



UC-One Connect Mobile

Configuration Guide

Release 3.9.35
Document Version 1

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Document Revision History

| Release | Version | Reason for Change | Date |
|---------|---------|---|--------------------|
| 3.8.1 | 1 | Edited changes and published document. | April 24, 2019 |
| 3.8.4 | 1 | Updated section 6.3.2 Guest Client . Updated section 6.1.25 Call Auto Recovery . Updated section 6.1.23 SIP rport Management for NAT Traversal . Updated section 6.10.5 Xtended Services Interface Mid-Call Controls Widget . Updated section 6.1.26.2 Secure Real-time Transport Protocol . Updated section 6.1.31 One-time VoIP Call . Updated section 6.1.2 Codec Configuration for Client . Updated section 6.10.3 Deployment Type . | June 25, 2019 |
| 3.8.4 | 1 | Edited changes and published document. | June 26, 2019 |
| 3.8.5 | 1 | Removed section 6.1.28.2 Update Emergency Location Dialog . Updated section 6.15.2 Directory Cache . | July 10, 2019 |
| 3.8.5 | 1 | Edited changes and published document. | July 11, 2019 |
| 3.8.6 | 1 | Added section 6.15.3 Contact Search . Added section 6.16 Login Informational Dialog . Added section 6.1.36 Number of Rings . Updated section 6.3.2 Guest Client . | August 19, 2019 |
| 3.8.6 | 1 | Edited changes and published document. | August 22, 2019 |
| 3.9.1 | 1 | Updated section 6.6 Landing Page . Updated section 6.1.36 Number of Rings . Updated section 6.1 VoIP Audio Calling . Updated section 8.3 Configuration File (config-uc1s.xml) . Added section 6.1.37 Single Alerting . Added section 2.8 Changes for Configuration Files for Release 3.9.1 . | September 16, 2019 |
| 3.9.1 | 1 | Edited changes and published document. | September 17, 2019 |
| 3.9.4 | 1 | Added section 6.17 Cisco BroadWorks Anywhere . Updated section 6.1.31 One-time VoIP Call . Updated section 6.1.21 User Selectable Control of VoIP Mode . Updated section 6.1.3 Force TCP, TLS, or UDP Usage . | November 19, 2019 |
| 3.9.4 | 1 | Edited changes and published document. | November 21, 2019 |
| 3.9.6 | 1 | Updated section 6.5 Web-based Call Settings . Added section 6.18 HTTP Proxy . Added section 7 Features and Configuration Dependencies and sub-section 7.1 MNO and OTT Deployments . Updated section 6.8.4 Call History . | January 17, 2020 |
| 3.9.6 | 1 | Edited changes and published document. | January 17, 2020 |
| 3.9.8 | 1 | Updated section 6.15.2 Directory Cache . Updated section 6.6 Landing Page . | February 7, 2020 |

| Release | Version | Reason for Change | Date |
|-----------------------|---------|---|--------------------|
| 3.9.8 | 1 | Edited changes and published document. | February 12, 2020 |
| 3.9.8 | 2 | Updated section 6.1.3 Force TCP, TLS, or UDP Usage . Updated section 11 Appendix B: DM Tag Provisioning Script . Updated section 6.5 Web-based Call Settings . Added section 6.1.38 Call Statistics . | February 14, 2020 |
| 3.9.8 | 2 | Edited changes and published document. | March 17, 2020 |
| 3.9.12 | 1 | Removed changes in versions older than 3.8.1. | April 3, 2020 |
| 3.9.12 | 1 | Updated section 6.5 Web-based Call Settings - updated Call Center tag names. Updated section 6.1.32 Incoming Audio Calls . Updated section 7.1 MNO and OTT Deployments . Added section 7.2 Automatic Deployment Mode Detection . | April 14, 2020 |
| UC 3.9.12 W Jun 20 | 2 | Renamed guide to include Webex. Added section 9 Webex for BroadWorks . Added Appendix C: Webex for BroadWorks DM Tag Provisioning Script . Updated section 6.1.6 SIP Failover and Failback . | June 3, 2020 |
| UC 3.9.12 W Jun 20 | 2 | Edited changes and published document. | June 11, 2020 |
| UC 3.9.12 W Jul 20 | 1 | Added section 9.6.27 Codec Configuration (which was subsequently removed). | July 1, 2020 |
| UC 3.9.12 W Jul 20 | 1 | Added section 9.6.25 XSI Root and Paths (which was subsequently removed). | July 2, 2020 |
| UC 3.9.12 W Jul 20 | 1 | Edited changes and published document. | July 20, 2020 |
| UC 3.9.14 W Jul 20 | 1 | Added section 6.19 Version Control . | July 27, 2020 |
| UC 3.9.14 W Aug 20 | 1 | Added section 9.6.21 Call Auto Recovery / Seamless Call Handover . | August 6, 2020 |
| UC 3.9.14 W Aug 20 | 1 | Removed section 6.6 UC-One Hub . | August 7, 2020 |
| UC 3.9.14 W Aug 20 | 1 | Edited changes and published document. | August 7, 2020 |
| UC 3.9.16 W Sep 20 | 1 | Added section 9.6.28.2 Call Pickup . Updated section 9.6.29.1 Emergency Calling . | September 4, 2020 |
| UC 3.9.16 W Sep 20 | 1 | Updated section 6.1 VoIP Audio Calling . Updated section 6.1.32 Incoming Audio Calls . | September 11, 2020 |
| UC 3.9.16 W Sep 20 | 1 | Edited changes and published document. | September 25, 2020 |
| UC 3.9.18 | 1 | Updated section 6.1 VoIP Audio Calling . Updated section 6.1.3 Force TCP, TLS, or UDP Usage . Updated section 6.1.26.2 Secure Real-time Transport Protocol . Removed content related to Webex. | January 5, 2021 |

| Release | Version | Reason for Change | Date |
|-----------|---------|--|------------------|
| UC 3.9.18 | 1 | Edited changes and published document. | January 7, 2021 |
| UC 3.9.20 | 1 | Updated document for Release 3.9.20. | March 22, 2021 |
| UC 3.9.20 | 1 | Updated copyright notice. Edited changes and published document. | March 22, 2021 |
| UC 3.9.22 | 1 | Added section 6.20 Token Expiry Warning . | May 29, 2021 |
| UC 3.9.22 | 1 | Edited changes and published document. | June 7, 2021 |
| UC 3.9.24 | 1 | Updated document for Release 3.9.24. | August 24, 2021 |
| UC 3.9.24 | 1 | Edited changes and published document. | August 24, 2021 |
| UC 3.9.30 | 1 | Updated document for Release 3.9.30. | December 2, 2021 |
| UC 3.9.35 | 1 | Updated document for Release 3.9.35 | October 6 2022 |

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1 Summary of Changes

This section describes the changes to this document for each release and document version.

1.1 Changes for UC-One Release 3.9.35, Document Version 1

This version of the document includes the following change:

- Added
"refreshXSIContactsLimit=%CONTACTS_CACHE_CONTROL_CACHE_refreshXSIContactsLimit%"

```
<services>
<contacts>
  <cache-control
cacheLimit=%CONTACTS_CACHE_CONTROL_CACHE_LIMIT%"
cacheInterval=%CONTACTS_CACHE_CONTROL_CACHE_INTERVAL%
refreshXSIContactsLimit=%CONTACTS_CACHE_CONTROL_CACHE_refreshXSICo
ntactsLimit%"/>
  </cache-control>
</contacts>
</services>
```

"%CONTACTS_CACHE_CONTROL_CACHE_refreshXSIContactsLimit%" this defines number of local contact cache size, 100 contacts default.

]

1.2 Changes for UC-One Release 3.9.30, Document Version 1

This version of the document includes the following change:

- Added <media-waiting-tone> under <calls>.

```
<config>
<services><calls>
<media-waiting-tone enabled=%ENABLE_MEDIA_WAITING_TONE_MOBILE%"/>
```

The following %TAG%s were added in both the Collaborate template and the BroadWorks (BWKS) config template (*config-uc1s.xml*) for UC-One SaaS version of the application:

- %ENABLE_MEDIA_WAITING_TONE_MOBILE%

1.3 Changes for UC-One Release 3.9.24, Document Version 1

This version of the document includes the following change:

- Updated document for Release 3.9.24.

1.4 Changes for UC-One Release 3.9.22, Document Version 1

This version of the document includes the following changes:

- Added section [6.20 Token Expiry Warning](#).
- Added <tokenExpiryWarnBeforeDays> under <settings>.

```
<config>
<settings>

<tokenExpiryWarnBeforeDays>%TOKEN_EXPIRY_WARN_BEFORE_DAYS%</tokenExpiryWarnBeforeDays>
```

The following %TAG% was added:

- %TOKEN_EXPIRY_WARN_BEFORE_DAYS%

1.5 Changes for UC-One Release 3.9.20, Document Version 1

This version of the document includes the following change:

- Updated document for Release 3.9.20.

1.6 Changes for UC-One Release 3.9.18, Document Version 1

This version of the document includes the following changes:

- Updated section [6.1 VoIP Audio Calling](#) – Renamed TLS_VERSION_MOBILE to TLS_VERSION.
- Updated section [6.1.3 Force TCP, TLS, or UDP Usage](#) – Renamed TLS_VERSION_MOBILE to TLS_VERSION.
- Updated section [6.1.26.2 Secure Real-time Transport Protocol](#) – Added note that SRTP is disabled by default.
- Removed content related to Webex.

1.7 Changes for UC-One Release 3.9.16, Document Version 2

This version of the document includes the following change:

- Updated section [6.1.32 Incoming Audio Calls](#).

1.8 Changes for UC-One Release 3.9.16, Document Version 1

This version of the document includes the following changes:

- Updated section [6.1 VoIP Audio Calling](#).
- Updated section [6.1.32 Incoming Audio Calls](#).
- Added section [9.6.28.2 Call Pickup](#).
- Updated section [9.6.29.1 Emergency Calling](#).

1.9 Changes for UC-One Release 3.9.14, Document Version 1

This version of the document includes the following changes:

- Added section [6.19 Version Control](#).
- Removed section [6.6 UC-One Hub](#).

1.10 Changes for UC-One Release 3.9.12, Document Version 1

This version of the document includes the following changes:

- Added section [9.6.27 Codec Configuration](#) (which was subsequently removed).
- Added section [9.6.25 XSI Root and Paths](#) (which was subsequently removed) .

1.11 Changes for UC-One Release 3.9.12, Document Version 2

This version of the document includes the following changes:

- Added section [9 Webex for BroadWorks](#).
- Added [Appendix C: Webex for BroadWorks DM Tag Provisioning Script](#).
- Updated section [6.1.6 SIP Failover and Failback](#).

1.12 Changes for Release 3.9.12, Document Version 1

This version of the document includes the following changes:

- Updated section [6.5 Web-based Call Settings](#) - updated Call Center tag names.
- Updated section [6.1.32 Incoming Audio Calls](#) – added new configuration parameter related to the iOS 13 push notifications changes.
- Updated section [7.1 MNO and OTT Deployments](#) – updated based on the new automatic deployment mode detection logic.
- Added section [7.2 Automatic Deployment Mode Detection](#).

1.13 Changes for Release 3.9.8, Document Version 2

This version of the document includes the following changes:

- Updated section [6.1.3 Force TCP, TLS, or UDP Usage](#) – updated the default value of %TRANSPORTS_TYPE_MOBILE% from TCP to TLS.
- Updated section [11 Appendix B: DM Tag Provisioning Script](#) – updated the tags renamed from Communicator to Connect and removed unused tags.
- Updated section [6.5 Web-based Call Settings](#).
- Added section [6.1.38 Call Statistics](#).

1.14 Changes for Release 3.9.8, Document Version 1

This version of the document includes the following changes:

- Updated section [6.15.2 Directory Cache](#).
- Updated section [6.6 Landing Page](#).

1.15 Changes for Release 3.9.6, Document Version 1

This version of the document includes the following changes:

- Updated section [6.5 Web-based Call Settings](#) – added a note about the URL configuration.
- Added section [6.18 HTTP Proxy](#).
- Added section [7 Features and Configuration Dependencies](#) and sub-section [7.1 MNO and OTT Deployments](#).
- Updated section [6.17 Cisco BroadWorks Anywhere](#) – fixed the default value of %ENABLE_BROADWORKS_ANYWHERE_ADVANCED_OPTIONS_MOBILE%.
- Updated section [6.1.31 One-time VoIP Call](#) – added the default value of %ONE_TIME_VOIP_REGISTER%.

- Updated section [6.8.4 Call History](#) – added new config parameter `%CALL_HISTORY_SHOW_DIRECTORY_SEARCH%`.

1.16 Changes for Release 3.9.4, Document Version 1

This version of the document includes the following changes:

- Added section [6.17 Cisco BroadWorks Anywhere](#).
- Updated section [6.1.31 One-time VoIP Call](#) – added new config parameter to control establishment of SIP REGISTER session before the VoIP call.
- Updated section [6.1.21 User Selectable Control of VoIP Mode](#) – added VoIP only mode and the always-on-warn option for VoIP calls using Mobile Network Data.
- Updated section [6.1.3 Force TCP, TLS, or UDP Usage](#) – fixed the TLS transport type configuration value.

1.17 Changes for Release 3.9.1, Document Version 1

This version of the document includes the following changes:

- Updated section [6.6 Landing Page](#) – fix the “messages” in the list of supported Values.
- Updated section [6.1.36 Number of Rings](#) – fixed the description of the configurable parameters.
- Updated section [6.1 VoIP Audio Calling](#) – fixed the default value of `%USE_PROXY_DISCOVERY_MOBILE%`.
- Updated section [8.3 Configuration File \(config-uc1s.xml\)](#) – updated the list of BroadWorks `%TAG%`s that can control features in the UC-One client.
- Added section [6.1.37 Single Alerting](#).
- Added section [2.8 Changes for Configuration Files for Release 3.9.1](#).

1.18 Changes for Release 3.8.6, Document Version 1

This version of the document includes the following changes:

- Added section [6.15.3 Contact Search](#).
- Added section [6.16 Login Informational Dialog](#).
- Added section [6.1.36 Number of Rings](#).
- Updated section [6.3.2 Guest Client](#) – removed the reset guest link as an available action for the user.

1.19 Changes for Release 3.8.5, Document Version 1

This version of the document includes the following changes:

- Removed section [6.1.28.2 Update Emergency Location Dialog](#) – not supported.
- Updated section [6.15.2 Directory Cache](#) – added clarification that this feature is intended for small and medium enterprises.

1.20 Changes for Release 3.8.4, Document Version 1

This version of the document includes the following changes:

- Updated section [6.3.2 Guest Client](#) – removed the tag that controls the availability of the application link in the invitation.
- Updated section [6.1.25 Call Auto Recovery](#) – the tags are applicable to both iOS and Android clients.
- Updated section [6.1.23 SIP rport Management for NAT Traversal](#) – added a note about IPv6.
- Updated section [6.10.5 Xtended Services Interface Mid-Call Controls Widget](#) – clarified that the widget availability depends on the enabled attribute of the <extended-call-control> tag.
- Updated section [6.1.26.2 Secure Real-time Transport Protocol](#) – added description for the “transport” mode.
- Updated section [6.1.31 One-time VoIP Call](#) – corrected the value of *denyCallsFromUnregisteredUsers* that should be configured.
- Updated section [6.1.2 Codec Configuration for Client](#) – added VP9 in the configuration example.
- Updated section [6.10.3 Deployment Type](#) – added recommendation for the Tablet configuration.

1.21 Changes for Release 3.8.1, Document Version 1

This version of the document includes the following changes:

- Added section [6.1.35 Call Recording](#).
- Added section [2.1 Changes for Configuration Files for Release 3.8.1](#).
- Updated section [6.3.2 Guest Client](#) – added information about the new tag, which controls the availability of the application link in the invitation.

2 Changes for Configuration Files

2.1 Changes for Configuration Files for UC-One Release 3.9.18

There were no updates in the configuration files for this version.

2.2 Changes for Configuration Files for UC-One Release 3.9.16

Added the ring-timeout-seconds attribute under the <push-notifications-for-calls> tag.

```
<config>
<services>
  <push-notifications-for-calls enabled="true"
  connect-sip-on-accept="%PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE%"
  use-registration-v1="%PN_FOR_CALLS_USE_REGISTRATION_V1%"
  ring-timeout-seconds="%PN_FOR_CALLS_RING_TIMEOUT_SECONDS_MOBILE%"/>
```

The following %TAG% was added:

- % PN_FOR_CALLS_RING_TIMEOUT_SECONDS_MOBILE %

2.3 Changes for Configuration Files for Release 3.9.14

- Added <version-control> under <services>.

```
<config>
<services>
  <version-control enabled="%ENABLE_VERSION_CONTROL%">
    <upgrade>
      <android>
        <must>%ANDROID_MUST_VERSION%</must>
        <recommended>%ANDROID_RECOMMENDED_VERSION%</recommended>
        <download>%ANDROID_DOWNLOAD_LINK%</download>
      </android>
      <ios>
        <must>%IOS_MUST_VERSION%</must>
        <recommended>%IOS_RECOMMENDED_VERSION%</recommended>
        <download>%IOS_DOWNLOAD_LINK%</download>
      </ios>
    </upgrade>
  </version-control>
```

The following %TAG%s were added:

- %ENABLE_VERSION_CONTROL%
- %ANDROID_MUST_VERSION%
- %ANDROID_RECOMMENDED_VERSION%
- %ANDROID_DOWNLOAD_LINK%
- %IOS_MUST_VERSION%
- %IOS_RECOMMENDED_VERSION%
- %IOS_DOWNLOAD_LINK%

Removed <uchub> under <config>.

```

<config>
<uchub>
  <enabled>${ENABLE_HUB}</enabled>
  <login>
    <url>${HUB_LOGIN_URL}</url>
  </login>
  <settings>
    <url>${HUB_SETTINGS_URL}</url>
  </settings>
  <contextual-gadget>
    <url>${CONTEXTUAL_GADGET_URL}</url>
  </contextual-gadget>
  <heartbeat>
    <url>${HUB_HEARTBEAT_URL}</url>
  </heartbeat>
  <metrics>
    <enabled>>false</enabled>
    <serviceProviderId></serviceProviderId>
    <resellerId></resellerId>
    <companyId></companyId>
  </metrics>
</uchub>

```

The following %TAG%s were removed:

- %ENABLE_HUB%
- %HUB_LOGIN_URL%
- %HUB_SETTINGS_URL%
- %CONTEXTUAL_GADGET_URL%
- %ENABLE_HUB_METRICS%
- %HUB_METRICS_SERVICE_PROVIDERID%
- %HUB_METRICS_RESELLERID%
- %HUB_METRICS_COMPANYID%
- %HUB_HEARTBEAT_URL%
- %HUB_ACTION_PATH%
- %HUB_DOMAIN%

2.4 Changes for Configuration Files for Release 3.9.12

The following %TAG%s were added:

- %WEB_CALL_STANDARD_SETTINGS_CC_VISIBLE%
- %WEB_CALL_BASIC_SETTINGS_CC_VISIBLE%
- %WEB_CALL_PREMIUM_SETTINGS_CC_VISIBLE%

```

<config>
<services>
<web-call-settings><service-settings>
<service>
  <name>Call Center - Standard</name>
  <visible>%WEB_CALL_STANDARD_SETTINGS_CC_VISIBLE%</visible>

```



```

</service>
<service>
  <name>Call Center - Basic</name>
  <visible>%WEB_CALL_BASIC_SETTINGS_CC_VISIBLE%</visible>
</service>
<service>
  <name>Call Center - Premium</name>
  <visible>%WEB_CALL_PREMIUM_SETTINGS_CC_VISIBLE%</visible>
</service>

```

- %PN_FOR_CALLS_USE_REGISTRATION_V1%

```

<config>
<services>
<push-notifications-for-calls enabled="true" connect-sip-on-
accept="%PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE%" use-registration-
v1="%PN_FOR_CALLS_USE_REGISTRATION_V1%"/>

```

2.5 Changes for Configuration Files for Release 3.9.8

- Added missing <service> tags under the section <web-call-settings>.

```

<config>
<services>
<web-call-settings>
  <enabled>%ENABLE_WEB_CALL_SETTINGS%</enabled>
  <url>%WEB_CALL_SETTINGS_URL%</url>
  <service-settings>

```

- Added <call-statistics> tag under section <calls>.

```

<config>
<services><calls>
<call-statistics enabled="%ENABLE_CALL_STATISTICS_MOBILE%"/>

```

The following %TAG%s were added:

- %CALL_HISTORY_SHOW_DIRECTORY_SEARCH%
- %WEB_CALL_SETTINGS_CFA_VISIBLE%
- %WEB_CALL_SETTINGS_ACR_VISIBLE%
- %WEB_CALL_SETTINGS_CFB_VISIBLE%
- %WEB_CALL_SETTINGS_CFNr_VISIBLE%
- %WEB_CALL_SETTINGS_CFNA_VISIBLE%
- %WEB_CALL_SETTINGS_SIMRING_VISIBLE%
- %WEB_CALL_SETTINGS_SEQRING_VISIBLE%
- %WEB_CALL_SETTINGS_RO_VISIBLE%
- %WEB_CALL_SETTINGS_ACB_VISIBLE%
- %WEB_CALL_SETTINGS_CW_VISIBLE%
- %WEB_CALL_SETTINGS_CLIDB_VISIBLE%
- %WEB_CALL_SETTINGS_PA_VISIBLE%
- %WEB_CALL_SETTINGS_BWA_VISIBLE%

- %WEB_CALL_SETTINGS_CC_VISIBLE%
- %WEB_CALL_SETTINGS_BWM_VISIBLE%
- %WEB_CALL_SETTINGS_VM_VISIBLE%
- %ENABLE_CALL_STATISTICS_MOBILE%

2.6 Changes for Configuration Files for Release 3.9.6

- Added show-directory-search attribute in the <call-history> tag.

```

<config>
<services>
<call-history enabled="%ENABLE_CALL_HISTORY%"
enhanced="%ENABLE_ENHANCED_CALL_HISTORY%" show-directory-
search="%CALL_HISTORY_SHOW_DIRECTORY_SEARCH%"/>
```

The following %TAG% was added:

- %CALL_HISTORY_SHOW_DIRECTORY_SEARCH%

2.7 Changes for Configuration Files for Release 3.9.4

- Added <broadworks-anywhere> tag under section <settings>.

```

<config>
<settings>
<broadworks-anywhere enable-advanced-
options="%ENABLE_BROADWORKS_ANYWHERE_ADVANCED_OPTIONS_MOBILE%">
  <call-control
default="%ANYWHERE_ADVANCED_OPTIONS_CALL_CONTROL_DEFAULT_MOBILE%" />
  <diversion-inhibitor
default="%ANYWHERE_ADVANCED_OPTIONS_DIVERSION_INHIBITOR_DEFAULT_MOBILE%" />
  <answer-confirmation
default="%ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ANSWER_CONFIRMATION_MOBILE%" />
  <alert-all-locations
default="%ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ALERT_ALL_LOCATIONS_MOBILE%" />
```

- Added register attribute under tag <one-time-voip>.

```

<config>
<services><calls>
<one-time-voip enabled="%ENABLE_ONE_TIME_VOIP%"
register="%ONE_TIME_VOIP_REGISTER%" />
```

The following %TAG%s were added in both the Collaborate template and the BroadWorks (BWKS) config template (*config-uc1s.xml*) for UC-One SaaS version of the application:

- %ENABLE_BROADWORKS_ANYWHERE_ADVANCED_OPTIONS_MOBILE%
- %ANYWHERE_ADVANCED_OPTIONS_CALL_CONTROL_DEFAULT_MOBILE%
- %ANYWHERE_ADVANCED_OPTIONS_DIVERSION_INHIBITOR_DEFAULT_MOBILE%
- %ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ANSWER_CONFIRMATION_MOBILE%
- %ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ALERT_ALL_LOCATIONS_MOBILE%
- %ONE_TIME_VOIP_REGISTER%

2.8 Changes for Configuration Files for Release 3.9.1

- Added <single-alerting> tag under section <calls>.

```

<config>
<services><calls>
<single-alerting enabled="%ENABLE_SINGLE_ALERTING%"/>

```

The following %TAG% was added:

- %ENABLE_SINGLE_ALERTING%

For UC-One SaaS version of the client, the following %TAG%s have been added in the *config-uc1s.xml* template:

