Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the British Telecom (Germany) SIP trunking service and an Avaya Distributed Office – Issue 1.0

Abstract

These Application Notes describe the steps to configure SIP trunking between the British Telecom (BT) Germany SIP trunking service and an Avaya SIP telephony solution consisting of Avaya Distributed Office, and various Avaya telephones.

Enterprise customers with this Avaya SIP-based solution can connect via dedicated Internet access to use British Telecom as a service provider to complete PSTN calls. This allows customers with a converged network to lower PSTN telecommunication costs, easily obtain local number presence without offices in each geographic area, and easily manage their network services using web-based tools.

Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps for configuring Session Initiation Protocol (SIP) trunking between the British Telecom SIP trunking service and an Avaya SIP telephony solution consisting of Avaya Distributed Office and Avaya telephones.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 [6] is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the network service offered by British Telecom. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

1.1. British Telecom SIP trunking service - IP Trunking Overview

The British Telecom SIP trunking service with IP trunking used within these Application Notes is serves as an interface between Avaya Telephones and ISDN, GSM and analog endpoints communicating with the British Telecom network.

1.2. SIP Trunking Configuration

Figure 1 illustrates a sample Avaya SIP telephony solution connected to British Telecom SIP trunking service using SIP trunking. This is the configuration that was used during the DevConnect compliance testing.

The Avaya SIP telephony solution used as a simulated customer location included:

- An Avaya Distributed Office i120 or i40
- Avaya 4600 Series IP telephones using the H.323 software bundle
- Avaya 9600 Series IP telephones using the SIP software bundle
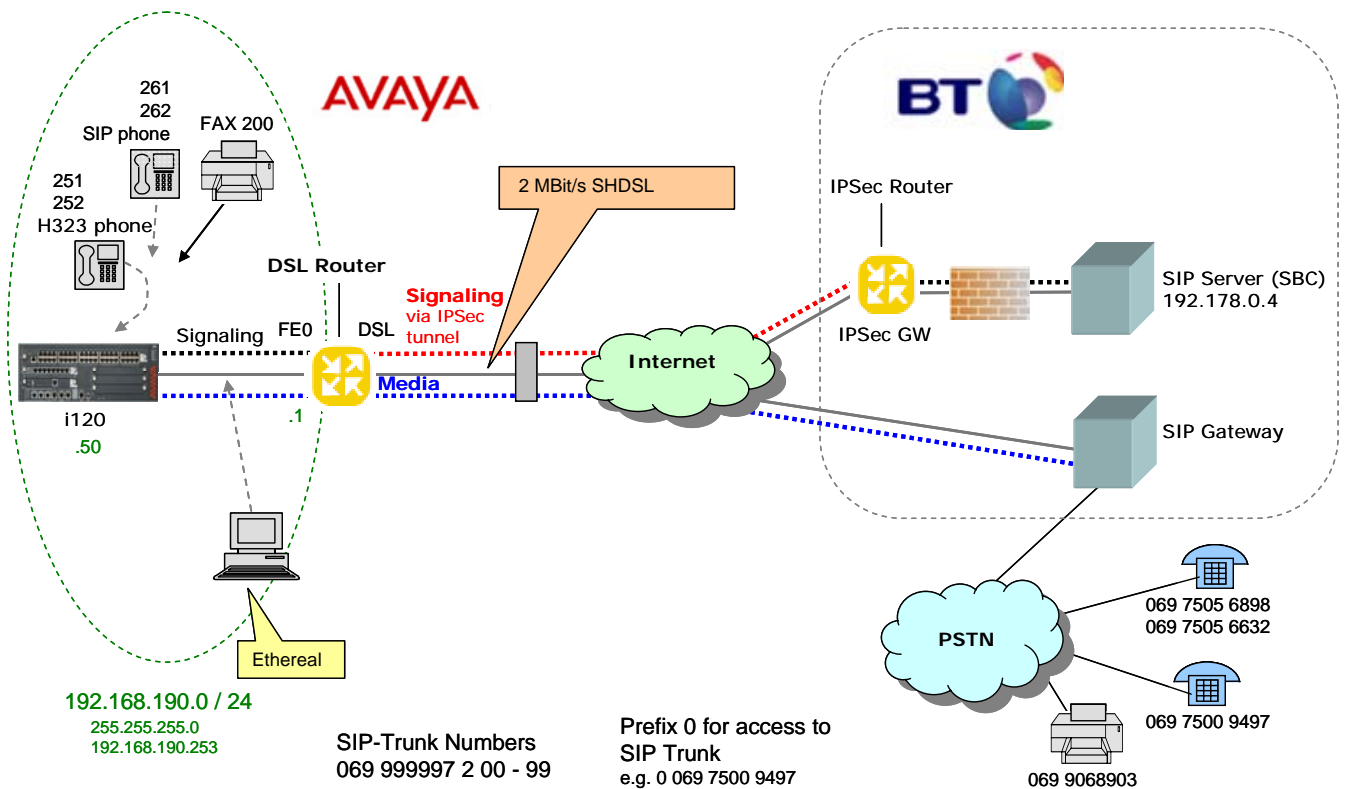- Avaya Analog phones

Figure 1: Avaya SIP Telephony Solution using British Telecom SIP trunking service

1.3. Call Flows

To better understand how calls are routed between the ISDN network and the enterprise site shown in
Figure 1 using SIP trunks, two call flows are described in this section.

The first call scenario illustrated in **Figure 2** is a PSTN call to the enterprise site terminating on a telephone supported by Avaya Distributed Office.

