Server Redundancy on Yealink IP Phones

This guide provides detailed information on how to configure and use server redundancy on Yealink IP phones.

This information applies to the following Yealink IP phones:

- SIP VP-T49G, SIP-T48G, SIP-T46G, SIP-T42G, SIP-T41P, SIP-T40P, SIP-T29G, SIP-T27P, SIP-T23P, SIP-T23G, SIP-T21(P) E2, SIP-T19(P) E2, CP860 and W56P IP phones running firmware version 80 or later
- SIP-T48S, SIP-T46S, SIP-T42S, SIP-T41S and SIP-T27G IP phones running firmware version 81 or later

Introduction

Server redundancy is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

Two types of server redundancy are possible. In some cases, a combination of the two may be used:

- **Failover:** In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down or off-line. This mode of operation should be done using the DNS mechanism from the primary to the secondary server.
- **Fallback:** In this mode, there are two types of the registration modes: Concurrent registration and Successive registration. IP phones support configurations of two servers per SIP registration for this purpose. For more information on two registration modes, refer to Phone Registration on page 3.

Glossary

The following terms may assist in understanding server redundancy feature:

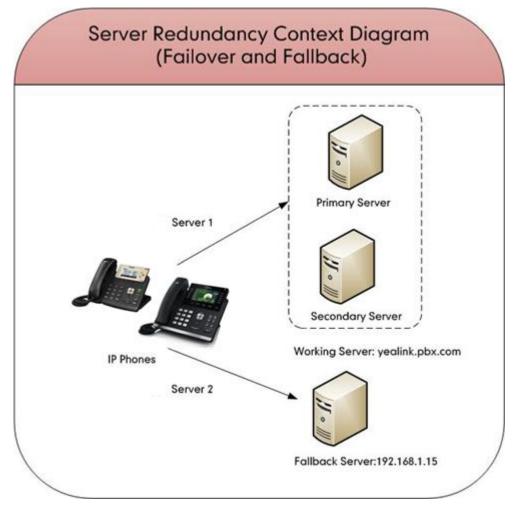
Working and Fallback Servers: The working and fallback servers are two separate servers used for per line registration.

Primary Server: The primary server has the highest priority in a group of servers gained from the DNS server.

Secondary Server: The secondary server backs up a primary server when the primary server fails. A secondary server may offer the same or less functionality than the primary server.

Server Redundancy Implementation

To assist in explaining the server redundancy behavior, an illustrative example of how an IP phone may be configured is shown as below. In the example, server redundancy for fallback and failover purposes is deployed. Two separate servers (a working server and a fallback server) are configured for per line registration.



Working Server: Server 1 is configured with the domain name of the working server. For example, yealink.pbx.com. DNS mechanism is used such that the working server is resolved to multiple servers for failover purpose. The working server is deployed in redundant pairs, designated as primary and secondary servers. The primary server has the highest priority in a cluster of servers resolved by the DNS server. The secondary server backs up a primary server when the primary server fails, and offers the same functionality as the primary server.

Fallback Server: Server 2 is configured with the IP address of the fallback server. For example, 192.168.1.15. A fallback server offers less functionality than the working server.

Phone Registration

Registration method of the failover mode:

The IP phone must always register to the primary server first except in failover conditions. If this is unsuccessful, the phone will re-register as many times as configured until the registration is successful. When the primary server registration is unavailable, the secondary server will serve as the working server.

Registration methods of the fallback mode include:

- Concurrent registration (default): The IP phone registers to two SIP servers (working server and fallback server) at the same time. In a failure situation, a fallback server can take over the basic calling capability, but without some advanced features (for example, shared lines, call recording and MWI) offered by the working server. It is not applicable to outbound proxy servers.
- Successive registration: The IP phone only registers to one server at a time. The IP phone
 first registers to the working server. In a failure situation, the IP phone registers to the
 failback server.

Server Domain Name Resolution

If a domain name is configured for a server, the IP address(es) associated with that domain name will be resolved through DNS as specified by RFC 3263. The DNS query involves NAPTR, SRV and A queries, which allows the IP phone to adapt to various deployment environments. The IP phone performs NAPTR query for the NAPTR pointer and transport protocol (UDP, TCP and TLS), the SRV query on the record returned from the NAPTR for the target domain name and the port number, and the A query for the IP addresses.

If an explicit port (except 0) is specified and the transport type is set to DNS-NAPTR, A query will be performed only. If a server port is set to 0 and the transport type is set to DNS-NAPTR, NAPTR and SRV queries will be tried before falling to A query. If no port is found through the DNS query, 5060 will be used.

For more information, refer to Appendix A: DNS SRV on page 18.

If your phone is not configured with a DNS server, or the DNS query returns no result from a DNS server, you can configure static DNS cache for the IP phone. The IP phone will attempts to resolve the domain name of the server with static DNS cache. For more information on static DNS cache, refer to Appendix B: Static DNS Cache on page 20.

Configuring Yealink IP Phones

Configuring Server Redundancy via Web User Interface

The followings take configurations of a SIP-T46G IP phone running firmware 81 as examples.

