Release Notes for Cisco Hosted Unified Communications Services, Release 7.1(a)

August 12, 2010

These release notes describe updated information, caveats and known issues for the Cisco Hosted Unified Communications Services (Hosted UCS), Release 7.1(a).

Contents

These release notes provide the following information:

Related Documentation

The following related documentation is available for Cisco Hosted Unified Communications Services, Release 7.1(a).

Getting Started with Cisco Hosted Unified Communications Services, Release 7.1(a)

Provides a high-level overview of the Hosted UCS platform and describes how to configure and apply static configuration to the platform components. This guide summarizes the options provided by VOSS Unified Services Manager (USM) for configuring and managing the platform components, explains how to use USM to load bulk data during initial configuration of the components, and how to backup, restore, and clear the platform components. To view this document, see the following URL:

http://www.cisco.com/en/US/docs/voice_ip_comm/hucs/7.1a/user/getstart7.1a.html

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Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Solution Reference Network Design for Cisco Hosted Unified Communications Services, Release 7.1(a)

Provides a detailed description of the Hosted UCS product design and architecture. The document describes the product components used to build the Hosted UCS solution, both Cisco products and partner products, and the suite of services that are provided by this solution. It describes the network architecture, including call scenarios, legacy PBX integration, and geographic redundancy.

The document also defines the supported Hosted UCS deployment models, provides guidelines regarding the required network infrastructure, and describes how the solution fulfills regulatory requirements, such as service provider requirements.

To obtain a copy of the Solution Reference Network Design document for Cisco Hosted Unified Communications Services, Release 7.1(a), contact your Cisco representative.

Software Matrix for Cisco Hosted Unified Communications Services, Release 7.1(a)

Provides a comprehensive list of the software and hardware components that are supported for the Cisco Hosted Unified Communications Services, Release 7.1(a). To view this document, see the following URL:

http://www.cisco.com/en/US/docs/voice_ip_comm/hucs/7.1a/softwarematrix/hucsmatrix71a.html

Provisioning Guide for Cisco Hosted Unified Communications Services, Release 7.1(a)

Describes how to use VOSS USM to provision the components of the Hosted UCS platform. To view this document, see the following URL:

http://www.cisco.com/en/US/docs/voice_ip_comm/hucs/7.1a/provision/provision7.1a.html

New Features and Changes

This section contains new and changed features introduced in Cisco Hosted UCS, Release 7.1(a). It includes the following topics:

Cisco Unified Communications Manager 7.1 and PGW 9.8 Support

Cisco Hosted Unified Communications Services, Release 7.1(a) introduces support for Cisco Unified Communications Manager 7.1.

- The USM 7.x driver can now be used to successfully add configurations to Cisco Unified Communications Manager 7.1.
- The Softkey Templates configured in Cisco Unified Communications Manager can be imported into USM, and then used when you are provisioning the IP Phone.

iFint Feature

The iFint Feature is introduced in HUCS7.1(a), to make use of FINT transparent to the user, by using only (SLC + EXT) as the Directory Number of the phone rather than the FINT as DN. Notice that FINTs are still used internally for routing from PGW to Cisco Unified CM and within Cisco Unified CM, but its use remains transparent to the end user.

Movius 4.2 - Voicemail and Auto Attendant Integration

The Movius AutoAttendant integration feature enables PGW and Cisco Unified Communications Manager to route calls to the Movius AutoAttendant from any PSTN or Hosted UCS phone. After selecting an option on a menu, these calls can be transferred to another Hosted UCS or PSTN phone.

The Movius voicemail dial plans were updated to enable making and receiving calls to and from Movius AutoAttendant. A new trunk was added to Cisco Unified CM, for Movius voicemail calls, to allow using a media termination point for the AutoAttendant calls.

Movius Auto Attendant can be integrated to PGW via SBC in Hosted UCS 7.1(a) Release.

Mobile Connect - Single Number Reach

The purpose of this feature is to provide support for the Cisco Unified CM (version 6.1 and 7.1) Mobile Connect feature. Hosted UCS 7.1 includes simultaneous ring only, specifically Desk Phone Pickup and Remote Destination Phone Pickup. The Cisco Unifed CM Mobile Connect allows users to manage business calls using a single phone number to pick up in-progress calls on the desktop phone or cellular phone without losing the connection, and to originate enterprise calls from the cellular phone.

