Technical specifications for connecting SIP PBX to the „Business Trunk" service by Slovak Telekom with registration of pilot account.

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1. Use of the service

„Business Trunk“ service with registration of pilot account is designed to provide connection of the customer's PBX to the Public Telecommunication Network (PTN) by Slovak Telekom, a.s.(ST) VoIP platforms. This method of connection is intended for connect of PBX with one SIP interfaces only, without redundancy. Signaling communication is running on SIP protocol, used transport protocol must by UDP. Telephone numbers assigned to the trunk do not need to be from the same range of numbers (DDI), but it is necessary that all numbers have to be created and assigned to the same trunk group.

2. General settings

DNS = 195.146.137.211; 195.146.137.212
NTP = 195.146.137.211; 195.146.137.212
DNS and NTP settings is valid for the connections via private MPLS access by ST. For other connections is necessary to use any reached NTP and DNS servers.
SIP domain = sip.vvn.telekom.sk (for using in SIP headers)
SIP proxy = sip.vvn.telekom.sk (for using in SIP headers)
Outbount proxy = 195.146.137.250 - for the connections via private MPLS access by ST
195.146.137.186 - for the connections via public internet (is need to allowed on IP firewall)
Realm = BroadWorks
Transport protocol = UDP
Supported audio codecs: G711 Alaw, G711 Ulaw, G729a, G722
Transport DTMF: RFC 2833, payload type 101 telephone-event.

The codec is negotiated between the calling and the called endpoint without the assistance of ST systems.

3. Autorization (authentication)

The authorization is required for all outgoing PBX calls. Authorization uses a common authentication name (pilot number without zero at the beginning) and a common SIP password, which are listed in the hand over protocol. Authentication credentials must by set in the PBX and used after receive SIP response „401 Unathorized”, or „407 Proxy Authentication Required“.
User name = pilot number (account)
Authentication name = pilot number (account)
Password = common SIP password
Example of REGISTER and INVITE messages authorization is showed in example of part 4 and 5b.

For security reasons is not possible turn off authentication of calls.

4. Registration

The trunk registration, regardless of the number of accounts included in this trunk, is performed by registering only a pilot account. This means that the REGISTER message is sent to this pilot acoount only. For other accounts, the registration is not sent. The values of the fields of SIP message header are showed in the example including combined with the authorization.

Example of REGISTER and authorization of the pilot trunk account 249119630:

User name (pilot number) = 249119630
Authentication name (pilot number) = 249119630
Domain names used in the SIP URI of the “FROM” and “TO” headers must by always **sip.vvn.telekom.sk**

PBX --> ST
REGISTER sip:sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKd11ebb6e76cdd5855
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=1db7a33261
To: <sip:249119630@sip.vvn.telekom.sk:5060>
Call-ID: e3f64831a50c5828
CSeq: 5627 REGISTER
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
Expires: 3600
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Length: 0

ST --> PBX
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKd11ebb6e76cdd5855
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=1db7a33261
To: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=1489687322-1438149919946
Call-ID: e3f64831a50c5828
CSeq: 5627 REGISTER
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXicodbw7eTcjvuh1BW",realm="BroadWorks",algorithm=MD5
Content-Length: 0

PBX --> ST
REGISTER sip:sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK43cbbd0a365fc8a4e
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=1db7a33261
To: <sip:249119630@sip.vvn.telekom.sk:5060>
Call-ID: e3f64831a50c5828
CSeq: 5628 REGISTER
Authorization: Digest ***username="249119630"****,realm="BroadWorks",nonce="BroadWorksXicodbw7eTcjvuh1BW",uri="sip:sip.vvn.telekom.sk:5060",response="951e9cb6468252a7c9d1fff6c390d93d",algorithm=MD5,qop=auth,cnonce="7262be96",nc=00000001*
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
Expires: 3600
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Length: 0

ST --> PBX
SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK43cbbd0a365fc8a4e
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=1db7a33261
To: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=2118676607-1438149919956
Call-ID: e3f64831a50c5828
CSeq: 5628 REGISTER
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>;expires=300;q=0.5
Allow-Events: call-info,line-seize,dialog,message-summary,as-feature-event,x-broadworks-hoteling,x-broadworks-call-center-status
Content-Length: 0

5. Format of numbers

a) Incoming calls to the PBX from the ST

- Called number = number in national format without zero on the first position. In the SIP URI is used domain „sip.vvn.telekom.sk" (e.g.: 249119635@sip.vvn.telekom.sk).
- Calling number = incoming national calls (originated in Slovakia) are in national format with zero on first position. In SIP URI is used domain „sip.vvn.telekom.sk" (e.g.: 0335920916@sip.vvn.telekom.sk). Incoming international calls are presented in the international format with double zeros on the begining. In the SIP URI is used domain „sip.vvn.telekom.sk" (e.g.: 00420705333555@sip.vvn.telekom.sk).