To configure server redundancy for fallback purpose via web user interface:

- 1. Click on Account->Register.
- 2. Select the desired account from the pull-down list of Account.
- 3. Configure registration parameters of the selected account in the corresponding fields.
- 4. Configure parameters of SIP server 1 and SIP server 2 in the corresponding fields.

			Log Out English(English) 🗸
Yealink 1466	Status Account Network	DSSKey Features Settings	
	Status Account Network	DSSKey Features Settings	Directory Security
Register	Account	Account 1 🔹 🕜	NOTE
Basic	Register Status	Registered	Account Registration
Dasic	Line Active	Enabled 🗸 🕜	Registers account(s) for the IP
Codec	Label	4605	phone.
Advanced	Display Name	4605	Server Redundancy It is often required in VoIP
	Register Name	4605	deployments to ensure continuity of phone service, for
	User Name	4605	events where the server needs to be taken offline for
	Password	•••••• 😢	maintenance, the server fails, or the connection between the IP
	SIP Server 1 🕜		phone and the server fails.
	Server Host	192.168.1.14 Port 5060	NAT Traversal A general term for techniques
	Transport	UDP 👻 🕐	that establish and maintain IP connections traversing NAT
	Server Expires	3600	gateways. STUN is one of the NAT traversal techniques.
	Server Retry Counts	3	Net develoar techniques.
	SIP Server 2 🕜		You can configure NAT traversal for this account.
	Server Host	192.168.1.15 Port 5060	for and account.
	Transport	UDP 👻 🕜	
	Server Expires	3600	
	Server Retry Counts	3	
	Enable Outbound Proxy Server	Disabled 🗸 🕜	
	Outbound Proxy Server 1	Port 5060	
	Outbound Proxy Server 2	Port 5060	
	Proxy Fallback Interval	3600	
	NAT	Disabled	
	Confirm	Cancel	

- 5. If you use outbound proxy servers, do the following:
 - 1) Select Enabled from the pull-down list of Enable Outbound Proxy Server.

2) Configure parameters of outbound proxy server 1 and outbound proxy server 2 in the corresponding fields.

alink 146G						Eng	Log (glish(English)	
	Status Account	Network	DSSKey	Features	Settings	Directory	Security	
Register	Account		Account 1	• ?		NOTE		
	Register Status		Registered					
lasic	Line Active		Enabled	- 🕜			unt(s) for the IP	
odec	Label		4605	0		phone.		
Advanced	Display Name		4605	0		Server Redu It is often requ		
	Register Name		4605	0		deployments to continuity of pl	o ensure hone service, fo	
	User Name		4605	0		events where t to be taken off	the server need: line for	
	Password		•••••	0			the server fails, between the IP	
	SIP Server 1 💡					phone and the		
	Server Host		192.168.1.14	Port 5	060 🕜	NAT Traversal A general term for technic that establish and maintai connections traversing NA gateways. STUN is one of NAT traversal techniques.		
	Transport		UDP	• 🕜			that establish and main connections traversing gateways. STUN is one	and maintain IP
	Server Expires		3600	0				IN is one of the
	Server Retry Counts		3	0				INAT traversal
	SIP Server 2 💡					You can config for this accoun	ure NAT travers	
	Server Host		192.168.1.15	Port 5	060 🕜	for this account	τ.	
	Transport		UDP	• 0				
	Server Expires		3600	0				
	Server Retry Counts		3	0				
	Enable Outbound Pro	xy Server	Enabled	• 🕜				
	Outbound Proxy Serv	ver 1	10.1.8.11	Port 5	060 🕜			
	Outbound Proxy Serv	ver 2	10.1.8.12	Port 5	060 🕜			
	Proxy Fallback Interv	al	3600	0				
	NAT		Disabled	• 🕜				
		nfirm		Cancel				

6. Click **Confirm** to accept the change.

To configure server redundancy for failover purpose via web user interface:

- 1. Click on Account->Register.
- 2. Select the desired account from the pull-down list of Account.
- 3. Configure registration parameters of the selected account in the corresponding fields.
- **4.** Configure parameters of the SIP server 1 or SIP server 2 in the corresponding fields. You must set the port of SIP server to 0 for NAPTR, SRV and A queries.

Register Basic	Account Register Status Line Active	Account 1 Registered	0	NOTE
	-	Registered		
Codec Advanced	Label Display Name	Enabled • 4605 4605	0 0 0	Account Registration Registers account(s) for the IP phone. Server Redundancy It is often required in VoIP deployments to ensure
	Register Name User Name Password SIP Server 1 7	4605 4605	0 0 0	continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.
	Server Host Transport Server Expires	yealink.pbx.com DNS-NAPTR 3600	Port 5060 00	NAT Traversal A general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

5. Select DNS-NAPTR from the pull-down list of Transport.

- 6. If you use outbound proxy servers, do the following:
 - 1) Select Enabled from the pull-down list of Enable Outbound Proxy Server.
 - **2)** Configure parameters of outbound proxy server 1 or outbound proxy server 2 in the corresponding fields.

