OpenScape Voice V9

How to Configure SIP Trunk for VoipVoice

Tested Scenario: Provider <–> OpenScape SBC <–> OpenScape Voice
Provider Topology: SIP Peer Trunk (Static trunk)

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Table of History

Date	Version	Changes
18.02.2020	0.1	First version
17.03.2020	0.9	Ready for review
13.05.2020	1.0	Released

Information

VoipVoice offers 2 products:

1. Registered trunk via domain **trunk.voipvoice.it** (IP 77.239.128.7)
2. SIP Peer Trunk (static trunk) via SIP Server IP **77.239.128.13**. This trunk is offered for customers with more than 10 simultaneous calls

This configuration guide described only the second product "SIP Peer Trunk"

Trunk Configuration Data provided by VoipVoice

Trunk connection: SIP Peer Trunk (static trunk). The trunk is linked to the SBC public Access IP.

Transport Protocol: UDP / TCP. UDP was tested only

Signaling IP: 77.239.128.0/24; **Signaling Port**: 5060

Media IP: 77.239.128.0/24

Provider settings: The trunk has no Openscape specific settings.

Documentation: Configurazione linee VoipVoice su VOIspeed

The configuration data needed to setup the SIP trunk will be provide via email.

The email can look like this:

VoipVoice

Gentile Cliente,

La informiamo che la data di possibile passaggio della portabilità dei numeri in oggetto di Sua proprietà, è prevista per il <%data1%>.

Di seguito i nuovi parametri di configurazione.

Il vostro numero Voip è: <%numeri_telefono_digits%> (Interni dal <%numeri_telefono_digits_da%> al <%numeri_telefono_digits_a%>)

PARAMETRI DI CONFIGURAZIONE:

Tipologia: GNR 10 in modalità SIP Peer Trunk
Numerazioni Assegnate: interni dal 123456789 al 123456789
Numerazione Primaria: 123456789
IP Trunk di destinazione: 77.239.128.13
IP Trunk di destinazione: IP DEL CLIENTE

CODIFICA AUDIO:

Codecs Supportati: G.729a / G729 / G.711a (PCMA) / G.711u (PCMU)
Codec con Suppporto Fax:
• T38 con commutazione esclusiva da codec G711a
• G711a passthrough
Codifica Toni DTMF:
• RFC2833 con G729, G711a, G711u (RACCOMANDATO)
• Inband con G711a e G711u

Si raccomanda l'uso dei toni in modalità RFC2833. Solo in caso di incompatibilità del PBX, utilizzare la modalità Inband.
Per la trasmissione dei FAX in T38 da PBX è necessario che la funzionalità T38 sia attiva e che sia selezionato il codec G711a.

CHIAMATE INTERNAZIONALI

La informiamo che le chiamate internazionali sono DISABILITATE.
Per richiederne l'attivazione si prega di inviare una mail a servizioclienti@voipvoice.it

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ATTENZIONE ALLE FRODI – PROTEZIONI ACL

VoipVoice non è responsabile dell'eventuale traffico anomalo verso le numerazioni estere.
Vi preghiamo di porre in atto tutte le eventuali politiche di Sicurezza e di monitorare gli apparati Voip con i quali state effettuando le chiamate.
Al fine di prevenire le frodi sul Vostro IP-PBX è raccomandabile inserire, tramite il vostro apparato router, una Access Control List (ACL) verso gli IP di negoziazione VoipVoice.
A tal fine si raccomanda di includere l'intera subnet 77.239.128.13/24 e:

- Utilizzare password alfanumeriche complesse cambiandole periodicamente di almeno 8 caratteri per:
- Tutti gli interni
- L'interfaccia di amministrazione
- Tutte le macchine fax
- Porre Attenzione agli interni remoti (VPN con IP Statici)

Buon VoIP!
A disposizione per qualsiasi richiesta.

