



Cisco TelePresence Multiway™

Deployment Guide

Cisco VCS X7.2
MCU version 4.3 and 4.4
Conductor XC2.2

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Introduction

Objectives and intended audience

This deployment guide is designed to guide the reader on how to configure a Cisco TelePresence Video Communication Server (VCS) Control and Expressway and a Cisco TelePresence MCU to use the Cisco TelePresence Multiway™ feature. If a Cisco TelePresence Conductor is being used to manage conference bridge resources this document describes how to integrate the TelePresence Conductor into the solution.

The deployment with a TelePresence Conductor supports both Cisco TelePresence MCU version 4.2 and later and TelePresence Server version 3.0 running in *Remotely managed* mode. The deployment without a TelePresence Conductor does not support TelePresence Server.

This document reflects the user interfaces presented on VCS X7.2, TelePresence Conductor XC2.2 and MCU version 4.3 software. Different software versions may require different navigation paths or use different field labels to refer to some settings.

Overview of Multiway™

Multiway conferencing enables video endpoint users to introduce a 3rd party into an existing call. It can be used in the following situations:

- you want to add someone else into your existing call
- you are called by a third-party while already in a call and you want to include that person in the call

To add a third-party while in an existing call:

1. Place the original party on hold.
2. Make a new call to the new party.
3. Press “Merge” to make the conference (may be “Join” on older endpoints).

To join an incoming caller into an existing conference:

1. Accept the new call and place the original party on hold.
2. Press “Accept and Merge” to make the conference (may be “Join” on older endpoints).

With VCS version X5 (or later) and with MCU version 3.1 (or later) additional calls can be added into the conference by any party — by putting the conference call on hold, receiving or making a call, then pressing “Merge”.

A TelePresence Server version 3.0 (or later) is only supported in a deployment with a TelePresence Conductor, if the TelePresence Server is configured to run in *Remotely managed* mode.

Endpoints invited to join the Multiway conference must support the H.323 routeToMC facility message if in an H.323 call, or SIP REFER message if in a SIP call.

Centralized resources for conferencing

Prior to Multiway the ability to add new callers into a conference was only available for video endpoints that had built-in conference bridges. Multiway uses centralized MCU resources; endpoints that do not have on-board conferencing capability (such as the Cisco IP Video Phone E20) can now initiate Multiway conferences.

- The conference capacity of Multiway conferences is only limited by the MCU providing the conferencing resource or by the resources available behind the TelePresence Conductor (if the TelePresence Conductor is used).
- The use of Multiway and MultiSite is mutually exclusive; turn off MultiSite if Multiway support is required.

Intuitive conferencing

With Multiway, making conference calls is intuitive. If you make a call to a new party, having put your first party on hold, or if you receive a call when you are already in a call and choose to accept it, you are offered the choice to “Merge” all parties into a Multiway conference, as well as to swap between talking to each party one at a time.

Cisco Multiway with TelePresence Conductor

The advantages of using TelePresence Conductor and Multiway

Multiway with TelePresence Conductor adds several improvements over Multiway without TelePresence Conductor, namely:

- Each user can be provisioned with their own personal Multiway (rendezvous) alias. The advantages of this are threefold:
 - This allows users to either dial directly into the conference or be joined to the conference by dialing other members of the conference. This limits one conference alias to a conference with a specific individual.
 - Administrators can provision different levels of service to different users.
 - Each user can have a personalized Multiway conference, with unique viewtypes, PINs and conference settings.
- Greater scalability and reliability of the video network.
 - Better handling of mixed SIP and H.323 conferences.
 - Added resiliency against conference bridge failure. If a conference bridge or the network connection to it fails, TelePresence Conductor will recognize this and mark them as unusable. This allows users to redial and arrive at a separate conference bridge.
- Support for TelePresence Server, which is not available in a deployment without the TelePresence Conductor.
- TelePresence Conductor manages conference bridges and distributes load across pools of conference bridges this allows:
 - Improved load balancing across multiple conference bridges, especially for SIP (the VCS supports basic load balancing for registered H.323 MCUs, but the VCS alone does not support SIP load balancing).
 - Conference bridges can be disabled from the TelePresence Conductor's pool of conference bridges allowing them to be taken offline for upgrades or network without a loss in service.
- If a conference exceeds the capability of a single MCU the conference can be cascaded to a second MCU. This is currently not supported on a TelePresence Server.
- Greater administrative control over the conferencing resources used: allowing the administrator to decide on a per user basis:
 - The maximum number of participants allowed in a conference.
 - If content is allowed.
 - If a conference can cascade across multiple MCUs.
 - Preferentially refer users to conference bridges local to their geographic region.
 - Allow some users HD services and some users SD services.

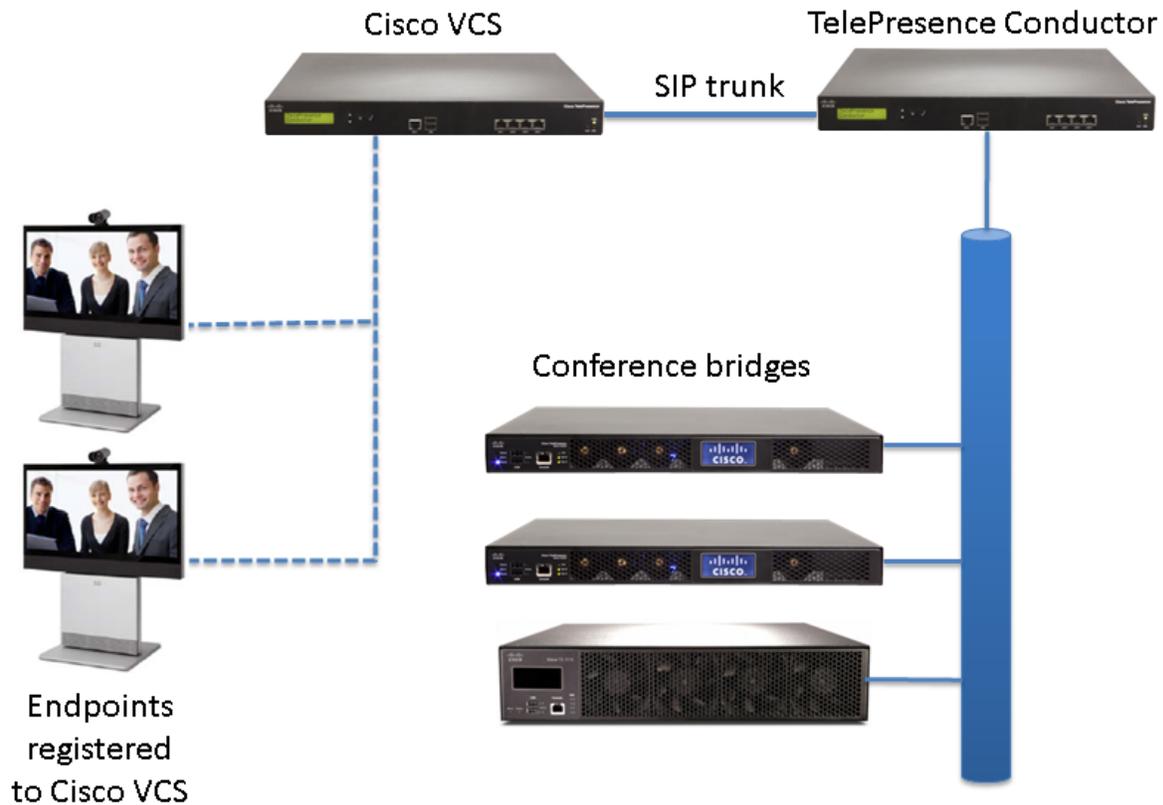
TelePresence Conductor is highly configurable, for more details of the benefits of TelePresence Conductor, see [Cisco TelePresence Conductor Administrator Guide](#) and the relevant deployment guide.

Deployment scenario

A company is currently using video for point-to-point calling but wants to support 3-way video calls and larger video conferences initiated by callers already in point-to-point calls. Although some of their larger video conference systems support this through conference bridges built into the video endpoints themselves (MultiSite), Jabber Video and EX90 users and those endpoints without the built-in conference feature

licensed do not. Furthermore conferences cannot grow beyond the capability of the multisite endpoint hosting the conference.

Multiway makes use of video conferencing more intuitive and seamless. Multiway allows point to point calls to be escalated to a multiparty conference and then allows further parties to be added to a call during the call as and when required, rather than requiring individual parties to call into a pre-setup video conference.



Prerequisites

This section describes how to configure the endpoints, the VCS, and TelePresence Conductor to use Multiway.

It assumes that TelePresence Conductor, the VCS and the conference bridges have already been configured and it is possible to dial directly into conferences using TelePresence Conductor. For details on doing this, see the relevant [Cisco TelePresence Conductor with Cisco VCS Deployment Guide](#) (B2BUA or Policy Service) depending on the deployment used. We recommend the deployment using the TelePresence Conductor's B2BUA.

It is assumed that the VCS Control is running X7.2 code (or later) and has at least the following option keys applied:

- H323-SIP interworking
- Non-traversal calls

The following software versions are required:

- The MCUs must be running code version 4.3 or later. We recommend the use of MCU version 4.4.
- The TelePresence Servers must be running code version 3.0 or later and must be configured to be running in *Remotely managed* mode. We recommend the use of TelePresence Server version 3.1.
- The VCS must be running X7.2 or later.
- The TelePresence Conductor must be running XC2.2 or later.
- If a Cisco TMS is to be used to provision endpoints with unique Multiway URIs, the supported method is to use Cisco TelePresence Management Suite Provisioning Extension in conjunction with a VCS running X7.2 or later in Provisioning Extension mode.
- The endpoints initiating a Multiway conference must have Multiway support. Endpoints that are joining a Multiway conference must have call hold and call forward functionality.

Before configuring the VCS, TelePresence Conductor and conference bridges, ensure that:

- You can access the VCS via a web browser
- You can access the TelePresence Conductor via a web browser
- The VCS is configured such that calls between video endpoints can be made.
- The VCS, TelePresence Conductor and conference bridges are configured such that it is possible to dial into a TelePresence Conductor conference.

Designing a dial plan

The things to take into consideration when designing a dial plan are:

- Each endpoint user is going to be configured with a unique Multiway alias.
- This alias needs to be routed to TelePresence Conductor using a search rule.

This is best achieved by making the unique Multiway alias configured on each users set of endpoints follow a simple pattern that can be matched against using a search rule. In this example we will be using the pattern `<username>.meet@cisco.com`.