- %CALL_RECORDING_MOBILE%
- %REJECT_WITH_XSI_MODE_MOBILE%
- %REJECT_WITH_XSI_DECLINE_REASON_MOBILE%
- %ENABLE_XSI_CALLS_MOBILE%
- %ENABLE_XSI_MIDCALL_CONTROLS_MOBILE%
- %ENABLE_XSI_MIDCALL_CONTROLS_WIDGET%
- %ENABLE_XSI_TRANSFER_CALLS_MOBILE%
- %VIDEO_SHOW_INCOMING_CALL_AS_AUDIO_MOBILE%
- %ENABLE_XSI_CONFERENCE_CALLS_MOBILE%
- %CONFERENCE_CALLS_AUTO_JOIN_PAUSE%
- %ENABLE_ONE_TIME_VOIP%
- %ENABLE_SINGLE_ALERTING%
- %CONFERENCE_BRIDGE_MEDIA_TYPE_MOBILE%
- %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE%
- %DEPLOYMENT_TYPE_MOBILE%
- %ENABLE_PERSONA_MANAGEMENT%
- %ENABLE_DIALING_DIALPAD%
- %ENABLE_DIALING_PREVIEW_URL_SCHEMA_NUMBER_TO_DIAL%
- %SIDE_MENU_CFA_VISIBLE%
- %SIDE_MENU_DND_VISIBLE%
- %ENABLE_WEB_CALL_SETTINGS%
- %WEB_CALL_SETTINGS_URL%
- %WEB_CALL_SETTINGS_CFA_VISIBLE%
- %WEB_CALL_SETTINGS_DND_VISIBLE%
- %ENABLE_PERSONAL_ASSISTANT_PRESENCE%
- %ENABLE_PERSONAL_ASSISTANT_PRESENCE_SILENT_ALERT%
- %ENABLE_PERSONAL_ASSISTANT_PRESENCE_ALERT_ME_FIRST%

- %ENABLE_MY_TELEPHONE_NUMBER_MOBILE%
- %CONTACTS_SORT_ORDER_VISIBLE%
- %CONTACTS_SORT_ORDER_DEFAULT%
- %CONTACTS_DISPLAY_NAME_ORDER_VISIBLE%
- %CONTACTS_DISPLAY_NAME_ORDER_DEFAULT%
- %CONTACTS_SHORT_NAMES_VISIBLE%
- %CONTACTS_SHORT_NAMES_DEFAULT%
- %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%
- %CONTACTS_CACHE_CONTROL_CACHE_INTERVAL%
- **CONTACTS_CACHE_CONTROL_CACHE_refreshXSIContactsLimit%**
- %ENABLE_CALL_HISTORY%
- %ENABLE_ENHANCED_CALL_HISTORY%
- %ENABLE_RTP_MUX_MOBILE%
- %ENABLE_AUTO_CONFIGURE_ANYWHERE%
- %DIALER_KEYBOARD_MODE_MOBILE%

2.9 Changes for Configuration Files for Release 3.8.6

- Added <xsi><search-sources> under section <contacts>.

```

<config>
<services><contacts>
<xsi><search-sources>
  <enterprise enabled="%ENABLE_CONTACTS_ENTERPRISE_SEARCH_MOBILE%"/>
  <enterprise-common
enabled="%ENABLE_CONTACTS_ENTERPRISE_COMMON_SEARCH_MOBILE%"/>
  <personal enabled="%ENABLE_CONTACTS_PERSONAL_SEARCH_MOBILE%"/>
  <group-common
enabled="%ENABLE_CONTACTS_GROUP_COMMON_SEARCH_MOBILE%"/>

```

- Added <login-informational-dialog> tag under section <settings>.

```

<config>
<settings>
<login-informational-dialog
enabled="%SHOW_LOGIN_INFORMATIONAL_DIALOG_MOBILE%"
recurring="%SHOW_INFORMATIONAL_DIALOG_ON_EVERY_LOGIN_MOBILE%"/>

```

- Added <number-of-rings-range> tag under section <services>.

```

<config>
<services>
<number-of-rings-range min="%NUMBER_OF_RINGS_RANGE_MIN_VALUE_MOBILE%"
max="%NUMBER_OF_RINGS_RANGE_MAX_VALUE_MOBILE%"/>

```

The following %TAG%s were added:

- %ENABLE_CONTACTS_ENTERPRISE_SEARCH_MOBILE%
- %ENABLE_CONTACTS_ENTERPRISE_COMMON_SEARCH_MOBILE%
- %ENABLE_CONTACTS_PERSONAL_SEARCH_MOBILE%

- %ENABLE_CONTACTS_GROUP_COMMON_SEARCH_MOBILE%
- %SHOW_LOGIN_INFORMATIONAL_DIALOG_MOBILE%
- %SHOW_INFORMATIONAL_DIALOG_ON_EVERY_LOGIN_MOBILE%
- %NUMBER_OF_RINGS_RANGE_MIN_VALUE_MOBILE%
- %NUMBER_OF_RINGS_RANGE_MAX_VALUE_MOBILE%

2.10 Changes for Configuration Files for Release 3.8.5

Removed section <non-phones> under section <emergency-dialing>.

The following %TAG%s were removed:

- %SHOW_EMERGENCY_WARNING_DIALOG_MOBILE%
- %EMERGENCY_DIALING_BEHAVIOR_MOBILE%

2.11 Changes for Configuration Files for Release 3.8.4

- Updated the <include-btbc-link-to-email-invitation> tag under section <myroom>: hardcoded the tag value to *false*.

```
<config>
<services><rooms><myroom>
<include-btbc-link-to-email-invitation>false</include-btbc-link-to-email-
invitation>
```

The following %TAG% was removed:

- %ENABLE_BTBC_LINK_INVITATION%

2.12 Changes for Configuration Files for Release 3.8.1

- Added <record> tag under section <calls>.

```
<config>
<calls>
<record enabled="%CALL_RECORDING_MOBILE%"/>
```

- Added <include-btbc-link-to-email-invitation> tag under section <myroom>.

```
<config>
<services><rooms><myroom>
<include-btbc-link-to-email-
invitation>%ENABLE_BTBC_LINK_INVITATION%</include-btbc-link-to-email-
invitation>
```

The following %TAG%s were added:

- %CALL_RECORDING_MOBILE%
- %ENABLE_BTBC_LINK_INVITATION%

3 Introduction

The purpose of this document is to provide a description of the configuration of the UC-One Connect for Android and iOS mobile clients.

NOTE: Android and iOS clients do not use all parameters available in the configuration template.

The *config.xml* file is common to iOS and Android Mobile.

The Android and iOS clients are configured using a configuration that is not visible to the end user. The *config.xml* provides server-specific information, such as server addresses and ports and runtime options for the client itself (for example, options visible in the *Settings* screen).

The configuration files are read by the client when it starts, after being retrieved from Device Management. The information from the configuration files is stored in the mobile phone's memory encrypted, thus making it invisible and inaccessible to the end user.

NOTE 1: The XML properties should not contain spaces (for example, `<audio enabled="%ENABLE_AUDIOCALLS_MOBILE%">` instead of `<audio enabled = "%ENABLE_AUDIOCALLS_MOBILE%">`).

NOTE 2: Previous versions of the UC-One Connect client have been using the *config.json* configuration file. This file was deprecated by the *config.xml* file.

NOTE 3: For users using Android foldable devices (for example, Samsung Fold) it is mandatory to configure both phone and tablet profiles since some of these devices use different profiles depending upon whether the device is open or closed.

4 Installation

4.1 Android Client

The Android client is installed as an application (Android application package [APK]), which keeps the settings-related data inside its private area and the configuration file data only inside its RAM.

There is version control based on the Google Play procedures. A standard Google Play notification is provided (that is, Android automatically indicates that there is a new version of software available).

When the new version is downloaded, the old software is overwritten; however, user data is kept by default.

Note that the user is not required to select any options for installation or un-installation.

4.2 iOS Client

The iOS client is installed as an application, which keeps the settings-related data inside its “sandbox” and the configuration file data is stored encrypted.

There is version control based on the Apple App Store procedures. The App Store icon is highlighted to indicate that there is a new version of software available.

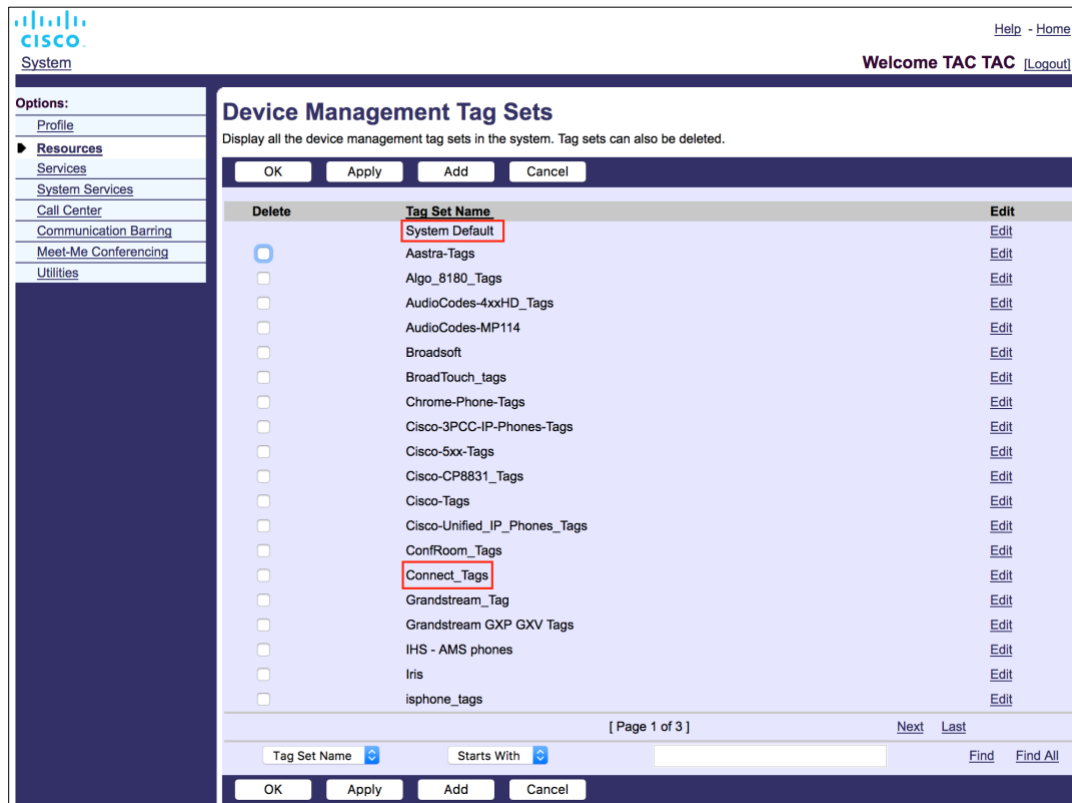
When the new version is downloaded, the old software is overwritten; however, user data is kept by default.

Note that the user is not required to select any options for installation or un-installation.

5 Connect Service Packages and Device Management

5.1 Device Management Tags

UC-One Connect uses the *Device Management Tag Sets* shown in the following figure. The *System Default* and custom *Connect_Tags* and *ConnectTablet_Tags* sets are required to provision specific device/client settings. Note that a system administrator must configure these tags through the *System* → *Resources* → *Device Management Tag Sets* option.



The screenshot displays the 'Device Management Tag Sets' configuration page. On the left is a navigation menu with options like Profile, Resources, Services, System Services, Call Center, Communication Barring, Meet-Me Conferencing, and Utilities. The main content area shows a table of tag sets:

| Delete | Tag Set Name | Edit |
|--------------------------|------------------------------|------|
| <input type="checkbox"/> | System Default | Edit |
| <input type="checkbox"/> | Aastra-Tags | Edit |
| <input type="checkbox"/> | Algo_8180_Tags | Edit |
| <input type="checkbox"/> | AudioCodes-4xxHD_Tags | Edit |
| <input type="checkbox"/> | AudioCodes-MP114 | Edit |
| <input type="checkbox"/> | Broadsoft | Edit |
| <input type="checkbox"/> | BroadTouch_tags | Edit |
| <input type="checkbox"/> | Chrome-Phone-Tags | Edit |
| <input type="checkbox"/> | Cisco-3PCC-IP-Phones-Tags | Edit |
| <input type="checkbox"/> | Cisco-5xx-Tags | Edit |
| <input type="checkbox"/> | Cisco-CP8831_Tags | Edit |
| <input type="checkbox"/> | Cisco-Tags | Edit |
| <input type="checkbox"/> | Cisco-Unified_IP_Phones_Tags | Edit |
| <input type="checkbox"/> | ConfRoom_Tags | Edit |
| <input type="checkbox"/> | Connect_Tags | Edit |
| <input type="checkbox"/> | Grandstream_Tag | Edit |
| <input type="checkbox"/> | Grandstream GXP GXV Tags | Edit |
| <input type="checkbox"/> | IHS - AMS phones | Edit |
| <input type="checkbox"/> | Iris | Edit |
| <input type="checkbox"/> | iphone_tags | Edit |

At the bottom of the interface, there is a search bar with 'Tag Set Name' and 'Starts With' filters, and buttons for 'OK', 'Apply', 'Add', and 'Cancel'.

Figure 1 UC-One Connect Device Management Tag Sets

5.2 Available Packages

Through the corresponding tags defined in the *Connect_Tags* and *ConnectTablet_Tags* sets, each of the communication services provided through Connect – Audio, Video, and Instant Messaging (Chat), and Presence can be independently controlled. Consequently, the following service packages can be created:

- Instant Messaging and Presence (IM&P) only
- IM&P + Audio
- IM&P + Audio + Video
- Audio + Video

5.2.1 Enable Package Features

| Service Package | IM&P | IM&P + Audio | IM&P + Audio + Video | Audio + Video |
|--------------------------------------|-------|--------------|----------------------|---------------|
| Tag | Value | | | |
| %DEPLOYMENT_MODEL_MESSAGING_ENABLED% | true | true | true | false |
| %DEPLOYMENT_MODEL_PRESENCE_ENABLED% | | | | |
| %ENABLE_AUDIOCALLS_MOBILE% | false | true | true | true |
| %ENABLE_VIDEOCALLS_MOBILE% | false | false | true | true |

5.2.2 Required Custom Tag Groups for Package Configuration

| Service Package | IM&P | IM&P + VoIP Calling | VoIP Calling |
|--|---------------------------|---------------------|--------------|
| Tag Group | Value (Y-Yes; O-Optional) | | |
| BroadWorks Xtended Service Interface (Xsi) | Y | Y | Y |
| VoIP Audio Calling | | Y | Y |
| VoIP Video Calling | | O | O |
| Instant Messaging and Presence | Y | Y | |
| Dialer Settings | O | O | O |
| Web-based Call Settings | | O | O |

5.3 Deployments of Mobile as Primary Device Without Shared Call Appearance

UC-One Connect supports deployments in which Mobile is the only and primary device for the user without the need to have Shared Call Appearance (SCA). For more information regarding configuration, service assignment, and provisioning steps for this deployment model, see the *UC-One Solution Guide*.

5.4 System Default Tags

As the system administrator, you can access the System Default tags through the *System* → *Resources* → *Device Management Tag Sets* option. The following System Default tags must be provisioned when the VoIP Calling package is installed.

| Tag | Description | Required For |
|---------------|---|--------------|
| %SBC_ADDRESS% | This should be configured as the fully qualified domain name (FQDN) or IP address of the session border controller (SBC) deployed in the network. Example: sbc.yourdomain.com | VoIP Calling |

| Tag | Description | Required For |
|-------------------|--|--------------|
| %SBC_PORT_MOBILE% | If the <i>SBC_ADDRESS</i> is an IP address, then this parameter should be set to the SBC port. If the <i>SBC_ADDRESS</i> is an FQDN, then it can be left unset. Example: 5075 | VoIP Calling |

5.5 Cisco BroadWorks System Tags

In addition to the default system tags and the custom tags that must be defined, there are existing Cisco BroadWorks System Tags that are typically used and are part of the recommended Device Type Archive File (DTAF). These tags are listed in this section. Depending on the installed solution package, not all system tags are used.

| Tag | Description | Required For |
|---------------------------------|---|--------------------|
| %BWNETWORK-CONFERENCE-SIPURI-1% | This is the server URI used to enable N-Way conferencing. | VoIP Calling |
| %BWVOICE-PORTAL-NUMBER-1% | This number is used for voice mail. The client dials this number when retrieving voice mail. | VoIP & XSI Calling |
| %BWLINERPORT-1% | SIP username used in SIP signaling, for example, in registration. | VoIP Calling |
| %BWAUTHPASSWORD-1% | SIP password used in SIP signaling. | VoIP Calling |
| %BWHOST-1% | Typically used as the SIP domain. | VoIP Calling |
| %BWAUTHUSER-1% | SIP username typically used in 401 and 407 signaling. Can be different from the default SIP username. | VoIP Calling |
| %BW_USER_IMP_ID-1% | This is the messaging username. | IM&P |
| %BW_USER_IMP_PWD-1% | This is the messaging password. | IM&P |
| %BW_IMP_SERVICE_NET_ADDRESS-1% | This is the messaging server domain. | IM&P |
| %BWE164-1% | This tag provides a phone number in international format that can also be used for federated calling (see section 6.1.17 Federated Calling in IM&P Deployments). | Calling |

5.6 Custom Tags

A new custom tag set must be created for the UC-One Connect client and then associated with the client's device type. This tag set provides flexibility in managing the client's network/service connectivity settings as well as feature activation controls.

This custom tag set is provisioned by a system administrator through the *System* → *Resources* → *Device Management Tag Sets* option. The administrator must add new tag sets called *Connect_Tags* and *ConnectTablet_Tags*, create each individual tag, and set its value. Section references provide detailed descriptions for each tag. In addition, tags are flagged as mandatory or optional (M/O). The custom tags are separated in groups based on the functionality and are discussed later in this document.

6 Custom Tags

This section describes deployment settings. Note, however, that some settings described in this section are supported only for the specific release of the client. To determine if a setting does not apply to an older client version, see the appropriate release-specific configuration guide.

6.1 VoIP Audio Calling

| Tag | Default Value | M/O |
|--|---|-----|
| %ENABLE_AUDIOCALLS_MOBILE% | true Enable/Disable VoIP audio calling | M |
| %ENABLE_AUDIO_QOS_MOBILE% | false For more information, see section 6.1.14 QoS – Diffserv Support . | M |
| %CONFERENCE_TYPE% | meetme For more information, see section 6.1.11 Conference Calls Bridge . | M |
| %ENABLE_SIP_INFO_MOBILE% | true For more information, see section 6.1.13 SIP Info . | M |
| %ENABLE_TRANSFER_TO_CIRCUIT_SWITCH_MOBILE% | true For more information, see section 6.1.10 Mid-Call Control Services . | M |
| %ENABLE_CONFERENCE_CALLS_MOBILE% | true For more information, see section 6.1.10.2 Conference Call . | M |
| %ENABLE_REJECT_WITH_486_MOBILE% | false For more information, see section 6.1.9 Reject Behavior . | M |
| %SBC_ADDRESS% | sbc.domain.com For more information, see section 6.1.1 SIP Server Configuration – Address, Port, and DNS Settings . | M |
| %SBC_PORT_MOBILE% | 5075 | M |
| %SOURCE_PORT% | 5075 | M |
| %TRANSPORTS_TYPE_MOBILE% | tls For more information, see section 6.1.3 Force TCP, TLS, or UDP Usage . | M |
| %SRTP_ENABLED_MOBILE% | false For more information, see section 6.1.26.2 Secure Real-time Transport Protocol . | M |
| %SRTP_MODE_MOBILE% | optional For more information, see section 6.1.26.2 Secure Real-time Transport Protocol Error! Reference source not found.. | O |

| Tag | Default Value | M/O |
|---------------------------------------|---|-----|
| %CALL_QUALITY_AUDIO_ENABLE_MOBILE% | false For more information, see section 6.1.26.4 Real-Time Control Protocol Extended Reports Error! Reference source not found.. | O |
| %CALL_QUALITY_SERVICE_URI% | For more information, see section 6.1.26.4 Real-Time Control Protocol Extended Reports Error! Reference source not found..Error! Reference source not found. | O |
| %CALL_QUALITY_LOCAL_GROUP% | For more information, see section 6.1.26.4 Real-Time Control Protocol Extended Reports Error! Reference source not found..Error! Reference source not found. | O |
| %ENABLE_ADAPTIVE_QUALITY_MOBILE% | true For more information, see section 6.1.15 Adaptive Call Quality . | O |
| %AUDIO_QOS_TYPE_MOBILE% | DSCP For more information, see section 6.1.15 Adaptive Call Quality . | O |
| %AUDIO_QOS_VALUE_MOBILE% | 46 For more information, see section 6.1.15 Adaptive Call Quality . | O |
| %ENABLE_CALL_PULL_MOBILE% | true For more information, see section 6.1.8 Call Pull (Call Retrieve) . | O |
| %CONFERENCE_BRIDGE_MEDIA_TYPE_MOBILE% | audio For more information, see section 6.1.11 Conference Calls Bridge . | O |
| %ENABLE_TRANSFER_CALLS_MOBILE% | true For more information, see section 6.1.10 Mid-Call Control Services . | O |
| %TRANSFER_CALL_TYPE_MOBILE% | full For more information, see section 6.1.10 Mid-Call Control Services . | O |
| %TLS_VERSION% | ssl23 For more information, see section 6.1.3 Force TCP, TLS, or UDP Usage . | O |
| %100_REL_ENABLED_MOBILE% | true For more information, see section 6.1.12 100rel – SIP Provisional Message Reliability . | O |
| %USE_EXPIRES_HACK_MOBILE% | false For more information, see section 6.1.16 Expires Hack . | O |
| %RTP_AUDIO_PORT_RANGE_START% | 8500 For more information, see section 6.1.26.1 6.1.26.1 Preferred Media Ports . | O |
| %RTP_AUDIO_PORT_RANGE_END% | 8598 For more information, see section 6.1.26.1 6.1.26.1 Preferred Media Ports . | O |

| Tag | Default Value | M/O |
|--|--|-----|
| %USE_MEDIASEC_MOBILE% | false For more information, see section 6.1.20 3GPP SIP Headers for SRTP . | O |
| %USE_PROXY_DISCOVERY_MOBILE% | false For more information, see section 6.1.4 Dynamic SIP Proxy Discovery . | O |
| %ENABLE_RE-KEYING_MOBILE% | true For more information, see section 6.1.26.3 SRTP Re-keying Configurability . | O |
| %ENABLE_VOIP_MODE_MOBILE% | false For more information, see section 6.1.21 User Selectable Control of VoIP Mode . | O |
| %CELLULAR_DATA_VOIP_CALLS_STATE_MOBILE% | always-on For more information, see section 6.1.21 User Selectable Control of VoIP Mode . | O |
| %ENABLE_SIP_UPDATE_SUPPORT_MOBILE% | false For more information, see section 6.1.22 SIP UPDATE . | O |
| %ENABLE_SIGN_IN_TIMER_MOBILE% | false For more information, see section 6.1.6 SIP Failover and Failback . | O |
| %SIGN_IN_TIMER_PERIOD_MOBILE% | 5000 For more information, see section 6.1.6 SIP Failover and Failback . | O |
| %ENABLE_CALLS_AUTO_RECOVERY_MOBILE% | false For more information, see section 6.1.25 Call Auto Recovery . | O |
| %CALLS_AUTO_RECOVERY_DATA_TO_WIFI_MODE_MOBILE% | keep For more information, see section 6.1.25 Call Auto Recovery . | O |
| %ENABLE_ONE_TIME_VOIP% | false For more information, see section 6.1.31 One-time VoIP Call . | O |
| %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE% | false For more information, see section 6.1.32 Incoming Audio Calls . | O |
| %ENABLE_MWI_MOBILE% | true For more information, see section 6.1.33 MWI . | O |
| %MWI_MODE_MOBILE% | empty For more information, see section 6.1.33 MWI . | O |
| %ENABLE_USE_RPORT_MOBILE% | false For more information, see section 6.1.23 SIP rport Management for NAT Traversal . | O |
| %RPORT_USE_LOCAL_PORT_MOBILE% | false For more information, see section 6.1.23 SIP rport Management for NAT Traversal . | O |

| Tag | Default Value | M/O |
|---|---|-----|
| %ENABLE_PROXY_DISCOVERY_RFC5922_CERT_VALIDATION_MOBILE% | true For more information, see section 6.1.4 Dynamic SIP Proxy Discovery . | O |
| %ENABLE_SIP_QOS_MOBILE% | false For more information, see section 6.1.34 Quality of Service – Diffserv Support for SIP . | O |
| %SIP_QOS_TYPE_MOBILE% | empty For more information, see section 6.1.34 Quality of Service – Diffserv Support for SIP . | O |
| %SIP_QOS_VALUE_MOBILE% | empty For more information, see section 6.1.34 Quality of Service – Diffserv Support for SIP . | O |
| %ENABLE_TCP_RESET_FAILOVER_MOBILE% | false For more information, see section 6.1.1 SIP Server Configuration – Address, Port, and DNS Settings . | O |
| %AUTODETECT_CONFERENCE% | true For more information, see section 6.1.11 Conference Calls Bridge . | O |
| %CONFERENCE_TITLE% | empty For more information, see section 6.1.11 Conference Calls Bridge . | O |
| %BRIDGE_ID% | empty For more information, see section 6.1.11 Conference Calls Bridge . | O |
| %CALL_RECORDING_MOBILE% | false For more information, see section 6.1.35 Call Recording . | O |
| %ENABLE_SINGLE_ALERTING% | true For more information, see section 6.1.37 Single Alerting . | O |
| %ENABLE_CALL_STATISTICS_MOBILE% | false For more information, see section 6.1.38 Call Statistics . | O |
| %PN_FOR_CALLS_USE_REGISTRATION_V1% | false For more information, see section 6.1.32 Incoming Audio Calls . | O |
| %PN_FOR_CALLS_RING_TIMEOUT_SECONDS_MOBILE% | 35 For more information, see section 6.1.32 Incoming Audio Calls . | O |
| %ENABLE_MEDIA_WAITING_TONE_MOBILE% | false For more information, see section 6.1.39 Media Waiting Tone . | O |

6.1.1 SIP Server Configuration – Address, Port, and DNS Settings

A common use is to configure the client to use another SIP network by modifying the *config.xml* file. Typically, the following parameters must be changed.

