TeleDynamic COMMUNICATIONS | we get it

Yealink EASY VoIP

Server Redundancy on Yealink IP Phones

Server Redundancy on Yealink IP Phones

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

The information applies to Yealink SIP-T28P, SIP-T26P, SIP-T22P, SIP-T20P, SIP-T21P, SIP-T19P, SIP-T46G, SIP-T42G and SIP-T41P IP phones running firmware version 71 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, a second less featured call server (fallback server) with SIP capability takes over call control to provide basic calling capability, but without some advanced features offered by the working server (for example, shared lines, call recording and MWI). IP phones support configurations of two SIP servers per SIP registration for this purpose.

Glossary

The following terms may assist in understanding server redundancy feature:

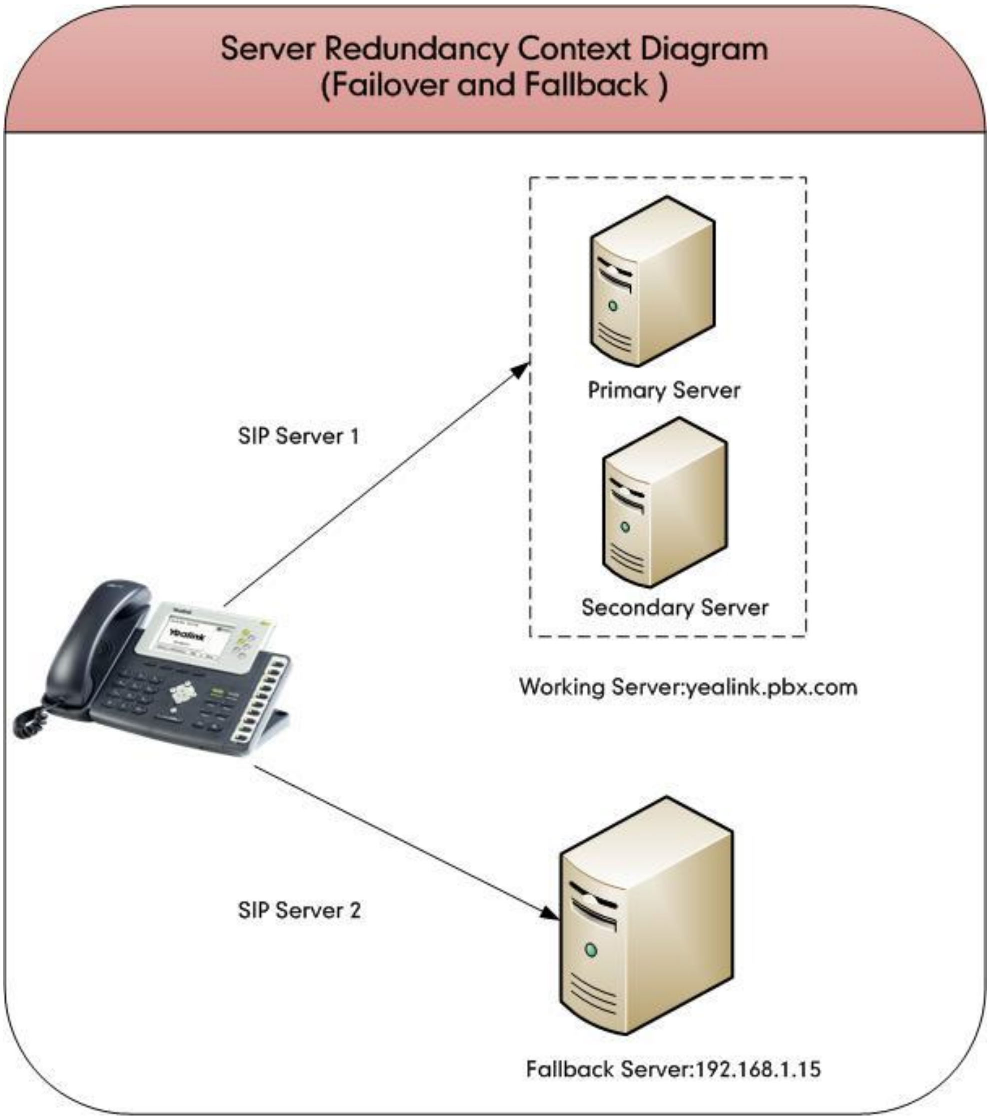
Working and Fallback Servers: The working and fallback servers are two separate servers used for each line registration.

