CUCM 8.0.3

BT SIP Trunk Configuration Guide

This document covers service specific configuration required for interoperability with the BT SIP Trunk service. Anything which could be considered as normal CUCM configuration (such as dial plan, device pools etc.) are not within the scope of this document unless a specific configuration parameter is required in order to ensure the greatest level of interoperability with the BT SIP Trunk service.

This configuration guidance can be split into four distinct areas:

1. Service parameters required for correct SIP behaviour (with regards to the BT SIP Trunk platform)
2. SIP Trunk configuration specific parameters
3. Hardware resources required for correct interaction of CUCM and the BT SIP Trunk platform
4. End device specific parameters required for correct operation

It should be further noted that the document reflects the configuration of the test environment used to execute the BT SIP Trunk platform compliance testing. Only configuration which is non-default will be covered.

1 Service Parameters

Cluster-wide parameters for the "**Cisco CallManager**" service:

Parameter	Default	New Setting
Transfer On-Hook Enabled	False	True
Retry Count for SIP Invite	6	3
SIP Trying Timer	500	300
SIP Min-SE Value	1800	450
Fail Call Over SIP Trunk if MTP Allocation Fails	False	True

More detailed information on these changes is as follows:

Transfer On-Hook Enabled

This parameter is set to "**True**" in order to enable the blind-transfer capabilities that were required during compliance testing.

Retry Count for SIP Invite

This parameter was reduced to "**3**" to decrease the number of SIP:INVITE messages sent by CUCM to the BT SIP Trunk platform SBC before failing over another platform SBC in a timely fashion (4.5 seconds rather than the default of 32 seconds).

SIP Trying Timer

This parameter was reduced to "**300ms**" to decrease the amount of time before re-sending a SIP:INVITE message to the BT SIP Trunk platform SBC so that failover could occur in a timely fashion (4.5 seconds rather than the default of 32 seconds).

SIP Min-SE Value

By default, a SIP:INVITE message sent from the BT SIP Trunk platform to CUCM had a Minimum Session Expiry (Min-SE) value set to 450ms. CUCM's default value is 1800ms. If CUCM receives a message with a Min-SE value lower than the configured service parameter it will reject the message with a SIP:422 – Session Time Too Small error. In order to prevent this, the service parameter was reduced to "**450ms**" to accommodate the requests coming from the platform.

Fail Call Over SIP Trunk if MTP Allocation Fails

By default if MTP is required for a connection CUCM will attempt to allocate one from its configured resources, however if one is unavailable it will still allow the call to proceed without allocating the MTP. For interaction with the BT SIP Trunk on outgoing (i.e. CUCM originated) calls, MTP is required to force Early-Offer and Early-Media such that the platform can play in-band network message announcements (e.g. Caller-Waiting notification) before the call is connected. As such, MTP must be allocated to guarantee that network messages will be received by the caller and so if MTP resource is unavailable the call should not proceed. Hence the parameter is set to "**True**".

2 SIP Trunk Configuration Specific Parameters

The configuration of a SIP Trunk in CUCM can be split into three distinct categories:

1. SIP Profile
2. SIP Trunk Security Profile
3. SIP Trunk Configuration

SIP Profile Configuration

A default SIP Profile is used.

SIP Trunk Security Profile Configuration

A default SIP Trunk Security Profile is mostly used, the core configuration parameters are:

Parameter	Default Setting	New Setting
Device Security Mode	Non Secure	Non Secure
Incoming Transport Type	TCP+UDP	TCP+UDP
Outgoing Transport Type	**TCP**	**UDP**
Enable Digest Authentication	Unchecked	Unchecked
Nonce Validity Time (mins)	600	600
X.509 Subject Name	<Blank>	<Blank>
Incoming Port	5060	5060
Enable Application Level Authorization	Unchecked	Unchecked
Accept Presence Subscription	Unchecked	Unchecked
Accept Out-of-Dialog REFER	Unchecked	Unchecked
Accept Unsolicited Notification	Unchecked	Unchecked
Accept Replaces Header	Unchecked	Unchecked
Transmit Security Status	Unchecked	Unchecked

The only real change from default parameters is the Outgoing Transport Type – this is set from TCP to UDP, which is the default platform access mechanism. This should be set dependent on the allocated access mechanism when the platform link is provisioned (implementation specific).

All other parameters essentially remain at default.