Note that search rules targeting TelePresence Conductor should be high priority as the alias chosen should be unique to TelePresence Conductor Multiway calls and should not need to be searched for anywhere else first.

Summary of configuration process

To make Multiway operational, configuration must be carried out on:

- VCS
- TelePresence Conductor
- conference bridge (MCU or TelePresence Server)
- endpoints that need the ability to initiate a Multiway call

Configuring a search rule on the VCS

Note that this guide has a prerequisite that the VCS, TelePresence Conductor and conference bridges are configured such that an endpoint can dial already into a TelePresence Conductor conference.

Configuring a search rule towards TelePresence Conductor:

For every VCS which is intended to route Multiway calls directly to TelePresence Conductor:

1. Log in to the VCS.
2. Go to **VCS configuration > Dial plan > Search rules**.
3. Click **New**.
4. Configure the search rule fields as follows:

Name	Multiway aliases to TelePresence Conductor
Priority	Enter a value appropriate to your dial plan. TelePresence Conductor search rules should be high priority as the alias pattern chosen should not be relevant to the other search rules.
Source	<i>Any</i>
Request must be authenticated	Configure this setting according to your authentication policy.
Mode	<i>Alias pattern match</i>
Pattern type	<i>Regex</i>
Pattern string	<i>.*\meet@cisco\com</i>
Pattern behavior	<i>Leave</i>
On successful match	<i>Stop</i>
Target zone	<i>Conductor Neighbor Zone</i> if using the TelePresence Conductor's B2BUA, (Recommended deployment) or <i>Conductor Policy Service</i> if using the TelePresence Conductor as a policy service.

5. Click **Create search rule**.

Configuring a Multiway conference alias on the TelePresence Conductor

This section uses a single conference alias for all Multiway calls. For per user differentiation of Multiway conferencing capabilities and properties it is necessary to define separate aliases and separate conference templates.

1. Log in to the TelePresence Conductor.
2. Go to **Conference configuration > Conference aliases**.
3. Click **New**.
4. Configure the fields as follows:

Name	Multiway conferences
Incoming alias (must use regex)	<i>(.*\meet)cisco\com</i>

Request must be authenticated	Configure this setting according to your authentication policy
Conference name	17
Priority	5
Pattern string	(.*)\meet@<SIP domain>
Conference template	Select a conference template appropriate to your TelePresence Conductor deployment. It must be a meeting-type conference.

5. Click **Create conference alias**.

Configuring the endpoints using Cisco TMSPE

This part of the guide does not cover configuring Cisco TMSPE on the VCS and the Cisco TMS. It assumes that the template schemas are already configured with enough information to create a point to point call between two provisioned endpoints.

1. Log in to Cisco TMS.
2. Go to **Systems > Provisioning > Users**.
3. Click **Configuration templates**.
4. Click on the template you want to edit.
5. Click **Edit configurations**.
6. Configure the Multiway parameters for the endpoint as per the table below.

E20	SIP Profile MultiWayConference (if using multiple SIP Profiles, this will be "SIP Profile # MultiWayConference" where # is the profile number)	
EX series	NetworkServices MultiWay Address	{username}.meet@<SIP domain>
	NetworkServices MultiWay Protocol	Configure as per your network, generally <i>Auto</i> (the default) suffices.
MX200	NetworkServices MultiWay Address	{username}.meet@<SIP domain>
	NetworkServices MultiWay Protocol	Configure as per your network, generally <i>Auto</i> (the default) suffices.
Jabber Video	Multiway Participant URI	{username}.meet@<SIP domain>

Configuring the endpoints manually for use with TelePresence Conductor

To configure the endpoints manually for use with VCS, see section [Endpoint configuration \(with or without TelePresence Conductor\) \[p.20\]](#).

Cisco Multiway without TelePresence Conductor

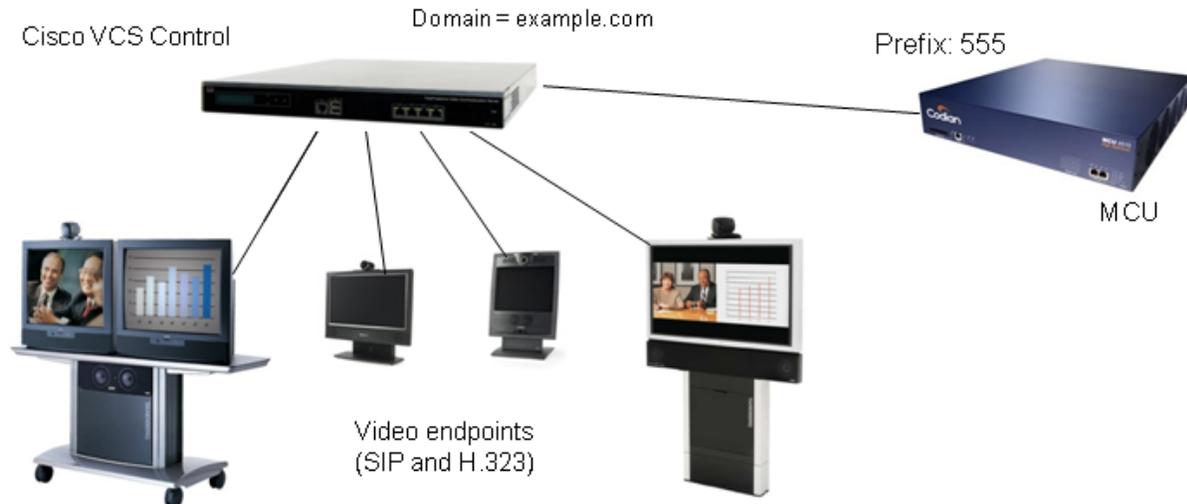
Deployment scenario

A company is currently using video for point-to-point calling but wants to support 3-way video calls and larger video conferences initiated by callers already in point-to-point calls. Although some of their larger video conference systems support this through conference bridges built into the video endpoints themselves (MultiSite), Jabber Video and EX90 users and those endpoints without the built-in conference feature licensed do not.

To implement Multiway:

- the VCS Control or VCS Expressway needs to be running X5 or later software
- a conference bridge must be available (MCU running software 3.1 or later)

Multiway makes use of video conferencing more intuitive and seamless, allowing multiple parties to be added to a call during the call as and when required, rather than requiring individual parties to call into a pre-setup video conference.



In this scenario, the MCU will be set up to receive calls with a prefix 555.

It is assumed that the VCS Control is running X5 code (or later) and has at least the following option keys applied:

- H323-SIP interworking
- Non-traversal calls
- Traversal calls (if interworking is to be used)

Summary of configuration process

To make Multiway operational, configuration must be carried out on:

- VCS
- MCU

- endpoints that need the ability to initiate a Multiway call

Prerequisites for system configuration

- VCS must be running X5 or later code
- MCU must be running code version 3.1 or later

Before configuring the VCS Control and the MCU, ensure that:

- you can access the VCS via a web browser
- the VCS is configured such that calls between video endpoints can be made

VCS configuration

To configure Multiway on the VCS:

1. Go to **Applications > Conference Factory** and configure the fields as follows:

Mode	<i>On</i>
Alias	URI of this Conference Factory (this is the Multiway ID that is configured into endpoints, that they call to initiate a Multiway conference)
Template	A template for a URI that will route calls to an MCU conference. It typically takes the form: <code><MCU service prefix><Conference prefix>%%@MCU.domain</code> for example 55577%%@example.com where %% is replaced by an incrementing number, 1, 2, ..., 9, 10, 11, ..., 99, 100..65535 etc each time Conference Factory is called. Note that the %% represents a fixed number of digits – with leading zeroes where required – based upon the length of the Number range end limit.
Number range start and end	You are recommended to leave these settings as their default values. These settings control the first and last numbers of the range used to replace %% in the Template . The maximum range is 1 to 65535. When the last value in the range has been used the system will loop round and start again at the first value in the range.

2. Click **Save**.

The Conference Factory **Alias** will be registered as a SIP registration on the VCS, so must not match any other registered devices.

Note that:

- The Conference Factory **Alias** must be a SIP URI that is routable to this VCS by every endpoint that wants to initiate a Multiway conference.
- The Conference Factory **Alias** will be registered to the VCS.
- The Conference Factory **Template** must result in a SIP URI that is routable by every endpoint that may be asked to join a conference – whether it is a SIP device or an H.323 device.
In an integrated environment where IP PBXs may be requested to join a conference, these IP PBXs must be able to route calls to this URI.

- If the **Template** is creating a URI that will be routed to an H.323 MCU, typically the name part of the URI (name@domain) must produce a numeric only ID, as H.323 MCUs typically only support E.164 dialing.
- If calls are to be made to an MCU that supports SIP, it will register <Conference prefix>%%@MCU.domain ... make sure that <Conference prefix> is unique for SIP calls, so that it does not match other dialable SIP destinations.
- Make sure that the Template cannot generate URIs which match existing endpoint URIs or other Multiway conference URIs – otherwise calls may end up ringing endpoints or joining another conference.
- The Template is not replicated across cluster nodes and must be configured independently and uniquely for each node. See the separate section on configuring Multiway in a clustered environment.

To ensure that the Conference Factory Alias is processed quickly, you are recommended to add a dedicated, high-priority search rule for this alias. The effect of this will be particularly beneficial in scenarios with many search rules and zones.

1. Go to **VCS configuration > Dial plan > Search rules**.
2. Add a new search rule with the following configuration:

Priority	1
Protocol	<i>Any</i>
Source	<i>Any</i>
Request must be authenticated	Configure this setting according to your authentication policy
Mode	<i>Alias pattern match</i>
Pattern type	<i>Exact</i>
Pattern string	Conference Factory Alias as configured under Applications > Conference Factory
Pattern behavior	<i>Leave</i>
On successful match	<i>Stop</i>
Target zone	<i>LocalZone</i>

VCS and H.323 MCU

The MCU will register its MCU service prefix (for example, 555) with the VCS as an MCU Prefix; VCS will route a call to this MCU if the call is made to an E.164 number whose first digits match the service prefix.

1. Check that the MCU is registering the MCU service prefix as an MCU Prefix registration.
When a call is made to an MCU with a dialed number:
<MCU service prefix><Conference prefix><extra digits>
The MCU will create a conference whose conference ID is:
<Conference prefix><extra digits>
The MCU will register the conference on the VCS as:
<Prefix for MCU registrations><Conference prefix><extra digits>
So that other callers being joined to the conference reach the same conference the
<Prefix for MCU registrations> must be the same as **<MCU service prefix>**.
2. As the Conference Factory **Template** must be a URI, it is necessary to convert the URI into an E.164 number when routing the call to the MCU. Converting the SIP URI into an E.164 number should be carried out in the VCS's Local Zone; it should strip the domain off any calls to

<MCU service prefix><Conference prefix><extra digits>@domain, so that the call to the MCU is placed as an E.164 dialed H.323 call.

