Calling/Connected Line Identification Presentation on Yealink IP Phones

This guide provides some enhancements of calling and connected line identification presentation supported on Yealink IP phones. Yealink IP phones support to derive calling and connected line identification from SIP headers and display the name associated with the telephone number on the LCD screen.

This guide applies to the following Yealink IP phones:

- SIP-T58A, SIP VP-T49G, SIP-T48G, SIP-T46G, SIP-T42G, SIP-T41P, SIP-T40P, SIP-T29G, SIP-T23P/G, SIP-T21(P) E2, SIP-T19(P) E2, CP860, CP960 and W56P IP phones running firmware version 80 or later

- SIP-T54S, SIP-T52S, SIP-T48S, SIP-T46S, SIP-T42S, SIP-T41S, SIP-T40G, SIP-T27G, CP920, W60P and W52P IP phones running firmware version 81 or later

- W53P and CP930W-Base IP phones running firmware version 83 or later

SIP Headers defined in RFC

FROM

FROM is defined in RFC 3261. This SIP header field indicates the logical identification of the initiator of the request, possibly the user's address-of-record. It contains a URI and optionally a display name.

P-Preferred-Identity

P-Preferred-Identity is defined in RFC 3325. This SIP header field is used from a user agent to a trusted proxy to carry the identification the user sending the SIP message wishes to be used for the P-Asserted-Header field value that the trusted element will insert.

PPreferredID = "P-Preferred-Identity" HCOLON PPreferredID-value

*(COMMA PPreferredID-value)

PreferredID-value = name-addr / addr-spec

A P-Preferred-Identity header field value must consist of exactly one name-addr or addr-spec. There may be one or two P-Preferred-Identity values. If there is one value, it must be a sip, sips, or tel URI. If there are two values, one value must be a sip or sips URI and the other must be a tel URI. It is worth noting that proxies can (and will) remove this header field.

P-Asserted-Identity

P-Asserted-Identity is defined in RFC 3325. This SIP header field is used among trusted SIP entities (typically intermediaries) to carry the identification of the user sending a SIP message as it was verified by authentication.

PAssertedID = "P-Asserted-Identity" HCOLON PAssertedID-value

*(COMMA PAssertedID-value)

PAssertedID-value = name-addr / addr-spec

A P-Asserted-Identity header field value must consist of exactly one name-addr or addr-spec. There may be one or two P-Asserted-Identity values. If there is one value, it must be a sip, sips, or tel URI. If there are two values, one value must be a sip or sips URI and the other must be a tel URI. It is worth noting that proxies can (and will) add and remove this header field.

Remote-Party-ID

Remote-Party-ID is defined in a SIP draft. This SIP header field provides information about the remote party.

Remote-Party-ID = "Remote-Party-ID" HCOLON rpid *(COMMA rpid)

A request or response may contain more than one Remote-Party-ID header field, with privacy requested independently for each.

Privacy

Privacy is defined in RFC 3323. The presence of this privacy type in a Privacy header field indicates that the user would like to be kept private with respect to SIP entities outside the Trust Domain with which the user authenticated.

Privacy-hdr = "Privacy" HCOLON priv-value *(";" priv-value)

Priv-values supported on Yealink IP phones:

Priv-value	Privacy header	Description
id	Privacy: id	Identification must be hidden. (defined in RFC 3325)
none	Privacy: none	Nothing must be hidden.
full	Privacy: full	Behind the Remote-Party-ID header. Both display name and URI from the untrusted entity must be hidden.
uri	Privacy: uri	Behind the Remote-Party-ID header. Only the URI from the untrusted entity must be hidden.
name	Privacy: name	Behind the Remote-Party-ID header. Only the display name from the untrusted entity must be hidden.
off	Privacy: off	Behind the Remote-Party-ID header. Nothing from the untrusted entity must be hidden.

UPDATE

UPDATE is defined in RFC 3311. It is a target refresh request. As specified in RFC 3261, this means that it can update the remote target of a dialog.

