



Telephony

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Call Drops on a TLS High-Availability System

Problem In a large high-availability environment that is configured for TLS (security encryption conferencing), a conference call drops.

Possible Cause The network connection between the primary and high-availability virtual machines drops for a few minutes, and then recovers during the meeting.

Solution Participants can manually rejoin the meeting.

Call-In Issues

Problem Users hear a reorder tone before or after they dial the complete number.

Problem The "Your call cannot be completed as dialed" message is played by the annunciator.

Possible Cause You need to reconfigure your CUCM servers.

Solution In CUCM, go to the route pattern being used for Cisco Webex Meetings Server, and check the configured partition. Then go to the device you are calling from and check the configured **Calling Search Space**. Select **Call Routing > Class of Control > Calling Search Space**, go to the configured calling search space and make sure it has the partition listed configured for the route pattern for Cisco Webex Meetings Server. If the partition is set to **<None>** any device configured in Cisco Unified Communications Manager would be able to call Cisco Webex Meetings Server.

Call-Back Issues

Problem Users try to have the system call their phone numbers, but the phone does not ring and they receive an error message: "Call back failed; no answer."

Possible Cause You need to reconfigure your CUCM servers.

Solution In CUCM, go to the SIP trunks configured for Cisco Webex Meetings Server, and check the configured **Calling Search Space**. Go to your phone under **Devices**, and check the configured partition. Select **Call Routing > Class of Control > Calling Search Space**, go to the configured calling search space and make sure it has the partition listed configured for your phone.

Cannot Enter Meeting

Problem During call-in, a call terminates or there is no sound after entering the meeting ID followed by #.

Problem During call-back, a call terminates after entering **1** to join the meeting.

Possible Cause Reconfigure your CUCM servers.

Solution In CUCM, check your SIP route patterns configured for Cisco Webex Meetings Server and check the configured route partition. Go to the SIP trunks configured for the load balancers and check the configured **Rerouting Calling Search Space** and **Out-Of-Dialog Refer Calling Search Space**. Select **Call Routing > Class of Control > Calling Search Space**. Go to the configured Rerouting Calling Search Space and Out-Of-Dialog Refer Calling Search Space. Verify that they each have the partition listed configured for the SIP Route Pattern for Cisco Webex Meetings Server.

EETone Does Not Play as Expected

Problem The Entry and Exit Tone for meetings does not play as expected.

Possible Cause The EETone parameters are not set correctly, or not set to pass to telephony.

Solution Check the following:

- Data from the database:

```
selectconfid,paramname,paramvalue fromwbxmmconfparamwhereconfid=(selectconfid
frommtgconferencewheremeetingtoolsid=?)andparamname='Orion_EntryExitTone';
```

meetingtoolsid is the number (ID) for the meeting.

- Check the return value for SQL:
 - **0: beep**
 - **1: announce name**
 - **2: no tone**

If it returns NULL, the meeting failed to set the EETone.

- Check the trace log (wbxtra*.wbt) and search for **RollCall**:

```
SubscribeName=QA;SubscribePassword=pass;PhoneLines=500;SpeakingLines=500;MajorCountryCode=1;MajorAreaCode=408;PublicName=Call-in
User;TeleServerName=Personal Conference
No.;LookingupTeleServerFlag=0;TimeoutWhenNetworkBroken=1;
PublicConfKey=993967382;ICR=orion;OriginalServerCallInNumber=202074%;TollFree=1;WbxSiteID=1;WbxHostID=44;
TelAdaptorUrlList=;Subscriberaccesscode=;Participantaccesscode=;RollCall=33;SubDisName=Host
Access Code;ParDisName=Participant
```

- Beep: RollCall=32, bit0 & bit1 set 0
 - Announce name: RollCall=33, bit0 set 1
 - No tone: RollCall=34, bit1 set 1.
- Check the tahoe log(wbxtahoeas_info*log).

Normal meeting:

Info:**RetrieveConfInfo** DBThreadEvent Return: conferenceid=400663694, parentid=0, accountid=1-55, conferencemode=1, conferencetype=0, conferencestatus=0, providerid=1, isrecord=N, starttime=03/22/2019 06:09:08, endtime=03/22/2019 07:09:08, duration=0, numbcallers=0, maxcaller=0, subject=, descriptions=, disabledflag=N, hostrequiredflag=N, broadcastonlyflag=N, ondemandflag=N, recurringflag=N, recurringtype=0, recurringenddate=, recurringdayofmonth=0, recurringdayofweek=0, datemodified=03/22/2019 06:09:08, datecreated=03/22/2019 06:09:08, locationid=1, meetingid=0, siteid=1, hostid=44, accounttype=0, accessnumber=*, operatorassist=2, **entryexittone=1**,