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 SIP servers (a working server and a fallback server) are configured for per line registration.



Working Server: SIP 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 SIP 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: SIP 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. 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**: 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 offered by the working server (default registration method).
- **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 fallback server.

SIP Server Domain Name Resolution

If a domain name is configured for a SIP 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 SIP 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.

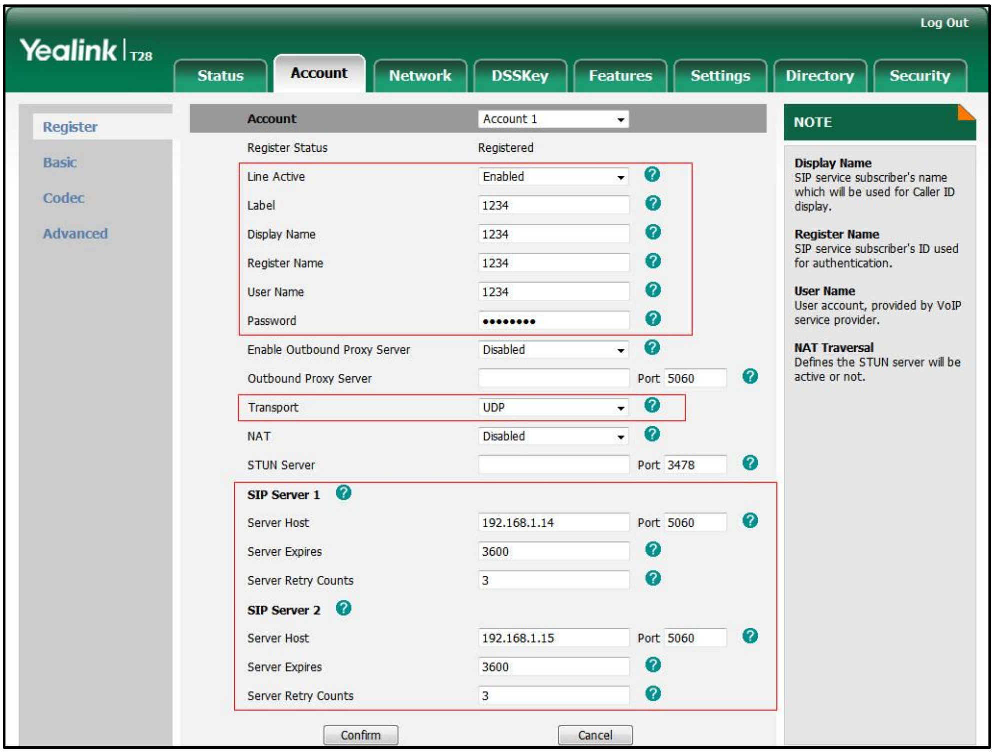
For more information, refer to Appendix: DNS SRV on page 13.

Configuring Yealink IP Phones

You can configure server redundancy feature for the IP phone via web user interface or using configuration files. The followings take configurations of a SIP-T28P IP phone running firmware version 71 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. Select the desired value from the pull-down list of **Transport**.
5. Configure parameters of SIP server 1 and SIP server 2 in the corresponding fields.