1. A user on the ISDN network dials a British Telecom SIP trunking service provided DID number that is assigned to an Avaya Distributed Office telephone at the enterprise site. The ISDN network routes the call to the British Telecom SIP trunking service network (as the local service provider), which in turn routes the DID number to the assigned customer.

2. Based on the DID number, the British Telecom SIP trunking service offers the call to Avaya Distributed Office using SIP signaling messages sent over the managed access facility. Note that the assignment of the DID number and the address of the Avaya

Distributed Office server was previously established during the ordering and provisioning of the service.

3. Avaya Distributed Office terminates the call to the Avaya telephone as shown in **Figure 2**.
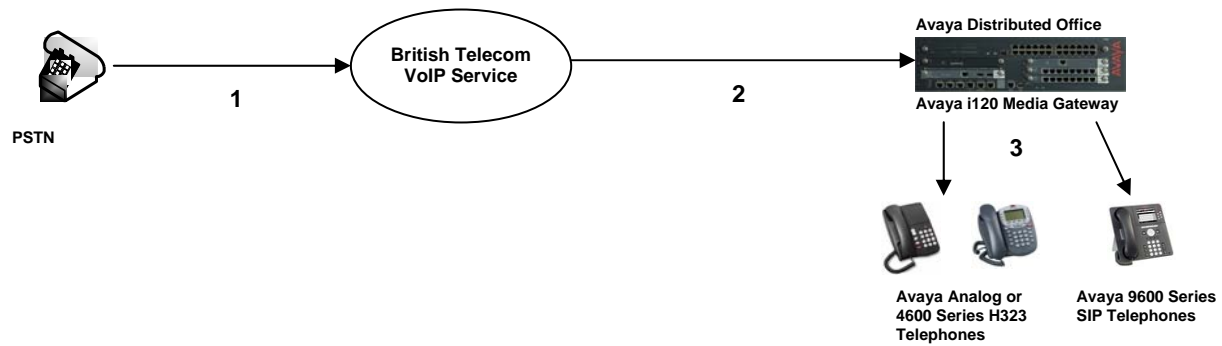

Figure 2: Incoming ISDN Calls to Avaya Distributed Office

The second call scenario illustrated in **Figure 3** is an outgoing call from an Avaya telephone at the enterprise site to the ISDN network via the SIP trunk to the British Telecom SIP trunking service.

1. An Avaya telephone served by Avaya Distributed Office originates a call to a user on the ISDN network.

2. The call request is handled by Avaya Distributed Office where origination treatment such as Automatic Route Selection (ARS) is performed. Avaya Distributed Office selects the SIP trunk and sends the SIP signaling messages to the British Telecom SIP trunking service.

3. The British Telecom SIP trunking service completes the call to the ISDN.
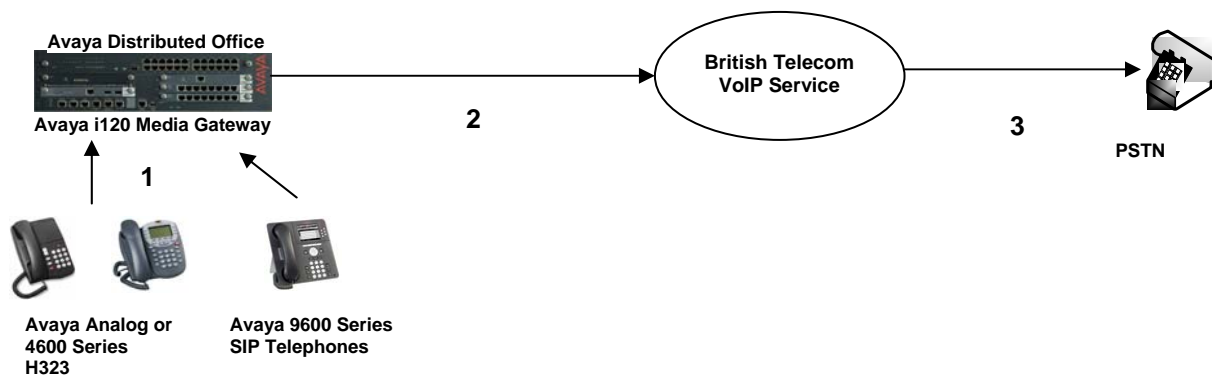
Figure 3: Outgoing Calls from Avaya Distributed Office to the ISDN

Appendix A illustrates examples of the SIP INVITE messages sent on the SIP trunk to begin each call.

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Avaya SIP Telephony Solution Components	
Component	**Software Version**
Avaya i120 Media Gateway	Avaya Distributed Office 1.2 1.2.0_17.03 *I120*: 28.10.50 HW-Suffix: B HW-Vintage: 3
Avaya MM711	FW-Version: 89 HW-Suffix: A HW-Vintage: 63
Avaya 4621 H.323 IP Telephones	*Application file:* a20d01b2_8_3_0.bin *Boot file:* b20d01b2_8_3.bin
Avaya 6221 Analog Telephones	n / a
Avaya 9620 SIP Telephones	*Application file:* SIP96xx_2_0_3_0.bin *Boot file:* Hb96xxual_50.bin
British Telecom SIP trunking service Components	
Component	**Version**
British Telecom VoIP Platform	Avaya SIP Trunk 1.0

Table 1: Equipment and Software used in the tested configuration

The specific configuration above was used for the British Telecom SIP trunking service interoperability testing. Note that this solution is extensible to other Avaya Distributed Office platform (e.g. i40) running a similar release.

