Abstract
This document describes Polycom’s SIP solution for multimedia conferencing consisting of Polycom’s SIP audio endpoints, video endpoints, audio and video bridges / MCU and Polycom’s WebOffice Portal. The document provides details of setting up and configuring the relevant equipment in specific environments, and outlines specific SIP-based conferencing scenarios. Main focus is given to the description of Polycom’s integration with Microsoft’s LCS and Nortel’s MCS environments.
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1 Introduction
This application note provides the required Information for configuring relevant Polycom and 3rd party products to enable their participation in the SIP-based multimedia conferencing scenarios as described in this document. While basic SIP-based scenarios should be possible using Polycom SIP products along with any 3rd party equipment, this application note focuses specifically on the scenarios made possible by using Polycom products within the Microsoft LCS and Nortel MCS environments, providing detailed information on configuring these environments in making such scenarios possible.

For a complete description of the Microsoft LCS or Nortel MCS configuration, please refer to the manufacturer’s documentation. For a complete description of Polycom products configuration, please refer to the relevant product’s administrator’s guide.
2 SIP conferencing terminology
Throughout this document basic conferencing terms, some defined by the IETF and others commonly used in the “SIP conferencing industry”, are used in describing the various SIP scenarios. This section provides a short description of these terms to ensure a clear understanding of the scenarios that follow.

- **Tightly Coupled Conference** - A conference in which a single user agent, referred to as a focus, maintains a dialog with each conference participant. The focus plays the role of the centralized manager of the conference, and is addressed by a conference URI.
- **Focus** - The focus is a SIP user agent that is addressed by a conference URI and identifies a conference. The focus maintains a SIP signaling relationship with each participant in the conference. The focus is responsible for ensuring that each participant receives the media that make up the conference.
- **Conference URI** - A URI, usually a SIP URI, which identifies the focus of a conference.
- **Factory URI** - In order to automatically create an arbitrary number of ad-hoc conferences (and subsequently their focuses) using SIP call control means, a globally routable Conference Factory URI can be allocated and published. A successful attempt to establish a call to this URI would result in automatic creation of a new conference and its focus.
- **Conference-Unaware Participant** - A participant in a conference that is not aware that it is actually in a conference. As far as the UA is concerned, it is a point-to-point call.
- **Conference-Aware Participant** - The participant in a conference that has learned, through automated means, that it is in a conference, and that can use a conference policy control protocol, media policy control protocol, or conference subscription, to implement advanced functionality.
- **Reserved / Scheduled conferences** – The conference is scheduled ahead of time to a specific time and date and usually resources are reserved and thus guaranteed.
- **Reservation less / Unscheduled / Meeting room (MR) conferences** – The conference has no specific time, no resources are reserved and therefore non are guaranteed. A MR is actually an “always available” dormant conference, which is activated upon first participant connecting subject to resources availability.
- **MeetMePer conference** – The conference has an associated / allocated direct dial-in number / address / SIP-URI all participants dial to enter the conference.
- **“Single dial” / Entry Queue (EQ) model** – All participants dial the same EQ number / address / SIP-URI regardless of their destination conference. Once connected they are being routed to their destination conference based on a unique conference ID they need to provide.
- **“MeetMePer MCU”** – All participants dial the same MCU number / address and are being routed to their corresponding conferences based on their source / CLI / “dial-from” number or address. This means participants must be predefined along with their respective number / address in the corresponding conference.
• **Dial out** – Once the conference is running, dial out can be initiated in various ways to join new participants into the conference. Participants can be predefined within the conference or defined dynamically while the conference is running.

• **Ad-hoc / a “spontaneous” creation of a new conference** - This type of conference is created upon a participant’s initiative. Unlike a MR, which does exist on the MCU (although it is dormant consuming no resources prior to its activation), an ad-hoc conference does not exist in any sense prior to its creation and “disappears” once it is terminated.

3 **Supported SIP-based scenarios**

The following table provides a list of the main SIP-based scenarios made possible using Polycom’s products. Subsequent sections detail the configurations required for running these scenarios in specific environments – Microsoft LCS and Nortel MCS.

<table>
<thead>
<tr>
<th>Scenario</th>
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| 1 MeetMe Conferencing<br>Direct dial in | • Dial in by conf-URI / alias.  
• Can join an active / running conference or activate then join a dormant Meeting room. |
| 2 “Single dial” conferencing<br>(Conference predefined on MCU) | • Dial in through an Entry Queue (EQ)  
• The conf-ID can be provided as part of the INVITE request or through using DTMF over the media channel once the call has been established.  
• The destination conference is predefined on the MCU. |
| 3 Ad-hoc conferencing | • Conference focus created ad-hoc upon initiator’s request with Conf-ID provided by initiator.  
• The initiating user dials a well-known Factory / ad-hoc EQ number / URI.  
• Other participants can then join the conference by either dialing the conference directly or going through the EQ and providing the conference ID. |
|   | **Dial out (conference reservation)** | • Participants can be defined by their IP address or SIP-URI  
|   |   | • To guarantee resources on the MCU the conference can be reserved along with its predefined users. At the scheduled time the conference is activated and dial out, to the predefined users, optionally occurs automatically or manually by the operator or conference chair. |
|   | **Ad-hoc**  
| 5 | **Multi party IM → A/V conference**  
|   | [Microsoft WM / LCS] | • Seamless escalation of a multi party IM session into a videoconference hosted on Polycom MCU with all IM participants automatically joined into the videoconference.  
|   |   | • All the initiating WM user needs to do is add a “MCU buddy” on his WM UI to the on-going IM session.  
|   |   | • The MCU creates an ad-hoc videoconference and dials out to all IM session participants joining them into the videoconference. |
|   | **Ad-hoc**  
| 6 | **“Buddy list selection”** | • Initiator selects conference participants from his buddy list and requests an ad-hoc conference creation. (See scenario 3 above)  
|   |   | • Once confirmation is received from the MCU for the Focus creation, the initiator’s UA can now refer the selected buddies into the newly created Focus / conference on the MCU. |
|   | **Point 2 Point** | • Any audio or video Polycom endpoint / SIP UA can call another
4 Polycom enhancements to Microsoft LCS environment
Polycom enhances the Microsoft LCS environment with enhanced multi-party multimedia conferencing capabilities, including high quality conference room video, and enables the SIP-only Microsoft environment to be bridged with legacy audio and video H.323 and H.320 / ISDN / PSTN equipment. Detailed Information for configuring relevant Polycom products as well as the LCS and WM is provided in subsequent sections. Polycom’s relevant products in this integrated solution include: Polycom VSX family of SIP video clients, Polycom SIP phones, Polycom Web Office Portal and the Polycom family of MGC MCUs. Microsoft’s Windows Messenger can be used as a desktop A/V/IM SIP UA. In addition to point-to-point calls the following conferencing scenarios are supported:

- MeetMe – including the ability to have a MCU conference on your WM buddy list and joining the conference by clicking on that buddy and selecting “Start video conversation”.
- Ad-hoc: Spontaneous creation of a new MCU conference from the UA. The initiator can specify the new conference name and ID.
- Reservations-based dial out. Scheduled conferences with optional predefined participants resulting in guaranteed allocated resources on the MCU.
- Seamless escalation of a multi party IM session between WM clients into an A/V conference on Polycom MCU.
- Multi selection of conference participants from WM buddy list and seamless initiation of an ad-hoc conference automatically joining the selected buddies into it.