Example of incoming call to the PBX :

Calling party number – 0910500374
Called party number – 249119635 (IP address 192.168.203.23, port 5060)
Pilot number -249119630

„Request URI"contains SIP URI in format **"pilot_number@IP_address_PBX"** (with values in the "CONTACT" header from REGISTER)
Header "FROM" contains SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
Header "TO" contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**

```
ST --> PBX
INVITE sip:249119630@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKll20i4200oc1br02a1a0.1
From: <sip:0910500374@sip.vvn.telekom.sk;user=phone>;tag=467902633-1438165478985-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW122438985290715107761699 3@10.20.60.10
CSeq: 749459237 INVITE
Contact: <sip:0910500374@195.146.137.250:5060;transport=udp>
Supported: 100rel
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: application/dtmf-relay,application/media_control+xml,application/sdp,multipart/mixed
Max-Forwards: 69
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 293

v=0
o=BroadWorks 192446892 1 IN IP4 195.146.137.250
s=-
c=IN IP4 195.146.137.250
t=0 0
m=audio 24182 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
a=maxptime:20
```

b) Outgoing calls from the PBX to the ST

- Called number = local, national, international, or E164 format, short numbers, the same as calls from standard fixed line. V SIP URI must by used public domain „sip.vvn.telekom.sk"
(e.g.: local: 58823254@sip. vvn.telekom.sk, national: 0258823254@sip. vvn.telekom.sk, international: 00436769550786@sip. vvn.telekom.sk, E164: +436769550786@sip. vvn.telekom.sk, short number: 1181@sip. vvn.telekom.sk ...).
- Calling number = number in national format without zero on the first position. In the SIP URI must by used domain "sip.vvn.telekom.sk" (e.g.: 249119910@sip.vvn.telekom.sk).

Example of outgoing call with authorization of this call:

Calling party number – 249119635 (IP address 192.168.203.23, port 5060)
Called party number - 0910500374
Pilot number -249119630

„Request URI" must contains SIP URI in format "**called_number@sip.vvn.telekom.sk**"
Hlavička "FROM" must contains SIP URI in format "**pilot_number@sip.vvn.telekom.sk**"
Header "TO" musí must contains SIP URI in format "**called_number@sip.vvn.telekom.sk**"
Header "CONTACT" must contains SIP URI in format "**pilot_number@IP_address_PBX**"
Header "P-PREFERRED-IDENTITY" must contains SIP URI in format "**calling_number@sip.vvn.telekom.sk**"

```
PBX --> ST
INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK0268fb33836068ede
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=e4886d9ff6
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: f1909d03c95094bf
CSeq: 13145 INVITE
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
P-Preferred-Identity: <sip:249119635@sip.vvn.telekom.sk:5060>
.
.
```

.
.
Supported: replaces
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Type: application/sdp
Content-Length: 271

v=0
o=MxSIP 0 13 IN IP4 192.168.203.23
s=SIP Call
c=IN IP4 192.168.203.23
t=0 0
m=audio 10014 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=sendrecv

ST --> PBX
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKd11ebb6e76cdd5855
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=e4886d9ff6
To: <sip:0910500374@sip.vvn.telekom.sk:5060>;tag=1489687322-1438149919946
Call-ID: f1909d03c95094bf
CSeq: 13145 INVITE
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXicodbw7eTcjvuh1BW",realm="BroadWorks",algorithm=MD5
Content-Length: 0

PBX --> ST
INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK0268fb33836068ede
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=e4886d9ff6
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: f1909d03c95094bf
CSeq: 13145 INVITE
Authorization: Digest **username="249119630"**,realm="BroadWorks",nonce="BroadWorksXicodbw7eTcjvuh1BW",uri=
"sip:sip.vvn.telekom.sk:5060",response="951e9cb6468252a7c9d1fff6c390d93d",algorithm=MD5,qop=auth,cnonce="7262be96",
nc=00000001
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
P-Preferred-Identity: <sip:249119635@sip.vvn.telekom.sk:5060>
Supported: replaces
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Type: application/sdp
Content-Length: 271

v=0
o=MxSIP 0 13 IN IP4 192.168.203.23
s=SIP Call
c=IN IP4 192.168.203.23
t=0 0
m=audio 10014 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=sendrecv

6. Call forwarding from the PBX back to the PTN

In the case when some PBX user is forwarded back to PTN is possible to use one of the methods for retention of origin Calling ID:

a) On the incoming **„INVITE"** request send response **„302 Moved Temporarily"**, where header **„Contact"** contains a new destination number. In the SIP URI must be used public domain "sip.vvn.telekom.sk".
SIP dialog between the ST an the PBX will finished after receive SIP response „302 Moved Temporarily" and call on the forwarded number will make on the ST platforms. Status info (succesfull, unsuccesfull, duration ...) about this new call will not sent to the PBX.

