



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring SIP Trunking between CenturyLink SIP Trunk (Legacy Qwest) Service and Avaya IP Office R8.0 (16) – Issue 1.0**

## **Abstract**

These Application Notes describe the steps required to configure Session Initiation Protocol (SIP) trunking between a CenturyLink SIP Trunk (Legacy Qwest) Service and an Avaya IP Office R8.0 (16) telephony solution.

Information in these Application Notes has been obtained through DevConnect 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 required to configure Session Initiation Protocol (SIP) trunking between CenturyLink and an Avaya IP Office telephony solution. The Avaya solution consists of Avaya IP Office R8.0 (16) with Avaya H.323, SIP, Digital and analog endpoints. These Application Notes correspond to the CenturyLink SIP Trunk Service offered using a Sonus Network Border Switch (NBS) in the network, otherwise known as Legacy Qwest SIP Trunk Service.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 is the primary specification governing this protocol. SIP manages the establishment and termination of connections, and the transfer of related information such as the desired codec, calling party identity, etc. Within these Application Notes, SIP is used as the signaling protocol between Avaya IP Office and the CenturyLink SIP Trunk Service solution.

Customers using this Avaya IP Office telephony solution with the CenturyLink SIP Trunk Service are able to place and receive PSTN calls via a dedicated broadband Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as ISDN-PRI. The text and coverage diagram (**Figure 1**) summarizes the CenturyLink SIP Trunk Service at the time of writing these Application Notes. Please consult CenturyLink for the most current description of capabilities.

## 1.1. CenturyLink SIP Trunk

CenturyLink SIP Trunk Service enables the origination and termination of local and long-distance calls, as well as domestic, international and toll-free traffic across a single broadband connection. It is designed to work in conjunction with CenturyLink's Networking service, which includes a secure, managed, and scalable suite of wide area network (WAN) services.

# 2. General Test Approach and Test Results

This section describes the interoperability compliance testing used to verify SIP trunk interoperability between the CenturyLink SIP Trunk Service and an Avaya IP office Telephony Solution.

A simulated enterprise site using an Avaya IP Office telephony solution was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the commercially available SIP Trunk Service provided by CenturyLink.

The compliance test included the following:

- Incoming calls to the enterprise site from the PSTN were routed to the DID numbers assigned by CenturyLink. Incoming PSTN calls were made to Avaya H.323, SIP, digital and analog telephones at the enterprise.
- Outgoing calls from the enterprise site were completed via CenturyLink to PSTN destinations. Outgoing calls from the enterprise to the PSTN were made from Avaya H.323, SIP, digital and analog telephones.

- Various call types were tested including: local, long distance, international, inbound/outbound toll-free, emergency, T.38 fax, operator and directory assistance.
- Calls were placed using both the G.729A and G.711MU codecs.
- DTMF transmission using RFC 2833 for inbound and outbound calls.
- User features such as hold and resume, forwarding, transfer and conference.
- Caller ID Presentation and Caller ID Restriction.
- Mobility Features: Mobile Twinning to a mobile phone was tested and verified.

## **2.1. Interoperability Compliance Testing**

A simulated enterprise site consisting of an Avaya IP Office telephony solution supporting SIP trunking was connected to the public Internet using a dedicated broadband connection. The enterprise site was configured to use the generally available SIP trunking solution provided by CenturyLink. This allowed the enterprise site to use SIP trunking for calls to and from the PSTN.

## **2.2. Test Results**

The test objectives were verified. Interoperability testing of the sample configuration was completed with successful results for the CenturyLink SIP Trunk Service, except as noted below.

Items not supported or not tested included the following:

- The use of the SIP REFER method was not fully supported at the time of this compliance test. While it was tested, there were a few issues that were actively being worked at the time of writing these Application Notes.