4.1 Polycom WebOffice / Portal enhancements to Microsoft Windows Messenger (WM)
Polycom WO conferencing portal (PCP) has been integrated with Microsoft Windows Messenger (WM) to enhance the WM capabilities mainly as related to conferencing scenarios. PCP plug-in SW running on the same PC along side WM, Polycom portal server, Polycom MCU and endpoints all work together in enhancing the Microsoft native LCS / WM environment enabling new unique scenarios. The main capabilities added by PCP are:

- The WM buddy list has been enriched such that in addition to your buddy’s presence indication (Online, Offline, Busy, Away, etc.) you can now see the endpoint type and status each of your buddies happen to be using as illustrated below.
WM users can associate themselves with a variety of personal or group conferencing endpoints – including SIP, H.323, H.320/ISDN and PSTN based endpoints. In addition, user’s can easily switch between their default associated endpoint (e.g., a Via Video desktop camera) and a room based system (e.g., a View Station FX). The user’s associated endpoint can be selected from a list pre-defined by the System Administrator, or it can be added manually. To do this, WM users simply select “Start Conference Preferences” (A new selection item added by Polycom to WM UI) from the WM “Actions” menu. The WebOffice “Properties” page appears and the user can select or manually specify his desired endpoint type from within the “Conference Endpoints” tab as illustrated below.
5 Polycom SIP-scenarios in Microsoft LCS environment

This section provides a detailed description of scenarios made possible using Polycom products in the Microsoft LCS / Windows Messenger (WM) environment. Polycom enhances the Microsoft environment with the ability to conduct multipoint audio and video conferencing and bridging to the “older” H323 and H320 worlds in addition to supporting the native Microsoft SIP environment. For each scenario a “cook book” style full description is provided including a scenario explanation, required configuration of all relevant participating entities, operation, limitations and user experience.

5.1 Scenario 1 – Joining a meeting room (MR) through a WM buddy list

5.1.1 Scenario General Description

1. In this scenario a video Meeting Room (MR=always available MeetMe conference) is created on Polycom MCU and is registered with LCS.
2. The MR is added to a user’s WM buddy list.
3. To activate and join the MR the WM user clicks on the corresponding MR buddy and selects “Start video conversation”. Other SIP user agents as well as H323 and H320 users can now join the running conference.
4. The conference video is displayed on WM.

5.1.2 Scenario Configuration

1. Create a video MR on Polycom MCU. (See Polycom MGC user’s guide for details). For example - MR name = “conf4” (its conf-ID=1000), defined with – Continuous Presence, 384K, H263, G711.
2 Configure the **MCU IP service** to use TCP for SIP signaling transport and the LCS as its outbound proxy and registrar. Specifically configure the MCU IP service to point to LCS IP address and the LCS-provisioned trusted listening port (See screen below).

3 Configure the MCU IP service to automatically register all Meeting rooms and select the registration scheme to “Redirect” as illustrated in the following screen. (See “MGC multi IP board configuration in LCS environment” for a complete description of the different registration options of MCU conferences when there are more than a single MCU IP board)

4 Configure a LCS trusted port all MCU requests will be directed to so LCS does not challenge them for authentication. Please refer to the “Configuring a LCS trusted port” section of this document.

5 Create an account on LCS under the SIP URI: `conf4@svt.co.il` so the conf4 MR can register with LCS and thus become routable from any SIP UA using LCS.

6 Once the MR was created and automatically registered with LCS add it to your WM buddy list as illustrated below. (In this example the MR name=conf4)

7 The MR=conf4 was added to WM buddy list and since it was registered by the MCU with LCS, it status=Online.
5.1.3 Scenario Operation

1. To actually activate and join the MR right click on the “conf4” buddy and select “start video conversation” as illustrated below:

2. Using MGC manager you should be able to see that the conf4 MR was activated and appears now under the on-going section as illustrated below:
3. Other clients can now join the active MR by dialing its name=conf4 or conf ID=1000. For example **H323 users can** dial the MCU prefix, as configured at GK, followed by the MR name or conf-ID. For example: “80” +”1000 or “80”+”conf4” where 80 is the MCU prefix instructing the GK to forward the request to the MCU.

4. Video should appear on WM video area according to conf4 continuous presence / layout configuration. In the illustration below the conference was configured for a 1+2 layout (you can dynamically modify the conference layout selecting from the rich set of layouts offered by Polycom MCU).
5.2 Scenario 2: WM multi party IM session → video conference

5.2.1 Scenario General Description
1. In this scenario a multi party IM session held between WM users is seamlessly escalated into a videoconference hosted on Polycom’s SIP MCU.
2. The “escalation initiator” adds an “ad-hoc video” buddy (added to his buddy list) to the on-going IM session, consequently a new ad-hoc conference is created on Polycom’s MCU and all IM participants are automatically joined into it so they can now see and talk to each other in addition to exchanging IM / text messages.

5.2.2 Scenario Configuration
1. Create an ad-hoc EQ (with an associated video profile) on Polycom MCU as illustrated below. (See Polycom MGC user’s guide for more details). In this example Ad-hoc EQ name = “ad-hoc-video”.

2. Perform the following configuration steps (see similar steps described for scenario 1 above)
   - Configure the MCU IP service to use TCP for SIP signaling transport and the LCS as its outbound proxy and registrar.
- Configure the MCU IP service to point to LCS IP address and the LCS-provisioned trusted listening port.
- Set up an account on LCS="ad-hoc-video" so the EQ can register with LCS.
- Set up a LCS trusted listening port the MCU will be using.

3. Configure the MCU IP service to automatically register all EQ and select the registration scheme to “Redirect”.

4. Once the ad-hoc EQ was created on the MCU and automatically registered with LCS add it to your WM buddy list following a similar procedure to that described in scenario 1 above for adding the MR=conf4 to WM buddy list.

5. The EQ="ad-hoc-video" should now appear on your buddy list (presence=Online) as illustrated below:
5.2.3 Scenario Operation
1. To escalate the multiparty IM session into a videoconference select “Invite Someone to this Conversation” at your WM UI, as illustrated below.

2. At the window that opens up select the “ad-hoc-video” (EQ) buddy.
3. Behind the scenes this causes a new / ad-hoc video conference to be created on Polycom MCU and all current IM session participants receive an invitation to join it as illustrated below (in this example the name of the ad-hoc conference created on the MCU=6741).

