



DEFENSE INFORMATION SYSTEMS AGENCY

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IN REPLY REFER TO: Joint Interoperability Test Command (JTE)

15 Sep 14

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Joint Interoperability Certification of the Acano Solution Software Release 1.5 with the Acano Client Software Release 1.5.1.1

- References: (a) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(b) DoD CIO, Memorandum, "Interim Guidance for Interoperability of Information Technology (IT) and National Security Systems (NSS)," 27 March 2012
(c) through (f), see Enclosure 1

1. Certification Authority. References (a) and (b) establish the Joint Interoperability Test Command (JITC) as the Joint Interoperability Certification Authority for the UC products.

2. Conditions of Certification. The Acano Solution Software Release 1.5 with the Acano Client Software Release 1.5.1.1; hereinafter referred to as the System Under Test (SUT), meets the critical requirements of the Unified Capabilities Requirements (UCR), Reference (c), and is certified for joint use on the Defense Information Systems Network (DISN) as a Video Only (VO) Unified Capabilities Conference System (UCCS) with the conditions described in Table 1. The SUT was tested and certified for use with Assured Services Session Initiation Protocol (AS-SIP) interface when configured on the line side of the REDCOM High Density Exchange (HDX) and SLICE Local Session Controllers (LSCs). The SUT was tested with the REDCOM Release 4.0AR3P9; however, the SUT is certified with any version of the REDCOM HDX and SLICE LSCs currently or previously on the UC Approved Products List (APL). This certification expires upon changes that affect interoperability, but no later than three years from the date of the UC APL memorandum.

Table 1. Conditions

Table with 3 columns: Condition, Operational Impact, Remarks. Rows include UCR Waivers (None), Conditions of Fielding (SUT video client unable to set DSCP tagging), and Open Test Discrepancies (SUT does not provide an online directory).

**Table 1. Conditions (continued)**

Condition	Operational Impact	Remarks												
The SUT does not provide a registration system or web access to the conference system	Minor	See note 1.												
The SUT does not provide a scheduling system.	Minor	See note 1.												
The SUT does not support all of the required audio and video codecs.	Minor	See note 1.												
The SUT does not fully comply with the chair control functionality requirements.	Minor	See note 2.												
The SUT does not support H.239 or H.261 annex D standards.	Minor	See note 3.												
The SUT is not capable of extending a conference.	Minor	See note 3.												
The SUT does not provide H.323 gateway capacity.	Minor	See note 3.												
The SUT does not provide H.239 capability.	Minor	See note 3.												
The SUT does not support dial-up video services.	Minor	See note 3.												
The SUT is unable to manage or monitor reservation or scheduling services.	Minor	See note 3.												
The SUT partially complies with the Fault Management Systems requirement.	Minor	See note 4.												
The SUT partially complies with Fault management functions requirement.	Minor	See note 5.												
The SUT does not support accounting management for dedicated or dial-up services.	Minor	See note 3.												
<p><b>NOTES:</b></p> <ol style="list-style-type: none"> <li>1. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR</li> <li>2. DISA has accepted the vendor's POA&amp;M and has adjudicated this discrepancy as having a minor operational impact.</li> <li>3. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.</li> <li>4. DISA has adjudicated this discrepancy as having a minor operational impact.</li> <li>5. DISA adjudicated this as minor and stated the intent to move this requirement to Section 15 and broaden by removing "dedicated and dial-up video services".</li> </ol> <p><b>LEGEND:</b></p> <table> <tr> <td>DISA</td> <td>Defense Information Systems Agency</td> <td>SUT</td> <td>System Under Test</td> </tr> <tr> <td>DSCP</td> <td>Differentiated Services Code Point</td> <td>UCR</td> <td>Unified Capabilities Requirements</td> </tr> <tr> <td>POA&amp;M</td> <td>Plan of Action and Milestones</td> <td></td> <td></td> </tr> </table>			DISA	Defense Information Systems Agency	SUT	System Under Test	DSCP	Differentiated Services Code Point	UCR	Unified Capabilities Requirements	POA&M	Plan of Action and Milestones		
DISA	Defense Information Systems Agency	SUT	System Under Test											
DSCP	Differentiated Services Code Point	UCR	Unified Capabilities Requirements											
POA&M	Plan of Action and Milestones													

3. **Interoperability Status.** Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. Table 4 provides a UC APL product summary.