3. Configure Avaya Distributed Office

This section describes the steps for configuring a SIP trunk on Avaya Distributed Office with the necessary signaling and media gateway characteristics for the SIP trunk connection with the British Telecom SIP trunking service.

In the Avaya SIP trunk architecture, the Avaya Distributed Office acts as a SIP proxy through which all incoming and outgoing SIP messages flow to the British Telecom SIP trunking service. There is a direct SIP signaling path between the British Telecom SIP trunking service and Avaya Distributed Office.

The SIP trunk configuration also allows a shared trunking solution that supports SIP endpoint devices (such as Avaya 9600 Series SIP telephones) using Avaya Distributed Office. In this configuration, the SIP endpoints are registered with Avaya Distributed Office and have calling privileges and features managed by Avaya Distributed Office. Avaya Distributed Office acts as a Back-to-Back SIP user agent for the SIP endpoint, when a call involves a SIP trunk to the British Telecom SIP Trunk Service.

Incoming SIP messages arrives at the Avaya Distributed Office, further incoming call treatment such as incoming digit translations, etc., may be performed.

All outgoing calls to the ISDN network are processed within Avaya Distributed Office and are subject to outbound features such as automatic route selection (ARS) and digit manipulation. Once Avaya Distributed Office selects a SIP trunk, the SIP signaling is going direct into outbound SIP messages to the British Telecom SIP trunking SIP Application Server.

The Avaya Distributed Office configuration was performed using the Distributes Office Local Manager (DOLM). The initial installation and configuration of the i120 and associated media modules are presumed to have been previously completed and are not discussed within these Application Notes.

3.1. SIP Trunk Configuration

3.1.1 Define SIP Trunk Group

Configure the **SIP Trunk Group** at DOLM Public Networking>Trunk Groups.
Using the button "Add new".

Figure 4: Add Trunk Group Form

- Set the **Trunk Type** field to *sip*.
- The **Trunk Group Number** field is set default to 31 and is not changeable.
- Choose a descriptive **Native Name**
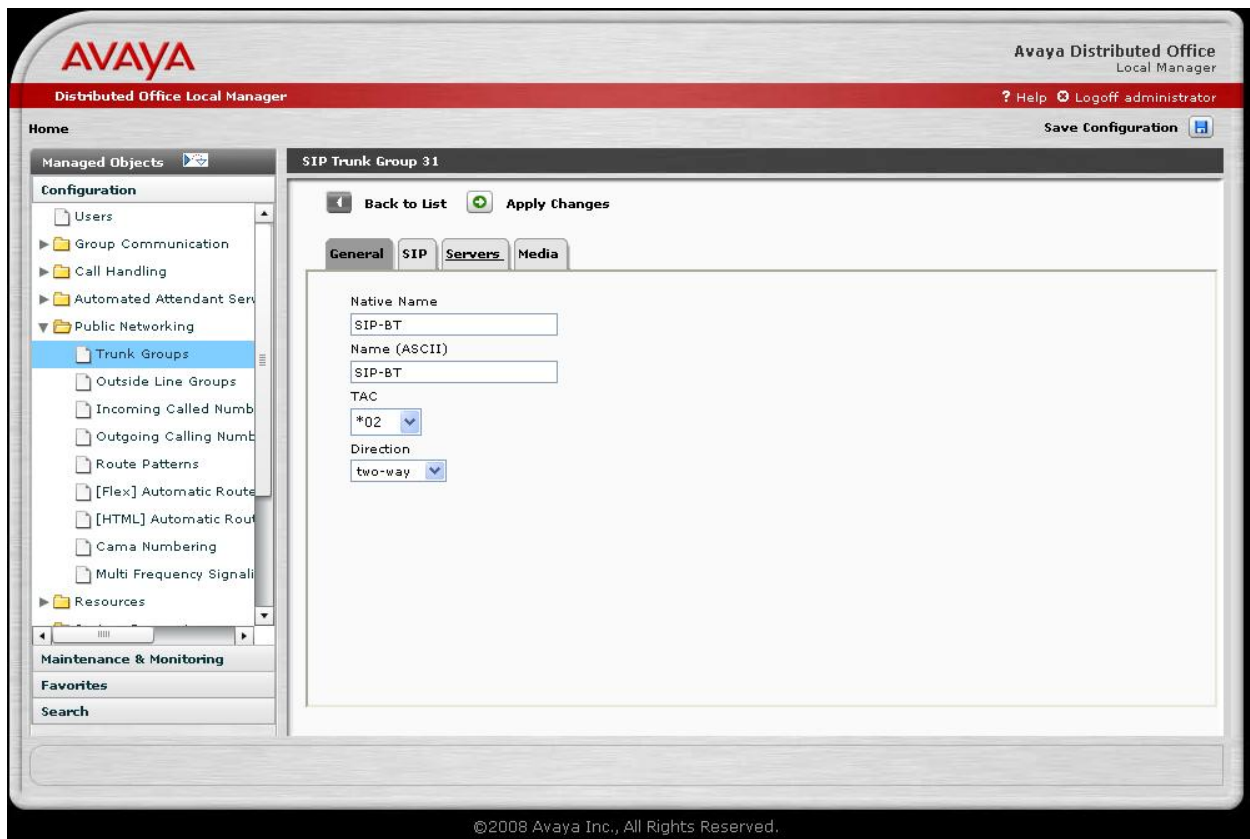- The **Name (ASCII)** field is set automatically.

Press the button "Continue".

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8 of 29
BT-Germany-SIP

Figure 5: SIP Trunk Group 31 General Form

- The **Native Name** field is set to the name of the trunk group.
- The **Name (ASCII)** field is set automatically.
- Specify an available trunk access code **TAC**
- Choose the **Direction** "two-way" for incoming and outgoing calls

Press the button "Apply Changes" to change the settings.

