

SIP Trunk configuration

Via a Cisco Unified Communications Manager SIP Trunk, incoming calls are forwarded to Callisto, where outgoing calls will be set up.

Add a new SIP Trunk via Device > Trunk

Trunk Information	
Trunk Type*	SIP Trunk
Device Protocol*	SIP
Trunk Service Type*	None(Default)

Check Transmit UTF-8 for Calling Party Name

Device Information	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Callisto_SIP_Trunk
Description	Callisto_SIP_Trunk
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input checked="" type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	

For Encrypted Communication: Check the SRTP option.

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security.
Consider Traffic on This Trunk Secure* Bei Verwendung von sRTP und TLS ▼

At Call Routing Information, check Remote-Party-ID and Asserted-Identity.

At Inbound Calls, check Redirecting Diversion Header Delivery – Inbound.

At Outbound Calls, check Redirecting Diversion Header Delivery – Outbound.

Outbound Calls

Called Party Transformation CSS < None > ▼

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None > ▼

Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator ▼

Calling Line ID Presentation* Default ▼

Calling Name Presentation* Default ▼

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None > ▼

Use Device Pool Redirecting Party Transformation CSS

Via SIP Information, enter the Callisto IP address at Destination Address.

For non-secure communication, set Destination Port to 5060.

To encrypt the communication, set this port to 5061.

SIP Information

Destination Address is an SRV

Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1* 192.168.100.90		5060	N/A	N/A	N/A

MTP Preferred Originating Codec* 711ulaw ▼

BLF Presence Group* Standard Presence group ▼

SIP Trunk Security Profile* Callisto SIP Trunk Security Profile ▼

Rerouting Calling Search Space < None > ▼

Out-Of-Dialog Refer Calling Search Space < None > ▼

SUBSCRIBE Calling Search Space < None > ▼

SIP Profile* Callisto SIP Profile ▼ [View Details](#)

DTMF Signaling Method* Keine Voreinstellung ▼

At SIP Profile, select the profile you have setup in point 1.1.

At SIP Trunk Security Profile, select the profile you have setup in point 1.2.

All Calling Search Space related settings have to be configured according the overall CUCM configuration.