

Enable DTMF Detection in BroadWorks 3-way/N-way Conferencing

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Introduction

This document describes how you config Cisco BroadWorks AS system level conferencing service to detect a joiner key in DTMF tone then send to other joiners.

Prerequisites

Cisco recommends that you have knowledge of these topics:

- Cisco BroadWorks System Administrator privilege
- SIP protocol and RTP protocol terminology

Requirements

There are no specific requirements for this document.

Components Used

This document is not restricted to specific software and hardware versions.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, ensure that you understand the potential impact of any command.

Background Information

Historically on one hand, it seems that clamp tones can be omitted from those join requests sent to the MS that are sent on behalf of those services that were transitioned from RFC-4240 to the Cr interface in R20.0; in order to maintain DTMF pass-through parity for those services in previous releases (detailed in legacy Cisco BroadWorks document BW-ReleaseNotes-R200.pdf - section 2.18). On the other hand allowing DTMF to pass through the bridge can sometimes result in random/spurious DTMF digits being heard on N-way / bridge calls as a result of some SIP GWs inadvertently producing RFC2833 packets from short periods of DTMF frequencies in human speech. Ultimately it seems there can be some way for the user to control whether tones are clamped on their bridge calls or not.



Caution: By default the recommended setup is set `confernceDisableClampTone` to **false**. If you are keen to enable this feature option for customer system wide, then you need to tune up some MS system resource to handle it. Besides once you `confernceDisableClampTone` to **true**, if a user makes a mistake to press a phone key, it could be incorrectly to be detected as user select a DTMF option and the MS sends that tone to all other participants in the conferencing.

Symptoms to DTMF Detection Disable in BroadWorks 3-way/N-way Conferencing

- All 3 or N parties have negotiated with the same telephone event in the SIP negotiation yet certain party could not detect and receive one other participant key in DTMF tone.

2026.10.01 11:08:20:813 AEST | Info | Sip | Sip EncodeQ 0 | 1845022 | +12409709002 | callhalf-399213:8 | 143c5db2-2993-4a7c-b99b-13009

f0216b5

udp 842 SIP Bytes OUT to 10.137.49.188:54609

INVITE sip:2409709002a@10.137.49.188:54609;ob SIP/2.0

Via:SIP/2.0/UDP 10.66.82.111;branch=z9hG4bKBroadWorks.1ecf9ns-10.137.49.188V54609-0-981507103-891032902-1790816892984-

From:<sip:9008@syd.cisco.com;user=phone>;tag=891032902-1790816892984-

To:<sip:2409709002a@syd.cisco.com;ob>;tag=tSGqysh9viBbdtxcSozvVuhz8pzwAQbI

Call-ID:BW110812984011026-1311916429@10.66.82.111

CSeq:981507103 INVITE

Contact:<sip:10.66.82.111:5060>

Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE

Supported:

Accept:application/media_control+xml,application/sdp,multipart/mixed

Max-Forwards:10

Content-Type:application/sdp

Content-Length:194

v=0

o=BroadWorks 6577 2 IN IP4 10.66.82.92

s=-

c=IN IP4 10.66.82.92

t=0 0

m=audio 10938 RTP/AVP 0 101

a=rtpmap:0 PCMU/8000

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15

a=ptime:20

2026.10.01 11:08:20:849 AEST | Info | Sip | Call Half Input Adapter 4 | 1845325 | +12409709002
| callhalf-399213:8 | 143c5db2-2993-4a7

c-b99b-13009f0216b5

udp 980 SIP Bytes IN from 10.137.49.188:54609

SIP/2.0 200 OK

Via: SIP/2.0/UDP 10.66.82.111;received=10.66.82.111;branch=z9hG4bKBroadWorks.1ecf9ns-
10.137.49.188V54609-0-981507103-891032902-1790816892984

-

Call-ID: BW110812984011026-1311916429@10.66.82.111

From: <sip:9008@syd.cisco.com;user=phone>;tag=891032902-1790816892984-

To: <sip:2409709002a@syd.cisco.com;ob>;tag=tSGqysh9viBbdtxcSozvVuhz8pzwAQbI

CSeq: 981507103 INVITE

Contact: "jianlianEntGrp1User2" <sip:2409709002a@10.137.49.188:54609;ob>

Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, INFO, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS

Supported: replaces, 100rel, timer, norefersub

Content-Type: application/sdp

Content-Length: 320

v=0

o=- 3999805696 3999805698 IN IP4 10.137.49.188

s=pjmedia

b=AS:117

t=0 0

a=X-nat:0

m=audio 4030 RTP/AVP 0 101

c=IN IP4 10.137.49.188

b=TIAS:96000

a=rtcp:4031 IN IP4 10.137.49.188

a=sendrecv

a=rtpmap:0 PCMU/8000

a=ssrc:1396647929 cname:01981b15291a0084

a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-16

- There is not packet loss, nor SRTP encrypt/decrypt discrepancy, yet certain party could not detect and receive another participant key in DTMF tone.
- Identify the CFW Control clamp tone "1 2 3 4 5 6 7 8 9 0 # *" in AS XSLog

2026.10.01 11:08:20:850 AEST | Info | Sip | Sip EncodeQ 0 | 1845348 | +12409709002 | callhalf-399213:8 | 143c5db2-2993-4a7c-b99b-13009

f0216b5

udp 460 SIP Bytes OUT to 10.137.49.188:54609

ACK sip:2409709002a@10.137.49.188:54609;ob SIP/2.0

Via:SIP/2.0/UDP 10.66.82.111;branch=z9hG4bKBroadWorks.1ecf9ns-10.137.49.188V54609-0-981507103A891032902-1790816892984-

