Technical specifications for connecting SIP PBX to the „Business Trunk“ service by Slovak Telekom.

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Version: 7.0

1. Use of the service

Service „Business Trunk“ is designed to provide connection of the customer's PBX to Public Telecommunication Network (PTN) by VoIP platform which is provided by Slovak Telekom, a.s. The signalling communication is running on SIP protocol. For transport is used UDP protocol. 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 should be created and assigned to the same trunk and also registered from/in the PBX based on section 3.

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 are only for the connections of private access ST lines e.g. MPLS, BIP, BCN, ... For the connections from public internet is necessary to use any public 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:
1.G729
2.G711 Alaw
3.G711 Ulaw
4.101 telephone-event (sending of DTMF in the voice channels).

3. Registration

a) **Dynamic – full**
Older method of trunk registration limited up to 150 accounts. In this way of registration is required to send “REGISTER” for each account (including the pilot numbers) separately. Field values of an individual headers in SIP messages are showed in attachment no. **1a.**

b) **Dynamic – pilot (new from september, 2015)**
The trunk is registered by pilot user only, that means a REGISTER is sent from pilot user only. Field values of an individual headers in SIP messages are described in attachment no. **1b.** This method of registration require to modify PBX settings for correct functionality of calls with correct presentation of caller's number. Field values of individual headers in SIP messages for incoming and outgoing calls are showed in attachments no. **2b** and **3b.**

c) **Static**
Older method of trunk registration for more than 150 accounts. For this registration is not sent REGISTER, but outgoing calls from PTN are routed staticaly to the IP address and port of the PBX. Incoming calls from the PBX are accepted only from this IP address and port.

From September 2015 as default option (independently on the number of trunks accounts) the **dynamic pilot registration** will be used**.** However, due to the high consumption of system resources for the static registration, this option will be provided only in the exceptional cases, e.g.: SIP trunk with more than 32 voice channels, or redundancy solution etc.
If is not possible to set dynamic registration on the PBX based on 3a) or 3b) and there are not conditions for static registration, there is necessary to include conversion gateway SIP/SIP between PBX and ST devices. This way to ensure correct registration and screening.

4. Authentication

All numbers assigned to the trunk are using common authentication name (pilot number without zero on the first position) and common SIP password which are mentioned in protocol of service delivered. Below mentioned authentification data has to be set for each number from the trunk and used for authorization of a registrations and calls by request „401 Unauthorized“.

User name *= Assigned Number*
Authentication name *= common authentication name (pilot number without zero on the position)*
Password *= common SIP password*

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

5. Format of numbers

Calls from the PTN to the PBX

- 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). For some older trunks may be used internal system domain “as” (e.g.: 249119635@as).
- 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). For some older trunks may be used a domain name IP address of the outbound proxy “195.146.137.250” (e.g.: 00420705333555@195.146.137.250).

Please look at the example of theincoming INVITE in the attachment **2a** and **2b,** based on registration method.

Calls from the PBX to the PTN

- Called number = any format (local, national, international, shortened numbers, ..., the same as calls from standard fixed line) in SIP URI must be used public domain „sip.vvn.telekom.sk (e.g.: 0910500374@sip.vvn.telekom.sk, 44638255@sip.vvn.telekom.sk, 1181@sip.vvn.telekom.sk, ...)
- Calling numbers = number in national format without zero on the first position, in SIP URI must be used public domain “sip.vvn.telekom.sk” (e.g.: 249119521@sip.vvn.telekom.sk)

Please look at the example of the outgoing INVITE in the attachment **3a** and **3b,** based on registration method.

6. Call forwarding from the PBX 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:

1) 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”. Please find example of this call in the attachment **4a** and **4b,** based on registration method.
2) Send new message **“INVITE**”, where new destination is placed in the **“Request Line”** and also in the header **“TO”**. The header **“FROM”** must contain the SIP URI of the origin called number or pilot number, based on registration method. Also this INVITE must contain the header **“DIVERSON”**with the SIP URI of the origin called number. The INVITE message should also contain the other parameters of the forwarding. The header **“P-Preffered-Identity”**, have to contain the SIP URI origin calling number. Please find example of this call forwarded back to the PTN in the attachment 5a and 5b, based on registration method.