As a system administrator, you can access the *System Default* tags through *System* → *Resources* → *Device Management Tag Sets*. The following *System Default* tags must be provisioned.

| Tag | Description | Values | M/O |
|-------------------|---|---------|-----|
| %SBC_ADDRESS% | This should be configured as the FQDN or IP address of the SBC deployed in the network. Example: sbc.yourdomain.com | string | M |
| %SBC_PORT_MOBILE% | If the <i>SBC_ADDRESS</i> is an IP address, then this parameter should be set to the SBC port. If the <i>SBC_ADDRESS</i> is an FQDN, then it can be left unset. Example: 5075 | integer | M |

Note that the configuration parameters described in the previous table have the following impact on client connectivity:

- If the proxy address is not set (that is, it is empty), then the client uses the address and port from the registrar parameter.
- If the proxy address is an IP address and the proxy port IS NOT empty, then the client uses the values as is, that is, no DNS SRV query is done. In this case, failover would not be possible.
- If the proxy address is an IP address and the port IS empty, then the client uses the IP and the default SIP port 5060. In this case, failover would not be possible.
- If the proxy address is an FQDN and the proxy port IS NOT empty, then the client does an A-query on the FQDN. If there are multiple results, the client uses them as the primary and secondary servers for the failover.

NOTE: It is strongly advisable for the SIP port to be different from 5060 (for example, 5075) due to known issues with using the standard SIP port (5060) with mobile devices.

Other parameters can also be changed; however, the previous settings enable the basic functionality:

- Registration to the network
- Audio/video calls

The setup of the SIP servers adheres to the following basic scheme:

- Registrar URI contains the URI of the registrar
- Proxy address contains the IP (in case DNS resolving fails)
- Only one proxy can be defined

UC-One Connect client also supports TCP reset failover. If TCP reset failover is enabled, and a SIP request fails with a TCP reset response, the SIP socket is recreated, the SIP REGISTER session is refreshed, and the failed SIP request is resent. If the TCP reset is disabled, a new REGISTER session is initiated when the socket is recreated.

| Tag | Description | Values |
|------------------------------------|---|-------------|
| %ENABLE_TCP_RESET_FAILOVER_MOBILE% | Enables the TCP reset failover. The default is "false". | true, false |

6.1.2 Codec Configuration for Client

UC-One Connect offers a variety of audio and video codecs. The respective lists of codecs are located under *config/services/calls/* in the *audio/codecs* and *video/codecs* sections. The priority of each codec can be changed via the *XML-attribute priority*, which is a value between 0.0 (lowest) and 1.0 (highest).

```
<video><codecs>
  <codec name="H264" payload="109" resolution="CIF" framerate="25"
  bitrate="768000" priority="1.0" />
  <codec name="VP9" payload="102" resolution="CIF" framerate="25"
  bitrate="150000" priority="0.5"/>
```

The UC-One Connect client supports H.264 and VP9 as video codecs. Video resolution attribute can be used to set one of the following available values: SUBQCIF, QCIF, CIF, 4CIF, VGA, and HD.

If the bit rate is not entered in the configuration, the default bit rate values are used. Default bit rate values, per resolution and frame rate, are listed in the following table.

| Resolution | Video Size * | FPS (Frames Per Second) | Default Bit Rate Values per Resolution and FPS |
|------------|--------------|----------------------------|--|
| SUBQCIF | 128 x 96 | 15 | 128000 |
| QCIF | 176 x 144 | 30 | 192000 |
| CIF | 352 x 288 | 15 | 384000 |
| CIF | 352 x 288 | 30 | 768000 |
| VGA | 640 x 460 | 15 | 2000000 |
| 4CIF | 704 x 576 | 25 | 2000000 |
| HD | 960 x 720 | 30 | 2000000 |

* Maximum advertised video resolution. The actual video resolution during a call between two UC-One Connect clients depends on the capabilities of both clients – it will be the lower of the two and will be the same on both clients.

Video resolution for a video call is negotiated during session setup and is based on the capabilities of the two endpoints. Video call resolution is the same on both UC-One Connect endpoints. That is, if the UC-One Connect endpoints have different capabilities (and therefore support different resolutions), then the lower resolution is negotiated for the call. Video resolution may change during a call if the network conditions deteriorate. In this case, the two Mobile endpoints may be using different video resolutions.

For iPad 2 and iPad Mini, limited video resolutions are supported – CIF and VGA.

Multiple telephone event configuration is also supported. During codec negotiation, the UC-One Connect client sends all the configured codecs, including multiple telephone events. After the audio codec is selected, it searches for telephone events in the offer. If the offer has the telephone event with the sample rate of the negotiated audio codec, then this telephone event is selected. Otherwise, the first telephone event in the list is used. If there are no telephone events in the offer at all, in-band dual-tone multi-frequencies (DTMFs) are used.

Example:

Configured codecs:

```
<codec name="telephone-event" payload="100" in-band="false" />
<codec name="telephone-event" payload="101" samplerate="48000" in-
band="false" />
```

If an audio codec with sample rate of 48kbps is negotiated, the telephone event with payload 101 is used.

6.1.3 Force TCP, TLS, or UDP Usage

The app can be configured to use TCP, Transport Layer Security (TLS), or UDP for both SIP signaling and RTP media. For mobile clients, it is strongly recommended to set the transport layer protocol to TCP/TLS.

The decision to use TCP/TLS or UDP for the client is up to the service provider; however, the suggested recommendation is to use TCP/TLS. The following tags, within the custom *Connect_Tags* or *ConnectTablet_Tags* set, control whether the client uses TCP, TLS, or UDP.

| Tag | Description | Values | M/O |
|------------------------------|--|--|-----|
| %TRANSPORTS_TY PE_MOBILE% | Specifies the chosen protocol for the SIP communication. The default value is TLS. | tcp, udp, tls | O |
| %TLS_VERSION% | Specifies the version of SSL used if TLS is configured as transport for SIP. NOTE: Used for Android only. The default value is ssl23. | ssl23, ssl3, tls1, tls11, tls12 | O |

In addition, when the transports type is set to "UDP", if the Mobile client sends a SIP message exceeding the preconfigured threshold setting (found in the configuration file) then the client defaults and uses TCP instead.

```
<tcp-size-threshold-bytes>18000</tcp-size-threshold-bytes>
```

For TLS, there is an additional option to ignore SSL certificate validation. If such behavior is desired, the value should be set to "true".

```
<tls>
  <ignore-ssl-errors enabled="false"/>
</tls>
```

If the TLS is configured as the transport for SIP, an additional parameter version can be used to specify the SSL version to be used. It can be one of SSL 2&3, SSL 3, TLS 1, TLS 1.1, and TLS 1.2.

SSL negotiation logic is different for iOS and Android. For more information, see the following:

- Android: https://www.openssl.org/docs/man1.0.2/ssl/SSL_CTX_new.html
- iOS: <https://developer.apple.com/documentation/security/sslprotocol?language=objc>

6.1.4 Dynamic SIP Proxy Discovery

To enable SIP dynamic proxy discovery functionality, see the following example.

```
<config version="20">
<protocols><sip>
<proxy-discovery enabled="%USE_PROXY_DISCOVERY_MOBILE%" rfc5922="
%ENABLE_PROXY_DISCOVERY_RFC5922_CERT_VALIDATION_MOBILE%">
<record-name>%SBC_ADDRESS%</record-name>
<domain-override>%DOMAIN_OVERRIDE%</domain-override>
</proxy-discovery>
```

The following tags, in the custom tag set, control this capability.

| Tag | Description | Values | M/O |
|---|--|--|-----|
| %USE_PROXY_DISCOVERY_MOBILE% | Enables dynamic SIP proxy discovery for audio and video calls. The recommended value is "true". The default value is "false". Example: <proxy-discovery enabled="true"> | true, false | O |
| %SBC_ADDRESS% | This Cisco BroadWorks tag is typically used for the record-name parameter. For more information, see the section that follows. Default value: empty Example: <record-name> sbc.domain.com</record-name> | Any valid URL (cannot be an IP address) | O |
| %DOMAIN_OVERRIDE% | This custom tag is used for the domain-override. For more information, see the section that follows. The default value is empty. Example: <domain-override> other.domain.com</domain-override> | Any valid URL (cannot be an IP address) | O |
| %ENABLE_PROXY_DISCOVERY_RFC5922_CERT_VALIDATION_MOBILE% | Controls if the server-side certificate is validated accordingly to RFC 5922. The default value is "true". Set to "false" for backwards compatibility. | true, false | O |

DNS allows UC-One Connect to get the IP address, port, and transport protocol for the SIP proxy as per RFC 3263.

UC-One Connect supports DNS SRV, Naming Authority Pointer (NAPTR), and A-record queries. At login, the 3-step flow is as follows:

- 1) Perform a NAPTR query using the <record-name> field to obtain the server URIs with the transport protocols if they exist. The value for the <record-name> parameter should be the full domain that the DNS is to resolve and cannot be an IP address.

- 2) Resolve items found in the NAPTR query using an SRV query to obtain the final server URI and port. The domain part used in the SRV query is taken from the result of the NAPTR query to find the final server URI (and port). The port received from DNS SRV query is used when the DNS SRV entries are available. Note that the port, only from the configuration file, applies to the static proxy in the configuration file, and not to the URIs resolved using SRV. See the following examples for the usage of the various record names.

If no NAPTR is found, then the client tries an SRV query with the record-name taken from the *<domain>* parameter unless there is a *<domain-override>* parameter present; in which case, *<domain-override>* is used and automatically tries to find separate entries for TCP, UDP, and TLS (*_sip_protocol* [UDP, TCP, or TLS]). Note that the Stream Control Transmission Protocol (SCTP) is not supported. If SRV queries do not yield any results, proxy discovery fails, and the end user is presented with an error indicating that calls are not available. In this case, there is no SIP registration. However, even if all SRV queries fail or if the servers received there do not work, as a fallback, UC-One Connect still checks if the configured static proxy works, only with A-queries to the URI specified in *<proxy address>* to see if it yields an IP address that provides a working SIP registration.

- 3) Resolve found URIs using the A-record query. The received final IP addresses are tried in the order in which they are received to get a working connection to the SIP proxy. This order can be defined by the service provider in the DNS. The first SIP proxy URI, with a successful A-record lookup, is selected and is used until it no longer works, or the client logs out. In the A-query step, only one IP address is used at a time even if many are received. However, all SRV entries are resolved until logout or the loss of the network.

Important notes:

- 1) If DNS proxy discovery is enabled and records are found, the default proxy is not used.
- 2) If the DNS proxy discovery results in transport protocol selection in the SRV step, by receiving a working SIP proxy URI for a transport protocol, then this selection overrides the *tcp-threshold* parameter typically used to select UDP or TCP in the configuration file.
- 3) Items received via SRV are prioritized over the static proxy in the configuration file. The NAPTR priority is not honored. Only SRV priority counts. If SRV priorities are identical, then the first transport is chosen. When SRV results in several items with equal transport protocol, priority, and weight, any one received is selected at random. NAPTR and SRV weights are not supported.
- 4) The optional *domain-override* parameter allows a record name other than the one in the SIP domain configuration parameter to be resolved with SRV. See the following examples for the usage of the *domain-override* parameter.
- 5) UC-One Connect uses operating system primitives for DNS operations and, typically, DNS responses are cached to honor the time to live (TTL) of the DNS response.

Example 1: Using DNS proxy discovery without domain-override configuration parameter

The following is an example of a configuration using SIP proxy discovery when only SIP over TCP is used and NAPTR query in step 1 returns results.

```

<config>
<protocols><sip>
<proxy address="domain.com" port="5060" lr="true" />
```

```
<proxy-discovery enabled="true" rfc5922="true">
  <record-name>record-domain.com</record-name>
  <domain-override>override-domain.com</domain-override>
</proxy-discovery>
<domain>sip-domain.com</domain>
```

This results in the following steps at the protocol level.

```
1. NAPTR query for record-domain.com, answer:
record-domain.com.
28591 IN NAPTR 100 10 "S" "SIPS+D2T" "" _sip._tcp.test.sip.record-
domain.com.
2. SRV query for _sip._tcp.test.sip.record-domain.com (received in the
NAPTR query), answer
_sip._tcp.test.sip.record-domain.com. 28635 IN SRV
10 10 5061 test.sipgeo.record-domain.com.
3. A-record query for test.sipgeo.record-domain.com, answer:
test.sipgeo.record-domain.com. 16 IN A 1.2.3.4
```

As a result, the SIP registration takes place over TCP using port 5061 (received in the SRV step) and towards the IP address 1.2.3.4.

Example 2: Using domain-override parameter in configuration file

The following is a second example of a configuration using SIP proxy discovery where the SIP domain is different from the proxy domain and only SIP over UDP is used and NAPTR query does not return results.

```
<config>
<protocols><sip>
<proxy address="domain.com" port="5060" lr="true" />
  <proxy-discovery enabled="true" rfc5922="true">
    <record-name>record-domain.com</record-name>
    <domain-override>override-domain.com</domain-override>
  </proxy-discovery>
</domain>sip-domain.com</domain>
```

This results in the following steps at the protocol level.

```
1. NAPTR query for record-domain.com, no answer.
2. SRV query for _sip._ucp.override-domain.com (from configuration file),
answer
_sip._ucp.override-domain.com. 28635 IN SRV
10 10 5061 test.override-domain.com.
3. A-record query for test.override-domain.com, answer:
test.sipgeooverride-domain.com. 16 IN A 4.3.2.1
```

As a result, the SIP registration takes place over UDP using port 5061 (received in the SRV step) and towards the IP address 4.3.2.1.

To disable DNS NAPTR and use only DNS SRV and A-record queries:

- Enable proxy discovery.
- Make sure the *record-name* tag is empty.
- Make sure the *domain-override* tag contains the domain for the DNS SRV.
- Configure the proper transport protocol in the corresponding section of the configuration file.

6.1.5 Preferred-Port Usage for SIP

There have been some cases when another software package has been running on the same machine as the client, occupying the default SIP port. To configure the client to use another port for SIP, the *preferred-port* parameter can be used. The client tries to use the configured port value specified in the *preferred-port* parameter, but if it is taken, the client incrementally tries port values above the configured value.

For example, if the value of the *preferred-port* is “6000” and that port is taken, the client tries 6001, 6002, 6003, and so on until it finds an unused port. Once an unused port is found, it uses that for its own SIP communication.

6.1.6 SIP Failover and Failback

SIP failover follows the Cisco BroadWorks procedures for failover and failback. For this, more than one proxy (typically the SBC) must be configured.

On the client side, the proxy should be resolved to multiple IP addresses. This can be achieved by either:

- SIP Proxy Discovery is enabled, and the DNS server has NAPTR and/or SRV records for the SBC FQDN, or
- The SIP proxy address is provided as an FQDN and it is resolved to multiple IP addresses.

At sign-in, if multiple IP addresses are resolved, they are ordered by priority. The UC-One Connect client starts using the first available IP address. SIP failover may be triggered by either a socket error or a request timeout error:

- Socket error – if the socket between the client and the server gets broken or is closed, as in the case of network connectivity loss, the clients react immediately and trigger a failover.
- Timeout (for example, when the SBC hangs) – based on the SIP T1:
 - SIP INVITE – if the INVITE request times out, the client registers to the next available SBC (IP) and retries the INVITE.
 - Other SIP request – the client tries to register to the next available SBC (IP).

When a failover is triggered, the client takes the next available IP address from the list. If all IP addresses fail, then the client displays a user interface error for SIP connectivity. If a VoIP call is in progress when the failover occurs, the call is terminated.

The SIP failover logic relies on several configuration parameters:

- SIP Failover Timers – SIP timers T1, T2, and T4 are exposed in the configuration file, but is not recommended to modify them.

```
<config><protocols><sip>
<timers>
  <T1>500</T1>
  <T2>4000</T2>
  <T4>5000</T4>
</timers>
```

- T1 – the amount of time, in milliseconds, for a network round trip delay
- T2 – the maximum amount of time, in milliseconds, before retransmitting non-invite requests and invite responses

- T4 – the maximum amount of time, in milliseconds, for a message to remain in the network
- SIP Proxy Address and SIP Proxy Discovery (for more information, see section 6.1.4 [Dynamic SIP Proxy Discovery](#)).

If the UC-One Connect client is connected to a proxy that is not first by priority, it tries to reconnect to the IP with the highest priority. The time for the failback is fixed on 15 minutes and it is not configurable. If a call is in progress when the failback timer is reached, the UC-One Connect client waits until all calls are completed and triggers the failback procedure.

For initial SIP REGISTER failover, there is a separate configuration parameter that can be used to override the standard timeout timer (64*T1) and after which the client will try next available address. This timer may be enabled, and the timeout value can be specified using the following tags in the custom *Connect_Tags* or *ConnectTablet_Tags* sets.

| Tag | Description | Values | M/O |
|-------------------------------|---|-------------|-----|
| %ENABLE_SIGN_IN_TIMER_MOBILE% | Enables the timer. The default value is "false". | true, false | O |
| %SIGN_IN_TIMER_PERIOD_MOBILE% | Specifies the timeout period in milliseconds. The default value is "5000". | Integer | O |

6.1.7 Q-Value

During a SIP failover, UC-One Connect clients can have multiple REGISTER sessions at the same time. In this case, different SBCs have different requirements for the *q-value*, defined in the *Contact* header of SIP REGISTER requests. Some require that the REGISTER session that is terminated should have higher weight, while others require lower weight.

To support different SBCs, UC-One Connect clients use the *q-value* defined in the sip section of the configuration file. The values range is 0.0 to 1.0, with the default value set to "1.0".

REGISTER sessions that should be terminated always use the *q-value* in the *Contact* header set to "0.5".

6.1.8 Call Pull (Call Retrieve)

The Call Pull function enables a user to retrieve an active call on any of their shared telephony endpoints or devices through the UC-One Connect Mobile client. Using the dedicated "Pull Call" option in the *Call* screen is equivalent to dialing the *11 (default) feature access code typically used to activate this function.

This feature is configurable for Mobile client deployments through the *Connect_Tags* or *ConnectTablet_Tags* sets identified in the following table.

| Tag | Description | Values | M/O |
|---------------------------|---|-------------|-----|
| %ENABLE_CALL_PULL_MOBILE% | Controls if the Call Pull/Retrieve option should be available for the user. The default value is "true". | true, false | O |

6.1.9 Reject Behavior

The UC-One Connect Mobile client offers the flexibility to reject a call with *486* or *603*.

Note that if the client is configured to reject a call with *603 Decline*, then the Call Forward Busy and Call Forward No Answer services may not work as expected.

| Tag | Description | Values | M/O |
|---------------------------------|---|----------------|-----|
| %ENABLE_REJECT_WITH_486_MOBILE% | Controls the SIP code and reason used to reject incoming SIP calls. If enabled, <i>486 Temporarily Unavailable</i> is used. Otherwise, <i>603 Decline</i> is used. The default value is "false" (<i>603 Decline</i>). | true, false | O |

6.1.10 Mid-Call Control Services

For the UC-One Connect Mobile client, Mid-Call Control services are accessible to the user through the device's touch screen. A user can invoke any of the following mid-call actions:

- Hold/Resume
- Transfer (blind and consultative)
- Transfer call to circuit-switched
- Add participants
- End call

Furthermore, creating the following Transfer Call and/or Conference Call (Add participants) tags in the *Connect_Tags* or *ConnectTablet_Tags* sets provides the ability to customize these Mid-Call Control options (instructions are described in section [5.6 Custom Tags](#)).

NOTE: These options are enabled by default. However, if no customization is required, then these tags are not necessary.

6.1.10.1 Transfer Call

The Transfer Call configuration node controls transfer options available for the user. Possible values for the type of transfer are:

- "full" – Both attended and blind transfers are enabled.
- "blind" – Only blind transfer is enabled.
- "talk-first" – Only attended transfer is enabled.

NOTE: Transfer Call is enabled by default. If no customization is required, then the tags are not necessary.

Creating the following Transfer Call tags in the *Connect_Tags* or *ConnectTablet_Tags* sets provides the ability to customize the Transfer Call option.

| Tag | Description | Values | M/O |
|--------------------------------|--|-------------------------|-----|
| %ENABLE_TRANSFER_CALLS_MOBILE% | Controls if the Transfer option should be enabled for the user. The default value is "true". | true, false | O |
| %TRANSFER_CALL_TYPE_MOBILE% | If the Transfer option is enabled for the user, this tag controls the transfer options provided to the user. The default value is "full". | full, blind, talk-first | O |

6.1.10.2 Conference Call

| Tag | Description | Values | M/O |
|----------------------------------|---|-------------|-----|
| %ENABLE_CONFERENCE_CALLS_MOBILE% | Controls if the Conference option should be enabled for the user. The default value is "true". | true, false | O |

NOTE: For a conference to work as expected, a proper conferencing URI must be configured for *%BWNWORK-CONFERENCE-SIPURI-1%*.

6.1.10.3 Transfer Call to Circuit-Switched

| Tag | Description | Values | M/O |
|--|---|-------------|-----|
| %ENABLE_TRANSFER_TO_CIRCUIT_SWITCH_MOBILE% | Controls if Transfer to Circuit-Switched option should be enabled for the user. The default value is "true". | true, false | O |

6.1.11 Conference Calls Bridge

| Tag | Description | Values | M/O |
|---------------------------------------|---|--------------|-----|
| %CONFERENCE_BRIDGE_MEDIA_TYPE_MOBILE% | This controls the bridge type. The default is "true". Used only for Meet-Me. | audio, video | O |
| %CONFERENCE_TYPE% | This should be set to the generic <i>Title</i> corresponding to the users' primary personal conference in Cisco BroadWorks. The <i>Title</i> is used (by the client) to search through the list of conferences associated with the user's primary bridge to establish a match. If a match is found, then the client subsequently sends an Xsi query to retrieve the conference's respective Moderator PIN and Security PIN (if one exists). If no match is found, then the client attempts to create one. NOTE: A <i>Title</i> must be defined; otherwise, auto provisioning is not possible. | meetme, uvs | O |

| Tag | Description | Values | M/O |
|-------------------------|--|-------------|-----|
| %AUTODETECT_CONFERENCE% | Controls whether the audio bridge associated with My Room should be auto provisioned via Meet-Me Conferencing. The default value is "true". Used only for Meet-Me. NOTE: UC-One Connect client version 3.6.5 supports only "true", where only auto provisioning is used in the client. | true, false | O |
| %CONFERENCE_TITLE% | The name of the conference. Example: "MyRoom". | string | O |
| %BRIDGE_ID% | If the UC-One Connect client needs to create a Meet-Me room, it should create it under the bridge given here (<bridged>). If the bridge is not found, the client should give up trying to create a room. If the attribute is empty or missing, the client should use the first active bridge it finds. Example: meet-me@company.com Used only for Meet-Me. | string | O |

6.1.12 100rel – SIP Provisional Message Reliability

The following UC-One Connect client tag, created in the tag set, controls insertion of the 100rel in the SIP header.

| Tag | Description | Values | M/O |
|-------------------------|---|-------------|-----|
| %100_REL_ENABLE_MOBILE% | This is used to enable insertion of 100rel. The default is "true". | true, false | O |

6.1.13 SIP Info

The following tag is intended to control SIP Info packets. Note that some Cisco BroadWorks services rely on SIP INFO. Therefore, the disabled state is intended only for specific environments.

| Tag | Description | Values | M/O |
|--------------------------|--|-------------|-----|
| %ENABLE_SIP_INFO_MOBILE% | This is used to enable SIP Info packets. The default is "true". | true, false | O |

6.1.14 QoS – Diffserv Support

The following UC-One Connect client tags enable the usage of Diffserv to improve the audio and video quality.

| Tag | Description | Values | M/O |
|---------------------------|--|-------------|-----|
| %ENABLE_AUDIO_QOS_MOBILE% | This is used to enable Quality of Service (QoS) for the audio stream. The default is "false". | true, false | O |

| Tag | Description | Values | M/O |
|--------------------------|--|-----------|-----|
| %AUDIO_QOS_TYPE_MOBILE% | This is the QoS type for audio. | dscp, tos | O |
| %AUDIO_QOS_VALUE_MOBILE% | This is the QoS value to be used for the audio stream. | integer | O |

6.1.15 Adaptive Call Quality

UC-One Connect clients provide the ability to choose best audio and video codec configuration based on device's capabilities. Adaptive call quality for audio helps selecting the most suitable codec prior establishing the call. However, there are no audio stream adjustments while the call is ongoing. For video, it not only helps selecting the most suitable video parameters prior to establishing the call, but also monitors the network conditions and CPU load of the device and can downscale the video parameters to achieve better user experience.

