



CLIP Selection Mode Description

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1 Introduction

1.1 Overview

Calling Line Identification Presentation (CLIP), also called caller ID, can be used in Voice over Internet Protocol (VoIP). During ringing, the calling number and name are sent to the called party. CLIP supports obtaining the calling number and name from the SIP header and displaying the calling number and name on the screen.

1.2 Applicable Models

This document applies to Fanvil X3S, X4, X5S, and X6.

2 Operation Description

2.1 CLIP Selection Mode Configuration

A telephone set can derive the calling number and name from the FROM, P-Asserted-Identity, or Remote-Party-ID field in an INVITE request. A user can configure a CLIP selection mode for a telephone set on the webpage. A CLIP selection mode can be configured for each line.

The screenshot displays the 'SIP Hotspot' configuration page. The left sidebar contains navigation options: System, Network, Line (selected), Phone settings, Phonebook, Call logs, Function Key, Application, Security, and Device Log. The main content area is titled 'Advanced Settings >>' and is divided into two columns of settings. The 'Caller ID Header' setting is highlighted with a red box and is currently set to 'FROM'. Other settings include various call forwarding options, encryption keys, session timers, and authentication parameters.

| Setting | Value |
|-------------------------------------|-------------------------------------|
| Use Feature Code: | <input type="checkbox"/> |
| Enable DND: | <input type="text"/> |
| Enable Call Forward Unconditional: | <input type="text"/> |
| Enable Call Forward on Busy: | <input type="text"/> |
| Enable Call Forward on No Answer: | <input type="text"/> |
| Enable Blocking Anonymous Call: | <input type="text"/> |
| Call Waiting On Code: | <input type="text"/> |
| Send Anonymous On Code: | <input type="text"/> |
| SIP Encryption: | <input type="checkbox"/> |
| RTP Encryption: | <input type="checkbox"/> |
| Enable Session Timer: | <input type="checkbox"/> |
| Enable BLF List: | <input checked="" type="checkbox"/> |
| Response Single Codec: | <input type="checkbox"/> |
| Keep Alive Type: | UDP |
| Keep Authentication: | <input type="checkbox"/> |
| User Agent: | <input type="text"/> |
| SIP Version: | RFC3261 |
| Local Port: | 5060 |
| Enable user=phone: | <input type="checkbox"/> |
| Auto TCP: | <input type="checkbox"/> |
| Enable Rport: | <input checked="" type="checkbox"/> |
| DNS Mode: | FROM |
| Enable Strict Proxy: | <input type="checkbox"/> |
| Use Quote in Display Name: | <input type="checkbox"/> |
| Sync Clock Time: | <input type="checkbox"/> |
| Caller ID Header: | FROM |
| Enable Feature Sync: | <input type="checkbox"/> |
| CallPark Number: | <input type="text"/> |
| DND Disabled: | <input type="text"/> |
| Disable Call Forward Unconditional: | <input type="text"/> |
| Disable Call Forward on Busy: | <input type="text"/> |
| Disable Call Forward on No Answer: | <input type="text"/> |
| Disable Blocking Anonymous Call: | <input type="text"/> |
| Call Waiting Off Code: | <input type="text"/> |
| Send Anonymous Off Code: | <input type="text"/> |
| SIP Encryption Key: | <input type="text"/> |
| RTP Encryption Key: | <input type="text"/> |
| Session Timeout: | 0 second(s) |
| BLF List Number: | bLF1 |
| BLF Server: | <input type="text"/> |
| Keep Alive Interval: | 60 second(s) |
| Blocking Anonymous Call: | <input type="checkbox"/> |
| Specific Server Type: | BroadSoft |
| Anonymous Call Standard: | None |
| Ring Type: | Default |
| Use Tel Call: | <input type="checkbox"/> |
| Enable PRACK: | <input type="checkbox"/> |
| Enable Long Contact: | <input type="checkbox"/> |
| Convert URI: | <input checked="" type="checkbox"/> |
| Enable GRUU: | <input type="checkbox"/> |
| Enable Use Inactive Hold: | <input type="checkbox"/> |
| Use 182 Response for Call waiting: | <input type="checkbox"/> |
| Enable SCA: | <input checked="" type="checkbox"/> |
| Server Expire: | <input checked="" type="checkbox"/> |

2.2 Configuration Items

A telephone set supports CLIP selection modes based on the FROM, P-Asserted-Identity, or Remote-Party-ID field in an INVITE request. The following describes enhanced functions of configuring CLIP selection modes based on the CLIP header on the telephone set.

2.2.1 FROM

The telephone set displays the calling number and name derived from the FROM field. If the calling number is a contact number, the contact name is displayed.

2.2.2 PAI-FROM

The telephone set checks and displays the calling number and name derived from the

P-Asserted-Identity field.

If the INVITE request does not carry the P-Asserted-Identity field, the telephone set displays the calling number and name derived from the FROM field.

If the calling number is a contact number, the contact name is displayed.

2.2.3 RPID-FROM

The telephone set checks and displays the calling number and name derived from the Remote-Party-ID field.

If the INVITE request does not carry the Remote-Party-ID field, the telephone set displays the calling number and name derived from the FROM field.

If the calling number is a contact number, the contact name is displayed.

2.2.4 PAI-RPID-FROM

The telephone set checks and displays the calling number and name derived from the P-Asserted-Identity field.

If the INVITE request does not carry the P-Asserted-Identity field, the telephone set displays the calling number and name derived from the Remote-Party-ID field.

If the INVITE request does not carry the Remote-Party-ID field, the telephone set displays the calling number and name derived from the FROM field.

If the calling number is a contact number, the contact name is displayed.

2.2.5 RPID-PAI-FROM

The telephone set checks and displays the calling number and name derived from the Remote-Party-ID field.

If the INVITE request does not carry the Remote-Party-ID field, the telephone set displays the calling number and name derived from the P-Asserted-Identity field.

If the INVITE request does not carry the P-Asserted-Identity field, the telephone set displays the calling number and name derived from the FROM field.

If the calling number is a contact number, the contact name is displayed.