7. Calling line identity restriction – CLIR

For “CLIR” calls from the PBX to PTN is necessary to add to the outgoing INVITE parameters „P-Preferred-Identity“ and „Privacy“ in following format:
P-Preferred-Identity: <sip:SIP URI calling number>
Privacy: id
Parameters „From“ and „Contact“ must stay in exactly the same format as by calls without CLIR.
Calls will be not accepted when calling number is not sent or the parameter SIP URI in the header „From“ will be anonymous@anonymous.invalid.

Please find example of outgoing INVITE from the PBX to the PTN with CLIR in the attachment **6a** and **6b,** based on registration method.

Please find example of incoming INVITE from the PTN to the PBX with CLIR in the attachment **7a** and **7b,** based on registration method.

8. Attachements

SIP messages, used as examples in the attachements are authentic and they are generated on the Patton SN4960/4E30V device. Fw. version R6.5.
SDP protocol is ommited in the INVITEs in attachements 4 and 5.

Attachment no. 1 – Registering

a) Dynamic - full

Example of registering the trunk account 249119635 (is not pilot)
User name = 249119635
Authentication name (pilot number) = 249119630
It is necessary to use same domain names specified in SIP URI of the the headers „FROM“ and „TO“.

REGISTER sip:sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK1c304c6f4fad52394
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=c7e4024878
To: <sip:249119635@sip.vvn.telekom.sk:5060>
Call-ID: 5a9d648f4a6c7075
CSeq: 25089 REGISTER
Contact: <sip:249119635@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

SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK1c304c6f4fad52394
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=c7e4024878
To: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=127025642-1438164349586
Call-ID: 5a9d648f4a6c7075
CSeq: 25089 REGISTER
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXicolx66qT90u6z9BW",realm="BroadWorks",algorithm=MD5
Content-Length: 0

REGISTER sip:sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK6786379e20184cf7e
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=c7e4024878
To: <sip:249119635@sip.vvn.telekom.sk:5060>
Call-ID: 5a9d648f4a6c7075
CSeq: 25090 REGISTER
Authorization: Digest
username="249119630",*realm="BroadWorks",nonce="BroadWorksXicolx66qT90u6z9BW",uri="sip:sip.vvn.telekom.sk:5060",response="6d185 b1cc84e158062d582f2306282aa",algorithm=MD5,qop=auth,cnonce="1a498b34",nc=00000001*
Contact: <sip:249119635@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

SIP/2.0 200 OK
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK6786379e20184cf7e
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=c7e4024878
To: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=1077249181-1438164349622
Call-ID: 5a9d648f4a6c7075
CSeq: 25090 REGISTER
Contact: <sip:249119635@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

b) Register pilot number only

Example of registering the pilot trunk account 249119630
User name (pilot number) = 249119630
Authentication name (pilot number) = 249119630
It is necessary to use same domain names specified in SIP URI of the the headers „FROM“ and „TO“.

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

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

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

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

Attachment no. 2 – Incoming INVITE (call from the PTN to the PBX)

a) Dynamic – full or static registration method

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

The “Request URI” contains SIP URI in format **“called_number@ PBX_IP_address”** (with values from the header “CONTACT” from Register)
The header “FROM” contains SIP URI in format **“calling_number@sip.vvn.telekom.sk”**
The header “TO” contains SIP URI in format **“called_number@sip.vvn.telekom.sk”**

```
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) Register pilot number only

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

The "Request URI" contains SIP URI in format **"pilot_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
The header "TO" contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**

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: BW1224389852907151077616993@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

Attachment no. 3 - Outgoing INVITE (call from the PBX to the PTN)

a) Dynamic – full or static registration method

Calling – 249119635 (IP address 192.168.203.23, port 5060)
Called – 0910500374

The "Request URI" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "FROM" must contain SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
The header "TO" must contain SIP URI in format **"called_number @sip.vvn.telekom.sk"**
The header "CONTACT" must contain SIP URI in format **"calling_number @PBX_IP_address"**

Calling party number must to be from the assigned range of numbers.

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bK0f9818fc7fa5d40ca
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=d7d0005c86
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: e19d814378517288
CSeq: 22102 INVITE
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: 271

v=0
o=MxSIP 0 17 IN IP4 192.168.203.23
s=SIP Call
c=IN IP4 192.168.203.23
t=0 0
m=audio 10018 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) Register pilot number only

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

The "Request URI" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "FROM" must contain SIP URI in format **"pilot_number @sip.vvn.telekom.sk**"
The header "TO" must contain SIP URI in format **"called_number @sip.vvn.telekom.sk"**
The header "CONTACT" must contain SIP URI in format **"pilot_number @PBX_IP_address"**
The header "P-PREFERRED-IDENTITY" must contain SIP URI in format **"calling_number@sip.vvn.telekom.sk"**

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