- Go to **VCS configuration > Dial plan > Search rules**.
- At match priority 50 there should be an *Always Match* search rule, or a match pattern that allows the VCS to route calls to registered endpoints.
- To convert the dialed URI to an E.164 number, add a new search rule with the following configuration:

Priority	48
Protocol	<i>Any</i>
Source	<i>Any</i>
Request must be authenticated	Configure this setting according to your authentication policy
Mode	<i>Alias pattern match</i>
Pattern type	<i>Regex</i>
Pattern string	(<MCU service prefix><Conference prefix>\d+)@domain e.g. (55577\d+)@example.com
Pattern behavior	<i>Replace</i>
Replace string	\1
On successful match	<i>Continue</i>
Target zone	<i>LocalZone</i>

For resilience, multiple H.323 MCUs may register the same service prefix with the VCS (or VCS cluster) and VCS will ensure that all calls to a conference are presented to the same MCU. (For details on VCS's handling of multiple MCUs registered with the same MCU service prefix, see [Cisco VCS – MCU Connection Using H323 Deployment Guide](#)).

VCS and SIP MCU

SIP does not support the concept of prefix registration; instead, configure a neighbor zone with a match that looks for prefix digits, and if found routes the call to the MCU.

The <MCU service prefix> needs to be removed from the called ID (<MCU service prefix><Conference prefix><extra digits>) so that the conference ID created by the MCU will be in the format:

<Conference prefix><extra digits>

(the same as would be created by an H.323 call to the same number)

- Go to **VCS configuration > Zones > Zones**.
- Configure a neighbor zone as follows:

SIP mode	<i>On</i>
SIP port	5060
SIP transport	<i>TCP</i> (or <i>TLS</i> – depending on how MCU is configured)
H.323 mode	<i>Off</i>

Authentication policy	Configure the authentication settings according to your authentication policy
Peer IP address	IP address of MCU
Zone profile	<i>Default</i>

- Go to **VCS configuration > Dial plan > Search rules**.
- Configure a search rule as follows:

Priority	48
Protocol	<i>Any</i>
Source	<i>Any</i>
Request must be authenticated	Configure this setting according to your authentication policy
Mode	<i>Alias pattern match</i>
Pattern type	<i>Regex</i>
Pattern string	<MCU service prefix>(<Conference prefix>\d+@MCU.domain)(:.*)? e.g. 555(77\d+@example.com)(:.*)?
Pattern behavior	<i>Replace</i>
Replace string	\1
On successful match	<i>Continue</i>
Target zone	Zone created above, for example Multiway - SIP

Note:

- When a SIP conference is set up, it will register <Conference prefix>%%@MCU.domain.
- Make sure that <Conference prefix> is unique for SIP calls, so that it does not match other dialable SIP destinations.

VCS and MCU supporting both H.323 and SIP

Follow the processes above and set up the VCS to work with an H.323 MCU and also to work with a SIP MCU.

It is very important that the priority of the neighbor zone search rule for the SIP MCU is the same value (48 in this example) as the priority for the Local Zone search rule that takes the received conference URI and converts it to an E.164 number.

When the VCS is routing calls, at any particular match priority value, it routes the call in native protocol in preference to interworking the call. By setting a priority of 48 for both the SIP route to the MCU and the H.323 route to the MCU, SIP calls will be routed using the SIP neighbor zone, and only if that were to fail would the VCS try to interwork the call to H.323. Similarly H.323 calls are routed to the H.323 registration in preference to being interworked to SIP.

To support both SIP and H.323 conferencing, the MCU must be running version 3 or later software.

VCS with H.323 endpoints initiating a Multiway conference

The Multiway Conference Factory functionality is SIP based. To allow H.323 endpoints to initiate a Multiway conference:

1. Go to **VCS configuration > Protocols > Interworking**.
2. Set **H.323 <-> SIP interworking mode** to *Registered only* (or *On* is also acceptable).

SIP endpoints with SIP MCU

No special configuration is required beyond that of setting up the VCS to work with a SIP MCU.

SIP endpoints with H.323 MCU

An H.323 MCU registers with E.164 service prefixes, therefore, for a SIP call to be placed onto an H.323 MCU, the dialed URI must be converted to an E.164 number and the leading digits of that number must match the service prefix. This is carried out using the local zone match and transform defined in the [“VCS and H.323 MCU \[p.13\]”](#) section above.

To enable the VCS to perform the SIP to H.323 interworking:

1. Go to **VCS configuration > Protocols > Interworking**.
2. Set **H.323 <-> SIP interworking mode** to *Registered only* (or *On* is also acceptable).

H.323 endpoints with H.323 MCU

No special configuration is required beyond that of setting up the VCS to work with an H.323 MCU, and setting up the configuration for [“VCS with H.323 endpoints initiating a Multiway conference \[p.16\]”](#).

H.323 endpoints with SIP MCU

No special configuration is required beyond that of setting up the VCS to work with a SIP MCU, and setting up the configuration for [“VCS with H.323 endpoints initiating a Multiway conference \[p.16\]”](#).

VCS configuration: in a cluster

If the VCS is in a cluster, each VCS in the cluster will be configured with the same alias, but, each template must be unique.

The %% value is not shared between cluster peers, so each peer must generate a unique conference ID (so that if conferences are set up by different VCS peers, they cannot issue the same conference ID). This is most easily done by adding a VCS ID between the <Conference prefix> and the %%.

For example, if the <MCU service prefix> is 555 and <Conference prefix> is 77, then a 3 digit VCS ID could be used, such as 001 for the first VCS, 002 for the second. Thus VCS 1 would have a template: 55577001%%@example.com, VCS 2 would have a template 55577002%%@example.com.

VCS configuration: in a network of VCSs

If the VCS is in a network with other VCSs, one or more VCSs may have Conference Factory enabled.

Each VCS with Conference Factory enabled, whether it has the same or different alias configured, must generate a unique conference ID. Therefore, just as with clusters, every VCS that has Conference Factory

enabled must have a unique template. Use a unique VCS ID after the <Conference prefix> to ensure that conference IDs are unique for each VCS.

VCS configuration: with Lync

- Multiway is supported with Lync.
- F8.1 or later must be used for MXP endpoints.
- Version TE1.0.1 or later must be used for E20 endpoints.

Note that:

- If there is no camera attached to Lync, E20 may not connect a call to that Lync as it needs to see video traffic to complete the connection.
- Lync must have a static domain route to allow Lync to be able to route calls to the Multiway MCU's domain (the domain specified in the Conference Factory's Template). Lync needs to be able to route calls to the MCU's domain to allow Lync to make a call to the conference that is defined in the REFER which is initiated when "Merge" is pressed.

For further details see *Microsoft Lync 2010 and VCS Deployment Guide*.

VCS configuration: with CUCM

For details about the configuration needed in CUCM to support Multiway, see the "Cisco TelePresence Multiway and CUCM" appendix in the *VCS and Cisco Unified Communications Manager Deployment Guide CUCM 6.1, 7, 8 and X7.2*.

Note that the CUCM must be configured to allow dialing to the VCS domain to which Multiway aliases are registered – this is documented in the "Allow dialing to VCS domain from Cisco phones" appendix in the same deployment guide.

Cisco TelePresence MCU configuration

Although many MCUs will support basic 3 way Multiway when used as an H.323 or SIP only MCU, for full Multiway functionality the use of Cisco TelePresence MCU is strongly recommended.

Use of software 3.1 or later is also recommended.

1. Go to **Settings > Conferences, Advanced settings** and configure the following fields:

Incoming calls to unknown conferences or auto attendants	<i>Create new ad hoc conference</i>
Require H.323 gatekeeper callers to enter PIN	Ensure this option is not selected
Time to wait when setting up ad hoc conference PIN	<i><never configure PIN></i>

2. Optional (these are recommended settings, but can be configured as required for the installation – they are only applicable if the **Encryption** option is installed on the MCU):
 - a. Go to **Settings > Encryption** and configure the following fields:

Encryption status	<i>Enabled</i>
SRTP encryption	<i>Secure transports (TLS) only</i>

- b. Go to **Conferences > Templates**, select the **Top level** template and configure the following field:

Encryption	<i>Optional</i>
-------------------	-----------------

- c. Go to **Conferences > Templates** select the **Ad hoc conferences** template and configure the following field:

Encryption	<i>Optional</i>
-------------------	-----------------

H.323 settings

1. Go to **Settings > H.323** and configure the following fields:

H.323 gatekeeper usage	<i>Required</i>
H.323 gatekeeper address	Set to the IP address of the VCS.
Gatekeeper registration type	<i>MCU (standard)</i>
Ethernet port association	Select both entries (ensures that MCU uses VCS as the gatekeeper for all calls)
(Mandatory) H.323 ID to register	The H.323 ID can be any unique alphanumeric ID – it will be registered onto the VCS.
Password	Select Use password and enter the relevant Password if the VCS has authentication enabled.
Prefix for MCU registrations	This is where < Prefix for MCU registrations > is configured e.g. 555 (same as MCU service prefix).
MCU service prefix	This is where < MCU service prefix > <is configured e.g. 555 (same as Prefix for MCU registrations).
Send resource availability indications	Select this check box.
Allow numeric ID registration for conferences	Select this check box.
RAI thresholds (conferences and video ports)	Set these based on the capacity of the MCU being configured, and expected ad hoc conference sizes.

2. Configure conference settings:

- For MCU version 4.1 or later:
 - i. Go to **Conferences > Templates** and select **Ad hoc conferences**.
 - ii. Ensure that **Numeric ID registration for H.323 gatekeeper** is set to *Enabled*.
 - iii. Click **Apply changes**.
- Prior to MCU version 4.1:
 - i. Go to **Settings > Conferences, Advanced settings**.
 - ii. Select the **Register ad hoc conferences with gatekeeper** check box.
 - iii. Click **Apply changes**.

