



Configuring Cisco UC560 for Spitfire SIP Trunks

This document is a guideline for configuring Spitfire SIP trunks onto Cisco UC560 and includes the settings required for Inbound DDI routing and Outbound CLI presentation. The settings contained within have been tested and are known to work at the time of testing.

SIP trunk details such as account number and password will be provided separately.

SIP Trunk Configuration

- Navigate to Configure/Telephony/Ports and Trunks/SIP Trunk.
- Under Service provider on the Left hand side click on “Add” and configure as below.
- **Account Information:**
- Voice Codec: G711ulaw
- Proxy Server (primary): spitfiresp.net
- Proxy Server (secondary): Leave Blank
- Registrar Server: 83.218.143.23
- Outbound Proxy Server: 83.218.143.23
- Maximum Number of Calls: Check Trunk Details
- Digest Authentication:
- Username – Base number dropping the first 0 and replacing it with 44 (example 442075013046)
- Password: Check Trunk Details
- Domain Name Service:
- SIP Domain Name – spitfiresp.net
- DNS Server Address – 217.13.128.17

SIP Trunk Configuration

- **Advanced Options:**
- Enable Toll Fraud Protection – Checked
- Timers and retries:
- Registrar Server Expiry – 1800
- Number of Register Retries – 10
- Number of Invite retries – 2
- Connect Timer – 100
- Proxy Server Keepalive Timer (active) - 100

SIP Trunk Configuration

- **Service Provider Settings:**
- General:
- Service Provider Name – spitfire
- Preferred Voice Codec – G711ulaw
- Fax Protocol – T38
- DTMF Method – RFC 2833 RTP Payload – 101
- DID Registration – Check (register Caller ID Main Number)
- Numbering Plan Locale – UK
- Alternate Voice Codec – G711alaw

Account Information

System Window Help

Home Configure

Ports Switching Routing Telephony System Ports and Trunks FXS Ports PSTN Trunks **SIP Trunk** Trunk Status Users/Extensions Phone Groups Voice Features Call Handling Dial Plan Site Management Phone Customization Security Device Properties Save Configuration...

Applications Monitor Troubleshoot Maintenance

SIP Trunk

It is recommended to use SIP Trunk on a managed network. However, if an external firewall is not provided, the UC500 Firewall should be configured on the WAN interface. The UC500 Firewall is currently configured.

Service Provider

- None
- spitfire**
- AT&T
- Broadview Networks
- BT
- Cbeyond
- Integra Telecom
- Keyyo
- Megapath
- nTelos
- NUWox
- One Communications
- OpenIP
- Optimum Business
- PAETEC
- Portugal Telecom
- Qwest Communications
- Skype for SIP
- WxC SIP
- XO Communications

Account Information Advanced Options Service Provider Settings

Voice Codec: G711ulaw

Proxy Server (primary): spitfiresp.net Proxy Server (secondary):

Registrar Server: 83.218.143.23

Outbound Proxy Server: 83.218.143.23

Maximum Number of Calls: 20 (1 - 40)

Digest Authentication

Username: 442031418000 Password: *****

Display Password as Plain Text

Domain Name Service

SIP Domain Name: spitfiresp.net DNS Server Address: 217.13.128.17

See also Routing > IP Addresses > Device Configuration.

User Credentials

Username	Password	Realm
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Total Rows: 0 Display Password as Plain Text Add Delete

