Trying |
17:06:32.348440	6223	192.168.80.23	192.168.50.19	SIP	516	Status: 180 Ringing |
17:06:34.042570	6346	192.168.80.23	192.168.50.19	SIP/SDP	1157	Status: 200 OK |

Call out/Call in failure

1. Make sure your SIP phones are registered successfully.

2. Check whether the account set SRTP on the PBX server, if so, the phone needs to enable SRTP.

3. Make sure the phone is disabled **Allow IP Calls** or **Outgoing Call without Registration** when he registers to the PBX server.

4. In the network environment involving NAT(Network Address Translation), set the "**RPort**" to "**Enable Direct Process**" and try again.

5. Check whether there are any other Models (brand) or softphones that work fine in the same scenario with the same account.

6. Contact Flyingvoice FAE for help. Provide the details below.

(1) Describe the detailed scenario. E.g. It's for dial-out failure or dial-in failure? When the call failed, what are the prompts on both phone LCD? Does it happen on the internal calls or external calls? Does it work fine before? When does the issue happen? How many phones does the customer have and how many of them have the issue.

(2) If the phone register to the cloud PBX, provide us with some test accounts to test and reproduce the issue.

(5) The phone's packet trace. (If another phone works fine, get a pcap trace on it to compare, send us the OK/NOK trace.)

Please confirm the file contain effective information, after filtering sip, you can see the related sip message.

Call Hold

During a call, the phone sends an INVITE message to the server to hold the call. In the SDP information, there is the keyword '**Sendonly**'.

When pressing the Resume key, the phone will send another INVITE message to the server which SDP information with '**Sendrecv**' to retrieve the call.

Source	Destination	Protocol	Length	Info
192.168.80.23	192.168.50.19	SIP/SDP	892	Request: INVITE sip:7006@192.168.50.19 |
192.168.50.19	192.168.80.23	SIP	534	Status: 401 Unauthorized |
192.168.80.23	192.168.50.19	SIP	549	Request: ACK sip:7006@192.168.50.19 |
192.168.80.23	192.168.50.19	SIP/SDP	1159	Request: INVITE sip:7006@192.168.50.19 |
192.168.50.19	192.168.80.23	SIP	347	Status: 100 Trying |
192.168.50.19	192.168.80.23	SIP	541	Status: 180 Ringing |
192.168.50.19	192.168.80.23	SIP	613	Status: 180 Ringing |
192.168.50.19	192.168.80.23	SIP/SDP	921	Status: 200 OK |
192.168.80.23	192.168.50.19	SIP	833	Request: ACK sip:7006@192.168.50.19:5060 |
192.168.80.23	192.168.50.19	SIP/SDP	1204	Request: INVITE sip:7006@192.168.50.19:5060, in-dialog |
192.168.50.19	192.168.80.23	SIP/SDP	859	Status: 200 OK |
192.168.80.23	192.168.50.19	SIP	833	Request: ACK sip:192.168.50.19:5060 |

```
Session Description Protocol Version (v): 0
> Owner/Creator, Session Id (o): F 1641377386 1641377388 IN IP4 192.168.80.23
Session Name (s): F
> Connection Information (c): IN IP4 0.0.0.0
> Time Description, active time (t): 0 0
> Media Description, name and address (m): audio 10000 RTP/AVP 8 0 101
> Media Attribute (a): ptime:20
> Media Attribute (a): rtpmap:8 PCMA/8000
> Media Attribute (a): fmtp:8 vad=no
> Media Attribute (a): rtpmap:0 PCMU/8000
> Media Attribute (a): fmtp:0 vad=no
> Media Attribute (a): rtpmap:101 telephone-event/8000
> Media Attribute (a): fmtp:101 0-15
Media Attribute (a): sendonly
[Generated Call-ID: 1750a8c061d56e6a0-2ea15180]
```

Call Transfer--Blind Transfer

7002 talk with 7007, 7002 perform the transfer, 7002 sends REINVITE message to hold the call, sends REFER message to transfer the call to 7001.

	No.	Source	Destination	Protocol	Length	Info
52.175972	486	192.168.50.19	192.168.80.20	SIP	432	Request: ACK sip:7002@192.168.80.20:5060 |
53.410834	788	192.168.80.20	192.168.50.19	SIP/SDP	1215	Request: INVITE sip:7007@192.168.50.19:5060, in-dialog |
53.414249	790	192.168.50.19	192.168.80.20	SIP/SDP	884	Status: 200 OK |
53.448395	795	192.168.80.20	192.168.50.19	SIP	583	Request: ACK sip:unknown@192.168.50.19:5060 |
57.953265	1675	192.168.80.20	192.168.50.19	SIP	701	Request: REFER sip:unknown@192.168.50.19:5060, in-dialog |
57.955527	1677	192.168.50.19	192.168.80.20	SIP	632	Status: 202 Accepted |
57.955882	1678	192.168.50.19	192.168.80.20	SIP/si…	657	Request: NOTIFY sip:7002@192.168.80.20:5060 | , with Sipfrag(SIP/2.0 100 Trying)
57.983849	1681	192.168.50.19	192.168.80.20	SIP/si…	658	Request: NOTIFY sip:7002@192.168.80.20:5060 | , with Sipfrag(SIP/2.0 180 Ringing)
57.984940	1683	192.168.50.19	192.168.80.20	SIP/si…	663	Request: NOTIFY sip:7002@192.168.80.20:5060 | , with Sipfrag(SIP/2.0 200 OK)
58.028211	1688	192.168.80.20	192.168.50.19	SIP	585	Status: 200 OK |
58.059643	1692	192.168.80.20	192.168.50.19	SIP	585	Status: 200 OK |
58.101064	1701	192.168.80.20	192.168.50.19	SIP	585	Status: 200 OK |
59.323483	1908	192.168.80.20	192.168.50.19	SIP	563	Request: BYE sip:unknown@192.168.50.19:5060 |
59.325584	1910	192.168.50.19	192.168.80.20	SIP	400	Status: 200 OK |

```
> From: "7002"<sip:7002@192.168.80.20>;tag=abbda3c5
> To: "7007"<sip:7007@192.168.50.19>;tag=6c234f4a-2ec6-472c-b0cd-3c42cdbf740a
  Call-ID: 00569b3f-69bf-4b2d-8c8d-28fd347965cf
  [Generated Call-ID: 00569b3f-69bf-4b2d-8c8d-28fd347965cf]
> CSeq: 2 REFER
> Via: SIP/2.0/UDP 192.168.80.20:5060;branch=z9hG4bK61dbf5ca4275861efa3f41;rport
  Max-Forwards: 70
  Supported: replaces
  Event: refer
  User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
> Contact: <sip:7002@192.168.80.20:5060>
  Referred-By: <sip:7002@192.168.50.19>
  Refer-To: <sip:7001@192.168.50.19:5060>
```

7002 talk with 7007, 7002 blind transfer the call to 7001.

Call Transfer--Attended Transfer

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7002 talk with 7007, 7002 perform the transfer, 7002 sends REINVITE message to hold the call, call 7001, sends REFER message to transfer the call.

	No.	Source	Destination	Protocol	Length	Info
17:20:24.773151	1693	192.168.80.20	192.168.50.19	SIP/SDP	1428	Request: INVITE sip:7001@192.168.50.19 |
17:20:24.777488	1696	192.168.50.19	192.168.80.20	SIP	349	Status: 100 Trying |
17:20:24.798508	1699	192.168.50.19	192.168.80.20	SIP	543	Status: 180 Ringing |
17:20:24.875479	1707	192.168.50.19	192.168.80.20	SIP	615	Status: 180 Ringing |
17:20:26.563266	1994	192.168.50.19	192.168.80.20	SIP/SDP	922	Status: 200 OK |
17:20:26.574621	1997	192.168.80.20	192.168.50.19	SIP	835	Request: ACK sip:7001@192.168.50.19:5060 |
17:20:28.941704	2573	192.168.80.20	192.168.50.19	SIP	802	Request: REFER sip:unknown@192.168.50.19:5060, in-dialog |
17:20:28.944220	2574	192.168.50.19	192.168.80.20	SIP	632	Status: 202 Accepted |
17:20:28.944666	2575	192.168.50.19	192.168.80.20	SIP/sip…	656	Request: NOTIFY sip:7002@192.168.80.20:5060 | , with Sipfrag(SIP/2.0 100 Trying)
17:20:28.952096	2576	192.168.50.19	192.168.80.20	SIP/sip…	662	Request: NOTIFY sip:7002@192.168.80.20:5060 | , with Sipfrag(SIP/2.0 200 OK)
17:20:28.962641	2581	192.168.50.19	192.168.80.20	SIP	444	Request: BYE sip:7002@192.168.80.20:5060 |
17:20:29.016399	2590	192.168.80.20	192.168.50.19	SIP	584	Status: 200 OK |

```
Message Header
  > From: "7002"<sip:7002@192.168.80.20>;tag=dc2b02b7
  > To: "7007"<sip:7007@192.168.50.19>;tag=e5d44b6b-65e1-4fc0-bf0c-160f054e1015
    Call-ID: 6091f7d2-6ab7-4920-9532-6d27e9db346b
    [Generated Call-ID: 6091f7d2-6ab7-4920-9532-6d27e9db346b]
  > CSeq: 2 REFER
  > Via: SIP/2.0/UDP 192.168.80.20:5060;branch=z9hG4bK61dbfa5d5453bd151ae339;rport
    Max-Forwards: 70
    Supported: replaces
    Event: refer
    User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
  > Contact: <sip:7002@192.168.80.20:5060>
    Referred-By: <sip:7002@192.168.50.19>
    Refer-To: <sip:7001@192.168.50.19?Replaces=1450a8c061dbfa590-28a15180%3Bto-tag%3D40157a8d-13a0-4111-ab04-93c049565b5f%3Bfrom-tag%3Da6171f17>
    Allow: ACK,BYE,CANCEL,OPTIONS,INVITE,MESSAGE,NOTIFY,SUBSCRIBE,REFER,INFO,UPDATE
```

7002 talk with 7007, 7002 attended transfer the call to 7001.