SIP Trunk Configuration

SIP Trunk specific configuration parameters that need to be changed from default are:

Parameter	Default Setting	New Setting
Media Termination Point Required	Unchecked	Checked
Call Routing Information – Asserted-Type	Default	PAI
MTP Preferred Originating Codec	711ulaw	G729/G729a
SIP Trunk Security Profile	-- Not Selected --	<Pre-configured profile>
SIP Profile	-- Not Selected --	<Pre-configured profile>
DTMF Signalling Method	No Preference	RFC 2833

More detailed information on these changes is as follows:

Media Termination Point Required

In order to play back network messages the platform requires CUCM to support Early Offer so that it can negotiate Early Media for in-band network message playback. By default CUCM does not do this, however forcing MTP to "**On**" configures CUCM to do that.

Call Routing Information – Asserted-Type

By default the BT SIP Trunk platform requires that the Privacy-Asserted-ID field in SIP messaging is populated to correctly populate caller identity, particularly for emergency calls. Therefore to ensure that CUCM populates this field this should be set from Default to "**PAI**".

MTP Preferred Originating Codec

CUCM version 8.0 and below are limited to offering a single codec when performing Early Offer. As the "MTP Required" was forced to on this therefore requires that a codec is specified. Due to the fact that some other customers on the platform may restrict calls to use the G.729a codec to save on bandwidth if the default codec of G.711µ-law is left on then calls could potentially fail (because only a single codec is offered). Therefore to ensure that all calls successfully proceed the lowest common denominator codec must be chosen and this parameter set to "**G729/G729a**".

SIP Trunk Security Profile / SIP Profile

Both a SIP Trunk Security profile and a SIP Profile need to be configured to reflect the BT SIP Trunk platform requirements (as detailed previously). These then need to be applied to the specific trunks when configured.

DTMF Signalling Method

The BT SIP Trunk platform requires that all DTMF signalling uses the RFC 2833 specified mechanism. Therefore to ensure that CUCM adheres to this requirement the SIP Trunk should be configured accordingly and the DTMF Signalling Method changed from No Preference to "**RFC 2833**".

3 Hardware Resources Required for the Correct Operation of CUCM

In order to successfully place outgoing calls across the BT SIP Trunk platform from CUCM, MTP is required to be configured. The use of software MTP does not scale well therefore hardware MTP must be used. Cisco IOS Enhanced Software MTP is sufficient for SIP Trunking purposes.

These hardware-based MTP devices (of which Cisco IOS Enhanced Software MTP is classed as hardware-based), must be assigned to the media resources of the SIP Trunk, and not device endpoints.

Additionally, hardware-based MTP devices are also required for devices which do not support RFC 2833 compliant DTMF signalling (such as the Cisco VG248 analogue voice gateway), to perform the necessary DTMF interworking. In this case the MTP resources may need to be included in the non-compliant devices' media resources as well.

Furthermore, because of the single codec (G.729a) restriction for outbound calls, hardware transcoding resource may be required for devices that do not natively support the G.729a codec, such as a software-based T.38 fax solution which supports the G.711 codec only. This hardware transcoding needs to be applied to the media resources of the devices which require transcoding.

Cisco IOS Enhanced Software MTP

The number of concurrent software MTP sessions which a platform can support will depend upon the hardware of the Cisco IOS device being configured. As such, the following configuration should be considered as guidance only. Additionally, each MTP resource can only support a singular codec and therefore multiple profiles must be configured for each and every codec expected to be in use across the BT SIP Trunk platform.

Note: In order to support T.38 faxing an additional codec can (and must) be configured for each MTP profile. This codec is "**pass-through**", failure to do so will result in T.38 fax calls failing.

Sample configuration for Cisco IOS Enhanced Software MTP is as follows:

```
dspfarm profile 1 mtp
 description Soft MTP - G.711ulaw
 codec g711ulaw
 codec pass-through
 maximum sessions software 16
 associate application SCCP
!
dspfarm profile 2 mtp
 description Soft MTP - G.711alaw
 codec g711alaw
 codec pass-through
 maximum sessions software 16
 associate application SCCP
!
dspfarm profile 3 mtp
 description Soft MTP - G.729a
 codec g729ar8
 codec pass-through
 maximum sessions software 16
 associate application SCCP
```

Cisco IOS Hardware Transcoding

The number of concurrent hardware transcoding sessions which a platform can support will depend upon the hardware of the Cisco IOS device being configured. As such, the following configuration should be considered as guidance only. Furthermore by default a hardware transcoder requires one call leg to be a G.711, however to cover all possibilities the following example uses "**Universal transcoding**". This enables transcoding from any codec to any codec (within the configured codec list), rather than G.711 to any codec which is the default behaviour.

