

# OpenScape Voice V9



## How to Configure SIP Trunk for TWT

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**Tested Scenario:** Provider <-> OpenScape SBC <-> OpenScape Voice

**Provider Topology:** Static Trunk

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## Table of History

<b>Date</b>	<b>Version</b>	<b>Changes</b>
02.04.2020	0.1	First version
09.04.2020	0.9	Ready for review
13.05.2020	1.0	Released

## Trunk Configuration Data provided by TWT

**Trunk connection:** Static Trunk. The trunk is linked to the SBC public Access IP.

**Transport Protocol:** UDP

**Signaling IP:** 77.239.128.6; **Signaling Port:** 5060

**Media IP:** 77.239.128.6; **Media Ports:** 25000-60000

**Provider settings:** The trunk has no Openscape specific settings.

**Documentation:** NGN TWT Documento di supporto tecnico

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

The email can look like this:

	TWT	OPERATOR
<b>City</b>	<b>Milano</b>	
<b>Platform</b>		
SBC or Gateway type	RIBBON C20	
Software Revision	R19	
<b>Signalling IP</b>		
IP address (Signalling) – 1 (IP1)	77.239.128.6	
IP address (Media) – 1	77.239.128.6	
Location IP	V.e Jenner, 33 - Milano - Italia	
Owner IP	TWT SPA	
<b>Transport</b>		
Protocol	SIP	SIP
Port	5060 (Provider Port)	5060 (Customer SBC Port)
OPTION	no	
<b>Codecs and Packetization</b>		
<b>Incoming (Operator to TWT)</b>		
G,729 a	First Priority	20 ms
G,711 A	Second Priority	20 ms
G,711 μ	Third Priority	20 ms
<b>Outgoing (TWT to Operator)</b>		
G,729 a	First Priority	20 ms
G,711 A	Second Priority	20 ms
G,711 μ	Third Priority	20 ms
<b>Other Services</b>		
DTMF	RFC 2833	
Fax Services	T.38 Fax Relay	
<b>Numbering Format</b>		
E.164	Preferred	
CALLING PARTY NUMBER	+cc+number	
CALLED PARTY NUMBER	+cc+number	
<b>Prefixes</b>		
Technical Prefixes		
<b>Capacity (Amount of sessions)</b>		
TWT to Operator		
<b>TRAFFIC PROFILE</b>		
<b>IP active</b>		
Operator to TWT		
<b>TRAFFIC PROFILE</b>		
<b>IP active</b>		

# 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 allow external to external transit calls.

The 'General' tab shows the following configuration:

- Name: EPP\_TWT\_IT
- Remark: (empty)
- Numbering Plan: NP\_InternetDialing
- Management Information: Please enter the data for the following fields in the corresponding screens.
- Class of Service: (empty)
- Routing Area: (empty)
- Calling Location: (empty)
- Time Zone: LOCAL
- SIP Privacy Support: Full
- Failed Calls Intercept Treatment: Disabled
- Language: Italian
- Impact Level: Unclassified

The 'Services' tab shows the following configuration:

- Message Waiting: Yes
- Call Transfer: Yes
- Call Forward Invalid Destination: No
- Toll and Call Restrictions: Yes
- Park to Server: No
- CSTA Network Interface Device: No

The 'Toll and Call Restrictions' configuration shows:

- Standard
- Blocked Call List
- Alternate
- Day S
- Toll and Call Restrictions
- Class of Restriction: COR\_Office

COR\_Office must have all external Traffic Types assigned

The 'Traffic Types' tab shows the following configuration:

- Enter the restricted Traffic Types.
- Add: (empty)
- Sel:0 | Items/Page: 200 | All:7
- Traffic Types
- Directory Assistance
- International
- Local
- National
- Premium Rate
- Public Operator
- Toll Free

## Endpoint

### General Tab

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

The screenshot shows the 'General' tab of the 'Endpoint' configuration page. The tabs at the top are 'General', 'SIP', 'Attributes', 'Aliases', 'Routes', and 'Account'. The main heading is 'Endpoint' with a sub-heading 'Define the connection data of an endpoint, e.g. you may use this to add a gate...'. The form contains the following fields:

