Multiway™

Cisco TelePresence Deployment Guide

Cisco VCS X6

MCU version 4.2

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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 (Cisco VCS) Control and Expressway and a Cisco TelePresence MCU to use the Cisco TelePresence Multiway™ feature.

This document reflects the user interfaces presented on VCS X6 and MCU version 4.2 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 in to your existing call
- you are called by a 3rd party while already in a call and you want to include that person in the call

To add a 3rd party while in an existing call:

1. Place the original party on hold.
2. Make a new call to the new party.
3. Press “Join” to make the conference.

To join an incoming caller into an existing conference:

1. Accept the new call and place the original party on hold.
2. Press “Join” to make the conference.

With Cisco VCS version X5 (or later) and with Cisco TelePresence 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 “Join”.

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
- endpoints that do not support Multisite conferencing can initiate / join in ad hoc conferences

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 “Join” all parties into a Multiway™ conference, as well as to swap between talking to each party one at a time.

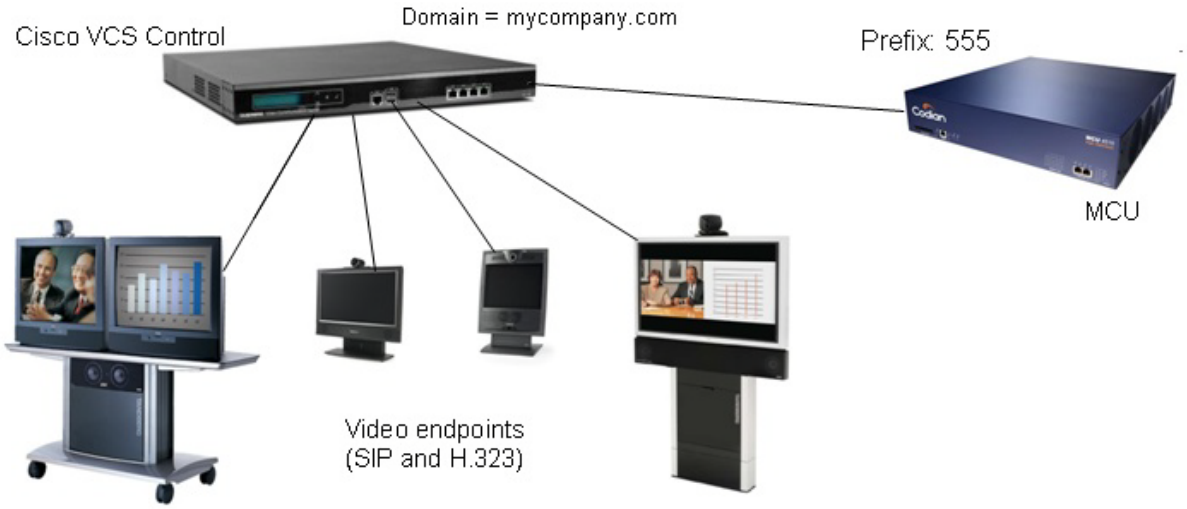
Deployment scenario

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, Movi and E20 users and those endpoints without the built-in conference feature licensed cannot.

To implement Multiway™:

- the VCS Control or VCS Expressway needs to be running X5 or later software
- a conference bridge needs to 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.



This deployment guide describes how to configure the Cisco VCS and the MCU.

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

It is assumed that the Cisco 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 will be carried out on:

- Cisco VCS
- MCU
- endpoints that need the ability to initiate a Multiway™ call

Prerequisites for system configuration

- the Cisco VCS must be running X5 or later code

- the MCU must be running code version 3.1 or later

Before configuring the Cisco 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

Cisco VCS configuration

To configure Multiway™ on the Cisco VCS:

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

Mode	*On*
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 ad hoc conference. It typically takes the form: <MCU service prefix><Conference prefix>%%@MCU.domain for example 55577%%@mycompany.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.

The Conference Factory **Alias** will be registered as a SIP registration on the Cisco VCS.

Note:

- The Conference Factory **Alias** must be a SIP URI that is routable to this Cisco VCS by every endpoint that wants to initiate a Multiway™ conference.
- The Conference Factory **Alias** will be registered to the Cisco 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 E164 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.