**Table 2. SUT Interface Status**

Interface (Protocol)	Threshold CR/FR Requirements (See note 1.)	Status	Remarks
Ethernet 10 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 100 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 1000 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 10000 Mbps (AS-SIP) (R) (See note 2.)	1	Not Tested	See note 3.
Ethernet 10 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
Ethernet 100 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
Ethernet 1000 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
T1 ISDN PRI (H.320) (C)	1	Not Tested	See note 4.
E1 ISDN PRI (H.320) (C)	1	Not Tested	See note 4.

**Table 2. SUT Interface Status (continued)**

<b>NOTES:</b>			
1. The UCR does not identify interface CR/FR applicability. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column are cross-referenced with Table 3.			
2. The UCR states that an Ethernet interface is required; however, it does not stipulate a specific rate. The SUT supports AS-SIP natively through the SUT Ethernet interfaces.			
3. Although the SUT supports this interface, it was not tested and is not covered under this certification			
4. The SUT does not support this conditional interface.			
<b>LEGEND:</b>			
AS-SIP	Assured Services Session Initiation Protocol	Mbps	Megabits per second
C	Conditional	PRI	Primary Rate Interface
CR	Capability Requirement	R	Required
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	UCR	Unified Capabilities Requirements
ISDN	Integrated Services Digital Network		

**Table 3. SUT Capability Requirements and Functional Requirements Status**

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	UC Audio and Video Conference System (R)	3.4	(See notes 2 and 3.)
<b>NOTES:</b>			
1. The annotation of ‘required’ refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.			
2. Security testing is accomplished by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).			
3. The SUT met the requirements for a Video Only (VO) Unified Capabilities Conferencing System (UCCS) with the exceptions noted in Table 1. DISA adjudicated these exceptions as minor.			
<b>LEGEND:</b>			
CR	Capability Requirement	R	Required
DISA	Defense Information Systems Agency	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements

**Table 4. UC APL Product Summary**

<b>Product Identification</b>			
Product Name	Acano Solution		
Software Release	1.5		
UC Product Type(s)	Unified Capabilities Conference System (UCCS) (Video Only)		
Product Description	The Acano Solution platform is an IP-based Multipoint Control Unit (MCU) and protocol gateway server for video collaboration.		
<b>Product Components (See note.)</b>	<b>Component Name</b>	<b>Version</b>	<b>Remarks</b>
MCU (Server)	Acano Solution	1.5	
Client Software (Laptop)	Acano Client	1.5.1.1	
Management Laptop	Dell Inspiron 3521	Microsoft Windows 7 SP1	
<b>NOTE:</b> The detailed component and subcomponent list is provided in Enclosure 3.			
<b>LEGEND:</b>			
APL	Approved Product List	SP	Service Pack
IP	Internet Protocol	UC	Unified Capabilities
MCU	Multipoint Conferencing Unit	UCCS	Unified Capabilities Conference System

4. **Test Details.** This certification is based on interoperability testing, review of the vendor’s Letters of Compliance (LoC), DISA adjudication of open test discrepancy reports (TDRs), and DISA Certifying Authority (CA) Recommendation for inclusion on the UC APL. Testing was

conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 23 June through 3 July 2014 using test procedures derived from Reference (d). Review of the vendor's LoC was completed on 5 June 2014. DISA adjudication of outstanding TDRs was completed on 22 July 2014. Information Assurance testing was conducted by DISA-led Information Assurance test teams and the results are published in a separate report, Reference (e). Enclosure 2 documents the test results and describes the tested network and system configurations. Enclosure 3 provides a detailed list of the interface, capability, and functional requirements.

