# TSP 3.5.3

Release Notes

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# 2 Executive Summary

TSP 3.5.3 is largely focused on new, and enhanced, signaling logic that is required to allow TDM cascading between Cisco audio platforms and third-party audio bridges. The existing cascade logic for connecting with the WebEx audio platform has been improved to allow more flexibility, making the Network Based Recording (NBR) and Audio Broadcast (AB) features more reliable. Even more important, TSP 3.5.3 adds support for brokering the cascade link between the Cisco TelePresence CTMS system.

In addition to the cascading functionality added in this release, several enhancements were made to improve performance, increase reliability, and improve operations team's ability to support TSP servers.

# 3 New Features

This section covers on the new features that were added to TSP 3.5.3.

## 3.1 TelePresence OneTouch Support

Cisco WebEx Business Suite 27 allows a WebEx session to be instantly launched as a part of a Cisco TelePresence Meeting, extending the collaboration experience to the desktop. This also allows remote attendees to participate remotely in a TelePresence meeting, allowing them to view TelePresence video and data sharing. Remote attendees will participate via the traditional audio bridge from an audio standpoint, which on its own, will not allow them to hear audio from TelePresence rooms. It is for this reason that all major WebEx integrated audio types must support cascading with the TelePresence CTMS system.

TSP 3.5.3 will act as a broker, providing third-party audio systems all the dial-in information that it requires to dial into the TelePresence CTMS. The signaling used to broker this dial-in information, including phone number and DTMF join sequence, is provided by several new commands in the TSP schema.

### 3.1.1 New Command: *W2A_CalloutToTP*

The *W2A_CalloutToTP* command is brand new to the TSP schema and has the following structure:

```
<WbxTSPSchema Name="WTSPDOM" Version="1.0">
  <TransID>xxx</TransID>
  <W2A_CalloutToTP MsgID=”xxx” WbxHostName=”xxx” ExtConfID=”xxx”
  ExtSubConfID=”xxx” ExtCallID=”xxx” Privilege=”x”
  ParticipantType=”x” DTMFSequence =”xxx” EntryTone=”xxx”
  TPConnectionID=”xxx”>
    <PhoneNum>
      <CountryCode>xxx</CountryCode>
      <AreaCode>xxx</AreaCode>
      <LocalNumber>xxx</LocalNumber>
      <Extension>xxx</Extension>
    </PhoneNum>
  </W2A_CalloutToTP>
</WbxTSPSchema>
```

### *W2A_CalloutToTP* Element Attributes:

| Attribute | Required? | Description |
|---|---|---|
| MsgID | Yes | Unique identifier of the TSP message |
| WbxHostName | Yes | Host name to send the asynchronous response message to |
| ExtConfID | Yes | External conference ID on the WebEx server side |
| ExtSubConfID | Yes | External Sub Conference ID on the WebEx server side |
| ExtCallID | Yes | External ID of the Call need to be added to the sub conference |
| Privilege | Yes | Initial privilege in the conference<br>0 – speak and listen<br>1 – listen only |
| ParticipantType | Yes | 1 – subscriber (host)<br>2 – participant |
| TPConnectionID | Yes | The Unique TelePresence connection ID for the connection operation. |
| DTMFSequence | Yes | The digit sequence used for DTMF handshaking between the third-party audio conference and CTMS. This handshake occurs immediately after the phone call is connected. The max length is 2047, terminated by 0.<br>The digits-timer peer is composed by either formats:<br>T[n]H[d]<br>or<br>P[n]D[d]<br>The letter 'T' represents the timer of WebEx audio system waiting for the digits to be received from CTMS, and 'n' is the number of seconds, from 1 to 999<br>The letter 'H' represents the digits expected from CTMS, 'd' is 1 or more DTMF digits which is composed of '0'-'9', '#', '*' (max length is 32);<br>The letter 'P' represents the pause time before the WebEx audio system sends out digits to CTMS, 'n' is the number of seconds, from 1 to 999<br>The letter 'D' represents the actual digits to be sent out, 'd' is 1 or more DTMF digits composed of '0'-'9', '#', '*' (max length is 32)<br>The T-H and P-D pairs must be in pairs |
| EntryTone | Yes | Entry and Exit Tone for the call<br>0: Inherit from conference, default<br>1: beep |

| | | |
|---|---|---|
| | | 2: name announcement<br>3: silent |

***W2A_CalloutToTP* Child Elements:**

There is one main child element that contains its own sub-elements, shown as follows:

```
<PhoneNum>
      <CountryCode>xxx</CountryCode>        (optional)
      <AreaCode>xxx</AreaCode>              (optional)
      <LocalNumber>xxx</LocalNumber>
      <Extension>xxx</Extension>            (optional)
</PhoneNum>
```

### 3.1.2 Response Command: A2W_RspCalloutToTP

This is an asynchronous command that is intended to be a response to W2A_CalloutToTP. The response command has the following structure:

```
<WbxTSPSchema Name="WTSPDOM Response" Version="1.0">
  <TransID>xxx</TransID>
   <A2W_RspCancelCalloutToTP MsgID="xxx"  TPConnectionID="xxx" Result="x"
   Description="xxx">
  </A2W_RspCancelCalloutToTP>
</WbxTSPSchema>
```

***A2W_RspCalloutToTP Element Attributes***

| Attribute | Required? | Description |
|---|---|---|
| MsgID | Yes | Message ID of the original W2A_CalloutToTP command. |
| TPConnectionID | Yes | The Unique TelePresence connection ID for the connection operation. |
| Result | Yes | Response Codes<br>0: Success<br>1: Conference is not running. The WebEx audio conference which CTMS is trying to connect to does not exist. |