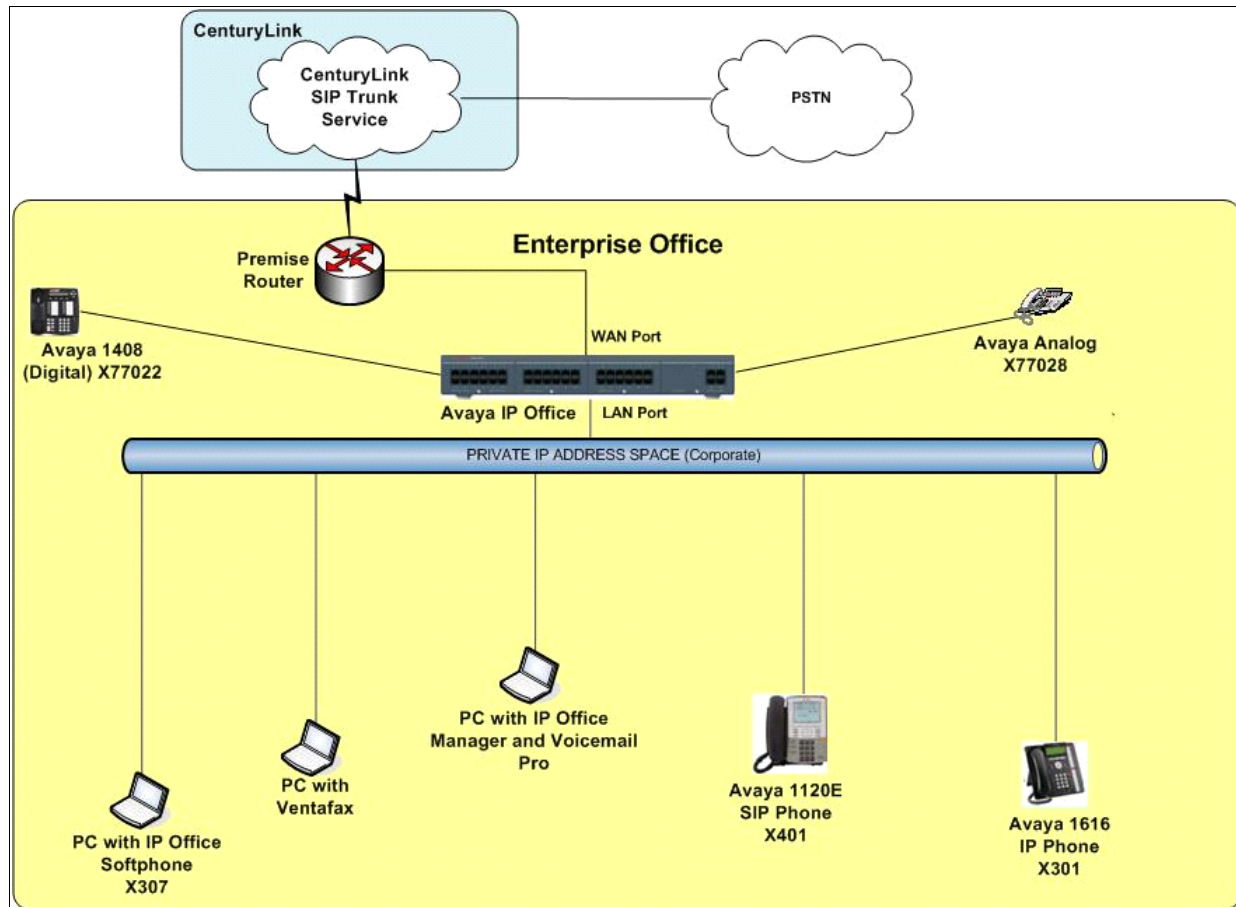
## **2.3. Support**

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on the CenturyLink SIP Trunk Service, contact CenturyLink at <http://www.CenturyLink.com/>.

### 3. Reference Configuration

**Figure 1** illustrates a sample Avaya IP Office telephony solution connected to CenturyLink SIP Trunk that was utilized for compliance testing. Since public IP addresses were used during compliance testing, those IP address are not shown in the figure below and are masked (at least partially) throughout the document.



**Figure 1: Avaya IP Office telephony system connected to CenturyLink SIP Trunk Service**

## 4. Equipment and Software Validated

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

<b>Avaya IP Telephony Solution Components</b>	
<b>Equipment</b>	<b>Software/Firmware</b>
Avaya IP Office 500	Avaya IP Office 500v2 R8.0.(16)
Avaya IP Office Manager (Windows PC)	Avaya IP Office Manager 10.0.(16)
Avaya IP Office Voicemail Pro	Avaya IP Office Voicemail Pro 7.0(17)
Avaya 1120E IP Telephone (SIP)	SIP1120 Load Version 04.01.13.00
Avaya 1616-I IP Telephone (H.323)	ha1616ua1_3000.bin
Avaya 1408 Digital Telephone	-
Analog Telephone and Fax	-
<b>CenturyLink Solution Components</b>	
<b>Equipment</b>	<b>Software/Firmware</b>
CenturyLink Network Border Switch (NBS)	7.3.5R6

## 5. Configure Avaya IP Office

This section describes the steps required for configuring a SIP trunk on Avaya IP Office.

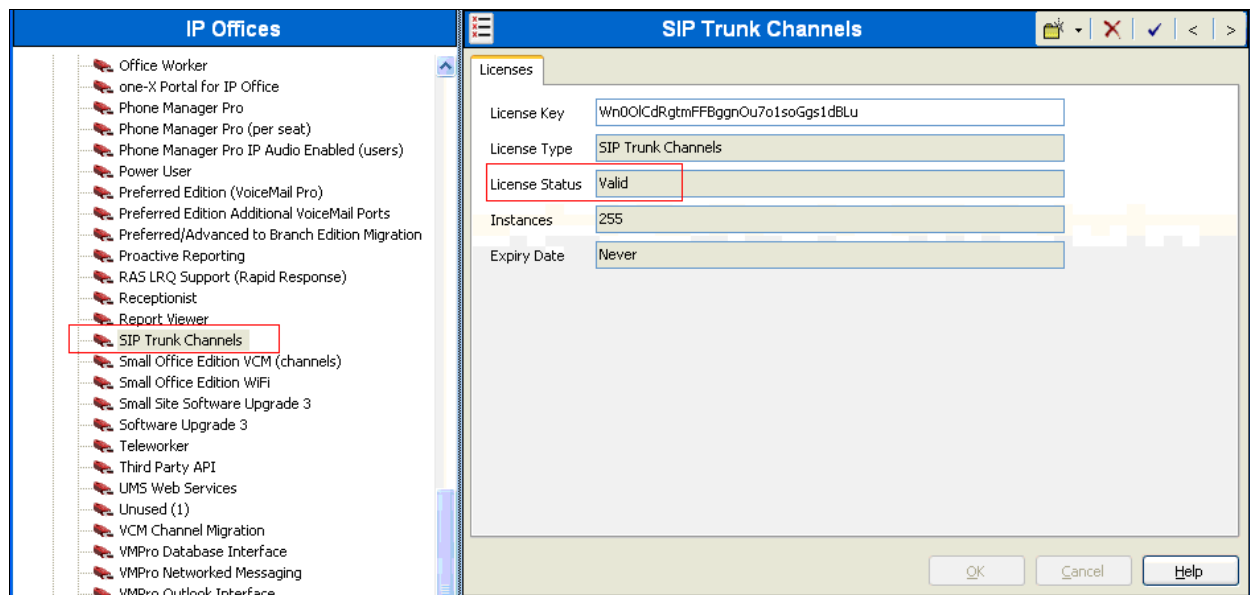
The procedures include the following areas:

- Verify SIP Trunk Channels License
- Configure WAN interface
- Configure SIP OPTIONS timer for “keep alive” function with CenturyLink
- Enable SIP Trunk
- Create the SIP line to CenturyLink
- Configure SIP URI parameters for the SIP Line
- Configure T38 Fax parameters for the SIP Line
- Configure a short code to route calls to CenturyLink
- Create an Incoming Call Route for the Inbound SIP calls
- Configure Users’ SIP names

## 5.1. Verify SIP Trunk Channels License

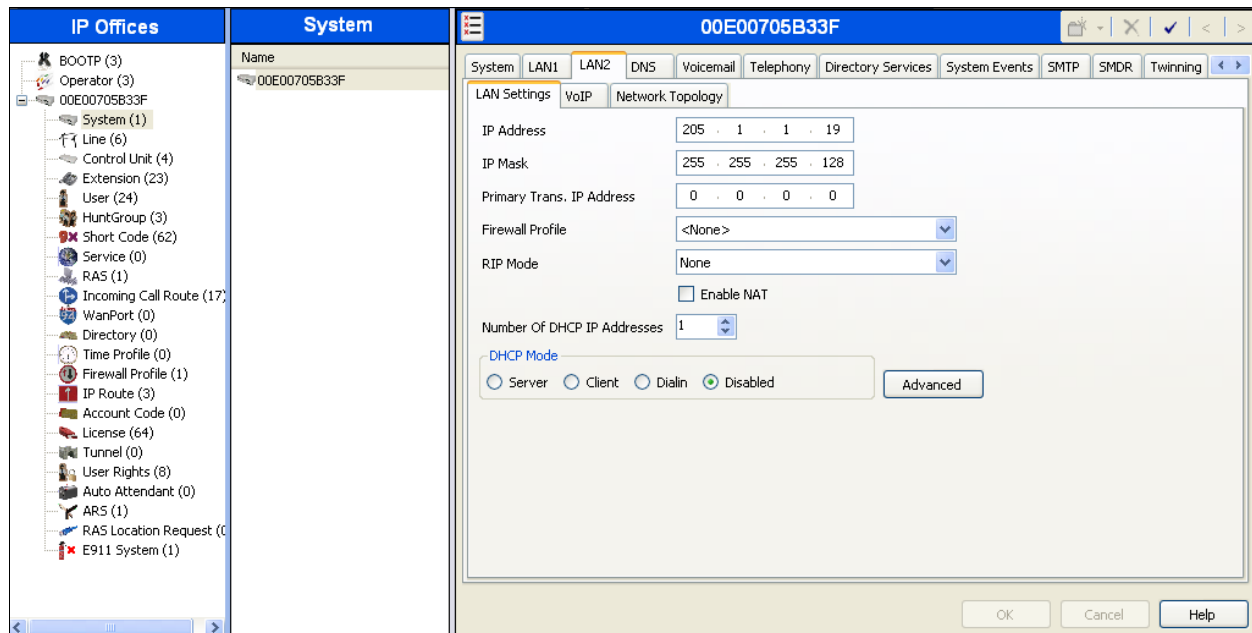
Avaya IP Office is configured via the IP Office Manager application. Log into the PC running the Avaya IP Office Manager application and select **Start → All Programs → IP Office → Manager** to launch the Manager application. Select the proper Avaya IP Office system if there is more than one Avaya IP Office system listed, and log in with the appropriate credentials. From the configuration tree in the left pane, select **License → SIP Trunk Channels** to display the SIP Trunk Channels screen in the right pane. Verify that the **License Status** field is set to “Valid” and that there are enough **Instances** available.

If a required feature is not enabled or there is insufficient capacity (**Instances**), contact an authorized Avaya sales representative to make the appropriate changes.



## 5.2. Configure WAN interface

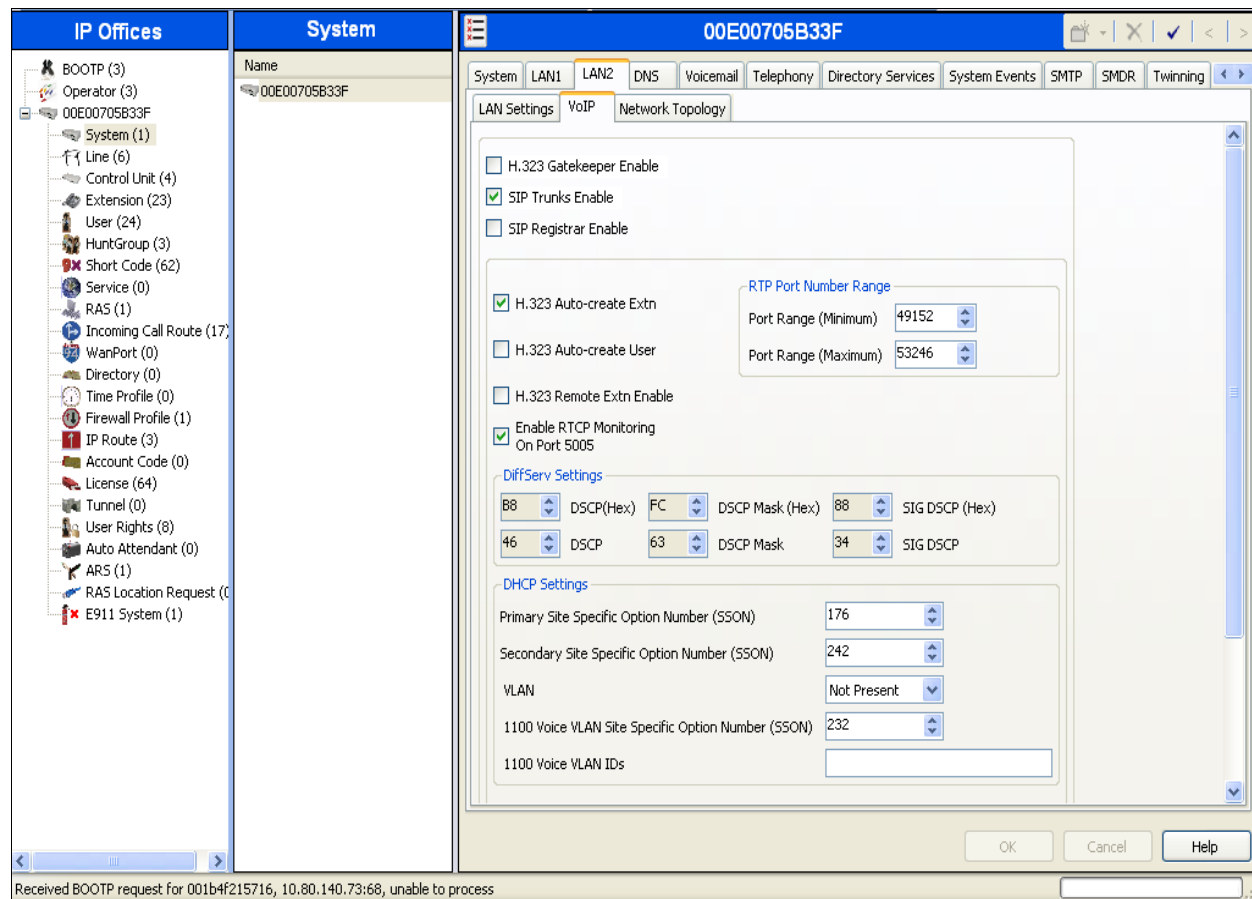
From the configuration tree in the left pane, select **System** to display the System screen in the right pane. Click the **LAN2** tab. Under the **LAN2** tab, select the **LAN Settings** sub-tab and provide an **IP Address** and **IP Mask**.