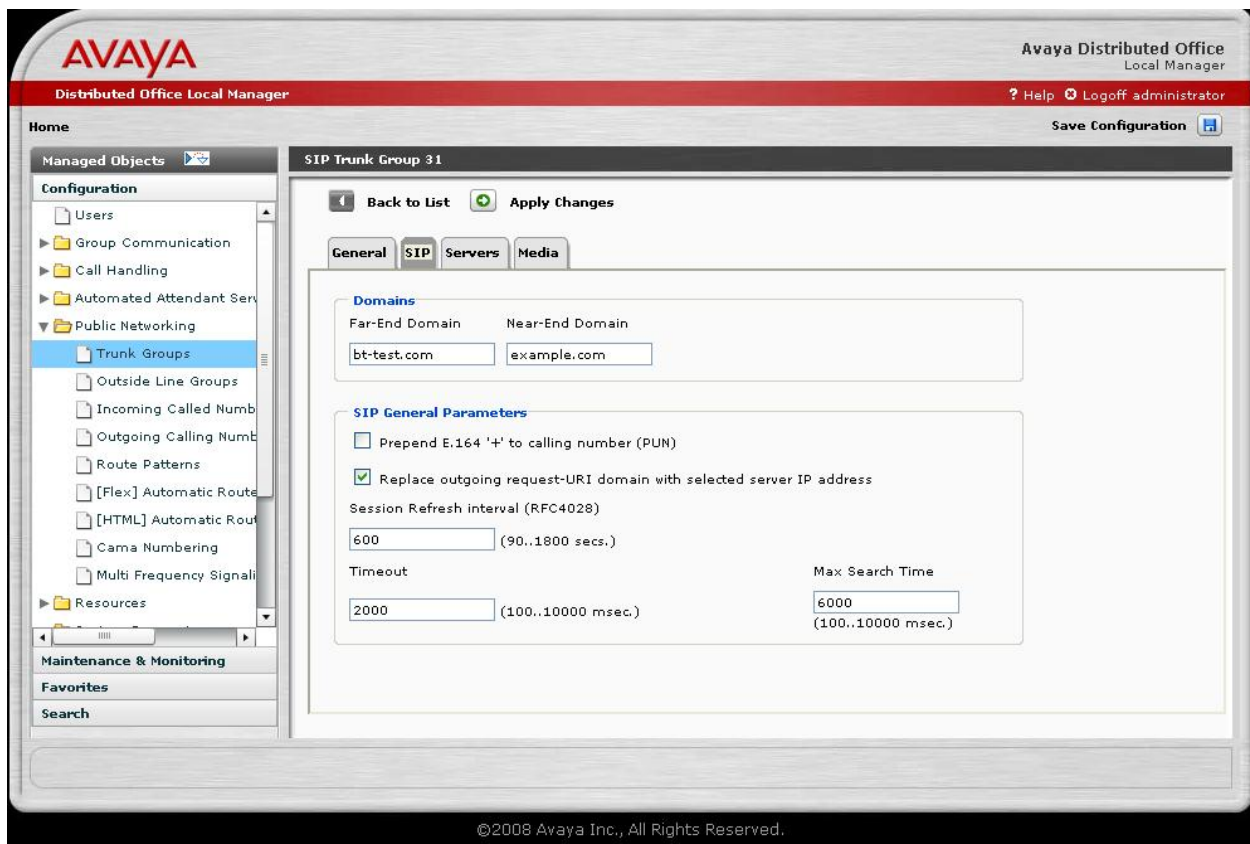Press the "SIP" tab for administering the SIP parameters

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9 of 29
BT-Germany-SIP

Figure 6: SIP Trunk Group 31 SIP Form

- Set the **Far-End Domain** field to the Domain Name got from Service Provider
- Set the **Near-End Domain** field to the customer Domain name
- Mark the field **Replace outgoing request-URI with selected server IP address**

Press the button "Apply Changes" to change the settings.