Call Forward--Always Forward

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7007 call 7002, 7002 forward the call to 7001, 7007 talk with 7001.

Forward Reason: 7002 enabled Always Forward.

	No.	Source	Destination	Protocol	Length	Info
1-10 17:34:24.549235	117	192.168.50.19	192.168.80.20	SIP	466	Request: OPTIONS sip:7002@192.168.80.20:5060 |
1-10 17:34:24.559018	122	192.168.80.20	192.168.50.19	SIP	550	Status: 200 OK |
1-10 17:34:30.634191	1690	192.168.50.19	192.168.80.20	SIP/SDP	1188	Request: INVITE sip:7002@192.168.80.20:5060 |
1-10 17:34:30.655588	1691	192.168.80.20	192.168.50.19	SIP	510	Status: 100 Trying |
1-10 17:34:30.672229	1696	192.168.80.20	192.168.50.19	SIP	657	Status: 302 Moved Temporarily |
1-10 17:34:30.674450	1697	192.168.50.19	192.168.80.20	SIP	432	Request: ACK sip:7002@192.168.80.20:5060 |

```
> Frame 1696: 657 bytes on wire (5256 bits), 657 bytes captured (5256 bits) on interface \Device\NPF_{2AA5746E-041C-4B4A-B11C-F487EA82AE64}, id 0
> Ethernet II, Src: EASY3CAL_22:b0:21 (00:21:f2:22:b0:21), Dst: NewH3CTe_97:7a:e8 (fc:60:9b:97:7a:e8)
> Internet Protocol Version 4, Src: 192.168.80.20, Dst: 192.168.50.19
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
v Session Initiation Protocol (302)
  > Status-Line: SIP/2.0 302 Moved Temporarily
  v Message Header   7007 call 7002
    > From: "7007"<sip:7007@192.168.50.19>;tag=024fd5f7-93b3-40ae-9574-55b0abffe9e0
    > To: <sip:7002@192.168.80.20>;tag=7bba066b
      Call-ID: 672a1412-a2b7-4ece-974a-161703c32000
      [Generated Call-ID: 672a1412-a2b7-4ece-974a-161703c32000]
    > CSeq: 17634 INVITE
    > Via: SIP/2.0/UDP 192.168.50.19:5060;branch=z9hG4bKPj4c1634b1-1bd8-478e-be4f-dc5ad3367e61;rport=5060
      Supported: replaces
      User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
      Diversion: <sip:7002@192.168.80.20:5060>;reason=unconditional  7002 forward the call
    > Contact: <sip:7001@192.168.50.19:5060>  forward to 7001
      Allow: ACK,BYE,CANCEL,OPTIONS,INVITE,MESSAGE,NOTIFY,SUBSCRIBE,REFER,INFO,UPDATE
      Content-Length: 0
```

Call Forward--Busy Forward

7007 call 7002, 7002 forward the call to 7001, 7007 talks with 7001.

Forward Reason: 7002 enabled Busy Forward--7002 is on a call, or 7002 enabled DND.

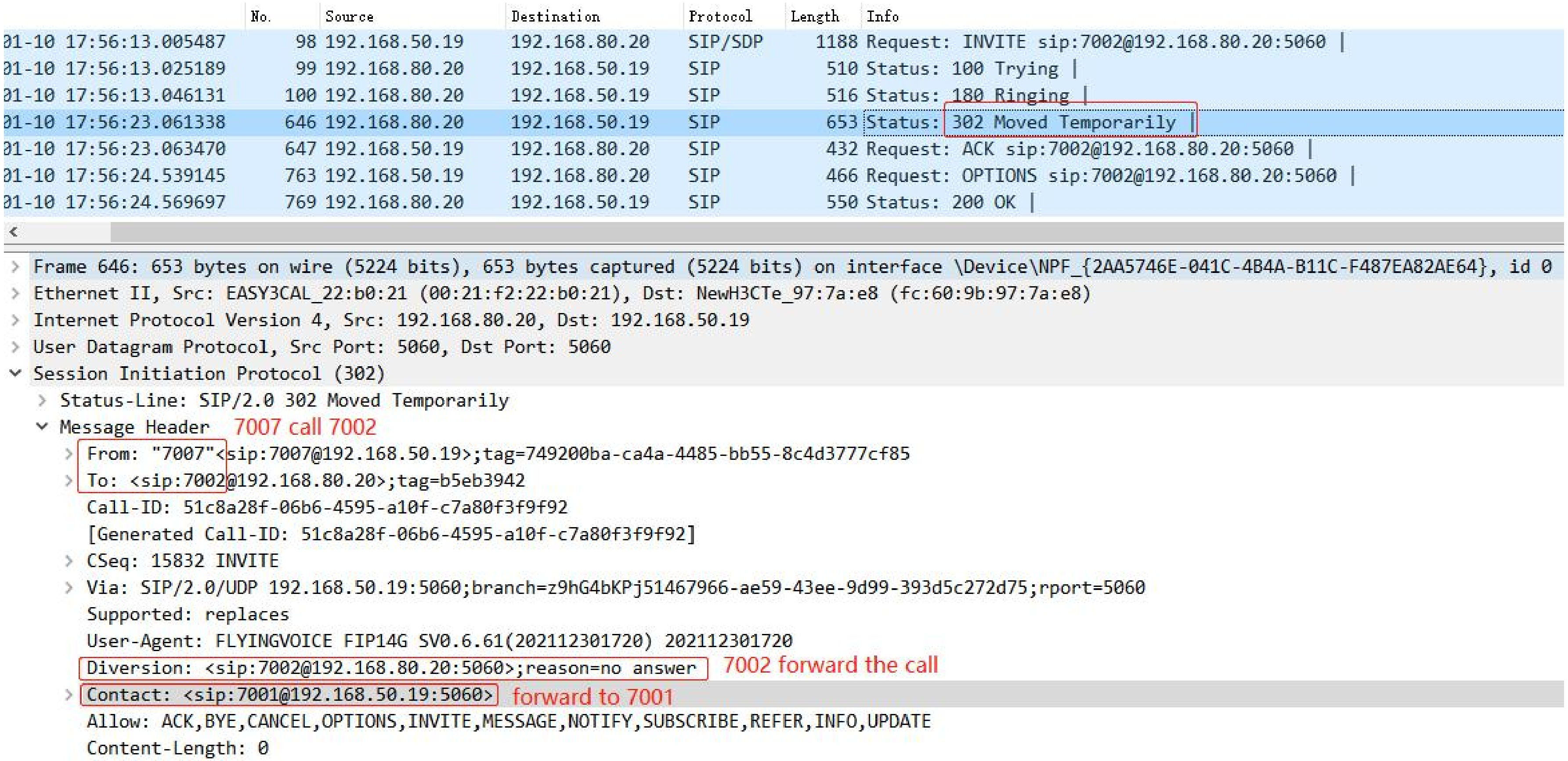
	No.	Source	Destination	Protocol	Length	Info
01-10 17:43:32.854154	752	192.168.50.19	192.168.80.20	SIP/SDP	1190	Request: INVITE sip:7002@192.168.80.20:5060 |
01-10 17:43:32.882517	753	192.168.80.20	192.168.50.19	SIP	510	Status: 100 Trying |
01-10 17:43:32.902611	754	192.168.80.20	192.168.50.19	SIP	516	Status: 180 Ringing |
01-10 17:43:37.175409	1172	192.168.80.20	192.168.50.19	SIP/SDP	1159	Status: 200 OK |
01-10 17:43:37.178219	1174	192.168.50.19	192.168.80.20	SIP	432	Request: ACK sip:7002@192.168.80.20:5060 |
01-10 17:43:39.430700	1828	192.168.50.19	192.168.80.20	SIP/SDP	1190	Request: INVITE sip:7002@192.168.80.20:5060 |
01-10 17:43:39.487699	1842	192.168.80.20	192.168.50.19	SIP	510	Status: 100 Trying |
01-10 17:43:39.533506	1859	192.168.80.20	192.168.50.19	SIP	653	Status: 302 Moved Temporarily |
01-10 17:43:39.535828	1862	192.168.50.19	192.168.80.20	SIP	432	Request: ACK sip:7002@192.168.80.20:5060 |
01-10 17:43:51.526384	5331	192.168.80.20	192.168.50.19	SIP	560	Request: BYE sip:7005@192.168.50.19:5060 |
01-10 17:43:51.528541	5332	192.168.50.19	192.168.80.20	SIP	400	Status: 200 OK |