The behavior of this feature can be controlled using the following tag.

| Tag | Description | Values | M/O |
|----------------------------------|---|-------------|-----|
| %ENABLE_ADAPTIVE_QUALITY_MOBILE% | This is used to enable adaptive call quality logic. The default is "false". | true, false | O |

- Adaptive quality logic uses different configurations of the audio/video settings specific for the network type.
- Default codecs order and configuration is optimized to achieve best performance.
- Attribute *type* of the *network* tag accepts values "wifi" or "3G". The "3G" mode is applied for all the cases where cellular network is used by the device (like 4G).

6.1.16 Expires Hack

UC-One Connect clients provide the ability to configure the re-register behavior to use either the value negotiated with the server or to always use the value from the configuration file. The latter option is needed only for specific server environments where the server is configured to return a smaller value after every register.

| Tag | Description | Values | M/O |
|---------------------------|---|-------------|-----|
| %USE_EXPIRES_HACK_MOBILE% | Enables the use of the "expires hack". If "true", then the expires value is always sent from the configuration file. If "false", then it uses the expires value returned by the server. The default is "false". | true, false | O |

6.1.17 Federated Calling in IM&P Deployments

UC-One Connect only uses the following data from the vCard: Avatar, SIP URI, My Room, bridge number, MUC room ID, Web Collab address.

Currently, UC-One Connect does not populate the vCard with any Meet-Me bridge information, and certainly no phone numbers.

To force the phone number in the vCard to be in international format, use the new DM tag, %BWE164-1%. Note that %BWE164-1% is currently only supported in Cisco BroadWorks Release 20.0.

```

<config version="20">
<protocols><sip>
  <phone-number>%BWE164-1%</phone-number>

```

The following table describes the DM tag for federated calling.

| Tag | Description | Value | M/O |
|------------|---------------------------------------|--------------|-----|
| %BWE164-1% | Phone number in international format. | Phone number | O |

6.1.18 Voice Mail Number

The number that the client dials when accessing the voice mail can be configured (see the following example). If the voice mail number is empty, UC-One Connect client will dial the number assigned to the current user.

There is no configuration option that controls the availability of the Voice Mail for the user. Instead, service assignment and a branding property are used. To have Voice Mail available for the user:

- Assign “Voice Messaging User” or “Third-Party Voice Mail Support” service to the user.
- The branding property *callvoicemaildisabled* should be set to “false” or left empty (defaults to “false”).

See the following table for more information on configuring voice mail.

| Tag | Description | Value | M/O |
|---------------------------|---|--------|-----|
| %BWVOICE-PORTAL-NUMBER-1% | If configured with a non-empty number, this number will be dialed to access the Voice Mail. If left empty, personal number assigned to the user will be dialed. | number | O |

6.1.19 Network Quality Indicator

UC-One Connect clients display a network quality indicator during a call. It is based on the Jitter Buffer statistics coming from the RTCP, provided by the Media Engine. There are three levels based on Jitter Buffer values:

- Below the normal jitter – excellent quality
- Between the normal and the max jitter – good quality
- Above the max jitter – bad quality

6.1.20 3GPP SIP Headers for SRTP

Newer 3GPP specifications require additional SIP headers to use Secure Real-time Transport Protocol (SRTP). For more information, see 3GPP TS 24.229 as well as the following:

<https://tools.ietf.org/html/draft-dawes-dispatch-mediasec-parameter-07>

The headers required by this specification may break SIP calling in deployments where this specification is not used. Therefore, these headers are recommended to be used only in environments where the server side supports them.

Only enabling the usage of the headers is configurable. No further configurability exists for individual headers. All headers are either enabled or disabled.

The following tag controls this capability.

| Tag | Description | Values | M/O |
|-----------------------|---|----------------|-----|
| %USE_MEDIASEC_MOBILE% | Enables 3GPP SIP headers for SRTP negotiation. The default is "false". | true, false | O |

6.1.21 User Selectable Control of VoIP Mode

The UC-One Connect client can be configured to control VoIP calls, depending on user's and customer's needs. If the VoIP mode functionality is enabled, the user can control if the calls go through VoIP or CS (using Call Back or Call Through). If the VoIP mode is turned ON, then all calls, incoming and outgoing, go as VoIP. If the VoIP mode is turned OFF, then all calls go through CS. If the feature is disabled through the configuration, it disables VoIP calls (the app is not registered for Push Notifications for calls) and hides the VoIP Calling UI.

There is a separate configuration option that can be used to configure VoIP call restrictions while using cellular data. The UC-One Connect client can be configured to allow or restrict VoIP calls over cellular data, or to allow users to configure these restrictions themselves.

Starting with Release 3.9.4, the UC-One Connect client can be configured so that only VoIP calls are available for the user. If the user-selectable VoIP mode is enabled, and only VoIP calls are enabled as a dialing option, the user will be always in VoIP mode turned ON mode, without the ability to switch it OFF.

VoIP Only configuration:

- %ENABLE_VOIP_MODE_MOBILE% = true
- %ENABLE_NATIVE_DIALER% = false
- %ENABLE_DIALING_CALL_THROUGH% = false
- %ENABLE_DIALING_CALL_BACK% = false
- %DEFAULT_DIALING_USE_VOIP% = true
- %DEFAULT_DIALING_METHOD% = empty

```
<services>
<calls>
  <voip-mode enabled="true"/>
...
<dialing>
```

```

<default></default>
<native>>false</native>
<callthrough>>false</callthrough>
<callback>>false</callback>
<defaultUseVoip>>true</defaultUseVoip>

```

Also, starting with Release 3.9.4, a new configuration option is added that allows the user to be warned for additional charges or roaming fees, when an outgoing VoIP call is initiated, using the Mobile Data Network.

| Tag | Description | Values | M/O |
|---|---|---|-----|
| %ENABLE_VOIP_MODE_MOBILE% | Allows the user to control the VoIP mode. The default is "false". | true, false | O |
| %CELLULAR_DATA_VOIP_CALLS_STATE_MOBILE% | Specifies if VoIP calls over Mobile Network can be controlled by the user or to be hidden. If they are hidden, they can be set to be always enabled or disabled. The default is "always-on". | always-on, always-on-warn, always-off, user-control | O |

NOTE: If User Selectable VoIP Mode is disabled for the user, the %DEFAULT_DIALING_USE_VOIP% should be set to "false".

6.1.22 SIP UPDATE

The UC-One Connect client can be configured to support incoming SIP UPDATE requests. The following tags are intended to control the SIP UPDATE availability in the allowed headers.

| Tag | Description | Values | M/O |
|------------------------------------|---|-------------|-----|
| %ENABLE_SIP_UPDATE_SUPPORT_MOBILE% | This is used to enable SIP UPDATE support. The default is "false". | true, false | O |

6.1.23 SIP rport Management for NAT Traversal

The client can be configured to use the SIP rport mechanism for NAT traversal. Note that, typically, it cannot be the only solution for NAT traversal and SBC is mainly used for this purpose. For a description of the rport specification, see *RFC 3581*.

For details on SIP port and transport protocol recommendations when SIP Application Layer Gateways (ALGs) are used in the network, see the *UC-One Solution Guide*.

Configuration parameter controls whether the "rport" string to be present in the outgoing SIP requests. This parameter only impacts the usage of IP address and port received from the server in the SIP "received" and "rport" headers. When the feature is enabled, the values from "received" and "rport" headers are used in the SIP Contact header of SIP requests (even when the "received" header is missing in the REGISTER response).

There is a separate configuration parameter *use-local-port* that forces local port of the client socket to be set in the *Contact* header. This is used for some SBCs that detect the client has a real IP (from the *Contact* header) and the SBC tries to establish a separate socket to the client for its requests. In most cases, a firewall sits between the SBC and the client, and it denies the incoming connections to the client.

NOTE: In IPv6 environments, all the addresses are real, and the SBC tries to establish a connection to the listening client address (from the *Contact* header).

To force the SBC to use the already established client's socket connection for its requests, enable the *use-local-port* parameter, to put the local port of the client's socket in the *Contact* header.

The *preferred-port* parameter is related in that it otherwise defines the port used in the SIP *Contact* header. For more information on SIP port allocation, see section [6.1.5 Preferred-Port Usage for SIP](#).

| Tag | Description | Values | M/O |
|-----------------------------------|---|----------------|-----|
| %ENABLE_USE_RP ORT_MOBILE% | Enables usage of rport. The default value is "false". | true, false | O |
| %RPORT_USE_LOC AL_PORT_MOBILE% | Controls whether the local port of the client's socket to be added in the SIP <i>Contact</i> header. The default value is "false". | true, false | O |

6.1.24 iOS CallKit Integration

iOS UC-One Connect adds support for CallKit, introduced in iOS 10. Using the CallKit, VoIP calls can be accepted on the iPhone lock screen without the need to unlock the device. In addition, VoIP calls are no longer interrupted by the incoming circuit-switched calls but are handled as standard second incoming call – the user can reject the incoming circuit-switched call without interrupting the ongoing VoIP call.

CallKit integration is not configurable. It is always enabled for UC-One Connect running on iOS 10 and later.

CallKit requires Push Notifications for calls.

6.1.25 Call Auto Recovery

UC-One Connect clients have support for call auto recovery at switching networks while the user has an ongoing VoIP call. Call auto recovery works in both directions – Cellular Data-to-WiFi and WiFi-to-Cellular Data. The call is tried to be recovered within a one-minute timeframe and then stops. If there are more than one ongoing VoIP call, just the active one is recovered.

In Cellular Data-to-WiFi transition, the client can be configured to keep the ongoing VoIP calls on cellular data, until terminated.

To enable call auto recovery and to select the Cellular Data-to-WiFi mode, use the tags in the following table, in the custom tags set.

| Tag | Description | Values | M/O |
|--|---|-------------|-----|
| %ENABLE_CALLS_AUTO_RECOVERY_MOBILE% | Controls if the auto recovery mechanism should be enabled for the user. The default value is "false". | true, false | O |
| %CALLS_AUTO_RECOVERY_DATA_TO_WIFI_MODE_MOBILE% | Controls the auto recovery mode for the Cellular Data-to-WiFi transition – if the call should remain on the cellular data or if it should be transferred automatically. The default value is "keep". | keep, auto | O |

NOTE: Chat and sharing connections are reconnected immediately in both Cellular Data-to-WiFi and WiFi-to-Cellular Data transitions.

6.1.26 Real Time Protocol

6.1.26.1 Preferred Media Ports

UC-One Connect Mobile clients provide the ability to configure the preferred audio and video port ranges.

| Tag | Description | Values | M/O |
|------------------------------|---|---------|-----|
| %RTP_AUDIO_PORT_RANGE_START% | The first possible port to be used for audio. The default is "8500". | integer | O |
| %RTP_AUDIO_PORT_RANGE_END% | The last possible port to be used for audio. The default is "8598". | integer | O |

6.1.26.2 Secure Real-time Transport Protocol

In addition to Real-time Transport Protocol (RTP), UC-One Connect also supports the Secure Real-time Transport Protocol (SRTP). Note that it should be configured on the server side for the client to operate successfully.

The current version of the media framework supports Advanced Encryption Standard (AES) 128 Counter Mode for protection and Hash Message Authentication Code (HMAC)-SHA-1 for authentication. The master key size is 16 bytes and the master salt size is 14 bytes. The media framework supports both the full (80-bit) and short (32-bit) authentication tags.

Different SRTP operating mode negotiations are supported.

When offering:

- Mandatory SRTP – Initial Session Definition Protocol (SDP) contains only the Secure Audio Video Profile (SAVP)-profile. The other party must honor this (that is, the call cannot be unencrypted).
- Preferred SRTP – Initial SDP first contains the SAVP-profile, then the Audio Video Profile (AVP)-profile, indicating an order of preference.

- Optional SRTP – Initial SDP first contains the AVP-profile, then the SAVP-profile, indicating an order of preference.
- Not enabled – Only the AVP-profile is in the SDP.

When answering to an offer:

- Mandatory SRTP – Only the media with crypto-attribute is accepted. SAVP if it exists, AVP if it has “crypto”.
- Preferred SRTP – SAVP is picked from the offer, no matter if it is the second or third option.
- Optional SRTP – First valid media is accepted with or without “crypto”.
- Not enabled – No SRTP is allowed.

Also, the UC-One Connect client can be configured to enable the SRTP dynamically, based on the transport used for the SIP communication – SRTP is enabled in mandatory mode only if TLS is used and in case of TCP or UDP – SRTP is disabled.

To select this mode of operation, configure the SRTP mode to “transport”.

NOTE: The Secure Real-Time Control Protocol (SRTCP) is also used when SRTP is enabled.

| Tag | Description | Values | M/O |
|-----------------------|---|--|-----|
| %SRTP_ENABLED_MOBILE% | This is used to enable SRTP usage. The default is “false”. | true, false | O |
| %SRTP_MODE_MOBILE% | This is the operational mode of the SRTP. The default is “optional”. | mandatory, preferred, optional, transport | O |

NOTE: The SRTP is by default disabled in the UC-One Connect application. It is always recommended to use maximum level of security. To achieve it, enable SRTP and configure “mandatory” mode.

6.1.26.3 SRTP Re-keying Configurability

In some deployments, re-keying for SRTP is not supported. The UC-One Connect client can be configured to enable/disable SRTP re-keying to match the deployment setup. New keys are always taken into use when received. Configurability only pertains to sending new keys.

The following tag controls this capability.

| Tag | Description | Values | M/O |
|---------------------------|---|-------------|-----|
| %ENABLE_RE-KEYING_MOBILE% | Enables SIP (SDP) re-keying for SRTP. The default is “true”. | true, false | O |

6.1.26.4 Real-Time Control Protocol Extended Reports

UC-One Connect can send Real-Time Control Protocol (RTCP) Extended Report (XR) when configuring the following tags.

| Tag | Description | Values | M/O |
|-------------------------------------|---|-------------|-----|
| %CALL_QUALITY_AUDIO_ENABLED_MOBILE% | This is used to enable an RTCP XR report for the audio stream. The default is "false". | true, false | O |
| %CALL_QUALITY_SERVICE_URI% | This is the location (or URI) where the reports should be sent. This is required if RTCP is enabled. | string | O |
| %CALL_QUALITY_LOCAL_GROUP% | This is the local group where the reports should be sent. | string | O |

6.1.27 Visual Voice Mail

Starting with Release 3.1.1, Visual Voice Mail (VVM) is supported in the UC-One Connect client for audio only. It allows users to view incoming voice mails in a list view, available from the Voicemail option in the left side navigation menu. To make this feature available:

- VVM should be enabled in the configuration file
- Voice Messaging User service should be assigned for the user

For Cisco BroadWorks release and patch requirements for Visual Voice Mail, see the *UC-One Solution Guide*.

The following tag is used to enable Visual Voice Mail.

| Tag | Description | Values | M/O |
|-----------------------------------|--|-------------|-----|
| %ENABLE_VISUAL_VOICE_MAIL_MOBILE% | This is used to enable VVM. The default is "false". | true, false | O |

6.1.28 Emergency Calls

6.1.28.1 Emergency Number Management

For every VoIP call, the client tries to determine whether the user dialed an emergency number by applying the following logic:

- Does the dialed number match the emergency number configured in the configuration file for the given user?
 - If no match is found in the configuration file, then the client searches the list of emergency numbers provided by the mobile device's operating system (available only through Android) and returns an emergency number by specific country.
 - If a match is found, then the client considers the dialed number to be the emergency number and initiates the call over the Public Land Mobile Network (PLMN). If a match is not established, the call is handled as a standard VoIP call.

```

<config>
  <services>
    <emergency-dialing>
      ...

```

```

        <numbers check-device-emergency-numbers="false">
            <number>911</number>
            <number>112</number>
            ... <!-- add more number nodes here -->
        </numbers>
    </emergency-dialing>
</services>
</config>

```

UC-One Connect Release 3.2.2 added a new configuration option for the Android client to control if the application should check the emergency numbers of the Android OS. If enabled, the UC-One Connect client checks if the dialed number is defined as emergency one by the OS and checks against the numbers listed in the configuration file. If disabled, just the dialed numbers are compared against the list of the emergency numbers in the configuration file.

| Tag | Description | Values | M/O |
|---|---|-------------|-----|
| %CHECK_DEVICE_EMERGENCY_NUMBERS_MOBILE% | Specifies whether the dialed number should be compared against the emergency numbers list provided by the Android OS. The default value is "true". | true, false | O |

6.1.28.2 Update Emergency Location Dialog

It is possible to enable a separate dialog to be shown after the login window, which is used to show the user important information about their emergency location. The text displayed in this dialog can be changed as part of the branding process.

Updating the user’s emergency location may be configured as required or optionally using the required attribute of the update-location-at-login tag. The buttons available in this dialog vary depending on configured mode:

- Required
 - Update Location – uses the configured URL to connect to the ECACS, described in section [6.1.28.3 Emergency Call Address Change Service \(ECACS\)](#).
 - Cancel – terminates the sign-in process.
- Optional
 - Update Location – opens an external browser with the configured URL.
 - OK – continues the sign-in process without emergency location update.
 - Cancel – terminates the sign-in process.

NOTE: The URL used for both optional and required modes is configured in the *url* tag of the *update-emergency-location* section. It is the administrator’s responsibility to configure the right URL depending on the emergency location update mode.

The URL should have as a final parameter a phone number (http://emergency_url/?tn=phone_number). It is always used for the required mode. For the optional mode, it may be discarded by the page that is being hit but should be part of the URL.

The phone number is configured using %BWDN-1%. It is user-specific, representing the primary phone number of the user without the country code.

If the emergency location update is configured as optional, there is a configuration parameter (the *recurring* attribute of the *update-location-at-login* tag) that can be used to specify if the emergency location update dialog should appear once or every time at sign-in. This applies per installation and not per user.

In required mode, there is a configuration parameter (the *verification-timeout* attribute of the *url* tag) that specifies a timeout in seconds, after which update location verification is considered as failed.

A separate option (tag *update-location-at-will*) controls if the user can update their emergency location after they sign in to UC-One Connect. If enabled, the user can see a separate menu option that can be used to initiate the emergency location update process.

Emergency calls dial behavior is configurable using the *dial-sequence* tag. Supported modes are:

- *cs-only* – UC-One Connect client places emergency calls only through the cellular network if the network is available.
- *cs-first* – Upon initiating an emergency call, the UC-One Connect client checks the network type to which the current device is connected. If the cellular network is available, the client places that call over the cellular network. If the cellular network is not available but a cellular data/WiFi network is available, the client places the call over the cellular data/WiFi network as a VoIP call. Also, if the emergency call is placed through the cellular network, the UC-One Connect client suggests to the user to retry the emergency call as VoIP.
- *voip-only* – UC-One Connect client places emergency calls only as VoIP if the cellular data/WiFi network is available.

The following table describes the DM tags.

| Tag | Description | Values | M/O |
|---|---|--------------|-----|
| %ENABLE_UPDATE_EMERGENCY_LOCATION_MOBILE% | Enables emergency location updates. The default value is "false". | true, false | O |
| %UPDATE_EMERGENCY_LOCATION_URL_MOBILE% | Used for both optional and required modes. The URL to the external update location service (web page or ECACS). Example: http:// emergency_url/?tn= The default value is empty. | string (URL) | O |
| %UPDATE_EMERGENCY_LOCATION_VERIFICATION_TIMEOUT_MOBILE% | Used in required mode. Specifies a timeout in seconds, after which the update location verification is considered to have failed. The defaults value is "900". | integer | O |
| %USE_ECACS_BASIC_AUTH_MOBILE% | Used in required mode. Specifies if basic authentication should be used for the web page displayed for 911 address change. The default value is "false". | true, false | O |
| %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_LOGIN_MOBILE% | Enables an additional step at login, during which the user can update their emergency location. The default value is "false". | true, false | O |

| Tag | Description | Values | M/O |
|---|---|------------------------------------|-----|
| %UPDATE_EMERGENCY_LOCATION_REQUIRED_MOBILE% | Specifies if the user is required to enter/update/confirm their emergency location before logging in. The default value is "false". | true, false | O |
| %UPDATE_EMERGENCY_LOCATION_RECURRING_MOBILE% | Used in optional mode only. Specifies if update of the emergency location should appear once or always at login. The default value is "always". | always , one-time | O |
| %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_WILL_MOBILE% | Specifies if the emergency location update should be possible from a menu item inside the application while the user is signed in. The default value is "false". | true, false | O |
| %EMERGENCY_CALL_DIAL_SEQUENCE_MOBILE% | Controls the dial-sequence mode for emergency calls. The default value is "cs-only". | cs-only, cs-first, voip-only | O |

Use the following configuration to configure the UC-One Connect client with:

- Required emergency location update.
- Allow emergency location update after sign-in.
- Initiate CS call with priority and then suggest emergency calls as VoIP.

```

<config><services>
<emergency-dialing>
  <update-emergency-location enabled="true">
    <url verification-timeout="900" use-basic-
auth="false">http://ecacs_url/?tn=9988776655</url>
    <update-location-at-login enabled="true" required="true"
recurring="always"/>
    <update-location-at-will enabled="true"/>
  </update-emergency-location>
  <dial-sequence mode="cs-first"/>

```

6.1.28.3 Emergency Call Address Change Service (ECACS)

The Emergency Call Address Change Service exposes web services that allow the UC-One Connect client to create a web browser session that allows a user to configure their emergency location address. At login, end users receive a dialog asking if the emergency call service address needs to be updated. If the end user decides to update their emergency location, the ECACS service is invoked. An external web browser is opened where the user can update their emergency location. At the same time, the UC-One Connect client monitors the emergency location update.

In the UC-One Connect client, the ECACS service is used only at login, if the emergency location update is a required step to use the application.

For information about the server-side implementation requirements from the client perspective, see the *Communicator Emergency Call Address Change Service Guide*.

6.1.29 RTP MUX

UC-One Connect supports RTP MUX. When enabled, it is advertised and negotiated in the SIP INVITE session. If negotiated, the RTP and RTCP streams use same port as per *RFC 5761*.

If SRTP is enabled, the RTP MUX is disabled.

| Tag | Description | Values |
|-----------------------------|--|-------------|
| %ENABLE_RTP_MUX_MO BILE% | Enables RTP MUX. The default value is "true". | true, false |

6.1.30 Bridge Dialing

Starting with Release 3.5.1, when dialing a bridge number, the UC-One Connect client adds support for configurable time to wait before the DTMFs for the bridge ID to be played.

The configurable time is specified as a number of pauses (",") added in the dialed number. The number of pauses can be set between 0 and 10 and specifies the number of "," added between the bridge number and the bridge ID.

| Tag | Description | Values | M / C |
|--|--|---------|-------|
| %CONFERENCE_CALLS_A UTO_JOIN_PAUSE% | The number of pauses (",") added between the bridge number and bridge ID. The default is "2". | integer | C |

6.1.31 One-time VoIP Call

Starting with Release 3.6.1, UC-One Connect allows the user to initiate one-time VoIP call in configurations where VoIP calls are not explicitly available, or VoIP mode is turned off, and cellular calls are used as the default calling option.

A VoIP audio or video call can be initiated in such configuration by using a long press on a Call button and selecting the VoIP call option from the menu.

(Single press on a Call button uses the default calling mode and initiates cellular call).

No SIP REGISTER session is established for the one-time VoIP calls. This requires certain configuration at the Application Server (AS):

- denyCallsFromUnregisteredUsers = false
- inviteAuthenticationRatio = 1

The one-time VoIP calls feature applies for outgoing calls only and is suitable for operators that use mobile (circuit switched) as the default calling option (for example, Call Through, Call Back, iPhone).