SIP settings

1. Go to **Settings > SIP** and configure the following fields:

SIP registrar usage	<i>Enabled</i>
SIP registrar domain	<domain for SIP calls> for example example.com
SIP registrar type	<i>Standard SIP</i>
Username	Username Used to create a default MCU registration and used for SIP authentication (if VCS authentication is enabled)
Password	Password used for SIP authentication (if authentication is enabled on the VCS)
Allow numeric ID registration for conferences	Select this check box
SIP proxy address	IP address of the VCS
Maximum bit rate from Microsoft OCS/LCS clients (does not apply to MCU 5300 series)	768kbit/s
Outgoing transport	<i>TCP or TLS</i> – this must match the VCS zone’s SIP transport setting
Use local certificate for outgoing connections and registrations	Select this check box

2. Configure conference settings:

- For MCU version 4.1 or later
 - i. Go to **Conferences > Templates** and select **Ad hoc conferences**.
 - ii. Ensure that **Numeric ID registration for SIP registrar** is set to *Enabled*.
 - iii. Click **Apply changes**.
- Prior to MCU version 4.1
 - i. Go to **Settings > Conferences, Advanced settings**.
 - ii. Select the **Register ad hoc conferences with SIP registrar** check box.
 - iii. Click **Apply changes**.

Note that using multiple MCUs registering the same service prefix to support a high capacity of Multiway conferences is only supported for H.323 MCUs.

When SIP conferences are being created, each MCU must have a unique service prefix to generate conferences.

Endpoint configuration (with or without TelePresence Conductor)

Endpoints that can initiate a Multiway conference

Cisco IP Video Phone E20

- E20 must be running TE1.0 or later code
- E20 supports SIP Multiway only

To configure Multiway through the menu:

1. Go to **Menu > Advanced Settings > System Configuration > SIP > Profile 1 > MultiWayConference**.
2. Configure the **MultiWayConference** setting:
 - If you are not using TelePresence Conductor, set **MultiWayConference** to the Conference Factory **Alias** as set up on the VCS.
 - If you are using TelePresence Conductor, set **MultiWayConference** to an alias matching the pattern used on the search rule to route Multiway aliases to TelePresence Conductor

Cisco TelePresence System EX Series

- EX60 and EX90 must be running TC4.2 or later code
- Cisco TelePresence Touch for EX Series must be running TC4.2 or later code

To configure Multiway through a web browser:

1. Go to **Configuration > Advanced Configuration > Network Services > Multiway > Address**.
2. Configure the **MultiWay Address** setting:
 - If you are not using TelePresence Conductor, set **MultiWay Address** to the Conference Factory **Alias** as set up on the VCS.
 - If you are using TelePresence Conductor, set **MultiWay Address** to an alias matching the pattern used on the search rule to route Multiway aliases to TelePresence Conductor

Cisco TelePresence System Profiles using C Series codecs

- C20 must be running TC3.0 or later code
- C40, C60 and C90 must be running TC4.0 or later code
- Cisco TelePresence Touch for C Series codecs must be running TC4.2 or later code

To configure Multiway through a web browser:

1. Go to **Configuration > Advanced Configuration > Network Services > Multiway > Address**.
2. Configure the **MultiWay Address** setting:
 - If you are not using TelePresence Conductor, set **MultiWay Address** to the Conference Factory **Alias** as set up on the VCS.

- If you are using TelePresence Conductor, set **MultiWay Address** to an alias matching the pattern used on the search rule to route Multiway aliases to TelePresence Conductor

Cisco TelePresence System MXP Series

- MXP must be running F8.0 or later code
- F8.1 or later is recommended for mixed H.323 and SIP Multiway conferences
- F8.2 or later is recommended if TLS is used to connect the MCU and VCS

You can configure Multiway through either a web browser or from the MXP itself.

To configure Multiway through a web browser:

1. Enter the IP address of the MXP into the web browser.
2. Go to **Endpoint configuration > General**.
3. Scroll to the bottom of page (MultiPoint Call Options).
4. Set **Multi point** to *Multiway*.
5. Configure the **MultiWay URI** setting:
 - If you are not using TelePresence Conductor, set **MultiWay URI** to the Conference Factory **Alias** as set up on the VCS.
 - If you are using TelePresence Conductor, set **MultiWay URI** to an alias matching the pattern used on the search rule to route Multiway aliases to TelePresence Conductor
6. Go to **Control panel (spanner) > General > Multipoint call options**.
7. Set **Multi point** to *Multiway*.
8. Set **Multiway URI** to the Conference Factory **Alias** as set up on the VCS.

Note that:

- If Multiway URI is configured with “sip:” at the start, e.g. sip:multiway@domain then the MXP will request the Multiway conference call using SIP.
If there is no “sip:” at the start of the Multiway URI, e.g. multiway@domain then the MXP will request the Multiway conference call using H.323. (For clarity if you want to force the Multiway conference call request to use H.323, an “h323:” can be put at the start of the Multiway URI.)
- The Multiway URI must be in the format [sip|h323:]name@domain, e.g. multiway@domain or h323:multiway@domain for H.323 or sip:multiway@domain for SIP.

Two additional advanced parameters are available on the command line interface:

- xconfiguration Multipoint MultiwayMultiprotocol: On|Off
 - *On* allows the “Merge” button when one call is H.323 and the other is SIP
 - *Off* only allows the “Merge” button if both calls are the same protocol
- xconfiguration Multipoint MultiwayStartupPeriod: 0..21
Defines the delay before audio and video is enabled after connecting to the MCU. By using a small number it reduces messages/sounds occurring as all 3 participants join the conference.

If MXP F8.1 code and SIP calling is used, either the connection from the MXP to the MCU must support secure connection, or the MXP must have **Security > Encryption** set to *Off*. If set to *On* or *Auto*, the MXP will demand a secure connection from the MCU. If any link between the MXP and MCU is TCP (rather than TLS) security cannot be supported and the Multiway join will fail for this party.

- MXP F8.0 can have encryption either set to *Off* or *Auto*
- MXP F8.1 can have encryption either set to *Off* or *Auto*, but if *Auto*, then a secure link to the MCU must be available
- MXP F8.2 and later can have encryption either set to *Off* or *Auto*

TANDBERG MXP – L Series

L Series currently do not support initiating Multiway calls.

Endpoint requirements (those that can be joined to a Multiway conference)

H.323

H.323 endpoints that can be joined into a Multiway call must support the H.323 routeToMC Facility Message.

Endpoints supporting this include:

- Cisco TelePresence System MXP (version F8.0 or later)
- Cisco TelePresence System EX Series (version TC4.2 or later)
- Cisco TelePresence System C Series (version TC4.0 or later)
- TANDBERG 150 MXP (version L6.0 or later)
- TANDBERG Classic endpoints (must have encryption disabled)
- Sony PCS G50 and Sony PCS TL30

SIP

SIP endpoints that can be joined into a Multiway call must support the SIP REFER message.

Endpoints supporting this include:

- Cisco IP Video Phone E20 (version TE 1.0 or later)
- Cisco TelePresence System MXP (version F8.0 or later)
- Cisco TelePresence Movi (version 3.1 or later)
- Cisco TelePresence System EX Series (version TC4.2 or later)
- Cisco TelePresence System C Series (version TC4.0 or later)
- most SIP endpoints (all that support REFER)

Other devices (IP PBXs, gateways and so on)

- H.323 devices need to support the routeToMC Facility Message
- SIP devices need to support the SIP REFER message
- the alias created by the Conference Factory must be an alias that is routable by the device

Note that the alias created by the Conference Factory (from the template) is in URI form (name@domain). The alias cannot be in E.164 format – devices that only support E.164 routing cannot be joined into a Multiway conference.

Optimizing the Multiway user experience on an MCU (without TelePresence Conductor)

To optimize the user experience for Multiway, you are recommended to configure the following settings on the MCUs that host the Multiway conferences.

If using a TelePresence Server in a TelePresence Conductor deployment, configure the equivalent settings on the TelePresence Conductor.

1. On the MCU, go to **Settings > Conferences**.
2. Ensure that the following options (indicated by  below) are configured as follows:

Overlaid icons: Audio participants	Select this check box
Overlaid text: Join and leave indications	Clear this check box
Overlaid logo duration	<i>Never show</i>
Conference welcome message duration	<i>Never show</i>
Time to show participant names	<i>10 seconds</i>
Don't see yourself in small panes	Select this check box
Don't duplicate in small panes	Select this check box

[Home](#) > [Settings](#) > [Conferences](#)

Conferences		H.323	SIP	Content	Encryption	Media ports	User interface	Time	Security	Upgrade	Shutdown
Conference settings											
Motion / sharpness tradeoff	Favor sharpness										
Transmitted video resolutions	Allow all resolutions										
Default bandwidth from MCU	4.00 Mbit/s										
Default bandwidth to MCU	4.00 Mbit/s										
Default view family	1 focused pane, many small panes										
Use full screen view for two participants	Disabled										
Active speaker display	Green border										
Media port reservation	Disabled										
Audio notifications	<input checked="" type="checkbox"/> Conference timing <input checked="" type="checkbox"/> Conference status <input checked="" type="checkbox"/> Join and leave indications										
Overlaid icons	<input checked="" type="checkbox"/> Important participant <input checked="" type="checkbox"/> Unsecured conferences <input checked="" type="checkbox"/> Tunneled camera control <input checked="" type="checkbox"/> Layout changes <input type="checkbox"/> Recording indicator <input checked="" type="checkbox"/> Audio participants <input type="checkbox"/> Media quality										
	Overlaid text: <input checked="" type="checkbox"/> Conference status <input checked="" type="checkbox"/> Conference timing <input checked="" type="checkbox"/> Join and leave indications <input checked="" type="checkbox"/> Text messages <input checked="" type="checkbox"/> Content channel text chat										
	Overlaid logo duration: <input type="checkbox"/> Never show										
Conference welcome message	<input type="text"/>										
Conference welcome message duration	<input type="checkbox"/> Never show no message set										
Time to show participant names	10 seconds										
Advanced settings											
Audio codecs from MCU	<input checked="" type="checkbox"/> G.711 <input checked="" type="checkbox"/> G.722 <input checked="" type="checkbox"/> G.722.1 <input checked="" type="checkbox"/> G.728 <input checked="" type="checkbox"/> G.729 <input checked="" type="checkbox"/> Polycom(R) Siren14(TM) <input checked="" type="checkbox"/> G.722.1 Annex C <input checked="" type="checkbox"/> AAC-LD <input checked="" type="checkbox"/> AAC-LC										
Audio codecs to MCU	<input checked="" type="checkbox"/> G.711 <input checked="" type="checkbox"/> G.722 <input checked="" type="checkbox"/> G.722.1 <input checked="" type="checkbox"/> G.728 <input checked="" type="checkbox"/> G.729 <input checked="" type="checkbox"/> Polycom(R) Siren14(TM) <input checked="" type="checkbox"/> G.722.1 Annex C <input checked="" type="checkbox"/> AAC-LD <input checked="" type="checkbox"/> AAC-LC										
Video codecs from MCU	<input checked="" type="checkbox"/> H.261 <input checked="" type="checkbox"/> H.263 <input checked="" type="checkbox"/> H.263+ <input checked="" type="checkbox"/> H.263 interlaced <input checked="" type="checkbox"/> H.264										
Video codecs to MCU	<input checked="" type="checkbox"/> H.261 <input checked="" type="checkbox"/> H.263 <input checked="" type="checkbox"/> H.263+ <input checked="" type="checkbox"/> H.263 interlaced <input checked="" type="checkbox"/> H.264										
ClearVision	<input type="checkbox"/>										
Video transmit size optimization	Dynamic codec and resolution										
Video resolution selection mode	Default										
Video format	NTSC-30 fps										
Maximum transmitted video packet size	1400 bytes										
Interlaced video optimization	<input checked="" type="checkbox"/>										
Video receive bit rate optimization	<input checked="" type="checkbox"/>										
Flow control on video errors	<input checked="" type="checkbox"/>										
Don't see yourself in small panes	<input checked="" type="checkbox"/>										
Don't duplicate in small panes	<input checked="" type="checkbox"/>										
Automatically make content channel important	<input checked="" type="checkbox"/>										
Loudest speaker pane placement behavior	Allow duplication of placed participants in small panes only										

