



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 Supporto 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

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 allow 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

The screenshot shows the 'General' tab of the 'Endpoint' configuration page. The 'Name' field contains 'ERES0128P50083'. The 'Registered' checkbox is checked. The 'Profile' dropdown is set to 'EPP_VOIPVOICE_IT'. The 'Branch Office' dropdown is set to 'BO_Trunks'. The 'Associated Endpoint' dropdown is set to 'D_EP_SharedTrks'. The 'Default Home DN' field contains '390553952120'. There are also fields for 'Remark', 'Location Domain', and 'Endpoint Template'.

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 'SIP Trunking' radio button is selected. The 'SIP Signaling' section has a note: 'For the static Endpoints the address of the SIP signaling interface can be s... Note that the address of the signaling interface cannot be modified unless has first been removed.' The 'Type' dropdown is set to 'Static'. The 'Signaling Address Type' dropdown is set to 'IP Address or FQDN'. The 'Endpoint Address' field contains '10.60.101.116'. The 'Port' field contains '50083'. The 'Transport protocol' dropdown is set to 'MTLS'. The 'Endpoint does not accept incoming TLS connections' checkbox is unchecked. The 'SRTP media mode' dropdown is set to '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
- **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

Code	Min/N	Prefix Access Code	Remark	Minimum Length	Maximum Length	Digit Position	Digits to insert	Prefix Type	Nature of Address	Destination Type
+	1 / 30	+		1	30	0		On-net	Unknown	BG Common
0	1 / 30	00		1	30	2	+	On-net	Unknown	BG Common
00	1 / 30	3		1	30	0	+39	On-net Access	Unknown	BG Common Destination
1	1 / 30									
2	1 / 30									
3	1 / 30									
4	1 / 30									
5	1 / 30									
6	1 / 30									
7	1 / 30									
8	1 / 30									
9	1 / 30									

DNM Prefix

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

	Public Network Access Code	Prefix
International		00
National		
Subscriber		

Business Group	ANY
Numbering Plan	ANY
Terminating Context Setting	
Selected numbering plan and/or endpoint.	
Business Group	Trunks
Numbering Plan	NP_39NatPrefix00_
Endpoint	ALL
Modification Rule	
Select Input Type of Number, Priority and define which number needs to be put out (Number format is (Output TON), how to optimize it (Optimize TON) and whether a prefix needs to presentation is restricted.	
Input Type Of Number:	International
Priority:	1
Output Type Of Number:	International
Number Source:	Input Number
Presentation Restricted:	<input type="checkbox"/>
Prefix Required:	<input checked="" type="checkbox"/>
Optimize Type Of Number:	National

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
<input type="checkbox"/>	Trunks	NP_39NatPrefix00_		Unknown	00-Z	International	{2}
<input type="checkbox"/>	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

Code	Prefix Access Code:	Prefix Access Code:	Prefix Access Code:	Prefix Access Code:
#	+	0	000	0112
*				
*001				
+				
0	Minimum Length: 1	Minimum Length: 2	Minimum Length: 2	Minimum Length: 4
0	Maximum Length: 30	Maximum Length: 30	Maximum Length: 30	Maximum Length: 4
000				
01	Digit Position: 0	Digit Position: 1	Digit Position: 3	Digit Position: 0
0112	Digits to insert:	Digits to insert: +39	Digits to insert: +	Digits to insert:
0113				
0115	Settings	Settings	Settings	Settings
0116	Specify additional parameters to d	Specify additional parameters to dete	Specify additional parameters to determ	Specify additional parameters to determine how t
0118				
012	Prefix Type: Off-net	Prefix Type: Off-net	Prefix Type: Off-net Acc	Prefix Type: Off-net Access
04	Nature of Address: Unkn	Nature of Address: Unkn	Nature of Address: Unknown	Nature of Address: Unknown
08				
084	Destination Type: BG C	Destination Type: BG Com	Destination Type: BG Common	Destination Type: None

DestinationCodes

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

1	International	BasedOnLocalToll	Destination	D_0128-123_T	No
---	---------------	------------------	-------------	--------------	----

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

800	International	Toll Free	Destination	D_0128-123_T	No
-----	---------------	-----------	-------------	--------------	----

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

01	Unknown	Toll Free	Destination	D_0128-123_S	No
0112	Unknown	Emergency	Destination	D_0128-123_S	No

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"

Destination Directory Number

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

Modification Type:

Local Toll Name:

Local Numbers only:

Prefixed:

Subscriber Prefix:

National Prefix:

International Prefix:

D_xxx_S

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

Destination Directory Number

Number of digits to delete: Leading digits are cut off from the D
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 (Optional)

Media Profile

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

Name VoipVoice

Media protocol **RTP only** 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 **G7221 16 kHz - 32Kbps** **Add**

Priority	Codec	Packetization Interval
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

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: VoipVoice **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 displ [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

1TR 118

Remote Endpoint Configuration

Fill all red marked fields.

Remote endpoint configuration

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

Remote Endpoint Settings

Name: VoipVoice_RES_0128_50083 Edit

Type: **SSP**

Profile: **VoipVoice**

Access realm profile: Main-Access-Realm - ipv4

Core realm profile: Main-Core-Realm - ipv4

Associated Endpoint:

Enable Call Limits

Maximum Permitted Calls: 0

Reserved Calls: 0

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
1	77.239.128.13	5060	UDP

Remote Location Domain

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

General

Remote URL: **77.239.128.13**

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: **VoipVoice**

Media realm subnet IP address:

Outbound Proxy Configuration

Outbound Proxy:

Outbound Proxy Port: 0

Registrar Server Configuration

Registrar Server:

Registrar Server Port: 0

PARAMETRI DI CONFIGURAZIONE:
 Tipologia: GNR 10 in modalità SIP Peer Trunk
 Numerazioni Assegnate: interni dal 123456789 al 1
 Numerazione Primaria: 77.239.128.13
 IP Trunk di destinazione: 77.239.128.13
 IP Trunk di destinazione: IP DEL CLIENTE

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 Expire 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): 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
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0553952124@35.225.252.67;user=phone SIP/2.0
  < 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: 8837377101732020155319@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
  < Message Body
    < 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
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0553952124@35.225.252.67;user=phone SIP/2.0
  < 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: 17837825441732020155757@77.239.128.13
      [Generated Call-ID: 17837825441732020155757@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
  < Message Body
    < 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: 17837825441732020155757@77.239.128.13]
```

International Outgoing Call:

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:004989700731994@77.239.128.13:5060;transport=udp SIP/2.0
  < 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_m0jsoJd5M3
    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
  < Message Body
    < 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
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:0654578634@77.239.128.13:5060;transport=udp SIP/2.0
  < Message Header
    > Via: SIP/2.0/UDP 35.225.252.67:5653;branch=z9hG4bK9a0d.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-Ux3031LSOY81
    [Generated Call-ID: SEC11-1e653c0a-1f653d0a-1-Ux3031LSOY81]
    > 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
  < Message Body
    < 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-Ux3031LSOY81]
```

Emergency Call:

```
> User Datagram Protocol, Src Port: 5653, Dst Port: 5060
< Session Initiation Protocol (INVITE)
  > Request-Line: INVITE sip:112@77.239.128.13:5060;transport=udp SIP/2.0
  < 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_7fMlxQoLYt
    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
< Session Initiation Protocol (OPTIONS)
  > Request-Line: OPTIONS sip:77.239.128.13:5060;transport=udp SIP/2.0
  < 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
< Session Initiation Protocol (200)
  > Status-Line: SIP/2.0 200 OK
  < 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
```