Starting with Release 3.9.4, the Android UC-One Connect client introduces an additional configuration option that controls whether SIP REGISTER session to be established before the SIP INVITE session. For some deployments, SIP INVITE sessions are not allowed if there is no active SIP REGISTER session for the user. For this type of deployment, use the new *register* configuration attribute to force SIP REGISTER session to be established before the VoIP call. Note that for this type of configuration, the Call Termination should be manually disabled for the Shared Call Appearance (Connect – Mobile/Connect – Tablet) used by the Mobile client. If the Call Termination is not disabled, the user may experience double alerting during call waiting.

| Tag | Description | Values | M/O |
|--------------------------|---|-------------|-----|
| %ENABLE_ONE_TIME_VOIP% | Enables the one-time VoIP calls. The default is "false". | true, false | O |
| %ONE_TIME_VOIP_REGISTER% | Controls if SIP REGSITER session should be established. The default is "false". | true, false | O |

NOTE: This feature relies only on this configuration tag and the availability of the corresponding license (service) assigned to the user. It does NOT rely on the values of the %ENABLE_AUDIOCALLS_MOBILE%, %ENABLE_VIDEOCALLS_MOBILE%, and %ENABLE_VOIP_MODE_MOBILE%.

6.1.32 Incoming Audio Calls

When an incoming call is received, the UC-One Connect client receives a push notification (PN) first. There is a config parameter that can be used to control when the SIP REGISTER session to be established:

- 1) When the push notification is received, OR
- 2) When the call is accepted by the user.

The second approach is recommended. However, compared to first case, it adds some delay before the call is established.

Starting with Release 3.9.12, UC-One Connect adds a configurability option to control the PN types used for the call-related events. According to the iOS13 requirements, the VoIP PNs should be used only for incoming calls. The rest of the call-related events should use regular PNs.

To meet this requirement, a new PN registration API is introduced which requires a corresponding patch to be applied on the Application Server. If the backend is not configured to support the iOS13 PNs, the configuration parameter can be used to enforce usage of the legacy push notifications, where all call-related events are delivered through VoIP PNs.

Starting with Release 3.9.16, UC-One Connect for iOS adds support for configurable ringing timer and introduces incoming call status check every 10 seconds. Due to the latest changes in the push notifications, introduced by Apple with iOS13 SDK, regular push notifications may be missed or delayed by the APNS. To handle this, the UC-One client behavior is modified to check the status of the incoming calls, while in ringing state. The ringing timer specifies the maximum duration of ring time controlled by the application. If the timer is set to 0, the check for the call state (every 10 seconds) will continue until the call is detected as accepted on another location or terminated (either at receiving the Call Update Push Notification or detecting it through the XSI requests).

| Tag | Description | Values | M/O |
|---|--|-------------|-----|
| %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE% | Controls when the SIP REGISTER session is established – upon receiving a Push Notification for incoming call or upon accepting it. The default value is “false”. | true, false | O |
| %PN_FOR_CALLS_USE_REGISTRATION_V1% | Controls the call-related PNs type. Set to “true” to use the legacy PNs – all call-related events are delivered as VoIP PNs. Set to “false” to be IOS13 compatible. The default is “false”. | true, false | O |
| %PN_FOR_CALLS_RING_TIMEOUT_SECONDS_MOBILE% | Controls the incoming VoIP call ringing time. The default value is “35”. | [0-180] | O |

NOTE: Starting with Release 3.9.12, if UC-One Connect is running on a device with iOS 13 or higher, and legacy PNs type is configured, the application may be terminated by the OS and APNS may stop sending PNs.

6.1.33 MWI

If you want to enable voice mail updates, you must enable MWI. MWI can be used even when VoIP calls are disabled. It just requires to be enabled in the configuration file, and the “Voice Messaging User” or “Third-Party Voice Mail Support” service to be assigned to the user.

| Tag | Description | Values |
|---------------------|--|--------------------|
| %ENABLE_MWI_MOBILE% | Enables MWI. The default value is “true”. | true, false |
| %MWI_MODE_MOBILE% | Used only if MWI is enabled and there is an ongoing VoIP call with active SIP REGISTER session. Set to “explicit” to send SIP SUBSCRIBE for MWI event package. Using “implicit” does not send a SIP SUBSCRIBE for MWI event package. If empty, MWI is disabled. The default value is empty. | implicit, explicit |

6.1.34 Quality of Service – Diffserv Support for SIP

UC-One Connect supports usage of Diffserv to control the SIP communication quality.

| Tag | Description | Values |
|-------------------------|---|-------------|
| %ENABLE_SIP_QOS_MOBILE% | This is used to enable Quality of Service (QoS) for SIP. The default value is "false". | true, false |
| %SIP_QOS_TYPE_MOBILE% | This is the QoS type. The default value is empty/not set. | dscp, tos |
| %SIP_QOS_VALUE_MOBILE% | This is the QoS value. The default value is empty/not set. | integer |

6.1.35 Call Recording

UC-One Connect Release 3.8.1 added support for call recording. It requires a properly configured server.

| Tag | Description | Values |
|-------------------------|--|-------------|
| %CALL_RECORDING_MOBILE% | This is used to enable Call Recording controls. The default value is "false". | true, false |

6.1.36 Number of Rings

UC-One Connect Release 3.8.6 added support for configurable Number of Rings range. The Number of Rings is shared between a several features like Personal Assistant, Call Forwarding, and others. The range specified by the new tags should apply on all the UI elements of the applications where the Number of Rings is available for the user.

| Tag | Description | Values | M/O |
|--|---|----------|-----|
| %NUMBER_OF_RINGS_RANGE_MIN_VALUE_MOBILE% | Defines the minimum value for number of rings, available for the user to select in the application. The default is "0". | [0;2-20] | O |
| %NUMBER_OF_RINGS_RANGE_MAX_VALUE_MOBILE% | Defines the maximum value for number of rings, available for the user to select in the application. The default is "20". | [0;2-20] | O |

NOTE: The administrator should configure the range to match the one set on the UC-One Portal.

6.1.37 Single Alerting

UC-Connect client supports VoIP and CS calls for calls. To avoid double alerting for incoming business calls, features like [User Selectable Control of VoIP Mode](#) automatically manage the alerting of the Cisco BroadWorks Mobility/Anywhere location and the Push Notification subscription.

Starting with Release 3.9.1, the UC-One Connect client adds a configurable option that can be used to turn off the automatic management of the Cisco BroadWorks Mobility/Anywhere location alerting. When the feature is disabled, the UC-One client will not perform check for double alerting and no Cisco BroadWorks service updates will be made on sign-in, sign-out, and on User Selectable VoIP Mode turning On/Off. Enable single alerting to make sure that incoming business calls will be received only as VoIP or CS.

| Tag | Description | Values | M/O |
|--------------------------|--|-------------|-----|
| %ENABLE_SINGLE_ALERTING% | Enables single alerting for incoming business calls. The default is "true". | true, false | O |

6.1.38 Call Statistics

UC-One Connect Release 3.9.8 adds support for End-of-Call Statistics.

Reporting End-of-Call Statistics in Session Initiation Protocol (SIP) BYE message enables sending call statistics to a remote end, when a call terminates. The call statistics are sent as a new header in the BYE message or in the 200 OK message (response to BYE message). The statistics include Real-time Transport Protocol (RTP) packets sent or received, total bytes sent or received, total number of packets that are lost, delay jitter, round-trip delay, and call duration.

If both End-of-Call Statistics and RTCP XR reporting (specified in section [6.2.1.1 Real-Time Control Protocol Extended Reports](#)) are enabled at the same time, just the End-of-Call report will be sent.

| Tag | Description | Values | M/O |
|---------------------------------|--|-------------|-----|
| %ENABLE_CALL_STATISTICS_MOBILE% | Enables capturing of calling metrics. The default is "false". | true, false | O |

6.1.39 Media Waiting Tone

UC-One Connect Release 3.9.30 adds support for Media Waiting Tone.

Media waiting tone if enabled is played once the user accepts an incoming call. The tone is automatically stopped once the RTP media is established. This feature helps user connected to Bluetooth device to get an audio indication in addition to the existing video indicator that the call is still connecting.

| Tag | Description | Values | M/O |
|------------------------------------|---|-------------|-----|
| %ENABLE_MEDIA_WAITING_TONE_MOBILE% | Enables playing of media waiting tone. The default is "false". | true, false | O |

6.2 VoIP Video Calling

The following tags control the setting specific to video calling.

| Tag | Default Value | M/O |
|-------------------------------------|---|-----|
| %ENABLE_VIDEOCALLS_MOBILE% | true Enable/disable the video calling feature. The default value is "true". | M |
| %VIDEO_QOS_TYPE_MOBILE% | DSCP For more information, see section 6.2.1 QoS – Diffserv Support . | O |
| %VIDEO_QOS_VALUE_MOBILE% | 34 For more information, see section 6.2.1 QoS – Diffserv Support . Quality of Service – Diffserv Support. | O |
| %ENABLE_VIDEO_QOS_MOBILE% | false For more information, see section 6.2.1 QoS – Diffserv Support . Quality of Service – Diffserv Support. | M |
| %RTP_VIDEO_PORT_RANGE_START% | 8600 For more information, see section 6.2.2.1 Preferred Media Ports . | O |
| %RTP_VIDEO_PORT_RANGE_END% | 8698 For more information, see section 6.2.2.1 Preferred Media Ports . | O |
| %CALL_QUALITY_VIDEO_ENABLED_MOBILE% | false For more information, see section 6.2.1.1 Real-Time Control Protocol Extended Reports . Quality of Service – Diffserv Support | O |
| %RTP_MTU% | 1150 For more information, see section 6.2.3 Video MTU . <i>Error! Reference source not found.</i> | O |
| %CALL_QUALITY_VIDEO_ENABLED_MOBILE% | false For more information, see section 6.2.4 Incoming Video Calls . | O |

6.2.1 QoS – Diffserv Support

The following Connect Mobile client tags enable the usage of Diffserv to improve audio and video quality.

| Tag | Description | Values | M/O |
|---------------------------|---|-------------|-----|
| %ENABLE_VIDEO_QOS_MOBILE% | This is used to enable QoS for the video stream. The default is "false". | true, false | O |

| Tag | Description | Values | M/O |
|--------------------------|--|-----------|-----|
| %VIDEO_QOS_TYPE_MOBILE% | This is the QoS type for video. | dscp, tos | O |
| %VIDEO_QOS_VALUE_MOBILE% | This is the QoS value to be used for the video stream. | integer | O |

6.2.1.1 Real-Time Control Protocol Extended Reports

Connect Mobile can send Real-Time Control Protocol (RTCP) Extended Reports (XR) for Video calls when configuring the following tag. It requires RTCP XR for audio calls to be configured.

| Tag | Description | Values | M/O |
|-------------------------------------|---|-------------|-----|
| %CALL_QUALITY_VIDEO_ENABLED_MOBILE% | This is used to enable an RTCP XR report for the video stream. The default is "false". | true, false | O |

6.2.2 Real Time Protocol

6.2.2.1 Preferred Media Ports

Connect Mobile clients provide the ability to configure the preferred audio and video port ranges.

| Tag | Description | Values | M/O |
|------------------------------|---|---------|-----|
| %RTP_VIDEO_PORT_RANGE_START% | The first possible port to be used for video. The default is "8600". | integer | O |
| %RTP_VIDEO_PORT_RANGE_END% | The last possible port to be used for video. The default is "8698". | integer | O |

6.2.3 Video MTU

Connect can be configured to set maximum size of the outgoing video RTP packets.

| Tag | Description | Values | M/O |
|-----------|---|---------|-----|
| %RTP_MTU% | MTU value for the outgoing video traffic. The default is "1150". | integer | O |

6.2.4 Incoming Video Calls

Starting with Release 3.6.1, the iOS version of the UC-One Connect client adds support for how the incoming VoIP video calls are presented to users. Incoming calls are handled by the CallKit, where video calls can be accepted only as audio calls. Since the video calls are to be accepted only as audio calls, based on customer preferences, now it is possible for the alerting visible in the native incoming call screen to show “Incoming Audio Call” for video calls.

| Tag | Description | Values | M/O |
|---|---|-------------|-----|
| %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE% | Controls when the SIP REGISTER session is established – upon receiving a Push Notification for an incoming call or upon accepting it. The default value is “false”. | true, false | O |
| %VIDEO_SHOW_INCOMING_CALL_AS_AUDIO_MOBILE% | Controls the alerting text for incoming VoIP video calls in the native incoming call screen. The default is “false” | true, false | O |

6.3 Instant Messaging and Presence

6.3.1 Server Configuration

| Tag | Description | Values | M/O |
|--------------------------|--|-------------|-----|
| %UMS_HTTP_PORT% | Indicates the port to use for communication with the Messaging Server (UMS) gateway. If the DNS lookup tag is set to “srv”, the port parameter is ignored, and the port value is obtained from the SRV record. If the SRV record does not provide a port, then no port is specified in the HTTP request. If the DNS lookup tag is <i><empty></i> or set to “default”, the port parameter defines the HTTP port to use. If unspecified or “0”, then no port is specified in the HTTP request. The default is “0”. | integer | O |
| %UMS_SRV_SERVICE% | Service name to be used for SRV lookup. The default is “http-client”. | string | M |
| %UMS_HTTP_CONTEXT% | HTTP URL context (that is, the path to reach the message service Application Programming Interface [API]). This path is relative to the domain. The default is “gateway”. | string | M |
| %ENABLE_MESSAGE_GATEWAY% | Enables usage of the Messaging Server (UMS) gateway. The default is “true”. | true, false | M |
| %MESSAGE_GATEWAY_SECURE% | Enables security for the communication with the Messaging Server (UMS) gateway. | true, false | O |

6.3.2 Guest Client

Starting with Release 3.4.1, the UC-One Connect client supports Guest Client.

This allows the owner of the My Room to generate guest client links. In addition, the owner can see, accept, or reject incoming Guest Client requests.

| Tag | Description | Values |
|--------------------------------------|--|-------------|
| %ENABLE_GUEST_CLIENT_SUPPORT_MOBILE% | Set this tag to "true" to enable Guest Client support. | true, false |
| %GUEST_CLIENT_URL% | URL used as a base to generate a link for Guest Clients to join My Room. When empty, disables the Guest Client feature. Example: <guest-client-url>https://kowabunga-guest.domain.com/cgc</guest-client-url>. | string |

6.3.3 My Room

Starting with Release 3.6.1, UC-One Connect client provides two separate parameters to control the availability of chat and call participation in own and buddy's My Room.

For the UC-One SaaS version of the client, Webex Meetings can be used instead of My Room. Users can choose between My Room and Webex Meetings to be used. If Webex Meetings is selected, it requires users to configure the URL to their Personal Room. When the feature is enabled, and the URL is configured, on selecting Join Room, the Webex Meetings client is started, and the user is placed into the meeting. The Call Room option is hidden for Webex Meetings configuration.

| Tag | Description | Values | M/O |
|--------------------------------------|---|-------------|-----|
| %MY_ROOM_SHOW_MOBILE% | Controls the availability of the Join Room and Call Room options. When disabled, both Join Room and Call Room are hidden and the My Rooms accessible from the message history are read only, without the ability to join or call. When enabled, the availability of the Join Room is controlled through the tag defined below. The default is "false". | true, false | O |
| %ENABLE_MY_ROOM_MOBILE% | Controls the availability of the Join Room option. When disabled, the user cannot join own and buddy's My Room with chat and the My Rooms accessible from the message history are read only. Depends on the tag defined above. The default is "false". | true, false | O |
| %ENABLE_WEBEX_MEETINGS_CROSS_LAUNCH% | Enables Webex Meetings cross-launch. Available for UC-One SaaS. The default is "false". | true, false | O |

6.4 Dialer Settings

There are various configuration parameters that can be used to control what dialing options should be available for the user.

Starting with UC-One Connect Release 3.6.1, the keyboard presented to the user in the Dialer is also configurable. It can display full keyboard, so users can type SIP URI or just the digits, where users can dial only numbers/extensions.

| Tag | Description | Values | M/O |
|--|---|---|-----|
| %ENABLE_NATIVE_DIALER% ¹ | Allows Business Line Native Dialing by enabling direct access dialing through the phone's native dialer. The default is "false". | true, false | O |
| %DEFAULT_DIALING_METHOD% | Enables presetting of the Dialing service. The default is "call-through". | call-through, callback, native ² | O |
| %ENABLE_DIALING_CALL_THROUGH% | This is used to control whether the Call Through option is visible and can be selected by the user. The default value is "true". | true, false | M |
| %ENABLE_DIALING_CALL_BACK% | This is used to control whether the Call Back option is visible and can be selected by the user. The default value is "true". | true, false | M |
| %DEFAULT_DIALING_USE_VOIP% | Specifies the default value of the User Selectable VoIP Mode at first sign-in. If User Selectable VoIP mode is disabled, this tag should be set to "false". For more information, see section 6.1.21 User Selectable Control of VoIP Mode . The default is "false" | true, false | M |
| %ENABLE_DIALING_DIALPAD% | Defines the availability of the dialpad in the left side navigation menu. The default is "true". | true, false | O |
| %ENABLE_DIALING_PREVIEW_URL_SCHEMA_NUMBER_TO_DIAL% | Controls if the number to call received through the custom URL scheme to be prepopulated in the dialer, so users can preview, edit, and manually dial it, or to be dialed automatically. When enabled, users can preview the number before dialing it. The default is "false". | true, false | O |
| %DIALER_KEYBOARD_MODE_MOBILE% | Controls the type of keyboard available for the user in the dialer screen when initiating new call. The default is "digits" | full, digits | O |

^{1,2} The native dialer settings should be enabled when deploying UC-One Connect as part of Cisco's (formerly BroadSoft's) mobility business solution – extending business services to the mobile handset through the Cisco BroadWorks Session Continuity Function (SCF) platform.

NOTE: IT is required to always set the three tags %ENABLE_DIALING_CALL_THROUGH%, %ENABLE_DIALING_CALL_BACK%, and %DEFAULT_DIALING_USE_VOIP% with a non-empty value.

6.5 Web-based Call Settings

UC-One Connect has the option to use web-based calling settings to allow the user to control the server-based call settings.

Separate tags can be used to control whether specific services should be visible in the web-based call settings.

| Tag | Description | Values | M/O/CM |
|-------------------------------------|--|-------------|--------|
| %ENABLE_WEB_CALL_SETTINGS% | Controls whether the Call Settings Web View should be available in the client Settings. The default is "false". | true, false | O |
| %WEB_CALL_SETTINGS_URL% | The URL to the web-based calling settings. | url string | CM |
| %WEB_CALL_SETTINGS_CFA_VISIBLE% | Controls whether Call Forwarding Always option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_DND_VISIBLE% | Controls whether Do Not Disturb (DND) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_ACR_VISIBLE% | Controls whether Anonymous Call Rejection (ACR) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_CFB_VISIBLE% | Controls whether Call Forwarding Busy (CFB) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_CFNR_VISIBLE% | Controls whether Call Forwarding Not Reachable (CFNR) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_CFNA_VISIBLE% | Controls whether Call Forwarding No Answer (CFNA) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_SIMRING_VISIBLE% | Controls whether Simultaneous Ring Personal (SIMRING) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_SEQRING_VISIBLE% | Controls whether Sequential Ring (SEQRING) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTINGS_RO_VISIBLE% | Controls whether Remote Office (RO) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |

| Tag | Description | Values | M/O/CM |
|---|---|----------------|--------|
| %WEB_CALL_SETTIN GS_ACB_VISIBLE% | Controls whether Automatic Callback (ACB) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_CW_VISIBLE% | Controls whether Call Waiting (CW) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_CLIDB_VISIBLE% | Controls whether Calling Line ID Delivery Blocking (CLIDB) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_PA_VISIBLE% | Controls whether Personal Assistant (PA) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_BWA_VISIBLE% | Controls whether BroadWorks Anywhere (BWA) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_STAND ARD_SETTINGS_CC_ VISIBLE% | Controls whether Call Center – Standard (CC) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_BASIC_ SETTINGS_CC_VISI BLE% | Controls whether Call Center – Basic (CC) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_PREMI UM_SETTINGS_CC_ VISIBLE% | Controls whether Call Center – Premium (CC) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_BWM_VISIBLE% | Controls whether BroadWorks Mobility (BWM) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |
| %WEB_CALL_SETTIN GS_VM_VISIBLE% | Controls whether Voice Management (VM) option should be visible for the user in the web-based settings. The default is "true". | true, false | O |

M=Mandatory; O=Optional; CM – Conditional Mandatory – mandatory if other options are enabled.

NOTE: The Call Settings WebView URL should always have a trailing "/" configured. For example: `http(s)://<XSP-FQDN>/<CSW-Context-Path>/`

6.6 Landing Page

Release 3.2.2 added configurability of the application landing page. It can be configured to display the messages, recent calls, or a custom web page.

If messages or recent calls are configured, then the entire `<webview-landing-page>` section is ignored.

When a Web URL is configured for a landing page, the messages are accessible from the side menu. The URL supports the following list of parameter replacements.

| Parameter | Description |
|---------------------------------|---------------------------------------|
| <code>%(BWUserID)</code> | Cisco BroadWorks User ID |
| <code>%(FirstName)</code> | Logged-in User's First Name |
| <code>%(LastName)</code> | Logged-in User's Last Name |
| <code>%(Username)</code> | SIP Username from DM Config |
| <code>%(JID)</code> | JID of Logged-in User |
| <code>%(XSP)</code> | Host Address of the Xsp |
| <code>%(BWToken)</code> | Cisco BroadWorks Token |
| <code>%(OptionalBWToken)</code> | Optional Cisco BroadWorks Token |
| <code>%(EmailAddress)</code> | Email Address of Logged-in User |
| <code>%(WorkPhone)</code> | Logged-in User's Work Phone Number |
| <code>%(MobilePhone)</code> | Logged-in User's Mobile Phone Number |
| <code>%(Extension)</code> | Logged-in User's Extension |
| <code>%(Language)</code> | Device's Current Language from Locale |

When a URL contains any of the above parameters, it should be prepended with “%” so that URL will be encoded correctly. For example, `%(JID)` must be defined as `%%(JID)`, `%(BWToken)` must be defined as `%%(BWToken)`.

There are various parameters that can be used to adjust the web page configuration, such as the URL encoding and refresh policy.

Three types of refresh policies are supported – manual, `appForeground`, and `pageFocus`. With the manual policy, the page is refreshed when user pulls to refresh, or uses web mechanisms such as polling to refresh content. With the `appForeground` policy, refresh happens when the app enters from the background to foreground mode. The `pageFocus` policy forces web page refresh when the landing page is brought into focus. This usually happens when closing the side menu, dismissing another view displayed on top, and when bringing the application in foreground mode.

For more information about the deployment parameters controlling the landing page, see the following table.

| Tag | Description | Values | M/O/CM |
|--|--|----------------------------------|--------|
| %DEPLOYMENT_MODEL_LANDING_PAGE% | Specifies which landing page to use within the app. If webview, then the webview-landing page configuration section is used. If webview-landing-page is missing, then the default value is used. The default value is “messages” or “recentcalls”, based on availability. “messages” is used with priority. | webview, messages, recentcalls | O |
| %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_URL% | The URL used to display a landing page. | string | CM |
| %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_USE_TITLE% | Controls whether the title derived from the landing page URL is used or the title text. The default is “false”. | true, false | O |
| %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_ENCODE% | Controls if the URL should be encoded by the application or not. The default is “true”. | true, false | O |
| %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_REFRESH_POLICY% | Controls how the web page should be refreshed. Three options are available. manual – User can pull to refresh or use web mechanisms such as polling to refresh content. | manual, appForeground, pageFocus | O |

M=Mandatory; O=Optional; CM – Conditional Mandatory – mandatory if other options are enabled

6.7 Deployment Model

UC-One Connect client Release 3.2.2 added the ability to specify if major functionalities such as messaging and presence to be enabled for the user. This allows the UC-One Connect client to be configured in a softphone-only mode.