Appendix 1: Troubleshooting

Problems connecting VCS Control local calls

Look at 'Search History' to check the applied transforms

Search history entries report on any searches initiated from a SETUP/ARQ/LRQ in H323 and from an INVITE/OPTIONS in SIP.

1. Go to **Status > Search history**.
The summary shows the source and destination call aliases, and whether the destination alias was found.
2. Select the relevant search attempt.
The **Search history** for that search attempt shows:
 - the incoming call's details
 - any transforms applied by admin or user policy or CPL
 - in priority order, the zones that matched the required (transformed) destination, reporting on:
 - any transforms the zone may apply
 - found or not found status
 - if not found, the error code as seen in the zone's search response repeated until a zone is found that can accept the call, or all prioritized zone matches have been attempted.
 (The search may be "not found" due to lack of bandwidth or because the search from the zone resulted in an H.323 rejection reason or a non 2xx response to a SIP request.)
3. If the search indicates:
 - Found: False
 - Reason: 480 Temporarily Not Available
 it is likely that the VCS's zone links are not correctly set up. From the command line execute:

```
xcommand DefaultLinksAdd
```

to set up the required links for VCS default zones; also check the links for other zones that have been created.

Note that:

- Each H.323 call will have 2 entries in the **Search history**:
 - the first for an ARQ to see if the endpoint can be found
 - the second for the SETUP to actually route the call
 The ARQ search does not worry about links or link bandwidth, and so if links do not exist or link bandwidth is insufficient it may still pass, even though the SETUP search will subsequently fail.
- Each SIP call will usually only have a single **Search history** entry for the SIP INVITE.

Look at 'Call History' to check how the call progressed

1. Go to **Status > Calls > History**.
The summary shows the source and destination call aliases, the call duration and whether the call is a SIP, H.323 or SIP< -->H.323 interworking call.
2. Select the relevant call attempt.

The entry will show the incoming and outgoing call leg details, the call's status and the zones that the VCS Control used to route the call.

Check for errors

Event Log

Check the VCS Event Log ([Status](#) > [Logs](#) > [Event Log](#)).

Tracing calls

Tracing calls at SIP / H.323 level

1. Log in to VCS Control web interface.
2. Go to [Maintenance](#) > [Diagnostics](#) > [Diagnostics logging](#).
3. Ensure all log levels are set to DEBUG and click **Start new log**.
4. Retry the action for which the problem occurs (such as setting up a call or similar).
5. Click **Stop logging** followed by **Download log**.

The log file will contain information related to the events triggered by the action performed in step 4.

MXP will not create a Multiway call

Whether the MXP uses H.323 signaling or SIP signaling to call the Multiway Conference Factory Alias is defined by the Multiway URI configured on the MXP.

If the Multiway URI is prefixed with "sip:", e.g. sip:multiway@domain then the MXP will always make the call to the Multiway™ Conference Factory Alias using SIP signaling; if it does not have the 'sip:' prefix e.g. multiway@domain or has an h323: prefix, e.g. h323:multiway@domain then the MXP will always make the call to the Multiway Conference Factory Alias using H.323 signaling.

The Multiway Conference Factory Alias must be in URI format name@domain as the underlying Conference Factory operation is SIP based and operates on SIP URIs.

The MXP must therefore be registered to the VCS, or be configured so that it can route calls to the VCS using the appropriate protocol (H.323 or SIP). Otherwise although "Merge" will be presented on the screen, when pressed the Multiway call will fail.

- If the Multiway URI has a "sip:" prefix, ensure that SIP calls can be made from the MXP to the VCS.
- If the Multiway URI does not have a "sip:" prefix, ensure that H.323 calls can be made from the MXP to the VCS.

Calls use less bandwidth than expected

When a Multiway conference is created, the bandwidth assigned to the original call is not released until after the merged Multiway call is established. Therefore, in limited bandwidth situations the Multiway call could connect at a lower bandwidth than the original call.

The following example explains what could happen:

1. Pipe01 is configured with a total bandwidth restriction of 900kbps.
2. EP01 in SZ01 calls EP02 in SZ02 consuming 384kbps on Pipe01.
3. EP01 puts EP02 on hold.
4. EP01 calls EP03 in SZ03 consuming another 384kbps on Pipe01.
At this time the total bandwidth used for 2 calls @ 384kbps each on Pipe 01 is 768kbps.
5. EP01 merges the calls using Multiway:
 - a. EP02 in SZ02 calls the MCU at 384kbps using Pipe02 (on the link from SZ02 to the Default Subzone).
 - b. EP03 in SZ03 calls the MCU at 384kbps using Pipe03 (on the link from SZ03 to the Default Subzone).
 - c. EP01 in SZ01 calls the MCU at 132kbps using Pipe01 (on the link from SZ01 to the Default Subzone).
132kbps was all that was left available in Pipe01 because EP01 is still called out to EP02 and EP03.
6. Post merge, the bandwidth used would be:
 - Pipe01: 132kbps
 - Pipe02: 384kbps
 - Pipe03: 384kbps

Appendix 2: Known interworking limitations without TelePresence Conductor

Multiple MCUs for capacity and resilience

For the VCS to work with multiple MCUs handling the same conference service prefixes for capacity and resilience purposes, the MCUs must be configured to support H.323 only calls.

For further details on configuring MCUs for capacity and resilience, and for details on the system operation, see [*Cisco VCS – MCU Connection Using H323 Deployment Guide*](#).

Appendix 3: Call flows for Multiway without TelePresence Conductor

Overview of Multiway call flow

1. When “Merge” Is pressed, one call is on hold and the other is live.
2. The initiator endpoint puts the live call on hold.
3. The initiator endpoint makes a call to its programmed Multiway URI.
4. This programmed Multiway URI matches a Conference Factory URI configured in a VCS.
5. The Conference Factory returns a “Temporarily moved to <URI of MCU>” message.
6. The initiator calls the <URI of MCU>.
7. Once answered the initiator tells one held party to now call <URI of MCU>.
8. Then the initiator tells the other held party to now call <URI of MCU>.
9. All 3 parties are now in a call.