You must set the port of outbound proxy to 0 for NAPTR, SRV and A queries.

a anti-a la l				Log (English(English)
alink 1466	Status Account Network	DSSKey	atures Settings	Directory Security
Register	Account	Account 1	- 0	NOTE
	Register Status	Registered		
Basic	Line Active	Enabled	▼ (?)	Account Registration Registers account(s) for the IF
Codec	Label	4605	0	phone.
Advanced	Display Name	4605	0	Server Redundancy It is often required in VoIP
	Register Name	4605	0	deployments to ensure continuity of phone service, for
	User Name	4605	0	events where the server need to be taken offline for
	Password	•••••	0	maintenance, the server fails, the connection between the IF
	SIP Server 1 🕜			phone and the server fails.
	Server Host	yealink.pbx.com	Port 5060	NAT Traversal A general term for techniques
	Transport	DNS-NAPTR	→ 🕜	that establish and maintain IP connections traversing NAT
	Server Expires	3600	0	gateways. STUN is one of the NAT traversal techniques.
	Server Retry Counts	3	0	
	SIP Server 2 🕜			You can configure NAT travers for this account.
	Server Host		Port 5060	
	Transport	UDP	- 🕜	
	Server Expires	3600	0	
	Server Retry Counts	3	0	
	Enable Outbound Proxy Server	Enabled	- 0	
	Outbound Proxy Server 1	yealink.sbc.com	Port 5060	
	Outbound Proxy Server 2		Port 5060 🥜	
	Proxy Fallback Interval	3600	0	
	NAT	Disabled	- 0	

7. Click **Confirm** to accept the change.

Configuring Server Redundancy Using Configuration Files

To configure server redundancy feature using configuration files:

 Add/Edit server redundancy parameters in configuration files. The following table lists the information of parameters: (For SIP VP-T49G/SIP-T48G/T48S/T46G/T46S/T29G: X ranges from 1 to 16; For SIP-T42G/S: X ranges from 1 to 12; For SIP-T41P/T41S/T27P/T27G: X ranges from 1 to 6; For W56P: X ranges from 1 to 5; For SIP-T40P/T23P/T23G: X ranges from 1 to 3; For SIP-T21(P) E2: X ranges from 1 to 2; For SIP-T19(P) E2/CP860: X is equal to 1; Y ranges from 1 to 2)

Parameters	Permitted Values	Default			
account.X.enable	0 or 1	0			
Description:					
Enables or disables the account X.	Enables or disables the account X.				
0 -Disabled					
1-Enabled					
Web User Interface:					
Account->Register->Line Active					
Phone User Interface:					
Menu->Advanced (default password: admin) ->Account->Activation					
account.X.label	String	Blank			
Description:					
Configures the label displayed on the LCD screen for account	Х.				
Web User Interface:					
Account->Register->Label	Account->Register->Label				
Phone User Interface:					
Menu->Advanced (default password: admin) ->Account->Lat	pel				
account.X.display_name	String	Blank			

Parameters	Permitted Values	Default
Description:		
Configures the display name for account X.		
Web User Interface:		
Account->Register->Display Name		
Phone User Interface:		
Menu->Advanced (default password: admin) ->Account->E	Display Name	
account.X.auth_name	String	Blank
Description:		
Configures the user name for register authentication for acc	count X.	
Web User Interface:		
Account->Register->Register Name		
Phone User Interface:		
Menu->Advanced (default password: admin) ->Account->R	Register Name	
account.X.user_name	String	Blank
Description:		
Configures the register user name for account X.		
Web User Interface:		
Account->Register->User Name		
Phone User Interface:		
Menu->Advanced (default password: admin) ->Account->L	Jser Name	
account.X.password	String	Blank
Description:		
Configures the password for register authentication for acco	ount X.	
Web User Interface:		
Account->Register->Password		
Phone User Interface:		
Menu->Advanced (default password: admin) ->Account->F	Password	
account.X.sip_server.Y.address	String within 256 characters	Blank
Description:		
Description.		
Configures the IP address or domain name of the SIP server	Y that accepts regist	rations for

Parameters	Permitted Values	Default			
Web User Interface:					
Account->Register->SIP Server Y->Server Host					
Phone User Interface:					
Menu->Advanced (default password: admin) ->Account->SIF	9 Server1				
	Integer				
account.X.sip_server.Y.port	from 0 to	5060			
	65535				
Description:					
Configures the port of the SIP server Y that specifies registrat	ions for account X.				
Web User Interface:					
Account->Register->SIP Server Y->Port					
Phone User Interface:					
None					
account.X.sip_server.Y.transport_type	0, 1, 2 or 3	0			
Description:					
Configures the transport method the IP phone uses to comm account X.	unicate with the SIP	server for			
0-UDP					
1-TCP					
2 -TLS					
3 -DNS-NAPTR					
Web User Interface:					
Account->Register->SIP Server Y->Transport					
Phone User Interface:					
None					
	Integer				
account.X.sip_server.Y.expires	from 30 to 2147483647	3600			
Description:					
Configures the registration expires (in seconds) of the SIP ser	ver Y for account X.				
Web User Interface:					
Account->Register->SIP Server Y->Server Expires					
Phone User Interface:					