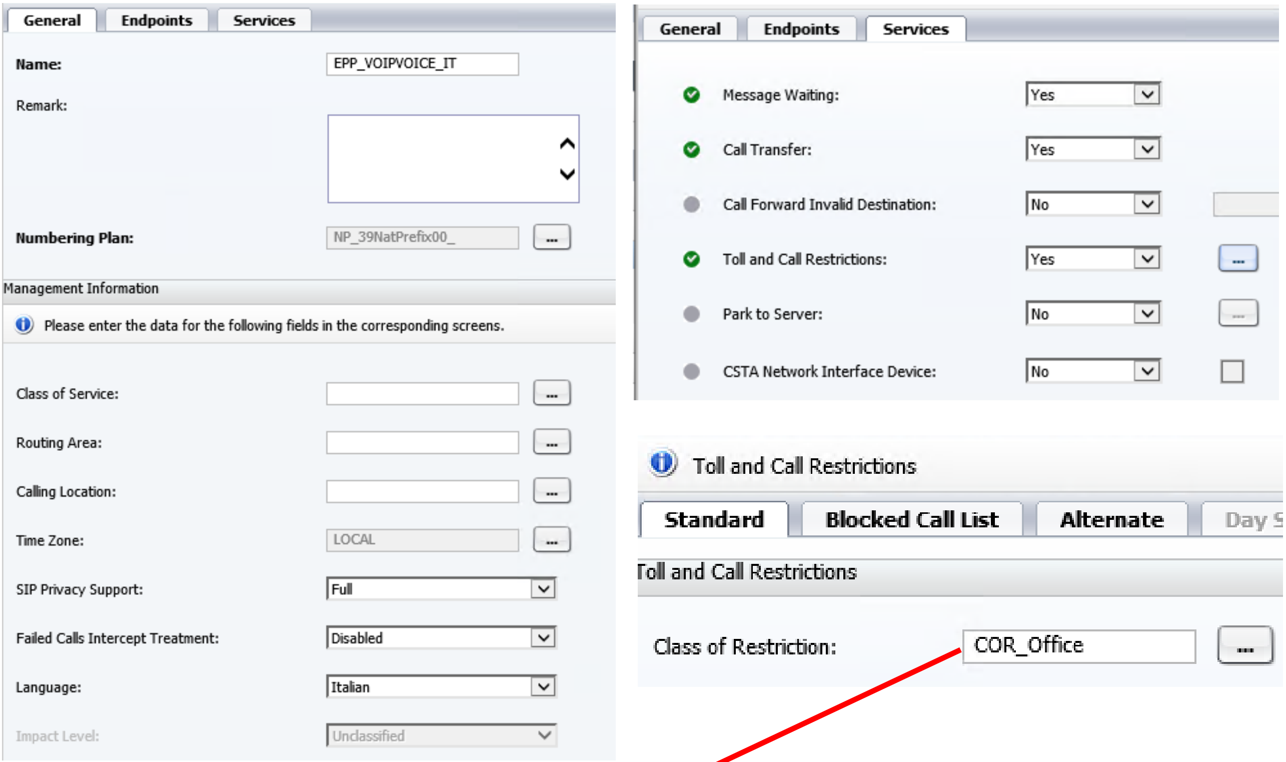
VoipVoice

Via del Lavoro, 8 50056 Montelupo Fiorentino (FI)
Num. VoIP: 0550935400 VoIPFax: 0550935522
PEC:
info@pec.voipvoice.it

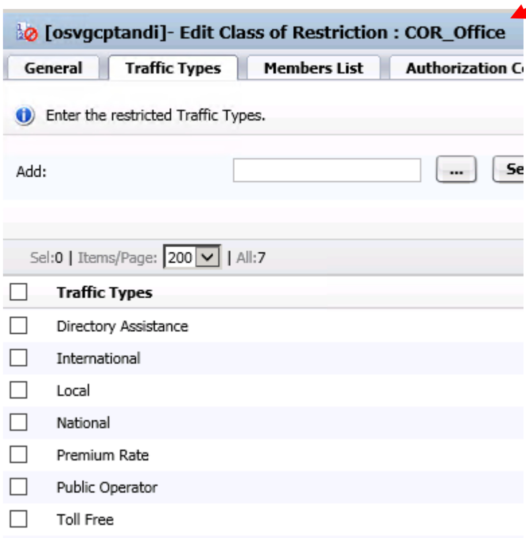
OpenScape Voice Configuration

Endpoint Profile

- **Numbering Plan**: create a specific gateway numbering plan. This helps you to be flexible for incoming routing and DNM rules based on the numbering plan.
- **SIP Privacy Support**: “Full”
- **Language**: select the country language
- **Toll and Call Restrictions**: use a COR which doesn’t allows external to external transit calls.



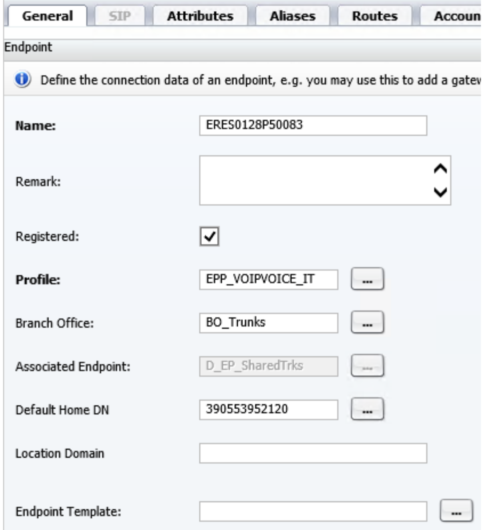
COR_Office must have all external Traffic Types assigned



Endpoint

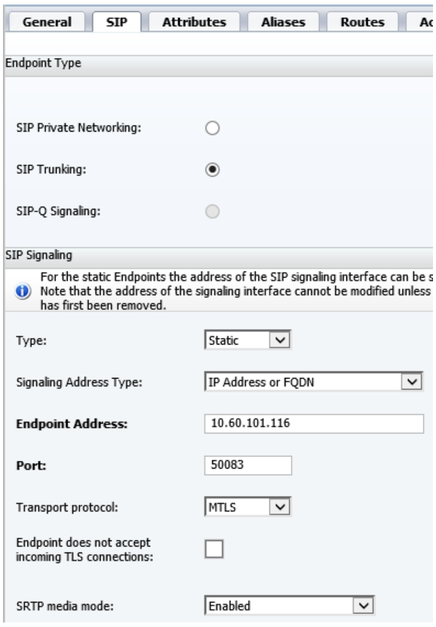
General Tab

- **Registered**: enable
- **Profile**: use the endpoint Profile configured before
- **Default Home DN**: use one number of the DID range



SIP Tab

- **Endpoint Type**: “SIP Trunking”
- **Type**: “Static”
- **Endpoint Address**: enter SBC IP Address
- **Port**: enter the SBC RemoteEndpoint Core Port
- **Transport protocol**: select “TCP” or “MTLS”



Attributes Tab

- **Public/Offnet Traffic**: enable
- **Send Redirect Number instead of calling number for redirected calls**: enable (Provider doesn’t support CLI numbers which are not owned by the customer)
- **Do not send Diversion header**: enable
- **Enable Session Timer**: optional