Calling Line Identification Presentation

Calling Line Identification Presentation (Abbreviation CLIP), also called Caller ID, Calling Number Identification, is a telephone service available in Voice over Internet Protocol (VoIP). That transmits a caller's number to the called party during the ringing signal. Calling line identification presentation can also provide a name associated with the calling line number. The information transmitted to the called party may be displayed on the phone LCD screen.
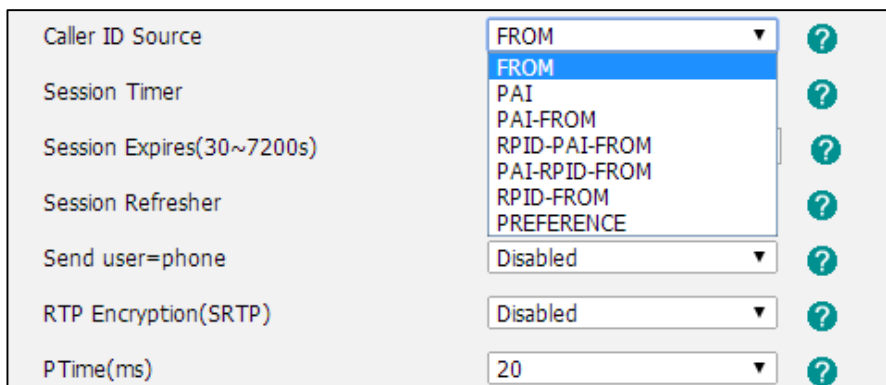
Configuring the Yealink IP Phones

Yealink IP phones support to derive calling line identification from the FROM, P-Preferred-Identity, P-Asserted-Identity and Remote-Party-ID SIP headers in the INVITE request.

Users can configure the calling line identification source for Yealink IP phones via web user interface or using the configuration file. The calling line identification source is configurable on a per-account basis.

To configure calling line identification source via web user interface:

1. Press ⊙ , or press **Menu**->**Status** on the IP phone to check its IP address.

2. Open the web browser on local computer.

3. Enter the IP address (e.g. http://192.168.0.10 or 192.168.0.10) in the browser's address bar, and then click **Enter**.

4. Enter the **User Name** (admin) and **Password** (admin) in the pop-up dialogue box.

5. Click **Log in** to log in the web user interface of the IP phone.

6. Click on **Account**->**Advanced**.

7. Select the desired value for calling line identification source from the pull-down list of **Caller ID Source**.

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

To configure calling line identification source using the configuration file:

1. Add/Edit the calling line identification source parameter in the configuration file.

   The following table lists the information of parameters:

   (For SIP-T58A/SIP VP-T49G/SIP-T54S/T48G/T48S/T46G/T46S/T29G: X ranges from 1 to 16;

   For SIP-T52S/T42G/T42S: X ranges from 1 to 12;

   For W53P/W60P/CP930W-Base: X ranges from 1 to 8;

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

   For W52P/W56P: X ranges from 1 to 5;

   For SIP-T40P/T40G/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/CP920/CP960: X is equal to 1)

Parameters	Permitted Values	Default
account.X.cid_source	0, 1, 2, 3, 4, 5 or 6	0
Description Provides the enhancement for calling line identification presentation. **0**-FROM **1**-PAI **2**-PAI-FROM **3**-PRID-PAI-FROM **4**-PAI-RPID-FROM **5**-RPID-FROM **6**-PREFERENCE If it is set to 6 (PREFERENCE), the IP phone uses the custom priority order for the sources of caller identity information (configured by the parameter "sip.cid_source.preference"). **Web User Interface:** Account->Advanced->Caller ID Source **Phone User Interface:** None		
sip.cid_source.preference	String	P-Preferred-Identity, P-Asserted-Identity, Remote-Party-ID, From
Description: Configures the priority order for the sources of caller identity information. The headers		

Parameters	Permitted Values	Default
can be in any order. **Note**: Yealink IP phones support deriving caller identity from the following SIP headers: From, P-Asserted-Identity (PAI), P-Preferred-Identity and Remote-Party-ID (RPID). **Web User Interface:** None **Phone User Interface:** None		

For example, configure the parameter of account 1 for supporting calling line identification from PAI and FROM SIP headers, the following parameter must be added in the configuration file:

   account.1.cid_source = 2

2. Upload the configuration file to the root directory of the download server and perform auto provisioning to configure the Yealink IP phones.