Cisco Emergency Responder

In Hosted UCS 7.1(a), Cisco Emergency Responder (Cisco ER) can be used to manage emergency calls in the telephony network so that it is possible to respond to these calls effectively. This enables the service provider to comply with local ordinances concerning the handling of emergency calls.

It is possible to provision some locations to handle emergency calls via the default Cisco Hosted UCS method and provision others to use Cisco ER. For the locations provisioned to use Cisco ER, emergency calls are sent to the PSTN with the Calling Party Number being replaced with the Emergency Location Identification Number (ELIN) which enables the Public Safety Answering Point (PSAP) operator to correctly identify the location of the caller. Cisco ER also enables the PSAP operator to call the emergency caller back.

Local PSTN Breakout Support Without SRST integrated to Cisco Unified Communications Manager

Local PSTN Breakout support in this release was tested to ensure that calls to PSTN can be handled correctly from/to locations provisioned on Unified CM with Local Breakout.

In Hosted UCS 7.1(a), the administrator can provision local gateways without SRST support integrated to Cisco Unified CM using H323 Protocol only. Calls to/from PSTN can be routed via PRI or BRI interfaces. The format of the calling and called party number and Nature of Address (NOA) can also be configured in various ways. Additionally, calls to and from PSTN from one location can be routed via a single trunk, or optionally, the administrator can provision two trunks to separate Local and National/International calls. On the Local Gateway, a number of TCL applications developed by Cisco are used to take over the role of the default application.

Note The default application is used to control voice dial-peers in IOS, which is part of the IOS built-in call control of IOS that basically binds two call legs whose characteristics are defined by the configured voice dial-peers.

These applications are then configured on each voice dial peer and each peer verifies on each voice call whether calling and/or called number needs to be translated in the same way as the voice translation rules did in the previous Hosted UCS releases.

Emergency Call Support (Non-CER)

Emergency call support (Non-CER) is tested in this release for feature parity with the earlier release of Hosted UCS. Also, this release has enhanced the emergency call support by providing the option for the customer to select either the DDI or the Emergency published number to be sent out on the emergency call. The emergency calls are handed over to the PSTN and then the PSTN network has the responsibility to correctly route them. The Emergency calls can be routed via both Central and Local Breakout depending on the Location configuration.

Per-Customer "Block Off-Net to Off-Net Call Transfer" Feature Support

The Block Off-Net to Off-Net Call Transfer (BO2OCT) feature blocks all transfers made by Cisco Unified CM phones from external incoming calls to outgoing external calls. In order to do this, a new trunk was introduced on the Cisco Unified CM in order to differentiate OffNet and OnNet calls.

In Hosted UCS 5.1(b) MR1, this feature could only be configured for a whole Cisco Unified CM cluster. In Hosted UCS 7.1(a), this feature can be enabled or disabled for each customer.

Extension dialing between two locations aka Linked Location

Extension dialing between sites aka Linked Location feature enables the customer to have extension dialling supported between two different physical locations. The site code for the linked location is same for the locations enabled for extension dialing between sites. All these locations must have to be provisioned on the same Cisco Unified CM cluster. Linked Location also supports Local Breakout and standard Cisco Unified CM features supported for HUCS standard location.

Forced Authorization code

Forced Authorization codes are used in the Cisco Unified CM to regulate the types of calls that certain users can place. The FAC feature requires changes to be made to the Cisco Unified CM route patterns. When the call hits a FAC enabled route pattern, the user is prompted to enter the FAC. Only on entering the correct code, the call is completed otherwise the call is rejected.

Legacy PBX integration

The HUCS7.1(a) release supports Legacy PBX integration at PGW using MGCP Controlled Cisco ISR gateways. Legacy PBX integration is supported for QSIG/Q931/DPNSS protocols.

Analog Phone support

Analog phones can be registered to Cisco Unified CM just like IP phones when connected via VG224 gateways using MGCP/SCCP protocol.