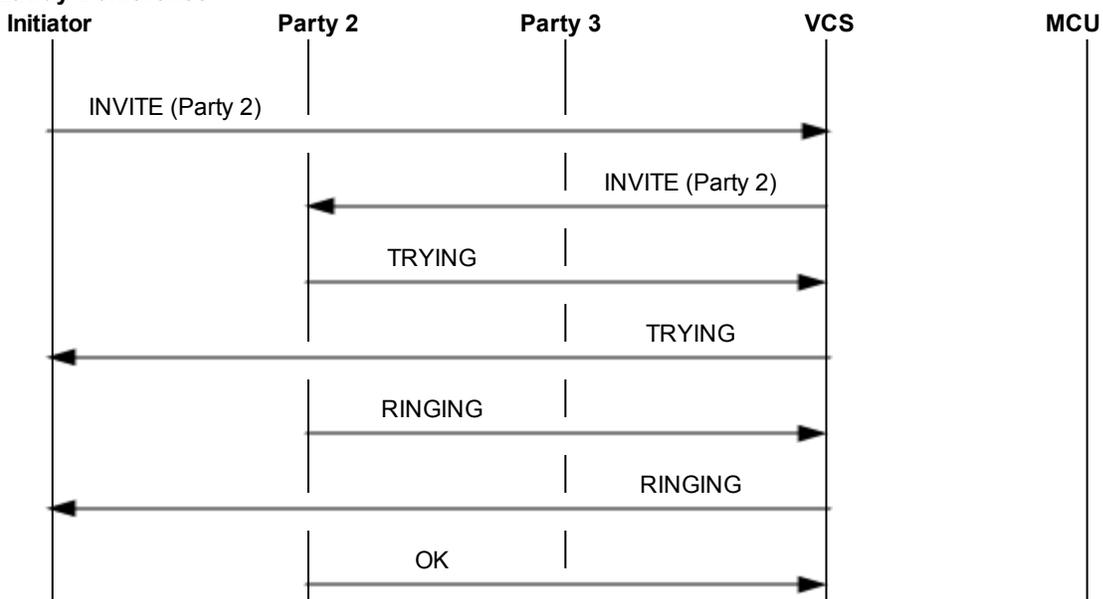
SIP

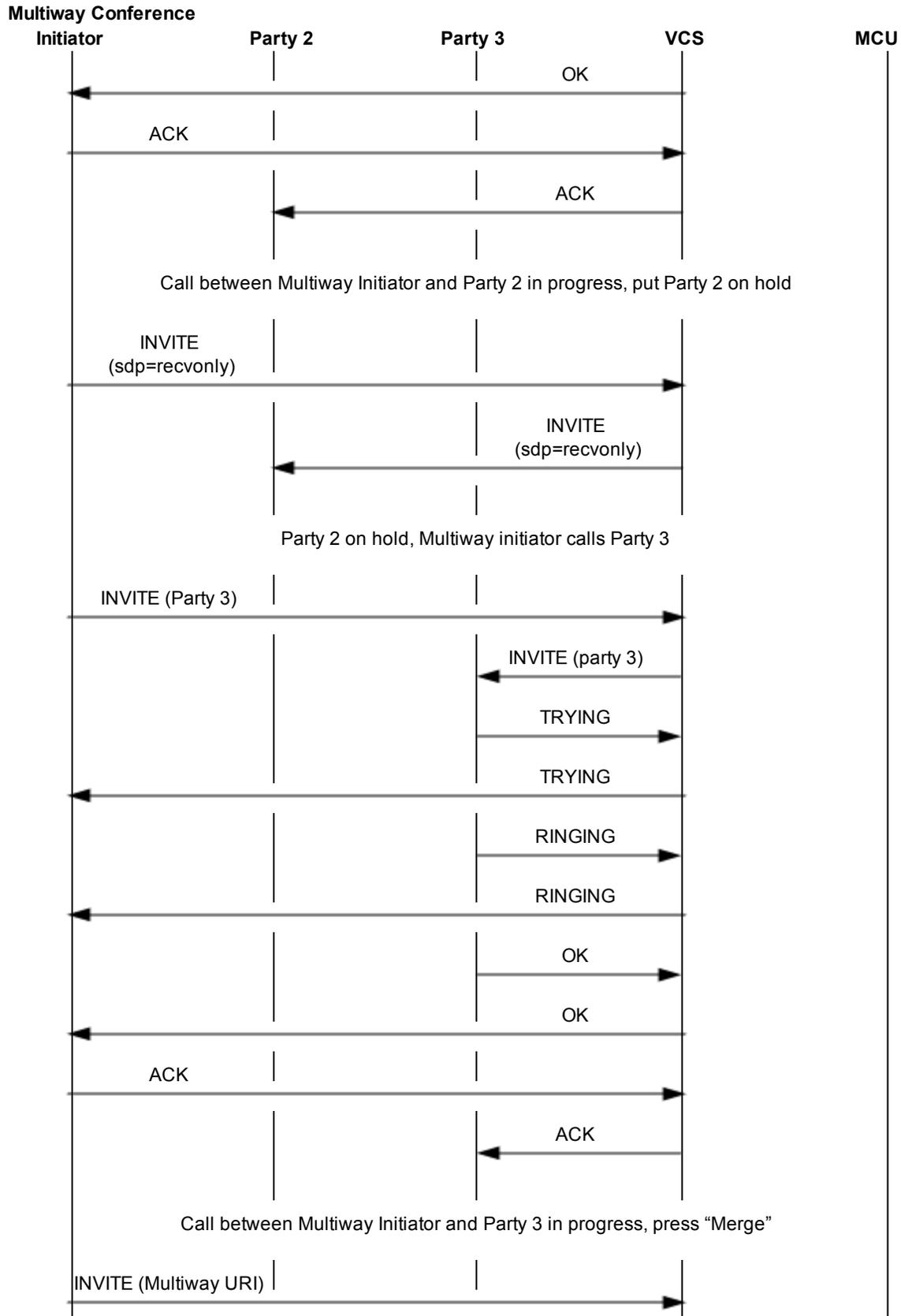
Example call flow, assuming:

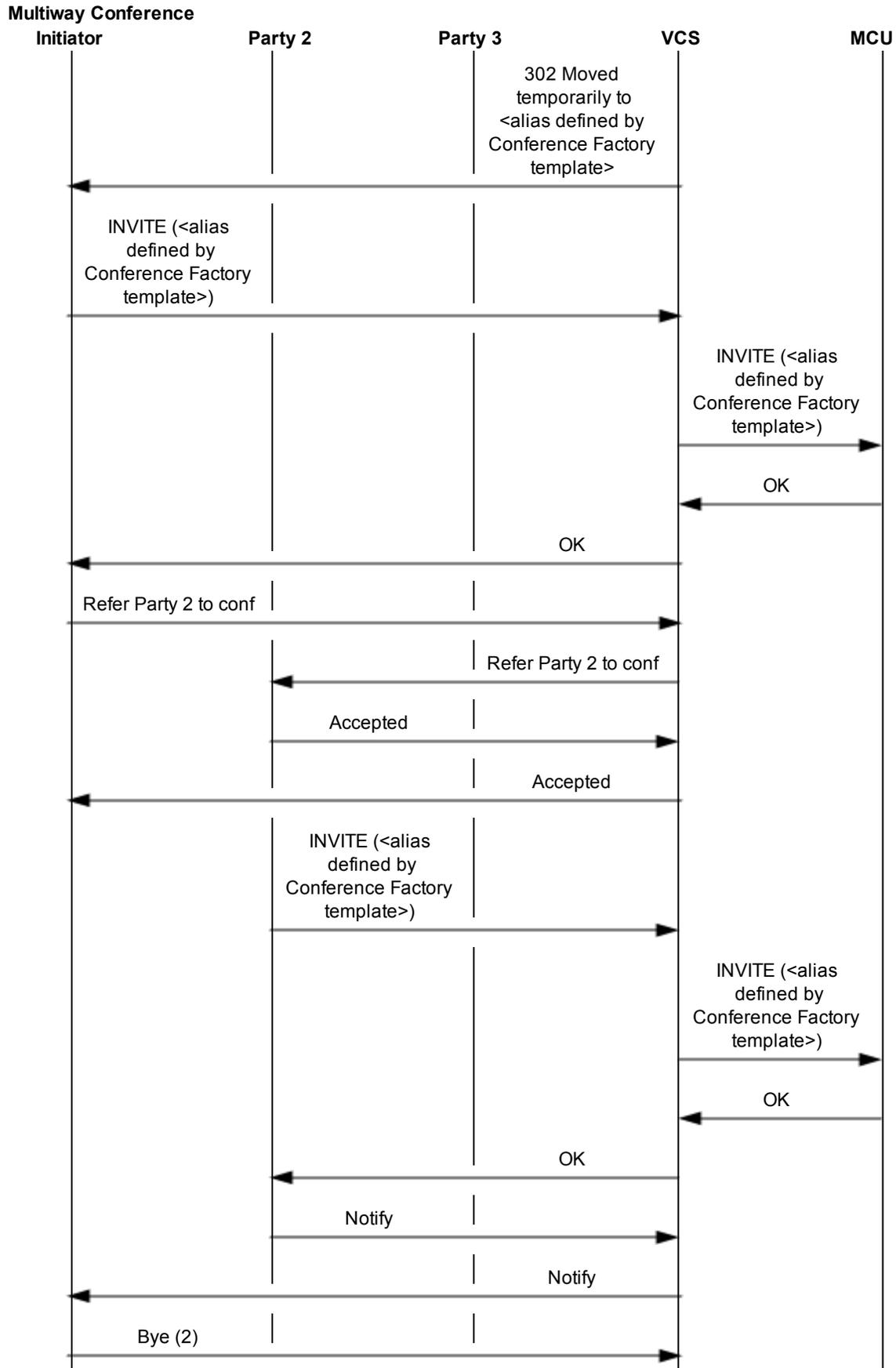
- Party 2 calls the Multiway initiator
- Multiway initiator answers call
- Multiway initiator puts Party 2 on hold
- Multiway initiator calls Party 3
- Multiway initiator presses “Merge”

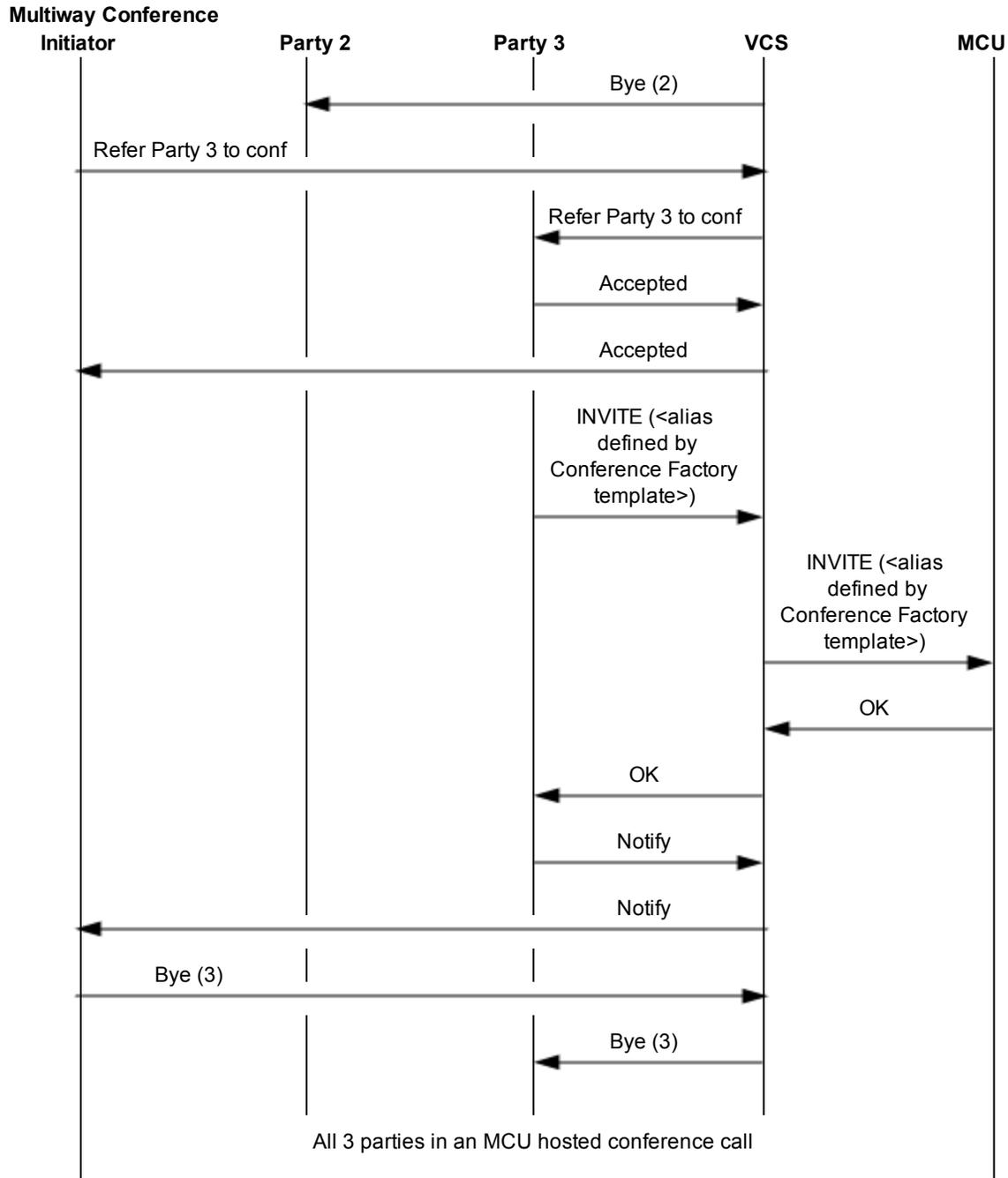
This sequence diagram shows the key SIP messaging, not all messages are shown for example no INFO (fast picture update) messages are shown.