### 5.3. Enable SIP Trunk

Under the **LAN2** tab, select the **VoIP** sub-tab, and check the **SIP Trunks Enable** box. Click the **OK** button.

**Note:** During the initial configuration of Avaya IP Office, **LAN1** was configured as a private network (LAN) and **LAN2** was configured as a public network (WAN). Avaya IP Office can support SIP extensions on the **LAN1** and/or **LAN2** interfaces. In the compliance test, the **LAN2** interface was used for the SIP trunk termination but not for remote SIP endpoints. On the public network, the **SIP Registrar Enable** check box was left unchecked for **LAN2** but was enabled for **LAN1** to support SIP phones on the private network.





## 5.4. Configure SIP OPTIONS Timer for “keep alive” Function with CenturyLink

Under the **LAN2** tab, select the **Network Topology** sub-tab. Set the **Binding Refresh Time** to the desired interval that OPTIONS messages will be sent to CenturyLink. For this compliance test, it is set to “60”. For **Public IP Address**, enter the Avaya IP Office system public IP address (WAN), confirm that **Public Port** is set to “5060” and accept the default values for all other fields. Click the **OK** button.

The screenshot displays the Avaya IP Office configuration window for system 00E00705B33F. The left sidebar shows the 'System' tab with the system name. The main window has a top navigation bar with tabs: System, LAN1, LAN2 (selected), DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, Twinning, VCM, CCR, and Codecs. Below this, there are sub-tabs: LAN Settings, VoIP, and Network Topology (selected). The 'Network Topology Discovery' section contains the following fields and controls:

- STUN Server IP Address: 69 . 90 . 168 . 13
- STUN Port: 3478 (dropdown)
- Firewall/NAT Type: Open Internet (dropdown)
- Binding Refresh Time (seconds): 60 (dropdown)
- Public IP Address: 205 . 1 . 1 . 19
- Public Port: 5060 (dropdown)
- Buttons: Run STUN, Cancel
- Checkbox: ☐ Run STUN on startup

At the bottom right of the window are buttons for OK, Cancel, and Help.

## 5.5. Create SIP Line to CenturyLink

Select **Line** in the left pane. Right-click and select **New → SIP Line** to create a new **SIP Line**. During the compliance test SIP line “17” was configured. Click the **OK** button.

**Note: REFER Support** was set to “Never” due to issues being worked regarding the use of the SIP REFER method at the time of writing these Application Notes, as described in **Section 2.2**.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'SIP Line' tab selected. The left pane shows a tree view with 'Line' selected. The main area contains the following fields:

Field	Value	Field	Value
Line Number	17	In Service	<input checked="" type="checkbox"/>
ITSP Domain Name	205.1.1.19	Use Tel URI	<input type="checkbox"/>
Prefix		Check OOS	<input checked="" type="checkbox"/>
National Prefix	0	Call Routing Method	Request URI
Country Code		Originator number for forwarded and twinning calls	
International Prefix	00	Name Priority	System Default
Send Caller ID	Diversion Header		
Association Method	By Source IP address		
<input checked="" type="checkbox"/> REFER Support			
Incoming	Never		
Outgoing	Never		

Select the **Transport** sub-tab, and provide the following information:

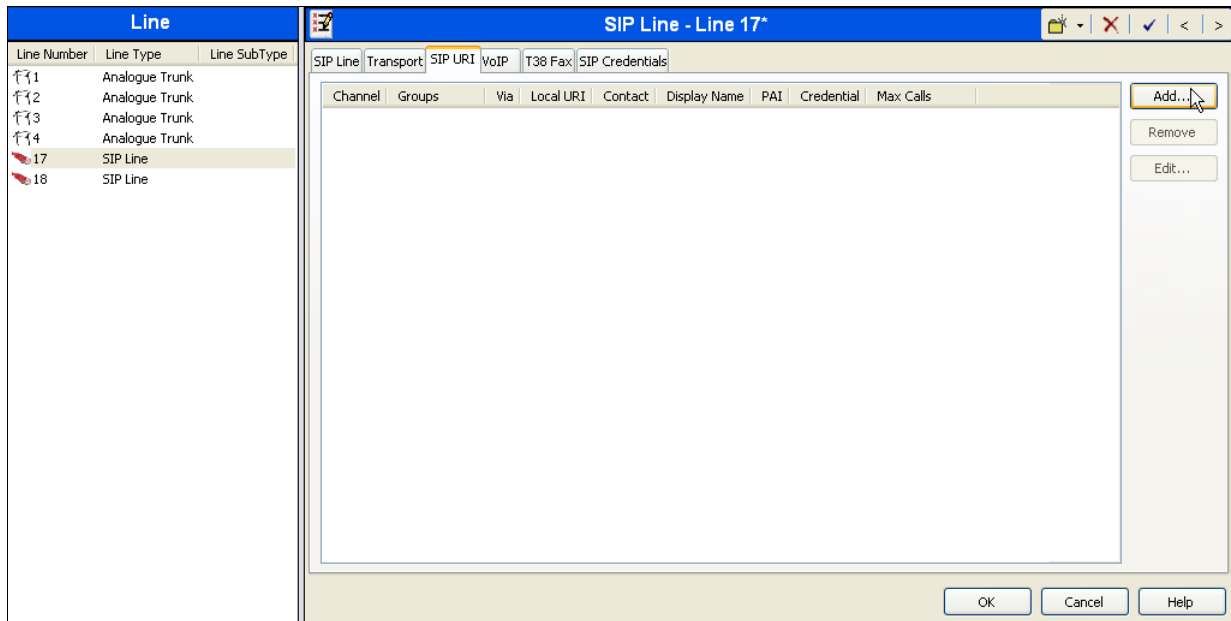
- **ITSP Proxy Address** – Enter the IP address of the CenturyLink SIP termination point.
- **Layer 4 Protocol** – Select **UDP**.
- **Use Network Topology Info** – Select **LAN2** (WAN interface).
- Click the **OK** button.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'Transport' sub-tab selected. The left pane shows a tree view with 'Line' selected. The main area contains the following fields:

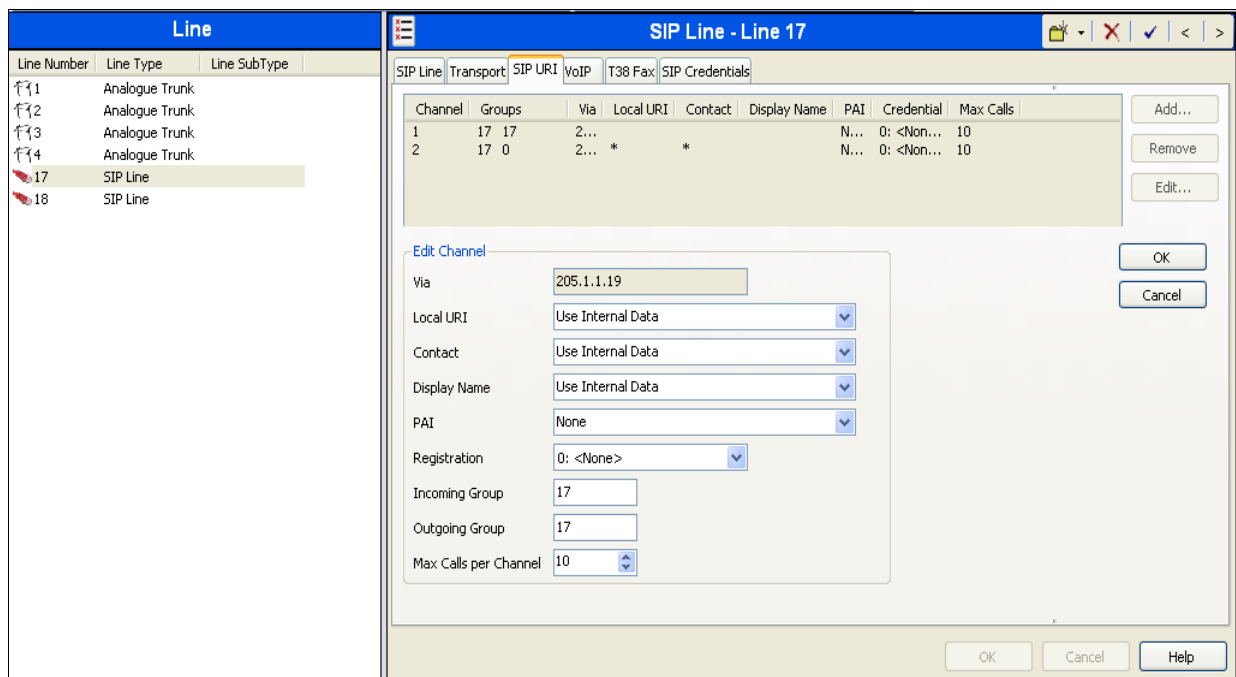
Field	Value	Field	Value
ITSP Proxy Address	67.1.1.8		
<b>Network Configuration</b>			
Layer 4 Protocol	UDP	Send Port	5060
Use Network Topology Info	LAN 2	Listen Port	5060
Explicit DNS Server(s)	0 . 0 . 0 . 0 . 0		
Calls Route via Registrar	<input checked="" type="checkbox"/>		
Separate Registrar			

## 5.6. Configure SIP URI Parameters for the SIP Line

Select the **SIP URI** tab to configure SIP URI parameters for the SIP Line. Click on the **Add** button.



Set the **Local URI**, **Contact** and **Display Name** fields to “Use Internal Data” so that outbound calls will use what is populated in the **SIP** tab for each user, for Caller ID. Be sure to set the **Incoming Group** and **Outgoing Group** fields to the appropriate SIP line number, in this case “17”.