Important Notes

This section includes the following topics related to Hosted UCS Release 7.1(a) features:

- Change in Static Configuration, page 5
- Feature Support for the SIP and SCCP Endpoints in Hosted UCS 7.1(a), page 6
- Product Provisioning Method, page 7

Change in Static Configuration

This section compares the static configuration in Cisco Hosted UCS 7.1(a) to the previous releases of Cisco Hosted UCS, and lists the changes:

- Earlier to PGW9.8, for every type of trunk (SIP/MGCP/EISUP), the trunk properties where set using the component trnkprop associated to the trunk group. From PGW 9.8, for the eisup trunk, these trunk properties are set using the component trunk group profile and then associating this trunk group profile to the trunk group. Example for the same is given below:

  ```
  prov-add: profile:name="lvl1eisupf-1001",type="EISUPPROFILE",custgrpid="ICCM",
  allowh323hairpin="1",gatewayrbtonesupport="1"
  prov-add:trnkgrpprof:name="1001",profile=" lvl1eisupf-1001"
  ```
- With the introduction of ifint, for the MWI to work, apart from the manual configuration that is required, we also need to set the the following service parameter for the CULM Service that needs to be set on the Cisco Unified CM 7.1 . This parameter indicates the Cisco Unified CM that the received Voicemailbox number needs to undergo a further translation to arrive at the right DN for which MWI has to be turned on/off.

  ```
  Multiple Tenant MWI Modes = True
  ```
- Whenever the locations have SNR enabled on the IP Phones. Make sure that the following service parameters are set on the Cisco Unified CM.
  - Matching Caller ID with Remote Destination—Partial Match

  - Number of Digits for Caller ID Partial Match Required Field—14 digits
- For the Movius AA integration to SBC, 100rel support needs to enabled on the SIP trunk incoming from SBC. For Direct integration this is not required.
- On the Movius IP unity for the AA integration to SBC, the transfer mode should be set to "Supervised" instead of "Monitoring" which is used for the direct Sip trunk integration of Movius to PGW.

Feature Support for the SIP and SCCP Endpoints in Hosted UCS 7.1(a)

Table 1 summarizes the Unified Communications Manager features that are supported with SIP and SCCP endpoints in Cisco Hosted Unified Communications Services Release 7.1(a):

Table 1 Feature Support for SIP and SCCP Endpoints in Cisco Hosted Unified Communications Services, Release 7.1(a)

Cisco Unified Communications Manager Feature Name	Support for SCCP Endpoints in Cisco Hosted UCS 7.1(a)	Support for SIP Endpoints in Cisco Hosted UCS 7.1(a)
Abbreviated dial	Yes	Yes
Answer and answer release	Yes	Yes
Auto-answer and intercom	Yes	Yes
Barge	No	No
Call connection	Yes	Yes
Call coverage	Yes	Yes
Call forward-All (off net and on net), busy, and no answer		Yes
Call hold and retrieve	Yes	Yes
Call park and pickup	Yes	Yes
Call waiting and retrieve	Yes	Yes
Calling line identification (CLID) and calling party name identification (CNID)	Yes	Yes
(View) Conference list and drop any party (impromptu conference)	Yes	Yes
Direct inward dial (DID) and direct outward dial (DOD)	Yes	Yes
Directories-Missed, placed, and received calls list stored on selected IP phones	Yes	Yes
Extension mobility support	Yes	Yes
Hands-free, full-duplex speakerphone	Yes	Yes
Last number redial (on and off net)	Yes	Yes
Multiple calls per line appearance	Yes	Yes
Multiple line appearances per phone	Yes	Yes
Music on hold	Yes	Yes
Mute capability from speakerphone and handset	Yes	Yes
On-hook dialing	Yes	Yes

Table 1 Feature Support for SIP and SCCP Endpoints in Cisco Hosted Unified Communications Services, Release 7.1(a) (continued)

Cisco Unified Communications Manager Feature Name	Support for SCCP Endpoints in Cisco Hosted UCS 7.1(a)	Support for SIP Endpoints in Cisco Hosted UCS 7.1(a)
Privacy	No	No
K-factor	Yes	No
Recent dial list-Calls to phone, calls from phone, autodial, and edit dial	Yes	Yes
Service URL-single-button access to IP phone service	No	No
Speed dial-Multiple speed dials per phone	Yes	Yes
Station volume controls (audio and ringer)	Yes	Yes
Transfer-Blind, consultative, and direct transfer of two parties on a line	Yes	Yes

Product Provisioning Method

Table 2 outlines the supported product model for the Cisco Hosted Unified Communications Services, Release 7.1(a); the provisioning method per product is shown:

Table 2 Provisioning Method for Cisco Hosted Unified Communications Services, Release 7.1(a)