Example of incoming call to the PBX, redirected back to other number by SIP response „302":

Calling party number – 0258823254 – **A_number**
Called party number – 249119635 (IP address 192.168.203.23, port 5060) - **B_number**
Redirected to 0910500374 - **C_number**
Pilot number -249119630

Incoming "INVITE"
„Request URI" contains SIP URI in format **"pilot number@IP_address_PBX"**
Header "FROM" contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**
Header "TO" contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**

Response "302 Moved Temporarily"
Header "FROM" must contains SIP URI in format "**A_number@sip.vvn.telekom.sk**"
Header "TO" must contains SIP URI in format "**B_number@sip.vvn.telekom.sk**"
Header "CONTACT" must contains SIP URI in format "**C_number@sip.vvn.telekom.sk**"

ST -->PBX
INVITE sip:249119630@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKqh3v7a2060rgoq4r54g1.1
From: <sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=1245563589-1438195960666-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW205240666290715573373036@10.20.60.10
CSeq: 764700078 INVITE
Contact: <sip:0258823254@195.146.137.250:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: application/dtmf-relay,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 293
.
.
SDP omitted

PBX --> ST

SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKqh3v7a2060rgoq4r54g1.1
From: <sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=1245563589-1438195960666-
To: "Display name" <sip:249119635@sip.vvn.telekom.sk:5060;user=phone>;tag=1703847302
Call-ID: BW205240666290715573373036@10.20.60.10
CSeq: 764700078 INVITE
Contact: "Display name" <sip:0910500374@sip.vvn.telekom.sk:5060;transport=udp>
Server: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Length: 0

b) Sending a new message **„INVITE"**, where the destination number of redirecting is placed in **„Request line"** and in the **„To"** header. The **„From"** header must contain the SIP URI of pilot number. This INVITE message must also contain the **„Diversion"** header, which must contains the SIP URI of the original dialed number and

other redirection parameters and the **„P-Preffered-Identity"** header, which must contain the SIP URI of the original calling number.
In this redirection mode, both streams (SIP and RTP) for both call directions go via PBX and therefore it has information about the status of these calls.
For correct identification of the caller ID when forwarding to a foreign number, we recommend that the „A" number in the **„P-Preferred-Identity"** header has been modified to the international format E164.

Example of incoming call to the PBX, redirected back to other number by new INVITE:

Calling party number – 0258823254 – **A_number**
Called party number – 249119635 (IP address 192.168.203.23, port 5060) - **B_number**
Redirected to 0910500374 - **C_number**
Pilot number -249119630

Incoming "INVITE"
„Request URI" contains SIP URI in format **"pilot_number@IP_address_PBX"**
Header "FROM" contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**
Header "TO" contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**

Outgoing "INVITE"
„Request URI" must contains SIP URI in format **"C_number@sip.vvn.telekom.sk"**
Header "FROM" must contains SIP URI in format **"pilot_number@sip.vvn.telekom.sk"**
Header "TO" must contains SIP URI in format **"C_number@sip.vvn.telekom.sk"**
Header "CONTACT" must contains SIP URI in format **"pilot_number@IP_address_PBX"**
Header "DIVERSION" must contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**
Header "P-PREFERRED-IDENTITY" must contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**

ST -->PBX
INVITE sip:249119630@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKqh3v7a2060rgoq4r54g1.1
From: <sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=1245563589-1438195960666-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW205240666290715573373036@10.20.60.10
CSeq: 764700078 INVITE
Contact: <sip:0258823254@195.146.137.250:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: application/dtmf-relay,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 293
.
.
SDP omitted

PBX --> ST
INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKa124df82b8ddab737
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=89c1b12534
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: e442ad177ce32054
CSeq: 1422 INVITE
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
Diversion: <sip:249119635@sip.vvn.telekom.sk:5060>;reason=unconditional;screen=no;privacy=off;counter=1
P-Preferred-Identity: <sip:+421258823254@sip.vvn.telekom.sk:5060>
Supported: replaces
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Type: application/sdp
Content-Length: 271
.
.
SDP omitted

7. Calling line identity restriction – CLIR

a) Outgoing calls from the PBX

For outgoing CLIR calls from the PBX is necessary to insert to the INVITE message **„P-Preferred-Identity"** header and **„Privacy"** header in following format:
P-Preferred-Identity: <sip:SIP URI calling number >
Privacy: id
The **„From"** and **„Contact"** headers must remain in exactly the same format as when calling without CLIR.