Alias Tab

- Enter <SBC-IP>:<CorePort>

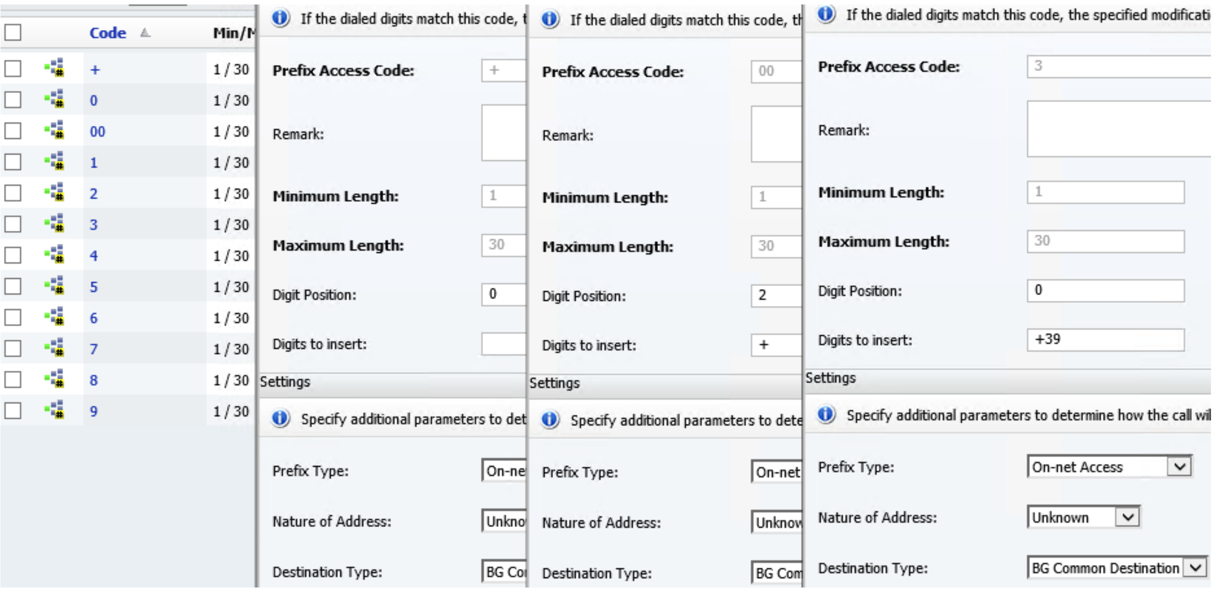
Digest Authentication

- Not used for this provider

Gateway Numbering Plan

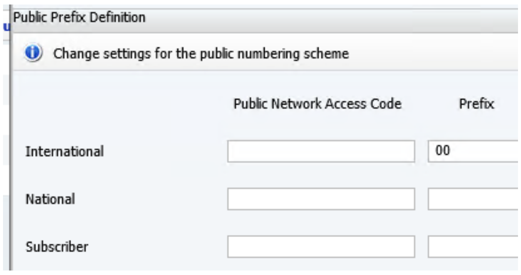
PrefixAccessCodes

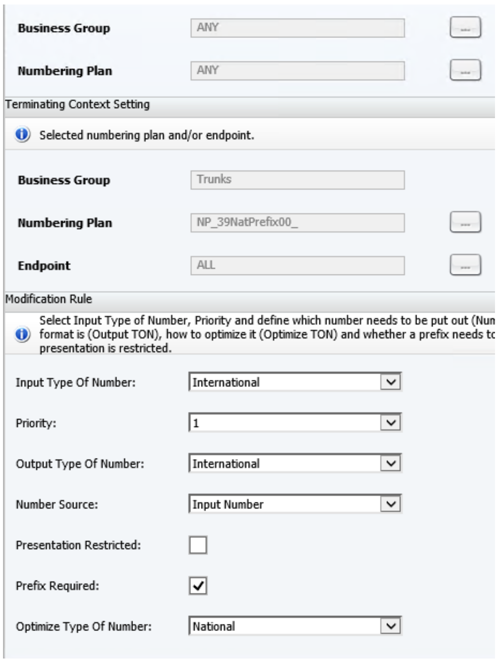
- **+**: delete 0 , don’t add
- **00**: delete 2, add +
- **0-9**: delete 0 , add +<CountryCode>
- In this example all numbers will be routed in GNF format to the CNP



DNM Prefix

- **International Prefix**: enter “00”
- **National Prefix**: do not enter a national prefix for Italy





DNM Modification

- **Input Type Of Number**: International
- **Priority**: 1
- **Output Type Of Number**: International
- **Number Source**: Input Number
- **Presentation Restricted**: none
- **Prefix Required**: enable
- **Optimize Type Of Number**: select “National”

DNM Normalization

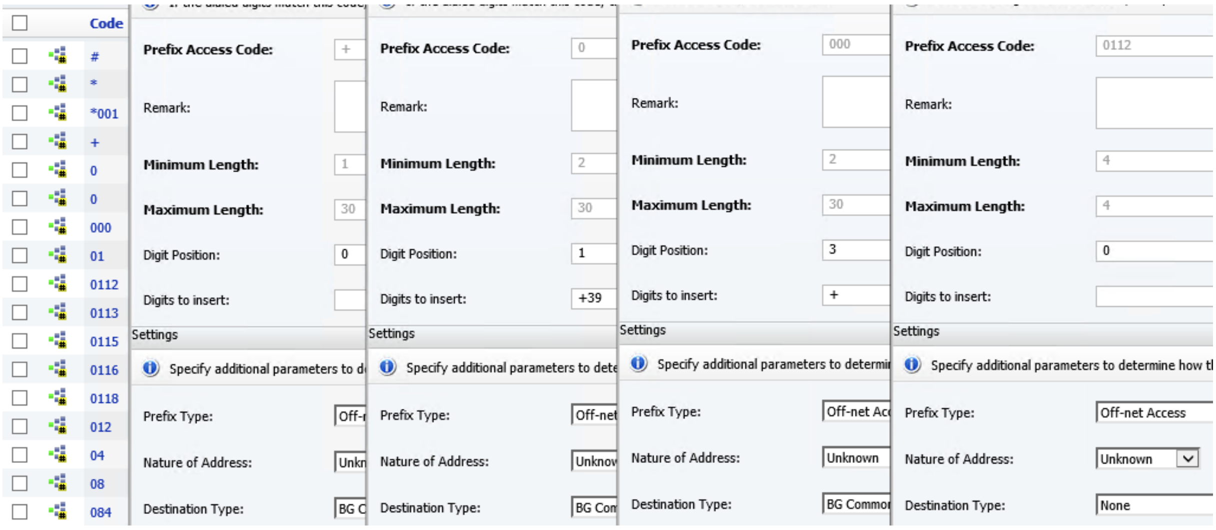
- **International rule**: Input Pattern= 00-Z, Normalized TON= International, NormalizedExpression= {2}
- **National rule**: Input Pattern= [0-9]Z, Normalized TON= International, NormalizedExpression= <CountryCode>{1}