	Manual Provision	USM Provision	Not Supported
Cisco Unified Communications Manager 7.1		Yes	
Cisco Unified Communications Manager 6.1		Yes	
Cisco Unified IP Phones		Yes	
Cisco Analog Telephone Adaptors		Yes	
Cisco IP Communicator		Yes	
Cisco Unified Personal Communicator			X
Cisco Unified Video Advantage	Yes		
Cisco Unity	Yes		
Cisco Unity Connection			X
Cisco Unity Express			X
Cisco Unified MeetingPlace			
Cisco Unified MeetingPlace Express			X
Cisco Unified Hosted Contact Center			X
Cisco Unified Presence			X
Cisco Unified Mobile Communicator			X
Cisco Unified Application Environment			X
Cisco Unified CallConnector Mobility			X
Cisco Unified Videoconferencing System			X

Table 2 Provisioning Method for Cisco Hosted Unified Communications Services, Release 7.1(a) (continued)

	Manual Provision	USM Provision	Not Supported
Cisco Unified Operations Manager	Yes		
Cisco Unified Service Monitor	Yes		
Cisco WebEx			X
Cisco Unified Communications Manager Express			X
Cisco Unified Communications 500 Series for Small Business			X
Cisco Unified Conferencing for Telepresence			X
Cisco Fax Server			X
Cisco PGW 2200 Softswitch		Yes	
Cisco Gatekeeper		Yes	
Cisco ISRouters for PSTN Breakout (H.323 protocol)		Yes	
Cisco H.323 Signaling Interface (HSI)	Yes		
Cisco AS5000 series routers as Central PSTN Breakout	Yes		
Cisco PSTN Gateways (local gateway)		Yes	
Cisco Emergency Responder	Yes		
Cisco Billing and Measurements Server (BAMS)	Yes		
Cisco Integrated Services Routers	Yes		
Cisco ASA/PIX/FWSM	Yes		
Movius (IP Unity) AutoAttendant	Yes		
Movius (IP Unity) VM		Yes	
Movius (IP Unity) Conferencing			X

Limitations and Observations in Hosted UCS Release 7.1(a)

This section includes observations about the features and enhancements introduced in Cisco Hosted Unified Communications Services, Release 7.1(a). It includes the following topics:

- Limitations and Observations for Analog Phone Support using VG224, page 14

Limitations and Observations In support for iFint

The limitations and observations affecting the support for iFint are as follows:

- Following number display issues where observed during the test. As understood this issues were caused due to limitation of the Cisco Unified CM in restricting the connected number from getting displayed.
  - Scenario 1: 000 calls 001 (SLC - 111) On Calling Phone the number is displayed as 111001 and alerting name is shown as 001 "001 (111001)". After the call is answered the connected number is shown as "To 111001". Where else the called party phone displays only the extension number of the Calling party that is "000". Also the issue noticed here is that the number is shown without a prefix of "8 ".
  - Scenario 2: During a conference, when the conference controller (A) dials ISP+SLC+EXT or E.164 number to reach the first party (B), FINT of the controller (A) is shown in the conferee list. Also in this scenario, no MOH is heard on B when it is put on hold while C is added to the conference.

    For example, A (112003 SCCP) dials 8112002 to reach B (112002 SIP). B answers it. A shows 'To 8112002'. A presses conf button and dials 000 to put C (112000 SIP) in conference (A shows 'To 112000' while calling), and presses conf button again after C answers it. Note that A dials an ISP+SLC-EXT first. In this case, FINT of A is displayed in the conferee list. During the conference A and C show 'To conference' and B shows 'From 8112003'. And when B (002) is disconnected from the conference, C shows 'From 1001008112003' and A shows 'To 112000'
  - Scenario 3: Phone A(111 002) dials e164 number i.e. 901402111003 to reach phone B (111 003). Both the phones are there in the same location. In Phone A, the number is displayed as 01402111003 and in Phone B it's displayed as 8111002. This happens due to DisableH323Notify parameter in HSI. Here the B will be sending back 111003 but the HSI will prevent that from being received at the caller's phone. After enabling "NotifyMessageEnabled" in HSI, for PSTN Calls (Forced on-net and also actual e164 dialing across customer), the calling party number is displayed as SLC+EXTN (i.e. 118002 etc). So this flag by default is disabled on the HSI.
  - Scenario 4: Suppose the hunt pilot number configured is 01637112105. Two phones i.e. 112102 and 112003 are configured in the number group in that hunt group. Another phone i.e. 112004 dials 105 (extension dialing) and the call is answered by 112003. After answering the call, the number displayed in the caller phone is To 003 (112 105) which should be To 005 (112 105). The proper number is getting displayed in case of SLC+EXT dialing and E164 dialing.
- Three-way conferencing in HUCS: If HUCS7.1a is deployed without Hardware conference bridge and Transcoder configured, the Cisco Unified CM cluster(s) must be configured with following for the conference to work using Cisco Unified CM software conference bridges as part of the static config
  - Check the "Regions" of s/w conference bridge and H225 GK controlled trunk to HSI on Cisco Unified CM cluster(s).
  - If there is no association between these two regions, configure the inter-region relationship codec to be G711. This is because, intra-region codec is G711 and inter-region codec is G729 by default on Cisco Unified CM. Hence, configuring G711 to be inter-region codec for these two regions enables the conference call to use G711 and makes it work.