Join Before Host Meeting (JBM):

Info:**SearchMeetingByMeetingkey** DBThreadEvent Return: o_result=3, o_desc=, o_siteName=orionf52vip, o_siteID=1, o_confID=10070, o_hostID=44, o_calloutType=2, o_priAccessNumbers=, o_bak1AccessNumbers=, o_bak2AccessNumbers=, **o_entryExitTone=1**, o_langID=1, o_mUponEntryFlag=0, o_AnonymousAttID=0, o_meetingStartTime=, o_scheduledEndTime=2019-03-22 08:00:00, o_accountID=, o_billingAccountID=, o_serviceCode=, o_subscriptionCode=, o_userLimit=0, o_meetingKey=996314212, o_ExtendParams=EnableAudioSecureAccess=1;AudioSecureAccessPwdAttempts=3, o_audioPassword=0,

Users Cannot Set EETone for Their Meetings

Problem Users cannot set the Entry and Exit Tone (EETone) at the meeting level.

Possible Cause Personal Conference meetings and Personal Room meetings do not support setting the EETone at the meeting level. These meetings follow the user-level setting for EETone.

Solution Check the meeting type. By design, Personal Conference meetings and Personal Room meetings follow user-level settings. Modern View does not support setting EETone when schedule meeting.

Caller ID Missing for Call-in Users

Problem Call-in users join meetings, and appear in the **Participants** list as Call-in User_X, instead of Call-in User_X (first name last name phone number ****).

Possible Cause This feature may not be enabled for your site.

Solution Sign in to Webex Administration and go to **Settings > Meetings**. Ensure that **Display Caller ID on Participants panel for Call-in Users joining by phone only**.

Solution If the problem persists, gather the following information and contact TAC for assistance:

- **Solution** CUCM log (must include a SIP trace)

Solution Use the CUCM RTMT tool.

- **Solution** CWMS SIP log

Solution Check the CWMS meeting log for a particular meeting, or check the Overall System log.



Note **Solution** If you need to check call ID text, you will need assistance from the CUCM TAC team.

User Calls Drop after a Failover

Problem Calls drop after a failover occurs on your high-availability system.

Possible Cause Your system has TLS enabled and uses a KPML IP phone. TAS attempts to send a subscribe SIP message to Cisco Unified Communications Manager (CUCM). The subscribe message cannot pass CUCM validation due to the change in the TAS IP address. To configure your CUCM settings, sign into Webex Administration and go to **Settings > Audio**.

Solution This is a known issue and there are no configuration changes that can fix this problem at this time. When calls are dropped because of this problem, users must rejoin the meeting by dialing back in.

Voice Activity Detection Support Issues

Problem Cisco Webex Meetings Server does not recognize the remote peer Voice Activity Detection (VAD) enable/disable condition and disables the VAD parameter by default. VAD, also known as speech activity detection or speech detection, is a technique used in speech processing in which the presence or absence of human speech is detected.

Possible Cause Cisco Webex Meetings Server does not perform the SDP-based negotiation for VAD support. Cisco Webex Meetings Server disables VAD.

Possible Cause By disabling VAD, codec bandwidth consumption does not exceed the standard bandwidth requirements for that codec. For example, the bandwidth consumption for G.711 is 64 kbps when VAD is disabled. VAD does not impact user experience in any way. When VAD is enabled, Cisco Webex Meetings Server helps to save network bandwidth depending on the active speech detected. When there is silence, Cisco Webex Meetings Server sends a special SID packet indicating the silence and stops sending packets, which helps to save network bandwidth. It starts sending audio packets again when there is voice activity detected.

Solution Cisco Webex Meetings Server does not support VAD negotiation through SDP.

Secure Teleconferencing Works Only on One Data Center

Problem Secure teleconferencing in a MDC system works only on one data center; calls to other data centers fail.

Possible Cause Cisco Unified Communications Manager (CUCM) cannot import multiple certificates with a common name and both data centers are using the global site URL in the certificate common name (the default behavior).

Solution Generate or purchase a new certificate for each data center that uses the Local Site URL as the common name:

- 1. Solution** Select **Settings > Security > Certificates > certificates on *datacentername* > Generate CSR**. The **Generate CSR** window appears.
- 2. Solution** In the **Common Name** drop down, select the local site URL of the data center.
- 3. Solution** Generate a CSR and submit to your Certificate Authority.
- 4. Solution** Install the certificate on the data centers and import it into CUCM.