OK Apply Refresh Cancel Help

Advanced Options

The screenshot displays the Cisco Unified Communications Manager (UCM) configuration interface for a SIP Trunk. The left-hand navigation pane shows the configuration tree with 'SIP Trunk' selected under 'Telephony'. The main content area is titled 'SIP Trunk' and contains a warning message: 'It is recommended to use SIP Trunk on a managed network. However, if an external firewall is not provided, the UCS500 Firewall should be configured on the WAN interface. The UCS500 Firewall is currently configured.' Below this, the 'Service Provider' list includes 'None' (selected), 'spitfire', 'AT&T', 'Broadview Networks', 'BT', 'Cbeyond', 'Integra Telecom', 'Keyyo', 'Megapath', 'nTelos', 'NuVox', 'One Communications', 'OpenIP', 'Optimum Business', 'PAETEC', 'Portugal Telecom', 'Qwest Communications', 'Skype for SIP', 'WxC SIP', and 'XO Communications'. The 'Advanced Options' tab is active, showing 'Toll Fraud Protection' and 'Timers and Retries' sections. The 'Toll Fraud Protection' section includes a checkbox for 'Enable Toll Fraud Protection (recommended)' which is checked, and a table for 'Additional Allowed IP Addresses' with 'Total Rows: 0' and 'Add'/'Delete' buttons. The 'Timers and Retries' section contains several settings with input fields and ranges: Registrar Server Expiry (1800, 60 - 65535 seconds), Number of Register Retries (10, 1 - 10), Number of Invite Retries (2, 1 - 10), Connect Timer (100, 100 - 1000 milliseconds), and Proxy Server Keepalive Timer (active) (100, 10 - 600 seconds). At the bottom, there are 'Add' and 'Delete' buttons for the service provider list, and 'OK', 'Apply', 'Refresh', 'Cancel', and 'Help' buttons for the configuration dialog.

system window Help

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FXS Ports

PSTN Trunks

SIP Trunk

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Voice Features

Call Handling

Dial Plan

Site Management

Phone Customization

Security

Device Properties

Save Configuration...

Applications

Monitor

Troubleshoot

Maintenance

SIP Trunk

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Service Provider

None

spitfire

AT&T

Broadview Networks

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Cbeyond

Integra Telecom

Keyyo

Megapath

nTelos

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Optimum Business

PAETEC

Portugal Telecom

Qwest Communications

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WxC SIP

XO Communications

Account Information Advanced Options Service Provider Settings

Toll Fraud Protection

It is recommended to use SIP Trunk on a managed network. However, if an external firewall is not provided, the UCS500 Firewall should be configured on the WAN interface. The UCS500 Firewall is currently configured.

Additional Allowed IP Addresses

IP Address

Enable Toll Fraud Protection (recommended)

Total Rows: 0 Add Delete

Timers and Retries

Registrar Server Expiry: 1800 (60 - 65535 seconds)

Number of Register Retries: 10 (1 - 10)

Number of Invite Retries: 2 (1 - 10)

Connect Timer: 100 (100 - 1000 milliseconds)

Proxy Server Keepalive Timer (active): 100 (10 - 600 seconds)

Add Delete

OK Apply Refresh Cancel Help

Service Provider Settings

The screenshot shows the Cisco Unified Communications Manager (CUCM) interface for configuring a SIP Trunk. The left sidebar contains navigation options: Home, Configure, Ports, Switching, Routing, Telephony (System, Ports and Trunks, Users/Extensions, Phone Groups, Voice Features, Call Handling, Dial Plan, Site Management, Phone Customization), Security, Device Properties, and Save Configuration... The main area is titled "SIP Trunk" and displays a list of service providers. The "spitfire" provider is selected and highlighted in green. The "Service Provider Settings" tab is active, showing configuration options for the selected provider. The "General" sub-tab is selected, displaying fields for Service Provider Name, Numbering Plan Locale, Preferred Voice Codec, Alternate Voice Codec, Fax Protocol, and DTMF Method. The "DID Registration" section includes radio buttons for Register Caller ID Main Number, Register All DIDs Using Same Password, DIDs Register Using Different Passwords, and Do Not Register DIDs. The "Import" and "Export" buttons are visible at the bottom of the configuration area.

system window help

CISCO

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PSTN Trunks

SIP Trunk

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SIP Trunk

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Service Provider

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spitfire

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Qwest Communications

Skype for SIP

WxC SIP

XO Communications

Account Information

Advanced Options

Service Provider Settings

General

Session Data

If a SIP Service Provider template file has been provided to you, select Import to update settings. It is also recommended to export new SIP Service Providers to a template file for backup and import to other CCA installations.

For calls to voicemail or auto attendant, transcoding may be required depending on voice codec and DTMF method. Transcoding is currently required.

Service Provider Name: spitfire

Numbering Plan Locale: UK

Preferred Voice Codec: G711ulaw

Alternate Voice Codec: G711alaw

Fax Protocol: T38

DTMF Method: RFC 2833

RTP Payload: 101

DID Registration:

Register Caller ID Main Number

Register All DIDs Using Same Password

DIDs Register Using Different Passwords

Do Not Register DIDs

Import

Export

OK

Apply

Refresh

Cancel

Help

Add

Delete

Configuring Incoming Calls and Outgoing Presentation

- Navigate to Configure/Telephony/Dial Plan and expand the options.
- Click on **Incoming** and select the **Direct Dialing tab**, from here you can configure the DID for both users and groups by selecting “Add”.