- DDI for Redirect" feature implementation has been moved from the Customer/Location level to the Provider in the current release of USM. This impacts the ability to customize this feature at Location level.
- USM does not manage the barge feature. For the feature to be activated, the 'Built in bridge' and the 'Privacy' option on the phone management page in the Cisco Unified CM have to be changed to 'on' and 'off' respectively from 'Default'. USM sets these options to 'Default' by default. This has to be done for all the phones with which the barge feature is tested.
- Callback feature is not supported in the HUCS7.1(a) release. Callback works only for the extension number dialing within the location.
- The Do Not Disturb (DND) feature support is limited on HUCS7.1(a). The DND feature can be activated/deactivated on the phone using the softkey if it is available as part of the Softkey template configured for the phone. But the phone specific configurations for DND such as whether call should be rejected or sent to Voicemail or whether it should ring or only flash are not available through USM. They have to be configured manually on the Cisco Unified CM if required.
- It is recommended that the published number is always provisioned for any location that is created. This is to make sure that the relevant PSTN number is sent out for the DN which does not have the DDI associated with it.

Limitation and Observations in Movius AutoAttendant Support

This section describes the limitations and observations affecting Movius AutoAttendant Support in Cisco Hosted Unified Communications Services, Release 7.1(a):

- USM cannot create the MWI On and MWI Off devices in Cisco Unified CM due to AXL API limitations. At present this has to be performed manually. Also, for the MWI to work, the Cisco Unified CM service parameter "Multiple Tenant MWI Modes" needs to be set to True.
- Movius AutoAttendant provisioning is primarily performed manually on the Movius side. PGW is provisioned by USM and AutoAttendant is enabled in the Voicemail organization by USM, but the rest of the configuration on Movius (creating the AutoAttendant on the organization, adding the Keys for the AA Pilot and configuring the AA menus) must be performed manually by the system integrator due to a restriction on Movius, which lacks support for these configurations via XML.
- Movius AutoAttendant is configured to monitor all the calls, that is, call flow requires two media ports throughout the duration of the call. Some call flows may fail, for example, forwarding calls transferred by the AutoAttendant. For this reason an MTP is required on the voicemail trunk on Unified CM.
- Movius AutoAttendant feature can transfer calls by AA to Hosted UCS and PSTN phones using the central gateway breakout. However, presently, calls cannot be routed out through a local gateway.
- For the Movius AutoAttendant to transfer a call, the administrator needs to configure the inter-site prefix if the AA is calling using site code + extension of a phone from the same customer or the PSTN breakout code is using a full E.164 number.
- In order to create an AutoAttendant pilot for a location, at least one location should have a Voicemail service configured.
- To be able to use the same DN across customer for creating the Voice Mailbox on the Movius, the Phone Number Type should be made Private on the Movius.
- If movius AA is provisioned to dial E.164 then the calling number displayed in the called phone will be the AA pilot number. If the AA is provisioned to dial a DN, then the calling number displayed in the called phone will be the A number which is received by AA i.e. original calling phone.