**5. Additional Information.** JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at <https://stp.fhu.disa.mil/>. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at <https://jit.fhu.disa.mil/>. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly from the Unified Capabilities Certification Office (UCCO), e-mail: [disa.meade.ns.list.unified-capabilities-certification-office@mail.mil](mailto:disa.meade.ns.list.unified-capabilities-certification-office@mail.mil). All associated information is available on the DISA UCCO website located at <http://www.disa.mil/Services/Network-Services/UCCO>.

**6. Point of Contact (POC).** The JITC point of contact is Mr. Dale Fulton, commercial telephone (520) 538-0507, DSN telephone 879-0507, FAX DSN 879-4347; e-mail address [dale.h.fulton.civ@mail.mil](mailto:dale.h.fulton.civ@mail.mil); mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Dale Fulton) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1330801.

FOR THE COMMANDER:



3 Enclosures a/s

for RIC HARRISON  
Chief  
Networks/Communications and UC Portfolio

JITC Memo, JTE, Joint Interoperability Certification of the Acano Solution Software Release 1.5  
with Acano Client Software Release 1.5.1.1

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DOT&E, Netcentric Systems and Naval Warfare

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UCCO

## **ADDITIONAL REFERENCES**

- (c) Office of the Department of Defense Chief Information Officer, "Department of Defense Unified Capabilities Requirements 2013, Errata-1," July 2013
- (d) Joint Interoperability Test Command, "Unified Capabilities Conference System (UCCS) Test Procedures for Unified Capabilities Requirements (UCR) 2013," Draft
- (e) Joint Interoperability Test Command, "Information Assurance (IA) Assessment Report For Acano Solution Software Release v.1.5 (Tracking Number 1330801)," Draft

## CERTIFICATION SUMMARY

**1. SYSTEM AND REQUIREMENTS IDENTIFICATION.** The Acano Solution Software Release 1.5 with Acano Client Software Release 1.5.1.1 is hereinafter referred to as the System Under Test (SUT). Table 2-1 depicts the SUT identifying information and requirements source.

**Table 2-1. System and Requirements Identification**

<b>System Identification</b>	
Sponsor	Army
Sponsor Point of Contact	Freddie Lee Pittman, Address: 93 <sup>rd</sup> Signal Brigade, 662 Sheppard Place, Fort Eustis, Virginia 23604, e-mail: <a href="mailto:freddie.l.pittman.civ@mail.mil">freddie.l.pittman.civ@mail.mil</a>
Vendor Point of Contact	Murl Coffey, Address: Acano Federal, 13800 Coppermine Rd, Suite 214, Herndon, Virginia 20171, e-mail: <a href="mailto:murl.coffey@acano.com">murl.coffey@acano.com</a>
System Name	Acano Solution
Increment and/or Version	Acano Solution v 1.5, Acano Client v 1.5.1.1
Product Category	Video Only (VO) Unified Capabilities Conference System (UCCS)
<b>System Background</b>	
Previous certifications	None.
<b>Tracking</b>	
UCCO ID	1330801
System Tracking Program ID	<a href="#">The government AO will provide this information.</a>
<b>Requirements Source</b>	
Unified Capabilities Requirements	Unified Capabilities Requirements 2013, Errata 1
Remarks	
<b>Test Organization(s)</b>	Joint Interoperability Test Command, Fort Huachuca, Arizona
<b>LEGEND:</b>	
ID	Identification
MCU	Multipoint Control Unit
UCCO	Unified Capabilities Connection Office

**2. SYSTEM DESCRIPTION.** The Unified Capabilities Conference System (UCCS) category comprises three types of conference systems. The Audio Only (AO) conference system includes systems that support audio only end instruments. The Video Only (VO) conference system includes systems that support video only end instruments (EI) (i.e., video teleconferencing [VTC] systems). The Audio/Video (A/V) conference system includes systems that support both dedicated AO and VO. An A/V conference system can be implemented as an integrated audio and video product, or as a combination of an audio conference system and a video conference system.