Press the "Servers" tab for administering the SBC parameters

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10 of 29
BT-Germany-SIP
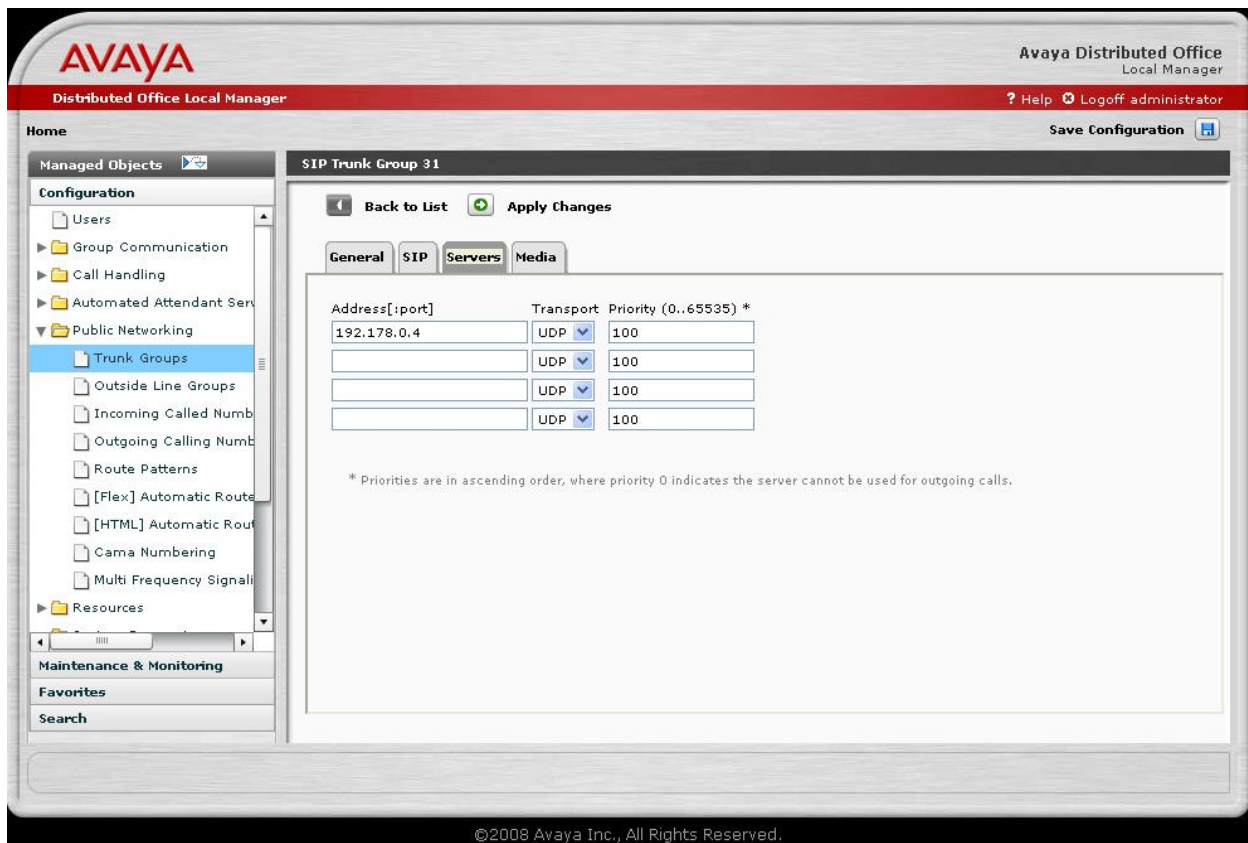
Figure 7: SIP Trunk Group 31 Servers Form

▪ Set the **Adddress [:port]** field to the IP address of the Session Border Controller (SBC) of BT
  If a backup SBC would be available, the IP address of the backup SBC must be added

Press the button "Apply Changes" for changing the settings.

Press the "Media" tab to specify the parameters of the Mediastream

Figure 8: SIP Trunk Group 31 Media Form

- Set the **Telephone Events RTP Payload Type (RFC2833)** field to 101
- Set the **Max. Concurrent Calls** field to 1-30 (in consultation with the Service Provider)
- **Direct Media** ("shuffling") is not activated, because shuffling doesn't work in all cases
- Set **Mode** to t.38-standard (FAX T38), if required
- Set **Redundancy** to 0-3 (in consultation with the Service Provider)
- Choose the **Codecs** G.711A and G.729ab, which are supported by BT

Press the button "Apply Changes" for changing the settings.

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12 of 29
BT-Germany-SIP

3.1.2 Configure Call Routing

3.1.2.1 Outbound Calls

In these Application Notes, the Automatic Route Selection feature is used to route calls via the SIP trunk to the British Telecom SIP trunking service, which in turn completes the calls to the ISDN network destination.

Use the **Dial Plan** and **Feature Access Code** form to specify **0** as the access code for external dialing (ARS ACCESS CODE).

Figure 9: Dial Plan Form

- ▪ Set **First Digit "0"** Length to 1 and Calltype to dac

Press the button "Apply Changes" to change the settings.

Figure 10: Feature Access Code Form

▪ Set **Feature Access Code "Automatic Route Selection"** Dial Code to 0 for accessing the SIP trunk

Press the button "Apply Changes" for changing the settings.

Now use the **Route Patterns** and **Automatic Route Selection** form to configure the route pattern selection rule based upon the number dialed after the digit "0" for accessing the SIP trunk. Note that further administration of ARS is beyond the scope of these Application Notes but is discussed in References [1] and [2].

Configure the **Route Pattern** at DOLM Public Networking > Route Patterns.

This Route Pattern is for national, international and Special Service numbers

Use the button "Add new" for creating national/international RP

Figure 11: Route Pattern Form for national, international and Special Service numbers

- Set the **Pattern Number** field to next free pattern number
- Choose a descriptive **Pattern Name**
- The **Trunk Group** field is set to SIP trunk group 31 (predefined for SIP trunk).
- Let the **Digits to Delete** and **Digits to insert** empty
  Dialed number must send unmodified to BT

Press the button "Apply Changes" for changing the settings.
Use the button "Add new" to create the RP for local numbers

Figure 12: Route Pattern Form for local numbers

- Set the **Pattern Number** field to next free pattern number
- Choose a descriptive **Pattern Name**
- The **Trunk Group** field is set to SIP trunk group 31(predefined for SIP trunk).
- Leave the **Digits to Delete** empty (no digits to delete)
- Insert in **Digits to Insert** the area code (example 069 for Frankfurt, Germany)

Press the button "Apply Changes" for changing the settings.

Configure the **Public Network Automatic Route Selection** at DOLM Public Networking > Automatic Route Selection for routing the outgoing calls.

Press the button "Add new".

Figure 13: Public Network Automatic Route Selection Form

- Set the **Dialed String** field to the dialing number (0 - 9)
- Set the **Minimum digits collected to match dialed string** field to necessary quantity
- Set the **Maximum digits collected to match dialed string** field to necessary quantity
- Set the **Call Type** field to necessary type
- Set the **Route Pattern** field to necessary route pattern

Figure 13 is an example that uses "0". In this case is "0" is dialed for national and international numbers in Germany.