4. Once participants accept the Invitation they are joined into the Videoconference on Polycom MCU, and receive the conference video on their WM UI as illustrated below. Note that the 6741 conference is now indicated as on-going on the MCU (MGC manager screen shot)
5.3 Scenario 3: Ad-hoc creation of audio / video conference from WM

5.3.1 Scenario General Description
1. In this scenario a WM user creates an ad-hoc audio / videoconference on Polycom MCU from his WM UI. Different from the MR scenario described above, in this scenario the conference does not exist on the MCU at the time the initiator calls the MCU Factory/ ad-hoc EQ, while
in the MR scenario the MR had to be pre defined on the MCU prior to its activation by the 1st caller.
2. The user can optionally specify the name and conf-ID of the new conference to be created as described below. If the user specifies no name or ID, the MCU will assign a pair of name – ID.
3. Once the A/V conference is created on Polycom MCU other users (SIP or others) can join it using the name or conf-ID assigned to it.

5.3.2 Scenario Configuration
1. Repeat the configuration steps described for scenario 2 above. (Configure the MCU IP service).
2. In this scenario the routing of the user’s request to Polycom MCU is based on LCS provisioned static routing so there is no need to set up an account on LCS or configure the MCU IP service to automatically register all EQ.
3. Configure a static route on LCS such that any request containing a URI user part="ad-hoc-video" would be routed to the MCU IP board. (See LCS configuration section for details of configuring a LCS static route).
4. Once the static route was created on LCS, the ad-hoc EQ on Polycom MCU becomes routable for LCS users to use for creating ad-hoc conferences on Polycom MCU as described below.

5.3.3 Scenario Operation
1. At your WM UI select “start video conversation” and type in the dial string. The user part consists of 3 parts: the ad-hoc EQ / Factory name, the name of the new conference to be created and its conf-ID in the format illustrated below: “EQ-name (conf-name)(conf-ID).
2. In the example below EQ-name=ad-hoc-video, conf-name=conf1 and conf-ID=007.
3. If the conf-name and / or conf-ID fields are left empty, the MCU will assign new unique values for these, for example – “ad-hoc-video()”.
4. Once the request is received by the MCU, it searches any existing conferences it hosts for a match in the specified conference name and ID fields. If such a conference is found, the caller is joined into it. If no match found, the MCU would create a new conference, using the name / ID specified, and join the caller into it.
5. Once joined into the conference the WM user gets the videoconference layout on his WM UI and can now invite others to join the on-going conference.
5.4 Scenario 4: Dial out from Polycom MCU to conference participants

5.4.1 Scenario General Description
1. In this scenario an A/V conference is created on Polycom MCU including predefined participants, WM and other SIP and non-SIP users.
2. At the scheduled time the conference is activated and the MCU automatically dials out to the predefined participants joining them into the conference.

5.4.2 Scenario Configuration
1. The relevant endpoints need be registered with LCS
2. A trusted listening port should be provisioned with LCS and the MCU IP service configured for the MCU to use it such that any requests sent from the MCU are sent to the LCS trusted port.
3. To define SIP participants at the MCU conference use the example illustrated below. In this example a user with a URI user part=Nancy is defined.
4. Once dial out is attempted by the MCU, the INVITE request will be routed to Nancy’s UA assuming Nancy is signed in with LCS at that point in time.
5.4.3 *Scenario Operation*

1. Once dial out is initiated the WM user is prompted to accept the call (see below) and would then be joined into the conference and get the video stream on his WM UI. (Note that the dial out was sent from “conf4” – name of MCU conference.)
5.5 Scenario 5: Multiparty IM→A/V conference using WO portal

5.5.1 Scenario General Description
1. In this scenario the WM user (involved in the IM session) clicks on the “Start Invite to Conference” new WM UI item added by Polycom to seamlessly escalate the IM session into a videoconference with all IM participants automatically joined in now sharing video and audio. There is no need to add a “video buddy” to the IM session as described in the previous scenario where the WO portal was not used and the scenario was based on the Polycom MCU only.

5.5.2 Scenario Configuration
1. You need to configure the MCU IP service to use the LCS as its outbound proxy. This is required since the WO portal interacts with the MCU behind the scenes (MCU API) in creating the conference and passing the participants addresses. It is the MCU, which actually dials out to the WM users and for this to work MCU, needs to send the INVITE requests through LCS.
2. You need to provision a trusted port on LCS where the MCU sends its requests that go unchallenged by LCS.

5.5.3 Scenario Operation
1. From the IM session click on “Start Invite to Conference” as illustrated below.
2. Selecting “Start Invite to Conference” causes all invited participants to receive an “Invite Message” on their WM as illustrated below.

3. Once invitees accept the invitation he will be joined into the ad-hoc conference that was created on Polycom MCU.
4. If a “Polycom conference portal video SW” is running on the invitees PC (along side WM) the portal can automatically refer the user into the conference hosted
on the MCU. In this case the conference video can be viewed and the conference managed within the standard WebOffice Meeting Manager UI (see below), offering the WM user the same powerful in-conference audio & video controls found in the standard WebOffice product - including Continuous Presence Layouts, Mute and Active Speaker indication, as well as owner controlled participant muting.

5. Please note that these video devices can be non-SIP such as Polycom’s H323 or H320 video endpoints supported by Polycom’s conferencing portal.

6. If the user has only WM with standard web cam then the Portal instructs the MCU to dial out to the WM user thus joining him into the conference.

5.6 Scenario 6: Buddy Multi-selection ➔ A/V conference using portal

5.6.1 Scenario General Description

1. This scenario, similar to the previous one, is made possible using Polycom’s conferencing portal server and plug-in SW running on each PC along side WM.

2. In this scenario Polycom enables the WM user to multi select the A/V conference participants from his WM contacts / buddy list and then click on a new WM UI item (added by Polycom) to seamlessly create an ad-hoc conference and join the selected buddies into it.

3. Except for the initiation step, described below, the rest of the scenario is identical to the previous IM ➔ video configuration scenario and therefore the full description is not repeated.
5.6.2 Scenario Operation
1. The WM user clicks on the buddies he wishes to invite into his A/V conference and then right clicks to open up a menu including the new “start invite to conference” selection added by Polycom
2. Once this option is selected the scenario proceeds similar to the IM→video scenario described above with an ad-hoc conference created on Polycom MCU and all selected buddies (similar to IM session participants in previous scenario) joined into it. (See below for buddies selection and conference initiation)

3. Alternatively you can select to initiate a conference by selecting the “Start Conference Organizer” option added by Polycom to WM UI. This will bring-up a “Conference Organizer” dialog box (see below) through which the user can simply point-and-click to “Add” conference participants from their Contacts list, much in the same way as was previously described for the “Start Invite to Conference” option. The one main advantage of this alternative is that WM users also have the option of manually adding “external” or non-WM based participants.
4. Electing to manually add a participant opens-up a “Participant Details” dialog box (see below) in which the WM user enters the specific endpoint details for the external participant. These endpoints may be SIP, H.323 or H.320/ISDN based personal or group video endpoints (e.g., a video conference room), or PSTN audio-only devices - such as an audio-endpoint (e.g., a Polycom SoundStation™) or even a cell phone.

