

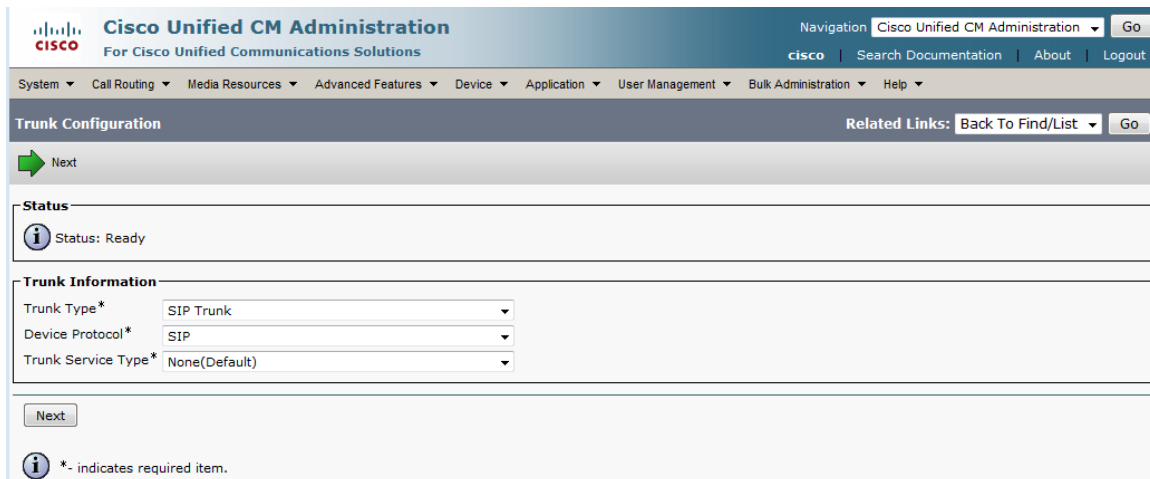
CUCM 11 - How to connect 2N® Helios IP via SIP trunk?

i This FAQ describes how to connect 2N® Helios IP intercom to CUCM 11 via SIP trunk. Connection via SIP trunk will enable calls from intercom to CUCM and its main advantage is that you don't need additional "device" licence for CUCM. FAQ is divided into the two parts. First part describes how to configure CUCM 11 and second part describes configuration of 2N® Helios IP intercom.

How to configure CUCM 11

Configuration of SIP trunk

Go to section "Device->Trunk", set parameters regarding the picture below and click on "Next".



The screenshot shows the Cisco Unified CM Administration interface. The page title is "Trunk Configuration". The navigation menu includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Device" menu is expanded, and "Trunk Configuration" is selected. The page contains a "Next" button with a green arrow, a "Status" section showing "Status: Ready", and a "Trunk Information" section with three dropdown menus: "Trunk Type*" (SIP Trunk), "Device Protocol*" (SIP), and "Trunk Service Type*" (None(Default)). A "Next" button is located below the "Trunk Information" section. A legend at the bottom indicates that "*" indicates a required item.

Continue in configuration according to the pictures below.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Trunk Configuration Related Links: [Back To Find/List](#)

Status
 Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="Helios_IP_trunk"/>
Description	<input type="text" value="Helios IP trunk"/>
Device Pool*	<input type="text" value="Testteam-dp"/>
Common Device Configuration	<input type="text" value="< None >"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="Testteam-mrgl"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="< None >"/>
Tunneled Protocol*	<input type="text" value="None"/>
QSIG Variant*	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

Media Termination Point Required
 Retry Video Call as Audio
 Path Replacement Support
 Transmit UTF-8 for Calling Party Name
 Transmit UTF-8 Names in QSIG APDU
 Unattended Port
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure*

Route Class Signaling Enabled*

Use Trusted Relay Point*

PSTN Access
 Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.164 Transformation Profile

MLPP and Confidential Access Level Information

MLPP Domain

Confidential Access Mode

Confidential Access Level

Call Routing Information

Remote-Party-Id

Asserted-Identity

Asserted-Type*

SIP Privacy*

Inbound Calls

Significant Digits*

Connected Line ID Presentation*

Connected Name Presentation*

Calling Search Space

AAR Calling Search Space

Prefix DN

Redirecting Diversion Header Delivery - Inbound

Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="None"/>	<input checked="" type="checkbox"/>

Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="None"/>	<input checked="" type="checkbox"/>

Connected Party Settings

Connected Party Transformation CSS

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

Use Device Pool Redirecting Party Transformation CSS

Caller Information

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status
1*	10.0.27.71		5060	N/A

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Configuratin of trunk security

In next step configure trunk security (password which will be used for registration to the trunk). Security is configured in section "System->Security->SIP Trunk Security Profile". Click on "Add New".

Cisco Unified CM Administration
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Navigation Cisco Unified CM Administration

System

Find and List SIP Trunk Security Profiles

SIP Trunk Security Profile

Find SIP Trunk Security Profile where begins with

No active query. Please enter your search criteria using the options above.