Parameters	Permitted Values	Default			
None					
account.X.sip_server.Y.retry_counts	Integer from 0 to 20	3			
from 0 to 20 Description: Configures the retry times for the IP phone to resend requests when the SIP server Y is unavailable or there is no response from the SIP server Y for account X. Web User Interface: Account->Register->SIP Server Y->Server Retry Counts Phone User Interface: None account.X.outbound_proxy_enable 0 or 1 0 Description: Enables or disables the phone to use the outbound proxy server for account X. 0-Disabled 1-Enabled					
Web User Interface: Account->Register->Enable Outbound Proxy Server Phone User Interface: Menu->Advanced (default password: admin) ->Account->Ou	thound Status				
account.X.outbound_host	IP address or domain name	Blank			
Description: Configures the IP address or domain name of the outbound proxy server 1 for account X. Note: It is only applicable to IP phones running firmware version 80 or prior. Web User Interface: Account->Register->Outbound Proxy Server 1 Phone User Interface: Menu->Advanced (default password: admin) ->Account->Outbound Proxy1 Integer account.X.outbound_port					
Description:					

Parameters	Permitted Values	Default			
Configures the port of the outbound proxy server 1 for account X.					
Note: It is only applicable to IP phones running firmware vers	ion 80 or prior.				
Web User Interface:					
Account->Register->Outbound Proxy Server 1->Port					
Phone User Interface:					
None					
account.X.backup_outbound_host	IP Address or Domain Name	Blank			
Description:					
Configures the IP address or domain name of the outbound p	proxy server 2 for ac	count X.			
Note: It is only applicable to IP phones running firmware vers	ion 80 or prior.				
Web User Interface:					
Account->Register->Outbound Proxy Server 2					
Phone User Interface:					
Menu->Advanced (default password: admin) ->Account->Ou	tbound Proxy2				
Integer					
account.X.backup_outbound_port	from 0 to	5060			
	65535				
Description:					
Configures the port of the outbound proxy server 2 for accou	nt X.				
Note: It is only applicable to IP phones running firmware vers	ion 80 or prior.				
Web User Interface:					
Account->Register->Outbound Proxy Server 2->Port					
Phone User Interface:					
None					
account.X.outbound_proxy.Y.address	IP Address or Domain Name	Blank			
Description:					
Configures the IP address or domain name of the outbound p	arow server V for ac	count X			
Note : It is only applicable to IP phones running firmware vers	-	Count A.			
Web User Interface:	וטו טב טו ומנפו.				
Account->Register->Outbound Proxy Server Y					
Phone User Interface:					

Parameters	Permitted Values	Default		
Menu->Advanced (default password: admin) ->Account->Outbound ProxyY				
account.X.outbound_proxy.Y.port	Integer from 0 to 65535	5060		
Description:				
Configures the port of the outbound proxy server Y for accou	int X.			
Note : It is only applicable to IP phones running firmware vers	ion 81 or later.			
Web User Interface:				
Account->Register->Outbound Proxy Server Y->Port				
Phone User Interface:				
None				
Fallback Mode				
account.X.fallback.redundancy_type	0 or 1	0		
Description:				
Configures the registration mode for the IP phone in fallback	mode.			
0-Concurrent Registration				
1-Successive Registration				
Note: It is not applicable to outbound proxy servers.				
Web User Interface:				
None				
Phone User Interface:				
None				
account.X.fallback.timeout	Integer from 10 to 2147483647	120		
Description:				
Configures the time interval (in seconds) for the IP phone to detect whether the working server is available by sending the registration request after the fallback server takes over call control.				
Note: It works only if the value of the parameter "account.X.fallback.redundancy_type" is set to 1 (Successive Registration). It is not applicable to outbound proxy servers.				
Web User Interface:				
None				

Parameters	Permitted Values	Default		
Phone User Interface:				
None				
account.X.outbound_proxy_fallback_interval	Integer	3600		
Description:				
Configures the time interval (in seconds) for the IP phone to detect whether the working outbound proxy server is available by sending the registration request after the fallback server takes over call control.				
Note: It is only applicable to outbound proxy servers.				
Web User Interface:				
Account->Register->Proxy Fallback Interval				
Phone User Interface:				
Menu->Advanced (default password: admin) ->Account->Pro	oxy Fallback Interval			
Failover Mode				
account.X.sip_server.Y.register_on_enable	0 or 1	0		
Description: Enables or disables the IP phone to register to the secondary to it for account X when encountering a failover. 0 -Disabled 1 -Enabled If it is set to 0 (Disabled), the IP phone won't attempt to regists since the phone assumes that the primary and secondary servinformation. So the IP phone will directly send the requests to If it is set to 1 (Enabled), the IP phone will register to the secondary servinformation. So the IP phone will directly send the requests to If it is set to 1 (Enabled), the IP phone will register to the second send the requests to it. Web User Interface: None Phone User Interface: None	ter to the secondary vers share registratic o the secondary serv	v server, on ver.		
account.X.sip_server.Y.only_signal_with_registered	0 or 1	0		
Description: Enables or disables the IP phone to only send requests to the	registered server fo	r account X		

when encountering a failover.