Click **Add** again under the **SIP URI** tab to add another entry for inbound calls. In this case the **Local URI** and **Contact** fields simply contain the wildcard character “\*” to accept inbound calls from any number. There is no need to populate the **Outgoing Group** field as this is for inbound calls only. Click **OK** when finished.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'SIP URI' tab selected. On the left, a 'Line' list shows lines 1-4 as 'Analogue Trunk' and lines 17-18 as 'SIP Line'. The main area contains a table with two entries:

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	17 17	2...	*	*		N...	0: <Non...	10
2	17 0	2...	*	*		N...	0: <Non...	10

Below the table is an 'Edit Channel' section with the following fields:

- Via: 205.1.1.19
- Local URI: \*
- Contact: \*
- Display Name: Use Internal Data
- PAI: None
- Registration: 0: <None>
- Incoming Group: 17
- Outgoing Group: 0
- Max Calls per Channel: 10

Buttons for 'Add...', 'Remove', 'Edit...', 'OK', and 'Cancel' are visible.

Select the **VoIP** tab to configure the appropriate codecs and fax settings. In the **Codec Selection** field, change the dropdown menu from “System Default” to “Custom” (not shown) and add the appropriate codecs to the **Selected** box. Make sure that the **Re-Invite Supported** check box is checked. In the **Fax Transport Support** field, select “T38” using the drop down list. Default values may be used for all other fields.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'VoIP' tab selected. The 'Codec Selection' dropdown is set to 'Custom'. Below it are two boxes: 'Unused' containing 'G.711 ALAW 64K' and 'Selected' containing 'G.729(a) 8K CS-ACELP' and 'G.711 ULAW 64K'. Arrows between the boxes allow moving items. On the right, several checkboxes are shown:

- ☐ VoIP Silence Suppression
- ☒ Re-Invite Supported
- ☐ Use Offerer's Preferred Codec
- ☐ Codec Lockdown
- ☒ PRACK/100rel Supported

At the bottom, the 'Fax Transport Support' dropdown is set to 'T38', 'Call Initiation Timeout (s)' is 6, and 'DTMF Support' is RFC2833. Buttons for 'OK', 'Cancel', and 'Help' are at the bottom right.

## 5.7. Configure T38 Fax Parameters for the SIP Line

Select the **T38 Fax** tab to configure T38 Fax parameters for the SIP Line. Uncheck the **Use Default Values** check box and change the **T38 Fax Version** field from “3” to “0” (At the time of writing these Application Notes CenturyLink only supported Fax Version 0). Default Values may be used for all other fields. Click the **OK** button.

The screenshot shows the 'SIP Line - Line 17' configuration window with the 'T38 Fax' tab selected. The left pane lists lines 1 through 18, with line 17 highlighted as a SIP Line. The main pane contains the following settings:

- T38 Fax Version: 0
- Transport: UDPTL
- Redundancy: Low Speed (0), High Speed (0)
- TCF Method: Trans TCF
- Max Bit Rate (bps): 14400
- EFlag Start Timer (msecs): 2600
- EFlag Stop Timer (msecs): 2300
- Tx Network Timeout (secs): 150
- Use Default Values: ☐
- Scan Line Fix-up: ☒
- TFOP Enhancement: ☒
- Disable T30 ECM: ☐
- Disable EFlags For First DIS: ☐
- Disable T30 MR Compression: ☐
- NSF Override: ☐
- Country Code: 0
- Vendor Code: 0

Buttons at the bottom: OK, Cancel, Help.

A fax machine was connected to an analog extension. The following screen shows the analog extension that was configured as a fax line. Set the Equipment Classification field as **Standard Telephone** not **Fax Machine**. Click the **OK** button.

The screenshot shows the 'Analogue Extension: 7 77028' configuration window. The left pane shows a tree of IP Offices and Extensions, with extension 77028 selected. The main pane contains the following settings:

- Equipment Classification: ☒ Standard Telephone (Other options: Quiet Headset, Paging Speaker, Door Phone 1, Door Phone 2, IVR Port, FAX Machine, MOH Source)
- Flash Hook Pulse Width: ☒ Use System Defaults (Minimum Width: 20 ms, Maximum Width: 500 ms)
- Message Waiting Lamp Indication Type: None
- Hook Persistence: 100 ms

Buttons at the bottom: OK, Cancel, Help.

## 5.8. Configure a Short Code to Route Calls to CenturyLink

Select **Short Code** in the left panel. Right click and select **Add**. Enter [x]N;, where [x] is a valid number in the **Code** text box. The number “9” is used for [x] in the example below. This code requires the user to dial the digit “9” followed by the destination’s telephone number, symbolized by N, in order to route the call out the SIP Trunk.

**Note:** N can be any number other than a local Avaya IP Office extension. For example, a 10-digit number, a toll-free number, directory assistance, information service, etc.

Select “Dial” for the **Feature**. Enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group ID** field. Enter “N” representing the dialed number followed by “@<IP address of CenturyLink NBS>” for the **Telephone Number** field. The **Telephone Number** field is used to construct the SIP URI in the **To** header of the outgoing SIP INVITE message. Use default values for all other fields. Click the **OK** button.

The screenshot displays the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree shows 'Short Code (62)' selected. The main panel shows a list of short codes with columns for Code, Telephone Number, and Feature. The '9N;; Dial' short code is highlighted. A configuration window for this short code is open on the right, showing the following fields:

Field	Value
Code	9N;;
Feature	Dial
Telephone Number	N@67.1.1.8
Line Group ID	17
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

At the bottom of the window are 'OK', 'Cancel', and 'Help' buttons.

Right click and select **Add** again to add another short code. This time enter “1xxxxxxxxx” in the **Code** field, select “Dial” for the **Feature**, and enter the **Outgoing Group** number created in **Section 5.5** for the **Line Group ID** field. Enter “1N” representing the dialed number followed by “@<IP address of CenturyLink NBS>” for the **Telephone Number** field. This **Short Code** allows users to dial 11 digits without the preceding 9. 10 and 7 digit dialing can be setup this way as well, if necessary. Use default values for all other fields. Click the **OK** button.