Cisco VCS and H.323 MCU

The MCU will register its MCU service prefix (e.g. 555) with the Cisco VCS as an MCU Prefix; Cisco VCS will route a call to this MCU if the call is made to an E164 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 an ad hoc conference whose conference ID is:

   <Conference prefix><extra digits>

The MCU will register the conference on the Cisco 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 E164 number when routing the call to the MCU. Converting the SIP URI into an E164 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 E164 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 E164 number, add a new **Search rule** with the following configuration:

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

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 MCU1.

Cisco 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)

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

[1] For details on VCS's handling of multiple MCUs registered with the same MCU service prefix, see the "VCS deployment guide - Connecting H 323 MCUs to VCSs".

SIP mode	*On*
SIP port	5060
SIP transport	*TCP* (or *TLS* – depending on how MCU is configured)
H.323 mode	*Off*
Authentication	Configure the authentication settings according to your authentication policy
Peer IP address	IP address of MCU
Zone profile	*Default*

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

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

Note: When a SIP ad hoc 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.

Cisco VCS and MCU supporting both H.323 and SIP

Follow the processes above and set up the Cisco 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 as the priority for the Local Zone search rule that takes the received conference URI and converts it to an E164 number (i.e. 48).

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

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

Cisco 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 Cisco VCS to work with a SIP MCU.

SIP endpoints with H.323 MCU

An H.323 MCU registers with E164 service prefixes, therefore, for a SIP call to be placed onto an H.323 MCU, the dialed URI must be converted to an E164 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 “Cisco VCS and H.323 MCU” section above.

To enable the Cisco 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 Cisco VCS to work with an H.323 MCU, and setting up the configuration for “Cisco VCS with H.323 endpoints initiating a Multiway™ conference”.

H.323 endpoints with SIP MCU

No special configuration is required beyond that of setting up the Cisco VCS to work with a SIP MCU, and setting up the configuration for “Cisco VCS with H.323 endpoints initiating a Multiway™ conference”.

Cisco VCS configuration: in a cluster

If the VCS is in a cluster, each VCS in the cluster must 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 ad hoc 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%%@mycompany.com**, VCS 2 would have a template **55577002%%@maycompany.com**.

Cisco 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 ad hoc 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.

Cisco VCS configuration: with OCS

OCS R1

- Multiway™ is not supported with OCS R1.
- OCS R1 does not accept being invited to join a Multiway conference — OCS invalidly rejects the INVITE to put it on hold before joining the conference with “421 Bad Extension – timer”.

OCS R2

- Multiway™ is supported with OCS R2.
- F8.1 or later must be used for MXP endpoints.
- Version TE1.0.1 or later must be used for E20 endpoints.

Note:

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

For further details about configuration with OCS, see the “Cisco VCS deployment guide for Microsoft OCS 2007 and Cisco VCS”.

Cisco VCS configuration: with CUCM

To enable CUCM registered endpoints to be joined into a Multiway™ conference set the Conference Factory **Template** to use the IP address of the VCS as the domain, for example 55501%%@<IPofVCS>.

Also configure a SIP Route pattern in CUCM that routes the domain @<IPofVCS> to the SIP trunk connecting to VCS.

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 MCUs 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	*Create new ad hoc conference*
Require H.323 gatekeeper callers to enter PIN	Ensure this option is not selected
Time to wait when setting up ad hoc conference PIN	*<never configure PIN>*

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	*Enabled*
SRTP encryption	*Secure transports (TLS) only*

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

Encryption	*Optional*

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

Encryption	*Optional*

H.323

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

H.323 gatekeeper usage	*Required*
H.323 gatekeeper address	Set to the IP address of the Cisco VCS
Gatekeeper registration type	*MCU (standard)*
Ethernet port association	Select both entries (ensures that MCU uses Cisco 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 Cisco 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

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

SIP registrar usage	*Enabled*
SIP registrar domain	<domain for SIP calls> for example mycompany.com
SIP registrar type	*Standard SIP*
Username	Username used for SIP authentication (if authentication is enabled on the VCS)
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	768kbit/s
Outgoing transport	*TCP* or *TLS* – 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: Using multiple MCUs registering the same service prefix to support a high capacity of Multiway™ ad hoc conferences is only supported for H.323 MCUs.

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

Endpoint configuration
(those that can initiate a Multiway™ conference)

Cisco IP Video Phone E20

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

To configure Multiway™ through the menu:

1. Go to **Menu > Advanced Settings > System Configuration > SIP > Profile 1 > MultiWayConference**.
2. Set **MultiWayConference** to the Conference Factory **Alias** as set up on the Cisco VCS.

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. Set the **Multiway Address** to the Conference Factory **Alias** as set up on the Cisco VCS.