Parameters	Permitted Values	Default			
0-Disabled					
1-Enabled					
Note: It is only applicable to IP phones running firmware vers	ion 81 or later.				
Web User Interface:					
None					
Phone User Interface:					
None					
account.X.sip_server.Y.invite_retry_counts	Integer from 1 to 10	3			
Description:					
Configures the number of retries attempted before sending reserver for account X when encountering a failover.	equests to the next a	available			
Note : It is only applicable to IP phones running firmware vers	ion 81 or later.				
Web User Interface:					
None					
Phone User Interface:					
None					
account.X.sip_server.Y.failback_mode	account.X.sip_server.Y.failback_mode 0, 1, 2 or 3 0				
Description:					
Configures the way in which the phone fails back to the prima	ary server.				
0 -newRequests: all requests are sent to the primary server first, regardless of the last server that was used.					
1 -DNSTTL: the IP phone will send requests to the last used se DNSTTL on the server expires, the phone will retry to send red		•			
2 -registration: the IP phone will send requests to the last used expires, the phone will retry to send requests to the primary s		egistration			
3 -duration: the IP phone will send requests to the last registered server first. If the time defined by the parameter "account.X.sip_server.Y.failback_timeout" expires, the phone will retry to send requests to the primary server.					
Note: DNSTTL, Registration and duration mode can only be p	processed when the I	IP phone is			
idle (that is, no incoming/outbound calls, no active calls or me	eetings, etc.).				
Web User Interface:					
None					
Phone User Interface:					
None					

Parameters	Permitted Values	Default
account.X.sip_server.Y.failback_timeout	0, Integer from 60 to 65535	3600
Description:		
Configures the timeout (in seconds) for the phone to retry to server after failing over to the current working server for acco		e primary
If you set the parameter to 0, the IP phone will not send requ a failover event occurs with the current working server.	ests to the primary s	server until
If you set the parameter from 1 to 59, the timeout will be 60	seconds.	
Note: It works only if the value of the parameter "account.X.s set to 3 (duration).	ip_server.Y.failback_	mode" is
Web User Interface:		
None		
Phone User Interface:		
None		
account.X.sip_server.Y.failback_subscribe.enable	0 or 1	0
Description:		
Enables or disables the IP phone to retry to re-subscribe afte server with different IP address for account X when encounte		econdary
0-Disabled		
1-Enabled		
If it is set to 1 (Enabled), the IP phone will immediately re-sub	oscribe to the second	dary server,
for ensuring the normal use of the features associated with s	ubscription (e.g., BLF	, SCA).
Note : It is only applicable to IP phones running firmware ver- the value of the parameter "account.X.sip_server.Y.failback_m		•
Web User Interface:		
None		
Phone User Interface:		

<y000000000xx.cfg> configuration file:

```
##Account1 Registration
account.1.enable = 1
account.1.label = 4605
account.1.display_name = 4605
```

```
account.1.auth name = 4605
account.1.user name = 4605
account.1.password = 4605
account.1.sip server.1.address = yealink.pbx.com
account.1.sip server.1.port = 0
account.1.sip server.1.expires = 3600
account.1.sip_server.1.retry_counts = 3
account.1.outbound proxy enable = 1
account.1.outbound proxy.1.address = yealink.pbx.com
account.1.outbound proxy.1.port = 0
##DNS SRV
account.1.sip_server.1.transport_type = 3
##Failover Mode
account.1.sip_server.1.register_on_enable = 0
account.1.sip_server.1.only_signal_with_registered = 1
account.1.sip server.1.invite retry counts = 5
account.1.sip server.1.failback mode = 3
account.1.sip server.1.failback timeout = 3600
account.1.sip server.1.failback subscribe.enable = 1
```

Upload configuration files to the root directory of the provisioning server and trigger IP phones to perform an auto provisioning for configuration update.
 For more information on auto provisioning, refer to *Yealink_SIP-T2_Series_T19(P)* E2_T4_Series_CP860_W56P_IP_Phones_Auto_Provisioning_Guide.

```
Yealink IP phones running firmware version 81 or later support a new boot file for auto provisioning. For more information, refer to Yealink_SIP-T2_Series_T19(P)
E2_T4_Series_IP_Phones_Auto_Provisioning_Guide_V81.
```

Using Server Redundancy on Yealink IP Phones

Fallback Scenario

The following introduces a REGISTER fallback scenario. The SIP server 1 (working server) and SIP server 2 (fallback server) are configured with the IP address respectively for account 1. The parameter "account.1.fallback.redundancy_type" is configured as 1 (Successive Registration). You do not use the outbound proxy servers.

REGISTER Fallback

The phone has ability to fail over to a fallback server when the working server has no response to a REGISTER request.

- 1. The phone sends a REGISTER request to the working server.
- 2. The phone retries to send REGISTER requests to the working server (three times by default).
- **3.** After no response from the working server, the phone sends a REGISTER request to the fallback server after the registration time defined for the working server expires.
- 4. The fallback server responds with 200 OK to the REGISTER request.

The phone sends REGISTER requests to the working server to detect whether the server is available at intervals defined by the "account.1.fallback.timeout" parameter after failing over to the fallback server. When the working server recovers, the phone has ability to fail back next REGISTER request to the working server.

The following introduces an INVITE fallback scenario. The SIP server 1 (working server) and SIP server 2 (fallback server) are configured with the IP address respectively for account 1. The parameter "account.1.fallback.redundancy_type" is configured as 0 (Concurrent Registration).

