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

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1. Use of the service

The „KVPS“ service is designed to provide connection of the customer’s PBX to the Public Telecommunication Network (PTN) by Slovak Telekom, a.s.(ST) VoIP platforms. Signaling communication is running on SIP protocol, used transport protocol must by UDP. Telephone numbers assigned to the trunk must be from the same range of numbers (DDI) 0912xxxxxx. On the connection is possible to use special data connect only.

For this service is not possible by use other data connection.

2. General settings

DNS = 195.146.137.211; 195.146.137.212
NTP = 195.146.137.211; 195.146.137.212
SIP domain = sip.mddi.telekom.sk (na on the DNS resolved as 195.146.137.253)
SIP proxy = sip.mddi.telekom.sk
Outbount proxy = sip.mddi.telekom.sk
SIP port on the ST side = 5060
Realm = BroadWorks
Transport protokol = UDP
Audio codecs: G711 Alaw, G711 Ulaw
DTMF transport: 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“.
Pilot number = first number from assigned number rage in national format (e.g.: 912911000)
User name = assigned number in national format (912xxxxxx)
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.mddi.telekom.sk”**
Header “TO” contains SIP URI in format **“ping@sip.mddi.telekom.sk”**

ST --> PBX
OPTIONS sip:MDDI-912911xxx:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bK9ac6e22010qh46tim6u0
Call-ID: 4d0ba393dfb67f91bbd07e0ce402274f080cbs3@195.146.137.253
To: <sip:ping@sip.mddi.telekom.sk:5060;user=phone>
From: <sip:ping@sip.mddi.telekom.sk;user=phone>;tag=79b5df44cbb51fae47a79f2982d3fef4080cbs3
Max-Forwards: 70
CSeq: 112376 OPTIONS
Route: <sip:192.168.203.22:5060;lr>
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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 195.146.137.253:5060;branch=z9hG4bK9ac6e22010qh46tim6u0
From: <sip:ping@sip.mddi.telekom.sk;user=phone>;tag=79b5df44cbb51fae47a79f2982d3fef4080cbs3
To: <sip:ping@sip.mddi.telekom.sk:5060;user=phone>;tag=2149581721
Call-ID: 4d0ba393dfb67f91bbd07e0ce402274f080cbs3@195.146.137.253
CSeq: 112376 OPTIONS
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, UPDATE, REFER, REGISTER
Server: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
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.mddi.telekom.sk" (e.g.: 912911335@sip.mddi.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. mddi.telekom.sk" (e.g.: 0335920916@sip. mddi.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. mddi.telekom.sk" (e.g.: 00420705333555@sip. mddi.telekom.sk).

Example of incoming call to the PBX :

Calling party number – 0258823254
Called party number – 912911335 (IP address 192.168.203.22, port 5060)

„Request URI" contains SIP URI in format **"called_number@IP_address_PBX"**
Header "FROM" contains SIP URI in format **"calling_number@sip.mddi.telekom.sk"**
Header "TO" contains SIP URI in format **"called_number@sip.mddi.telekom.sk"**

ST --> PBX
INVITE sip:912911335@192.168.203.22:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bK5b2nac202ogh06l2m0g1.1
From: <sip:0258823254@sip.mddi.telekom.sk;user=phone>;tag=597325752-1519204917387-
To: <sip:912911335@sip.mddi.telekom.sk:5060;user=phone>
Call-ID: BW1021573872102181282053884@10.20.60.50
CSeq: 466989126 INVITE
Contact: <sip:0258823254@195.146.137.253:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Call-Info: <sip:10.20.60.50>;appearance-index=1
Recv-Info: x-broadworks-client-session-info
Accept: application/btbc-session-info,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 290

v=0
o=BroadWorks 806584452 1 IN IP4 195.146.137.253
s=-
c=IN IP4 195.146.137.253
t=0 0
m=audio 20780 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) Outgoing calls from the PBX to the ST

- Called number = national, international, or E164 format. In the SIP URI must by used public domain „sip.mddi.telekom.sk” (e.g.: national: 0258823254@sip.mddi.telekom.sk, international: 00436769550786@sip.mddi.telekom.sk, E164: +436769550786@sip.mddi.telekom.sk).
- Calling number = number in national format without zero on the first position. In the SIP URI must by used domain „sip.mddi.telekom.sk” (e.g.: 912911335@sip.mddi.telekom.sk)

Example of outgoing call with authorization of this call:

Calling party number – 912911335 (IP address 192.168.203.22, port 5060)
Called party number – 0258823254

„Request URI“ must contains SIP URI in format “**called_number@sip.mddi.telekom.sk**”
Header “FROM” must contains SIP URI in format “**calling_number@sip. mddi.telekom.sk**”
Header “TO” must contains SIP URI in format “**called_number @sip. mddi.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.mddi.telekom.sk”**