- Multiple E.164 mapping to the Internal Voicemail Pilot number is supported in this release. If the location specific E.164 number needs to be mapped to Voicemail Pilot number. This can be done by moving one of the external numbers from the location to be moved to Voicemail services location and then it can be mapped under the PSTN number Mgmt.
- For SBC integration with Movius AA, SIP 100rel support have to be enabled on the PGW and for the successful call transfer to the PSTN phone over SS7 network, Support for183 had to be disabled for the SIP profile associated to SIP trunks configured for the Movius.
- It is important to note that only the outcall for AA is send via SBC, the incoming call to the AA is send via direct trunk from PGW to Movius.

Limitations and Observations for Mobile Connect - Single Number Reach

The following section describes some of the limitation and observations affecting the support of Mobile Connect - Single Number Reach feature:

- Mobility key used to send the call to remote destination for the progress call, is supported only for the EM user currently. This limitation is due to provisioning restriction on the USM. This is not a Cisco Unified CM limitation.
- When the SNR call picked up by the desk phone tries to send the call to the Mobile phone by pressing the mobility key, the call fails. This issues has been observed on 7.1.5.11901-1 release, but the fix for the same is available on the 7.1.5 ES release 7.1.5.12009-1

Limitations and Observations in Cisco Emergency Responder Integration

This section describes the limitations and observations affecting Cisco Emergency Responder Integration in Cisco Hosted Unified Communications Services, Release 7.1(a):

- There is only one instance of a Default ERL within Cisco ER which must be used by all Cisco Unified CM clusters regardless of the end customer for each site.
- To support PSAP Callback for different customers using the same CER group within the Cisco Unified CM cluster a new partition called "CustPsapCallback<custid>" is created for every customer using the CER group. This Partition is assigned to the 913XXXXXXXXXX DN associated to the CTI route point. Also the CTI route RELIN913 must be configured to use the CSS "EmergencyCust<custid>", to route the call to the phone which originated the 911 call. This configuration is manual and is not done by the USM
- Cisco ER server is manually configured. There is no interaction between USM and the Cisco ER server. In future releases of Cisco ER the API may be opened up and it may be possible for USM to apply configuration direct on the Cisco ER server. The Cisco ER settings entered into USM should correspond with the settings in Cisco ER.
- For the integration of Cisco ER into Hosted UCS, it is necessary to disable the modification of the Calling Party Number via Cisco ER. Instead, the modification of the CgPN is done on the Route Point on Cisco Unified CM.
- USM cannot add additional partitions to an existing CSS. Because of this, the administrator needs to manually add the created Cisco ER partition (for example EUSA), to the IncomingToCluster CSS.
- It is recommended that every ERL/ELIN is associated with the Onsite Security Personnel DN. This number typically should be one of the internal numbers of the customer. But for provisioning it on the CER, this number has to be the FINT of the internal number of the Onsite Security Personnel's DN.

- If a user makes an emergency call and Cisco ER is not available or the phone is unallocated in Cisco ER, the call will be sent out of the PGW Central gateway trunk towards the PSAP operator servicing the Default ERL (CdPN: 911, CgPN: ELIN for the Default ERL). Because there is only one configurable number for the Default ERL, this number will have to be sent out as the Calling Party Number for all Hosted UCS customers. If the PSAP operator tries to call back, the call will be routed to a SP representative servicing the Default ERL.
- If a user makes an emergency call and the PSAP operator answers the call and after that the PSAP operator calls back, but then Cisco ER (the PSAP callback CTI Route Point) is unavailable, the call will be routed to the SP representative servicing the Default Onsite security, which will be the ELIN for the Default ERL (CdPN will be the SP representative E.164 number (ELIN for the Default ERL), and the CdPN will be the PSAP operator number).
- The SP representative phone should be placed in a separate customer/location. This customer / location needs to be created as per standard Hosted UCS location. The DID of the service provider Representative's Phone DN should be used as the ELIN for the default ERL.

Limitations and Observations for the Emergency Call Support (Non-CER)

- Non CER Emergency calls for the UK need to be dialed with PSTN dialing prefix. Direct dialing is not supported in the base model. The customization for the direct dialing of the emergency numbers 999 and 112 needs to be done by the SI if required.
- Non CER emergency calls for the PBX location can be configured with only one Emergency Number using USM, if the country has multiple Emergency numbers, it needs to be provisioned manually.