INVITE Fallback

The phone has ability to fail over to a fallback server when the working server has no response to an INVITE request.

- 1. Phone A places a call to Phone B.
- 2. Phone B answers the call.

The following SIP messages appear:

- Phone A sends an INVITE request to the working server.
- Phone A retries INVITE requests to the working server (three times by default).
- After no response from the working server, the phone sends an INVITE request to the fallback server.
- The fallback server responds with 200 OK to the INVITE request.

Phone A sends REGISTER requests to the working server to detect whether the server is available. When the working server recovers, the phone has ability to fail back the INVITE request to the working server.

Failover Scenario

The following introduces a REGISTER failover scenario. The SIP server 1 is configured with the domain name of the working server for account 1. The working server is resolved to two SIP servers (primary server and secondary server) using the DNS mechanism. The parameter "account.1.sip_server.1.failback_mode" is configured as 0 (newRequests) and "account.1.sip_server.1.register_on_enable" is configured as 0 (Disabled). You do not use the outbound proxy servers.

REGISTER Failover

The phone has ability to fail over to a secondary server when the primary server has no response to a REGISTER request.

- 1. The phone sends REGISTER request to the primary server.
- 2. The phone retries REGISTER requests to the primary server (three times by default).
- **3.** After no response from the primary server, the phone sends a REGISTER request to the secondary server.
- 4. The secondary server responds with 200 OK to the REGISTER request.

The phone waits until next REGISTER attempt and then sends next REGISTER request to the primary server. When the primary server recovers, the phone has ability to fail back next REGISTER request to the primary server.

INVITE Failover

The phone has ability to fail over to a secondary server when the primary server has no response to an INVITE request.

- 1. Phone A places a call to Phone B.
- 2. Phone B answers the call.

The following SIP messages appear:

- Phone A sends an INVITE request to the primary server.
- Phone A retries INVITE requests to the primary server (three times by default).
- After no response from the primary server, the phone sends an INVITE request to the secondary server.
- The secondary server responds with 200 OK to the INVITE request.

When phone A places a call to Phone B again, the phone sends an INVITE request to the primary server first. When the primary server recovers, the phone has ability to immediately fail back INVITE request to the primary server after failing over to the secondary server.

Appendix A: DNS SRV

The following details the procedures of DNS query for the IP phone to resolve the domain name (e.g., yealink.pbx.com) of working server into the IP address, port and transport protocol.

NAPTR (Naming Authority Pointer)

First, the IP phone sends NAPTR query to get the NAPTR pointer and transport protocol. Example of NAPTR records:

	order	pref	flags	service	regexp	replacement	
IN NAPTR	90	50	"s"	"SIP+D2T"		_siptcp.yealink.pbx.c	com
IN NAPTR	100	50	"s"	"SIP+D2U"		_sipudp.yealink.pb	x.com

Parameters are explained in the following table:

Parameter	Description
order	Specifies preferential treatment for the specific record. The order is from lowest to highest, lower order is more preferred.
pref	Specifies the preference for processing multiple NAPTR records with the same order value. Lower value is more preferred.
flags	The flag "s" means to perform an SRV lookup.
	Specify the transport protocols supported:
	SIP+D2U: SIP over UDP
service	SIP+D2T: SIP over TCP
	SIP+D2S: SIP over SCTP
	SIPS+D2T: SIPS over TCP
regexp	Always empty for SIP services.
replacement	Specifies a domain name for the next query.

The IP phone picks the first record, because its order of 90 is lower than 100. The pref parameter is unimportant as there is no other record with order 90. The flag "s" indicates performing the SRV query next. TCP will be used, targeted to a host determined by an SRV query of "_sip._tcp.yealink.pbx.com". If the flag of the NAPTR record returned is empty, the IP phone will perform NAPTR query again according to the previous NAPTR query result.

SRV (Service Location Record)

The IP phone performs an SRV query on the record returned from the NAPTR for the host name and the port number. Example of SRV records:

	Priority	Weight	Port	Target
IN SRV	0	1	5060	server1.yealink.pbx.com
IN SRV	0	2	5060	server2.yealink.pbx.com

Parameters are explained in the following table:

Parameter	Description
Priority	Specifies preferential treatment for the specific host entry. Lower priority is more preferred.
Weight	When priorities are equal, weight is used to differentiate the preference. The preference is from highest to lowest. Again, keep the same to load balance.
Port	Identifies the port number to be used.
Target	Identifies the actual host for an A query.

SRV query returns two records. The two SRV records point to different hosts and have the same priority 0. The weight of the second record is higher than the first one, so the second record will be picked first. The two records also contain a port "5060", the IP phone uses this port. If the Target is not a numeric IP address, the IP phone performs an A query. So in this case, the IP phone uses "server1.yealink.pbx.com" and "server2.yealink.pbx.com" for the A query.

A (Host IP Address)

The IP phone performs an A query for the IP address of each target host name. Example of A records:

Server1.yealink.pbx.com IN A 192.168.1.13 Server2.yealink.pbx.com IN A 192.168.1.14 The IP phone picks the IP address "192.168.1.14" first.