Configure the rest according to the pictures below.

Cisco Unified CM Administration
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Navigation: Cisco Unified CM Administration | Go

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SIP Trunk Security Profile Configuration | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Status
Status: Ready

SIP Trunk Security Profile Information

Name* Non Secure SIP Trunk Profile
 Description Non Secure SIP Trunk Profile authenticated by null String
 Device Security Mode Non Secure
 Incoming Transport Type* TCP+UDP
 Outgoing Transport Type UDP

Enable Digest Authentication
 Nonce Validity Time (mins)* 600
 X.509 Subject Name
 Incoming Port* 5060

Enable Application level authorization
 Accept presence subscription
 Accept out-of-dialog refer**
 Accept unsolicited notification
 Accept replaces header
 Transmit security status
 Allow charging header
 SIP V.150 Outbound SDP Offer Filtering* Use Default Filter

Save | Delete | Copy | Reset | Apply Config | Add New

*. indicates required item.
 **If this profile is associated with an EMCC SIP trunk, Accept Out-of-Dialog REFER is enabled regardless of the setting on this page

Add user

In this step add a user in section "User Management->Application User". Click on "Add New".

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Find and List Application Users

+ Add New

Application User

Find Application User where User ID begins with [] Find Clear Filter [] []

No active query. Please enter your search criteria using the options above.

Add New

Continue with configuration according to pictures below.

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Application User Configuration | Related Links: Back To Find/List | Go

Save

Status
Status: Ready

Application User Information

User ID*: 4999
 Password: ●●●●
 Confirm Password: ●●●●
 Digest Credentials: ●●●●
 Confirm Digest Credentials: ●●●●
 BLF Presence Group*: Standard Presence group

Accept Presence Subscription
 Accept Out-of-dialog REFER
 Accept Unsolicited Notification
 Accept Replaces Header

Device Information

Available Devices: Auto-registration Template, SEP111122221000, SEP7C1EB3010D70, SEPACA0166F2266, Sample Device Template with TAG usage examples

Controlled Devices: [Empty list]

Available Profiles: [Empty list]

CTI Controlled Device Profiles: [Empty list]

Device Association | Find more Route Points

Add route pattern

In next step create route pattern (number which will be called by the intercom). Add Route pattern in section "Call Routing->Route->Route Pattern". Click "Add New".

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System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Find and List Route Patterns

+ Add New

Route Patterns

Find Route Patterns where: Pattern | begins with | Find | Clear Filter | + | -

No active query. Please enter your search criteria using the options above.

Add New

Continue with configuration according to pictures below.

Cisco Unified CM Administration
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Route Pattern Configuration Related Links: Back To Find/List Go

Save

Status
Status: Ready

Pattern Definition

Route Pattern* 4999
 Route Partition < None >
 Description To Helios IP
 Numbering Plan -- Not Selected --
 Route Filter < None >
 MLPP Precedence* Default
 Apply Call Blocking Percentage
 Resource Priority Namespace Network Domain < None >
 Route Class* Default
 Gateway/Route List* Helios_IP_trunk (Edit)
 Route Option
 Route this pattern
 Block this pattern No Error
 Call Classification* OffNet
 External Call Control Profile < None >
 Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority
 Require Forced Authorization Code
 Authorization Level* 0
 Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask
 Calling Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Calling Line ID Presentation* Default
 Calling Name Presentation* Default
 Calling Party Number Type* Cisco CallManager
 Calling Party Numbering Plan* Cisco CallManager

Connected Party Transformations

Connected Line ID Presentation* Default
 Connected Name Presentation* Default

Called Party Transformations

Discard Digits < None >
 Called Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Called Party Number Type* Cisco CallManager
 Called Party Numbering Plan* Cisco CallManager

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --
 Carrier Identification Code
 Network Service -- Not Selected -- Service Parameter Name < Not Exist > Service Parameter Value

Save

* - indicates required item.

How to configure 2N® Helios IP intercom

Now you can register the intercom to the newly created SIP trunk. Credentials for this example are 4999/4999 an

address of CUCM 11 is 10.0.27.36. Instead of this address fill here address of your CUCM.

2N Helios IP Force CZ | EN | DE | FR | IT | ES | RU Logout

SIP 1 SIP 2 Calls Audio Video 2N Indoor Touch

Services

- Phone >
- Streaming
- Onvif
- E-Mail
- Automation
- HTTP API
- User Sounds
- Web Server
- Audio Test
- SNMP

Intercom Identity ▾

Display Name	2N Helios IP Force
Phone Number (ID)	4999
Domain	10.0.27.36
	Test Call

Authentication ▾

Use Authentication ID	<input checked="" type="checkbox"/>
Authentication ID	4999
Password	••••••••

SIP Proxy ▾

Proxy Address	10.0.27.36
Proxy Port	5060

SIP Registrar ▾

Registration Enabled	<input type="checkbox"/>
Registrar Address	192.168.1.1
Registrar Port	5060
Registration Expires	120 [s]

Advanced Settings >