Business Group	Numbering Plan	Endpoint	Input TON	Input Pattern	Normalized TON	Normalized Expression
Trunks	NP_39NatPrefix00_		Unknown	00-Z	International	{2}
Trunks	NP_39NatPrefix00_		Unknown	[0-9]Z	International	39{1}

Subscriber Numbering Plan

PrefixAccessCodes

- **+**: delete 0 , don’t add and route it to the Common Numbering Plan (CNP)
- **0**: delete 1, add +<CountryCode> and route it to the CNP
- **00**: don’t create this rule as Italy not has a national prefix
- **000**: delete 3, add + and route it to the CNP
- **Non E.164 numbers**: <PNAC><non E.164 numbers> e.g. emergency must be routed directly to the local DestinationCode

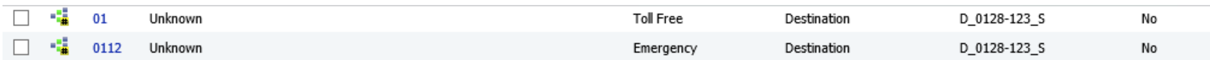


DestinationCodes

- **1-9**: NoA=International , TrafficType=BasedOnLocalToll, Destination= D_xxx_T e.g.:

- **Other numbers which need specific TrafficType**: enter International numbers and route to Destination D_xxx_T e.g.:

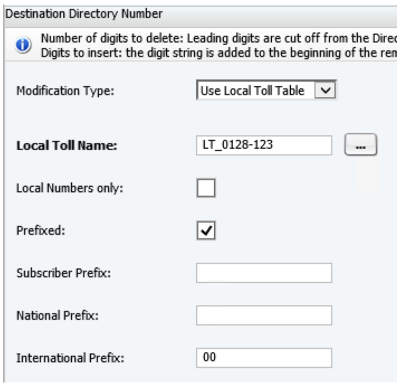
- **Non E.164 numbers**: <PNAC><non E.164 numbers> route it to Destination D_xxx_S e.g.:



Destination

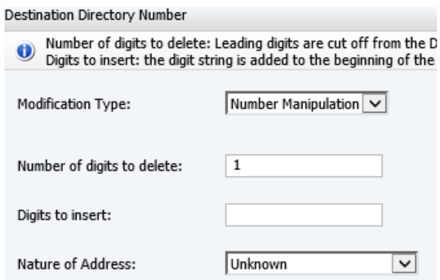
D_xxx_T

- **Modification Type**: “Use Local Toll Table”
- **Local Toll Name**: select Local Toll Table of this site
- **Prefixed**: enable
- **International Prefix**: enter “00”



D_xxx_S

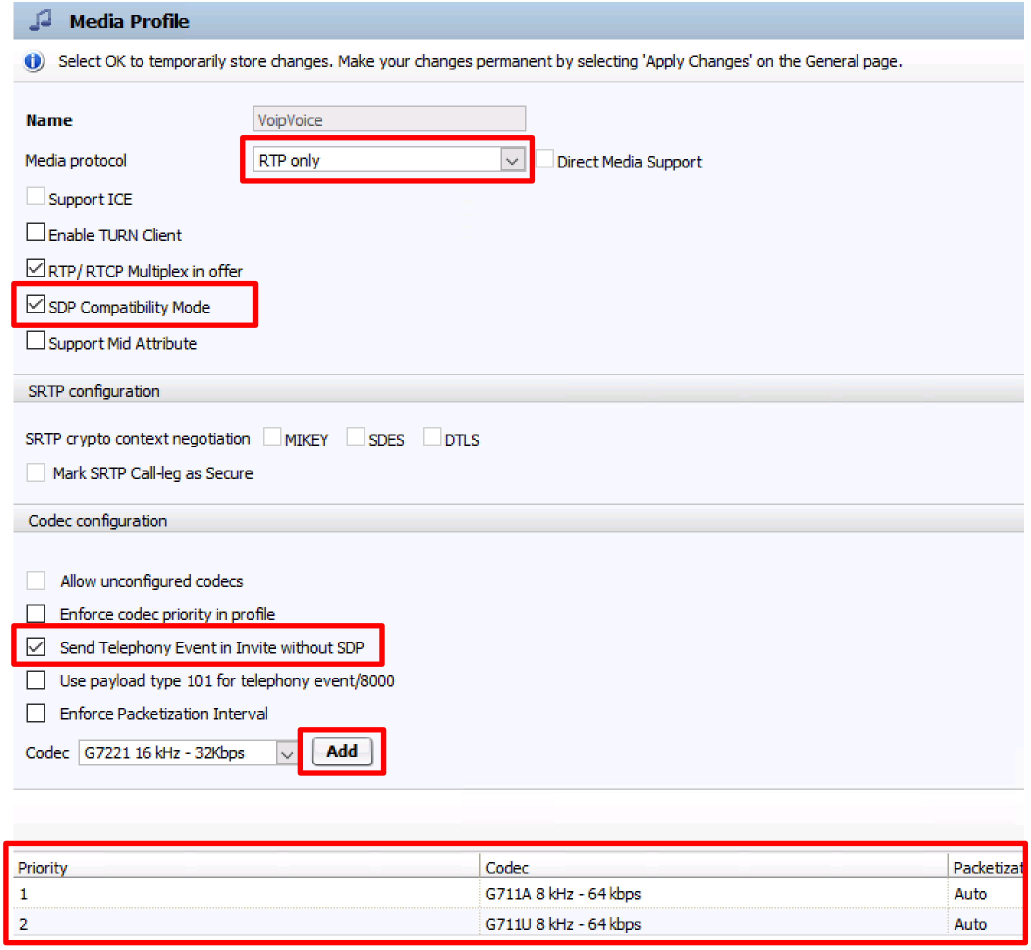
- **Modification Type**: “Number Manipulation”
- **Number of digits to delete**: enter “1” (delete PNAC)
- **Nature of Address**: select “Unknown”