- Name: ERES0066P50050
- Remark: (empty dropdown)
- Registered:
- Profile: EPP\_TWT\_IT
- Branch Office: BO\_Trunks
- Associated Endpoint: D\_EP\_SharedTrks
- Default Home DN: (empty dropdown)
- Location Domain: (empty text box)
- Endpoint Template: (empty dropdown)

### 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"

The screenshot shows the 'SIP' tab of the 'Endpoint Type' configuration page. The tabs at the top are 'General', 'SIP', 'Attributes', 'Aliases', 'Routes', and 'Acc'. The main heading is 'Endpoint Type'. The form contains the following fields:

- SIP Private Networking:
- SIP Trunking:
- SIP-Q Signaling:
- SIP Signaling: Note that the address of the signaling interface cannot be modified unless has first been removed.
- Type: Static
- Signaling Address Type: IP Address or FQDN
- Endpoint Address: 10.60.101.116
- Port: 50050
- Transport protocol: MTLS
- Endpoint does not accept incoming TLS connections:
- SRTP media mode: Enabled

### 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
- **Send International Numbers in Global Number Format (GNF):** 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

<input type="checkbox"/>	Code ▲	Min/Max	<b>i</b> If the dialed digits match this code, the specified modification	<b>i</b> If the dialed digits match this code, the specified modification
<input type="checkbox"/>	+	1 / 30	<b>Prefix Access Code:</b> +	<b>Prefix Access Code:</b> 3
<input type="checkbox"/>	1	1 / 30	<b>Remark:</b>	<b>Remark:</b>
<input type="checkbox"/>	2	1 / 30	<b>Minimum Length:</b> 1	<b>Minimum Length:</b> 1
<input type="checkbox"/>	3	1 / 30	<b>Maximum Length:</b> 30	<b>Maximum Length:</b> 30
<input type="checkbox"/>	4	1 / 30	<b>Digit Position:</b> 0	<b>Digit Position:</b> 0
<input type="checkbox"/>	5	1 / 30	<b>Digits to insert:</b>	<b>Digits to insert:</b> +
<input type="checkbox"/>	6	1 / 30	<b>Settings</b>	<b>Settings</b>
<input type="checkbox"/>	7	1 / 30	<b>i</b> Specify additional parameters to determine how the call will	<b>i</b> Specify additional parameters to determine how the call will
<input type="checkbox"/>	8	1 / 30	<b>Prefix Type:</b> On-net Access	<b>Prefix Type:</b> On-net Access
<input type="checkbox"/>	9	1 / 30	<b>Nature of Address:</b> Unknown	<b>Nature of Address:</b> Unknown
			<b>Destination Type:</b> BG Common Dest	<b>Destination Type:</b> BG Common Destination

### DNM Prefix

- No Gateway specific rule needed

### DNM Modification

- **Input Type Of Number:** ANY
- **Priority:** 1
- **Output Type Of Number:** International
- **Number Source:** Input Number
- **Presentation Restricted:** none
- **Prefix Required:** none
- **Optimize Type Of Number:** select "None"

<b>Business Group</b>	ANY	...
<b>Numbering Plan</b>	ANY	...
<b>Terminating Context Setting</b>		
<b>i</b> Selected numbering plan and/or endpoint.		
<b>Business Group</b>	ANY	
<b>Numbering Plan</b>	ANY	...
<b>Endpoint</b>	ALL	...
<b>Modification Rule</b>		
<b>i</b> Select Input Type of Number, Priority and define which number needs to be put out (Num format is (Output TON), how to optimize it (Optimize TON) and whether a prefix needs to presentation is restricted.		
<b>Input Type Of Number:</b>	ANY	▼
<b>Priority:</b>	1	▼
<b>Output Type Of Number:</b>	International	▼
<b>Number Source:</b>	Input Number	▼
<b>Presentation Restricted:</b>	<input type="checkbox"/>	
<b>Prefix Required:</b>	<input type="checkbox"/>	
<b>Optimize Type Of Number:</b>	None	▼

### DNM Normalization

- **International rule:** Input Pattern= [1-9] , Normalized TON= International, NormalizedExpression= {1}