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
v Session Initiation Protocol (302)
  > Status-Line: SIP/2.0 302 Moved Temporarily
  v Message Header   7007 call 7002
    > From: "7007"<sip:7007@192.168.50.19>;tag=c4e19a9a-ced2-42e3-9f5a-15bcde5ea08d
    > To: <sip:7002@192.168.80.20>;tag=78918ad8
      Call-ID: a237b80c-3b64-461b-819a-fef8a9379929
      [Generated Call-ID: a237b80c-3b64-461b-819a-fef8a9379929]
    > CSeq: 15747 INVITE
    > Via: SIP/2.0/UDP 192.168.50.19:5060;branch=z9hG4bKPjeede5dd2-07d2-4e09-a931-a67749e042ac;rport=5060
      Supported: replaces
      User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
      Diversion: <sip:7002@192.168.80.20:5060>;reason=user-busy   7002 forward the call because 7002 is busy
    > Contact: <sip:7001@192.168.50.19:5060>   forward to 7001
      Allow: ACK,BYE,CANCEL,OPTIONS,INVITE,MESSAGE,NOTIFY,SUBSCRIBE,REFER,INFO,UPDATE
```

Call Forward--No Answer Forward

7007 call 7002, 7002 forward the call to 7001, 7007 talks with 7001.

Forward Reason: 7002 enabled No Answer Forward, 7002 didn't answer the call.



	No.	Source	Destination	Protocol	Length	Info
01-10 17:56:13.005487	98	192.168.50.19	192.168.80.20	SIP/SDP	1188	Request: INVITE sip:7002@192.168.80.20:5060 |
01-10 17:56:13.025189	99	192.168.80.20	192.168.50.19	SIP	510	Status: 100 Trying |
01-10 17:56:13.046131	100	192.168.80.20	192.168.50.19	SIP	516	Status: 180 Ringing |
01-10 17:56:23.061338	646	192.168.80.20	192.168.50.19	SIP	653	Status: 302 Moved Temporarily |
01-10 17:56:23.063470	647	192.168.50.19	192.168.80.20	SIP	432	Request: ACK sip:7002@192.168.80.20:5060 |
01-10 17:56:24.539145	763	192.168.50.19	192.168.80.20	SIP	466	Request: OPTIONS sip:7002@192.168.80.20:5060 |
01-10 17:56:24.569697	769	192.168.80.20	192.168.50.19	SIP	550	Status: 200 OK |

```
> Frame 646: 653 bytes on wire (5224 bits), 653 bytes captured (5224 bits) on interface \Device\NPF_{2AA5746E-041C-4B4A-B11C-F487EA82AE64}, id 0
> Ethernet II, Src: EASY3CAL_22:b0:21 (00:21:f2:22:b0:21), Dst: NewH3CTe_97:7a:e8 (fc:60:9b:97:7a:e8)
> Internet Protocol Version 4, Src: 192.168.80.20, Dst: 192.168.50.19
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
v Session Initiation Protocol (302)
  > Status-Line: SIP/2.0 302 Moved Temporarily
  v Message Header   7007 call 7002
    > From: "7007"<sip:7007@192.168.50.19>;tag=749200ba-ca4a-4485-bb55-8c4d3777cf85
    > To: <sip:7002@192.168.80.20>;tag=b5eb3942
      Call-ID: 51c8a28f-06b6-4595-a10f-c7a80f3f9f92
      [Generated Call-ID: 51c8a28f-06b6-4595-a10f-c7a80f3f9f92]
    > CSeq: 15832 INVITE
    > Via: SIP/2.0/UDP 192.168.50.19:5060;branch=z9hG4bKPj51467966-ae59-43ee-9d99-393d5c272d75;rport=5060
      Supported: replaces
      User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
      Diversion: <sip:7002@192.168.80.20:5060>;reason=no answer   7002 forward the call
    > Contact: <sip:7001@192.168.50.19:5060>   forward to 7001
      Allow: ACK,BYE,CANCEL,OPTIONS,INVITE,MESSAGE,NOTIFY,SUBSCRIBE,REFER,INFO,UPDATE
      Content-Length: 0
```

SIP Call--DND

7007 call 7002, can't get through, get error 480.

Reason: 7002 enabled DND.

	No.	Source	Destination	Protocol	Length	Info
-10 18:07:57.995929	130	192.168.50.19	192.168.80.20	SIP/SDP	1189	Request: INVITE sip:7002@192.168.80.20:5060 |
-10 18:07:58.015613	132	192.168.80.20	192.168.50.19	SIP	509	Status: 100 Trying |
-10 18:07:58.034581	133	192.168.80.20	192.168.50.19	SIP	515	Status: 180 Ringing |
-10 18:07:59.077975	212	192.168.80.20	192.168.50.19	SIP	588	Status: 480 Temporarily Unavailible |
-10 18:07:59.080024	213	192.168.50.19	192.168.80.20	SIP	431	Request: ACK sip:7002@192.168.80.20:5060 |