5. Microsoft® MSN Messenger Clients - Microsoft Windows® Messenger has the embedded ability to allow users to include MSN Messenger (MSN) users within their “Contacts” list. This is possible when a user elects to configure their WM client for dual account registrations with both LCS and Microsoft’s .NET Messenger Service.
In this scenario, MSN users will be shown in the Contacts list along with their presence status, however there is no endpoint detection or status displayed. MSN users can be invited to participate in multipoint conferences, but the invite and join methods are slightly different than that for WM users. As with WM users, MSN users can be invited to participate either directly from within an IM session or through the “Start Invite to Conference” option (Note: they cannot be invited via the “Conference Organizer” option). In both cases, all invited MSN participants will receive an IM-based “Invite Message” which contains a URL hyperlink to the WM user’s WebOffice Lobby page. After clicking on the URL and being directed to the WebOffice Lobby page, the MSN user will need to select “Join a Meeting” from the list of available actions. At this point, the MSN user will be prompted to download and install the WebOffice meeting client software, after which they will be placed in the meeting.

6 Polycom enhancements to the Nortel MCS environment
1. Polycom’s relevant products in this integrated solution include: Polycom VSX family of SIP video clients, Polycom SIP phones and the Polycom family of MGC MCUs. In addition Polycom enables the SIP-only Nortel environment to be bridged with legacy audio and video H323 and H320 / ISDN / PSTN equipment.
2. In addition to the Nortel MCS Nortel also provides the MCS SIP video desktop client and Nortel’s SIP phones.
3. All these different endpoints using different network protocols and A/V codecs can seamlessly be joined into the same conference hosted on Polycom MCU with the MCU taking care of all required signaling and media transcoding / mixing.
4. In addition to point to point calls two main conferencing scenarios are supported – MeetMe and reservations-based dial out.
5. The rich set of Polycom MCU features (Continuous presence with many different layouts to choose from, full transcoding, flexible / configurable video IVR services, C&V, etc) is available for users of this integrated Nortel-Polycom environment.

7 Using Nortel’s MCS client
1. Nortel MCS PC client software Version 3.0 is required since it is the first version, which supports H.263 video.
2. The PC client is typically used with a Logitech camera, and can be set up to use the Nortel i2004 IP phone for all audio.
3. The client needs to be set for H.263 / CIF or QCIF.
4. The operation of MCS client is modified using the “Preferences” toolbar. The following options should be set under “Video”:
   1. Select “Custom Setting” and “Configure”
   2. Set the video codec to H.263
   3. Set the Video bit rate to 320kbps,
   4. Set the frame rate to 15 fps
   5. Set the resolution to CIF or QCIF
8 Joining a conference from a Nortel MCS Video client

The following describes the steps including relevant screen shots, of joining a conference through an EQ.

1. Dial from the Nortel client using “Make a Call”…

2. You will get the screen below…
3. Enter the EQ number as illustrated below…(78378)

4. Select “Make a Video Call”…

5. On the MCU side – before making the call:
6. After making the call – notice that EQ-NORTEL is now under the “On going conferences” which means it was activated and the calling user is in it going through the IVR phase getting his destination conference ID.
7. User get a prompt played by MGC to enter his conf-ID and using DTMF provides the corresponding number.
9 VSX V.7.0– SIP Setting up and operation

The VSX is a multi-protocol video conferencing endpoint. It can be configured to use H.323, H.320 and SIP and the priority order of these protocols can be specified. Thus for example, if the VSX is configured to enable H.323 and SIP and if H.323 is given priority, then when a call is attempted, an H.323 call will be attempted first, if that call fails, then a SIP call will be attempted. The following subsections provide detailed information concerning the required setup and operation of the VSX as a SIP UA video endpoint.

9.1 VSX settings

- In order to utilize SIP in VSX Version 7.0, two primary setup functions must be performed:
  1. The SIP protocol must be enabled and its priority must be established.
  2. SIP parameters such as transport protocol, proxy server, and registrar server must be set.
- In order to enable the SIP protocol, the user must enter the “Call Preferences” screen (System → Admin Settings → IP → Network → Call Preferences) and enable SIP as shown in the Figure below. Then, the user can set the protocol priority by selecting “Next” and setting the relative priorities by using the drop-down lists as shown below.
- Due to the complexities of routing calls in a mixed H.323 and SIP network, at this time, it is recommended that either SIP or H.323 be selected, but not both.
Enabling SIP on the Call Preference Screen

- To set the SIP parameters, the user must enter the SIP Settings screen (System → Admin Settings → IP → Network → SIP) shown below.
- The user may configure the system to use a proxy and registrar server using this screen. Note that SIP does not require the use of a proxy server and registrar server, so these fields can be left blank.
  1. **SIP Proxy Server** – the user may enter the DNS name or IP address of the SIP Proxy Server. If this field is left blank, no proxy server will be used. If no port is entered, then SIP signaling is sent to the proxy server on port 5060. In order to specify a port, the user enters an IP address (or DNS name) and the port as follows: 216.54.150.112:5070. In this case, all SIP signaling will be sent to Port 5070 of the SIP Proxy Server at IP address 216.54.150.112.
  2. **SIP Registrar Server** – the user may enter the DNS name or IP address of the SIP Registrar server. In some cases the SIP Proxy server and SIP Registrar server will be the same, but they can be different. If this field is left blank, no registrar server will be used. If no port is entered, then SIP signaling is sent to the registrar server on port 5060. In order to specify a port, the user enters an IP address (or DNS name) and the port as follows: 216.54.150.112:5070. In this case, all SIP registration messages will be sent to Port 5070 of the SIP Registrar Server at IP address 216.54.150.112.
  3. **SIP User Name** – the user may enter the SIP user name for the system. If this field is left blank, the system’s IP address will be used as the SIP user name.
4. **SIP Password** – the user may enter a password, which is used to generate a response to a Digest or NTLM challenge from a Registrar server or peer endpoint. Since authentication in not implemented in this release, this field is currently ignored.

5. **Transport Protocol** – the user must choose either TCP or UDP as the transport protocol used for SIP signaling (the Microsoft LCS requires TCP and the Nortel MCS requires UDP). Only one transport protocol can be selected, so selecting UDP, unselects TCP and vice versa.
Setting SIP Protocol Priority

- Note that the VSX will only register with the SIP Registrar Server when it starts (i.e. re-boots). Navigating to the SIP Settings screen and entering Information in the SIP Proxy Server field or the SIP Registrar Server field will not cause the VSX to re-register. The user must re-start the system to initiate the registration process.