   For more information on auto provisioning, refer to the latest Auto Provisioning Guide for your phone on Yealink Technical Support.

Calling Line Identification Presentation Enhancement

Yealink IP phones support calling line identification presentation through the FROM, P-Preferred-Identity, P-Asserted-Identity, Remote-Party-ID and Privacy SIP headers. Configuration of calling line identification source decides the feature enhancement on Yealink IP phones. The following sessions show the enhancements of calling line identification presentation according to the calling line identification source configured on the IP phones.

Caller ID source = FROM

1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the calling line identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.
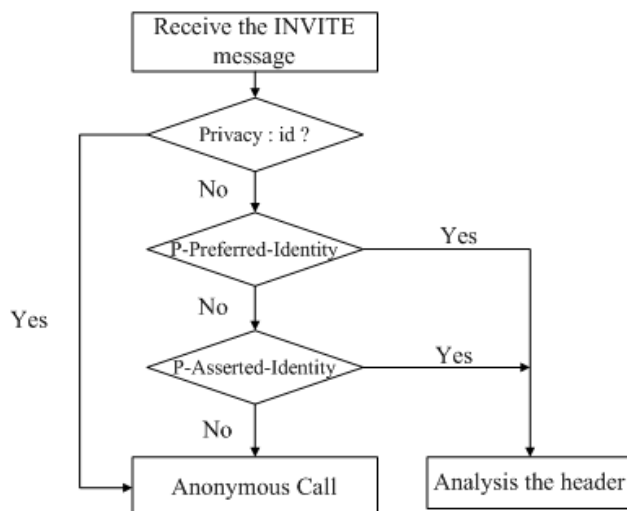
The following process flow diagram for reference:

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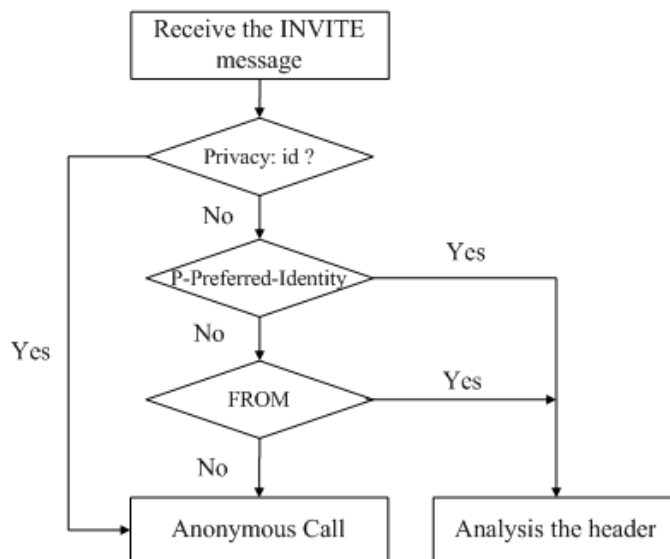
Caller ID source = PAI