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. Set the **Multiway Address** to the Conference Factory **Alias** as set up on the Cisco VCS.

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 Cisco 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. Set **Multiway URI** to the Conference Factory **Alias** as set up on the Cisco VCS.

To configure Multiway™ on the MXP itself:

1. Go to **Control panel (spanner) > General > Multipoint call options**.
2. Set **Multi point** to *Multiway*.
3. Set **Multiway URI** to the Conference Factory **Alias** as set up on the Cisco VCS.

Note:

- 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 “Join” button when one call is H.323 and the other is SIP
  - *Off* only allows the “Join” 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 TE1.0 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: 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.

Appendix 1 - Troubleshooting

Problems connecting Cisco 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.

- 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. Select **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 Cisco VCS Control used to route the call.

Check for errors

Event log

Check the event log which is accessible from the web browser: **Status > Logs > Event log**.

Real time detailed event log

To obtain a more detailed log of key events and errors, start up syslog level 1 logging and then try the call or initiate a presence action.

1. Log in to Cisco VCS Control as **admin** using an SSH or Telnet connection.
2. At the prompt type:
   `syslog 1`
3. To turn off tracing, at the prompt type:
   `syslog off`

Information displayed between typing `syslog 1` and `syslog off` contains the key events and error messages that occurred between those two times.

Tracing calls

Tracing calls at SIP / H.323 level

1. Log in to Cisco VCS Control as **admin** using an SSH or Telnet connection.
2. At the prompt type:
   `syslog 2`
3. To turn off tracing, at the prompt type:
   `syslog off`

Information displayed between typing `syslog 2` and `syslog off` contains the SIP and H.323 messaging received and sent out by the Cisco VCS.

Information displayed by `syslog 2` includes the key event and error message information reported by `syslog 1`. Viewing `syslog 1` and `syslog 2` information separately can be useful so that `syslog 1` messages are not lost within the detailed SIP / H.323 messaging.

My 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.

Note: 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 Cisco VCS, or be configured so that it can route calls to the Cisco VCS using the appropriate protocol (H.323 or SIP). Otherwise although "Join" 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 Cisco VCS.
- If the Multiway URI does not have a “sip:” prefix, ensure that H.323 calls can be made from the MXP to the Cisco VCS.

Appendix 2 – Known interworking limitations

Multiple MCUs for capacity and resilience

- For the Cisco VCS to work with multiple MCUs handling the same ad hoc 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 deployment guide – Connecting H.323 MCUs to Cisco VCSs”.

Appendix 3 – Call flows for Multiway™

Overview of Multiway™ call flow

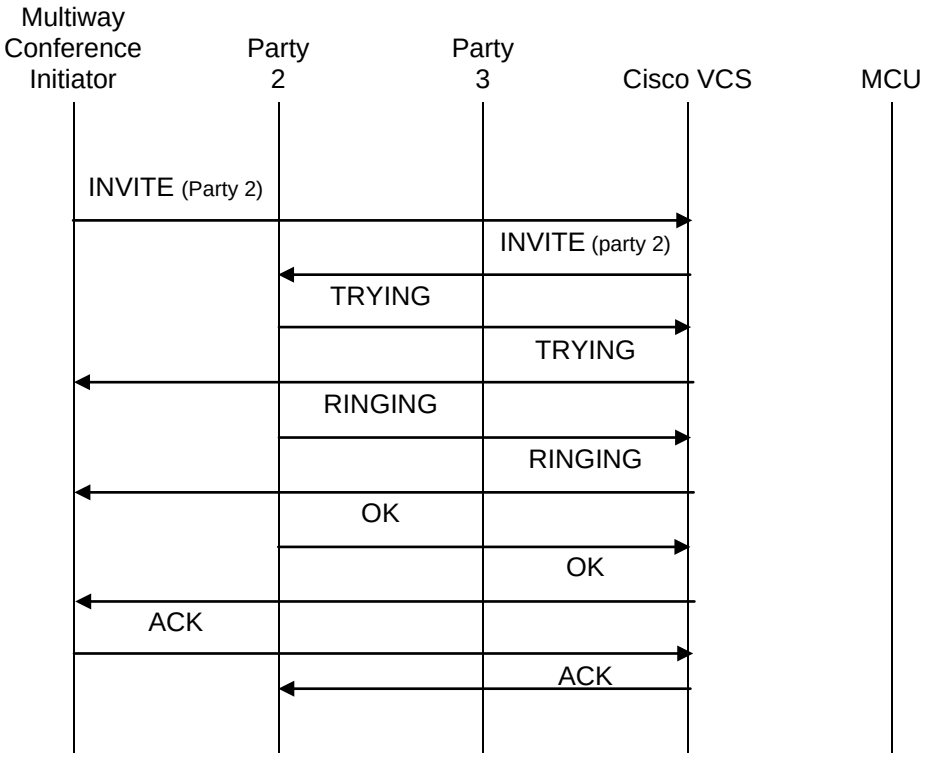
1. When “Join” Is pressed, one call is on hold and the other is live.
2. The initiator endpoint puts the live call on hold.
3. The initiator endpoint makes a call to its programmed Multiway URI.
4. This programmed Multiway URI matches a Conference Factory URI configured in a Cisco VCS.
5. The Conference 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.