Limitations and Observations in Local PSTN Breakout Support

This section describes the limitations and observations affecting Local PSTN Breakout support in Cisco Hosted UCS Release 7.1(a):

- Local PSTN breakout in this release only supports H.323 integration of the gateways with the Cisco Unified CM.
- Two scripts hucsntpstn.tcl and hucsntvoip.tcl are needed for the Local gateway to provide the correct features of the Local PSTN breakout. These scripts are loaded onto the gateway and configured as services and invoked from dialpeer to manipulate the incoming/outgoing numbers.
- It is recommended that the external number mapping is done post the activation of the gateway on the location. This ensures that the E.164 to Internal number mapping is pushed correctly onto the Gateway.
- AutoAttendant service is not supported for LBO enabled location in this release of Hosted UCS.
- The forced onnet call for the LBO location is supported only within the location. LBO location cannot make a forced on-net call to the location enabled for Central Breakout. The forced on-net call from the Central Breakout enabled location to the LBO enabled location is supported.
- The E.164 number mapping to the pilot number needs to be done under **Location > Advanced Mgmt > Voicemail Mgmt** for the Voicemail access from the PSTN phone via the Local Gateway.

Limitations and Observations for Extension Dialling between two locations aka Linked Location

The following section describes the Limitations and observations affecting the support extension dialing between two locations:

- The linked locations should be configured on the same Cisco Unified CM cluster, since the partitions are shared by the linked location child and parent.
- The same shared line number cannot be configured across linked locations. The configured shared lines in parent location are not visible in child location and the vice-versa
- Following observations were made during Call park feature testing on the linked location:
  - If the child location doesn't have any call park extension configured and configured only in parent location. Then the parked call can not be retrieved from child location.
  - If the parent location doesn't have any call park extension configured and configured only in child location. Then the parked call can not be retrieved from parent location.
  - If both the locations i.e. child and parent have call park extensions configured then the parked call can be retrieved from across locations.
  - If both the locations have call park extensions configured say child location: 060-065, parent: 025-030, then the extension to park the call at child location is taken from parent extension pool i.e. 025. If all the extensions in parent location are used, then extension is taken from child extension pool.
  - Similarly say child location has only one parking extension i.e. 060 and the parent have one extension i.e. 025. If the call is parked at child location then the extension used to park the call is 025. Again if the call is parked at parent location, parking extension used is 060 though it's associated with child location.

Limitations and Observations for Per-Customer "Block Off-Net to Off-Net Call Transfer" Support

The following section describes the Limitations and observations affecting the support for the Per Customer "Block offnet to offnet call transfer"

The block offnet to offnet restrict call transfers made within customer locations using the E.164 number. This is because the Cisco Unified CM feature of "Block offnet to offnet" is leveraged to implement this feature in Hosted UCS.

Limitations and Observations for Per-Customer Forced Authorization Code

The following section describes the Limitations and observations affecting the support for the Per Customer Forced Authorization Code

- The current implementation of FAC only allows the user to configure the FAC code and assign it to a location/customer. It does not allow the user to enable, disable call for the route patterns. The enhancement 6160 is filed to improve the implementation of FAC and make is user friendly. FAC was tested by enabling FAC manually on the required route pattern on the Cisco Unified CM.
- Cisco Unified CM does not support the use of FAC on the Forwarded calls. So if the FAC is enabled for the forwarded call, the call will be dropped.

Limitations and Observations for Legacy PBX integration

The following section describes the Limitations and observations affecting the support Legacy PBX Integration:

When activating legacy gateway E1 ports, the configurations pertaining to IOS gateway available in the IOS model requires both Master and Slave PGWs (active and standby) must be configured on the Hosted UCS deployment. Otherwise gateway port activation would fail with IOS command. Hence all Hosted UCS deployments must have active-standby PGW configured (Standalone PGW is not supported for Legacy PBX provisioning).

Voicemail services config using USM for the PBX phones is not supported in the current release of Hosted UCS.

Limitations and Observations for Analog Phone Support using VG224

The following section describes the Limitations and observations affecting Analog Phone Support using VG224

A Voicemail service for Analog phones is not supported due to limitation of the USM to associate the user to the line of the Analog phone.

Caveats

This section describes the open and unresolved caveats affecting Cisco Hosted UCS Release 7.1(a), This section contains the following topics:

- Using the Bug Toolkit, page 14
- Unresolved Caveats, page 15
- Resolved Caveats, page 16

Using the Bug Toolkit

You can search for problems by using the Cisco Software Bug Toolkit. To access Bug Toolkit, you need the following:

- Internet connection
- Web browser
- Cisco.com user ID and password

To use the Software Bug Toolkit, complete the following steps:

Procedure

Step 1 To access the Bug Toolkit, go to http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl.