OpenScape SBC Configuration

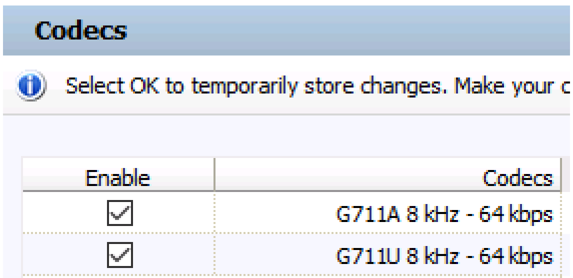
Media Profile

- **Media protocol**: RTP only
- **RTP/ RTCP Multiplex in offer**: enable
- **SDP Compatibility Mode**: enable (This allow us to eliminate unsupported codec in the SDP)
- **Send Telephony Event in Invite without SDP**: enable
- **Codec List**: G.711A, G711U, G729 (Optional)



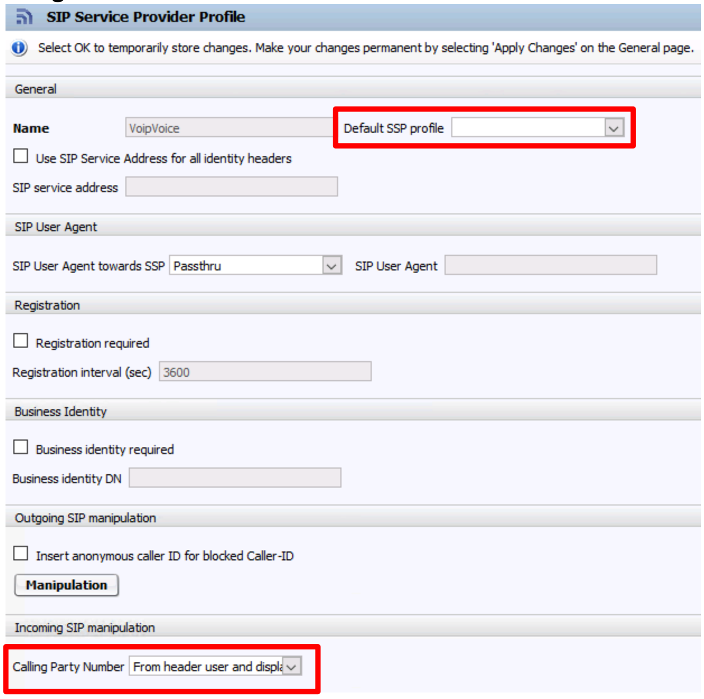
Enable Codec Support for transcoding

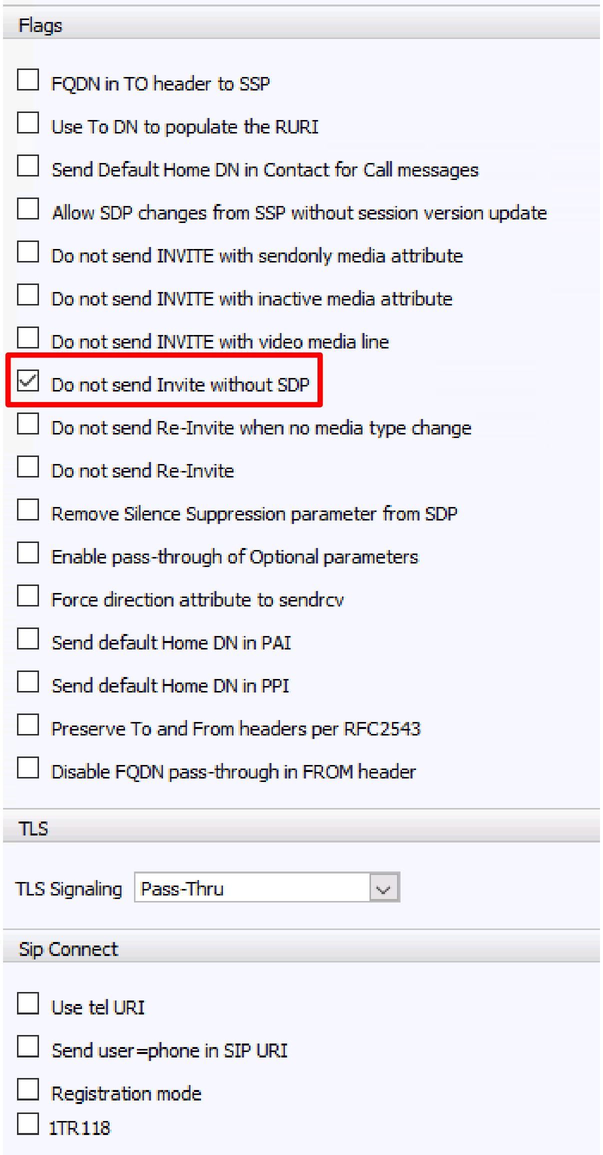
Enable all needed Codecs. At minimum G711



