**Document Purpose and Target Audience**

This document will serve as a reference guide to configure the Microsoft OCS 2007 IP PBX to interoperate with Time Warner Cable (TWC) SIP Trunk Service.

This guide is not intended to be a replacement of the PBX manufacture's user or configuration guide. It is intended to provide additional guidance on configuring the PBX in preparation to receive voice service from the SIP Trunk. It provides detailed instructions and best practices for a successful installation with TWC SIP Trunks.

There are many options for establishing and maintaining service using the Microsoft OCS 2007 series. This guide focuses on the minimum configurations essential for successful interoperability with Time Warner Cable Business Class SIP Trunks.

This configuration guide is based on:

**Customer Premise Equipment:**

<table>
<thead>
<tr>
<th>Model</th>
<th>Microsoft OCS 2007</th>
</tr>
</thead>
<tbody>
<tr>
<td>Firmware</td>
<td>Release 2</td>
</tr>
</tbody>
</table>

**TWC Network Equipment:**

| ESG         | InnoMedia ESBC 9378-4B |

2
SIP Trunk Components

The Time Warner Cable Business Class (TWCBC) SIP Trunks product is an IP-based, voice only trunk that uses Session Initiation Protocol (SIP) to connect an IP PBX to the PSTN. The IP PBX uses SIP to exchange signaling information with the service provider and to deliver and receive voice in IP packets.

The IP PBX is connected to the TWC Enterprise SIP Gateway (ESG), which provides network access for voice traffic. The customer is responsible for the LAN infrastructure and configuration, including the physical connection to the LAN port 2 on the ESG.

The ESG is the demarcation point to the TWC network. The ESG is connected to a dedicated router for SIP Trunks delivered over a fiber connection or to a cable modem when delivered over a DOCSIS connection.

SIP Trunk components located on the customer premise, including connections to the TWC network, are illustrated below.

All TWC SIP Trunk calls are routed over Time Warner Cable’s IP network and are not routed over the public internet.
Getting Started

You will need to have the TWC “SIP Trunk Questionnaire” and “Business Class (BC) SIP Trunks: Customer Cut Sheet” in order to configure your IP PBX for TWC Business Class SIP Trunk service.

Confirm that your IP PBX model number and software versions recorded on the Customer Cut Sheet match those associated with your current equipment. If they do not, be sure to alert your TWC sales engineer or TWC project manager as this can impact how TWC designs your service configuration.

Example from Customer Cut Sheet for Cisco UC 560:

<table>
<thead>
<tr>
<th>SERVICE INFORMATION</th>
</tr>
</thead>
<tbody>
<tr>
<td>PRODUCT</td>
</tr>
<tr>
<td>IP-PBX MAKE</td>
</tr>
<tr>
<td>IP-PBX MODEL</td>
</tr>
<tr>
<td>IP-PBX SOFTWARE Version</td>
</tr>
</tbody>
</table>

While configuring your IP PBX for BC SIP Trunk service, you will need to know your Lead Telephone Number and the IP address of your IP PBX.

The Lead Number is confirmed on the Customer Cut Sheet as seen below:

<table>
<thead>
<tr>
<th>TWC TRUNK Group ID</th>
<th>DID Range</th>
<th>Lead Number</th>
<th>Inbound Call Blocking</th>
<th>Outbound Call Blocking</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The IP Address of the IP PBX was recorded on the SIP Trunk Questionnaire, Section 5. Signaling and Media as shown below:

<table>
<thead>
<tr>
<th>5. Signaling and Media</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address for PBX or SBC</td>
</tr>
<tr>
<td>To setup LAN configuration for signaling of voice traffic to the ESG</td>
</tr>
</tbody>
</table>

This document is intended as an aid to help configure a customer’s IP PBX for interoperability with TWCBC SIP Trunk Service.
Microsoft Office Communication Server 2007 R2 configuration

The instructions provided in this section are intended to configure the OCS R2 to connect to the TWC ESG. They are not intended for advanced functionality setups. It is further assumed that there is already knowledge of OCS R2 operations.

1. Log on to a Communications Server 2007 Mediation Server.
2. Click Start, point to Administrative Tools, and then click Office Communications Server 2007

Create Location Profiles

1. Right-click the Forest node, point to Properties, and then click Voice Properties.
2. On the Location Profiles tab.

![Figure 1 Location Profiles](image)

3. Click Add to create a new location profile.
1. **Name**: type a useful descriptive name.

2. **Description**: type the common, recognizable name of the geographic location to which the location profile applies.

4. Click **Add** to open the **Add Phone Number Normalization Rule** dialog box.
3. **Name**: type a name that describes the number pattern being normalized (for example, 5 Digit Extension or 7 digit calling Toronto).

4. **Description**: type a brief description of the normalization rule.

5. **Phone pattern regular expression**: use .NET Framework regular expressions to describe a phone number pattern (for example, ^9\d{7}$, which describes a phone number pattern consisting of the number 9 followed by any seven digits).

6. **Translation pattern regular expression**: use .NET Framework regular expressions to describe the E.164 phone number corresponding to the number entered in the **Phone pattern regular expression** box.

7. **Use translation when dialing from device**: check.

5. Click **OK**.

6. Create as many normalization rules as your location profile requires.

7. When you are done, click **OK**.
Assign Location Profile to Pool

1. Configure an Enterprise Edition Pool, expand Enterprise Pools, right-click the pool that you want to configure, click Properties, and then click Front End Properties.

2. On the Voice tab, in the Location Profile box, select the default location profile for the pool.

3. Click OK.
Configure Mediation Server

1. Expand the **Mediation Servers** node, right-click the Mediation Server to be configured, click **Properties**, and then click the **General** tab.