Step 2 Log on with your Cisco.com user ID and password.

Step 3 Click the **Launch Bug Toolkit** hyperlink.

Step 4 To look for information about a specific problem, enter the bug ID number in the "Enter known bug ID" field and click **Search**.

The following tables list defects that are unresolved in Cisco Hosted UCS Release 7.1(a). For more information about an individual defect, you can access the online record for the defect by clicking the Identifier (CSC caveats). You must be a registered Cisco.com user to access this online information.

Defect status frequently changes, so these tables list the defects that were unresolved at the time of the release of Cisco Hosted UCS Release 7.1(a). To view a current list of unresolved defects, use the Bug Toolkit as described in the "Using the Bug Toolkit" section on page 14.

Unresolved Caveats

This section lists defects that are unresolved in Cisco Hosted Unified Communications Services,release 7.1(a).

Note For more information about an individual defect, you can access the online record for the defect by clicking the Identifier (CSC caveats). You must be a registered Cisco.com user to access this online information.

Note Be aware that defect status frequently changes, these tables list the defects that were unresolved at the time of the release of Cisco Hosted Unified Communications Services, Release 6.1(a). To view a current list of unresolved defects, use the Bug Toolkit as described in the "Using the Bug Toolkit" section on page 14.

Note For details and information about the current status of these defects, and for any workaround available, contact VisionOSS.

Table 10 lists the defects that are unresolved in Hosted UCS Release 7.1(a).

Table 3 Unresolved Caveats in Cisco Hosted Unified Communications Services, Release 6.1(a)

Identifier	Headline
PGW	
CSCsr67050	Code porting from 9.7 to 9.8 for CSCsq26195 and CSCsq55277.
Cisco Unified Communication Manager	
CSCtc68651	Dusting and Send Call to Mobile doesn't work with NuRD call via LRG.
USM	
6083	BVSM shows incorrect version of PGW (Transit Switch).
6085	Description field of TFTP server not updated on Manage PBX server page.

Table 3 Unresolved Caveats in Cisco Hosted Unified Communications Services, Release 6.1(a)

Identifier	Headline
6126	Unable to delete location when authorisation codes are present.
6156	External Mask on Phone's DN not updated correctly.
6160	FAC implementation.
6331	SNR Mobile Connect should be enabled even if the user is not EM logged on.
6400	IOS Model change requires change in USM gateway provisioning.
6450	HUCS71A: During creation and deletion of extensions in unmanaged locations AddLine and DelLine, respectively, should be called to populate/remove the extensions in the X10 database.
6459	HUCS71A:Add/Mod/DelQ931,Trunk-Pre/Post transactions require change.
6499	HUCS71A: USM should support provisioning of multiple non-cer emergency numbers for an unmanaged location
6507	HUCS71A: Label on shared line is not set by USM during phone registration.
6835	HUCS71A: Incorrect setting of Analog line details in Cisco Unified CM by USM during Analog line registration.
6990	HUCS71A: Local Gateway translation rule not provisioned properly.
7038	HUCS71A: Unable to delete a Service Type.
7262	HUCS71A: Delete roaming profile transaction failure - api parameter missing.
7338	HUCS71A: Unable to unregister analog gateway through USM.
7355	Unable to Convert Linked Location Child to Parent.

Resolved Caveats

This section describes the resolved caveats affecting Cisco Hosted UCS Release 7.1(a):

Identifier	Headline
General	
CSCsx26169	FINT displayed on Calling phone display.
Cisco Emergency Responder	
CSCsv63726	Emergency calls failing if Primary Cisco ER server not available.
CSCsw36017	Masking digits when Cisco ER plays announcements to Onsite Security.
Local PSTN Breakout	

Identifier	Headline
CSCsl72528	PSTN->IP->CFU->PSTN via local gateway fails.
CSCsk97744	Unified CM call from loc X to Unified CM in loc Y CF to PSTN - wrong cgpn sent.

Features not supported in this release

- Shared Building Feature
- Inter-tenant Lawful Intercept
- Unified Contact Center Hosted (UCCH) Integration