The SUT is a Video Only UCCS that provides a Multipoint Conferencing Unit (MCU) capabilities using Assured Services Session Initiation Protocol (AS-SIP) natively while registering on the line side of the REDCOM High Density Exchange (HDX) and SLICE Local Session Controllers (LSCs). The SUT was tested and certified for use with AS-SIP interface when configured on the line side of the REDCOM HDX and SLICE LSCs. The SUT was tested with the REDCOM Release 4.0AR3P9; however, the SUT is certified with any version of the REDCOM HDX and SLICE LSCs currently or previously on the Unified Capabilities (UC) Approved Products List (APL).

The Acano Platform is an IP-based and protocol gateway server for video collaboration. The Acano Platform supports Session Initiation Protocol (SIP)-based communications between endpoints and hosts using the Acano coSpace collaborative interface. Endpoint instruments (EIs) and infrastructure supporting SIP and AS-SIP interface with the Acano Platform as a single unified capabilities hub for audio, video, and web-based conferencing. Using Extensible Messaging and Presence Protocol (XMPP)-based presence, the Acano Platform enables persistent chat and Instant Messaging for enterprise web clients within the coSpace interface.

**Acano Solution (MCU).** The Acano Platform is optimized for deployment of the Acano software. Built on an Intel X86 platform, the Acano Core module handles management, conferencing, media, and external interoperability. The Acano Edge module handles web services and the WebRTC client; manages XMPP communications; and provides ICE, TURN, and STUN for secure NAT traversal where applicable.

**Acano Client.** The site must provide a STIG-compliant workstation for remote web access to the Acano Platform as a client-side user. The workstation may support any DoD-compliant OS. The SUT supports Windows 7 SP1. Communication requires a DoD-compliant web browser and is provided via HTTPS and XMPP. Users deploy the Acano Client Software to provide real-time video collaboration using WebRTC and HTML5.

**Management Workstation.** The management workstation is a site-provided, Security Technical Implementation Guide (STIG)-compliant, Public Key (PK)-enabled workstation running on the Microsoft Windows 7 Professional Service Pack 1 Operating System used to manage the SUT.

**3. OPERATIONAL ARCHITECTURE.** The Unified Capabilities (UC) architecture is a two-level network hierarchy consisting of Defense Information Systems Network (DISN) backbone switches and Service/Agency installation switches. The Department of Defense (DoD) Chief Information Officer (CIO) and Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The UC architecture, therefore, consists of several categories of switches. Figure 2-1 depicts the notional operational UC architecture in which the SUT may be used and Figure 2-2 the UCCS functional model.

**4. TEST CONFIGURATION.** The test team tested the SUT at JITC, Fort Huachuca, Arizona in a manner and configuration similar to that of a notional operational environment. Testing of the system's required functions and features was conducted using the test configuration depicted in Figure 2-3. Information Assurance (IA) testing used the same configuration.

**5. METHODOLOGY.** Testing was conducted using UCCS requirements derived from the Unified Capabilities Requirements (UCR) 2013, Reference (c), and UCCS test procedures, Reference (d). Any discrepancies noted were written up in Test Discrepancy Reports (TDRs). The vendor submitted Plan of Action and Milestones (POA&M) as required. The remaining open TDRs were adjudicated by Defense Information Systems Agency (DISA) as minor. Any new discrepancy noted in the operational environment will be evaluated for impact on the

existing certification. These discrepancies will be adjudicated to the satisfaction of DISA via a vendor POA&M, which will address all new critical TDRs within 120 days of identification.