```
> Frame 212: 588 bytes on wire (4704 bits), 588 bytes captured (4704 bits) on interface \Device\NPF_{2AA5746E-041C-4B4A-B11C-F487EA82AE64}, id 0
> Ethernet II, Src: EASY3CAL_22:b0:21 (00:21:f2:22:b0:21), Dst: NewH3CTe_97:7a:e8 (fc:60:9b:97:7a:e8)
> Internet Protocol Version 4, Src: 192.168.80.20, Dst: 192.168.50.19
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
v Session Initiation Protocol (480)
  > Status-Line: SIP/2.0 480 Temporarily Unavailible
  v Message Header
    > From: "7007"<sip:7007@192.168.50.19>;tag=4249337f-4cbc-4a22-a94c-0d45c0e3b66e
    > To: <sip:7002@192.168.80.20>;tag=9366bd22
      Call-ID: 690bd6b7-90b9-4742-b955-4733ebbad5a1
      [Generated Call-ID: 690bd6b7-90b9-4742-b955-4733ebbad5a1]
    > CSeq: 4794 INVITE
    > Via: SIP/2.0/UDP 192.168.50.19:5060;branch=z9hG4bKPj6e3998f1-b15a-4ead-9014-5855e3506e03;rport=5060
      User-Agent: FLYINGVOICE FIP14G SV0.6.61(202112301720) 202112301720
    > Reason: Q.850;Cause=18;text="No User Responding"
      Allow: ACK,BYE,CANCEL,OPTIONS,INVITE,MESSAGE,NOTIFY,SUBSCRIBE,REFER,INFO,UPDATE
      Content-Length: 0
```

Auto Provisioning

What is Auto Provisioning?

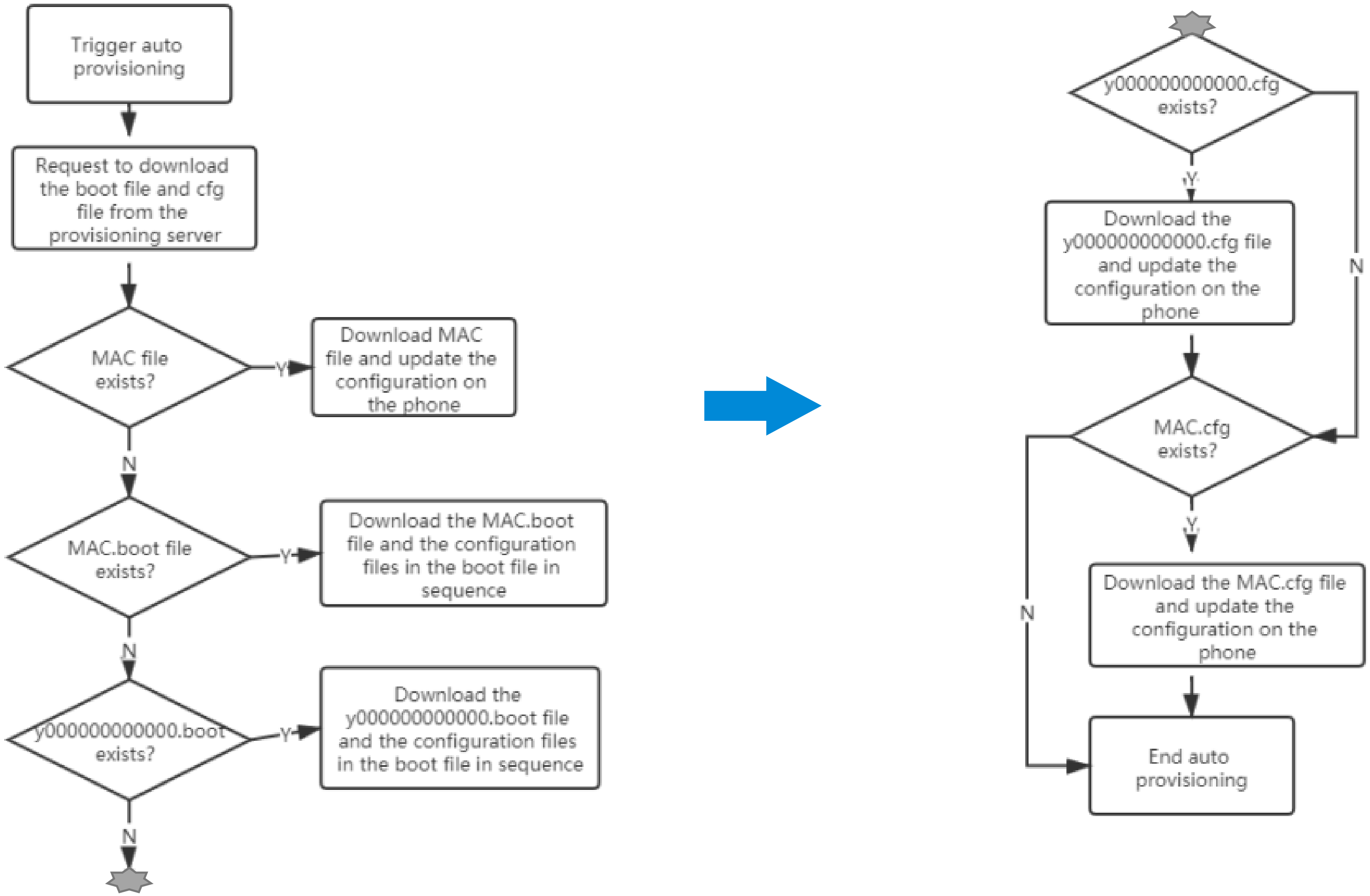
Auto-provisioning is a method to update the phone settings or upgrade the firmware automatically. Allows users to deploy the phones without having to do the configuration setting manually. The phone can obtain the configurations from the server once it has been powered on.

Advantages

With auto-provisioning, a large number of phones can be remotely configured and upgraded concurrently, saving time and labor.

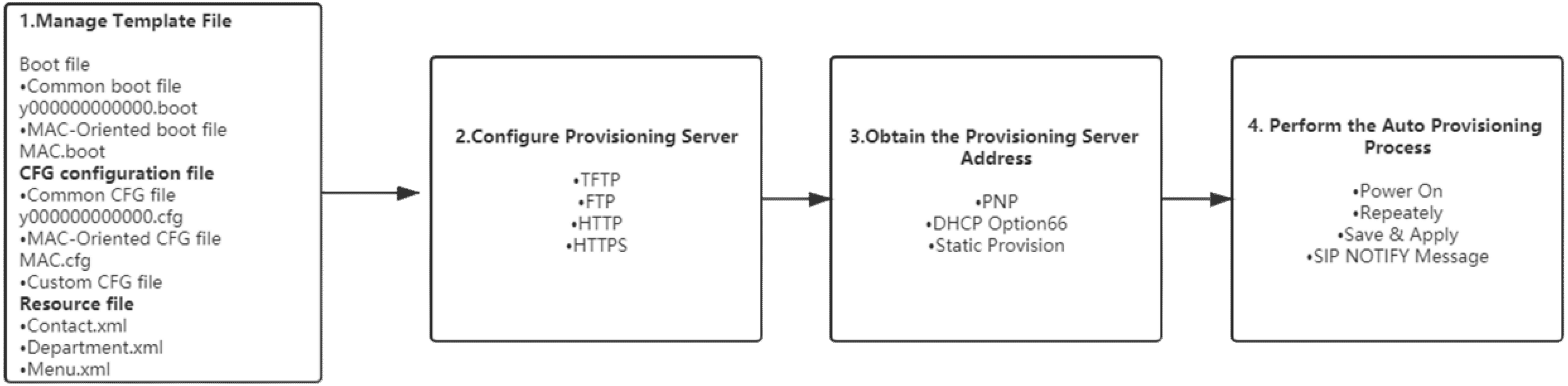
Auto Provisioning Process

FLYINGVOICE



Steps for Auto Provisioning

There are 4 major steps to do auto-provisioning.



Step1-Manage Template File

Boot Files

The phone tries to download the boot file first, then download the configuration files referenced in the boot file during auto-provisioning.

Flyingvoice supports the following two types of boot files:

- **MAC-Oriented boot file** (for example 0021f2000001.boot)
- **Common boot file**(y000000000000.boot)

y000000000000.boot

```
include:config <time.cfg>
include:config "xxx.cfg"
```

Step1-Manage Template File

Configuration Files

The phone supports the following two types of boot files:

➢ **MAC-Oriented cfg file** (for example 0021f2000001.cfg)

The file contains configuration parameters that are expected to be updated per phone, such as the registration information.

➢ **Common cfg file**(y000000000000.cfg)

The common CFG includes the configuration parameters for all phones, such as firmware upgrade, phonebook, and volume setting.

```
mac.cfg
##account.X.*:(FIP10(P)/FIP12WP/FIP16: X ranges from 1 to 2. FIP11C(P): X ranges f
##################################################################################
##                              Account Register                                ##
##################################################################################
#Enable or disable the account1, 0-Disabled (default), 1-Enabled;
account.1.enable=

#Configure the label displayed on the LCD screen for account1.
account.1.label=

#Configure the display name of account1.
account.1.display_name=

#Configure the username and password for register authentication.
account.1.auth_name=
account.1.password=
```

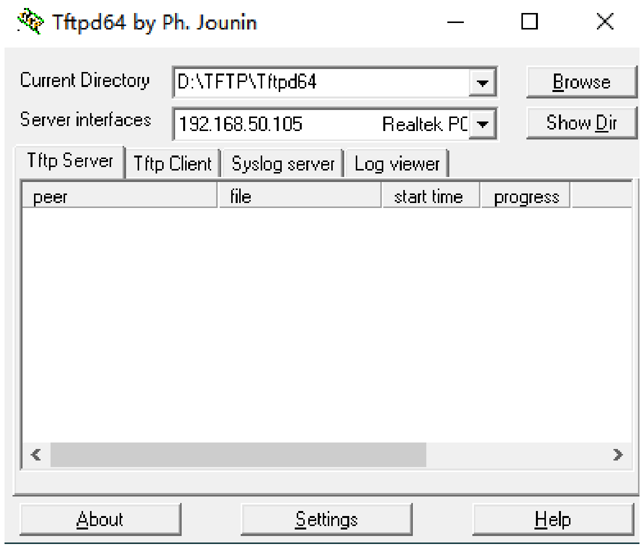
MAC cfg

```
y000000000000.cfg
##################################################################################
##                              WEB Advanced                                    ##
##################################################################################
wui.http_enable=
wui.https_enable=