SIP Service Provider Profile

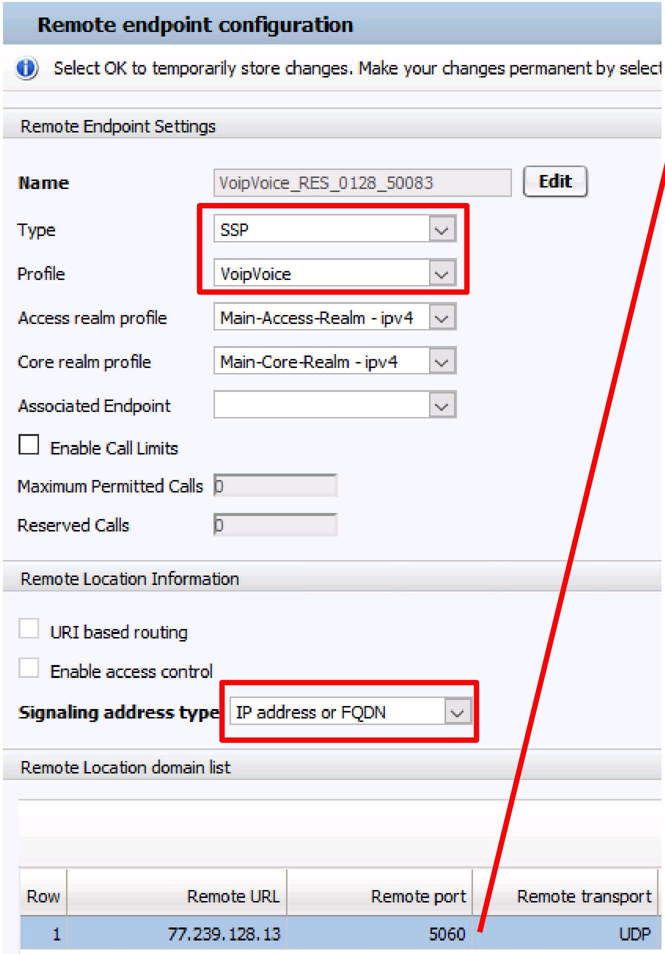
- **Default SSP profile:** empty
- **Incoming SIP manipulation - Calling Party Number:** From header user and display name part
- **Flag - Do not send Invite without SDP:** enable