Presence can be enabled separately from messaging. This allows messaging to be disabled, while retaining presence.

| Tag | Description | Values | M/O/CM |
|--------------------------------------|---|-------------|--------|
| %DEPLOYMENT_MODEL_MESSAGING_ENABLED% | Specifies if messaging should be available to the user. The default is “false”. | true, false | O |
| %DEPLOYMENT_MODEL_PRESENCE_ENABLED% | Specifies if presence should be available to the user. The default is “false”. | true, false | O |

6.8 Side Menu and Settings

6.8.1 Configurable Web Button

UC-One Connect client Release 3.1.1 added the ability for web links to be configured and displayed for the user in the side menu. Various parameters can be set to specify what is displayed to the user, the URL, and where to load it. For more information, see the following table.

| Tag | Description | Values | M/O/CM |
|---|---|-------------|--------|
| %ENABLE_WEB_BUTTON% | Controls whether the configured web links are to be available for the user. The default is "true". If not specified in the configuration file and there are valid links configured, they are presented to the user. | true, false | O |
| %WEB_BUTTON_DISPLAY_MODE% | Controls how the links are presented to the user. In "ext" mode, links are listed in a separate Web View page with their titles and description. In "direct" mode, links are displayed directly in the side menu with their titles. The default is "ext". | ext, direct | O |
| %WEB_BUTTON_LIST_LINK_URL_1% | This is either the link to a single web page or a link to an XML document listing many links. For example: url="http://domain.com:80/docs/webbutton_config.xml") The default is "empty". | string | M |
| %WEB_BUTTON_LIST_LINK_ENCODE_1% | Specifies if the parameters of the URL should be encoded or not. The default is "true". | true, false | O |
| %WEB_BUTTON_LIST_LINK_TARGET_1% | Controls how the links are presented to the user. If "main" is configured, the selected link is opened inside the application. If "ext" is configured, links are opened in the default external browser. The default is "ext". | main, ext | O |
| %WEB_BUTTON_LIST_LINK_KEEP_LOADED_1% | Controls whether opened links should be cached when loaded. If enabled, the next time the same link is opened, it will not be reloaded, but the cached data will be displayed to the user. This works only when the target is set to "main" and the links are opened inside the application. The default is "false". | true, false | O |
| %WEB_BUTTON_LIST_LINK_USE_PAGE_TITLE_1% | Controls if the actual web page title should be displayed, replacing the one set in the configuration file. The default is "false". | true, false | O |
| %WEB_BUTTON_LIST_LINK_TITLE_1% | Defines the name of the side navigation menu item or the name displayed in the Web Links view, corresponding to the configured link. The default is "empty". | string | M |

| Tag | Description | Values | M/O/CM |
|--------------------------------------|---|--------|--------|
| %WEB_BUTTON_LIST_LINK_DESCRIPTION_1% | Defines the description of the configured link, displayed in the Web Links view. The default is “empty”. | string | O |
| %WEB_BUTTON_LIST_LINK_REF_1% | Specifies an identifier that can be used by the landing page to access the links defined in the web button list of the side menu, using the internal:// schema. | string | O |

If the display mode is set to “direct” and there are more than three configured links, then the “extended” mode is used and the links are displayed in a separate Web View.

The list of supported web link parameters can be found in section [6.6 Landing Page](#).

To add a link, a separate <link> section should be added in web button list section (<config><services><side-menu><web-button><web-button-list>).

```

<link>
  <url>%WEB_BUTTON_LIST_LINK_URL_1%</url>
  <encode>%WEB_BUTTON_LIST_LINK_ENCODE_1%</encode>
  <target>%WEB_BUTTON_LIST_LINK_TARGET_1%</target>
  <keepLoaded>%WEB_BUTTON_LIST_LINK_KEEP_LOADED_1%</keepLoaded>
  <usePageTitle>%WEB_BUTTON_LIST_LINK_USE_PAGE_TITLE_1%</usePageTitle>
  <title language="en">%WEB_BUTTON_LIST_LINK_TITLE_1%</title>
  <description language="en">%WEB_BUTTON_LIST_LINK_DESCRIPTION_1%</description>
  <ref>%WEB_BUTTON_LIST_LINK_REF_1%</ref>
</link>

```

To make the configuration of a link dynamic, add the new %TAG%s in the template – use the previous example and replace the ending digit with a new number. Then the system administrator can configure these tags through the *System* → *Resources* → *Device Management Tag Sets* option.

For each language, you can specify a title and description. To configure them for a new language, copy the title and description tags, specify the language in the corresponding attribute, and set the text to display. To configure the title and description dynamically, set a %TAG% that can be controlled by the system administrator through the *System* → *Resources* → *Device Management Tag Sets* option.

If language and title are not set for the current device language, English is used as the default.

6.8.2 Call Forwarding

Use the following tag to control Call Forwarding availability in the *Settings*¹.

| Tag | Description | Values | M/O |
|-------------------------|--|----------------|-----|
| %SIDE_MENU_CFA_VISIBLE% | Controls whether the Call Forwarding option is visible for the user. The default is “true”. | true, false | O |

¹ The tag name refers to Side Menu; however, starting with Release 3.1, Call Forwarding and DND options are moved from the Side Menu to App Settings.

6.8.3 Do Not Disturb

Use the following tag to control DND availability in the *Settings*¹.

| Tag | Description | Values | M/O |
|-------------------------|--|-------------|-----|
| %SIDE_MENU_DND_VISIBLE% | Controls whether the DND option to be visible for the user. The default is “true” | true, false | O |

¹ The tag name refers to Side Menu; however, starting with Release 3.1, Call Forwarding and DND options are moved from the Side Menu to App Settings.

6.8.4 Call History

Starting with Release 3.6.1, the UC-One Connect client adds configurable option to control the Call History availability and the type of Call History retrieved and provided to the user.

Release 3.9.6 adds a new config option that controls what action button is available on the top right corner of the screen (when Call History is set as default landing page). Based on this setting, either Directory Search or Delete All is displayed to the user.

| Tag | Description | Values | M/O |
|--------------------------------------|--|-------------|-----|
| %ENABLE_CALL_HISTORY% | Controls whether the Call History is visible for the user. The default is “true”. | true, false | O |
| %ENABLE_ENHANCED_CALL_HISTORY% | Specifies if basic or enhanced Call History to be presented to the user. The default value is “false”. | true, false | O |
| %CALL_HISTORY_SHOW_DIRECTORY_SEARCH% | Specifies if the Directory Search button should be displayed (when Call History is configured as landing page). The default is “false”. | true, false | O |

6.9 Custom URL Scheme for Calling and Messaging

UC-One Connect supports the ability to place calls or send chat messages when launched with a custom URL scheme from another application (third party app). This is also known as deep linking.

The following three actions are supported:

- Open a chat to one or more contacts.
- Call a number or contact.
- Start a new broadcast message to many contacts.

The URL scheme is defined as `scheme://action/path[/resource][?query=parm]`.

Note that it should contain only URL-escaped characters (valid URL). These are some of the required encodings. Where:

- Scheme is the brandable part of the URL (currently defined as “iris”).
- Action is host part of the URL and represents the requested call action with the following possible values:
 - message

– call

The message action initiates a chat or broadcast message. The following URL parts are supported:

- Message – The action part of the URL must be “message”.
- Chat – When the path part of the URL is “chat”, UC-One Connect sends a chat message to the recipient identified by the *to* query parameter. If multiple recipients are specified, UC-One Connect sends a group message instead of a one-on-one chat.
- Broadcast – When the path part of the URL is “broadcast”, UC-One Connect sends a broadcast message to the recipients identified by the *to* query parameters.
- To – The *to* query parameter of the URL identifies the IM&P ID or email address of a chat recipient. This parameter is repeated for each recipient, and this approach is consistent with industry standards.

Examples:

```
iris://message/chat?to[]=smith@example.com
iris://message/chat?to[]=patel@example.com&to[]=ahmed@example.com
iris://message/broadcast?to[]=smith@example.com&to[]=ahmed@example.com&to[]=martin@example.com&to[]=wang@example.com&to[]=garcia@example.com
```

The call action initiates a call to either a number, person, or room. The following URL parts are supported:

- Call – The action part of the URL must be “call”.
- Number/resource – The number path of the URL identifies a call to a phone number. The resource is the number of the party to call. The resource can include a PIN, conference, or room codes. The number can include any of the following characters.

| Characters | % Encoding | Usage |
|-------------|------------|--|
| 0 through 9 | none | Phone number digits. |
| - | none | Optional hyphen digit separators. |
| + | %2B | The plus sign can be used as the first character to denote a country code. |
| , | %2C | The comma inserts a pause (typically two seconds for each comma). |
| ; | %3B | The semicolon inserts a wait. |
| # | %23 | The octothorpe character (#) is used as a continuation character in many PIN and room codes. |
| * | %2A | The asterisk character can also be used as a continuation character in PIN codes. |

- Contact/resource – The contact path of the URL identifies a call to a contact. The resource is the IM&P ID or email address of the party to call. It is assumed UC-One Connect can find a valid phone number for the recipient.
- Room/resource – The room path of the URL identifies a call to a room. The resource is the IM&P ID or email address of the contact whose room to call. It is assumed UC-One Connect can find a valid My Room number for the recipient.

Examples:

```
iris://call/number/5555555555
iris://call/number/5555555555%2C12345%23
iris://call/contact/patel@example.com
iris://call/room/wang@example.com
```

6.10 Cisco BroadWorks Xtended Service Interface (Xsi)

6.10.1 Xtended Services Interface Root and Paths

The UC-One Connect client uses the following tags to control the Xsi root, actions, and events path, if they are different from the one used for sign-in.

| Tag | Description | Values |
|---------------------------|---|--------|
| %XSI_ROOT% | The XSI root for all Xsi operations. Example: https://domain.com/ | string |
| %XSI_ACTIONS_PATH_MOBILE% | Specifies the Xsi-actions path. It should start and end with "/" and contain only the actions context (for example, /com.broadsoft.xsi-actions/). Example: /com.domain.xsi-actions/ | string |
| %XSI_EVENTS_PATH_MOBILE% | Specifies the Xsi Events path. It should start and end with "/" and contain only the events context (for example, /com.broadsoft.xsi-events/). Example: /com.domain.xsi-events/ | string |

6.10.2 Xtended Services Interface Mid-Call Controls and Events Channel

The UC-One Connect client can be configured to provide Mid-Call Control services for Click To Dial calls (that is, calls from a phone). When enabled, users can perform mid-call actions such as:

- Hold or Resume
- Transfer
- Conference
- Add participants
- End call

Note that the call control is available only for the calls terminated on the same device.

To provide Xtended Services Interface mid-call controls, the Xsi Events channel must be enabled. The Xsi Events channel is opened only when an Xtended Services Interface call is started as shown in the following example.

```
<config>
<services><calls>
<reject-with-xsi mode="decline_false" declineReason="" />
<extended-call-control enabled="true" call-control-window="true" />
<transfer-call enabled="true" xsi-enabled="true" type="full" />
<conference enabled="true" xsi-enabled="true" />
...
<protocols><xsi>
<xsi-call-control>
```



```

    <deployment type="ott"/>
</xsi-call-control>
<paths>
  <root>https://btbc.ihs.broadsoft.com/</root>
  <actions>/com.broadsoft.xsi-actions/</actions>
  <events>/com.broadsoft.xsi-events/</events>
</paths>
<event-channel>
  <heartbeatInterval>10000</heartbeatInterval>
</event-channel>

```

New properties controlling Transfer and Conference actions availability for Xsi calls are introduced so they can be configured separately from SIP calls.

The heartbeat is used to keep the Xsi Events channel open and the heartbeat interval can be specified using the parameter in the previous example.

The Xsi call control depends on the deployment type. For more information about how to set the deployment type, see section [6.10.3 Deployment Type](#).

In Mobile Network Operator (MNO) mode with Cisco BroadWorks Mobility, all circuit-switched calls on the mobile device are anchored on Cisco BroadWorks and are considered business calls. In this mode, while in Xsi Mid-Call Control, a new call, attended transfers, and conference uses a native call or Call Back for the second leg and not Call Through.

In Over The Top (OTT) deployments, only the calls received or initiated through the Cisco BroadWorks Anywhere or Remote Office services are anchored on Cisco BroadWorks and considered as business calls. In this mode, if a business call is detected and Xsi Mid-Call Control is provided to the user, a new call, attended transfer, and conference use either Call Back or Call Through for the second leg, and not a native call.

A separate configuration option is added to specify the call decline behavior when an incoming call is received as a Push Notification. The client can be configured to ignore the call or to respond to the server through Xsi with decline set to "true" or "false", in which case, the assigned Cisco BroadWorks call treatment services will be applied. If "decline_false" is configured, the call continues ringing until the originator abandons or the no-answer timer expires, and the associated call treatment services start. If "decline_true" is configured, the decline reason specifies the call processing. If the decline reason is set to "busy", the server immediately forces the busy treatment service. If "temp_unavailable" is configured, the temporary unavailable treatment service is applied.

The following tags are used to enable Mid-Call Control services.

| Tag | Description | Values | M/O/CM |
|---------------------------------------|---|-------------|--------|
| %ENABLE_XSI_CALLS_MOBILE% | Should be enabled to have the Xsi Mid-Call Control in-app and widget enabled. The default value is "false". | true, false | O |
| %ENABLE_XSI_MIDC_ALL_CONTROLS_MOBILE% | Enables in-app Xsi Mid-Call Control. The default value is "false". | true, false | O |
| %CHANNEL_HEARTBEAT_MOBILE% | Xsi Event channel heartbeat interval in milliseconds. The default value is "10000". | integer | O |
| %ENABLE_XSI_TRANSFER_CALLS_MOBILE% | Enables the transfer option for Xsi calls. The default value is "false". | true, false | O |

| Tag | Description | Values | M/O/CM |
|---|--|---|--------|
| %ENABLE_XSI_CONFERENCE_CALLS_MOBILE% | Enables the conference option for Xsi calls. The default value is "false". | true, false | O |
| %REJECT_WITH_XSI_MODE_MOBILE% | Specifies the call decline behavior. The default value is "decline_false". | ignore, decline_true, decline_false | O |
| %REJECT_WITH_XSI_DECLINE_REASON_MOBILE% | Specifies the call decline reason if the reject mode is set to "decline_true". The default value is "busy". | busy, temp_unavailable | O |

6.10.3 Deployment Type

A separate configuration option is added to specify the deployment type of the UC-One Connect client. Supported deployment types are Over The Top (OTT) and Mobile Network Operator (MNO) with Cisco BroadWorks Mobility.

The deployment type influences UC-One Connect client behavior for features related to native calling and how the device is integrated in the mobile network. For more information, see section [6.10.2 Xtended Services Interface Mid-Call Controls and Events Channel](#).

| Tag | Description | Values | M/O/CM |
|--------------------------|---|-----------------|--------|
| %DEPLOYMENT_TYPE_MOBILE% | Specifies the deployment type. The default is "ott". | ott, mno_scf | O |

NOTE: For Tablet, it is recommended to use the OTT deployment type. MNO deployments imply that the user will be able to initiate circuit-switched calls, and some tablets (like iPad) do not support native calling.

6.10.4 Call From

UC-One Connect client Release 3.0.1 allowed users to select their Calling Line Identity (CLID) presented to the remote party upon initiating an outgoing call. The persona management is available only for Mobile Network Operator (MNO) deployments with Cisco BroadWorks Mobility.

Starting with Release 3.4.1, the availability of the persona management and the Call From menu option is configurable. If persona management is enabled, the Call From option is available in the *Settings*.

| Tag | Description | Values |
|-----------------------------|--|-------------|
| %ENABLE_PERSONA_MANAGEMENT% | Enables persona management. The default value is "false". | true, false |

6.10.5 Xtended Services Interface Mid-Call Controls Widget

Starting with Release 3.5.1, the UC-One Connect client provides a widget that can be used by the user to control native circuit-switched calls anchored on Cisco BroadWorks. If enabled, the widget can be added by the user through the OS. The widget has three actions buttons – Transfer Call, Move Call, and Settings.

For Android only, from the Settings, the user can enable Call Control Notification, available in the Notification drawer and on the lock screen. The Call Control Notification provides similar call control functions as the widget.

| Tag | Description | Values |
|--------------------------------------|---|-------------|
| %ENABLE_XSI_MIDCALL_CONTROLS_WIDGET% | Together with %ENABLE_XSI_CALLS_MOBILE%, controls if the call control widget should be visible for the user. For Android, also controls the availability of the Call Control Notification option in Call Settings. The default value is "true". | true, false |

6.11 Enhanced Presence Options with Personal Assistant Support

Starting with Release 3.3.1, UC-One Connect can be configured to allow users to set additional presence options from a predetermined list. Additionally, these presence states can be integrated with the Cisco BroadWorks Personal Assistant feature when the user is assigned the Personal Assistant service and the Messaging Server (UMS) has been upgraded to Release 22.0 and the appropriate patches have been applied.

Personal Assistant provides to the user an option to inform the callers of the reason the called party is not available, optionally providing information on when the called party will return and whether there is an attendant to attend the call.

Starting with Release 3.6.1, the UC-One Connect client adds support for configurable Ring Reminder (also referred to as Silent Alert) and Alert Me First options of the Personal Assistant.

For more information about all the config parameters of the Personal Assistant, see the following table.

| Tag | Description | Values | M/O/CM |
|--|--|-------------|--------|
| %ENABLE_PERSONAL_ASSISTANT_PRESSENCE% | Controls whether the Enhanced Presence feature is visible for the user. Optionally, if the Cisco BroadWorks Personal Assistant service is assigned to the user – the Enhanced Presence Settings will also include duration, call transfer, and silent alert. The default is "false". | true, false | O |
| %ENABLE_PERSONAL_ASSISTANT_PRESSENCE_SILENT_ALERT% | Controls whether the Silent Alert option is visible for the user. The default is "true". | true, false | O |
| %ENABLE_PERSONAL_ASSISTANT_PRESSENCE_ALERT_ME_FIRST% | Controls whether the Alert Me First and Number of Rings options are visible for the user. The default is "true". | true, false | O |

6.12 Password Update

UC-One Connect supports password update and notification for the user when the password is going to expire. Notification is displayed only if a positive value is configured for how many days before the expiration user should be warned and the expiration day is in less than the configured time.

The following tags control the password update feature.

| Tag | Description | Values | M/O/CM |
|------------------------------------|---|-------------|--------|
| %ENABLE_PASSWORD_UPDATE% | Controls whether the Update Password feature is visible for the user. The default is "true". | true, false | O |
| %PASSWORD_UPDATE_WARN_BEFORE_DAYS% | Controls the number of days before the expiration, when the user should start seeing a warning. The default value is "14". | integer | O |

6.13 Auto-configure Anywhere Location

At sign-in, if Cisco BroadWorks Anywhere is enabled for the user, the UC-One Connect client checks the Anywhere locations with the phone number specified by the user is configured. If there is no location, the application suggests to the user to auto-configure it.

The auto-configuration of the Anywhere location is controlled through the tag specified in the following table.

| Tag | Description | Values | M/O/CM |
|----------------------------------|---|-------------|--------|
| %ENABLE_AUTO_CONFIGURE_ANYWHERE% | Controls whether the auto-configuration of the Cisco BroadWorks Anywhere location should be available for the user. The default value is "true". | true, false | O |

6.14 My Telephone Number

Starting with Release 3.5.1, the UC-One Connect client adds the option own telephone number to be optional for the sign-in. Until now, the user had to provide their own number to sign in.

| Tag | Description | Values | M/O/CM |
|-------------------------------------|--|-------------|--------|
| %ENABLE_MY_TELEPHONE_NUMBER_MOBILE% | Controls the availability of the phone number screen at sign-in and editing it from the Call Settings. The default value is "true". | true, false | O |

6.15 Contacts

6.15.1 Sort Order, Display Name Order, and Short Names

Starting with Release 3.6.1, the UC-One Connect client adds support for sort order, display order, and short names. These settings are part of the application itself, and if allowed, the user can configure them. As these settings are configurable through the application, they can be hidden for the user allowing default values to be used.

The configuration tags used only by the Android version of UC-One Connect, can be found in the following table.

| Tag | Description | Values |
|---------------------------------------|--|--|
| %CONTACTS_SORT_ORDER_VISIBLE% | Allows the user to overwrite the default sort order. The default value is "false". | true, false |
| %CONTACTS_SORT_ORDER_DEFAULT% | Defines the default sort order. The default value is "first". | first, last |
| %CONTACTS_DISPLAY_NAME_ORDER_VISIBLE% | Allows the user to overwrite the default display name format. The default value is "false". | true, false |
| %CONTACTS_DISPLAY_NAME_ORDER_DEFAULT% | Defines the default display name order. The default value is "first". | first, last |
| %CONTACTS_SHORT_NAMES_VISIBLE% | Allows the user to overwrite the default short names format. The default value is "false". | true, false |
| %CONTACTS_SHORT_NAMES_DEFAULT% | Defines the default short names format. The default value is "fullname". | fullname, firstinit, lastinit, firstonly, lastonly |

NOTE: On iOS, there are system settings for sort and display order, and short names, which are not respected by the iOS UC-One Connect client.

6.15.2 Directory Cache

UC-One Connect client provides the ability to search for contacts in the Enterprise Directory. Starting with Release 3.6.1, the UC-One Connect client adds support to downloading and cache a number of directory contacts, so that the Directory view is not empty.

This feature is intended for small and medium enterprises, and it is recommended to configure the directory cache to be 50 or less.

| Tag | Description | Values | M/O/C M |
|--|--|---------|------------|
| %CONTACTS_CACHE_CONTROL_CACHE_LIMIT% | Specifies the number of contacts to be retrieved from the Directory search sources. The default value is "50". | Integer | O |
| %CONTACTS_CACHE_CONTROL_CACHE_INTERVAL% | Controls the interval of time at which the cached Directory contacts result should be updated. The default value is "86400" (24 hours). | Integer | O |
| %CONTACTS_CACHE_CONTROL_CACHE_refreshXSIContactsLimit% | This value defines number of local contact cache size. The default value is "100" (100 Contacts). | Integer | O |

When the application is running in normal mode, if the total number of contacts in the Enterprise Directory is lesser than %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%, all the contacts will be cached. If the total number of contacts is more than the %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%, none of the contacts will be cached, and only the contacts with whom the user has interacted with (Favorites, Groups, chats), will be displayed to the user.

When the application is running in Softphone mode, the behavior is different. If the total number of contacts in the Enterprise Directory is more than the %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%, the application will cache number of contacts equal to the value specified by %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%.

6.15.3 Contact Search

UC-One Connect client has Cisco BroadWorks Telephony Directory integration, providing to the user the ability to search for contacts in the following contact sources:

- Enterprise
- Personal
- Enterprise Common
- Group Common

Starting with Release 3.8.6, the contact sources available for the user can be configured separately, using the tags defined in the following table.

| Tag | Description | Values |
|---|--|----------------|
| %ENABLE_CONTACTS_ENTERPRISE_SEARCH_MOBILE% | Controls if Enterprise should be included in the Directory search sources. The default value is "true". | true, false |
| %ENABLE_CONTACTS_ENTERPRISE_COMMON_SEARCH_MOBILE% | Controls if Enterprise Common should be included in the Directory search sources. The default value is "false". | true, false |
| %ENABLE_CONTACTS_PERSONAL_SEARCH_MOBILE% | Controls if Personal Phone should be included in the Directory search sources. The default value is "false". | true, false |
| %ENABLE_CONTACTS_GROUP_COMMON_SEARCH_MOBILE% | Controls if Group Common should be included in the Directory search sources. The default value is "false". | true, false |

NOTE: If all the search sources are disabled, the user can initiate a directory search, but no results will appear.

6.16 Login Informational Dialog

Starting with Release 3.8.6, UC-One Connect client can be configured to display dialog with some information at sign-in. The user is forced to acknowledge a given message (for example, information that the client shall not be used for E911 calls), before using the application. To configure the login dialing, use the tags defined in the following table.

| Tag | Description | Values |
|---|--|-------------|
| %SHOW_LOGIN_INFORMATIONAL_DIALOG_MOBILE% | This is used to display an informational message during the user login procedure (before the client is usable). The default value is "false". | true, false |
| %SHOW_INFORMATIONAL_DIALOG_ON_EVERY_LOGIN_MOBILE% | This is used to control if the dialog should be displayed only the first time or on every user login. The default value is "false". | true, false |

NOTE: Message content is configurable through branding.

6.17 Cisco BroadWorks Anywhere

The UC-One Connect client can provide the ability to the user to configure and use the Cisco BroadWorks Anywhere service. To enable this feature for the user, the Cisco BroadWorks Anywhere service should be assigned.

Once enabled, user can configure the Cisco BroadWorks Anywhere Location settings in the application. Release 3.9.4 adds a configuration parameter that controls Basic or Advanced set of Cisco BroadWorks Anywhere location settings to be available for the user. In Basic mode, the user controls the Telephone number, Description, and Active state of the location. The Advanced mode adds to the Basic, Call Control, Don't Divert Calls, and Answer Confirmation settings. Also, four new configuration parameters are added to control the default values of the Cisco BroadWorks Anywhere location parameters and if the client should enable the generic Alert All Locations setting for the entire service.

| Tag | Description | Values |
|--|---|-------------|
| %ENABLE_BROADWORKS_ANYWHERE_ADVANCED_OPTIONS_MOBILE% | Controls whether the Basic or Advanced set of Cisco BroadWorks Anywhere options are available for the user. By default, the Basic set is displayed to the user. The default value is "true". | true, false |
| %ANYWHERE_ADVANCED_OPTIONS_CALL_CONTROL_DEFAULT_MOBILE% | Controls the default value of the Call Control setting for a Cisco BroadWorks Anywhere (BWA) location. The default value is "true". | true, false |
| %ANYWHERE_ADVANCED_OPTIONS_DIVERSION_INHIBITOR_DEFAULT_MOBILE% | Controls the default value of the <i>Don't Divert Calls</i> setting of a BWA location. The default value is "false". | true, false |
| %ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ANSWER_CONFIRMATION_MOBILE% | Controls the default value of the Answer Confirmation setting of a BWA location. The default value is "false". | true, false |

| Tag | Description | Values |
|--|---|-------------|
| %ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ALERT_ALL_LOCATIONS_MOBILE% | Controls if the Alert All Locations setting should be enabled by the application or if it should remain as configured. The default value is "false". | true, false |

6.18 HTTP Proxy

The UC-One Connect Mobile client does not support proxy configuration on the application level. However, the HTTP proxy could be configured on the system level.