9.2 VSX SIP Operation
- Placing a SIP call is accomplished in a manner identical to placing a H.323 call or an ISDN call. From the “Place a Call” screen shown below, the user enters the SIP URI (which is routable by the SIP Proxy server, or the IP address of another endpoint and then presses the call button on the remote.
Placing a SIP Call

- **Important Notes**
  1. On the VSX 8000, VSX 7000 and the VSX 3000, the user can manually designate the protocol to be used for the call by using the drop-down list shown in the Figure above. By manually selecting the protocol, the user overrides the protocol preferences which were designated on the “Network Dialing” screen.
  2. On the V500, the protocol selection drop-down list is not present on the “Place a Call” screen. Thus, the user cannot override the protocol preference order set on the “Network Dialing” screen.
  3. Due to the complexities of routing calls in a mixed H.323 and SIP network, at this time it is recommended that either the SIP or H.323 protocol be selected, but not both. Selecting only one protocol will simplify SIP, or H.323 testing.

9.3 **Operating the VSX with the LCS and Windows Messenger 5.0**
- In order to operate the VSX in the Microsoft LCS SIP infrastructure, four primary setup operations must be performed:
  1. The LCS must be configured to trust any SIP signaling or registrations on a port other than 5060 (e.g. 5070). This must be done because the VSX does not currently support NTLM authentication. The steps required to set up this trusted port are described below in the “LCS configuration” section.
  2. The LCS server must be configured with an account with a SIP user name to be assigned to the VSX. The steps required to set up a LCS account are described below in the “LCS configuration” section.
3. The VSX must be configured to operate with the LCS. This requires the following steps:
   - SIP must be enabled on the Call Preferences screen.
   - The proxy server must point to the trusted port of the LCS (e.g. 216.54.150.112:5070).
   - The registrar server must point to the trusted port of the LCS (e.g. 216.54.150.112:5070).
   - TCP must be selected as the SIP transport protocol.
   - The user name must be configured to match the account set up on the LCS.
   - The password must be configured to match the account set up on the LCS.
     (this field is actually ignored in VSX Release 7.0 because the VSX does not perform authentication.
     In future releases, the password in this field will be required to match the account password on the LCS).

4. The account used by the VSX must be configured to enable the LCS to send presence Information to all the Windows Messenger clients which are registered to the LCS and which have added the VSX account to their contact list. The steps required to achieve that are described below in the “LCS configuration” section.

9.4 Operating the VSX with the Nortel SIP MCS server and PC Multimedia Client

- Setting up the Nortel Multimedia Communication Server (MCS) to enable SIP Proxy services is outside the scope of this document. That Information is available from the Nortel website. Once the MCS is configured, two primary setup operations must be performed to enable the VSX to operate in this SIP infrastructure:
  1. The MCS must be configured with a trusted SIP account. The account must be trusted because the VSX does not support authentication in this release.
  2. The VSX must be configured to operate with the MCS. This requires the following steps:
     - SIP must be enabled on the Call Preferences screen.
     - The proxy server must point to the MCS (e.g. 216.54.150.112).
     - The registrar server must point to the MCS (e.g. 216.54.150.112).
     - UDP must be selected as the SIP transport protocol.
     - The user name must be configured to match the account set up on the MCS.
     - The password must be configured to match the account set up on the MCS.
     - The steps required to setup the MCS PC client are described below in the “Nortel client setup” section.

10 MGC V.7.0 – SIP Features table, Set up and operation
10.1 MGC V.7.0 “SIP features” table

<table>
<thead>
<tr>
<th>Feature</th>
<th>Supported Y/N</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Supported Y/N</td>
<td>Comments</td>
</tr>
<tr>
<td>------------------------------------------------------------------------</td>
<td>--------------------------------</td>
<td>-----------------------------------------------</td>
</tr>
<tr>
<td><strong>Common IP service for H.323 &amp; SIP</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>o Same board can serve H.323, SIP or both</td>
<td>√</td>
<td>IP48 V.4.23 or higher</td>
</tr>
<tr>
<td>o Each port on IP board can serve H.323 or SIP</td>
<td>√</td>
<td></td>
</tr>
<tr>
<td>o An IP service can contain multiple IP boards</td>
<td>√</td>
<td></td>
</tr>
<tr>
<td><strong>Optionally different default IP service for H.323 &amp; SIP</strong></td>
<td>√</td>
<td></td>
</tr>
<tr>
<td><strong>QOS - Same as for the current H.323</strong></td>
<td>√</td>
<td></td>
</tr>
<tr>
<td><strong>Boards</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>o SIP is supported on IP48 &amp; IPN</td>
<td>√</td>
<td>IP48 V.4.23 or higher</td>
</tr>
<tr>
<td><strong>SIP infrastructure</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>o SDP support</td>
<td></td>
<td></td>
</tr>
<tr>
<td>o Media parameters (codec, resolution, frames per second, etc.) + dynamic payload type</td>
<td>√</td>
<td></td>
</tr>
<tr>
<td>o Offer/Answer</td>
<td>√</td>
<td></td>
</tr>
<tr>
<td>o Session timer</td>
<td>NO</td>
<td></td>
</tr>
<tr>
<td><strong>Proxy manager</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>o Locate SIP servers by name</td>
<td>√</td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Supported Y/N</td>
<td>Comments</td>
</tr>
<tr>
<td>---------</td>
<td>---------------</td>
<td>----------</td>
</tr>
<tr>
<td>Outbound proxy</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Registrar</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Register ongoing conferences, Meeting Rooms, entry queues</td>
<td>✓</td>
<td>Configurable</td>
</tr>
<tr>
<td>Register</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Multiple boards support</td>
<td>✓</td>
<td>Redirect, Polling, Forking</td>
</tr>
</tbody>
</table>

- **Intra request**

  | By Info (signaling) | ✓ |

- **TCP and UDP SIP signaling transport**

  | ✓ |

---

**Media**

- **Audio**

  | G.711, G.722, G.722.1, G.723.1, G.729A | ✓ |

- **Video**

  | H.261 QCIF, CIF | ✓ |
  | H.263 QCIF, CIF, 4CIF (VSW) | ✓ |
  | Annex-D | ✓ |

- **Video modes**

<p>| Video switching | ✓ |
| Transcoding / Continuous | ✓ |</p>
<table>
<thead>
<tr>
<th>Feature</th>
<th>Supported Y/N Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>presence</td>
<td></td>
</tr>
<tr>
<td>o FECC</td>
<td>NO</td>
</tr>
</tbody>
</table>

**Conference modes**

| o Dial-in & dial-out                   | √                      |
| o Ad-hoc by entry queue               | √                      |

**Interoperability**

- **Live Communications**

| o Conf / EQ appear in WM buddy list along with its presence information. | √ |
| o IM --> A/V multi-point                                                       | √ |
| o T.120 for Windows Messenger clients                                           | NO |

- **Nortel MCS**

| o MCS video client, Nortel SIP phone, Polycom VSX and SIP phone – all join a videoconference on Polycom MCU. Dial in (MeetMe) and dial out (reservation) are both supported. | √ |

- **Proxies**

| o LCS                                  | √ |
## Polycom SIP-based Multimedia Conferencing

<table>
<thead>
<tr>
<th>Feature</th>
<th>Supported Y/N Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>o Nortel MCS</td>
<td>√</td>
</tr>
<tr>
<td>o Iptel</td>
<td>√</td>
</tr>
<tr>
<td>o Any other standard based SIP proxy</td>
<td>√</td>
</tr>
</tbody>
</table>