2. In the **Default location profile** list, select the default location profile for this Mediation Server, as shown in Figure 5. The IP address entered for **Communication Server listening IP address** indicates the SIP Contact of PBX Static Registration on ESG page, as shown in Error! Reference source not found..

3. On the **Next Hop Connections** tab, under **PSTN Gateway next hop**: 

4. **Address:** specify the IP address of the ESG LAN interface.

Figure 5 Configure location profile

Figure 6 Configure PSTN Gateway address
9. **Transport**: select TCP.
10. **Port**: accept the default of 5060 for TCP.
4. Click **OK**.

## Configure Call Authorization

### Create a phone usage record

1. Right-click the **Forest** node, point to **Properties**, and then click **Voice Properties**.
2. On the **Phone Usages** tab, click **Add** to create a new phone usage record.

![Create phone usage record](image)

3. Continue adding phone usage records as needed, and then click **OK**.

### Create a Voice Policy
1. Right-click the **Forest** node, point to **Properties**, then click **Voice Properties**.
2. On the **Properties** dialog, click the **Policy** tab

![Figure 8. policy](image)

3. Click **Add**. Type a name in the **Policy name** box. **Select the Allow simultaneous ringing of phones** check box.
4. Click Configure. Select a phone usage from the Available list, and then click the right arrow to add the usage to the Configured list.

5. Click OK, and verify that the phone usages you added now appear in the list of Phone usage records in the Add Policy or Edit Policy dialog boxes.

6. Click OK.

To select a global voice policy

- On the Policy tab, open the Global policy list, select Use per user policy.
Configure Outbound Call Routing

1. Right-click the Forest node, point to Properties, and click Voice Properties.
2. Click the Routes tab
3. Click Add.

Figure 11 Select global voice policy

- Click OK.

Figure 12 Routes list
11. **Name**: type a unique name for the new call route.

12. **Description**: type an optional description of the new call route.

13. **Target phone numbers**: use .NET Framework Regular Expressions to describe the pattern of the phone numbers that are to use this route.

4. Click **Add** to add a media gateway or Mediation Server to the **Gateways** list.

5. Click **Configure**. Select the phone usage record you want to add from the **Available** list, and then click the **Right Arrow** key (>) to add the record to the **Configured** list.
6. Create as many routes as you need, and then click OK.

Enable Users for Enterprise Voice

1. Expand the pool or server where your users reside, and then click the Users node.
2. In the right pane, right-click one or more users whom you want to configure, and then click Configure users.
3. Follow the Welcome to the Configure Users Wizard to configure user.

Enabling Enterprise Voice disables remote call control for the selected users. Only enabled users with a valid line URI can use Enterprise Voice.
Figure 16 Configure Users

Configuring Telephony Setting of User

1. Expand the pool or server where your users reside, and then click the Users node.
2. Double-click the user you want to configure. Click Configures of Telephony Setting.
Figure 17. Configure Telephony setting for user

14. Line URI: Enter Pilot Number User ID.
15. Select the policy and location profile.
3. Click OK.
# Mediation Server Overview

<table>
<thead>
<tr>
<th>Status</th>
<th>Event Log</th>
<th>Performance</th>
<th>Resources</th>
</tr>
</thead>
</table>

## General Settings
- **Mediation Service**: Running
- **Certificate Settings**
  - **Name**: msrd.com
  - **Issuer Name**: rd
  - **Expiration Date**: 2011-7-21

## Location Profile
- **osipbx-digimap**: osi digimap - 3 digit extension
  - **3 digit extension**
    - **Phone Pattern**: ^\d(3)\$\n    - **Translation**: +1
  - **1 DIGITAL EXT**
    - **Phone Pattern**: ^\d(4)\$\n    - **Translation**: +1
  - **all digits**
    - **Phone Pattern**: ^\d+\$\n    - **Translation**: +1

- **AV Edge Server FQDN**: <None>
- **AV Edge Server Port**: <None>

## Listening Connections
- **Listening address for Communications Server**: 172.16.1.59
  - **Communications Server listening port**: 5061
  - **Listening address for Gateway traffic**: 172.16.1.67
  - **PSTN Gateway Listening Port**: 5060
  - **Media port range**: 60000 - 64000

## Next Hop Connections
- **Communications Server Next Hop FQDN**: pocrd.com
- **Communications Server Next Hop Port**: 5061
- **PSTN Gateway Address**: 172.16.1.20
- **PSTN Gateway Port**: 5060
- **Gateway Transport Type**: TCP
- **Gateway Encryption Level**: Do Not Support Encryption

## Route Information
The following routes are served by this Mediation Server. Please use the Route tab on the Voice property page to add, modify or delete a route. To access the Voice property page, right click the forest node of the MMC tree view pane.

- **route all external call to osi gw**
  - **Phone Number Pattern**: ^[^x]+(\d+)\$
  - **Phone Usage**: ospi

## Description:
Appendix

TWC Turn-up Testing Procedure

To ensure proper service between the IP PBX and the TWC network, test calls from the IP PBX will be made. Typically, the following call types will be used (call testing varies depending on service configuration)

1. Outbound/Inbound call to a local number
2. Outbound/Inbound call to a long distance number
3. Calls to 411 and 611
4. Outbound calls to a blocked number to verify call blocking settings
5. Other calls based on customer request, e.g. FAX testing using T.38 or calls to an auto-attendant to verify DTMF

Questions

If you have questions, please contact your Time Warner Cable Business Class Account Executive.