Direct Dial to Internal User Extensions

Description:	DID Range Start	DID Range End	Internal Number Start	Internal Number End	Trunks
8001	442031418001	442031418001	8001	8001	SIP Trunk
8002	442031418002	442031418002	8002	8002	SIP Trunk
8003	442031418003	442031418003	8003	8003	SIP Trunk
8004	442031418004	442031418004	8004	8004	SIP Trunk
8005	442031418005	442031418005	8005	8005	SIP Trunk
8011	442031418011	442031418011	8011	8011	SIP Trunk

Total Rows: 21

Direct Dial to Auto-Attendant, Groups, Operator

Description:	DID Range Start	DID Range End	Destination Type	Destination	Trunks
TeamWealth	442031418020	442031418020	BLAST_GROUP	Blast Group:3(8020)	SIP Trunk
TESTDIV	442031418050	442031418050	HUNT_GROUP	Hunt Group:10(5009)	SIP Trunk
TeamSys	442031418010	442031418010	BLAST_GROUP	Blast Group:2(8010)	SIP Trunk
GQR	442031418000	442031418000	BLAST_GROUP	Blast Group:1(8000)	SIP Trunk
CityIntern	442031418080	442031418080	BLAST_GROUP	Blast Group:4(8080)	SIP Trunk
Rostron	442031418100	442031418100	HUNT_GROUP	Hunt Group:1(8100)	SIP Trunk

Outgoing Call Options

Outgoing Call Handling PSTN Trunk Groups Caller ID

Numbering Plan Locale: UK-8-Digit-Local-Numbers

Default Access Code: 9

Digit Collection Timeout [2-120]: 5 seconds

Outgoing Numbers

Permissions	Description	Access Code ▲	Begins With	Number of Digits	Dial Pattern	Trunk Priority	Configure Priority
International	International	9	00[1-9]	Variable	900[1-9]T	SIP then PSTN	Configure Priority
Local	Non-Emergency Police/Health Ser	9	1[01]1	3	91[01]1	SIP then PSTN	Configure Priority
Local	Local Rate NGN - <= 10 digits	9	0845	Variable	90845T	SIP then PSTN	Configure Priority
Local	local direct dial	9	[2-9]	8	9[2-9]xxxxxxx	SIP then PSTN	Configure Priority
Local	Local Rate NGN - 11 digit	9	0845	11	90845xxxxxxx	SIP then PSTN	Configure Priority
Local-Plus	Level1 Services	9	15[0-4]	Variable	915[0-4]T	SIP then PSTN	Configure Priority
Local-Plus	Speaking Clock	9	123	3	9123	SIP then PSTN	Configure Priority
National	Services of Social Value	9	116	6	9116xxx	SIP then PSTN	Configure Priority
National	National Non Geo - 11 digits	9	0870	11	90870xxxxxxx	SIP then PSTN	Configure Priority

OK Apply Refresh Cancel Help

Setting Caller ID

- Click on the spitfire SIP Trunk to highlight it.
- To program individual DID presentation you will need to select “Add” and then insert the extension number along with the DID you wish to present on Outgoing Calls.

The screenshot displays a configuration interface for an 'Outgoing Dial Plan'. The left sidebar shows a navigation tree with 'Outgoing' selected under 'Dial Plan'. The main area shows a table with one row: 'SIP(spitfire)' with 'Internal Extensions' and a 'Caller ID' of '442031418000'. Below the table, a message states: 'The default caller ID for calls going out through SIP(spitfire) is 442031418000. Caller IDs for specific extensions to override 442031418000 may be added here.' A table with columns 'Primary Extension' and 'Caller ID' is visible. An 'Add Caller ID for Internal Extensions' dialog box is open, containing the following fields:

- Internal Extension Start Number:
- Internal Extension End Number:
- Caller ID Start Number:
- Caller ID End Number:

Buttons for 'Add', 'Modify', 'Delete', 'OK', 'Cancel', and 'Help' are present in the dialog and the main interface.