```
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
```

Attachment no. 4 - Incoming call transferred by “302 Moved Temporaily”

a) Dynamic – full or static registration method

Calling – 0258823254 (VTS) – **A**
Called – 249119635 (IP address 192.168.203.23, port 5060) - **B**
Forwarded to 0910500374 - **C**

Incoming “INVITE”
The “Request URI” contains SIP URI in format **“B_number@ PBX_IP_address”** (with values from the header “CONTACT” from Register)
The header “FROM” contains SIP URI in format **“A_number@sip.vvn.telekom.sk”**
The header “TO” contains SIP URI in format **“B_number@sip.vvn.telekom.sk”**

Response “302 Moved Temporaily”
The header “FROM” must contain SIP URI in format “**A_number@sip.vvn.telekom.sk**”
The header “TO” must contain SIP URI in format “**B_number@sip.vvn.telekom.sk**”
The header “CONTACT” must contain SIP URI in format “**C_number@sip.vvn.telekom.sk**”

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

```
INVITE sip:249119635@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bK4ig24130cg7hno0sk4s0.1
From: <sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=993683331-1438195539122-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW204539122290715-1086428789@10.20.60.10
CSeq: 764489306 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
.
.
.

SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bK4ig24130cg7hno0sk4s0.1
From: "<sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=993683331-1438195539122-
To: "Display name" <sip:249119635@sip.vvn.telekom.sk:5060;user=phone>;tag=3742156777
Call-ID: BW204539122290715-1086428789@10.20.60.10
CSeq: 764489306 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

20:45:38  SIP_TR> [STACK] < 429 Stack: from 195.146.137.250
ACK sip:249119635@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bK4ig24130cg7hno0sk4s0.1
CSeq: 764489306 ACK
From: "<sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=993683331-1438195539122-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>;tag=3742156777
Call-ID: BW204539122290715-1086428789@10.20.60.10
Max-Forwards: 69
Content-Length: 0
```

b) Register pilot number only

Calling – 0258823254 (VTS) – **A**
Called – 249119635 (IP address 192.168.203.23, port 5060) **- B**
Forwarded to 0910500374 **- C**
Pilot number -249119630

Incoming "INVITE"
The "Request URI" contains SIP URI in format **"pilot_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**
The header "TO" contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**

Response "302 Moved Temporaily"
The header "FROM" must contain SIP URI in format "**A_number@sip.vvn.telekom.sk**"
The header "TO" must contain SIP URI in format "**B_number@sip.vvn.telekom.sk**"
The header "CONTACT" must contain SIP URI in format "**C_number@sip.vvn.telekom.sk**"

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

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
.
.
.

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

20:52:40 SIP_TR> [STACK] < 428 Stack: from 195.146.137.250
ACK 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
CSeq: 764700078 ACK
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
Max-Forwards: 69
Content-Length: 0

Attachment no. 5 – Incoming call transferred by new INVITE

a) Dynamic – full or static registration method

Calling – 0258823254 (VTS) – **A**
Called – 249119635 (IP address 192.168.203.23, port 5060) **- B**
Forwarded to 0910500374 **- C**

Incoming "INVITE"
The "Request URI" contains SIP URI in format **"B_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**
The header "TO" contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**

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

In the case of redirection to foregin number I would like to advice to set A-Party number in header "P-Preferred-Identity" to international format E164.

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

```
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: BW214455806290715679078785@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
.
.
.
```