For iOS devices, proxy configuration can be done by navigating to native *Settings* → *Wi-Fi* → *Select Network* → "Configure Proxy". The user can either select Auto configure or manually configure the proxy details.

For Android devices, based on the make, model, and OS version, the configuration option can be found on one of the following locations:

- *Settings* → *Connections* → *Wi-Fi* → *Long Press on Wi-Fi network name* → *Advanced*.
- *Settings* → *Wi-Fi* → *Long Press on Wi-Fi network name* → *Modify network* → *Advanced options*.
- *Settings* → *Network & internet* → *Wi-Fi* → *Tap on Connected network* → *Edit (Icon)* → *Advanced options*
- *Settings* → *Connections* → *Wi-Fi* → *Long Press on network name* → *Manage Network Settings* → *Show Advanced Options*

6.19 Version Control

The UC-One Connect Mobile client has a mechanism that informs users for optional or mandatory application upgrade. This can be used to install a new version of the current application or install a different one. This new functionality allows service providers to enable it when they are ready with the migration.

The optional interstitial web page will be developed and hosted by the service provider and will include more information and download links.

| Tag | Description | Values | M/O/CM |
|-------------------------------|--|-------------|--------|
| %ENABLE_VERSION_CONTROL% | This is used to enable/disable version control. The default value is "false". | true, false | O |
| %ANDROID_MUST_VERSION% | Mandatory version string for Android client. | string | O |
| %ANDROID_RECOMMENDED_VERSION% | Recommended version string for Android client. | string | O |
| %ANDROID_DOWNLOAD_LINK% | Download link for the latest Android client. | string | O |
| %IOS_MUST_VERSION% | Mandatory version string for iOS client. | string | O |
| %IOS_RECOMMENDED_VERSION% | Recommended version string for iOS client. | string | O |

| Tag | Description | Values | M/O/CM |
|---------------------|--|--------|--------|
| %IOS_DOWNLOAD_LINK% | Download link for the latest iOS client. | string | O |

NOTE: The download link specified must be a fully formatted URL along with the protocol such as http://, https://, and so on.

6.20 Token Expiry Warning

The UC-One Connect Mobile client has a mechanism to display a warning to the user when the token is about to expire. The following tag can be used to configure the time period (number of days) before the warning message should be displayed.

| Tag | Description | Values | M/O/CM |
|---------------------------------|---|---------|--------|
| %TOKEN_EXPIRY_WARN_BEFORE_DAYS% | This is used to display a warning message before the token expiry. The default value is 3. | Integer | O |

7 Features and Configuration Dependencies

This section lists Cisco BroadWorks services and client features that may have dependency on each other, as well as related dependency on configuration settings.

7.1 MNO and OTT Deployments

Cisco BroadWorks supports two deployment models (MNO and OTT) with regards to how the user mobile phone works with the mobile network.

For MNO deployments, the phone is company provided and integrated with the mobile network. The phone number is provisioned in the Cisco BroadWorks Mobility service.

For OTT deployments, the phone is BYOD and the phone number is provisioned in the BroadWorks Anywhere service.

UC-One Mobile applications are intended to run on single mobile phone per user and support either MNO- or OTT-type provisioned users. UC-One Mobile is not intended to support users that can switch between MNO and OTT on the same mobile device or to use two separate phones (or switch two SIM cards).

In other words, a user configured for an MNO deployment type and running the client on an integrated device, should not use the UC-One Mobile application running on a non-integrated device.

Users running the application on a non-integrated device should be configured with deployment type OTT and not run the application on an integrated device.

UC-One Mobile has partial support for automatic deployment mode detection, allowing user configured with MNO deployment to occasionally use the application on non-integrated device (OTT). For more information, see the following section.

7.2 Automatic Deployment Mode Detection

Starting with Release 3.9.12, the UC-One Mobile client adds support for dynamic detection of the deployment mode. Only if the deployment mode is configured as MNO, at sign in, the application will search for the BroadWorks Mobility (BWM) location, configured with the number set by the user as “This phone number” / “This Phone”.

- If there is a BWM location found, the application will work in MNO mode.
- If there is no BWM location found, the application will work in OTT mode.
 - Persona management – disabled even if enabled through configuration (%ENABLE_PERSONA_MANAGEMENT%). This will remove the “Call From” in *Settings* and Persona selection from the *Call* screen.
 - Block My Caller ID in Call Settings Web View is not affected. If enabled (%WEB_CALL_SETTINGS_CLIDB_VISIBLE%), it will remain available for the user. CLID blocking will work only for calls anchored on BroadWorks. In OTT mode, it works only for the Call Back and Call Through calls.
- Upon updating the phone number in Settings, to update the deployment mode the application is running, users must sign out and sign back in.

8 UC-One SaaS

8.1 Client Configuration

The UC-One Software as a Service (SaaS) version of the client uses the *config-uc1s.xml* file for configuration of its calling functionality. There is a separate configuration file that resides on the BroadCloud and controls all the remaining functionality apart from calling and it is not covered in this document.

8.2 Deployment of config-uc1s.xml

Follow the deployment steps for the UC-One Connect *config.xml* for the UC-One SaaS *config-uc1s.xml*.

The UC-One SaaS *config-uc1s.xml* configuration file is independent from the UC-One Connect *config.xml* file and both can be deployed at the same time.

8.3 Configuration File (config-uc1s.xml)

The configuration file used for UC-One SaaS version of client is a subset of the UC-One Connect *config.xml* configuration file.

The following is a list with all the tags that are used in the Cisco BroadWorks part of the UC-One SaaS configuration file. For more information on each tag, see the corresponding subsections under section [6 Custom Tags](#):

- %CALL_RECORDING_MOBILE%
- %ENABLE_REJECT_WITH_486_MOBILE%
- %REJECT_WITH_XSI_MODE_MOBILE%
- %REJECT_WITH_XSI_DECLINE_REASON_MOBILE%
- %ENABLE_XSI_CALLS_MOBILE%
- %ENABLE_XSI_MIDCALL_CONTROLS_MOBILE%
- %ENABLE_XSI_MIDCALL_CONTROLS_WIDGET%
- %ENABLE_TRANSFER_CALLS_MOBILE%
- %ENABLE_XSI_TRANSFER_CALLS_MOBILE%
- %TRANSFER_CALL_TYPE_MOBILE%
- %ENABLE_TRANSFER_TO_CIRCUIT_SWITCH_MOBILE%
- %ENABLE_AUDIOCALLS_MOBILE%
- %ENABLE_AUDIO_QOS_MOBILE%
- %AUDIO_QOS_TYPE_MOBILE%
- %AUDIO_QOS_VALUE_MOBILE%
- %ENABLE_VIDEOCALLS_MOBILE%
- %ENABLE_VIDEO_QOS_MOBILE%
- %VIDEO_QOS_TYPE_MOBILE%
- %VIDEO_QOS_VALUE_MOBILE%
- %VIDEO_SHOW_INCOMING_CALL_AS_AUDIO_MOBILE%

- %ENABLE_CONFERENCE_CALLS_MOBILE%
- %ENABLE_XSI_CONFERENCE_CALLS_MOBILE%
- %CONFERENCE_CALLS_AUTO_JOIN_PAUSE%
- %ENABLE_ADAPTIVE_QUALITY_MOBILE%
- %ENABLE_CALL_PULL_MOBILE%
- %ENABLE_VOIP_MODE_MOBILE%
- %CELLULAR_DATA_VOIP_CALLS_STATE_MOBILE%
- %ENABLE_CALLS_AUTO_RECOVERY_MOBILE%
- %CALLS_AUTO_RECOVERY_DATA_TO_WIFI_MODE_MOBILE%
- %ENABLE_ONE_TIME_VOIP%
- %ENABLE_SINGLE_ALERTING%
- %CONFERENCE_BRIDGE_MEDIA_TYPE_MOBILE%
- %ENABLE_UPDATE_EMERGENCY_LOCATION_MOBILE%
- %UPDATE_EMERGENCY_LOCATION_VERIFICATION_TIMEOUT_MOBILE%
- %UPDATE_EMERGENCY_LOCATION_URL_MOBILE%
- %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_LOGIN_MOBILE%
- %UPDATE_EMERGENCY_LOCATION_REQUIRED_MOBILE%
- %UPDATE_EMERGENCY_LOCATION_RECURRING_MOBILE%
- %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_WILL_MOBILE%
- %EMERGENCY_CALL_DIAL_SEQUENCE_MOBILE%
- %SHOW_EMERGENCY_WARNING_DIALOG_MOBILE%
- %EMERGENCY_DIALING_BEHAVIOR_MOBILE%
- %CHECK_DEVICE_EMERGENCY_NUMBERS_MOBILE%
- %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE%
- %DEPLOYMENT_TYPE_MOBILE%
- %ENABLE_PERSONA_MANAGEMENT%
- %ENABLE_DIALING_DIALPAD%
- %ENABLE_DIALING_PREVIEW_URL_SCHEMA_NUMBER_TO_DIAL%
- %SIDE_MENU_CFA_VISIBLE%
- %SIDE_MENU_DND_VISIBLE%
- %ENABLE_WEB_CALL_SETTINGS%
- %WEB_CALL_SETTINGS_URL%
- %WEB_CALL_SETTINGS_CFA_VISIBLE%
- %WEB_CALL_SETTINGS_DND_VISIBLE%
- %ENABLE_PERSONAL_ASSISTANT_PRESENCE%
- %ENABLE_PERSONAL_ASSISTANT_PRESENCE_SILENT_ALERT%

- %ENABLE_PERSONAL_ASSISTANT_PRESENCE_ALERT_ME_FIRST%
- %ENABLE_MY_TELEPHONE_NUMBER_MOBILE%
- %CONTACTS_SORT_ORDER_VISIBLE%
- %CONTACTS_SORT_ORDER_DEFAULT%
- %CONTACTS_DISPLAY_NAME_ORDER_VISIBLE%
- %CONTACTS_DISPLAY_NAME_ORDER_DEFAULT%
- %CONTACTS_SHORT_NAMES_VISIBLE%
- %CONTACTS_SHORT_NAMES_DEFAULT%
- %CONTACTS_CACHE_CONTROL_CACHE_LIMIT%
- %CONTACTS_CACHE_CONTROL_CACHE_INTERVAL%
- %CONTACTS_CACHE_CONTROL_CACHE_refreshXSIContactsLimit%
- %ENABLE_CALL_HISTORY%
- %ENABLE_ENHANCED_CALL_HISTORY%
- %ENABLE_MWI_MOBILE%
- %MWI_MODE_MOBILE%
- %ENABLE_SIP_INFO_MOBILE%
- %ENABLE_VIDEO_FIR_SIP_INFO_MOBILE%
- %SBC_PORT_MOBILE%
- %SBC_ADDRESS%
- %USE_PROXY_DISCOVERY_MOBILE%
- %DOMAIN_OVERRIDE%
- %100_REL_ENABLED_MOBILE%
- %TRANSPORTS_TYPE_MOBILE%
- %TLS_VERSION%
- %ENABLE_SIGN_IN_TIMER_MOBILE%
- %SIGN_IN_TIMER_PERIOD_MOBILE%
- %USE_EXPIRES_HACK_MOBILE%
- %USE_MEDIASEC_MOBILE%
- %ENABLE_SIP_UPDATE_SUPPORT_MOBILE%
- %SRTP_MODE_MOBILE%
- %SRTP_ENABLED_MOBILE%
- %ENABLE_RE-KEYING_MOBILE%
- %CALL_QUALITY_AUDIO_ENABLED_MOBILE%
- %CALL_QUALITY_VIDEO_ENABLED_MOBILE%
- %CALL_QUALITY_SERVICE_URI%
- %CALL_QUALITY_LOCAL_GROUP%

- %RTP_AUDIO_PORT_RANGE_START%
- %RTP_AUDIO_PORT_RANGE_END%
- %RTP_VIDEO_PORT_RANGE_START%
- %RTP_VIDEO_PORT_RANGE_END%
- %RTP_MTU%
- %CONFERENCE_CALLS_AUTO_JOIN_PAUSE%
- %ENABLE_USE_RPORT_MOBILE%
- %RPORT_USE_LOCAL_PORT_MOBILE%
- %ENABLE_PROXY_DISCOVERY_RFC5922_CERT_VALIDATION_MOBILE%
- %ENABLE_SIP_QOS_MOBILE%
- %SIP_QOS_TYPE_MOBILE%
- %SIP_QOS_VALUE_MOBILE%
- %ENABLE_TCP_RESET_FAILOVER_MOBILE%
- %VIDEO_SHOW_INCOMING_CALL_AS_AUDIO_MOBILE%
- %ENABLE_RTP_MUX_MOBILE%
- %ENABLE_AUTO_CONFIGURE_ANYWHERE%
- %DIALER_KEYBOARD_MODE_MOBILE%

Some of the Cisco BroadWorks System Tags are also used in the configuration file. For more information on each of the following tags, see section [5.5 Cisco BroadWorks System Tags](#).

- %BWNETWORK-CONFERENCE-SIPURI-1%
- %BWLINEPORT-1%
- %BWAUTHPASSWORD-1%
- %BWE164-1%
- %BWAUTHUSER-1%
- %BWHOST-1%
- %SOURCE_PORT%

In general, in the UC-One SaaS calling configuration file, as in the Collaborate version of the client configuration file, some settings in the configuration file are hardcoded and cannot be provisioned through the Device Management TAG settings.

NOTE: Some features in the UC-One SaaS client can be controlled through the BAM Portal.

9 Bandwidth Requirements

The following table describes the estimated bandwidth consumption for audio calls (RTP).

| Audio Codec | Bit Rate (kbps) | Sampling Frequency (kHz) | Bit Rate Including IP/UDP/RTP Overhead * |
|-------------|-----------------|--------------------------|--|
| G.711 | 64 | 8 | 80 kbps |
| G.722 | 64 | 16 | 80 kbps |
| G.729 | 8 | 8 | 24 kbps |
| Opus | 6 to 510 | 48 | 22 kbps to 526 kbps |

* Based on a packetization interval of 20 milliseconds.

For the video codecs, the maximum bit rate can be set through the configuration file.

10 Appendix A: TLS Ciphers

All the Xsi and HTTP connections used by the UC-One Connect client are ATS-compliant and require only TLS1.2 with ATS compliance ciphers.

SIP can use any of the ciphers in this section; however, using D and F grades is not recommended. Using SSLv3, SSLv2, and TLS1.0 is also not recommended.

10.1 Android Client

The Android client supports the following protocol versions:

- SSLv2
- SSLv3
- TLS1.0
- TLS1.1
- TLS1.2

The Android client supports the following ciphers:

- ECDHE-RSA-AES256-GCM-SHA384
- ECDHE-ECDSA-AES256-GCM-SHA384
- ECDHE-RSA-AES256-SHA384
- ECDHE-ECDSA-AES256-SHA384
- ECDHE-RSA-AES256-SHA
- ECDHE-ECDSA-AES256-SHA
- SRP-DSS-AES-256-CBC-SHA
- SRP-RSA-AES-256-CBC-SHA
- DHE-DSS-AES256-GCM-SHA384
- DHE-RSA-AES256-GCM-SHA384
- DHE-RSA-AES256-SHA256
- DHE-DSS-AES256-SHA256
- DHE-RSA-AES256-SHA
- DHE-DSS-AES256-SHA
- ECDH-RSA-AES256-GCM-SHA384
- ECDH-ECDSA-AES256-GCM-SHA384
- ECDH-RSA-AES256-SHA384
- ECDH-ECDSA-AES256-SHA384
- ECDH-RSA-AES256-SHA
- ECDH-ECDSA-AES256-SHA
- AES256-GCM-SHA384
- AES256-SHA256

- AES256-SHA
- PSK-AES256-CBC-SHA
- ECDHE-RSA-DES-CBC3-SHA
- ECDHE-ECDSA-DES-CBC3-SHA
- SRP-DSS-3DES-EDE-CBC-SHA
- SRP-RSA-3DES-EDE-CBC-SHA
- EDH-RSA-DES-CBC3-SHA
- EDH-DSS-DES-CBC3-SHA
- ECDH-RSA-DES-CBC3-SHA
- ECDH-ECDSA-DES-CBC3-SHA
- DES-CBC3-SHA
- PSK-3DES-EDE-CBC-SHA
- ECDHE-RSA-AES128-GCM-SHA256
- ECDHE-ECDSA-AES128-GCM-SHA256
- ECDHE-RSA-AES128-SHA256
- ECDHE-ECDSA-AES128-SHA256
- ECDHE-RSA-AES128-SHA
- ECDHE-ECDSA-AES128-SHA
- SRP-DSS-AES-128-CBC-SHA
- SRP-RSA-AES-128-CBC-SHA
- DHE-DSS-AES128-GCM-SHA256
- DHE-RSA-AES128-GCM-SHA256
- DHE-RSA-AES128-SHA256
- DHE-DSS-AES128-SHA256
- DHE-RSA-AES128-SHA
- DHE-DSS-AES128-SHA
- ECDH-RSA-AES128-GCM-SHA256
- ECDH-ECDSA-AES128-GCM-SHA256
- ECDH-RSA-AES128-SHA256
- ECDH-ECDSA-AES128-SHA256
- ECDH-RSA-AES128-SHA
- ECDH-ECDSA-AES128-SHA
- AES128-GCM-SHA256
- AES128-SHA256
- AES128-SHA
- PSK-AES128-CBC-SHA

- ECDHE-RSA-RC4-SHA
- ECDHE-ECDSA-RC4-SHA
- ECDH-RSA-RC4-SHA
- ECDH-ECDSA-RC4-SHA
- RC4-SHA
- RC4-MD5
- PSK-RC4-SHA
- EDH-RSA-DES-CBC-SHA
- EDH-DSS-DES-CBC-SHA
- DES-CBC-SHA
- EXP-EDH-RSA-DES-CBC-SHA
- EXP-EDH-DSS-DES-CBC-SHA
- EXP-DES-CBC-SHA
- EXP-RC2-CBC-MD5
- EXP-RC4-MD5

10.2 iOS Client

The iOS client supports the following protocol versions:

- SSLv2
- SSLv3
- TLS1.0
- TLS1.1
- TLS1.2

The iOS client supports the following ciphers:

- SSL_RSA_EXPORT_WITH_RC4_40_MD5
- SSL_RSA_WITH_RC4_128_MD5
- SSL_RSA_WITH_RC4_128_SHA
- SSL_RSA_EXPORT_WITH_RC2_CBC_40_MD5
- SSL_RSA_WITH_IDEA_CBC_SHA
- SSL_RSA_EXPORT_WITH_DES40_CBC_SHA
- SSL_RSA_WITH_DES_CBC_SHA
- SSL_RSA_WITH_3DES_EDE_CBC_SHA
- SSL_DH_DSS_EXPORT_WITH_DES40_CBC_SHA
- SSL_DH_DSS_WITH_DES_CBC_SHA
- SSL_DH_DSS_WITH_3DES_EDE_CBC_SHA
- SSL_DH_RSA_EXPORT_WITH_DES40_CBC_SHA

- SSL_DH_RSA_WITH_DES_CBC_SHA
- SSL_DH_RSA_WITH_3DES_EDE_CBC_SHA
- SSL_DHE_DSS_EXPORT_WITH_DES40_CBC_SHA
- SSL_DHE_DSS_WITH_DES_CBC_SHA
- SSL_DHE_DSS_WITH_3DES_EDE_CBC_SHA
- SSL_DHE_RSA_EXPORT_WITH_DES40_CBC_SHA
- SSL_DHE_RSA_WITH_DES_CBC_SHA
- SSL_DHE_RSA_WITH_3DES_EDE_CBC_SHA
- SSL_DH_anon_EXPORT_WITH_RC4_40_MD5
- SSL_DH_anon_WITH_RC4_128_MD5
- SSL_DH_anon_EXPORT_WITH_DES40_CBC_SHA
- SSL_DH_anon_WITH_DES_CBC_SHA
- SSL_DH_anon_WITH_3DES_EDE_CBC_SHA
- SSL_FORTEZZA_DMS_WITH_FORTEZZA_CBC_SHA
- TLS_RSA_WITH_AES_128_CBC_SHA
- TLS_DH_DSS_WITH_AES_128_CBC_SHA
- TLS_DH_RSA_WITH_AES_128_CBC_SHA
- TLS_DHE_DSS_WITH_AES_128_CBC_SHA
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA
- TLS_DH_anon_WITH_AES_128_CBC_SHA
- TLS_RSA_WITH_AES_256_CBC_SHA
- TLS_DH_DSS_WITH_AES_256_CBC_SHA
- TLS_DH_RSA_WITH_AES_256_CBC_SHA
- TLS_DHE_DSS_WITH_AES_256_CBC_SHA
- TLS_DHE_RSA_WITH_AES_256_CBC_SHA
- TLS_DH_anon_WITH_AES_256_CBC_SHA
- TLS_ECDH_ECDSA_WITH_RC4_128_SHA
- TLS_ECDH_ECDSA_WITH_3DES_EDE_CBC_SHA
- TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA
- TLS_ECDH_ECDSA_WITH_AES_256_CBC_SHA
- TLS_ECDHE_ECDSA_WITH_RC4_128_SHA
- TLS_ECDHE_ECDSA_WITH_3DES_EDE_CBC_SHA
- TLS_ECDHE_ECDSA_WITH_AES_128_CBC_SHA
- TLS_ECDHE_ECDSA_WITH_AES_256_CBC_SHA
- TLS_ECDH_RSA_WITH_RC4_128_SHA
- TLS_ECDH_RSA_WITH_3DES_EDE_CBC_SHA

- TLS_ECDH_RSA_WITH_AES_128_CBC_SHA
- TLS_ECDH_RSA_WITH_AES_256_CBC_SHA
- TLS_ECDHE_RSA_WITH_RC4_128_SHA
- TLS_ECDHE_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA
- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA
- TLS_ECDH_anon_WITH_RC4_128_SHA
- TLS_ECDH_anon_WITH_3DES_EDE_CBC_SHA
- TLS_ECDH_anon_WITH_AES_128_CBC_SHA
- TLS_ECDH_anon_WITH_AES_256_CBC_SHA
- TLS_RSA_WITH_RC4_128_MD5
- TLS_RSA_WITH_RC4_128_SHA
- TLS_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_RSA_WITH_AES_128_CBC_SHA256
- TLS_RSA_WITH_AES_256_CBC_SHA256
- TLS_DH_DSS_WITH_3DES_EDE_CBC_SHA
- TLS_DH_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_DHE_DSS_WITH_3DES_EDE_CBC_SHA
- TLS_DHE_RSA_WITH_3DES_EDE_CBC_SHA
- TLS_DH_DSS_WITH_AES_128_CBC_SHA256
- TLS_DH_RSA_WITH_AES_128_CBC_SHA256
- TLS_DHE_DSS_WITH_AES_128_CBC_SHA256
- TLS_DHE_RSA_WITH_AES_128_CBC_SHA256
- TLS_DH_DSS_WITH_AES_256_CBC_SHA256
- TLS_DH_RSA_WITH_AES_256_CBC_SHA256
- TLS_DHE_DSS_WITH_AES_256_CBC_SHA256
- TLS_DHE_RSA_WITH_AES_256_CBC_SHA256
- TLS_DH_anon_WITH_RC4_128_MD5
- TLS_DH_anon_WITH_3DES_EDE_CBC_SHA
- TLS_DH_anon_WITH_AES_128_CBC_SHA256
- TLS_DH_anon_WITH_AES_256_CBC_SHA256
- TLS_RSA_WITH_AES_128_GCM_SHA256
- TLS_RSA_WITH_AES_256_GCM_SHA384
- TLS_DHE_RSA_WITH_AES_128_GCM_SHA256
- TLS_DHE_RSA_WITH_AES_256_GCM_SHA384
- TLS_DH_RSA_WITH_AES_128_GCM_SHA256

- TLS_DH_RSA_WITH_AES_256_GCM_SHA384
- TLS_DHE_DSS_WITH_AES_128_GCM_SHA256
- TLS_DHE_DSS_WITH_AES_256_GCM_SHA384
- TLS_DH_DSS_WITH_AES_128_GCM_SHA256
- TLS_DH_DSS_WITH_AES_256_GCM_SHA384
- TLS_DH_anon_WITH_AES_128_GCM_SHA256
- TLS_DH_anon_WITH_AES_256_GCM_SHA384
- TLS_ECDHE_ECDSA_WITH_AES_128_CBC_SHA256
- TLS_ECDHE_ECDSA_WITH_AES_256_CBC_SHA384
- TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA256
- TLS_ECDH_ECDSA_WITH_AES_256_CBC_SHA384
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256
- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384
- TLS_ECDH_RSA_WITH_AES_128_CBC_SHA256
- TLS_ECDH_RSA_WITH_AES_256_CBC_SHA384
- TLS_ECDHE_ECDSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_ECDSA_WITH_AES_256_GCM_SHA384
- TLS_ECDH_ECDSA_WITH_AES_128_GCM_SHA256
- TLS_ECDH_ECDSA_WITH_AES_256_GCM_SHA384
- TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256
- TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384
- TLS_ECDH_RSA_WITH_AES_128_GCM_SHA256
- TLS_ECDH_RSA_WITH_AES_256_GCM_SHA384
- SSL_RSA_WITH_RC2_CBC_MD5
- SSL_RSA_WITH_IDEA_CBC_MD5
- SSL_RSA_WITH_DES_CBC_MD5
- SSL_RSA_WITH_3DES_EDE_CBC_MD5

10.3 OpenSSL

10.3.1 Usage

Connect Mobile clients use OpenSSL for encrypting the data traffic for:

- Android – SIP

10.3.2 Ciphers Translation

To translate a cipher name to an OpenSSL cipher name, use the following link:

<https://www.openssl.org/docs/man1.0.2/apps/ciphers.html>

11 Appendix B: UC-One Connect DM Tag Provisioning Script

The number of custom DM tags has increased with each release, as many customers prefer tags for the new configuration parameters. To offer mechanisms for provisioning those custom DM tags more easily, this section contains a script that can be run on the Xtended Services Platform (Xsp) side to assign values to the custom DM tags. This script is especially intended for new deployments where most of the custom DM tags are intended to be used.

