Zažime to spolu

Technical specifications for connecting SIP PBX to the „Business Trunk“ service by Slovak Telekom without registration, with static routing.

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1. Use of the service

The „Business Trunk“ service without register, with static routing 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 PBX with two or more SIP interfaces. Call hunting method to this interfaces is needed to negotiate for each case individually. 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 (on the DNS resolved as 195.146.137.250)
SIP proxy = sip.vvn.telekom.sk
Outbount proxy = sip.vvn.telekom.sk
Realm = BroadWorks
Transport protocol = UDP
Audio codecs: G711 Alaw, G711 Ulaw, G729a, G722
Transport DTMF: RFC 2833, payload type 101 telephone-event.

3. Autorization (authentication)

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 = calling number
Authentication name = pilot number
Password = common SIP password
Example of INVITE messages authorization is showed in example of part 5b.

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

4. Registration

This connection method does not use SIP message REGISTER. All calls to the PBX are routed from the ST systems on the specified static IP addresses and ports of the PBX SIP intefaces. Outgoing calls from the PBX are accepted from this IP addresses and ports only. If the PBX send SIP message REGISTER, the ST systems will send response „403 Forbidden”.
For checking of the connection status and the SIP communication the ST systems send to the PBX SIP message „OPTIONS“ in the time period 60 sec. If the SIP communication is OK, the "200 OK" response is expected.

Example of successfull SIP checkig by send message OPTIONS:

„Request URI“ contains **„trunkgroupname:port”,** Where **„trunkgroupname”** is name of trunk group is the ST.
Header “FROM” contains SIP URI in format **“ping@sip.vvn.telekom.sk”**
Header “TO” contains SIP URI in format **“ping@sip.vvn.telekom.sk”**

```
ST --> PBX
OPTIONS sip:ST-ViVieN-tkg1:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKvc8maj009ophi3v665k0
Call-ID: fbdc6f2bd0cc68b15fdc8464e5289ca9080slh1@195.146.137.250
To: <sip:ping@sip.vvn.telekom.sk:5060;user=phone>
From: <sip:ping@sip.vvn.telekom.sk;user=phone>;tag=1113776743e9ced37db3f43bec038dbc080slh1
Max-Forwards: 70
CSeq: 89699 OPTIONS
Route: sip:192.168.203.23:5060;lr
.
.
```

.
.
PBX --> ST
SIP/2.0 200 OK
Accept: application/sdp, application/dtmf-relay, application/hook-flash, application/QSIG, application/broadsoft, application/vnd.etsi.aoc+xml
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKvc8maj009ophi3v665k0
From: <sip:ping@sip.vvn.telekom.sk;user=phone>;tag=1113776743e9ced37db3f43bec038dbc080slh1
To: <sip:ping@sip.vvn.telekom.sk:5065;user=phone>;tag=1342690562
Call-ID: fbdc6f2bd0cc68b15fdc8464e5289ca9080slh1@195.146.137.250
CSeq: 89699 OPTIONS
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, UPDATE, REFER, REGISTER
Server: Patton SN4960 4E120V 00A0BA043CFB R6.11 2018-09-06 H323 RBS SIP M5T SIP Stack/4.2.22.37
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)

„Request URI" contains SIP URI in format **"called_number@IP_address_PBX"**
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:249119635@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKadp8vu00fgfglp08f720.1
From: <sip:0910500374@sip.vvn.telekom.sk;user=phone>;tag=1899919165-1438165129980-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW121849980290715-1560306303@10.20.60.10
CSeq: 749284735 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 192416707 1 IN IP4 195.146.137.250
s=-
c=IN IP4 195.146.137.250
t=0 0
m=audio 24204 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. In the 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

„Request URI" must contains SIP URI in format "**called_number@sip.vvn.telekom.sk**"
Header "FROM" must contains SIP URI in format "**calling_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 "**calling_number@IP_address_PBX**"
Header "P-PREFERRED-IDENTITY" does not have to be sent, if is the header sent, 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=z9hG4bK4a838e31121d2d975
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=666b6a6fd9
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: 6041fa00c1988346
CSeq: 31247 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, UPDATE, REFER, REGISTER
Contact: <sip:249119635@192.168.203.23:5060;transport=udp>
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: 284

v=0
o=MxSIP 0 762 IN IP4 192.168.203.30
s=SIP Call
c=IN IP4 192.168.203.30
t=0 0
m=audio 10826 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=ptime:20
a=sendrecv

ST --> PBX
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK4a838e31121d2d975
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=666b6a6fd9
To: <sip:0910500374@sip.vvn.telekom.sk:5060>; tag=1806272326-1545908494909
Call-ID: 6041fa00c1988346
CSeq: 31247 INVITE
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXjq6i04sdTvym55bBW",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=z9hG4bK4a838e31121d2d975
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=666b6a6fd9
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: 6041fa00c1988346
CSeq: 31247 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, UPDATE, REFER, REGISTER
Authorization: Digest username="249119630",realm="BroadWorks",nonce="BroadWorksXjq6i04sdTvym55bBW",
uri="sip:0910500374@sip.vvn.telekom.sk:5060",response="063914628c6550d6816f4e3c7be3f5f4",algorithm=MD5,qop=auth,
cnonce="3fcb6d3f",nc=00000002
Contact: <sip:249119635@192.168.203.23:5060;transport=udp>
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: 284

v=0
o=MxSIP 0 762 IN IP4 192.168.203.30
s=SIP Call
c=IN IP4 192.168.203.30
t=0 0
m=audio 10826 RTP/AVP 8 0 18 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:18 annexb=no
a=fmtp:101 0-16
a=ptime:20
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".
In this issue the 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**

Incoming "INVITE"
„Request URI" contains SIP URI in format **"B_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:249119635@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: BW2052406662907155733730 36@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: BW2052406662907155733730 36@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 original dialed 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**

Incoming "INVITE"
„Request URI" contains SIP URI in format **"B_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 **"B_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 **"B_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:249119635@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKjsudju10eon0rpsp64p1.1
From: <sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=1177363245-1438199095806-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW2144558062907156790787 85@10.20.60.10
CSeq: 766267648 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=z9hG4bK029298284e76b2f4c
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=cf2a49e5b3
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: 8b427d9b9c1efcc9
CSeq: 920 INVITE
Contact: <sip:249119635@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 then PBX

For outgoing CLIR calls from the PBX is necessary to insert to the INVITE message **„Privacy”** header in following format:
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

„Request URI“must contains SIP URI in format **“called_number@sip.vvn.telekom.sk”**
Header “FROM” must contains SIP URI in format **“calling_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 **“calling_number@IP_address_PBX”**
Header “P-PREFERRED-IDENTITY” does not have to be sent, if is the header sent, 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:249119635@sip.vvn.telekom.sk:5060>;tag=593b5b54bf
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: a8cc7c1863c78100
CSeq: 7536 INVITE
Contact: <sip:249119635@192.168.203.23:5060;transport=udp>
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)

„Request URI“ contains SIP URI in format **“called_number@IP_address_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”**

```
ST -->PBX
INVITE sip:249119635@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
.
.
SDP omitted
```

8. Routing of RTP stream

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 architectury.

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.