##security.user_password=
security.user_password=

##################################################################################
##                              Language                                        ##
##################################################################################
##Support language: English, Chinese_S, Chinese_T, French, German, Italian, Polish,
lang.gui=
```

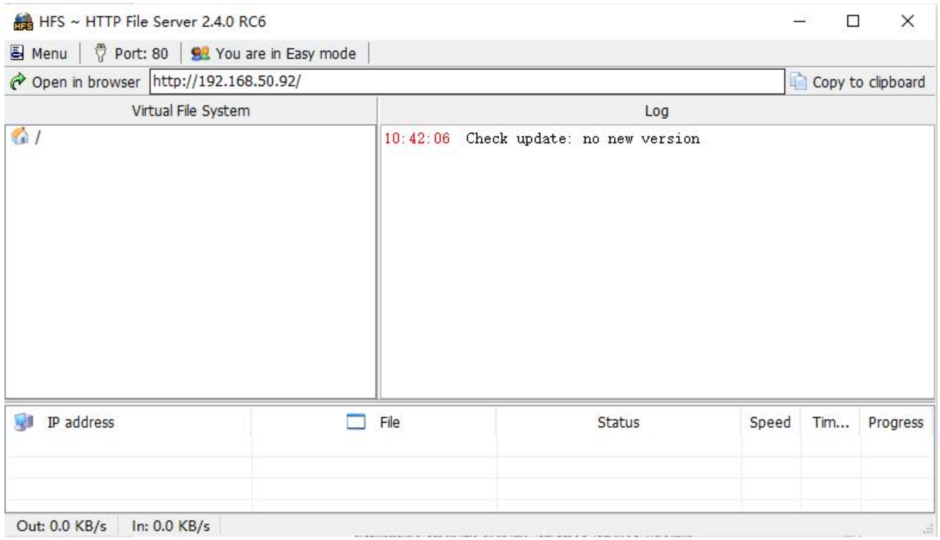
Common cfg

Step2-Configure a Provisioning Server

The phones support using FTP, TFTP, HTTP, and HTTPS protocols to download boot files and configuration files. You can use one of these protocols for provisioning.



TFTP server(TFTPD64)

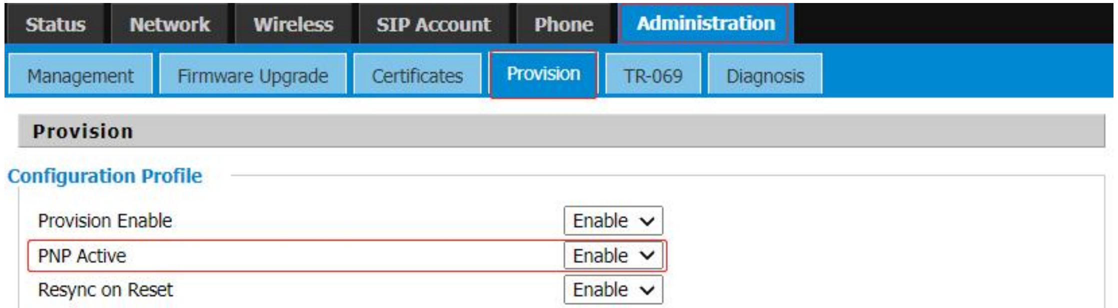


HTTP server(HFS)

Step3-Obtain the Provisioning Server Address

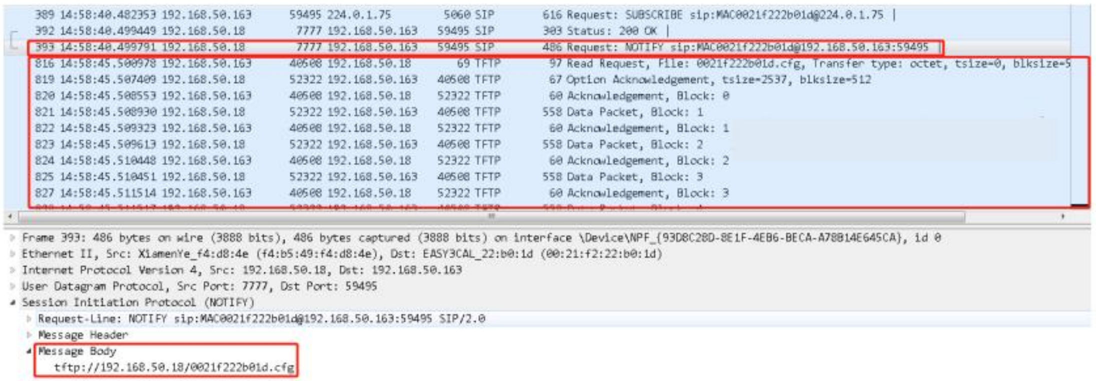
◆ **Plug and Play (PnP)** Server

The phones support obtaining the provisioning server address from the PnP server



The phone sends the PnP SUBSCRIBE message to the broadcast address 224.0.1.75 to obtain the provisioning server address during startup.

The following figure indicates the phone obtain the provision server's address from the NOTIFY message from the PNP server.



◆ DHCP Option66

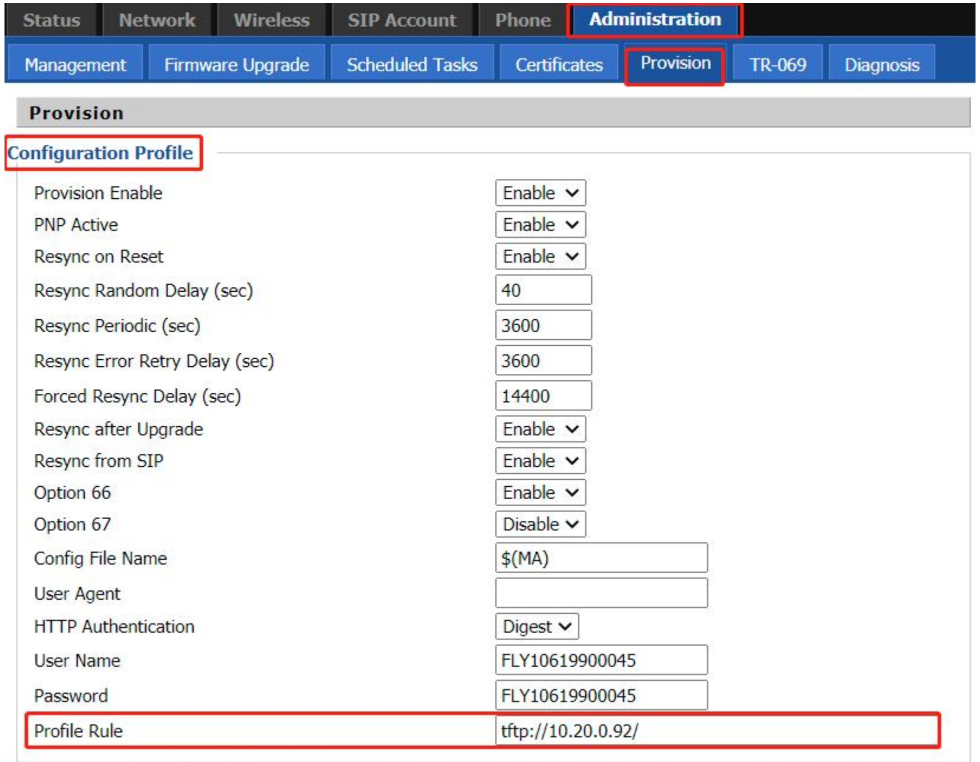
The phones can obtain the provisioning server address by detecting DHCP option 66 during startup. The following figure indicates the phone obtain the provision server's address by detecting DHCP option 66

```
485 14:29:06.842740 0.0.0.0            68 255.255.255.255    67 DHCP      590 DHCP Discover - Transaction ID 0xf8448829
489 14:29:06.876257 192.168.50.92      67 255.255.255.255    68 DHCP      343 DHCP Offer    - Transaction ID 0xf8448829
490 14:29:06.879025 0.0.0.0            68 255.255.255.255    67 DHCP      590 DHCP Request  - Transaction ID 0xf8448829
491 14:29:06.911088 192.168.50.92      67 255.255.255.255    68 DHCP      343 DHCP ACK      - Transaction ID 0xf8448829
573 14:29:30.719753 192.168.50.4    42952 192.168.50.92      69 TFTP       93 Read Request, File: 0021f222b01d, Transfer type: octet, tsize=0, blksize=512, timeout=
580 14:29:34.954594 192.168.50.4    42952 192.168.50.92      69 TFTP       93 Read Request, File: 0021f222b01d, Transfer type: octet, tsize=0, blksize=512, timeout=
581 14:29:34.956266 192.168.50.92   64902 192.168.50.4    42952 TFTP       62 Error Code, Code: File not found, Message: File not found
583 14:29:35.321573 192.168.50.4    54546 192.168.50.92      69 TFTP       98 Read Request, File: 0021f222b01d.boot, Transfer type: octet, tsize=0, blksize=512, ti
584 14:29:35.322920 192.168.50.92   64903 192.168.50.4    54546 TFTP       62 Error Code, Code: File not found, Message: File not found
585 14:29:35.518512 192.168.50.4    52506 192.168.50.92      69 TFTP       99 Read Request, File: y000000000000.boot, Transfer type: octet, tsize=0, blksize=512, ti
586 14:29:35.520433 192.168.50.92   64904 192.168.50.4    52506 TFTP       62 Error Code, Code: File not found, Message: File not found
587 14:29:35.676215 192.168.50.4    55823 192.168.50.92      69 TFTP       98 Read Request, File: y000000000000.cfg, Transfer type: octet, tsize=0, blksize=512, ti
588 14:29:35.677966 192.168.50.92   64905 192.168.50.4    55823 TFTP      116 Data Packet, Block: 1 (last)