From:<sip:9008@syd.cisco.com;user=phone>;tag=891032902-1790816892984-

To:<sip:2409709002a@syd.cisco.com;ob>;tag=tSGqysh9viBbdtxcSozvVuhz8pzwAQbI

Call-ID:BW110812984011026-1311916429@10.66.82.111

CSeq:981507103 ACK

Contact:<sip:10.66.82.111:5060>

Max-Forwards:10

Content-Length:0

2026.10.01 11:08:20:850 AEST | Info | MediaCr | MediaCrInputAdapter 0 | 1845349 | cfw-control@10.66.82.92 | localhost75850 | 143c5db2-

2993-4a7c-b99b-13009f0216b5

Processing Event: com.broadsoft.io.mediacr.events.MediaCrJoinEvent (participant - 0)
{CHCallManagerService.callhalf-399213,SessionKe

y.localhost75850}

2026.10.01 11:08:20:850 AEST | Info | MediaCr | MediaCrInputAdapter 0 | 1845350 | cfw-control@10.66.82.92 | localhost75850 | 143c5db2-

2993-4a7c-b99b-13009f0216b5

409 CFW Bytes OUT to 10.66.82.92:7575

CFW 230 CONTROL

Content-Type: application/msc-mixer/1.0+xml

Control-Package: msc-mixer/1.0

Content-Length: 292

<mscmixer version="1.0" xmlns="urn:ietf:params:xml:ns:msc-mixer">

<join id1="1999115148-1790816900797-:359393922" id2="1_51a417e0-a6c8-4fd9-b285-a7b496773709_1@10.66.82.111">

<stream media="audio" direction="sendrecv">

<clamp tones="0 1 2 3 4 5 6 7 8 9 # *"/>

</stream>

</join>

</mscmixer>

2026.10.01 11:08:20:850 AEST | Info | CallP | Call Half Input Adapter 4 | 1845351 | Media Manager | Media Conference | +12409709002 |

jianlianEntGrp1 | callhalf-399213 | 143c5db2-2993-4a7c-b99b-13009f0216b5

com.broadsoft.events.media.ConferenceParticipantAddedEvent(callhalf-399213:8)

processed by - com.broadsoft.services.conference.ConferenceServiceInstance

2026.10.01 11:08:20:850 AEST | Info | Sip | Call Half Input Adapter 4 | 1845352 | +12409709002 | callhalf-399213:8 | 143c5db2-2993-4a7

c-b99b-13009f0216b5

udp 428 SIP Bytes IN from 10.137.49.188:54609

SIP/2.0 500 Unhandled by dialog usages

Via: SIP/2.0/UDP 10.66.82.111;received=10.66.82.111;branch=z9hG4bKBroadWorks.1ecf9ns-10.137.49.188V54609-0-981507104-891032902-1790816892984

-

Call-ID: [BW110812984011026-1311916429@10.66.82.111](tel:BW110812984011026-1311916429@10.66.82.111)

-

Call-ID: BW110812984011026-1311916429@10.66.82.111

From: <sip:9008@syd.cisco.com;user=phone>;tag=891032902-1790816892984-

To: <sip:2409709002a@syd.cisco.com;ob>;tag=tSGqysh9viBbdtxcSozvVuhz8pzwAQbI

CSeq: 981507104 INFO

Content-Length: 0

Information TAC Needs for Initial Triage

- XSLog analyse of the AS XSLog to verify all participants have negotiated with the same telephone-event.
- Network topology.
- Packet capture of that DTMF pressing participant phone device.
- Output of these commands when the 3-way/N-way conferencing DTMF undetect from one or more of the other parties when a participant has pressed in the DTMF tone.

AS_CLI/System/CallP/DTMFTransmission> **get**

transmissionMethod = signaling

signalingContentType = application/dtmf-relay

MS_CLI/Applications/MediaStreaming/Interfaces/SIP> **get**

listeningPort = 5060

listeningPortTLS = 5061

retransmissionDelay200OKInMilliseconds = 500

maxRetransmission200OK = 3

t1 = 500

t2 = 4000

supportTCP = false

supportUDP = true

supportTLS = false

tlsClientAuthenticationRequired = false

tlsHandshakeTimeoutInMilliseconds = 1000

acceptDTMFRelay = true

routeAdvanceTimeoutInSeconds = 5

sessionAuditIntervalInSeconds = 1200

enableStrictSdpValidation = false

AS_CLI/SubscriberMgmt/Policy/CallProcessing/Conferencing> **get**

maxConferenceParties = 6

conferenceURI = labconfAS@syd.cisco.com

conferenceDisableClampTones = false

Configuration Step

Post troubleshooting to rule out other factors like SIP negotiation and packet drop, and other misconfigurations to accept DTMF Relay in the 3-way/N-way conferencing bridge, you check AS system level parameter **conferenceDisableClampTones** has been set to **false**. Change this System level parameter **conferenceDisableClapTones** to **true** then the DTMF detection can be enabled in 3-way/N-way conferencing.



Note: In certain extreme situations, you need to restartbw AS (one at a time in a cluster) to overcome the possible delay due to CLI updates the configuration file but does not immediately update the running cache.

Related Information

[Cisco BroadWorks CR Interface Enhancements DTMF Relay Support](#)