### Application

- **DTMF**
  - o In-band / audio samples | √ |
  - o RFC 2833 | √ |
- **CDR** | √ |
- **Move participants between conferences** | √ |
- **IVR** | √ |

### Management

- **MGC manager** | √ |
- **Web commander** | √ |

### 10.2 MGC Setup – IP service

- The MGC is a multi-protocol MCU that can be configured to concurrently use H.323, H.320 and SIP. In order to utilize SIP in the MGC an IP service MUST be defined on the MCU supporting SIP and IP cards need be associated with this service.
- The procedure for configuring a SIP IP service and associating IP cards with it is similar to the way this has been supported for a H.323 service prior to the introduction of SIP on Polycom’s MCU.
- To have SIP support on your Polycom MCU you need to have an IP48 card V.4.23 or higher and have MGC V.7.0 or higher.
- The same IP service can be configured to support both SIP as well as H.323.
- A typical SIP IP service configuration is illustrated below. For a complete description of the MGC IP service please refer to Polycom’s MGC administrator’s guide documentation.
- Through configuring the MCU IP service you can:
  1. Select between Redirect, Forking and Polling registration mode for supporting multiple MCU IP cards.
  2. Select registering any or all of – on going conferences, meetings rooms and entry queues.
3. Select the registration refresh interval
4. Select the SIP signaling transport (TCP / UDP)
5. The SIP server – outbound proxy and registrar.

The rest of this subsection provides a detailed description of the steps you need to follow to configure the above.

- In the screen below you can configure:
  1. Launch the SIP servers screen
  2. Determine which conferences should be registered with the Registrar (Ongoing, MR, EQ)
  3. The registration mode (Redirect, Forking or Polling). This is important for MCU multi IP cards support and determines how conferences on the MCU are registered with the registrar and consequently how SIP requests are routed to MCU IP cards.
  4. The MCU registration refresh interval.
• In the following screen you can configure:
  1. SIP transport layer to be UDP or TCP
  2. Outbound proxy IP / DNS address and port
  3. Registrar IP / DNS address and port
  4. Optionally and alternate SIP server
10.3 SIP Operation

- Placing a SIP call (Dial out) is similar to placing a H.323 call. The participant needs be designated as “SIP” and the destination URI needs be provided.
- This URI can be a fully qualified “user part@domain” address (domain can be an explicit IP address) or contain just the user part. In the later case the MCU will automatically append the domain / host part taken from the IP service configuration.
- In the following example the user part was configured to be “007”, the MCU would append the domain part, taken from the IP service, to create a fully qualified SIP URI and send the dial out / INVITE request to the MCU outbound (assuming one was defined in the IP service).
10.3.1 MGC in the LCS / WM 5.0 environment
Polycom’s integration and enhancement of Microsoft’s LCS environment is illustrated in the following diagram.
In order to operate the MGC in the Microsoft LCS SIP infrastructure, the following setup operations must be performed on the LCS (detailed instructions of how to perform these configurations is provided in the “LCS configuration section of this document):

1. A trusted listening address / port need be provisioned with LCS such that the LCS does not challenge any requests coming from the MCU.
2. For a conference / EQ on the MGC to be routable and be added by WM users as a buddy on a WM client, the LCS server must be configured with a corresponding account including the conference SIP user name to be assigned to that conference.
3. For all MGC conferences to become routable through LCS without explicit registration, that required provisioning corresponding accounts with LCS, – **static routing** can be provisioned at the LCS. This basically means that the LCS will make routing decisions forwarding requests towards the MCU based on implicit URI pattern matching rather than explicit registrations. For example you can provision the LCS such that any request with “80*” as its user part will be forwarded to the MGC at the provisioned IP address.

The MGC must be configured to operate with the LCS. This requires the following steps:

1. In the MGC SIP-enabled IP service the LCS must be configured as the outbound proxy and registrar. The IP address and port must match the provisioned Trusted port provisioned on the LCS. For example point to port 5070 of the LCS.
2. TCP must be selected as the SIP transport protocol.
3. Conference names to be added to users’ buddy lists must be configured to match the corresponding accounts set up on the LCS.
4. If static route is provisioned with LCS to route callers to MCU conferences the corresponding conference names must match the pattern provisioned with LCS.
5. The registration scheme in the MCU IP service should be configured to Redirect.
• To add a “trusted” connection to LCS or configure a static route (as detailed in the LCS configuration sections below) you need to launch the LCS application, as illustrated in the following screen shot.

10.3.2 MGC multi IP board configuration in LCS environment
• When the MGC has multiple IP boards the following configuration needs to be done on MGC and LCS to have LCS route an incoming call to a MGC IP board with available resources.
• The required configuration is explained through a simple example.
  1. Assuming there are 2 IP boards in the MCU with IP addresses: IP1=172.22.128.10 and IP2=172.22.128.11 respectively.
  2. Set the MGC IP service to use SIP redirect. This means conferences on MGC will register with the provisioned registrar (LCS in our case) through or in association with a single IP board, say IP1.
  3. Let’s assume a meeting room with a name “Conf1” was defined on MGC and registered with LCS with a IP1=172.22.128.10 contact.
  4. On LCS you need to configure a static route to the MGC IP boards as follows. Using our example a static route need be configured with LCS pointing to MGC IP2 board with a user part='*' and a domain part='172.22.128.11”. This means any request received by LCS with a request URI matching the *@172.22.128.11 pattern, will be forwarded to MGC IP2 board.
  5. Now let’s assume a user is trying to join Conf1 by dialing in from his SIP UA. An INVITE is sent from the UA to LCS including the following SIP request URL=Conf1@polycom.conf.com.
  6. This INVITE shall be routed by LCS to MGC IP1 based on the explicit conference registration done by the MGC when the meeting room was created.
7. Lets assume that at this point resources are unavailable on MGC IP1 board so MGC replies with a REDIRECT response including the address of IP2 board: Conf1@172.22.128.11. (assuming resources are available on IP2)
8. Since this URI matches the static route provisioned with LCS the original INVITE will be routed by LCS to IP2.
9. In case there are additional MGC IP cards corresponding static routes should be configured in LCS following the same format.

10.3.3 MGC in the Nortel MCS server / Multimedia Client environment
- The integrated Nortel-Polycom solution is illustrated in the following diagram

The integrated solution includes the following equipment: (see diagram above)

**Polycom**
- MCU V.7.0 (MGC25/50/100)
- SIP video clients: VSX7000, VSX3000, V500
- Polycom SIP phones SP300, 600
- Legacy H323 and H320 video endpoints. This is not part of any interop scenarios and Nortel’s equipment is actually not aware of these endpoints that simply connect to the MCU directly not through the Nortel infrastructure.

**Nortel**
- MCS 5100 / 5200
- MCS Soft video client
- SIP phone
All these different endpoints can join the same conference on the MCU.

