Templates Cisco UC560 :

ENREGISTREMENT TRUNK SIP

```
********************************************************
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
supplementary-service h450.12
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
sip
 registrar server expires max 120 min 60
localhost dns:openip.local
no update-callerid

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
sip-ua
credentials username LOGIN_SIP password 0 PASSWORD realm sip5.voip-centrex.net
authentication username LOGIN_SIP password 0 PASSWORD
no remote-party-id
retry invite 2
retry register 10
timers connect 100
registrar dns:sip5.voip-centrex.net:5060 expires 120
sip-server dns:sip5.voip-centrex.net:5060
host-registrar
********************************************************
```

APPEL SORTANT

```
********************************************************
voice translation-rule 1
rule 9 /.*/ /0123456789/
!
voice translation-profile CallerID_Calling_1
translate calling 1
!
dial-peer voice 1001 voip
description outgoing_sip_call
translation-profile outgoing CallerID_Calling_1
destination-pattern 0T
session protocol sipv2
session target sip-server
dtmf-relay rtp-nte
ip qos dscp cs5 media
```

ip qos dscp cs4 signaling
codec g711ulaw
no vad
****

APPEL ENTRANT
****
voice translation-rule 1
rule 1 /0123456789/ /100/
rule 15 /.*/ /100/
!
voice translation-profile 0123456789_Called_1
translate called 1
!
dial-peer voice 3000 voip
description 0123456789
translation-profile incoming 0123456789_Called_1
session protocol sipv2
session target sip-server
incoming called-number 0123456789
voice-class codec 1
voice-class sip dtmf-relay force rtp-nte
dtmf-relay rtp-nte
ip qos dscp cs5 media
ip qos dscp cs4 signaling
no vad
****