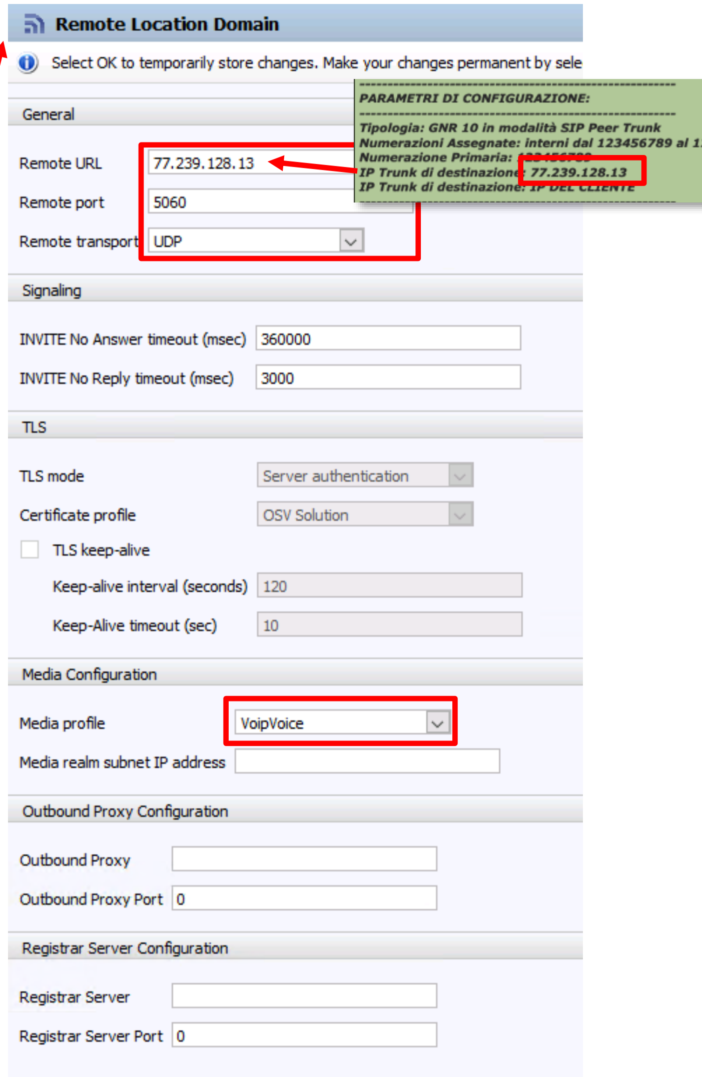


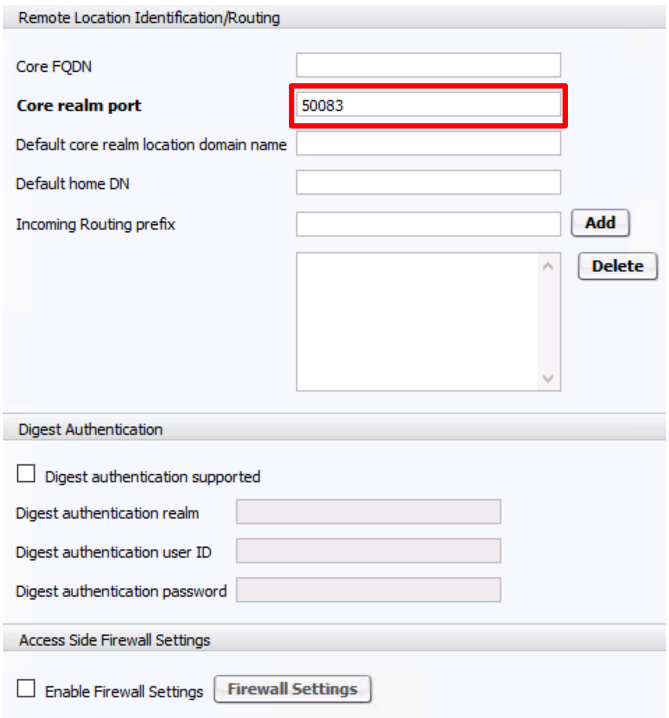


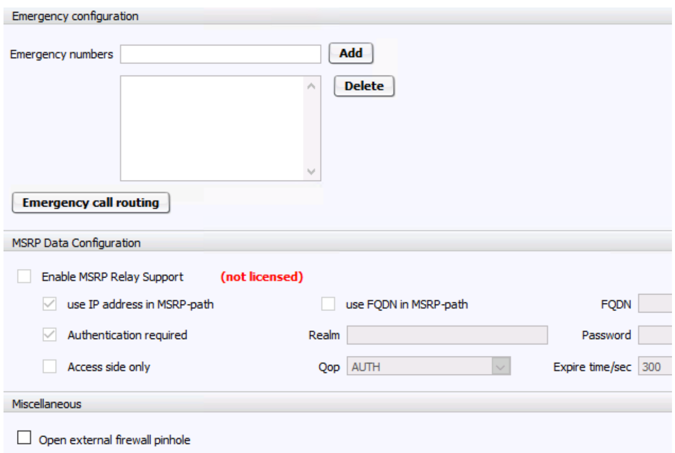
Remote Endpoint Configuration

Fill all red marked fields.









Known limitations, restrictions and things to know:

- FROM and PAI number should not be differ
- CLIP no screening is not supported by Italian law

Appendix

Supported Numbering Formats

Incoming: Called Party (REQUEST, TO): national number
Incoming: Calling Party (FROM, PAI, Diversion): national: national number, international: 00+E.164

Outgoing: Called Party (REQUEST, TO): national: national number, international: 00+E.164
Outgoing: Calling Party (FROM, PAI, Diversion): national number

Outgoing Emergency Call - Called Party (REQUEST, TO): unknown number
Outgoing Emergency Call - Calling Party (FROM, PAI, Diversion): national number

Outgoing OPTIONs: supported (answer: 200 OK)

Incoming OPTIONs: Provider doesn’t monitor the trunk via OPTION per default.

Call examples:

International Incoming Call:

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 5653
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0553952124@35.225.252.67;user=phone SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 77.239.128.13:5060;branch=z9hG4bKac1309308266
      Max-Forwards: 69
    > From: <sip:00498970070@10.39.31.20;user=phone>;tag=1c809563038
    > To: <sip:0553952124@10.39.31.123;user=phone>
      Call-ID: 883737710173202015531 9@77.239.128.13
    > CSeq: 1 INVITE
    > Contact: <sip:77.239.128.13:5060>
      Supported: 100rel,resource-priority,sdp-anat
      Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
    > Remote-Party-ID: <sip:00498970070@10.39.31.20;user=phone>;party=calling;privacy=off;screen=yes
      Diversion: <sip:unknown@unknown.invalid>;reason=unconditional;counter=1
      User-Agent: CS2000_NGSS/8.0
    > P-Asserted-Identity: <sip:00498970070@10.39.31.20;user=phone>
      Content-Type: application/sdp
      Content-Length: 344
  v Message Body
    v Session Description Protocol
        Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): PVG 770448296 1623073684 IN IP4 77.239.128.13
        Session Name (s): -
        Phone Number (p): +1 6135555555
      > Connection Information (c): IN IP4 77.239.128.13
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 27700 RTP/AVP 18 8 0 111 101
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmtp:101 0-15
      > Media Attribute (a): ptime:20
      > Media Attribute (a): fmtp:18 annexb=no
      > Media Attribute (a): rtpmap:111 OPUS/48000/2
      > Media Attribute (a): fmtp:111 minptime=20; maxplaybackrate=8000; maxaveragebitrate=50000; useinbandfec=1
```

National Incoming Call:

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 5653
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0553952124@35.225.252.67;user=phone SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 77.239.128.13:5060;branch=z9hG4bKac1055692658
      Max-Forwards: 69
    > From: <sip:0654578634@10.39.31.20;user=phone>;tag=1c1495147818
    > To: <sip:0553952124@10.39.31.123;user=phone>
      Call-ID: 178378254417320201557570@77.239.128.13
      [Generated Call-ID: 178378254417320201557570@77.239.128.13]
    > CSeq: 1 INVITE
    > Contact: <sip:77.239.128.13:5060>
      Supported: 100rel,resource-priority,sdp-anat
      Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
    > Remote-Party-ID: <sip:0654578634@10.39.31.20;user=phone>;party=calling;privacy=off;screen=yes
      User-Agent: CS2000_NGSS/8.0
    > P-Asserted-Identity: <sip:0654578634@10.39.31.20;user=phone>
      Content-Type: application/sdp
      Content-Length: 344
  v Message Body
    v Session Description Protocol
        Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): PVG 220317538 1730341299 IN IP4 77.239.128.13
        Session Name (s): -
        Phone Number (p): +1 6135555555
      > Connection Information (c): IN IP4 77.239.128.13
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 34368 RTP/AVP 18 8 0 111 101
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmtp:101 0-15
      > Media Attribute (a): ptime:20
      > Media Attribute (a): fmtp:18 annexb=no
      > Media Attribute (a): rtpmap:111 OPUS/48000/2
      > Media Attribute (a): fmtp:111 minptime=20; maxplaybackrate=8000; maxaveragebitrate=50000; useinbandfec=1
        [Generated Call-ID: 178378254417320201557570@77.239.128.13]
```

International Outgoing Call:

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:004989700731994@77.239.128.13:5060;transport=udp SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5653;branch=z9hG4bK9a0d.1da167d1ccb620038cc7e2780b2eb12d.0;i=2381
      Max-Forwards: 69
    > Contact: <sip:0553952124@35.225.252.67:5653;transport=udp>
    > To: <sip:004989700731994@77.239.128.13:5060;transport=udp>
    > From: <sip:0553952124@35.225.252.67;transport=udp>;tag=snl_mOjsoJd5M3
      Call-ID: SEC11-1e653c0a-1f653d0a-1-CK0P8jPl61e1
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Tue, 17 Mar 2020 15:10:35 GMT
      Supported: resource-priority
      Privacy: none
    > P-Asserted-Identity: <sip:0553952124@35.225.252.67>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 319
    > X-Siemens-OSS: OpenScape SBC V9 R4.14.03-99
  v Message Body
    v Session Description Protocol
        Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): SBC-WebRTC 6318162130489211634 2 IN IP4 10.60.100.11
        Session Name (s): SBC-WebRTC
      > Connection Information (c): IN IP4 35.238.247.221
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 18758 RTP/AVP 0 8 106 105 13 126
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:106 CN/32000
      > Media Attribute (a): rtpmap:105 CN/16000
      > Media Attribute (a): rtpmap:13 CN/8000
      > Media Attribute (a): rtpmap:126 telephone-event/8000
        Media Attribute (a): sendrecv
        Media Attribute (a): rtcp-mux
```

National Outgoing Call:

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0654578634@77.239.128.13:5060;transport=udp SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5653;branch=z9hG4bKea8c.8b0fcd3c6067b4f3dad1a8263cbb67c0.0;i=2381
      Max-Forwards: 69
    > Contact: <sip:0553952124@35.225.252.67:5653;transport=udp>
    > To: <sip:0654578634@77.239.128.13:5060;transport=udp>
    > From: <sip:0553952124@35.225.252.67;transport=udp>;tag=snl_546Tpg6t09
      Call-ID: SEC11-1e653c0a-1f653d0a-1-Ux3031LSOY8l
      [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-Ux3031LSOY8l]
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Tue, 17 Mar 2020 15:12:24 GMT
      Supported: resource-priority
      Privacy: none
    > P-Asserted-Identity: <sip:0553952124@35.225.252.67>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 319
    > X-Siemens-OSS: OpenScape SBC V9 R4.14.03-99
  v Message Body
    v Session Description Protocol
        Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): SBC-WebRTC 1600800272922600825 2 IN IP4 10.60.100.11
        Session Name (s): SBC-WebRTC
      > Connection Information (c): IN IP4 35.238.247.221
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 19714 RTP/AVP 0 8 106 105 13 126
      > Media Attribute (a): rtpmap:0 PCMU/8000
      > Media Attribute (a): rtpmap:8 PCMA/8000
      > Media Attribute (a): rtpmap:106 CN/32000
      > Media Attribute (a): rtpmap:105 CN/16000
      > Media Attribute (a): rtpmap:13 CN/8000
      > Media Attribute (a): rtpmap:126 telephone-event/8000
        Media Attribute (a): sendrecv
        Media Attribute (a): rtcp-mux
        [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-Ux3031LSOY8l]
```

Emergency Call:

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:112@77.239.128.13:5060;transport=udp SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5653;branch=z9hG4bK4814.3212977212c79087c1f87166d3cdd306.0;i=9028
      Max-Forwards: 67
    > Contact: <sip:0553952125@35.225.252.67:5653;transport=udp>
    > To: <sip:112@77.239.128.13:5060;transport=udp>
    > From: <sip:0553952125@35.225.252.67;transport=udp>;tag=snl_7fMixQoLYt
      Call-ID: SEC11-1e653c0a-1f653d0a-1-AuQn70aWW5Wt
      [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-AuQn70aWW5Wt]
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Mon, 18 Nov 2019 08:59:02 GMT
      Supported: resource-priority
      Privacy: none
    > P-Asserted-Identity: <sip:0553952125@35.225.252.67>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 296
    > X-Siemens-OSS: OpenScape SBC V9 R4.12.03-1
  > Message Body
```

Outgoing OPTION:

OSV -> Provider

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
v Session Initiation Protocol (OPTIONS)
  > Request-Line: OPTIONS sip:77.239.128.13:5060;transport=udp SIP/2.0
  v Message Header
      Call-ID: SEC11-471a440a-481a440a-1-tv5MmOU506qM
      [Generated Call-ID: SEC11-471a440a-481a440a-1-tv5MmOU506qM]
    > CSeq: 1 OPTIONS
    > To: <sip:77.239.128.13:5060;transport=udp>
    > From: <sip:ERES0128P50083@5.10.69.132;transport=udp>;tag=snl_DR9L5IJ489
      Content-Length: 0
      Max-Forwards: 70
    > Via: SIP/2.0/UDP 5.10.69.132:5653;branch=z9hG4bK3842.47f8ee86c5d43ba3424700335d73b557.0;i=1987
```

Provider -> OSV

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 5653
v Session Initiation Protocol (200)
  > Status-Line: SIP/2.0 200 OK
  v Message Header
    > Via: SIP/2.0/UDP 5.10.69.132:5653;i=1987;branch=z9hG4bK3842.47f8ee86c5d43ba3424700335d73b557.0
    > From: <sip:ERES0128P50083@5.10.69.132;transport=udp>;tag=snl_DR9L5IJ489
    > To: <sip:77.239.128.13;transport=udp>;tag=1c1304515750
      Call-ID: SEC11-471a440a-481a440a-1-tv5MmOU506qM
      [Generated Call-ID: SEC11-471a440a-481a440a-1-tv5MmOU506qM]
    > CSeq: 1 OPTIONS
      Content-Length: 0
```