- **MeetMe scenarios**
  1. A conference, Meeting room (MR) or Entry Queue (EQ) is configured on the MCU. This conference has a name /alias / SIP URI and it is routable for all SIP clients through the MCS.
  2. To become routable a conference needs to be provisioned with the MCS.
  3. To join a conference directly the user needs to type in its name / URI.
  4. A conference might have a password and chair password, which are provided by the user using DTMF as part of the conference joining IVR interaction. The UA INVITE request is routed by the MCS to the MCU based on MCS admin provisioning.
  5. To avoid MCS provisioning of multiple conferences, a “single dial” scheme is recommended where a single EQ is provisioned with MCS and initially joined by all users. Through an IVR-DTMF interaction a user then provides his specific conf-ID and is then joined into it.
  6. While in the conference the rich set of MGC supported DTMF activated functions is available including C&V used for changing the video display layout.
  7. H323 and H320 can join the conference directly or through the EQ by dialing its alias / number. The MCS is unaware of this activity.
  8. SIP phones can join a videoconference and will get the audio portion of it (“secondary”).

- **Direct Meeting Room (MR) calling Example**:
  1. Configure a MR on MCU; name = “007”.
  2. The MR is provisioned with MCS under the SIP URI: 007@Nortel.conf.com and thus becomes routable from any SIP EP.
  3. To actually activate and join the MR one needs to type in “007” at the SIP client. The “007” user part is completed into a fully qualified SIP URI by the UA and sent to the MCS.
  4. Other clients can now join the active MR by dialing its name.
  5. H323 users dial the MCU prefix, as configured at GK, followed by the MR name. For example: “80”+”007” where 80 is the MCU prefix instructing the GK to forward the request to the MCU.

- **“Single Dial” Entry Queue (EQ) calling Example**
  1. Configure an EQ on the MCU; name = “296”.
  2. The EQ is provisioned with MCS under the SIP URI: 296@Nortel.conf.com and thus becomes routable from any SIP EP.
  3. To actually activate and join one of the MRs defined on the MCU (note that there could be many MRs defined on the MCU and none of these need be provisioned with MCS since users do not call them directly, only the EQ need be routable for users to join) a user needs to type in “296” at the SIP client. The “296” user part is completed into a fully qualified SIP URI by the UA and sent to the MCS.
  4. Once connected to the EQ users provide their conf-ID (using DTMF)
  5. If this is the 1st caller the MR will be activated and the caller joined into it. Other clients can now join the active MR.
6. H323 users dial the MCU prefix, as configured at GK, followed by the EQ name. For example: “80”+”296” where 80 is the MCU prefix instructing the GK to forward the request to the MCU.

- **Multi MGC IP boards support**
  1. In the Nortel integrated solution Polling is the preferred way through MCS provisioning as detailed below.
  2. Every conference on the MCU that needs to be directly routable through MCS needs to first be provisioned with MCS. The MCS supports the provisioning of multiple IP addresses, for the same account / URI, along with assigned priorities / weights.
  3. The different MCU IP cards addresses need be provisioned with MCS under the corresponding account / URI. When a call gets to MCS including such a URI (for example – EQ name) it forwards the request to the highest priority MCU IP board and will continue sequentially until an OK comes back from one of the MCU IP cards.
  4. Since all MCS routing to the MCU is based on provisioning there is no need for any registration of MCU conferences with MCS.
  5. To make the admin MCS provisioning effort reasonable it is recommended to provision a single MCU EQ all users go through on their way to their destination conference.

- **Multi MCU support**
  1. There is no “Single dial” service offered at this phase across multiple MCUs.
  2. It is recommended that a single EQ be provisioned with MCS for each MCU and users will be dialing different EQ numbers to get into conferences on different MCUs.

- **Dial out scenario**
  1. A conference is configured / scheduled on the MCU.
  2. The conference might have a list of predefined participants or participants can be defined / added while the conference is running. Either way dialing out to these participants and joining them into the conference can be initiated automatically at conference activation or at the chair initiative any time during the conference.
  3. The INVITE requests are sent by the MCU to the MCS and then forwarded to the corresponding user’s registered devices / contacts.
  4. MCS forwards any request from the MCU unchallenged for authentication based on the request’s source IP being one of the MCU’s IP cards that were provisioned to be trusted with MCS.
  5. H323 and H320 users can be joined to the same conference.

11 **Microsoft LCS configuration**

11.1 **Introduction**

- To enable some of Polycom’s SIP-based scenarios outlined in this document within the Microsoft LCS environment, some pre-configuration of the LCS is required as outlined in the following sub sections.
- The LCS configuration described in this section includes:
1. **Configuring a LCS static route** – this enables making SIP entities / UAs routable without explicit registration. Routing is performed by LCS based on comparing the received URI with the provisioned static route pattern, should a match be identified the requests is forwarded based on the provisioned “next hop” address. This makes sense for example when making many MCU-hosted conferences routable for users to join without explicitly registering all these conferences with LCS. Yet another example is for ad-hoc conference creation where the user provides a URI containing the name of the new conference to be created. In this case the URI is obviously not registered with LCS, since the conference was not created at that point, and the only way for the request to be routed to the factory / MCU is based on LCS provisioned static routing.

2. **Configure a LCS Trusted listening port** – requests received by LCS on such a port go unchallenged for authentication. This makes sense mainly for a MCU, which similar to a GW is a network device representing many SIP entities (conferences) and can be considered to be trusted.

3. **Configuring a LCS user account** – for a SIP UA / conference to be able to register with LCS and possibly appear on other UA buddy lists, a corresponding account needs to first be provisioned with LCS.

4. **VSX on users buddy lists** – to enable VSX to be added as a buddy to LCS UAs and consequently be able launch calls to a VSX from your buddy list, the procedure outlined below should be followed.

### 11.2 Configuring a LCS Static Route

- The need for provisioning a static route on LCS arises in scenarios where an entity needs to become routable through LCS without explicit registration. This usually makes sense for a GW or MCU which are network entities representing many SIP UAs / conference. Static routing makes less sense for endpoints which represent a single UA and can use explicit SIP registration.

- One scenario is for users to dial into and join conferences on the MCU without having to register all these conferences with LCS. So a static route can be provisioned such that all SIP URIs having a user part with a “80” prefix would be routed to the MCU. All conference names on the MCU will be configured to start with 80 correspondingly. When a specific request is received by the MCU it would examine the full user part (the “80” part and the rest), determine the destination conference and join the caller into it.