Multiway Conference









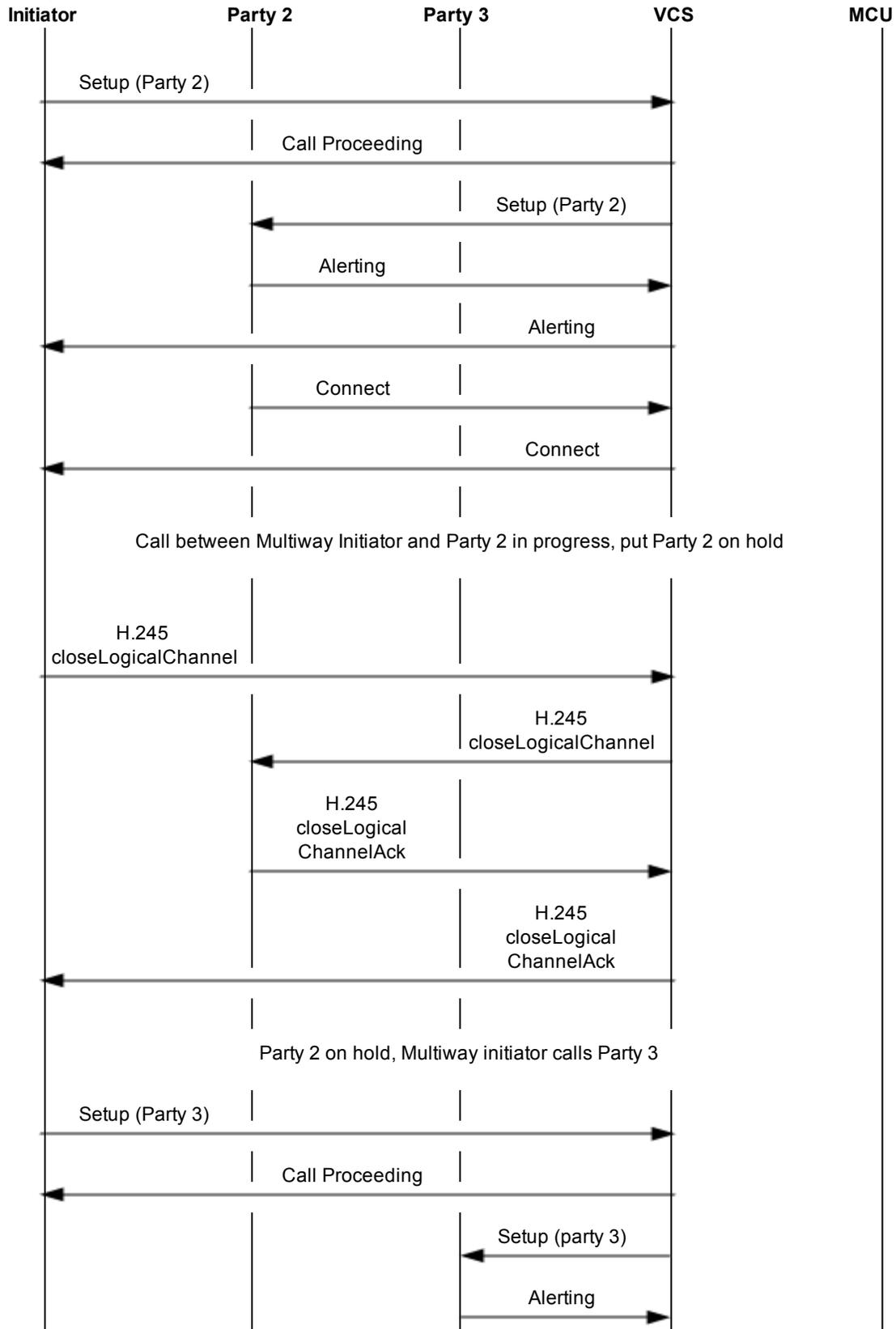
H.323

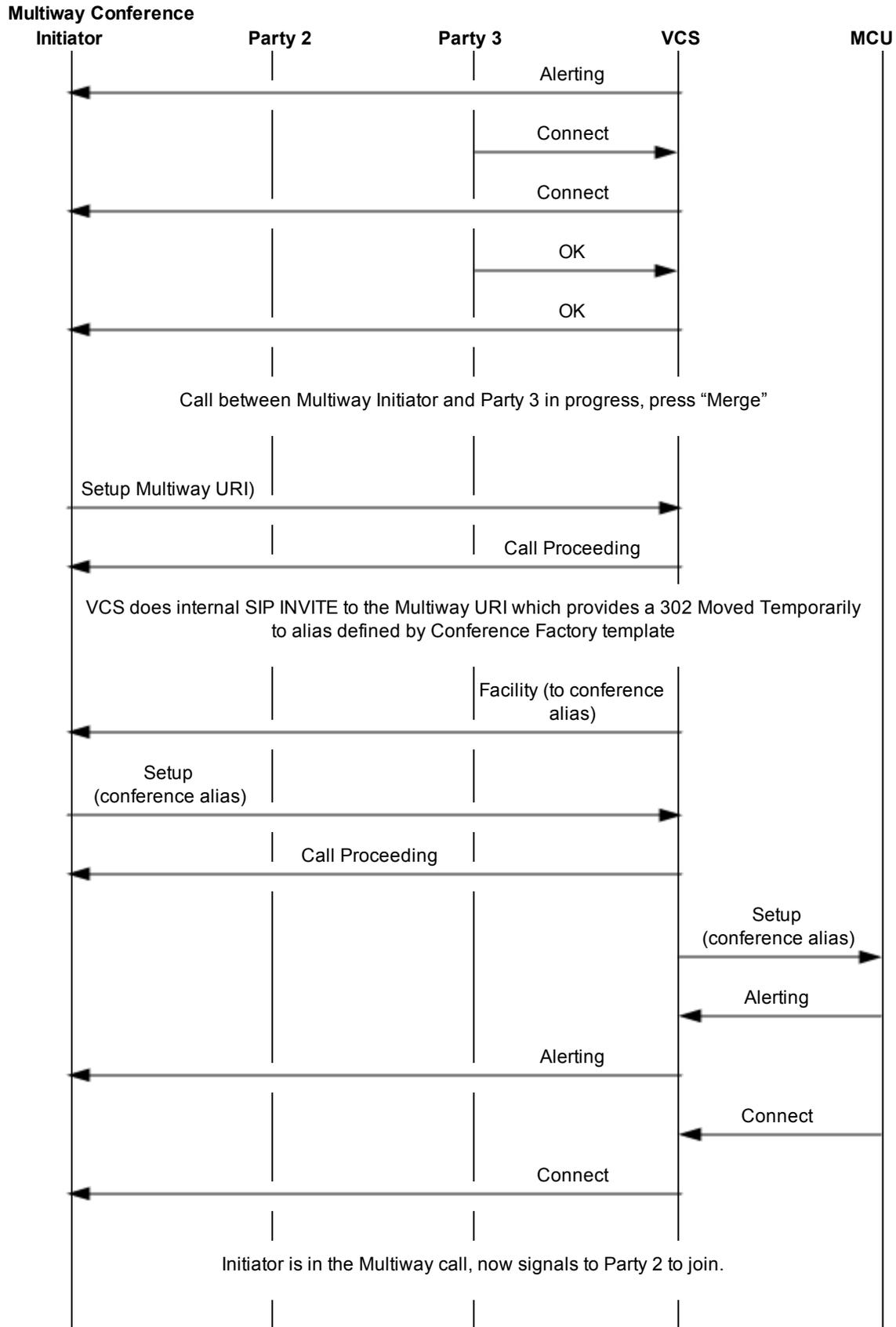
H.323 works in a similar manner, using routeCallToMC H.225 'Facility' messages to indicate which conference ID to dial:

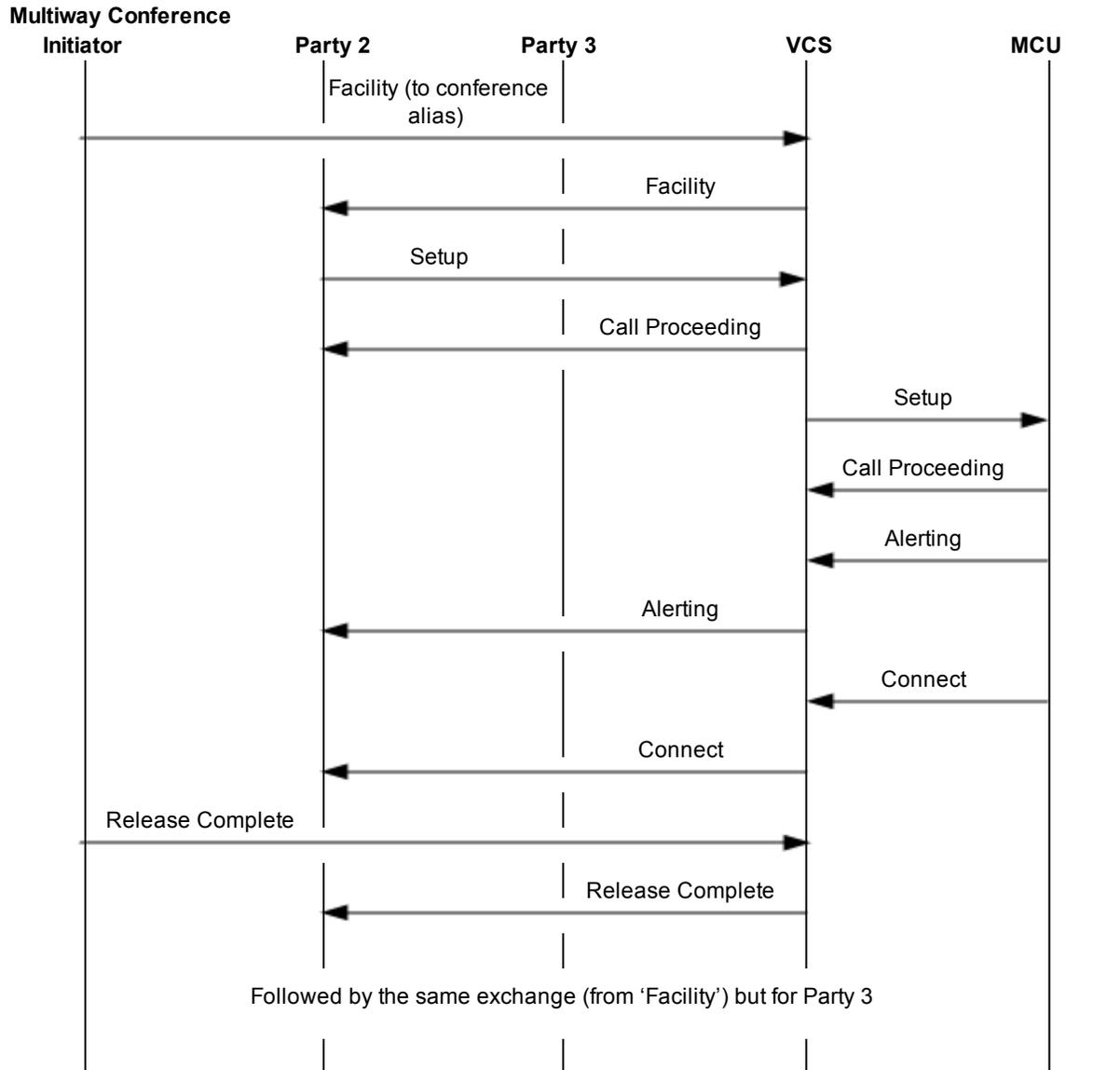
- VCS interworks H.323 calls to Conference Factory into SIP, so the Conference Factory always sees an INVITE when "Merge" is pressed, and responds with the 302 temporarily moved. VCS interworking converts the 302 into a routeCallToMC facility message.
- The initiating H.323 endpoint then sends routeCallToMC facility messages to the endpoints that are to join the Multiway conference.

The following sequence diagram shows the key H.323 messaging, although not all messages are shown.

Multiway Conference







Appendix 4: Cisco TelePresence product integration

Using Cisco TelePresence ISDN Gateway with Multiway

To join calls into a Multiway conference from an Cisco TelePresence ISDN Gateway, the Cisco TelePresence ISDN Gateway must be running version 2.1(1.22) or later code.

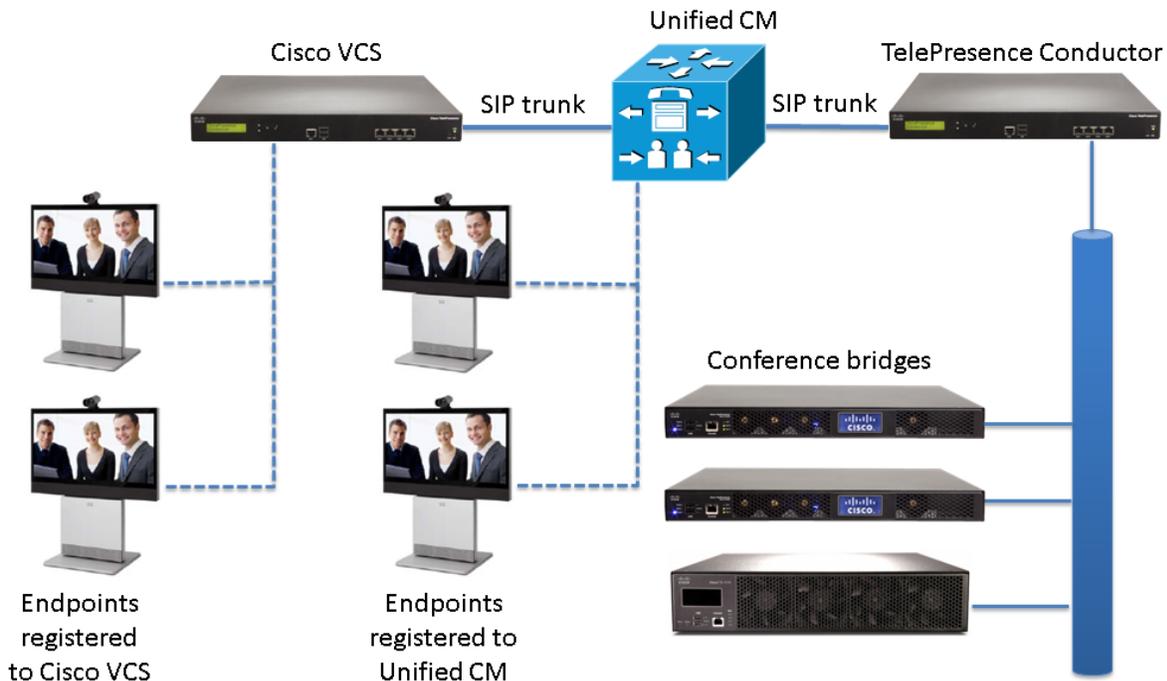
Appendix 5: Deployment with TelePresence Conductor connected to Cisco Unified CM

Deployment scenario

The following alternative deployment scenario is supported when running VCS version X7.2 or later and TelePresence Conductor version XC2.2 or later:

A VCS is connected to a Cisco Unified CM via a SIP trunk and a TelePresence Conductor with its conference bridges is connected to the Cisco Unified CM via another SIP trunk. Endpoints can be registered to both VCS and Cisco Unified CM.

Only endpoints registered to the VCS can initiate a Multiway conference. Cisco Unified CM-registered endpoints can be joined to the Multiway conference, but cannot initiate or add other participants into an existing Multiway conference. The **join/conference** button on Cisco Unified CM-registered endpoints initiates an ad hoc conference - this works independently from Multiway and cannot add participants to Multiway calls.



Conference creation process

The process of creating a Multiway conference involving the devices in the deployment scenario described above, is:

1. When an endpoint registered to the VCS creates a Multiway conference, the VCS will generate a new conference ID with a prefix (defined in ...).
2. The SIP trunk configured between the VCS and Cisco Unified CM routes calls with this prefix to Cisco Unified CM.

3. A route group in Cisco Unified CM routes the call to a SIP trunk between Cisco Unified CM and TelePresence Conductor.
4. TelePresence Conductor creates a Rendezvous conference for the Multiway conference on one of its conference bridges.

Note: for Cisco Unified CM route groups to work, the prefix must be numeric only.

Configuration tasks

For this deployment to work, the following configuration tasks must be followed:

Task 1: Checking that the VCS Conference Factory application uses numeric prefixes only

On the VCS:

1. Go to **Applications > Conference Factory**.
2. Check that the following settings have been applied

Task 2: Configuring a SIP trunk between VCS and Cisco Unified CM

Follow the [Cisco TelePresence Cisco Unified Communications Manager with Cisco VCS \(SIP Trunk\) Deployment Guide](#) to configure a SIP trunk between the VCS and Cisco Unified CM.

Task 3: Adding additional configuration on the VCS

On the VCS

1. Go to **VCS configuration > Zones > Zones** and add a neighbor zone towards Cisco Unified CM that has the following settings:

Name	<i>CUCM neighbor</i>
Type	<i>Neighbor</i>
Hop count	15
H.323 mode	<i>Off</i> (H.323 access is not required for communication with Cisco Unified CM)
SIP mode	On
SIP port	'5061' (if the SIP access port on Cisco Unified CM is not 5061, change the SIP Port value to be the same as used by Cisco Unified CM)
Transport	<i>TLS</i>
Accept proxied registrations	<i>Allow</i>
Media encryption mode	<i>Auto</i>
Authentication policy	Configure this setting according to your authentication policy.

SIP authentication trust mode	<i>Off</i>
Peer 1 address	FQDN of the Cisco Unified CM
Zone profile	<i>Cisco Unified Communications Manager or Custom</i> with Call signaling routed mode set to <i>Always</i> .

- Go to **VCS configuration > Dial plan > Search rules** and add a search rule with a conference alias that points to the Cisco Unified CM SIP trunk. This is the conference alias that should match the Multiway URI that a VCS-registered endpoint will dial. The search rule should have the following settings:

Name	<i>Route to UCM</i>
Priority	Enter a value appropriate to your dial plan.
Protocol	<i>Any</i>
Source	<i>Any</i>
Request must be authenticated	Configure this setting according to your authentication policy.
Mode	<i>Alias pattern match</i>
Pattern type	<i>Regex</i>
Pattern string	<i>(3\d{3})@vcs.domain(.*)</i>
Pattern behavior	<i>Leave</i>
On successful match	<i>Stop</i>
Target zone	<i>CUCM neighbor</i>
State	<i>Enabled</i>

Task 4: Configuring a SIP trunk between Cisco Unified CM and TelePresence Conductor

Follow the [Cisco TelePresence Conductor with Cisco Unified Communications Manager Deployment Guide](#) to configure a SIP trunk between Cisco Unified CM and TelePresence Conductor.

Task 5: Adding additional configuration on Cisco Unified CM

On the Cisco Unified CM

- Go to **Call Routing > Route/Hunt > Route Pattern** and add a route pattern with the following settings (all other settings can be left as default):

Route Pattern	<i>3xxx</i> (the conference alias that matches the Multiway URI configured on a VCS-registered endpoint)
Gateway/Route List	Select the TelePresence Conductor SIP trunk.

- Go to **Device > Device Settings > SIP Profile** and check that the SIP profile used by the VCS SIP trunk has **Redirect by application** ticked.

Task 6: Adding additional configuration on the TelePresence Conductor

On the TelePresence Conductor

1. Go to **Conference configuration > Conference templates** and add a conference template with the following settings (all other settings can be left as default):

Name	Multiway conference template
Conference type	<i>Meeting</i>
Service Preference	Select the Service Preference that contains the conference bridges used for Multiway conferences
Number of cascade ports to reserve	(Only available if the Service Preference selected is for TelePresence MCU(s)) Enter '0' to disable cascade port reservation.

2. Go to **Conference configuration > Conference aliases** and add a conference alias with the following settings:

Name	Multiway conferences
Incoming alias (must use regex)	<code>(3\d{3})@vcs.domain(.*)</code>
Conference name	<code>\1</code>
Priority	'5'
Conference template	<i>Multiway conference template</i>
Role type	<i>Participant</i>

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