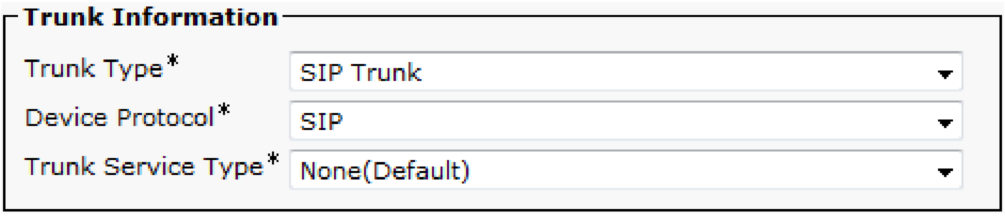
SIP Trunk configuration

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

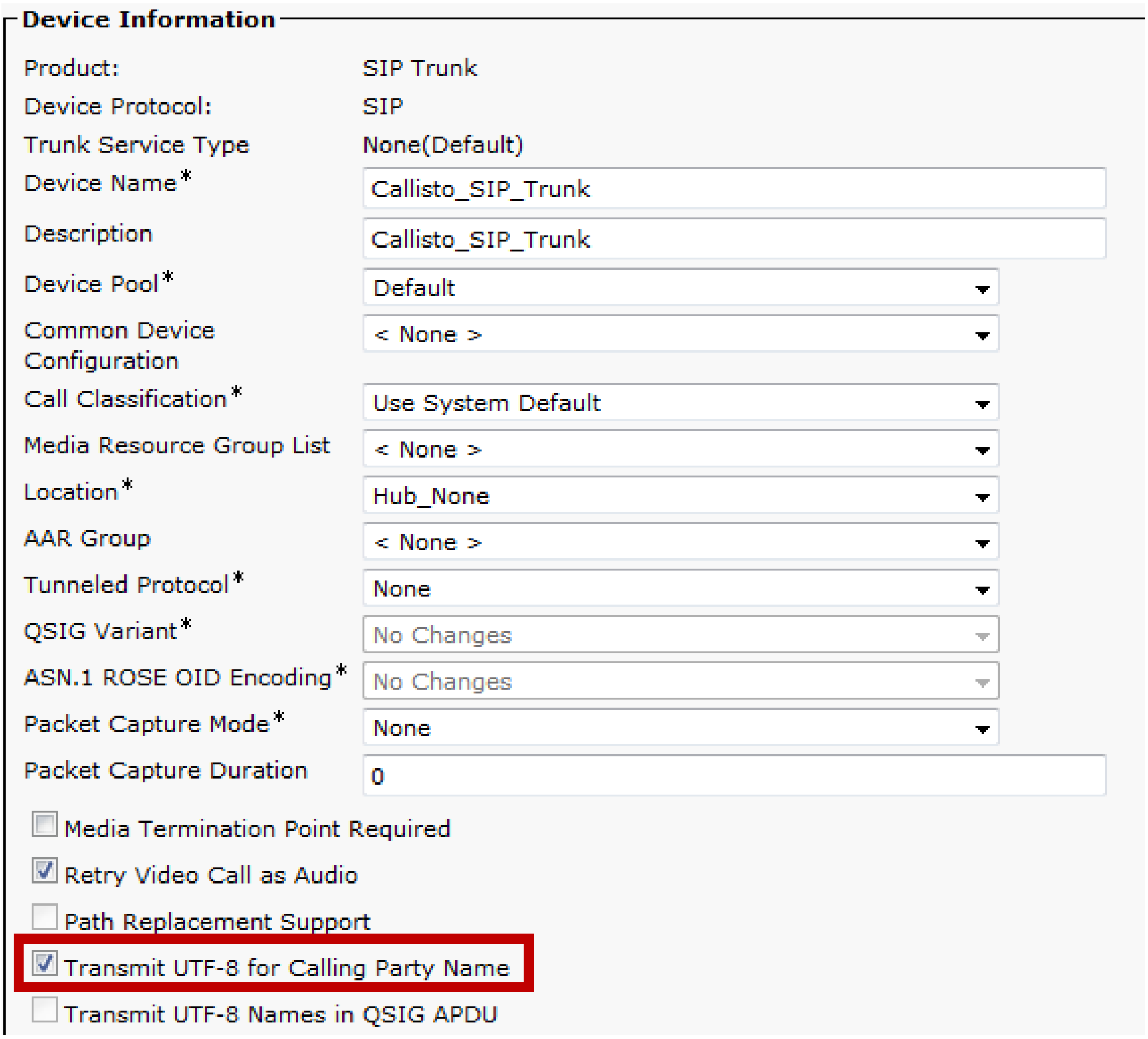
Add a new SIP Trunk via Device > Trunk



Trunk Information

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

Check Transmit UTF-8 for Calling Party Name

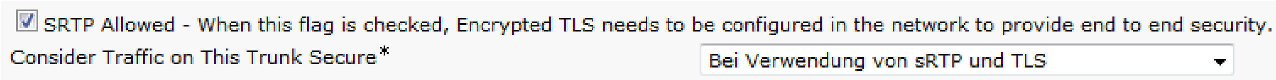


Device Information

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

- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [x] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU

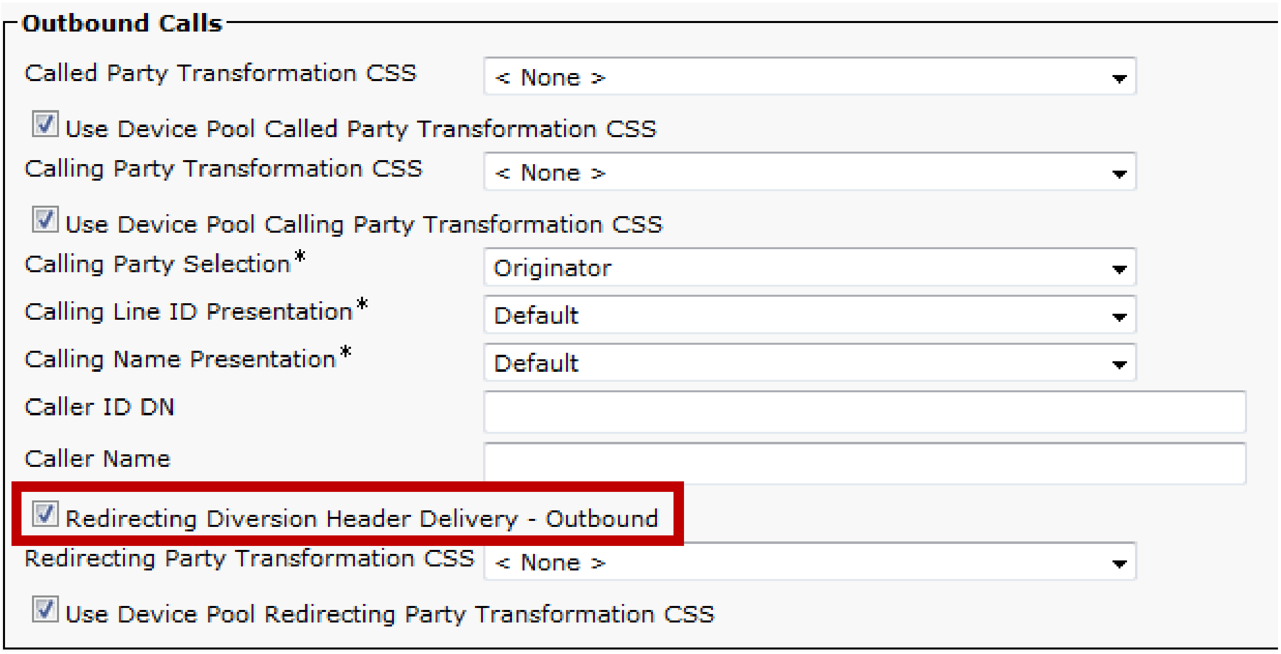
For Encrypted Communication: Check the SRTP option.



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

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

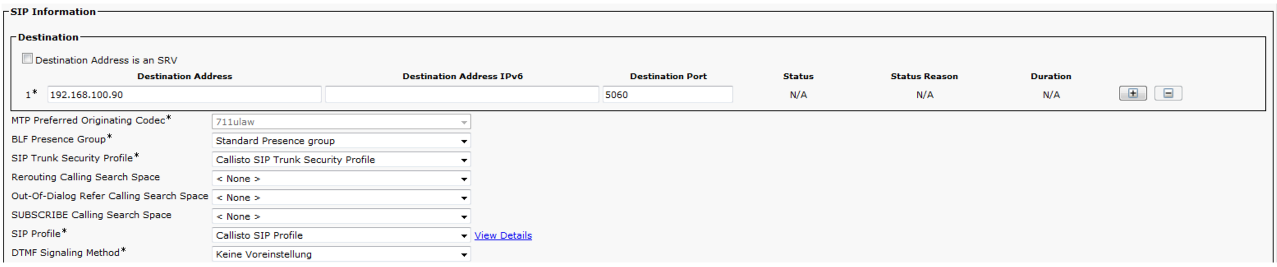
At Outbound Calls, check Redirecting Diversion Header Delivery – Outbound.



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.



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.