  Server host name not given
  Boot file name not given
  Magic cookie: DHCP
▷ Option: (53) DHCP Message Type (ACK)
▷ Option: (54) DHCP Server Identifier (192.168.50.2)
▷ Option: (1) Subnet Mask (255.255.255.0)
▷ Option: (3) Router
▷ Option: (6) Domain Name Server
▷ Option: (51) IP Address Lease Time
▷ Option: (58) Renewal Time Value
▷ Option: (59) Rebinding Time Value
▷ Option: (66) TFTP Server Name
▷ Option: (255) End
```

Step3-Obtain the Provisioning Server Address

◆ **Static Provision**

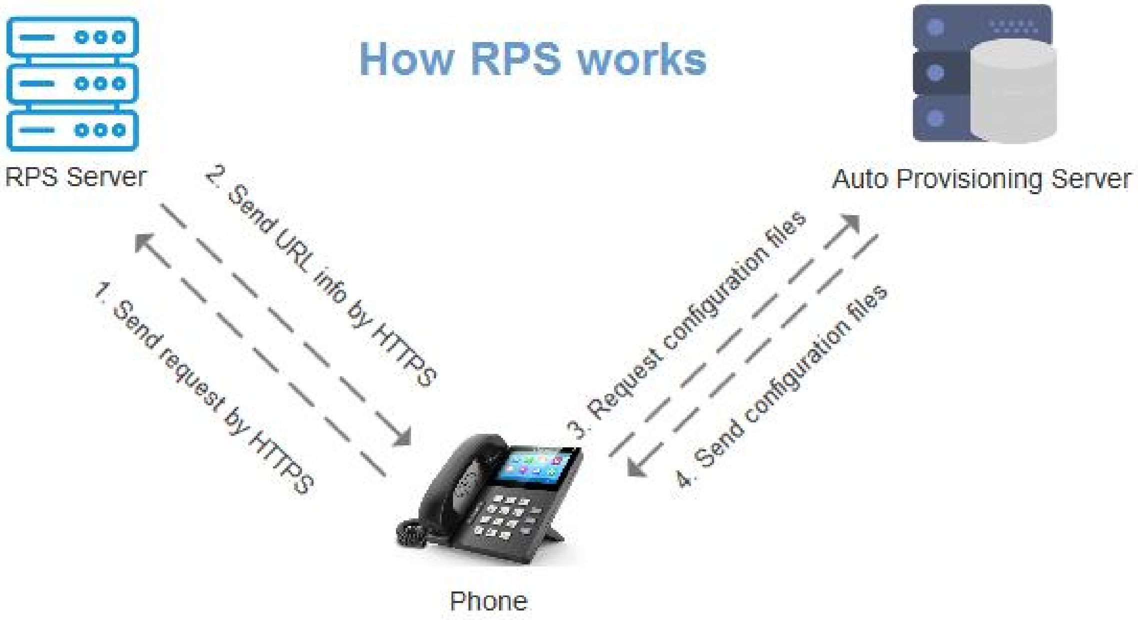
The phones can obtain the provisioning server address by static provision. Go to **Administration** > **Provision** > **Configuration Profile**, type in the access URL of the provisioning server in the Profile Rule field.



Step3-Obtain the Provisioning Server Address

◆ **RPS**

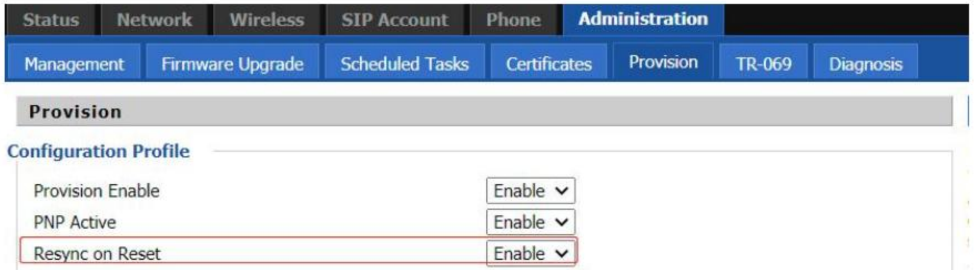
The phones can obtain the provisioning server address from the RPS server.



Step4-Triggering the Phone to Perform Auto Provision

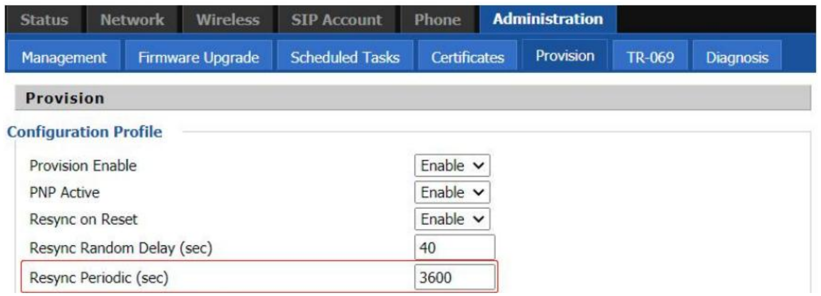
- **Power ON**

The IP phone performs the auto provisioning when the IP phone is powered on.



- **Repeatedly**

The IP phone performs the auto provisioning at regular intervals. You can configure the interval for the repeatedly mode. The default interval is 3600 seconds



Step4-Triggering the Phone to Perform Auto Provision

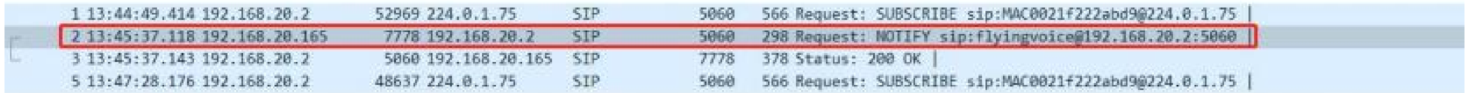
➢ **Save & Apply**

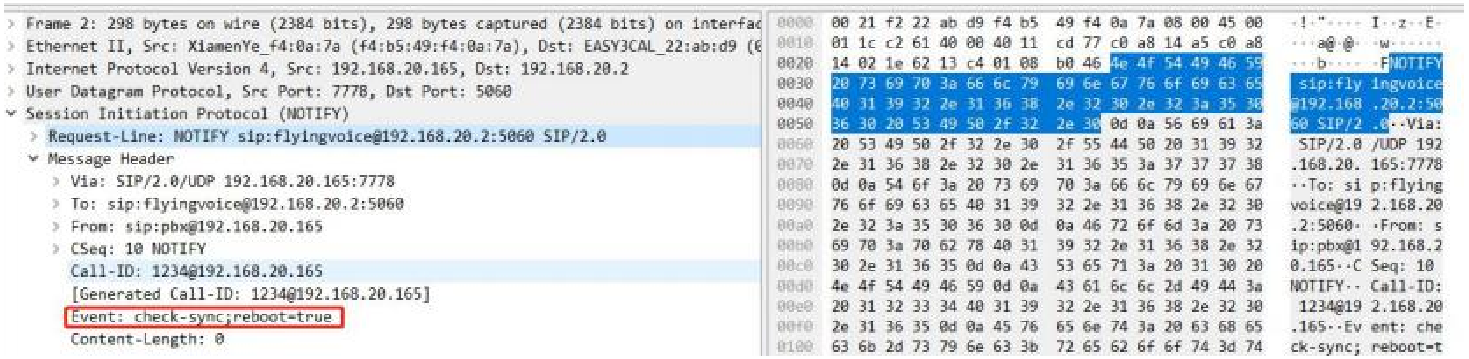
The IP phone performs the auto provisioning when you click Save & Apply on the provision page.

➢ **SIP NOTIFY Message**

The IP phone will perform auto provisioning when receiving a SIP NOTIFY message which contains the header “Event: check-sync”.

Whether the IP phone reboots or not depends on the value of the parameter “reboot = true”. If the header of the SIP NOTIFY message contains this string “ reboot=true ”, the IP phone will reboot immediately.





The phone doesn't apply the auto provisioning settings

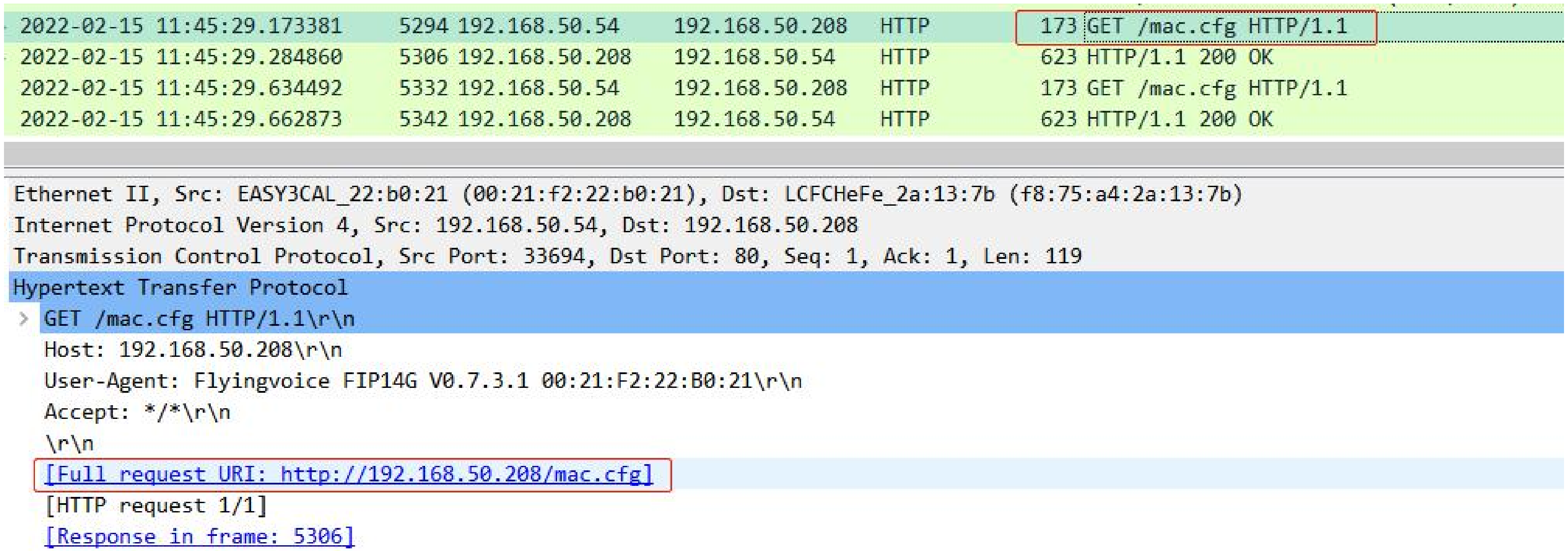
1. Check the phone's system log. Check the provisioning URL and provisioning result. Make sure that the phone has not been provisioned by other servers before.