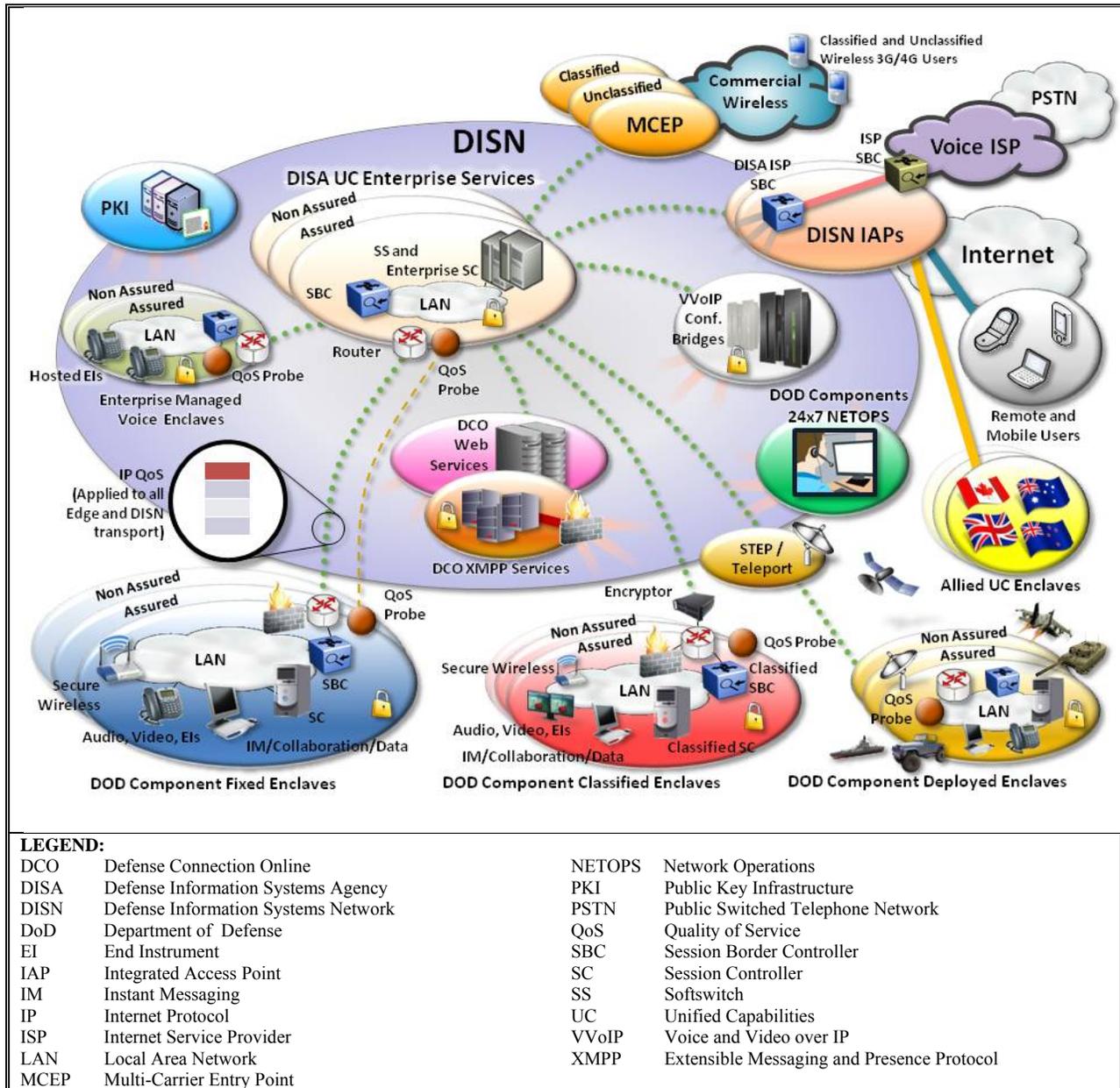