Appendix B: Static DNS Cache

Yealink IP phones allow you to statically configure a set of NAPTR/SRV/A records. The following details the configuration parameters of the static DNS cache for the IP phone to resolve the domain name of the server.

You can specify the preference of the records used by IP phones. To use static DNS cache preferentially, set the following parameter to 1.

(For SIP VP-T49G/SIP-T48G/T48S/T46G/T46S/T29G: X ranges from 1 to 16;

For SIP-T42G/S: X ranges from 1 to 12;

For SIP-T41P/T41S/T27P/T27G: X ranges from 1 to 6;

For W56P: X ranges from 1 to 5;

For SIP-T40P/T23P/T23G: X ranges from 1 to 3;

For SIP-T21(P) E2: X ranges from 1 to 2;

For SIP-T19(P) E2/CP860: X is equal to 1)

Parameters	Permitted Values	Default			
account.X.dns_cache_type	0, 1 or 2	1			
Description:					
Configures whether the IP phone uses the DNS cach	Configures whether the IP phone uses the DNS cache for domain name resolution of the				
server and caches the additional DNS records for account X.					
0 -Perform real-time DNS query rather than using DNS cache.					
1 -Use DNS cache, but do not cache the additional DNS records.					
2 -Use DNS cache and cache the additional DNS records.					
Web User Interface:					
None					

Parameters	Permitted Values	Default		
Phone User Interface:				
None				
account.X.static_cache_pri	0 or 1	0		
Description:				
Configures whether preferentially to use the static D	NS cache for domain name re	esolution of		
the server for account X.				
0 -Use domain name resolution from server preferentially				
1-Use static DNS cache preferentially				
Web User Interface:				
None				
Phone User Interface:				
None				

Specifying DNS A Parameters

The following table lists the configuration parameters for specifying the domain name, IP address, and Time to Live (TTL) for A record (X ranges from 1 to 12):

Parameters	Permitted Values	Default				
dns_cache_a.X.name	Domain name	Blank				
Description:						
Configures the domain name in A record X.						
Web User Interface:						
None						
Phone User Interface:						
None						
dns_cache_a.X.ip	String	Blank				
Description:						
Configures the IP address that the domain name in A record X maps to.						
Web User Interface:						
None						
Phone User Interface:						
None						
dns_cache_a.X.ttl	Integer from 30 to 2147483647	300				

Parameters	Permitted Values	Default
Description:		
Configures the time interval (in seconds) that A reco should be consulted again.	rd X may be cached before tl	ne record
Web User Interface:		
None		
Phone User Interface:		
None		

Specify DNS SRV Parameters

The following table lists the configuration parameters for specifying the domain name, port, priority, target, weight and Time to Live (TTL) for SRV record (X ranges from 1 to 12):

Parameters	Permitted Values	Default
dns_cache_srv.X.name	Domain name	Blank
Description:		
Configures the domain name in SRV record X.		
Web User Interface:		
None		
Phone User Interface:		
None		
dns_cache_srv.X.port	Integer from 0 to 65535	0
Description:		
Configures the port to be used in SRV record X.		
Web User Interface:		
None		
Phone User Interface:		
None		
dns_cache_srv.X.priority	Integer from 0 to 65535	0
Description:		
Configures the priority for the target host in SRV rec	ord X.	
Lower priority is more preferred. For example, SRV re	ecord with the priority value () is more
preferred than that with the priority value 1 because	0 is lower than 1.	
Note : For more information, refer to RFC 2782.		

Parameters	Permitted Values	Default
Web User Interface:		
None		
Phone User Interface:		
None		
dns_cache_srv.X.target	Domain name	Blank
Description:		
Configures the domain name of the target host for a	in A query in SRV record X.	
Note: For more information, refer to RFC 2782.		
Web User Interface:		
None		
Phone User Interface:		
None		
dns_cache_srv.X.weight	Integer from 0 to 65535	0
Configures the weight of the target host in SRV reco When priorities are equal, weight is used to different is more preferred. Note: For more information, refer to RFC 2782. Web User Interface: None Phone User Interface:		eight value
None		
None dns_cache_srv.X.ttl	Integer from 30 to 2147483647	300
	-	300
dns_cache_srv.X.ttl	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV rea	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV rea should be consulted again.	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV re- should be consulted again. Web User Interface:	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV reasonable of the should be consulted again. Web User Interface: None	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV red should be consulted again. Web User Interface: None Phone User Interface:	2147483647	
dns_cache_srv.X.ttl Description: Configures the time interval (in seconds) that SRV red should be consulted again. Web User Interface: None Phone User Interface: None	2147483647	the record

Parameters	Permitted Values	Default
returned from NAPTR query for account X.		
0 -SRV query using UDP only		
1 -SRV query using UDP, TCP and TLS		
Web User Interface:		
None		
Phone User Interface:		
None		

Specify DNS NAPTR Parameters

The following table lists the configuration parameters for specifying the domain name, order, flags, preference, replacement, service and Time to Live (TTL) for NAPTR record (X ranges from 1 to 12):