PBX --> ST
INVITE sip:0258823254@sip.mddi.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.22:5060;branch=z9hG4bK3d5a4c2eecebf252e
Max-Forwards: 70
From: <sip:912911335@sip.mddi.telekom.sk:5060>;tag=fec7f15c8b
To: <sip:0258823254@sip.mddi.telekom.sk:5060>
Call-ID: ad31e08bd3d91d20
CSeq: 18604 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, REFER, REGISTER
Contact: <sip:912911335@192.168.203.22:5060;transport=udp>
Supported: replaces
User-Agent: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
Content-Type: application/sdp
Content-Length: 226

v=0
o=MxSIP 0 1134 IN IP4 192.168.203.22
s=SIP Call
c=IN IP4 192.168.203.22
t=0 0
m=audio 11160 RTP/AVP 8 0 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=sendrecv

ST --> PBX
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 192.168.203.22:5060;branch=z9hG4bK3d5a4c2eecebf252e
From: <sip:912911335@sip.mddi.telekom.sk:5060>;tag=fec7f15c8b
To: <sip:0258823254@sip.mddi.telekom.sk:5060>
Call-ID: ad31e08bd3d91d20
CSeq: 18604 INVITE
WWW-Authenticate: DIGEST qop="auth",nonce="BroadWorksXjq6i04sdTvym55bBW",realm="BroadWorks",algorithm=MD5
Content-Length: 0
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PBX --> ST
INVITE sip:0258823254@sip.mddi.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.22:5060;branch=z9hG4bK3d5a4c2eecebf252e
Max-Forwards: 70
From: <sip:912911335@sip.mddi.telekom.sk:5060>;tag=fec7f15c8b
To: <sip:0258823254@sip.mddi.telekom.sk:5060>
Call-ID: ad31e08bd3d91d20
CSeq: 18604 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, REFER, REGISTER
Authorization: Digest username="912911000",realm="BroadWorks",nonce="BroadWorksXjq6i04sdTvym55bBW", uri="sip:0258823254@sip.mddi.telekom.sk:5060",response="063914628c6550d6816f4e3c7be3f5f4",algorithm=MD5,qop=auth, cnonce="3fcb6d3f",nc=00000002
Contact: <sip:912911335@192.168.203.22:5060;transport=udp>
Supported: replaces
User-Agent: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
Content-Type: application/sdp
Content-Length: 226

v=0
o=MxSIP 0 1134 IN IP4 192.168.203.22
s=SIP Call
c=IN IP4 192.168.203.22
t=0 0
m=audio 11160 RTP/AVP 8 0 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16
a=sendrecv

6. Call forwarding from the PBX back to the PTN

The presentation of a origin Calling ID when a call is forwarded back to PTN is possible by two ways:

a) On the incoming **„INVITE”** request send response **„302 Moved Temporarily”**, where new target is in the header **„Contact”.** In the SIP URI must be used public domain „sip.mddi.telekom.sk”.
SIP dialog between the ST an the PBX will finished after receive SIP response „302 Moved Temporarily”. Redirection is realized on the voice system of ST. Status info (succesfull, unsuccesfull, duration ...) about this redirected call is 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 – 912911350 (IP address 192.168.203.22, 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.mddi.telekom.sk”**
Header “TO” contains SIP URI in format **“B_number@sip.mddi.telekom.sk”**

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

ST -->PBX
INVITE sip:912911350@192.168.203.22:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bK08d977208o7h031np3n0.1
From: <sip:0258823254@sip.mddi.telekom.sk;user=phone>;tag=568154802-1519243945605-
To: <sip:912911350@sip.mddi.telekom.sk:5060;user=phone>
Call-ID: BW2112256052102181635557623@10.20.60.50
CSeq: 486503235 INVITE
Contact: <sip:0258823254@195.146.137.253:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Call-Info: <sip:10.20.60.50>;appearance-index=1
Accept: application/btbc-session-info,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 219
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.
SDP omitted

PBX --> ST
SIP/2.0 302 Moved Temporarily
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bK08d977208o7h031np3n0.1
From: <sip:0258823254@sip.mddi.telekom.sk;user=phone>;tag=568154802-1519243945605-
To: <sip:912911350@sip.mddi.telekom.sk:5060;user=phone>;tag=4011807640
Call-ID: BW2112256052102181635557623@10.20.60.50
CSeq: 486503235 INVITE
Contact: <sip:0910500374@sip.mddi.telekom.sk:5060;transport=udp>
Server: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
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 case, the call passes via PBX, which has all informations about this call.
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.