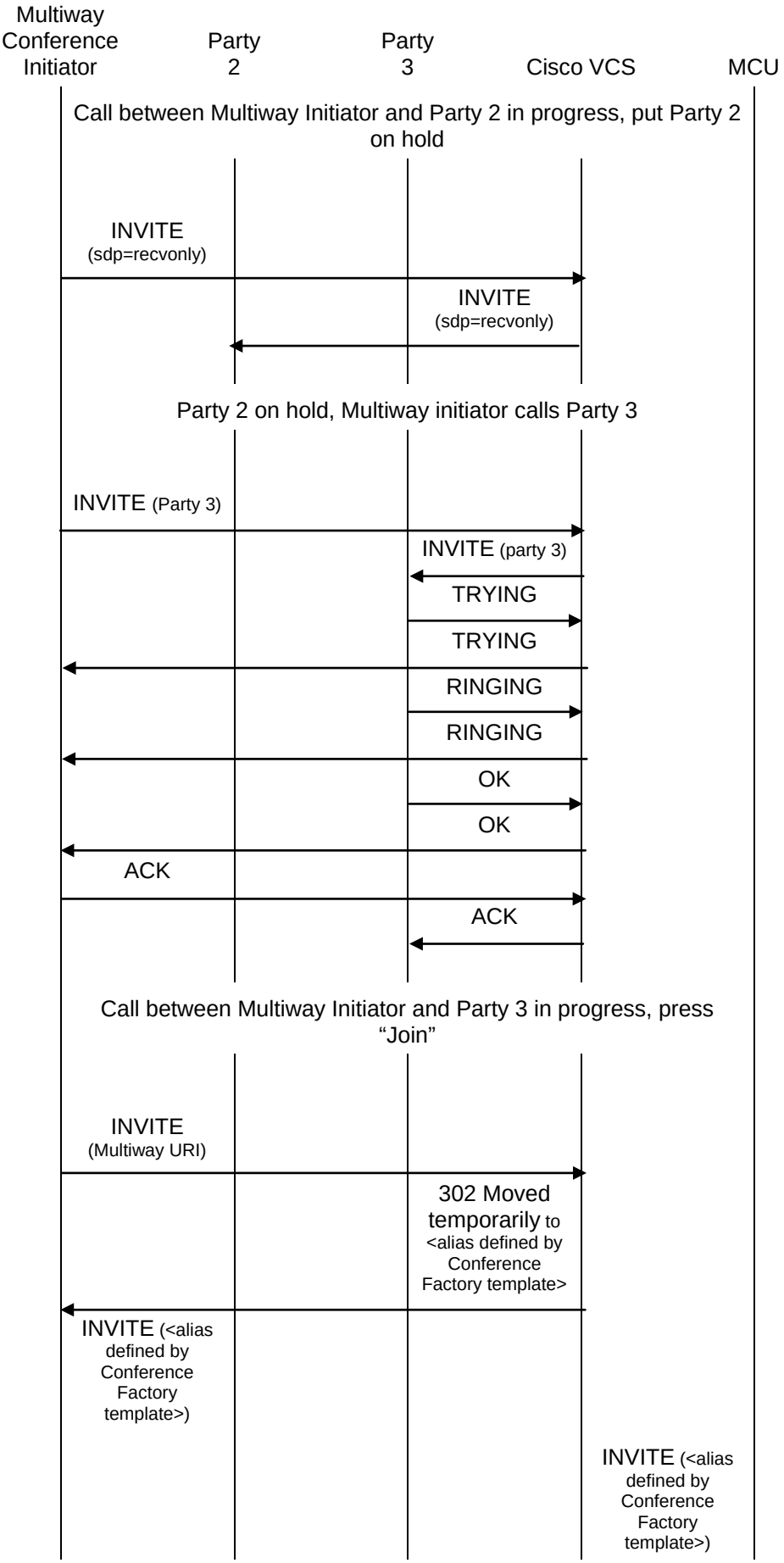
SIP