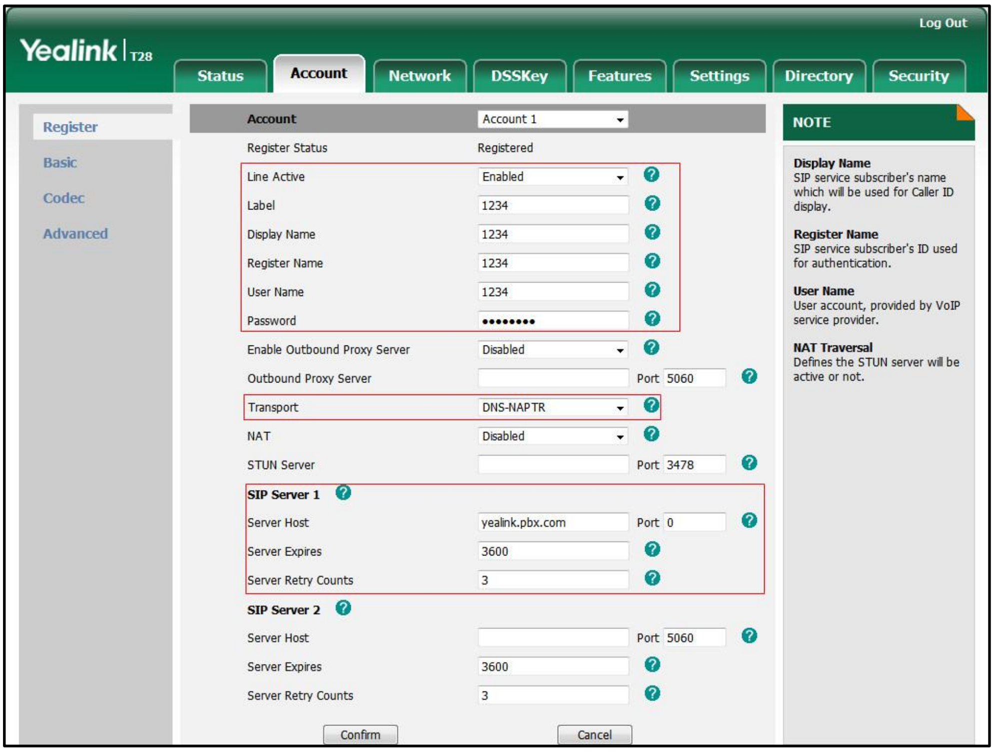
6. Click **Confirm** to accept the change.

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

5. Configure parameters of 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.



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

Note

If the outbound proxy server is required and the transport is set to DNS-NAPTR, you must set the port of outbound proxy server to 0 for NAPTR, SRV and A queries.

To configure server redundancy feature using configuration files:

1. Add/Edit server redundancy parameters in configuration files.

   The following table shows the information of parameters: (For SIP-T28P/T46G, x ranges from 1 to 6; For SIP-T26P/T22P/T42G/T41P, x ranges from 1 to 3; For SIP-T21P/T20P, x ranges from 1 to 2; For SIP-T19P, x is 1):

Parameter	Description	Valid Value	Web Path
Phone Registration			
account.x.enable	Enables or disables the account x. **0**-Disabled **1**-Enabled The default value is 0.	0 or 1	Account->Register->Line Active

account.x.label	Configures the label displayed on the LCD screen for account x.	String	Account-> Register ->Label
account.x.display_name	Configures the display name for account x.	String	Account-> Register-> Display Name
account.x.auth_name	Configures the user name for register authentication for account x.	String	Account-> Register-> Register Name
account.x.user_name	Configures the register user name for account x.	String	Account-> Register->User Name
account.x.password	Configures the password for register authentication for account x.	String	Account-> Register-> Password
account.x.outbound_proxy_enable	Enables or disables the phone to use the outbound proxy server for account x. **0**-Disabled **1**-Enabled The default value is 0.	0 or 1	Account->Register->Enable Outbound Proxy Server
account.x.outbound_host	Configures the IP address or domain name of the outbound proxy server for account x.	IP Address or Domain Name	Account->Register->Outbound Proxy Server
account.x.outbound_port =	Configures the port of the outbound proxy server for account x.	Integer from 0 to 65535	Account->Register->Outbound Proxy Server->Port
account.x.sip_server.1.address	Configures the IP address or domain name of the SIP server 1 for account x.	IP Address or Domain Name	Account-> Register->SIP Sever 1->Server Host
account.x.sip_server.1.port	Configures the port of the SIP server 1 for account x. The default value is 5060.	Integer from 0 to 65535	Account-> Register->SIP Sever 1->Server Host->Port