Each hardware transcoding profile must be configured for each and every codec expected to be in use across the BT SIP Trunk platform.

Note: In order to support T.38 faxing an additional codec can (and must) be configured for each hardware transcoder profile. This codec is "**pass-through**", failure to do so will result in T.38 fax calls failing.

Sample configuration for Cisco IOS Hardware Transcoding is as follows:

```
dspfarm profile 4 transcode universal
 description Hardware transcoder
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec pass-through
 maximum sessions 3
 associate application SCCP
!
```

4 End Device Specific Parameters Required for Correct Operation

Within the test solution an MGCP gateway was configured as an analogue fax gateway and also as a local PSTN gateway (to test for potential migration scenarios). This required some configuration to be applied on CUCM and some configuration to be applied on the MGCP controlled Cisco IOS gateway as well.

CUCM MGCP Gateway Configuration – Product Specific Configuration Layout

The following parameters were changed from default during testing:

Parameter	Default Setting	New Setting
Type of DTMF Relay	Current GW Config	NTE-CA
Modem Passthrough	Enable	Disable
Cisco Fax Relay	Disable	Disable
T38 Fax Relay	Disable	Enable

More detailed information on these changes is as follows:

Type of DTMF Relay

DTMF relay towards the BT SIP Trunk is extremely restrictive (i.e. the use of RFC 2833 compliant signalling is required). Therefore to ensure that CUCM instructs the MGCP gateway to use the correct DTMF signalling the signalling needs to be set under Call Agent control by setting the Type of DTMF Relay parameter to “**NTE-CA**”.

Modem Passthrough

As only T.38 fax support was mandated the potential to use G.711 pass-through for faxing needed to be restricted and therefore Modem Passthrough was set to “**Disable**”.

Cisco Fax Relay

Cisco Fax Relay uses a proprietary signalling mechanism to signal switchover to using Fax Relay and therefore will not interoperate with the BT SIP Trunk platform. Accordingly the capability must be kept disabled by setting the Cisco Fax Relay parameter to “**Disable**” (which is the default setting).

T38 Fax Relay

The BT SIP Trunk platform requires the use of T.38 fax relay to allow faxes to successfully transfer and therefore this capability must be enabled by setting the T38 Fax Relay parameter to “**Enable**”.

IOS MGCP Gateway Configuration

By default a Cisco IOS voice gateway will drop the negotiated fax rate to a rate commensurate with the negotiated voice codec (i.e. 7,200 bps for a G.729a call or 14,400 bps for a G.711 call). This can be overridden using the “**mgcp fax rate <rate>**” command should solution specific configurations require it.

Additionally redundancy can also be added to the T.38 fax data stream to counter the effects of lost packets (the important parameter being high speed redundancy) using the “**mgcp fax t38 ls_redundancy <value>**” and “**mgcp fax t38 hs_redundancy <value>**” commands.

As with other devices, the MGCP gateway must also transmit RFC 2833 compliant DTMF signalling. By default it does not and an additional capability package must be enabled to do so, this is done with the use of the “**mgcp dtmf-relay voip codec all mode nte-ca**” and “**mgcp package-capability fm-package**” commands.

The following example reflects the Cisco IOS voice gateway configuration that corresponds to the CUCM MGCP configuration, but further expands it to add RFC 2833 compliant DTMF signalling, implement full T.38 fax redundancy and override the negotiated fax rate beyond the default G.729a codec restriction:

```
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server <ip address list>
ccm-manager config
!
mgcp
mgcp call-agent <ccm server> 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode nte-ca
mgcp rtp unreachable timeout 1000 action notify
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax rate 14400
mgcp fax t38 ecm
mgcp fax t38 ls_redundancy 5
mgcp fax t38 hs_redundancy 2
!
mgcp profile default
!
!
```

Note: This configuration assumes automatic CUCM configuration rather than manual MGCP configuration via the Cisco IOS voice gateway CLI. Any additional IOS configuration commands that were manually entered are highlighted in red.