```
<Tue Feb 15 17:54:14 2022> provision[3939]: provision v1.2 start
<Tue Feb 15 17:54:14 2022> provision[3939]: adapter_yphone_enable = 0
<Tue Feb 15 17:54:14 2022> goahead[3761]: webs start https on port <443>!
<Tue Feb 15 17:54:17 2022> provision[3939]: SIPPNP_RECV_MSG
<Tue Feb 15 17:54:17 2022> provision[3939]: Profile Rule [tftp://192.168.50.165/0021f222b021.cfg]
<Tue Feb 15 17:54:18 2022> lldpd[1243]: MSAP has changed for port eth0.1, sending a shutdown LLDPDU
<Tue Feb 15 17:54:19 2022> provision[3939]: Profile Rule [tftp://192.168.50.165/0021f222b021.cfg]...
<Tue Feb 15 17:54:20 2022> admin: [config_manager.sh]Update the configuration file successfully!
<Tue Feb 15 17:54:20 2022> admin: [config_manager.sh]Update the configuration file successfully!
<Tue Feb 15 17:54:20 2022> admin: [config_manager.sh]Update the configuration file successfully!
<Tue Feb 15 17:54:20 2022> provision[3939]: Update other configuration...
<Tue Feb 15 17:54:21 2022> admin: [config_manager.sh]Update the configuration file successfully!
<Tue Feb 15 17:54:21 2022> provision[3939]: Update the configuration file successfully....
<Tue Feb 15 17:55:51 2022> ipphone[3386]: ***system booting***
<Tue Feb 15 17:55:52 2022> ipphone[3386]: SW:142(120106174008)
```

Auto Provisioning-Troubleshooting

2. Check auto provisioning URL and network environment.

If provisioning server address is domain name, please make sure DNS works normally so phone can resolve IP address correctly.



Use HTTP auto provisioning server

Auto Provisioning-Troubleshooting

Source	Destination	Protocol	Length	Info
192.168.50.54	192.168.50.165	TFTP	97	Read Request, File: 0021f222b021.cfg, Transfer type: octet, tsize=0, blksize=512, timeou…
192.168.50.165	192.168.50.54	TFTP	67	Option Acknowledgement, tsize=2536, blksize=512
192.168.50.54	192.168.50.165	TFTP	60	Acknowledgement, Block: 0
192.168.50.165	192.168.50.54	TFTP	558	Data Packet, Block: 1
192.168.50.54	192.168.50.165	TFTP	60	Acknowledgement, Block: 1
192.168.50.165	192.168.50.54	TFTP	558	Data Packet, Block: 2
192.168.50.54	192.168.50.165	TFTP	60	Acknowledgement, Block: 2
192.168.50.165	192.168.50.54	TFTP	558	Data Packet, Block: 3
192.168.50.54	192.168.50.165	TFTP	60	Acknowledgement, Block: 3
192.168.50.165	192.168.50.54	TFTP	558	Data Packet, Block: 4
192.168.50.54	192.168.50.165	TFTP	60	Acknowledgement, Block: 4
192.168.50.165	192.168.50.54	TFTP	534	Data Packet, Block: 5 (last)

Use TFTP auto provisioning server

3. Check the configuration file.

Make sure the parameter value and format are correct in the configuration file.

4. Contact Flyingvoice FAE for help. Provide the details below.

① The provisioning configuration file.

② The phone's system log.

③ The phone's configuration files (before and after provisioning).

④ The packet trace.

BLF(Busy Lamp Field)

Subscribe Process

1. The subscriber (IP Phone) sends a SUBSCRIBE message to the SIP server to monitor the extension.
2. The SIP server sends back a 200 OK once it successfully processed. (If authentication is configured, authentication takes place and if the subscriber is successfully authenticated a 200 OK SIP message response is sent back to the subscriber)

Notify Process

1. The SIP server sends a NOTIFY message including XML body to the subscriber informing the subscriber of the status of the monitored extension.
2. The subscriber sends back a 200 OK message to the SIP server.

Source	Destination	Protocol	Length	Info
192.168.80.23	192.168.50.19	SIP	822	Request: SUBSCRIBE sip:7007@192.168.50.19 |
192.168.50.19	192.168.80.23	SIP	619	Status: 200 OK |
192.168.50.19	192.168.80.23	SIP	868	Request: NOTIFY sip:7002@192.168.80.23:5060 |
192.168.80.23	192.168.50.19	SIP	550	Status: 200 OK |

Note:

Subscriber: 192.168.80.23

Server: 192.168.50.19

Subscribe Pcap Trace

	No.	Source	Destination	Protocol	Length	Info
13 15:05:36.307209	4351	192.168.80.23	192.168.50.19	SIP	825	Request: SUBSCRIBE sip:7007@192.168.50.19 |
13 15:05:36.311126	4352	192.168.50.19	192.168.80.23	SIP	621	Status: 200 OK |
13 15:05:36.312820	4353	192.168.50.19	192.168.80.23	SIP	870	Request: NOTIFY sip:7002@192.168.80.23:5060 |
13 15:05:36.331583	4356	192.168.80.23	192.168.50.19	SIP	552	Status: 200 OK |

```
> Frame 4351: 825 bytes on wire (6600 bits), 825 bytes captured (6600 bits) on interface \Device\NPF_{2AA5746E-041C-4B4A-B11C-F487EA82AE64}, id 0
> Ethernet II, Src: EASY3CAL_23:8a:d5 (00:21:f2:23:8a:d5), Dst: NewH3CTe_97:7a:e8 (fc:60:9b:97:7a:e8)
> Internet Protocol Version 4, Src: 192.168.80.23, Dst: 192.168.50.19
> User Datagram Protocol, Src Port: 5060, Dst Port: 5060
v Session Initiation Protocol (SUBSCRIBE)
  > Request-Line: SUBSCRIBE sip:7007@192.168.50.19 SIP/2.0 The monitored extension
  v Message Header
    > From: <sip:7002@192.168.50.19>;tag=d939dbf1 The subscriber
    > To: <sip:7007@192.168.50.19>
      Call-ID: a0505c-0-13c4-61dfcf42-10cc-1c956679@192.168.50.19
      [Generated Call-ID: a0505c-0-13c4-61dfcf42-10cc-1c956679@192.168.50.19]
    > CSeq: 2 SUBSCRIBE
    > Via: SIP/2.0/UDP 192.168.80.23:5060;branch=z9hG4bK61dfcf4210dd672f09ca;rport
      Expires: 3600 The Expires header, where the duration of the subscription in seconds is stated
      Event: dialog
      Max-Forwards: 70
      Supported: 100rel,replaces,timer
```

The subscriber can unsubscribe the extension by sending a SUBSCRIBE message to the PBX server with **Expires header** set to **0**.

Notify Pcap Trace

	No.	Source	Destination	Protocol	Length	Info
:41:18.608468	24229	192.168.80.23	192.168.50.19	SIP	822	Request: SUBSCRIBE sip:7007@192.168.50.19 |
:41:18.612217	24230	192.168.50.19	192.168.80.23	SIP	619	Status: 200 OK |
:41:18.614118	24232	192.168.50.19	192.168.80.23	SIP	868	Request: NOTIFY sip:7002@192.168.80.23:5060 |
:41:18.632889	24233	192.168.80.23	192.168.50.19	SIP	550	Status: 200 OK |

```
Session Initiation Protocol (NOTIFY)
> Request-Line: NOTIFY sip:7002@192.168.80.23:5060 SIP/2.0
v Message Header
```

The sip notify message send to this subscriber

```
  > Via: SIP/2.0/UDP 192.168.50.19:5060;rport;branch=z9hG4bKPj23c0d0c8-449e-42db-9ce4-8fc875a211a6
  > From: <sip:7007@192.168.50.19>;tag=83cb4319-ab5c-44b8-bc39-f1013c2505df
  > To: <sip:7002@192.168.50.19>;tag=cc953a19
  > Contact: <sip:192.168.50.19:5060>
    Call-ID: a051cc-0-13c4-61de861e-f95-7f787c84@192.168.50.19
    [Generated Call-ID: a051cc-0-13c4-61de861e-f95-7f787c84@192.168.50.19]
  > CSeq: 8285 NOTIFY
    Event: dialog
```

The subscription state and how many seconds are left for subscription to expire are stated.

```
    Subscription-State: active;expires=3599
    Allow-Events: message-summary, presence, dialog, refer
    Max-Forwards: 70
    User-Agent: Yeastar P550
    Content-Type: application/dialog-info+xml
    Content-Length:   232
v Message Body
```

The monitored extension

```
    <?xml version="1.0" encoding="UTF-8"?>\n
    <dialog-info xmlns="urn:ietf:params:xml:ns:dialog-info" version="0" state="full" entity="sip:7007@192.168.50.19:5060">\n
     <dialog id="7007">\n
```

Phone status, terminated means it is available, confirmed means it is on a call, early means it is ringing.