Code	Telephone Number	Feature	Line Group...
*42	2	Relay On	0
*43	2	Relay Off	0
*44	2	Relay Pulse	0
*45*N#	N	Call Steal	0
*46		Call Steal	0
*47		Conference Add	0
*48		Voicemail Ringback On	0
*49		Voicemail Ringback Off	0
*50		Forward HuntGroup Calls On	0
*51		Forward HuntGroup Calls Off	0
*52		Clear Call	0
*53*N#	N	Call Pickup Members	0
*57*N#	N	Forward Busy Number	0
*70*N#	N	Dial Physical Extn by Number	0
*71*N#	N	Dial Physical Extn by Id	0
*9000*	"MAINTENANCE"	Relay On	0
*91N;	N".1"	Record Message	0
*92N;	N".2"	Record Message	0
*DSSN	";[0]151/ERR - "N	Display Msg	0
*SDN	";[0]151/ERR - "N	Display Msg	0
*SKN	";[0]151/ERR - "N	Display Msg	0
1xxxxxxxxx	1N"@67.1.1.8"	Dial	17
5xxxxxx	3035N"@67.1.1.8"	Dial	17
6N;	N"@67.1.1.8	Dial	50: Main
8N;	WN"@67.1.1.8"	Dial	17
9N;	N"@67.1.1.8"	Dial	17
FNE00	00	FNE Service	0

Short Code	
Code	1xxxxxxxxx
Feature	Dial
Telephone Number	1N"@67.1.1.8"
Line Group ID	17
Locale	United States (US English)
Force Account Code	<input type="checkbox"/>

(Note: The configuration of the Short Codes above is shown only as examples and is not meant to be prescriptive)

## 5.9. Create an Incoming Call Route for the Inbound SIP Calls

Select **Incoming Call Route** in the left pane. Right-click and select **New**. Enter the following:

- “Any Voice” for the **Bearer Capability** field.
- The **Incoming Group** number created for the URI in **Section 5.5** in the **Line Group ID** field.
- The 10-digit DID number provided by CenturyLink that is mapped back to a local Avaya IP Office extension, in the **Incoming Number** field.
- Use default values for all other fields.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'IP Offices' tree is visible, with 'Incoming Call Route (17)' selected. The main window displays the configuration for 'Incoming Call Route' with the title bar '17 3035557104'. The 'Standard' tab is active, showing the following fields:

Field	Value
Bearer Capability	Any Voice
Line Group ID	17
Incoming Number	3035557104
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

Next, navigate to the **Destinations** tab and select the desired local extension number from the drop down list. This is the extension that the DID will map to when receiving an inbound call for that DID. Click the **OK** button.

The screenshot shows the Avaya IP Office configuration interface. On the left, the 'Incoming Call Route' configuration window is visible, with the 'Destinations' tab active. The main window displays the configuration for 'Incoming Call Route' with the title bar '17 3035557104'. The 'Destinations' tab is active, showing a table with columns for Line Group ID, Incoming Number, and Destination. The 'Default Value' is set to '77028 Analog Phone' and the 'Fallback Extension' is set to '100 DID Hunt Group'.

Line Group ID	Incoming Number	Destination
0		DialIn
17	614555688	401 SIP 1120E
17	614555687	301 H323 1616
17	614555686	77022 Digital 1408
17	614555685	77028 Analog Phone
17	3035557104	77028 Analog Phone
17	3035557105	77022 Digital 1408
17	3035557106	301 H323 1616
17	3035557107	401 SIP 1120E
17	3035557108	VoiceMail
18	8555555224	100 DID Hunt Group
18	614555715	401 SIP 1120E
18	614555714	301 H323 1616
18	614555711	77022 Digital 1408
18	3035557133	301 H323 1616
18	3035557127	77022 Digital 1408
18	3035557109	77028 Analog Phone