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|---|---|---|
| | | 2: No privilege. The WebEx audio conference is not allowed to make a call out, or the call out destination number beyond the privilege.<br>*Example:* the call out number is a international number while the WebEx audio conference has only domestic call out privileges.<br>3: Invalid parameter.<br>4: Phone number not found. The destination CTMS, call out phone is not reachable.<br>5: Busy. The phone line to CTMS is busy<br>6: No answer. The phone line to CTMS is not answering the call.<br>7: Hand shaking failed.<br>10: Internal WebEx error. |
| Description | Yes | The error description |

### 3.1.3 New Command: W2A_CancelCalloutToTP

This command tells the audio bridge to cancel its TDM cascade connection to the CTMS system. The command has the following structure:

```
<WbxTSPSchema Name="WTSPDOM" Version=“1.0”>
  <TransID>xxx</TransID>
  <W2A_CancelCalloutToTP MsgID=”xxx” WbxHostName=”xxx”
  ExtConfID=”xxx” ExtCallID=”xxx” TPConnectionID=”xxx” >
  </W2A_ CancelCalloutToTP>
</WbxTSPSchema>
```

***W2A_CancelCalloutToTP* Element Attributes:**

| Attribute | Required? | Description |
|---|---|---|
| MsgID | Yes | Unique identifier of the TSP message |
| WbxHostName | Yes | Host name to send the asynchronous response message to |
| ExtConfID | Yes | External conference ID on the WebEx server side |
| ExtCallID | Yes | External ID of the Call need to be added to the sub conference |
| TPConnectionID | Yes | The Unique TelePresence connection ID for the connection operation |

### 3.1.4 Response Command: A2W_RspCancelCalloutToTP

This is an asynchronous command that is intended to be a response to W2A_CancelCalloutToTP. The response command has the following structure:

```
<WbxTSPSchema Name="WTSPDOM Response" Version="1.0">
   <TransID>xxx</TransID>
   <A2W_RspCancelCalloutToTP MsgID="xxx"  TPConnectionID="xxx" Result="x"
   Description="xxx">
   </A2W_RspCancelCalloutToTP>
</WbxTSPSchema>
```

***A2W_RspCancelCalloutToTP* Element Attributes:**

| Attribute | Required? | Description |
|---|---|---|
| MsgID | Yes | Message ID of the original W2A_CancelCalloutToTP command. |
| TPConnectionID | Yes | The Unique TelePresence connection ID for the connection operation |
| Result | Yes | Response Codes:<br>0: Success<br>1: Failed because TelePresence connection is already established<br>2: Internal WebEx error |
| Description | Yes | The description of the error. |

## 3.2 Dynamic NBR Dial Sequence

TSP 3.5.3, allows audio integrated partners to pass the NBR/AB phone number and DTMF join-sequence to the WebEx platform at the start of, and during the course of, each WebEx Meeting. The passing of these cascading join instructions will result in an override of the sequence stored in the Telephony Domain.

The following commands have been updated to allow the passing of the new cascade join-instructions for NBR and Audio Broadcast:

### 3.2.1 Updated Command: *A2W_RspCreateConference*

The *A2W_RspCreateConference* command will receive the following new attributes, and will facilitate the override of NBR/AB join instructions at the start of the WebEx Meeting.

| Attribute | Required? | Description |
|---|---|---|
| NBRPhoneNumber | No | NBR dial-out phone number<br>Max Length: 63<br>Format (comma separated):<br>*CountryCode, AreaCode, LocalNumber*<br>Example: 1,408,9041708 |
| NBRDialSequence | No | The dial-out sequence that an end-user would enter to join the conference.<br>Max Length: 255<br>Composed by the following characters:<br>P – Pause (followed by number of seconds to pause)<br>D – Dial (followed by digits to dial)<br># – Pound (Enter)<br>Example: P10D234234234#P10D909#P0D# |

### 3.2.2 Updated Command: *A2W_NotifyConferenceChange*

The *A2W_NotifyConferenceChange* command will receive the following new attributes, and will facilitate the override of NBR/AB join instructions during the course of the WebEx Meeting.

| Attribute | Required? | Description |
|---|---|---|
| NBRPhoneNumber | No | NBR dial-out phone number, the max length is 63<br>Format (comma separated):<br>*CountryCode, AreaCode, LocalNumber*<br>Example: 1,408,9041708 |
| NBRDialSequence | No | The dial-out sequence that an end-user would enter to join the conference.<br>Composed by the following characters: (Max of 255)<br>P – Pause (followed by number of seconds to pause)<br>D – Dial (followed by digits to dial)<br># – Pound (Enter) |

| | | |
|---|---|---|
| | | Example: P10D234234234#P10D909#P0D# |

# 4 Operational Enhancements

The following changes were made to TSP 3.5.3 for the purpose of improving the manageability of TSP.

## 4.1 Log File Timestamp Improvement

One of the log files utilized to capture various inbound request details will now capture timestamps, allowing for improved troubleshooting in the event of an integration issue or outage. This is particularly valuable when troubleshooting issues that may be a partner adapter defect.

# 5 Bug Fixes

## 5.1 WebEx Node Compatibility

A small fix relating to the byte-order sent to TSP, whether as a WebEx Node component, or hosted in the WebEx cloud. This issue has been resolved which avoids a system crash.