1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.
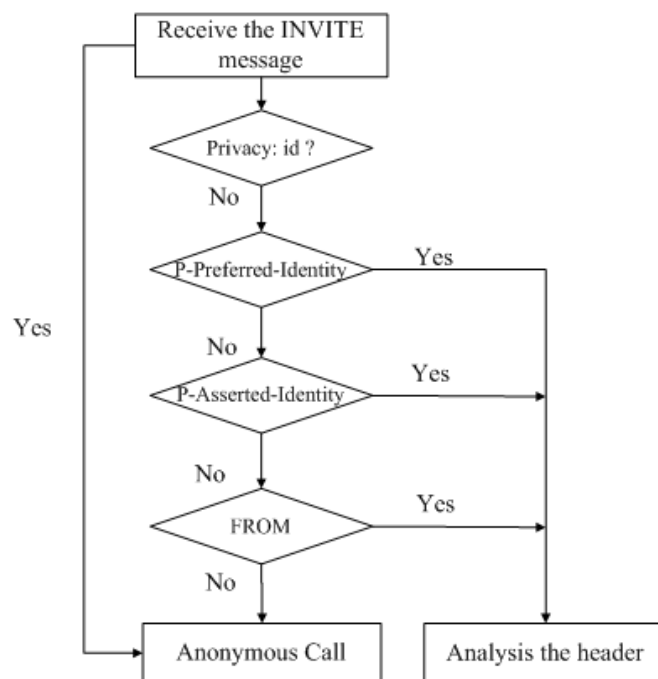
The following process flow diagram for reference:

Caller ID source = PAI-FROM

1) The IP phone checks Privacy: id header preferentially, If there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.

4) If there is not P-Asserted-Identity header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

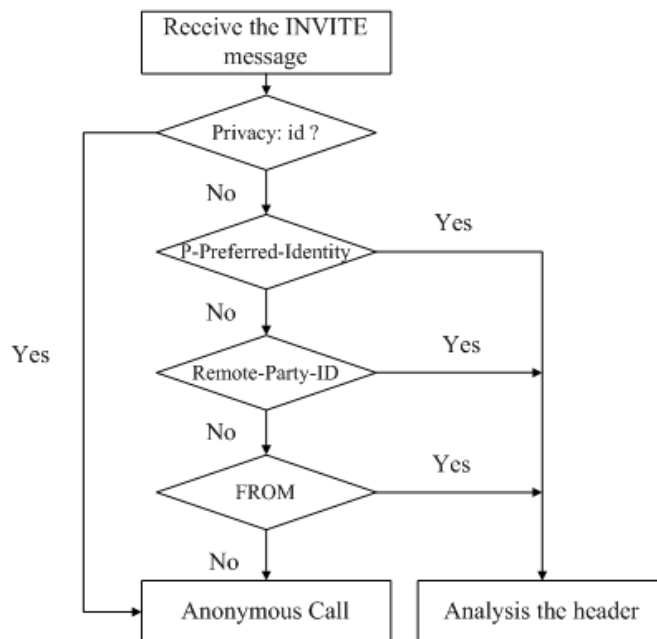The following process flow diagram for reference:

Caller ID source = RPID-FROM

1) The IP phone checks Privacy: id header preferentially, If there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.

4) If there is not Remote-Party-ID header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.
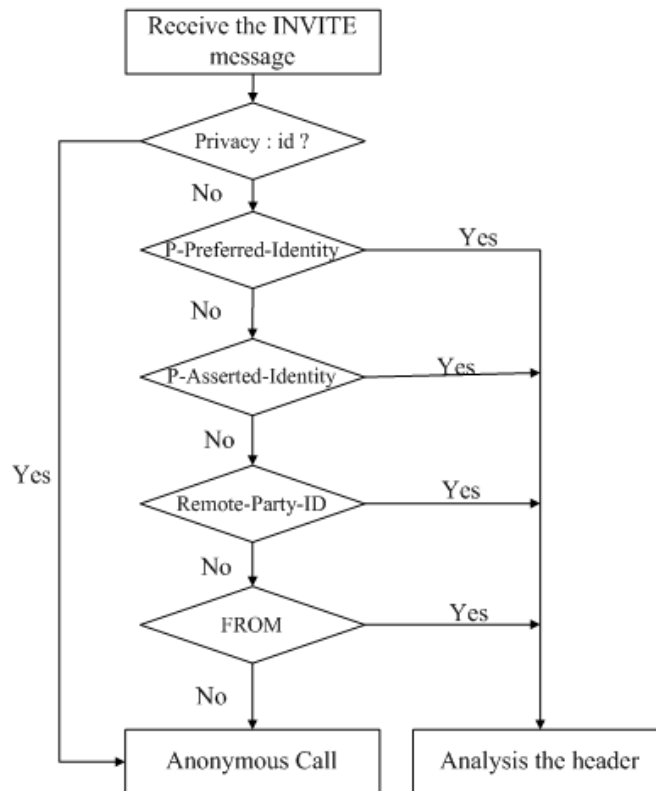
The following process flow diagram for reference:

Caller ID source = PAI-RPID-FROM

1) The IP phone checks Privacy: id header preferentially, If there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.