Príklad volania z ST do PBX, presmerovaného späť na iné číslo novým volaním :

Calling party number – 0258823254 – **A_number**
Called party number – 912911350 (IP address 192.168.203.22, 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.mddi.telekom.sk"**
Header "TO" contains SIP URI in format **"B_number@sip.mddi.telekom.sk"**

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

ST -->PBX
INVITE sip:912911350@192.168.203.22:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bKb336sk200gcgcvg9p300.1
From: <sip:0258823254@sip.mddi.telekom.sk;user=phone>;tag=1879937546-1519247221704-
To: <sip:912911350@sip.mddi.telekom.sk:5060;user=phone>
Call-ID: BW2207017042102181885125877@10.20.60.50
CSeq: 488141285 INVITE
Contact: <sip:0258823254@195.146.137.253:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Accept: application/btbc-session-info,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 219
.
..
SDP omitted

PBX --> ST
INVITE sip:0910500374@sip.mddi.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.22:5060;branch=z9hG4bK5a722a7341738351c
Max-Forwards: 70
From: <sip:912911350@sip.mddi.telekom.sk:5060>;tag=ef8b3dfb7e
To: <sip:0910500374@sip.mddi.telekom.sk:5060>
Call-ID: b117ec0c68b1690f
CSeq: 19266 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, REFER, REGISTER
Contact: <sip:912911350@192.168.203.22:5060;transport=udp>
Diversion: <sip:912911350@sip.mddi.telekom.sk:5060>;reason=unconditional;screen=no;privacy=off;counter=1
P-Preferred-Identity: <sip:+421258823254@sip.mddi.telekom.sk:5060>
Supported: replaces
User-Agent: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
Content-Type: application/sdp
Content-Length: 226
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.
.
SDP omitted

7. Calling line identity restriction – CLIR

a) Odchádzajúce volania z PBX do ST

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 – 912911350 (IP address 192.168.203.22, port 5060)
Called party number – 0258823254

„Request URI“must contains SIP URI in format **“called_number@sip.mddi.telekom.sk”**
Header “FROM” must contains SIP URI in format **“calling_number@sip.mddi.telekom.sk”**
Header “TO” must contains SIP URI in format **“called_number@sip.mddi.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.mddi.telekom.sk”**
Header “PRIVACY” must contains value **“id”**

PBX --> ST

INVITE sip:0258823254@sip.mddi.telekom.sk:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.203.22:5060;branch=z9hG4bKcf91f16e54d153457
Max-Forwards: 70
From: <sip:912911350@sip.mddi.telekom.sk:5060>;tag=614c5bd33c
To: <sip:0258823254@sip.mddi.telekom.sk:5060>
Call-ID: 420be66cfbdfdba6
CSeq: 19171 INVITE
Allow: ACK, BYE, CANCEL, INVITE, NOTIFY, OPTIONS, INFO, REFER, REGISTER
Contact: <sip:912911350@192.168.203.22:5060;transport=udp>
Privacy: id
Supported: replaces
User-Agent: Patton SN4970 1E30V 00A0BA09C6CE R6.9 2016-09-01 H323 RBS SIP M5T SIP Stack/4.2.14.18
Content-Type: application/sdp
Content-Length: 226

v=0
o=MxSIP 0 1159 IN IP4 192.168.203.22
s=SIP Call
c=IN IP4 192.168.203.22
t=0 0
m=audio 11192 RTP/AVP 8 0 101
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
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”.

Príklad prchádzajúceho volania z ST do PBX s potlačením zobrazenia čísla volajúceho (CLIR) :

Calling party number – any number with CLIR (anonymous)
Called party number – 912911350 (IP address 192.168.203.22, 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.mddi.telekom.sk”**
Header “CONTACT” contains SIP URI **“Restricted@195.146.137.253”**

ST -->PBX

INVITE sip:912911350@192.168.203.22:5060 SIP/2.0
Via: SIP/2.0/UDP 195.146.137.253:5060;branch=z9hG4bK08d977208o7h031np3n0.1
From: "Anonymous"<sip:anonymous@anonymous.invalid;user=phone>;tag=568154802-1519243945605-
To: <sip:912911350@sip.mddi.telekom.sk:5060;user=phone>
Call-ID: BW2112256052102181635557623@10.20.60.50
CSeq: 486503235 INVITE
Contact: "Anonymous"<sip:Restricted@195.146.137.253:5060;transport=udp>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Call-Info: <sip:10.20.60.50>;appearance-index=1
Recv-Info: x-broadworks-client-session-info
Accept: application/btbc-session-info,application/media_control+xml,application/sdp,multipart/mixed
Supported:
Max-Forwards: 69
Content-Type: application/sdp
Content-Length: 219
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.
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.
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.