- Another relevant scenario is creating an ad-hoc conference from a SIP UA; in this example the initiating user types in a URI identifying the name of an ad-hoc EQ defined on the MCU, the name of the ad-hoc new conference to be created and optionally the new Conf-ID. For example the user might type “80EQ1(conf77)(007)”. LCS will forward the corresponding request (containing this URI) towards the MCU based on the provisioned static route “80” prefix. The MCU will internally route this request to the ad-hoc EQ called “80EQ1”, the result will be a new conference created under the name “conf77” with an assigned ID=”007”. This scenario is impossible without using LCS static routing To configure a static route, perform the following steps: (see screen shot below)
1. Open the Computer Management console.
2. In the console tree, double-click Services and Applications.
3. Right-click Real-Time Communications Server and then click Properties.
4. Click the Routing tab.
5. Click Add to create a new static route.
6. Enter wildcards in the User and Domain boxes according to the static route you are provisioning. In the example illustrated below the user part was set to “80*” which means any request with a URI starting with “80” matches the pattern and, therefore will be routed based on the static route parameters for the “next hop”.
7. Click “Next hop” IP Address.
8. Enter an IP address where you want all requests matching this static route pattern to be forwarded. This can be for example the IP address of a Polycom MCU IP card where requests are to be sent for users attempting to join a conference on the MCU.
9. In the Transport dropdown list box, click TCP.
10. In the Port box, enter the next hop / destination TCP port (5060). This can be for example the TCP port a Polycom MCU IP card is listening on for SIP messages.
11. Click the OK button to save your settings.
   - In the following example any request containing 80, for the URI user part prefix, will be forwarded to IP=172.22.130.22 port=5060.
11.3 Configuring a LCS Trusted Port

- Similar to a GW, being a network infrastructure device representing many endpoint entities, it is reasonable to assume that the LCS should not challenge a MCU for authentication.
- To achieve the above trust settings must be applied to the listening address where MGC requests are to be sent. (Trust settings need to be used by Polycom SIP UA / endpoints as well at this point since they do not support the LCS authentication scheme). Any SIP request (INVITE, REGISTER or other) sent to the LCS trusted port address would go unchallenged by the LCS.
- To add a new Connection / listening port to the LCS perform the following steps: (In this example the new listening port is 5050, which is added to the already existing 5060 and 5070 LCS listening ports.

1. Open the Computer Management console.
2. In the console tree, double-click Services and Applications.
3. Right-click Real-Time Communications Server and then click Properties.

Click the Connections tab.
Windows Management Instrumentation (WMI) Tester, also called WBEMTest, is the tool used to view and modify the settings for the connections between the LCS Server and SIP user agents.

To apply “Trusted settings” to a listening address and port combination perform the following steps:
1. On the taskbar, click **Start**, and then click **Run**
2. In the Open text box, type **Wbemtest**.
3. Click **Connect**
4. In the Namespace box, type **root\cimv2**, and then click **Connect**.
5. Under IWbemServices, click Enum Classes

6. Select Recursive and then click OK
The Superclass Information dialog box

7. Double-click **MSFT_SIPLISTeningAddressData** to open the Object Editor dialog box

The Recursive Classes Query Result dialog box

8. Click **Instances** to open the Query Result dialog box
The Object Editor dialog box for MSFT_SIPListeningAddressData

9. Double-click each instance ID until you have chosen the one for transport type **TCP**.
The Instances Query Result dialog box for MSFT_SIPLookingAddressData

10. The window shown below will appear when you click on one of the listening address instances. To determine if it is the correct one, look in the Properties window and verify that Port line indicates the connection / port number you are trying to set to become trusted. In the example window below is shows Port=5060 but you might be configuring another listening port such as the 5050 connection / port used in the above example. In addition to the port number, to identify you are configuring the right instance, make sure TransportType is TCP. If you have opened the wrong instance, click Close and select the other listening address instance. If there are more than two listed, search through the instances until you find the appropriate one. If you have more than one IP address, use the IPAddress property to ensure you are configuring the appropriate listening address.

11. Highlight the TreatAllConnectionsAsServer property and click the Edit Property button
The Object Editor dialog box for the TCP instance of MSFT_SIPListeningAddressData

12. Select **Not NULL** and type **TRUE** in the **Value** text box.
13. Click **Save Property**
The Property Editor dialog box for the TreatAllConnectionsAsServer property

14. Highlight the TreatAllConnectionsAsTrusted property and click Edit Property
The Object Editor dialog box for the TCP instance of MSFT_SIPListingAddressData

15. Select **Not NULL** and type **TRUE** in the **Value** text box.
16. Click **Save Property**
The Property Editor dialog box for the TreatAllConnectionsAsTrusted property

17. Highlight the **DropRouteHeaders** property and click **Edit Property**.
The Object Editor dialog box for the TCP instance of MSFT_SIPListingAddressData

18. Select Not NULL and type TRUE in the Value text box.
19. Click Save Property.
The Property Editor dialog box for the DropRouteHeaders property

20. Click **Save Object** in the Object Editor for MSFT_SIPIPLteningAddressData dialog box for this instance.
21. Click **Close** to close the Query Result dialog box.
22. Click **Close** to close the Object Editor for MSFT_SIPIPLteningAddressData dialog box.

11.4 Setting up a SIP Account on the LCS

- In order to register a SIP UA or conference (hosted on Polycom MGC) with LCS a SIP account must be pre configured by:
  1. A user must be set up in the domain on which the LCS is running,
  2. The user must be enabled to use Live Communication Services.
- To set up a user in the domain on which the LCS is running, use the following steps. First open up the “Manage Your Server” dialog and click on “Manager users and computers in Active Directory” as shown in the Figure below.
Manage Your Server Dialog

- Right click on “Users”, select “New” and then “User” as shown in the next Figure.
Creating a new User

- Follow the steps indicated in the next 3 figures to create a new user on LCS.
Creating a new User-1
Creating a new User-2
Creating a New User-3

- Right-click on the new user and select “Properties” as shown below.
Selecting the new User

- In the dialog box, click on the “Live Communication Server” tab, enable LCS services for the user, enter a SIP user name (URI) and select the home server for this user as shown below.
Enabling LCS for the new user

- The LCS server may need to be stopped and restarted before the LCS recognizes this user. That is accomplished through the LCS management console.

11.5 Enabling the VSX to appear on LCS users buddy lists

- To enable LCS user agents, such as Windows Messenger clients, to add a VSX UA to their buddy lists and initiate video calls with VSX, the following steps should be followed which result in user being able to add VSX to their buddy list without the need for the VSX to approve it.
- The account used by the VSX must be configured to enable the LCS to send presence information to all the Windows Messenger clients which are registered to the LCS and which have added the VSX account to their contact list. (VSX Watchers)
- The simplest way to do this is to sign on using the VSX’s SIP account using a Windows Messenger client. Then, open the Options Dialog by selecting “Tools → Options” from the main Messenger menu as shown below.
Opening the Options Dialog

- Next select the Privacy Tab in the Options Dialog. For new accounts, “Alert me when other people add me to their contact list” will normally be selected as shown in the next Figure. This option should be unchecked as shown in Figure that follows.
Select Privacy Tab on the Options Dialog
Deselect “Alert me when other people....”

- Click on “OK” to close the Options Dialog and then sign out from the VSX’s account.
12 References

1. “WebOffice/Windows Messenger Integration” Application Note
2. “Polycom Video Conferencing and SIP in VSX 7.0” White paper
3. For more information on Polycom products, visit www.polycom.com.
4. For more information on IETF drafts, visit www.ietf.org.