Press the button "Apply Changes" for changing the settings.

Example of Public Network Automatic Route Selection for SIP trunking to BT in Germany

Figure 14: Public Network Automatic Route Selection Form

3.1.3 Configure Calling Party Number Information for outgoing calls

Configure the **Outgoing Calling Number Manipulation** form to send the correct calling party number to the British Telecom SIP trunking service for calls originating from Avaya Distributed Office. The calling party number (sent in the SIP "From" header) must match the value expected by the British Telecom SIP trunking service.

In these Application Notes, the extensions on Avaya Distributed Office use a 3 digit dialing plan with extensions between 200 and 299 that match the last three digits of the DID numbers assigned by the British Telecom SIP trunking service. In these Application Notes, the British Telecom SIP trunking requirement is that the calling party number be 3 digits long between 200 and 299. Note this requirement may vary with other numbering plans provided by British Telecom.

Configure the **Outgoing Calling Number Manipulation** at DOLM Public Networking > Outgoing Calling Number Manipulation.

Use the button "Add new".

Figure 15: Outgoing Calling Number Manipulation Form

- Set the **Extension Leading Digits** field to the leading extension number
- Set the **Trunk Group** field to the trunk group of the SIP trunk
- Enter the digits into the **Digits to Prefix to Form Calling Party Number (CPN Prefix)** field, that should be prepended to the extensions number
- Set the **Total CPN Length** field to CPN Prefix length + extension length, in this example the CPN Prefix length is 9 and the extension length is 3

Press the button "Apply Changes" for changing the settings.

3.1.3.1 Incoming Calls

This step configures the mapping of incoming DID numbers to the proper extensions.

In these Application Notes, the British Telecom SIP trunking service sends exactly 12 digits in the incoming INVITE message. The incoming digits received do not match the intended extension. Therefore the incoming call handling treatment is used to modify the number in order to reach the desired extension. The example below illustrates the technique to assign the incoming DID numbers beginning with 069999997 to the corresponding extensions between 211 and 269.

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19 of 29
BT-Germany-SIP

Use the **Incoming Called Number Manipulation** form to administer this assignment.

Press the button "Add new".

Figure 16: Incoming Called Number Manipulation Form

- Set the **Trunk Group** field to the trunk group of the SIP trunk
- Set the **Called Number** field to the CPN prefix
- Set the **Called Number Length** field to CPN Prefix length + extension length, in this example the CPN Prefix length is 9 and the extension length is 3
- Set the **# of Digits to Delete** field to CPN Prefix length to cut off the head number of the called party number, in this example the CPN Prefix length is 9

Press the button "Apply Changes" for changing the settings.

3.1.4 Save Avaya Distributed Office Changes

Press the button "Save Configuration" to save the changes.

4. British Telecom SIP trunking service Configuration

In order to use the British Telecom SIP trunking service, a customer must order service from British Telecom SIP trunking using their sales processes. The process can be started by contacting a British Telecom SIP trunking sales representative.

British Telecom SIP trunking provided the information contained in **Table 2** to complete the configuration in these Application Notes. This information was necessary to complete the Avaya Communication Manager and Avaya SIP Enablement Services administration discussed in the previous sections.

Required Information	Values Used In These Application Notes
British Telecom SIP trunking service SIP Application Server	*192.178.0.4*
Assigned DID Numbers	*069 999997 200 to 299*
Codecs Supported	*G.729, G.711A, T38*

Table 2: British Telecom SIP trunking service Configuration Information

5. Interoperability Compliance Testing

This section describes the interoperability compliance testing used to verify SIP trunking interoperability between the British Telecom SIP trunking service and an Avaya SIP telephony solution using SIP trunking. This section covers the general test approach and the test results.

5.1. General Test Approach

A simulated enterprise site consisting of an Avaya SIP telephony solution supporting SIP trunking was connected to a laboratory version of the British Telecom SIP trunking service via simulated managed access facilities. The enterprise site was configured as if using the generally available service provided by British Telecom.

The following features and functionality were covered during the SIP trunking interoperability compliance test. All calls involved various Avaya telephones, the British Telecom SIP trunking service and various ISDN telephones.

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21 of 29
BT-Germany-SIP

- Outgoing calls to ISDN telephones
- Incoming calls to British Telecom SIP trunking provided DID numbers from PSTN telephones
- Calls using Avaya 4600 Series IP Telephones with the H.323 firmware configuration
- Calls using Avaya 9600 Series IP Telephones with the SIP firmware configuration
- G.729AB and G.711A codecs for voice calls
- G.711A and T38 codecs for fax calling
- DTMF tone transmission using RFC 2833 with successful Voice Mail navigation
- Telephone features such as hold, transfer and conference
- Direct IP-to-IP media (also known as "shuffling") with SIP telephones

5.2. Test Results

Interoperability testing of the sample configuration was completed with partial unsuccessful results. Some observations are described in **Table 3**.