4) If there is not P-Asserted-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.

5) If there is not Remote-Party-ID header in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

The following process flow diagram for reference:

Caller ID source = RPID-PAI-FROM

1) The IP phone checks Privacy: id header preferentially, If there is a Privacy: id in the INVITE request, the caller identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the INVITE request, the IP phone checks and presents the caller identification from the P-Preferred-Identity header.

3) If there is not P-Preferred-Identity header in the INVITE request, the IP phone checks and presents the caller identification from the Remote-Party-ID header.

4) If there is not Remote-Party-ID header in the INVITE request, the IP phone checks and presents the caller identification from the P-Asserted-Identity header.

5) If there is not P-Asserted-Identity in the INVITE request, the IP phone presents the caller identification derived from the FROM header.

The following process flow diagram for reference:
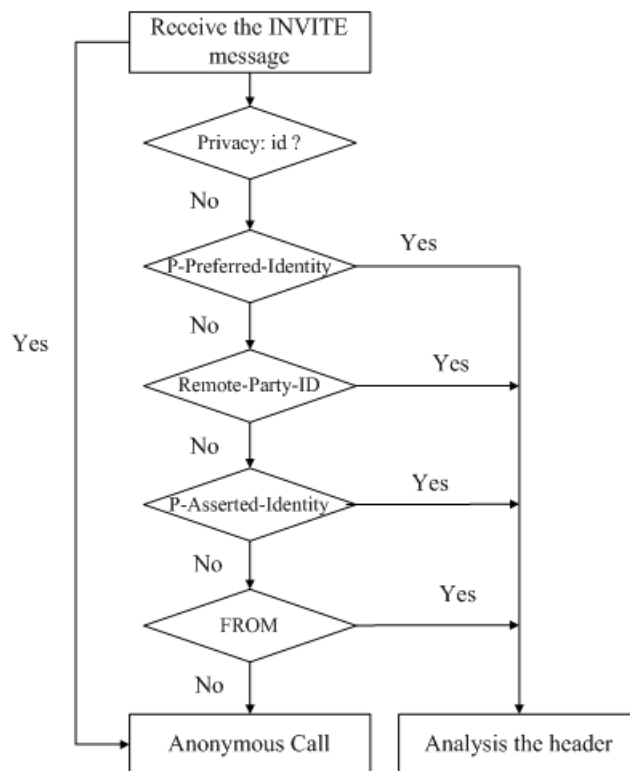
Connected Line Identification Presentation

Connected Line Identification Presentation (Abbreviation COLP) service allows the IP phone to display the identification of the connected party. That transmits a connected line number to the calling party. COLP can also provide a name associated with the connected line number. The information transmitted to the calling party may be displayed on the phone LCD screen.

Configuring the Yealink IP Phone

Yealink IP phones display the dialed digits on the phone LCD screen by default during outgoing calls. During a call establishment, Yealink IP phones support to derive connected line identification from the FROM, P-Preferred-Identity, P-Asserted-Identity and Remote-Party-ID SIP headers in the 18X, 200 OK response. Yealink IP phones also support to derive associated connected line identification from UPDATE message following the RFC 4916.

Users can configure the connected line identification source for Yealink IP phones using the configuration file. The connected line identification source is configurable on a per-account basis.