<input type="checkbox"/>	Business Group	Numbering Plan	Endpoint	Input TON	Input Pattern	Normalized TON	Normalized Expression
<input type="checkbox"/>	ANY	ANY		Unknown	[1-9]Z	International	{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

Code	Prefix Access Code	Minimum Length	Maximum Length	Digit Position	Digits to insert	Prefix Type	Nature of Address	Destination Type
#	+	1	30	0		Off-net	Unkn	BG C
*001								
0		2	30	1	+39	Off-net	Unkn	BG C
00								
000								
01				3	+	Off-net	Unkn	BG C
0112								
0113								
0115								
0116								
0118								
012						Off-net	Unkn	BG C
04							Unkn	BG C
08							Unkn	BG C
084							Unkn	None

## DestinationCodes

- **1-9**: NoA=International , TrafficType=BasedOnLocalToll, Destination= D\_xxx\_T  
 e.g.: 

1	International	BasedOnLocalToll	Destination	D_0066-123_T
---	---------------	------------------	-------------	--------------
- **Other numbers which need specific TrafficType**: enter International numbers and route to Destination D\_xxx\_T  
 e.g.: 

800	International	Toll Free	Destination	D_0066-123_T
-----	---------------	-----------	-------------	--------------
- **Non E.164 numbers**: <PNAC><non E.164 numbers> route it to Destination D\_xxx\_S  
 e.g.: 

01	Unknown	Toll Free	Destination	D_0066-123_S
0112	Unknown	Emergency	Destination	D_0066-123_S

## Destination

### D\_xxx\_T

- **Modification Type**: "None"

Destination Directory Number

Number of digits to delete: Leading digits are cut off from  
 Digits to insert: the digit string is added to the beginning of

Modification Type:

### D\_xxx\_S

- **Modification Type**: "Number Manipulation"
- **Number of digits to delete**: enter "1" (delete PNAC)
- **Digits to insert**: <CountryCode>
- **Nature of Address**: select "Unknown"

Destination Directory Number

Number of digits to delete: Leading digits are cut off from the  
 Digits to insert: the digit string is added to the beginning of the

Modification Type:

Number of digits to delete:

Digits to insert:

Nature of Address:



# 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 (This is the preferred codec for TWT)

Media Profile

i Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.

---

General

**Name**

Media protocol   Direct Media Support

Support ICE

Enable TURN Client

RTP/ RTCP Multiplex in offer

SDP Compatibility Mode

Support Mid Attribute

---

SRTP configuration

SRTP crypto context negotiation  MIKEY  SDES  DTLS

Mark SRTP Call-leg as Secure

---

Codec configuration

Allow unconfigured codecs

Enforce codec priority in profile

Send Telephony Event in Invite without SDP

Use payload type 101 for telephony event/8000

Enforce Packetization Interval

Codec

---

Priority	Codec	Packeti
1	G711A 8 kHz - 64 kbps	Auto
2	G711U 8 kHz - 64 kbps	Auto

## Enable Codec Support for transcoding

Enable all needed Codecs. At minimum G711

Codecs

i Select OK to temporarily store changes. Make your c

Enable	Codecs
<input checked="" type="checkbox"/>	G711A 8 kHz - 64 kbps
<input checked="" type="checkbox"/>	G711U 8 kHz - 64 kbps

## 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

**SIP Service Provider Profile**

Select OK to temporarily store changes. Make your changes permanent by selecting 'Apply Changes' on the General page.

**General**

Name: TWT Default SSP profile: [dropdown]

Use SIP Service Address for all identity headers

SIP service address: [text box]

**SIP User Agent**

SIP User Agent towards SSP: Passthru [dropdown] SIP User Agent: [text box]

**Registration**

Registration required

Registration interval (sec): 3600 [text box]

**Business Identity**

Business identity required

Business identity DN: [text box]

**Outgoing SIP manipulation**

Insert anonymous caller ID for blocked Caller-ID

**Manipulation**

**Incoming SIP manipulation**

Calling Party Number: From header user and display name [dropdown]