Parameters	Permitted Values	Default		
dns_cache_naptr.X.name	Domain Name	Blank		
Description:				
Configures the domain name to which NAPTR record X refers.				
Web User Interface:				
None				
Phone User Interface:				
None				
dns_cache_naptr.X.flags	S, A, U or P	Blank		
Description:				
Configures the flag of NAPTR record X. (Always "S" for SIP, which means to do an SRV				
lookup on whatever is in the replacement field).				
S-Do an SRV lookup next.				
A-Do an A lookup next.				
U -No need to do a DNS query next.				
P-Service customized by the user				
Note : For more details of the permitted flags, refer to RFC 2915.				
Web User Interface:				
None				
Phone User Interface:				
None				

Parameters	Permitted Values	Default		
dns_cache_naptr.X.order	Integer from 0 to 65535	0		
Description:				
Configures the order of NAPTR record X.				
NAPTR record with lower order is more preferred. For example, NAPTR record with the order				
90 has the higher priority than that with the order 100 because 90 is lower than 100.				
Web User Interface:				
None				
Phone User Interface:				
None				
dns_cache_naptr.X.preference	Integer from 0 to 65535	0		
Description:				
Configures the preference of NAPTR record X.				
NAPTR record with lower value is more preferred when the multiple NAPTR records have the				
same order value.				
Web User Interface:				
None				
Phone User Interface:				
None				
dns_cache_naptr.X.replace	Domain name	Blank		
Description:				
Configures a domain name to be used for the next	SRV query in NAPTR record X			
Web User Interface:				
None				
Phone User Interface:				
None				
dns_cache_naptr.X.service	String within 32 characters	Blank		
Description:				
Configures the transport protocol available for the server in NAPTR record X.				
SIP+D2U: SIP over UDP				
SIP+D2T: SIP over TCP				
SIP+D2S: SIP over SCTP				
SIPS+D2T: SIPS over TCP				

Parameters	Permitted Values	Default		
Web User Interface:				
None				
Phone User Interface:				
None				
dns_cache_naptr.X.ttl	Integer from 30 to	300		
	2147483647	300		
Description:				
Configures the time interval (in seconds) that NAPTR record X may be cached before the				
record should be consulted again.				
Web User Interface:				
None				
Phone User Interface:				
None				

Example Configuration

The following three examples show you how to configure the static DNS cache.

Example 1

This example shows how to configure static DNS cache when your DNS server does not return A records. In this case, the static DNS cache on the phone provides A records.

When the static DNS cache is used, the configurations would look as below:

```
account.1.sip_server.1.address = yealink.pbx.com
account.1.sip_server.1.port = 5060
account.1.sip_server.1.transport_type = 3
dns_cache_a.1.name = yealink.pbx.com
dns_cache_a.1.ip = 192.168.1.13
dns_cache_a.1.ttl = 3600
dns_cache_a.2.name = yealink.pbx.com
dns_cache_a.2.ip = 192.168.1.14
dns_cache_a.2.ttl = 3600
```

Example 2

This example shows how to configure static DNS cache when your DNS server returns A records but not SRV records. In this case, the static DNS cache on the phone provides SRV records.

When the static DNS cache is used, the configurations would look as below:

```
account.1.sip_server.1.address = yealink.pbx.com
account.1.sip_server.1.port = 0
account.1.sip_server.1.transport_type = 3
dns_cache_srv.1.name = _sip._tcp.yealink.pbx.com
dns_cache_srv.1.port = 5060
dns_cache_srv.1.priority = 0
dns_cache_srv.1.target = server1.yealink.pbx.com
dns_cache_srv.1.weight = 1
dns_cache_srv.1.ttl = 3600
dns_cache_srv.2.name = _sip._tcp.yealink.pbx.com
dns_cache_srv.2.port = 5060
dns_cache_srv.2.priority = 0
dns_cache_srv.2.target = server2.yealink.pbx.com
dns_cache_srv.2.weight = 2
dns_cache_srv.2.ttl = 3600
```

```
Note
```

The parameter "account.1.sip_server.1.port" is set to 0 to force SRV query.

Example 3

This example shows how to configure static DNS cache when your DNS server returns A and SRV records but not NAPTR records. In this case, the static DNS cache on the phone provides NAPTR records.

When the static DNS cache is used, the configurations would look as below:

```
account.1.sip_server.1.address = yealink.pbx.com
account.1.sip_server.1.port = 0
account.1.sip_server.1.transport_type = 3
dns_cache_naptr.1.name = yealink.pbx.com
dns_cache_naptr.1.flags = S
dns_cache_naptr.1.order = 90
dns_cache_naptr.1.preference = 50
dns_cache_naptr.1.replace = _sip._tcp.yealink.pbx.com
dns_cache_naptr.1.service = SIP+D2T
dns_cache_naptr.1.ttl = 3600
```

```
dns_cache_naptr.2.name = yealink.pbx.com
dns_cache_naptr.2.flags = S
dns_cache_naptr.2.order = 100
dns_cache_naptr.2.preference = 50
dns_cache_naptr.2.replace = _sip._udp.yealink.pbx.com
dns_cache_naptr.2.service = SIP+D2U
dns_cache_naptr.2.ttl = 3600
```

Note The parameter "account.1.sip_server.1.port" is set to 0 to force NAPTR query.

Customer Feedback

We are striving to improve our documentation quality and we appreciate your feedback. Email your opinions and comments to DocsFeedback@yealink.com.