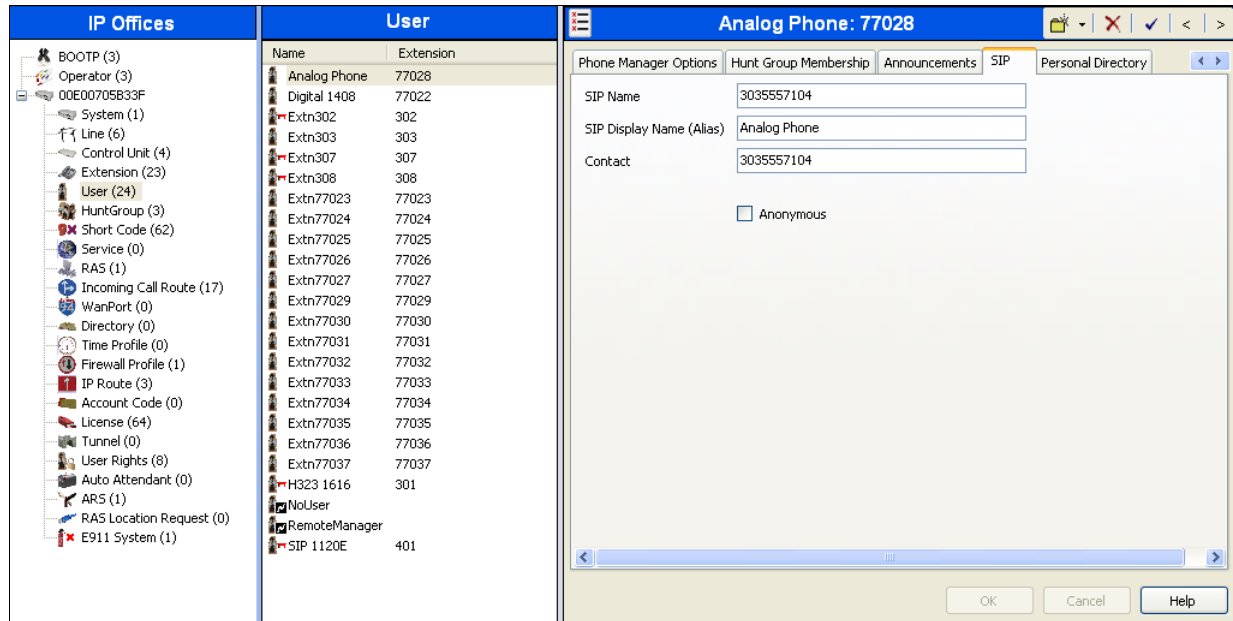


## 5.10. Configure SIP Parameters for Users

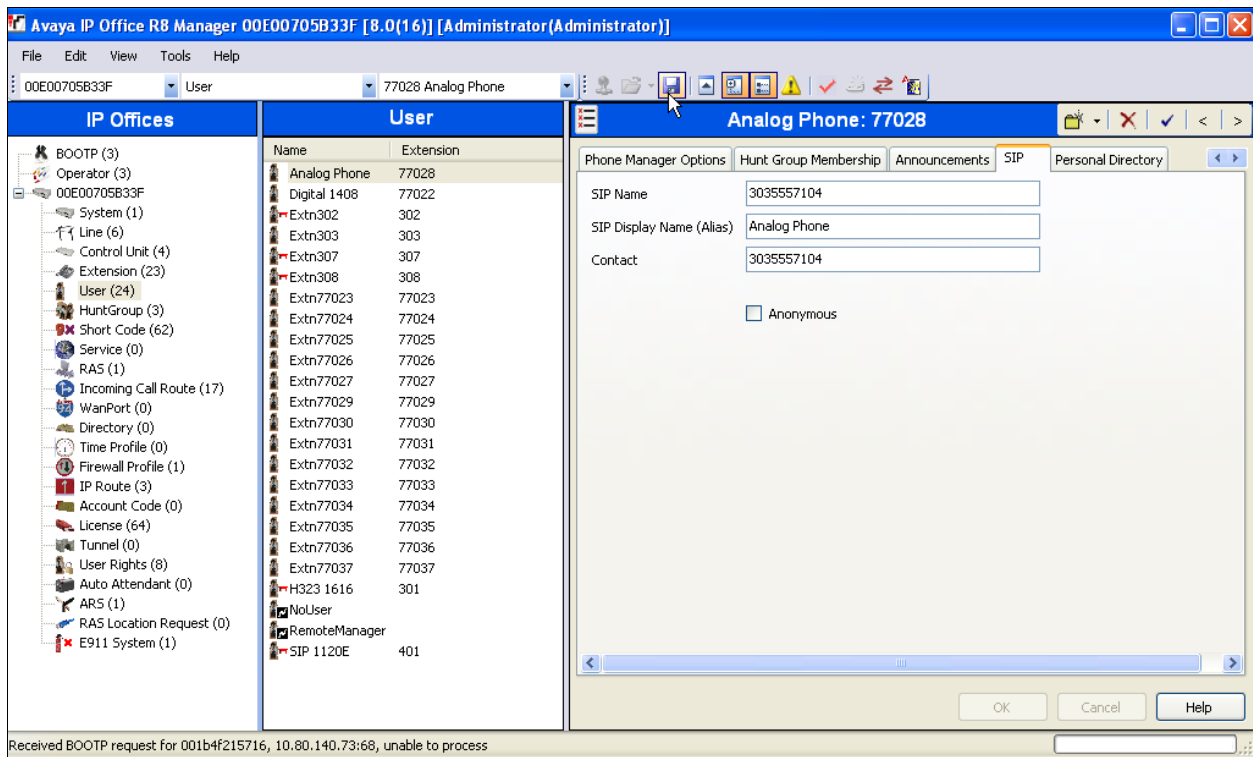
Select **User** in the left panel. Select the desired user by clicking on an entry in the middle panel. Select the **SIP** tab. Modify the **SIP Name** and **Contact** fields to the DID number provided by CenturyLink that is used for this particular extension. These settings instruct the system to use this DID number to construct the:

- user part of the SIP URI in the From header of an outgoing SIP INVITE message
- user part of the SIP URI in the Contact header of an outgoing SIP INVITE message

Modify the **SIP Display Name (Alias)** that will be used for the SIP Display info, to a descriptive name. Click the **OK** button.



After making changes or completing the configuration, click on the floppy disk icon to push the changes to the Avaya IP Office system and have them take effect.



*(Note: Changes will not take effect until this step is completed. This may cause a reboot of Avaya IP Office causing service disruption.)*

## 6. Configure CenturyLink

To use the CenturyLink SIP Trunk Service, a customer must request service from CenturyLink using their sales processes. The process can be started by contacting CenturyLink via the corporate web site at <http://www.CenturyLink.com> and requesting information via the online sales links or telephone numbers.

## 7. Verification Steps

This section provides verification steps that may be performed to verify that the Avaya H.323, SIP, digital and analog endpoints can place outbound calls and receive inbound calls through CenturyLink's SIP Trunk Service solution.

- Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on the PSTN can terminate an active call by hanging up.
- Verify that an endpoint at the enterprise site can terminate an active call by hanging up.

## 8. Conclusion

These Application Notes describe the configuration steps required to connect customers using an Avaya IP Office telephony system to a CenturyLink SIP Trunk Service solution. The CenturyLink SIP Trunk Service is a SIP-based Voice over IP solution. CenturyLink SIP Trunk Service passed compliance testing with the exception of items noted in **Section 2.2**.

## 9. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Avaya IP Office 8.0 Installation, Issue 24i*, December 2011, Document Number 15-601042
- [2] *Avaya IP Office Release 8 Manager 8.0, Issue 04i*, January 2012, Document Number 15-601011
- [3] RFC 3261 *SIP: Session Initiation Protocol* <http://www.ietf.org/>
- [4] RFC 2833 *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals* <http://www.ietf.org/>

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