```
      <state>terminated</state>\n
```

BLF--Troubleshooting

BLF Icons and Status

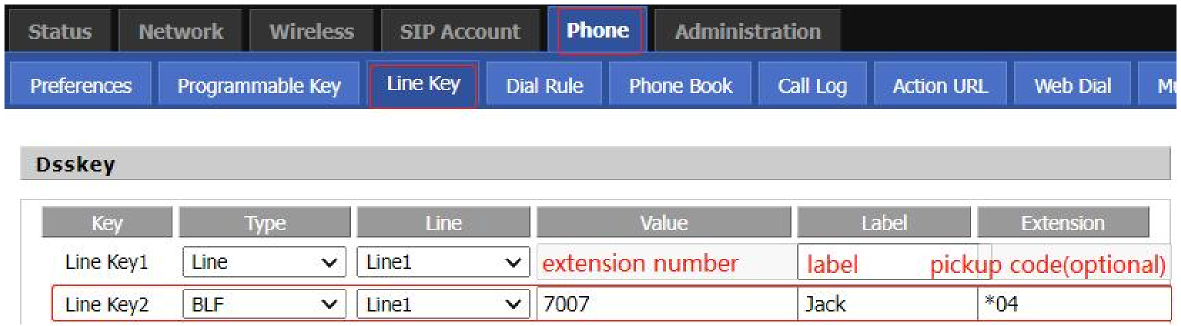
Status of the monitored phone	BLF Key Status	BLF Icon
Available	Steady Green	
Ringing	Flashing Red	
Busy	Steady Red	

BLF subscribe failed

1. Make sure the extension number you monitored is registered successfully. It is in the same PBX server as the Line you choose for the BLF setting.

2. Make sure the SIP server supports BLF.

3. Check the BLF settings.



4. Check package trace.

403 – Forbidden. Server did not authorize the request.

423 – Interval to small. The Expires value specified in the SIP message is too short.

481 – Subscription does not exist.

489 – Bad event. The event package designated is not supported.

5. Check whether there is any other Models (brand) works fine in the same scenario with same account.

6. Contact Flyingvoice FAE for help. Provide the details below.

(1) Describe the detailed steps to reproduce the issue. E.g. A monitor B, C call B, A's BLF light is solid green.

(2) Send us the following debug file to check.

①The phone's configuration file.

②The phone's packet trace. (If another phone can subscribe successfully, get a pcap trace on it to compare, send us the OK/NOK trace.)

Please confirm the file contain effective information, after filtering sip, you can see trace like below.

sip

	No.	Source	Destination	Protocol	Length	Info
-13 15:05:36.307209	4351	192.168.80.23	192.168.50.19	SIP	825	Request: SUBSCRIBE sip:7007@192.168.50.19 |
-13 15:05:36.311126	4352	192.168.50.19	192.168.80.23	SIP	621	Status: 200 OK |
-13 15:05:36.312820	4353	192.168.50.19	192.168.80.23	SIP	870	Request: NOTIFY sip:7002@192.168.80.23:5060 |
-13 15:05:36.331583	4356	192.168.80.23	192.168.50.19	SIP	552	Status: 200 OK |

Network

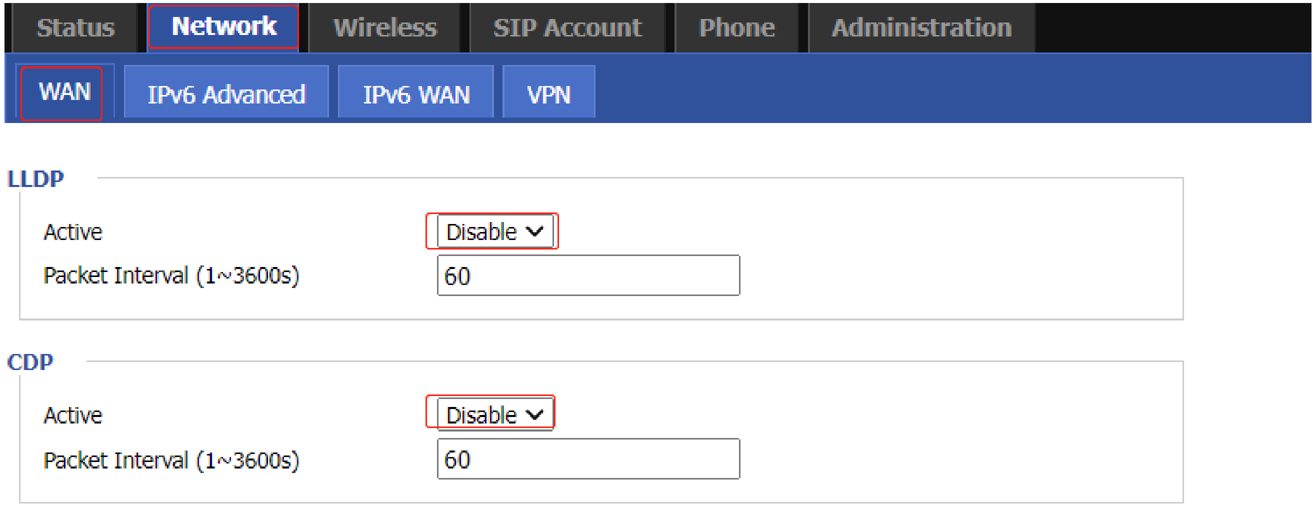
Q: Configured a VLAN ID on phone manually, but it can't obtain the correct IP from this VLAN.

A: Flyingvoice phone has three ways to obtain VLAN ID , the priority is LLDP>manually VLAN >DHCP Option

1. Disable LLDP and CDP and check whether the phone can get IP address.

2. Check whether there is any other Models (brand) works fine in the same scenario.

3. Get a packet trace with Wireshark. Contact Flyingvoice FAE for help, send us the phone's configuration file and the packet trace. (If another phone works, get a pcap trace on it to compare, send us the OK/NOK trace)



Q: Set up LLDP or CDP on the phone, but it can't obtain the VLAN from LLDP or CDP.

A: Get a packet trace with Wireshark. Check whether the LLDP packet has included the VLAN information.

If there is no VLAN information, check the setting of the Switch.

If it has the VLAN information, do the following.

1. Make clear the customer's network topology(model number and connection).
2. Disable LLDP/CDP, set up the VLAN on the phone manually and check whether it works.
3. Check whether there is any other Models (brand) works fine in the same scenario.
4. Contact Flyingvoice FAE for help, send us the phone's configuration file and the packet trace. (If another phone works, get a pcap trace on it to compare, send us the OK/NOK trace)

Q: Phone can't obtain the IP address.

1. Restart/Reset the phone.

2. Connect this Network cable to other phones, check whether other phones work.

3. Set up a static IP address on the phone.

4. Check whether the customer's network environment has VLAN, if not, disable LLDP/CDP have a try. If there is VLAN in the environment, check whether the phone set up the correct VLAN settings.

5. Get a packet trace with Wireshark to check the DHCP progress.

Phone become Slow/Stuck/ Reboot

Phone becomes Slow/Stuck or Reboot--Troubleshooting

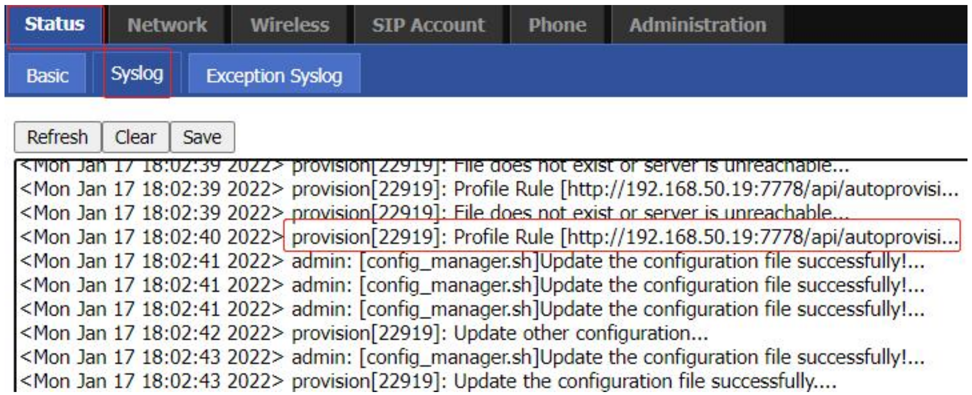
When the phone becomes slow or stuck or sometimes will reboot automatically.

1. Check whether the phone has used correct power, if the customer use POE, try to test with the power adapter.

5V/1A adapter: FIP10(P), FIP11C(P), FIP12WP, FIP13G, FIP14G, FIP16

5V/2A adapter: FIP15G, FIP16Plus

2. Check whether the phone has set up auto-provisioning, the phone will reboot when updating the configurations from the server.



3. Check whether the phone has enabled a feature but the server disabled this feature, causing the phone to continue to resend the related information to the server.

4. If the phone just reboots automatically without being slow, please check whether the phone has received Reboot information, like below.

5. Upgrade the phone's firmware version to the latest, then factory reset the phone and test again.

6. Check whether there are any other Models (brand) work fine in the same scenario.

7. Contact Flyingvoice FAE for help. Provide the details of the issue scenario.

- In what scenario the phone will become slow, stuck, restart? For example, the phone has configured with LDAP, remote phonebook, BLF, XML browser?
- Describe specific operations and problem symptoms. Please take a video if it is complicated to describe clearly.

Phone becomes Slow/Stuck or Reboot--Troubleshooting

FLYINGVOICE

- How often the issue happen? How many phones customers have? How many phones have the issue?
- Does the customer do any special settings or operations before the issue happens? Try to find out the probabilistic.
- If the phone is slow or stuck, check whether reboot the phone can fix the issue, check whether the web interface is still accessible.
- Provide us the following information for debug.

① Phone's configuration file.

② Phone's system log.(Syslog and Exception Syslog)

③ Pcap trace(better to have).

Thank You!

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