Item	Issue Observed	Discussion / Workaround
Incoming call from a PBX phone to a H323 or SIP phone through the BT Germany network.	The Calling and connected party is only the pilot number: 069 7505 0. Instead of the pbx extension number (e.g. 666) 0 is displayed.	BT Germany can't send the pbx extension number
Outgoing call from a H323 or SIP phone to a PSTN phone	Calling number is displayed Connected number not displayed, Message "SIPD (unavailable)" on display	PAI is not supported by BT The information "SIPD" in the Contact header can't be changed by BT to the connected number. Connected number is sent in the FROM header.
Outgoing call to an unassigned number	BT sends SIP message "404 not found" instead of "503 service unavailable". Busy tone is received instead of the announcement "Kein Anschluss unter dieser Nummer"	No workaround is available at this time.
Outbound Calling Number Restriction	Call not established BT sends "403 no permission for this kind of call". The calling party hears busy tone	BT does not support the kind of anonymous "FROM" Header that is delivered by Avaya (BT needed P-Asserted-Identity and Privacy)

Item	Issue Observed	Discussion / Workaround
FAX SIP/T38 is not working in all cases	FAX SIP/T38 is not working with a baud rate greater than 9600 for incoming calls.	If the baud rate is larger than 9600, the T38 packet size, sent from BT, is up to 500 bytes. The Media gateway supports a T38 packet size of 350 bytes. Workaround: Set the baud rate from sending Fax machine to 9600 or change the voip parameter with DOLM, System Parameters>VoIP Parameters, ID 53 value to 1 and ID 51 to 9600.
Shuffling scenarios only worked for SIP phones by outgoing calls	The RTP data were not shuffled for H323 phones by incoming or outgoing calls and for SIP phones only by incoming calls. After hold - resume no talk path	Test must repeat after Software update by BT

Table 3: Summary of problems identified during Interoperability Testing

6. Verification Steps

This section provides the verification steps that may be performed to verify operation of the Avaya SIP telephony solution with the British Telecom SIP trunking service.

- Verify outbound voice calls can be placed to ISDN phones and that the calls remain connected for 3 minutes.
- Verify incoming voice calls can be received from ISDN phones and that the calls remain connected for 3 minutes.
- Verify that calls are properly disconnected when either end disconnects first.
- Verify outbound and inbound fax calls can be placed to and from ISDN fax machines.

7. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya Distributed Office to the British Telecom SIP trunking service using SIP trunking. The British Telecom SIP trunking service is a Voice over IP solution for customers ranging from small businesses to large enterprises. SIP trunking uses the Session Initiation Protocol to connect private company networks to the public telephone network via converged IP access. It provides businesses a flexible, cost-saving alternative to current TDM-based telephony trunk lines.

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23 of 29
BT-Germany-SIP

8. References

This section references documentation relevant to these Application Notes.

The Avaya product documentation is available at http://support.avaya.com.

[1] *Design and Implementation Guide for Avaya Distributed Office*, May 2008, Release 1.2, Issue 2, Document Number 03-602023.
[2] *Feature Description for Avaya Distributed Office,* May 2008, Issue 2, Document Number 03-602027
[3] *4600 Series IP Telephone LAN Administrator Guide,* February 2007, Issue 6, Document Number 555-233-507
[4] Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide Release 2.0, December 2007, Issue 2, Document Number 16-601944
[5] Avaya one-X Deskphone Value Edition 1600 Series IP Telephones Administrator Guide Release 1.0, June 2007, Issue 1, Document Number 16-601443

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[6] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard
[7] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard
[8] RFC 2782 - *A DNS RR for Specifying the Location of Services (DNS SRV)*, February 2000, Proposed Standard

APPENDIX A: Sample SIP INVITE Messages

This section displays the format of the SIP INVITE messages sent by British Telecom SIP trunking and the Avaya SIP network at the enterprise site. These INVITE messages may be used for comparison and troubleshooting purposes. Differences in these messages may indicate that different configuration options were selected.

Sample SIP INVITE Message from British Telecom SIP trunking to Avaya Distributed Office:

```
Session Initiation Protocol
    Request-Line: INVITE sip:069999997261@192.168.190.50 SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 192.178.0.4:5060;branch=z9hG4bKf7eucu202gc0ob89t2s1.1
            Transport: UDP
            Sent-by Address: 192.178.0.4
            Sent-by port: 5060
            Branch: z9hG4bKf7eucu202gc0ob89t2s1.1
        From: "06975050" <sip:06975050@62.180.193.17>;tag=6616016d
            SIP Display info: "06975050"
            SIP from address: sip:06975050@62.180.193.17
            SIP tag: 6616016d
        To: <sip:69999997261@62.180.193.26>
            SIP to address: sip:69999997261@62.180.193.26
        Call-ID: 0347f1d350000000012d@csgmc5
        CSeq: 42 INVITE
            Sequence Number: 42
            Method: INVITE
        Date: Tue, 01 Apr 2008 06:16:48 GMT
        User-Agent: CVX_SS7_Gateway/CSG.6.0.b192o.E
        Accept: application/sdp
        Expires: 90
        Max-Forwards: 29
        Contact: "SIPD" <sip:sipdintavaya2004-di4t40p95qsof@192.178.0.4:5060;transport=udp>
            Contact Binding: "SIPD" <sip:sipdintavaya2004-
di4t40p95qsof@192.178.0.4:5060;transport=udp>
                URI: "SIPD" <sip:sipdintavaya2004-di4t40p95qsof@192.178.0.4:5060;transport=udp>
                    SIP Display info: "SIPD"
                    SIP contact address: sip:sipdintavaya2004-di4t40p95qsof@192.178.0.4:5060
        Content-Type: application/sdp
        Content-Length: 387
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): SIPD 1 1 IN IP4 192.178.0.4
                Owner Username: SIPD
                Session ID: 1
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 192.178.0.4
            Session Name (s): SIPD
            Connection Information (c): IN IP4 192.178.0.4
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 192.178.0.4
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 10160 RTP/AVP 8 0 4 2 18 125 96 101
                Media Type: audio
                Media Port: 10160
```

```
        Media Proto: RTP/AVP
        Media Format: ITU-T G.711 PCMA
        Media Format: ITU-T G.711 PCMU
        Media Format: ITU-T G.723
        Media Format: ITU-T G.721
        Media Format: ITU-T G.729
        Media Format: DynamicRTP-Type-125
        Media Format: DynamicRTP-Type-96
        Media Format: DynamicRTP-Type-101
    Media Attribute (a): rtpmap:8 PCMA/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 8
        MIME Type: PCMA
    Media Attribute (a): rtpmap:0 PCMU/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 0
        MIME Type: PCMU
    Media Attribute (a): rtpmap:4 G723/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 4
        MIME Type: G723
    Media Attribute (a): rtpmap:2 G726-32/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 2
        MIME Type: G726-32
    Media Attribute (a): rtpmap:18 G729/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 18
        MIME Type: G729
    Media Attribute (a): rtpmap:125 octet-stream/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 125
        MIME Type: octet-stream
    Media Attribute (a): rtpmap:96 red/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 96
        MIME Type: red
    Media Attribute (a): fmtp:96 101
        Media Attribute Fieldname: fmtp
        Media Format: 96 [red]
        Media format specific parameters: 101
    Media Attribute (a): rtpmap:101 telephone-event/8000
        Media Attribute Fieldname: rtpmap
        Media Format: 101
        MIME Type: telephone-event
    Media Attribute (a): fmtp:101 32,33,34,35,36,37,38,39,40
        Media Attribute Fieldname: fmtp
        Media Format: 101 [telephone-event]
        Media format specific parameters: 32,33,34,35,36,37,38,39,40
    Media Attribute (a): ptime:20
        Media Attribute Fieldname: ptime
        Media Attribute Value: 20
```