To configure connected line identification source using the configuration file:

1. Add/Edit the connected line identification source parameter in the configuration file.

   The following table lists the information of parameters:

   (For SIP-T58A/SIP VP-T49G/SIP-T54S/T48G/T48S/T46G/T46S/T29G: X ranges from 1 to 16;

For SIP-T52S/T42G/T42S: X ranges from 1 to 12;

For W53P/W60P/CP930W-Base: X ranges from 1 to 8;

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

For W52P/W56P: X ranges from 1 to 5;

For SIP-T40P/T40G/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/CP920/CP960: X is equal to 1)

Parameters	Permitted Values	Default
account.X.cp_source	0, 1 or 2	0

Description:
Provides the enhancement for connected line identification presentation.   **0**-PAI-RPID   **1**-Dialed Digits   **2**-RFC4916   **Web User Interface:**   None   **Phone User Interface:**   None

For example, configure the parameter of account 1 for supporting connected line identification from PAI and RPID SIP headers, the following parameter must be added in the configuration file:

account.1.cp_source = 0

2. Upload the configuration file to the root directory of the download server and perform auto provisioning to configure the Yealink IP phones.

For more information on auto provisioning, refer to the latest Auto Provisioning Guide for your phone on Yealink Technical Support.
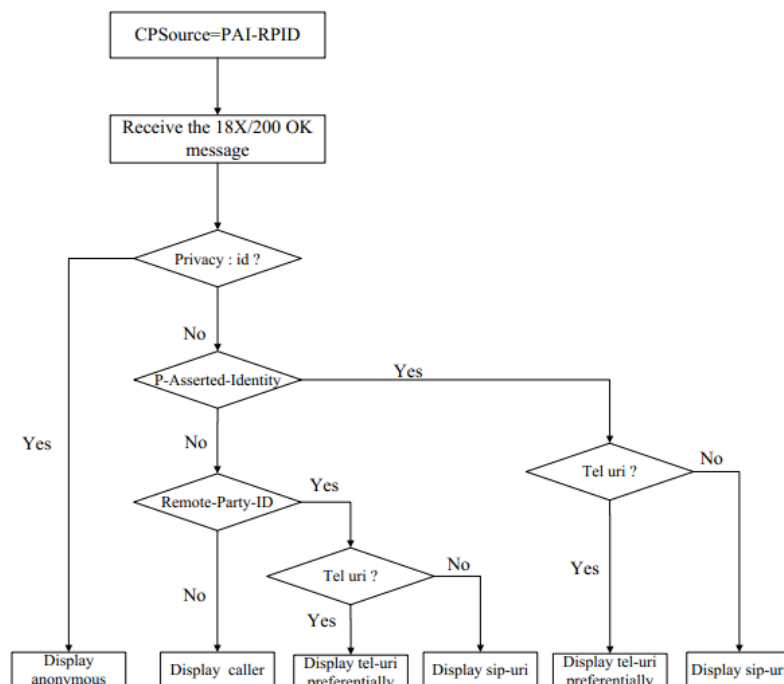
Connected Line Identification Presentation Enhancement

Yealink IP phones support connected line identification through the P-Asserted-Identity, Remote-Party-ID, Privacy SIP headers and UPDATE message following RFC 4916. Configuration of connected line identification source decides the feature enhancement on Yealink IP phones. The following sessions show the enhancements of connected line identification according to the connected line identification source configured on the IP phones.

Connected Line Identification source = PAI-RPID

1) The IP phone checks Privacy: id header preferentially, if there is a Privacy: id in the 18X or 200OK response, the connected line identification information will be hidden and the IP phone LCD screen presents anonymous.

2) If there is not any Privacy: id header in the 18X or 200OK response, the IP phone checks and presents the connected line identification from the P-Asserted-Identity header.

3) If there is not P-Asserted-Identity header in the I8X or 200OK response, the IP phone presents the connected line identification from the Remote-Party-ID header. If no, the IP phone presents the connected line identification according to the dialed digits.

The following process flow diagram for reference:

Connected Line Identification source = Dialed digits

Yealink IP phones present the connected line identification according to the dialed digits.

Connected Line Identification source = RFC4916

Yealink IP phones support to present the connected line identification from UPDATE message following the RFC 4916.

1) The IP phone receives an UPDATE message during a call, the connected line identification on the LCD screen should be refreshed according the FROM SIP carried in the UPDATE message.