Cisco WebEx Web Conferencing

Audio Controls Release Notes

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Cisco WebEx™ Web Conferencing, provided by InterCall, allows for users with an InterCall Reservationless-Plus® audio account to control their audio conference from within the web interface.

This document includes a summary of the Audio Controls feature including known issues, limitations and recommendations.

Key Features of the Audio Controls

- Schedule and send invitations to your web conferences that include your Reservationless-Plus dial-In information without manually typing it in each time.
- Start instant meetings with your Reservationless-Plus audio information included.
- Select call-in or call-back options for attendees.
- See participants' connection status – both on the phone and on the web – using the participant list phone indicator.
- Mute and unmute audio participants' lines – participants can also perform this function for their own lines.

Audio Controls Known Issues and Limitations

SCHEDULING A MEETING

- When scheduling a meeting via the Advanced Scheduler or the Productivity Tools, such as the Microsoft Outlook add-in, you cannot specify the entry and exit tone for the teleconference. To select the entry and exit tone for your teleconference, choose ***3** on your telephone keypad or modify your account settings on www.intercallonline.com.

JOINING A MEETING

- The Join the teleconference without pressing **1** option to receive a call back is not currently supported. If you are unable to press **1** to receive a call back, you must dial directly into the teleconference.

INVITE BY PHONE

When using the Invite by Phone option, if the host wants to hang up before inviting a participant to join the teleconference, the host is not automatically returned to the main teleconference. In order to return to the main teleconference, you must select ***** on your telephone keypad.

MUTE ATTENDEES UPON ENTRY

- The mute on entry option does not automatically mute attendees' phone lines. To mute attendees upon entry to the meeting, select ***5** to mute and **#5** to unmute on your telephone keypad.

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PARTICIPANT LIST DISPLAY

- When using the Attendees Call-In option, participants will not see a phone indicator by their name. Instead, they will see Call-In User to indicate they are in the audio conference and they will see their name to indicate they are in the web conference portion. To reconcile the Call-In User with the name of the person connected to the web conference, the user must enter the Attendee ID and/or Identity Code once they have joined the conference. This code is shown on the Join Teleconference dialog box, the Info tab, and under the Meeting information menu. There is no audio prompt that asks the participant to enter the Attendee ID.
    - **Note:** The Attendee ID is available on certain bridge types. If you have any questions about this functionality, please contact InterCall technical support.
- When renaming the Call-In User in the participant list, it will only display for the individual that changed it.
- When using Cisco WebEx Event Center®, the participant list does not automatically show the phone indicator icon and status changes (i.e. muted) next to each participant's name. You must select **Refresh** in order to see the participant's phone status.

BREAKOUT SESSIONS (CISCO WEBEX TRAINING CENTER ONLY)

- When attendees use the Call-In option to join a WebEx Training Center session, the Call-In user will not be joined automatically by the integrated audio. It is recommended that all attendees identify using the Attendee ID and/or Identity Code option.
- When the presenter attempts to put attendees back into breakout sessions that have already ended, the sub-conference audio does not follow back into the previously used breakout sessions. The presenter must allow attendees to leave the breakout session and sub-conference audio before closing it. Then, each instance of the breakout session needs to be deleted before another one is created and attendees are added to it.
- There is a maximum of nine breakout sessions allowed.

AUDIO BROADCAST (EVENT CENTER ONLY)

- When scheduling an event using Reservationless-Plus and Audio Broadcast, the Mute upon entry for all participants and Entry & Exit tone options do not synchronize with your Reservationless-Plus account. You must select **Mute upon Entry** after you have started the conference or you can select ***5** on your telephone keypad. To change the Entry & Exit method, select ***3** on your telephone keypad.
- When using Reservationless-Plus integration, the audio broadcast starts as soon as the event is started online. It is recommended that the host select **Mute upon Entry** from your telephone keypad (***5**) to mute all lines during the conference to prevent any side conversations from broadcasting to all attendees.
- When the host selects to use Event Center Audio Broadcast with Other Teleconference, the service defaults to using country code 1 (or a U.S. default telephone number). You can obtain the U.S. conference number by contacting your sales representative or InterCall customer service.

Additional Recommendations

InterCall recommends using alternative audio conference services such as Operator-Assisted or Direct Event for the audio portion of your meeting if you are conducting an event or have more than 125 participants in your meeting. Reservationless-Plus is not designed for large meetings or events. Please contact your sales representative for more information on InterCall's event audio conferencing services.

Contacting InterCall

For information about InterCall's conferencing services, please contact your sales representative or visit www.intercalleurope.com.