Sample SIP INVITE Message from Avaya Distributed Office to British Telecom:

```
Session Initiation Protocol
    Request-Line: INVITE sip:06975056085@192.178.0.4;transport=udp SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Call-ID: 01652833514dd17e947f0e35300
        CSeq: 1 INVITE
            Sequence Number: 1
            Method: INVITE
        From: "DO SIP 21" <sip:069999997261@example.com:6002>;tag=01652833514dd17d947f0e35300
            SIP Display info: "DO SIP 21"
            SIP from address: sip:069999997261@example.com:6002
            SIP tag: 01652833514dd17d947f0e35300
        Record-Route: <sip:192.168.190.50:5060;lr>,<sip:192.168.190.50:6002;lr;transport=tls>
        To: "06975056085" <sip:06975056085@bt-test.com>
            SIP Display info: "06975056085"
            SIP to address: sip:06975056085@bt-test.com
        Via: SIP/2.0/UDP 192.168.190.50:5060;branch=z9hG4bK030303131313636363631578.0,SIP/2.0/TLS
192.168.190.50:6002;psrrposn=2;received=192.168.190.50;branch=z9hG4bK01652833514dd17f947f0e35300,
SIP/2.0/TCP 192.168.190.118;psrrposn=1;received=192.16
            Transport: UDP
            Sent-by Address: 192.168.190.50
            Sent-by port: 5060
            Branch: z9hG4bK030303131313636363631578.0,SIP/2.0/TLS
        Content-Length: 159
        Content-Type: application/sdp
        Contact: "DO SIP 21" <sip:069999997261@192.168.190.50:6002;transport=tls>
            Contact Binding: "DO SIP 21" <sip:069999997261@192.168.190.50:6002;transport=tls>
                URI: "DO SIP 21" <sip:069999997261@192.168.190.50:6002;transport=tls>
                    SIP Display info: "DO SIP 21"
                    SIP contact address: sip:069999997261@192.168.190.50:6002
        Max-Forwards: 67
        User-Agent: Avaya CM/R013w.01.2.017.0
        Allow: INVITE,CANCEL,BYE,ACK,PRACK,SUBSCRIBE,NOTIFY,REFER,OPTIONS
        History-Info: <sip:06975056085@bt-test.com>;index=1
        History-Info: "06975056085" <sip:06975056085@bt-test.com>;index=1.1
        Supported: 100rel,timer,replaces,join,histinfo
        Min-SE: 3600
        Session-Expires: 3600;refresher=uac
        P-Asserted-Identity: "DO SIP 21" <sip:069999997261@example.com:6002>
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): - 1 1 IN IP4 192.168.190.50
                Owner Username: -
                Session ID: 1
                Session Version: 1
                Owner Network Type: IN
                Owner Address Type: IP4
                Owner Address: 192.168.190.50
            Session Name (s): -
            Connection Information (c): IN IP4 192.168.190.111
                Connection Network Type: IN
                Connection Address Type: IP4
                Connection Address: 192.168.190.111
            Time Description, active time (t): 0 0
                Session Start Time: 0
                Session Stop Time: 0
            Media Description, name and address (m): audio 2746 RTP/AVP 8 127
                Media Type: audio
                Media Port: 2746
                Media Proto: RTP/AVP
                Media Format: ITU-T G.711 PCMA
                Media Format: DynamicRTP-Type-127
            Media Attribute (a): rtpmap:8 PCMA/8000
```

```
            Media Attribute Fieldname: rtpmap
            Media Format: 8
            MIME Type: PCMA
      Media Attribute (a): rtpmap:127 telephone-event/8000
            Media Attribute Fieldname: rtpmap
            Media Format: 127
            MIME Type: telephone-event
```