**Flags**

- FQDN in TO header to SSP
- Use To DN to populate the RURI
- Send Default Home DN in Contact for Call messages
- Allow SDP changes from SSP without session version update
- Do not send INVITE with sendonly media attribute
- Do not send INVITE with inactive media attribute
- Do not send INVITE with video media line
- Do not send Invite without SDP
- Do not send Re-Invite when no media type change
- Do not send Re-Invite
- Remove Silence Suppression parameter from SDP
- Enable pass-through of Optional parameters
- Force direction attribute to sendrcv
- Send default Home DN in PAI
- Send default Home DN in PPI
- Preserve To and From headers per RFC2543
- Disable FQDN pass-through in FROM header

**TLS**

TLS Signaling: Pass-Thru [dropdown]

**Sip Connect**

- Use tel URI
- Send user=phone in SIP URI
- Registration mode
- ITR118

# Remote Endpoint Configuration

Fill all red marked fields.

**Remote endpoint configuration**

Select OK to temporarily store changes. Make your changes permanent by selecting OK.

**Remote Endpoint Settings**

Name: TWT\_RES\_0066\_50050 Edit

Type: **SSP**

Profile: **TWT**

Access realm profile: Main-Access-Realm - ipv4

Core realm profile: Main-Core-Realm - ipv4

Associated Endpoint:

Enable Call Limits

Maximum Permitted Calls:

Reserved Calls:

**Remote Location Information**

URI based routing

Enable access control

Signaling address type: **IP address or FQDN**

**Remote Location domain list**

Row	Remote URL	Remote port	Remote transport	Media IP	Media profile
1	77.239.128.6	5060	UDP		TWT

**Remote Location Domain**

Select OK to temporarily store changes. Make your changes permanent by selecting OK.

**Signalling IP**

IP address (Signalling) - 1 (IP1): **77.239.128.6**

IP address (Media) - 1: 77.239.128.6

Location IP: **77.239.128.6**

Owner: V.8 janna, 33 - Milano - Italia

TWT SPA

**General**

Remote URL: **77.239.128.6**  Shared domain

Remote port: 5060

Remote transport: UDP

**Signaling**

INVITE No Answer timeout (msec): 360000

INVITE No Reply timeout (msec): 3000

**TLS**

TLS mode: Server authentication

Certificate profile: OSV Solution

TLS keep-alive

Keep-alive interval (seconds): 120

Keep-Alive timeout (sec): 10

**Media Configuration**

Media profile: **TWT**

Media realm subnet IP address:

**Outbound Proxy Configuration**

Outbound Proxy:

Outbound Proxy Port: 0

**Registrar Server Configuration**

Registrar Server:

Registrar Server Port: 0

**Remote Location Identification/Routing**

Core FQDN:

Core realm port: **50083**

Default core realm location domain name:

Default home DN:

Incoming Routing prefix:  Add

Delete

**Digest Authentication**

Digest authentication supported

Digest authentication realm:

Digest authentication user ID:

Digest authentication password:

**Access Side Firewall Settings**

Enable Firewall Settings Firewall Settings

**Emergency configuration**

Emergency numbers:  Add

Delete

**Emergency call routing**

**MSRP Data Configuration**

Enable MSRP Relay Support **(not licensed)**

use IP address in MSRP-path  use FQDN in MSRP-path FQDN:

Authentication required Realm:  Password:

Access side only Qop: AUTH Expires time/sec: 300

**Miscellaneous**

Open external firewall pinhole

## **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):** E.164 number

**Incoming: Calling Party (FROM, PAI, Diversion):** E.164 number

**Outgoing: Called Party (REQUEST, TO):** GNF

**Outgoing: Calling Party (FROM, PAI, Diversion):** GNF

**Outgoing Emergency Call - Called Party (REQUEST, TO):** 39<emergency number>

**Outgoing Emergency Call - Calling Party (FROM, PAI, Diversion):** GNF

**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: 5652
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:3909311790820@35.225.252.67:5652;user=phone SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 77.239.128.6:5060;branch=z9hG4bKd085qdu36fqmdtq432ot1u6cu2
    > From: <sip:498970070@10.39.31.20:5060;user=phone>;tag=SD0hjof01-c4-65014-a4b041-4f098425-a4b041
    > To: <sip:3909311790820@10.39.6.16:5060;user=phone>
    Call-ID: SD0hjof01-173775e94e0ce695724c0797f115c686-ao92gd1
    [Generated Call-ID: SD0hjof01-173775e94e0ce695724c0797f115c686-ao92gd1]
    > CSeq: 1 INVITE
    User-Agent: CS2000_NGSS/8.0
    Max-Forwards: 69
    Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
    > Contact: <sip:77.239.128.6:5060;transport=UDP>
    Supported: 100rel,resource-priority
    Content-Type: application/sdp
    Content-Length: 231
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): PVG 1584617210260 1584617210260 IN IP4 77.239.128.6
      Session Name (s): -
      Phone Number (p): +1 6135555555
      > Connection Information (c): IN IP4 77.239.128.6
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 57306 RTP/AVP 18 8 0 101
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15
      > Media Attribute (a):ptime:20
      > Media Attribute (a): fmp:18 annex=no
      [Generated Call-ID: SD0hjof01-173775e94e0ce695724c0797f115c686-ao92gd1]
```

### National Incoming Call:

```
> User Datagram Protocol, Src Port: 5060, Dst Port: 5652
v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:3909311790820@35.225.252.67:5652;user=phone SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 77.239.128.6:5060;branch=z9hG4bKn0m9r3c86fht92ecr6mc5auu94
    > From: <sip:390654578634@10.39.31.20:5060;user=phone>;tag=Sdb4s9f01-3c4-65014-c02858-d67af55-c02858
    > To: <sip:3909311790820@10.39.6.16:5060;user=phone>
    Call-ID: Sdb4s9f01-eec0607a0bd13693d9e0cc09552fe85b-ao92gd1
    [Generated Call-ID: Sdb4s9f01-eec0607a0bd13693d9e0cc09552fe85b-ao92gd1]
    > CSeq: 1 INVITE
    User-Agent: CS2000_NGSS/8.0
    Max-Forwards: 69
    Allow: ACK,BYE,CANCEL,INVITE,OPTIONS,INFO,SUBSCRIBE,REFER,NOTIFY,PRACK
    > Contact: <sip:77.239.128.6:5060;transport=UDP>
    Supported: 100rel,resource-priority
    Content-Type: application/sdp
    Content-Length: 231
  v Message Body
    v Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): PVG 1586432268030 1586432268030 IN IP4 77.239.128.6
      Session Name (s): -
      Phone Number (p): +1 6135555555
      > Connection Information (c): IN IP4 77.239.128.6
      > Time Description, active time (t): 0 0
      > Media Description, name and address (m): audio 46714 RTP/AVP 18 8 0 101
      > Media Attribute (a): rtpmap:101 telephone-event/8000
      > Media Attribute (a): fmp:101 0-15
      > Media Attribute (a):ptime:20
      > Media Attribute (a): fmp:18 annex=no
      [Generated Call-ID: Sdb4s9f01-eec0607a0bd13693d9e0cc09552fe85b-ao92gd1]
```

## International Outgoing Call:

```
> User Datagram Protocol, Src Port: 5652, Dst Port: 5060
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+4989700711355@77.239.128.6:5060;transport=udp SIP/2.0
  < Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5652;branch=z9hG4bKf593.c2d9d567c64cf41346ed104c4aae7509.0;i=3166
      Max-Forwards: 69
    > Contact: <sip:+3909311790820@35.225.252.67:5652;transport=udp>
    > To: <sip:+4989700711355@77.239.128.6:5060;transport=udp>
    > From: <sip:+3909311790820@35.225.252.67;transport=udp>;tag=sn1_ayF90UPI86
      Call-ID: SEC11-1e653c0a-1f653d0a-1-6D0sJq92JhW3
      [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-6D0sJq92JhW3]
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Thu, 09 Apr 2020 10:16:56 GMT
      Supported: resource-priority
      Privacy: none
    > P-Asserted-Identity: <sip:+3909311790820@35.225.252.67>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 319
    > X-Siemens-OSS: OpenScape SBC V9 R4.14.04-99
  < Message Body
    < Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): SBC-WebRTC 1021745738972081581 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 17704 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-1-6D0sJq92JhW3]
```

## National Outgoing Call:

```
> User Datagram Protocol, Src Port: 5652, Dst Port: 5060
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:+390654578634@77.239.128.6:5060;transport=udp SIP/2.0
  < Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5652;branch=z9hG4bK911e.db4c5461ed04218d39f40cd35023f295.0;i=3166
      Max-Forwards: 69
    > Contact: <sip:+3909311790820@35.225.252.67:5652;transport=udp>
    > To: <sip:+390654578634@77.239.128.6:5060;transport=udp>
    > From: <sip:+3909311790820@35.225.252.67;transport=udp>;tag=snl_13Yc7y13s8
      Call-ID: SEC11-1e653c0a-1f653d0a-1-8HE0h5fDx67m
      [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-8HE0h5fDx67m]
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Thu, 09 Apr 2020 10:29:54 GMT
      Supported: resource-priority
      Privacy: none
    > P-Asserted-Identity: <sip:+3909311790820@35.225.252.67>
    > X-Siemens-Call-Type: Org-NWid-external-public
      Content-Length: 318
    > X-Siemens-OSS: OpenScape SBC V9 R4.14.04-99
  < Message Body
    < Session Description Protocol
      Session Description Protocol Version (v): 0
      > Owner/Creator, Session Id (o): SBC-WebRTC 329283763948432992 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 11948 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-1-8HE0h5fDx67m]
```

## Emergency Call:

```

v Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:39118@77.239.128.6:5060;transport=udp SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 5.10.65.38:5101;branch=z9hG4bKac55.ecb9a543450936945995c38c1935eefd.0;i=d602
      Max-Forwards: 67
    > Contact: <sip:+3909311790820@5.10.65.38:5101;transport=udp>
    > To: <sip:39118@77.239.128.6:5060;transport=udp>
    > From: <sip:+3909311790820@5.10.65.38;transport=udp>;tag=snl_jA92T48T7q
      Call-ID: SEC11-471a440a-481a440a-1-e7D78p8W79qP
      [Generated Call-ID: SEC11-471a440a-481a440a-1-e7D78p8W79qP]
    > CSeq: 1235 INVITE
      Accept-Language: en;q=0.0
      Allow: REGISTER, INVITE, ACK, BYE, CANCEL, NOTIFY, REFER, INFO
      Content-Type: application/sdp
      Date: Thu, 21 Jun 2018 07:50:12 GMT
      Privacy: none
    > P-Asserted-Identity: <sip:+3909311790820@5.10.65.38>
    > X-Siemens-Call-Type: Org-Nwid-external-public
      Content-Length: 362
    > X-Siemens-OSS: OpenScape SBC V9 R3.26.00-2
  > Message Body

```

## Outgoing OPTION:

### OSV -> Provider

```

v Session Initiation Protocol (OPTIONS)
  > Request-Line: OPTIONS sip:77.239.128.6:5060;transport=udp SIP/2.0
  v Message Header
    Call-ID: SEC11-1e653c0a-1f653d0a-1-Iu6ekxMg61Hp
    [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-Iu6ekxMg61Hp]
    > CSeq: 1 OPTIONS
    > To: <sip:77.239.128.6:5060;transport=udp>
    > From: <sip:ERES0066P50050@35.225.252.67;transport=udp>;tag=snl_K6682w10jw
      Content-Length: 0
      Max-Forwards: 70
    > Via: SIP/2.0/UDP 35.225.252.67:5652;branch=z9hG4bK080f.b90001217e3608d1adde6e2422d4f24d.0;i=24879

```

### Provider -> OSV

```

v Session Initiation Protocol (OPTIONS)
  > Request-Line: OPTIONS sip:35.225.252.67:5652 SIP/2.0
  v Message Header
    > Via: SIP/2.0/UDP 77.239.128.6:5060;branch=z9hG4bK0gira310980h9j8ut6e0
    > To: sip:ping@35.225.252.67
    > From: <sip:ping@77.239.128.6>;tag=rg58100fik300-77m1000
      Call-ID: rg58100fik30gap99ih15718nqn2khdch9c0jpk6ehc4jqo5i9d43no6n1-77m1000@77.239.128.6
      [Generated Call-ID: rg58100fik30gap99ih15718nqn2khdch9c0jpk6ehc4jqo5i9d43no6n1-77m1000@77.239.128.6]
    > CSeq: 55527 OPTIONS
      Max-Forwards: 70
      Content-Length: 0

```