Note that this script is only valid for new deployments where custom DM tags are being created. To modify existing custom DM tags, the command in the following script must be changed from “add” to “set”.

Script template with only a few custom tags set (in a real deployment, you would need to populate a bigger list of custom tags).

```

%% ***** Connect_Tags - read file *****
%%
%% Instructions:
%% -----
%% - This read file can be used to create, add and set UC-One Connect
%% client custom tags
%% - Use %% to comment out any steps not required based on deployment specific
%% service requirements:
%% Step 1 -- for new deployments only, create initial tag set label
%% Step 2 -- add a new custom tag (an entry is required for each new tag)
%% Step 3 -- set value for an existing custom tag (entry required for each applicable tag)
%% Step 4 -- display and visually verify tag settings
%%
%% - Edit, modify file as needed respecting command syntax. Save file (e.g. Connect_Tags.txt)
%% - SFTP read file to AS under directory /tmp
%% - Login to AS, bwcli (login as admin)
%% - Execute the following command from bwcli: AS_CLI> r /tmp/Connect_Tags.txt
%% - Verify results
%%
%% -----
%% Step 1: Create Connect tag set label - Connect_Tags
%% -----
quit all;System;DeviceTagSet
add Connect_Tags
%% -----
%% Step 2: Add Connect custom tags
%% EXAMPLE – for all mobile tags see the list below-----
quit all;System;DeviceTagSet;Tags
add tagSetName Connect_Tags %ENABLE_AUDIOCALLS_MOBILE% true
%% -----
%% Step 3: Set Connect custom tags (if tag already exists)
%% EXAMPLE – for all mobile tags see the list below
set tagSetName Connect_Tags %ENABLE_CALL_PULL_MOBILE% tagvalue true
%% -----
%% Step 4: Verify custom tags have been correctly defined and set
%% -----
quit all;System;DeviceTagSet;Tags
get tagSetName Connect_Tags
quit all

```

The following lists all custom tags used by UC-One Connect with example (default) values. Note that some of the tags require values specific for the corresponding deployment (like server addresses). That is why these tags are added at the end of the script, but left empty, and additional set commands should be added to specify them.

```

add tagSetName Connect_Tags %ENABLE_REJECT_WITH_486_MOBILE% false
add tagSetName Connect_Tags %REJECT_WITH_XSI_MODE_MOBILE% decline_false
add tagSetName Connect_Tags %REJECT_WITH_XSI_DECLINE_REASON_MOBILE% busy
add tagSetName Connect_Tags %ENABLE_XSI_CALLS_MOBILE% true
add tagSetName Connect_Tags %ENABLE_XSI_MIDCALL_CONTROLS_MOBILE% false
add tagSetName Connect_Tags %ENABLE_XSI_MIDCALL_CONTROLS_WIDGET% true
add tagSetName Connect_Tags %ENABLE_TRANSFER_CALLS_MOBILE% true
add tagSetName Connect_Tags %ENABLE_XSI_TRANSFER_CALLS_MOBILE% false
add tagSetName Connect_Tags %TRANSFER_CALL_TYPE_MOBILE% full
add tagSetName Connect_Tags %ENABLE_TRANSFER_TO_CIRCUIT_SWITCH_MOBILE% true
add tagSetName Connect_Tags %ENABLE_AUDIOCALLS_MOBILE% true
add tagSetName Connect_Tags %ENABLE_AUDIO_QOS_MOBILE% false
add tagSetName Connect_Tags %AUDIO_QOS_TYPE_MOBILE% dscp
add tagSetName Connect_Tags %AUDIO_QOS_VALUE_MOBILE% 46
add tagSetName Connect_Tags %ENABLE_VIDEOCALLS_MOBILE% true
add tagSetName Connect_Tags %ENABLE_VIDEO_QOS_MOBILE% false
add tagSetName Connect_Tags %VIDEO_QOS_TYPE_MOBILE% dscp
add tagSetName Connect_Tags %VIDEO_QOS_VALUE_MOBILE% 34
add tagSetName Connect_Tags %VIDEO_SHOW_INCOMING_CALL_AS_AUDIO_MOBILE% false
add tagSetName Connect_Tags %ENABLE_CONFERENCE_CALLS_MOBILE% true
add tagSetName Connect_Tags %ENABLE_XSI_CONFERENCE_CALLS_MOBILE% false
add tagSetName Connect_Tags %CONFERENCE_CALLS_AUTO_JOIN_PAUSE%
add tagSetName Connect_Tags %ENABLE_ADAPTIVE_QUALITY_MOBILE% true
add tagSetName Connect_Tags %ENABLE_CALL_PULL_MOBILE% true
add tagSetName Connect_Tags %ENABLE_VOIP_MODE_MOBILE% false
add tagSetName Connect_Tags %CELLULAR_DATA_VOIP_CALLS_STATE_MOBILE% always-on
add tagSetName Connect_Tags %ENABLE_CALLS_AUTO_RECOVERY_MOBILE% true
add tagSetName Connect_Tags %CALLS_AUTO_RECOVERY_DATA_TO_WIFI_MODE_MOBILE% keep
add tagSetName Connect_Tags %ENABLE_ONE_TIME_VOIP% false
add tagSetName Connect_Tags %ENABLE_MY_ROOM_MOBILE% false
add tagSetName Connect_Tags %MY_ROOM_SHOW_MOBILE% false
add tagSetName Connect_Tags %ENABLE_GUEST_CLIENT_SUPPORT_MOBILE% false
add tagSetName Connect_Tags %CONFERENCE_BRIDGE_MEDIA_TYPE_MOBILE% audio
add tagSetName Connect_Tags %CONFERENCE_TYPE% uvs
add tagSetName Connect_Tags %ENABLE_WEBEX_MEETINGS_CROSS_LAUNCH% false
add tagSetName Connect_Tags %UPDATE_EMERGENCY_LOCATION_VERIFICATION_TIMEOUT_MOBILE% 900
add tagSetName Connect_Tags %USE_ECACS_BASIC_AUTH_MOBILE% false
add tagSetName Connect_Tags %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_LOGIN_MOBILE% false
add tagSetName Connect_Tags %UPDATE_EMERGENCY_LOCATION_REQUIRED_MOBILE% false
add tagSetName Connect_Tags %UPDATE_EMERGENCY_LOCATION_RECURRING_MOBILE% always
add tagSetName Connect_Tags %ENABLE_UPDATE_EMERGENCY_LOCATION_AT_WILL_MOBILE% false
add tagSetName Connect_Tags %EMERGENCY_CALL_DIAL_SEQUENCE_MOBILE% cs-only
add tagSetName Connect_Tags %PN_FOR_CALLS_CONNECT_SIP_ON_ACCEPT_MOBILE% false
add tagSetName Connect_Tags %DEPLOYMENT_TYPE_MOBILE% ott
add tagSetName Connect_Tags %ENABLE_NATIVE_DIALER% false
add tagSetName Connect_Tags %DEFAULT_DIALING_METHOD% call-through
add tagSetName Connect_Tags %ENABLE_DIALING_CALL_THROUGH% true
add tagSetName Connect_Tags %ENABLE_DIALING_CALL_BACK% true
add tagSetName Connect_Tags %DEFAULT_DIALING_USE_VOIP% false
add tagSetName Connect_Tags %ENABLE_PERSONA_MANAGEMENT% false
add tagSetName Connect_Tags %ENABLE_DIALING_DIALPAD% true
add tagSetName Connect_Tags %ENABLE_DIALING_PREVIEW_URL_SCHEMA_NUMBER_TO_DIAL% false
add tagSetName Connect_Tags %SIDE_MENU_CFA_VISIBLE% true
add tagSetName Connect_Tags %SIDE_MENU_DND_VISIBLE% true
add tagSetName Connect_Tags %ENABLE_WEB_BUTTON% true
add tagSetName Connect_Tags %ENABLE_WEB_CALL_SETTINGS% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CFA_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_DND_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_ACR_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CFB_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CFNr_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CFNA_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_SIMRING_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_SEQRING_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_RO_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_ACB_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CW_VISIBLE% true

```

```

add tagSetName Connect_Tags %WEB_CALL_SETTINGS_CLIDB_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_PA_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_BWA_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_STANDARD_SETTINGS_CC_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_BASIC_SETTINGS_CC_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_PREMIUM_SETTINGS_CC_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_BWM_VISIBLE% true
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_VM_VISIBLE% true
add tagSetName Connect_Tags %ENABLE_PERSONAL_ASSISTANT_PRESENCE% false
add tagSetName Connect_Tags %ENABLE_PERSONAL_ASSISTANT_PRESENCE_SILENT_ALERT% true
add tagSetName Connect_Tags %ENABLE_PERSONAL_ASSISTANT_PRESENCE_ALERT_ME_FIRST% true
add tagSetName Connect_Tags %ENABLE_PASSWORD_UPDATE% true
add tagSetName Connect_Tags %PASSWORD_UPDATE_WARN_BEFORE_DAYS% 14
add tagSetName Connect_Tags %ENABLE_MY_TELEPHONE_NUMBER_MOBILE% true
add tagSetName Connect_Tags %CONTACTS_SORT_ORDER_VISIBLE% false
add tagSetName Connect_Tags %CONTACTS_SORT_ORDER_DEFAULT% first
add tagSetName Connect_Tags %CONTACTS_DISPLAY_NAME_ORDER_VISIBLE% false
add tagSetName Connect_Tags %CONTACTS_DISPLAY_NAME_ORDER_DEFAULT% first
add tagSetName Connect_Tags %CONTACTS_SHORT_NAMES_VISIBLE% false
add tagSetName Connect_Tags %CONTACTS_SHORT_NAMES_DEFAULT% fullname
add tagSetName Connect_Tags %CONTACTS_CACHE_CONTROL_CACHE_LIMIT% 100
add tagSetName Connect_Tags %CONTACTS_CACHE_CONTROL_CACHE_INTERVAL% 86400
add tagSetName Connect_Tags %ENABLE_CALL_HISTORY% true
add tagSetName Connect_Tags %ENABLE_ENHANCED_CALL_HISTORY% false
add tagSetName Connect_Tags %DIALER_KEYBOARD_MODE_MOBILE% digits
add tagSetName Connect_Tags %ENABLE_MWI_MOBILE% true
add tagSetName Connect_Tags %ENABLE_SIP_INFO_MOBILE% true
add tagSetName Connect_Tags %ENABLE_USE_RPORT_MOBILE% false
add tagSetName Connect_Tags %RPORT_USE_LOCAL_PORT_MOBILE% false
add tagSetName Connect_Tags %SBC_PORT_MOBILE% 5075
add tagSetName Connect_Tags %USE_PROXY_DISCOVERY_MOBILE% false
add tagSetName Connect_Tags %ENABLE_PROXY_DISCOVERY_RFC5922_CERT_VALIDATION_MOBILE% true
add tagSetName Connect_Tags %SOURCE_PORT% 5075
add tagSetName Connect_Tags %100_REL_ENABLED_MOBILE% true
add tagSetName Connect_Tags %TRANSPORTS_TYPE_MOBILE% tls
add tagSetName Connect_Tags %TLS_VERSION% ssl23
add tagSetName Connect_Tags %ENABLE_SIGN_IN_TIMER_MOBILE% false
add tagSetName Connect_Tags %SIGN_IN_TIMER_PERIOD_MOBILE% 5000
add tagSetName Connect_Tags %USE_EXPIRES_HACK_MOBILE% false
add tagSetName Connect_Tags %USE_MEDIASEC_MOBILE% false
add tagSetName Connect_Tags %ENABLE_SIP_UPDATE_SUPPORT_MOBILE% false
add tagSetName Connect_Tags %ENABLE_SIP_QOS_MOBILE% false
add tagSetName Connect_Tags %ENABLE_TCP_RESET_FAILOVER_MOBILE% false
add tagSetName Connect_Tags %SRTP_MODE_MOBILE% optional
add tagSetName Connect_Tags %SRTP_ENABLED_MOBILE% false
add tagSetName Connect_Tags %ENABLE_RE-KEYING_MOBILE% true
add tagSetName Connect_Tags %CALL_QUALITY_AUDIO_ENABLED_MOBILE% false
add tagSetName Connect_Tags %CALL_QUALITY_VIDEO_ENABLED_MOBILE% false
add tagSetName Connect_Tags %RTP_AUDIO_PORT_RANGE_START% 8500
add tagSetName Connect_Tags %RTP_AUDIO_PORT_RANGE_END% 8598
add tagSetName Connect_Tags %RTP_VIDEO_PORT_RANGE_START% 8600
add tagSetName Connect_Tags %RTP_VIDEO_PORT_RANGE_END% 8698
add tagSetName Connect_Tags %RTP_MTU% 1150
add tagSetName Connect_Tags %ENABLE_RTP_MUX_MOBILE% true
add tagSetName Connect_Tags %CHANNEL_HEARTBEAT_MOBILE% 10000
add tagSetName Connect_Tags %ENABLE_VISUAL_VOICEMAIL_MOBILE% false
add tagSetName Connect_Tags %ENABLE_AUTO_CONFIGURE_ANYWHERE% true
add tagSetName Connect_Tags %ENABLE_MESSAGE_GATEWAY% true
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_MESSAGING_ENABLED% true
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_PRESENCE_ENABLED% true
add tagSetName Connect_Tags %ENABLE_UPDATE_EMERGENCY_LOCATION_MOBILE% false
add tagSetName Connect_Tags %AUTODETECT_CONFERENCE% true
add tagSetName Connect_Tags %CALL_RECORDING_MOBILE% false
add tagSetName Connect_Tags %ENABLE_CONTACTS_ENTERPRISE_SEARCH_MOBILE% true
add tagSetName Connect_Tags %ENABLE_CONTACTS_ENTERPRISE_COMMON_SEARCH_MOBILE% false
add tagSetName Connect_Tags %ENABLE_CONTACTS_PERSONAL_SEARCH_MOBILE% false
add tagSetName Connect_Tags %ENABLE_CONTACTS_GROUP_COMMON_SEARCH_MOBILE% false
add tagSetName Connect_Tags %SHOW_LOGIN_INFORMATIONAL_DIALOG_MOBILE% false
add tagSetName Connect_Tags %SHOW_INFORMATIONAL_DIALOG_ON_EVERY_LOGIN_MOBILE% false
add tagSetName Connect_Tags %NUMBER_OF_RINGS_RANGE_MIN_VALUE_MOBILE% 0
add tagSetName Connect_Tags %NUMBER_OF_RINGS_RANGE_MAX_VALUE_MOBILE% 20
add tagSetName Connect_Tags %ENABLE_SINGLE_ALERTING% true
add tagSetName Connect_Tags %ENABLE_CALL_STATISTICS_MOBILE% false
add tagSetName Connect_Tags %ENABLE_BROADWORKS_ANYWHERE_ADVANCED_OPTIONS_MOBILE% false

```



```

add tagSetName Connect_Tags %ANYWHERE_ADVANCED_OPTIONS_CALL_CONTROL_DEFAULT_MOBILE% true
add tagSetName Connect_Tags
%ANYWHERE_ADVANCED_OPTIONS_DIVERSION_INHIBITOR_DEFAULT_MOBILE% false
add tagSetName Connect_Tags
%ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ANSWER_CONFIRMATION_MOBILE% false
add tagSetName Connect_Tags
%ANYWHERE_ADVANCED_OPTIONS_DEFAULT_ALERT_ALL_LOCATIONS_MOBILE% false
add tagSetName Connect_Tags %CALL_HISTORY_SHOW_DIRECTORY_SEARCH% false
add tagSetName Connect_Tags %ONE_TIME_VOIP_REGISTER% false
add tagSetName Connect_Tags %PN_FOR_CALLS_USE_REGISTRATION_V1% false
add tagSetName Connect_Tags %ENABLE_VERSION_CONTROL% false
add tagSetName Connect_Tags %PN_FOR_CALLS_RING_TIMEOUT_SECONDS_MOBILE% 35
add tagSetName Connect_Tags %TOKEN_EXPIRY_WARN_BEFORE_DAYS% 3
add tagSetName Connect_Tags %ENABLE_MEDIA_WAITING_TONE_MOBILE% false
add tagSetName Connect_Tags %ANDROID_MUST_VERSION%
add tagSetName Connect_Tags %ANDROID_RECOMMENDED_VERSION%
add tagSetName Connect_Tags %ANDROID_DOWNLOAD_LINK%
add tagSetName Connect_Tags %IOS_MUST_VERSION%
add tagSetName Connect_Tags %IOS_RECOMMENDED_VERSION%
add tagSetName Connect_Tags %IOS_DOWNLOAD_LINK%
add tagSetName Connect_Tags %UPDATE_EMERGENCY_LOCATION_URL_MOBILE%
add tagSetName Connect_Tags %SBC_ADDRESS%
add tagSetName Connect_Tags %CALL_QUALITY_SERVICE_URI%
add tagSetName Connect_Tags %CALL_QUALITY_LOCAL_GROUP%
add tagSetName Connect_Tags %UMS_HTTP_PORT%
add tagSetName Connect_Tags %UMS_SRV_SERVICE%
add tagSetName Connect_Tags %UMS_HTTP_CONTEXT%
add tagSetName Connect_Tags %CHECK_DEVICE_EMERGENCY_NUMBERS_MOBILE%
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_LANDING_PAGE%
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_URL%
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_USE_TITLE%
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_ENCODE%
add tagSetName Connect_Tags %DEPLOYMENT_MODEL_WEBVIEW_LANDING_PAGE_REFRESH_POLICY%
add tagSetName Connect_Tags %WEB_BUTTON_DISPLAY_MODE%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_URL_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_ENCODE_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_TARGET_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_KEEP_LOADED_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_USE_PAGE_TITLE_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_TITLE_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_DESCRIPTION_1%
add tagSetName Connect_Tags %WEB_BUTTON_LIST_LINK_REF_1%
add tagSetName Connect_Tags %MESSAGE_GATEWAY_SECURE%
add tagSetName Connect_Tags %GUEST_CLIENT_URL%
add tagSetName Connect_Tags %WEB_CALL_SETTINGS_URL%
add tagSetName Connect_Tags %MWI_MODE_MOBILE%
add tagSetName Connect_Tags %DOMAIN_OVERRIDE%
add tagSetName Connect_Tags %SIP_QOS_TYPE_MOBILE%
add tagSetName Connect_Tags %SIP_QOS_VALUE_MOBILE%
add tagSetName Connect_Tags %XSI_ROOT%
add tagSetName Connect_Tags %XSI_ACTIONS_PATH_MOBILE%
add tagSetName Connect_Tags %XSI_EVENTS_PATH_MOBILE%
add tagSetName Connect_Tags %CONFERENCE_TITLE%
add tagSetName Connect_Tags %BRIDGE_ID%

```

The following lists all system tags used by UC-One Connect.

```

%BWDN-1%
%BWLINEPORT-1%
%BWAUTHPASSWORD-1%
%BWE164-1%
%BWAUTHUSER-1%
%BWHOST-1%
%BWVOICE-PORTAL-NUMBER-1%
%BW_USER_IMP_ID-1%
%BW_USER_IMP_PWD-1%
%BW_IMP_SERVICE_NET_ADDRESS-1%
%BWNETWORK-CONFERENCE-SIPURI-1%

```

12 Acronyms and Abbreviations

This section lists the acronyms and abbreviations found in this document. The acronyms and abbreviations are listed in alphabetical order along with their meanings.

| | |
|-------|-----------------------------------|
| ACB | Automatic Callback |
| ACD | Automatic Call Distribution |
| ACR | Anonymous Call Rejection |
| AES | Advanced Encryption Standard |
| ALG | Application Layer Gateway |
| API | Application Programming Interface |
| APK | Application Package |
| ARS | Automatic bit Rate Selection |
| AS | Application Server |
| AVP | Audio Visual Profile |
| BW | BroadWorks |
| BWA | BroadWorks Anywhere |
| BWKS | BroadWorks |
| BWM | BroadWorks Mobility |
| BYOD | Bring Your Own Device |
| CC | Call Center |
| CFB | Call Forwarding Busy |
| CFNA | Call Forwarding No Answer |
| CFNR | Call Forwarding Not Reachable |
| CIF | Common Intermediate Format |
| CLI | Command Line Interface |
| CLID | Calling Line Identity |
| CLIDB | Calling Line ID Delivery Blocking |
| CM | Conditional Mandatory |
| CRLF | Carriage Return Line Feed |
| CS | Circuit-Switched |
| CW | Call Waiting |
| DB | Database |
| DM | Device Management |
| DND | Do Not Disturb |
| DNS | Domain Name System |
| DTAF | Device Type Archive File |

| | |
|-------|--|
| ECACS | Emergency Call Address Change Service |
| FQDN | Fully Qualified Domain Name |
| HMAC | Hash Message Authentication Code |
| IM | Instant Messaging |
| IM&P | Instant Messaging and Presence |
| IOT | Interoperability Testing |
| IP | Internet Protocol |
| JID | Jabber Identifier |
| M/O | Mandatory/Optional |
| MNO | Mobile Network Operator |
| MTU | Maximum Transmission Unit |
| MUC | Multi-User Chat |
| MWI | Message Waiting Indicator |
| NAPTR | Naming Authority Pointer |
| NAT | Network Address Translation |
| OTT | Over The Top |
| PA | Personal Assistant |
| PAI | P-Asserted-Identity |
| PEM | P-Early Media |
| PLI | Picture Loss Indication |
| PLMN | Public Land Mobile Network |
| PN | Push Notification |
| QCIF | Quarter Common Intermediate Format |
| QoS | Quality of Service |
| RO | Remote Office |
| RTCP | Real-Time Control Protocol |
| RTP | Real Time Protocol |
| SaaS | Software as a Service |
| SAN | Subject Alternative Name |
| SASL | Simple Authentication and Security Layer |
| SAVP | Secure Audio Video Profile |
| SBC | Session Border Controller |
| SCA | Shared Call Appearance |
| SCF | Session Continuity Function |
| SCTP | Stream Control Transmission Protocol |
| SDP | Session Definition Protocol |

| | |
|---------|--|
| SEQRING | Sequential Ring |
| SIMRING | Simultaneous Ring |
| SIP | Session Initiation Protocol |
| SRTCP | Secure Real-Time Control Protocol |
| SRTP | Secure Real-time Transport Protocol |
| SSL | Secure Sockets Layer |
| SUBQCIF | Sub Quarter CIF |
| TCP | Transmission Control Protocol |
| TLS | Transport Layer Security |
| TTL | Time To Live |
| UDP | User Datagram Protocol |
| UI | User Interface |
| UMS | Messaging Server (Cisco BroadWorks) |
| URI | Uniform Resource Identifier |
| UVS | Video Server (Cisco BroadWorks) |
| VGA | Video Graphics Array |
| VoIP | Voice Over IP |
| VVM | Visual Voice Mail |
| XMPP | Extensible Messaging and Presence Protocol |
| XR | Extended Report |
| Xsp | Xtended Services Platform |
| Xsi | Xtended Services Interface |