```
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
.
.
.
```

b) Register pilot number only

Calling – 0258823254 (VTS) – **A**
Called – 249119635 (IP address 192.168.203.23, port 5060) **- B**
Forwarded to 0910500374 **- C**
Pilot number -249119630

Incoming "INVITE"
The "Request URI" contains SIP URI in format **"pilot_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI in format **"A_number@sip.vvn.telekom.sk"**
The header "TO" contains SIP URI in format **"B_number@sip.vvn.telekom.sk"**

Outgoing "INVITE"
The "Request URI" must contain SIP URI in format "**C_number@sip.vvn.telekom.sk**"
The header "FROM" must contain SIP URI in format "**pilot_number@sip.vvn.telekom.sk**"
The header "TO" must contain SIP URI in format "**C_number@sip.vvn.telekom.sk**"
The header "CONTACT" must contain SIP URI in format "**pilotne _number@IP_address_PBX**"
The header "DIVERSION" must contain SIP URI in format "**B_number@sip.vvn.telekom.sk**"
The header "P-PREFERRED-IDENTITY" must contain SIP URI in format **A_number@sip.vvn.telekom.sk**

In the case of redirection to foregin number I would like to advice to set A-Party number in header "P-Preferred-Identity" to international format E164.

The port number must be specified only if is used different from 5060. Its not necessary to specify default port number: 5060.

INVITE sip:249119630@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKu8nmrf0060tglqk4g611.1
From:<sip:0258823254@sip.vvn.telekom.sk;user=phone>;tag=543364054-1438204702593-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW2318225932907155461796 77@10.20.60.10
CSeq: 769071041 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
.
.
.

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
.
.
.

Attachment no. 6 Outgoing call from the PBX to the PTN with CLIR

a) Dynamic – full or static registration method

Calling – 0249119635 (IP address 192.168.203.23, port 5060)
Called – 0910500374

The "Request URI" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "FROM" must contain SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
The header "TO" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "CONTACT" must contain SIP URI in format **"calling_number@IP_address_PBX"**
The header "P-PREFERRED-IDENTITY" must contain SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
The header "PRIVACY" must contain value **"id"**

INVITE sip:0910500374@sip.vvn.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.23:5060;branch=z9hG4bKff6609fa347bf1bd1
Max-Forwards: 70
From: <sip:249119635@sip.vvn.telekom.sk:5060>;tag=7b913e90c3
To: <sip:0910500374@sip.vvn.telekom.sk:5060>
Call-ID: e3f48cb833a97e09
CSeq: 22825 INVITE
Contact: <sip:249119635@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 59 IN IP4 192.168.203.23
s=SIP Call
c=IN IP4 192.168.203.23
t=0 0
m=audio 10064 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) Register pilot number only

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

The "Request URI" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "FROM" must contain SIP URI in format **"pilot_number@sip.vvn.telekom.sk"**
The header "TO" must contain SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "CONTACT" must contain SIP URI in format **"pilot_number @IP_address_PBX"**
The header "P-PREFERRED-IDENTITY" must contain SIP URI in format **"calling_number@sip.vvn.telekom.sk"**
The header "PRIVACY" must contain value **"id"**

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

Attachment no. 7 – Incoming call from the PTN to the PBX with CLIR

a) Dynamic – full or static registration method

Calling – Anonymous
Called – 249119635 (IP address 192.168.203.23, port 5060)

The "Request URI" contains SIP URI in format **"called_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI **"anonymous@anonymous.invalid"**
The header "TO" contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "CONTACT" contains SIP URI **"Restricted@195.146.137.250"**

INVITE sip:249119635@192.168.203.23:5060;transport=udp SIP/2.0
Via: SIP/2.0/UDP 195.146.137.250:5060;branch=z9hG4bKoev30j0038a18qgmd4s0.1
From: "Anonymous"<sip:anonymous@anonymous.invalid;user=phone>;tag=46314389-1438277342623-
To: "Display name"<sip:249119635@sip.vvn.telekom.sk:5060;user=phone>
Call-ID: BW192902623300715-162156763@10.20.60.10
CSeq: 805391056 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 201257108 1 IN IP4 195.146.137.250
s=-
c=IN IP4 195.146.137.250
t=0 0
m=audio 24698 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

b) Register pilot number only

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

The "Request URI" contains SIP URI in format **"pilot_number@ PBX_IP_address"** (with values from the header "CONTACT" from Register)
The header "FROM" contains SIP URI **"anonymous@anonymous.invalid"**
The header "TO" contains SIP URI in format **"called_number@sip.vvn.telekom.sk"**
The header "CONTACT" contains SIP URI **"Restricted@195.146.137.250"**

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: BW192526417300715149452566 0@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