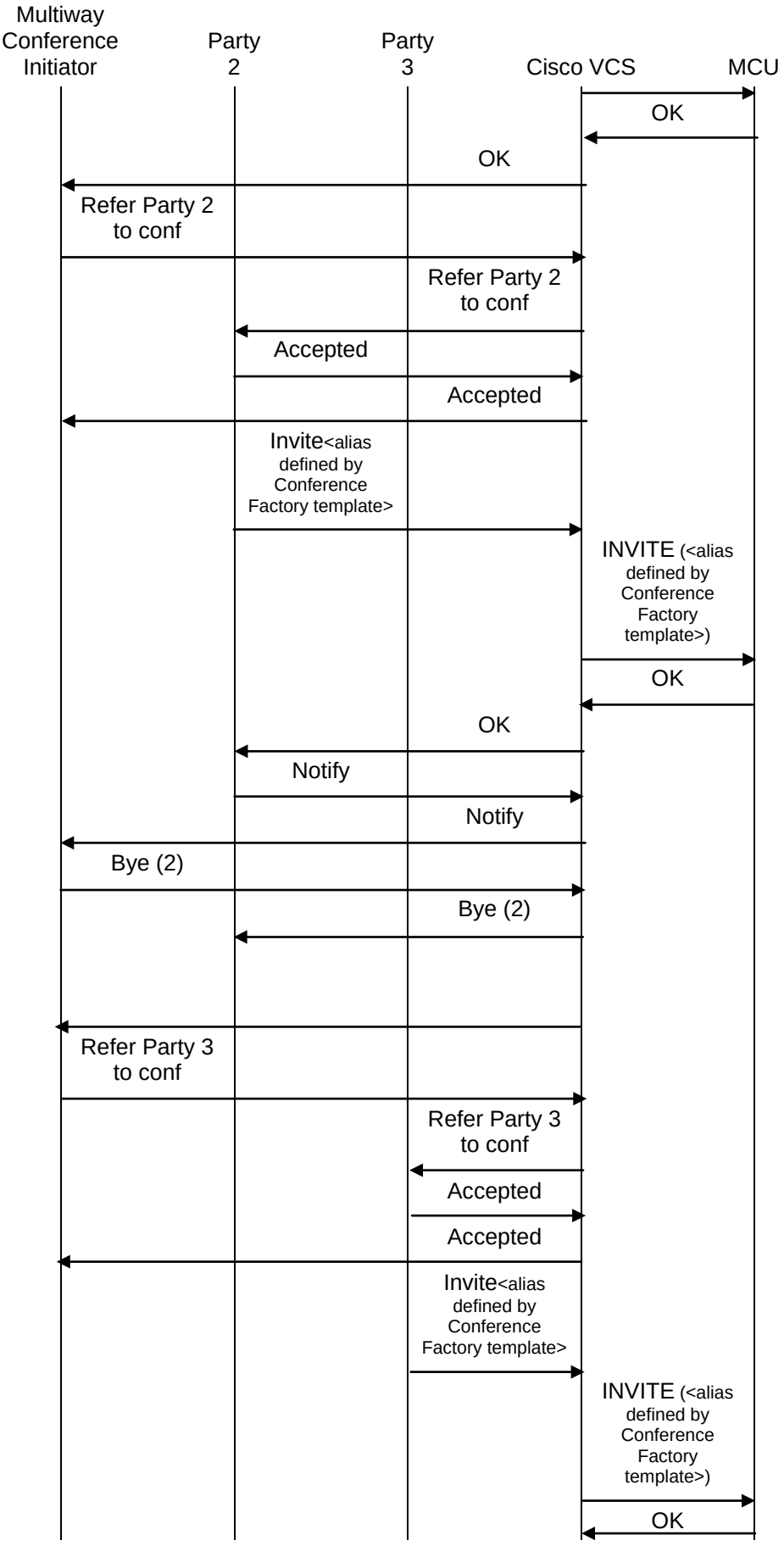
Example call flow, assuming:

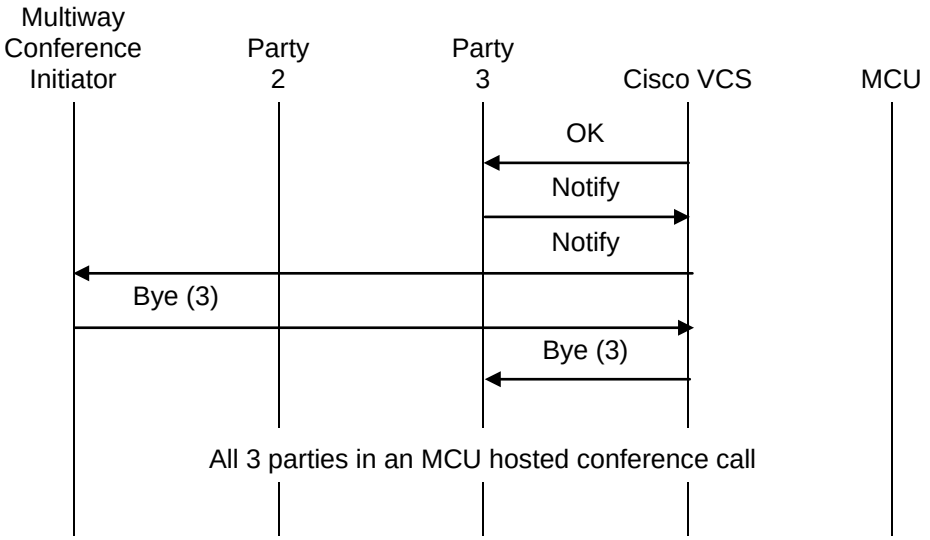
- Party 2 calls the Multiway initiator
- Multiway initiator answers call
- Multiway initiator puts Party 2 on hold
- Multiway initiator calls Party 3
- Multiway initiator presses “Join”

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 Initiator
Party 2
Party 3
Cisco VCS
MCU
Call between Multiway Initiator and Party 2 in progress, put Party 2 on hold
INVITE (sdp=recvonly)
INVITE (sdp=recvonly)
Party 2 on hold, Multiway initiator calls Party 3
INVITE (Party 3)
INVITE (party 3)
TRYING
TRYING
RINGING
RINGING
OK
OK
ACK
ACK
Call between Multiway Initiator and Party 3 in progress, press "Join"
INVITE (Multiway URI)
302 Moved temporarily to <alias defined by Conference Factory template>
INVITE (<alias defined by Conference Factory template>)
INVITE (<alias defined by Conference Factory template>)


Multiway Conference Initiator
Party 2
Party 3
Cisco VCS
MCU
OK
OK
Refer Party 2 to conf
Refer Party 2 to conf
Accepted
Accepted
Invite<alias defined by Conference Factory template>
INVITE (<alias defined by Conference Factory template>)
OK
OK
Notify
Notify
Bye (2)
Bye (2)
Refer Party 3 to conf
Refer Party 3 to conf
Accepted
Accepted
Invite<alias defined by Conference Factory template>
INVITE (<alias defined by Conference Factory template>)
OK



H.323

H.323 works in a similar manner, using routeToMC facility messages to indicate which conference ID to dial.

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

Appendix 4 – Multiway Beta on TANDBERG Gatekeeper

TANDBERG's Gatekeeper supports Multiway Beta – an early version of Multiway™. This is not compatible with Cisco VCS Multiway™, and is not compatible with modern versions of endpoint software.

When using Multiway in a mixed environment of Cisco VCSs and Gatekeepers, turn off Multiway and Call transfer on all gatekeepers:

1. Log in to the Gatekeeper web interface.
2. Go to **Gatekeeper Configuration > Services**.
3. In the Call Transfer section, clear the **Allow call transfer** check box.
4. In the Multiway section, clear the **Allow ad hoc conferencing** check box.
5. Click **Save**.

Appendix 5 – Using Cisco TelePresence Conductor and Multiway

You can use both the Cisco TelePresence Conductor and Multiway within the same deployment providing that you:

- ensure that any MCUs that are part of the Cisco TelePresence Conductor's MCU pool are not also used for Multiway. Any MCUs used by the Cisco TelePresence Conductor must be reserved for its sole use and must not be used by any other system or feature.
- ensure that your dial plan is such that there are no conflicts between the **Conference aliases** and **Call policy prefix** used by the Cisco TelePresence Conductor, and the **MCU service prefix** used by the Multiway MCU.

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