If the caller ID will not sent or if in the "From" header will be „anonymous@anonymous.invalid" in the sense of §60 of the Electronic Communications Act no. 351/2011 Z.z. (SK), such calls will by rejected.

Example of outgoing call from the PBX with CLIR:

Calling party number – 0249119635 (IP address 192.168.203.23, port 5060)
Called party number – 0910500374
Pilot number -249119630

„Request URI"must contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**
Header "FROM" must contains SIP URI in format **"pilot_number@sip.vvn.telekom.sk"**
Header "TO" must contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**
Header "CONTACT" must contains SIP URI in format **"pilot_number@IP_adresa_PBX"**
Header "P-PREFERRED-IDENTITY" must contains SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
Header "PRIVACY" must contains value **"id"**

```
PBX --> ST
INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKb042e3d30efbfe87b
Max-Forwards: 70
From: <sip:249119630@sip.vvn.telekom.sk:5060>;tag=593b5b54bf
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: a8cc7c1863c78100
CSeq: 7536 INVITE
Contact: <sip:249119630@192.168.203.23:5060;transport=udp>
P-Preferred-Identity: <sip:249119635@sip.vvn.telekom.sk:5060>
Privacy: id
Supported: replaces
User-Agent: Patton SN4960 4E30V UI 00A0BA026610 R6.5 2014-07-10 H323 RBS SIP M5T SIP Stack/4.2.8.10
Content-Type: application/sdp
Content-Length: 271

v=0
o=MxSIP 0 63 IN IP4 192.168.203.23
s=SIP Call
c=IN IP4 192.168.203.23
t=0 0
m=audio 10068 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=sendrecv
```

b) Incoming calls to the PBX

If the caller has activated the CLIR service, the incoming calls to the PBX suppress the number by replacing SIP URI values in the **„From”** and **„Contact”** headers. The caller ID is replaced to “anonymous” or “Restricted”.

Example of incoming call to the PBX with CLIR:

Calling party number – any number with CLIR (anonymous)
Called party number – 249119635 (IP address 192.168.203.23, port 5060)
Pilot number -249119630

„Request URI“ contains SIP URI in format **“pilot_number@IP_adresa_PBX”**
Header “FROM” contains SIP URI in format **“anonymous@anonymous.invalid”**
Header “TO” contains SIP URI in format **“called_number@sip.vvn.telekom.sk”**
Header “CONTACT” contains SIP URI **“Restricted@195.146.137.250”**

```
PBX --> ST
INVITE sip:249119630@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKse2q22100o70kq8sh5b1.1
From: "Anonymous"<sip:anonymous@anonymous.invalid;user=phone>;tag=1554756345-1438277126417-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW1925264173007151494525660@10.20.60.10
CSeq: 805282953 INVITE
Contact: "Anonymous"<sip:Restricted@195.146.137.250:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: application/dtmf-relay,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 293

v=0
o=BroadWorks 201241282 1 IN IP4 195.146.137.250
s=-
c=IN IP4 195.146.137.250
t=0 0
m=audio 24686 RTP/AVP 18 8 0 101
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
a=sendrecv
```

8. Routing of RTP stream (not valid for internet connections)

For routing of RTP stream on the VoIP systems by ST is used “direct routing” method. It means, if calling party and called party are connected via the same network segment, are inserted to the SDP protocol IP addresses both calling and called users. RTP streams will routed after answer between connected users directly. Possibility of this direct routing is evaluated on the ST VoIP system based on network architecturry.
We strongly recommend to deactivate all firewalls or iptables on the SIP interfaces of PBX, which are used on the communication with ST. IP routing should by on this interfaces to the default gateway, wich is specified in the hand over protocol.

9. Postscript

All SIP messages which are used in examples are authentic and they were generated by device Patton SN4960/4E30V, fw version R6.5.
If is not possible to set the PBX for compatibility with described scenarious, must by connected any SBC between the PBX and the ST VoIP interface. This SBC must to eliminate incopatibilities in the SIP communication. Based on business agreement is possible to provide this SBC by ST. In this issues must by negotiated individually all details of SIP comunication, IP addresses, number formats, domain names e.t.c. for communication between this SBC and PBX.