account.x.sip_server.1.expires	Configures the registration expires (in seconds) of the SIP server 1 for account x. The default value is 3600s.	Integer from 30 to 2147483647	Account->Register->SIP Sever 1->Server Expires
account.x.sip_server.1.retry_counts	Configures the retry times for the IP phone to resend requests when the SIP server 1 does not respond for account x. The default value is 3.	Integer from 0 to 20	Account->Register->SIP Sever 1->Server Retry Counts
account.x.sip_server.2.address	Configures the IP address or domain name of the SIP server 2 for account x.	IP Address or Domain Name	Account->Register->SIP Sever 2->Server Host
account.x.sip_server.2.port	Configures the port of the SIP server 2 for account x. The default value is 5060.	Integer from 0 to 65535	Account->Register->SIP Sever 2->Server Host->Port
account.x.sip_server.2.expires	Configures the registration expires (in seconds) of the SIP server 2 for account x. The default value is 3600s.	Integer from 30 to 2147483647	Account->Register->SIP Sever 2->Server Expires
account.x.sip_server.2.retry_counts	Configures the retry times for the IP phone to resend requests when the SIP server 2 does not respond for account x. The default value is 3.	Integer from 0 to 20	Account->Register->SIP Sever 2->Server Retry Counts
DNS SRV			
account.x.transport	Configures the transport type for account x. **0**-UDP **1**-TCP **2**-TLS **3**-DNS-NAPTR The default value is 0.	0, 1, 2 or 3	Account->Register->Transport

account.x.naptr_build	Specifies UDP SRV query or TCP/TLS SRV query for the IP phone to be performed when no result is returned from NAPTR query. **0**-UDP **1**-TCP or TLS. The default value is 0.	0 or 1	
Failover Mode			
account.x.sip_server.y.failback_mode	Configures the way in which the phone fails back to the primary server for call control in the failover mode. **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 retry to send requests to the primary server after the timeout equal to the DNSTTL configured for the server that the IP phone is registered to. **2**-registration: the IP phone will retry to send REGISTER requests to the primary server when registration renewal. **3**-duration: the IP phone will retry to send requests to the primary server after the timeout defined by the account.x.sip_server.y.failback_timeout parameter. The default value is 0.	0, 1, 2 or 3	

account.x.sip_server.y.failback_timeout	Configures the time (in seconds) for the phone to retry to send requests to the primary server after failing over to the current working server when the parameter account.x.sip_server.y.failback_mode is set to duration. If you set the parameter to 0, the IP phone will not send requests to the primary server until a failover event occurs with the current working server.	0, 60 to 65535	
account.x.sip_server.y.register_on_enable	Enables or disables the IP phone to register to the secondary server before sending requests to the secondary server in the failover mode. **0**-Disabled **1**-Enabled The default value is 0.	0 or 1	
Fallback Mode			
account.x.fallback.redundancy_type	Configures the registration mode for the IP phone in fallback mode. **0**-Concurrent Registration **1**-Successive Registration The default value is 0.	0 or 1	

account.x.fallback .timeout	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. It is only applicable to the Successive Registration mode. The default value is 120s.	Integer from 10 to 2147483647	

The following shows an example of failover configurations for account 1 in configuration files:

```
##Phone Registration
account.1.enable = 1
account.1.label = 1234
account.1.display_name = 1234
account.1.auth_name = 1234
account.1.user_name = 1234
account.1.password = 1234
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
##DNS SRV
account.1.transport = 3
account.1.naptr_build = 0
##Failover Mode
account.1.sip_server.1.failback_mode = 3
account.1.sip_server.1.failback_timeout = 120
account.1.sip_server.1.register_on_enable = 0
account.1.sip_server.2.failback_mode = 0
account.1.sip_server.2.register_on_enable = 0
```

2. 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 IP Phones Auto

Provisioning Guide.

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).

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.
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).

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: 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"	""	_sip._tcp.yealink.pbx.com
IN NAPTR	100	50	"s"	"SIP+D2U"	""	_sip._udp.yealink.pbx.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.
service	Specify the transport protocols supported: SIP+D2U: SIP over UDP 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.

Customer Feedback

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