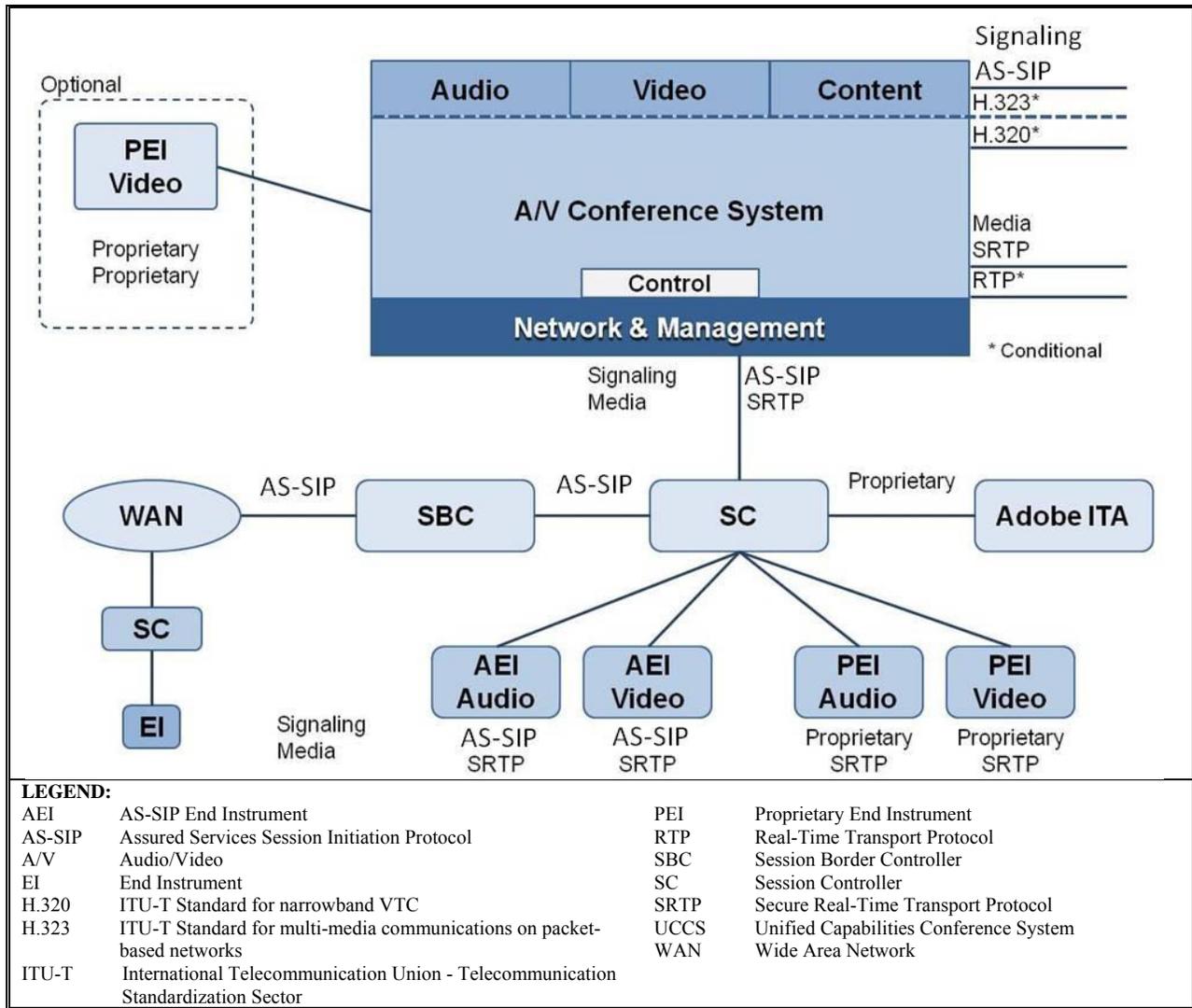
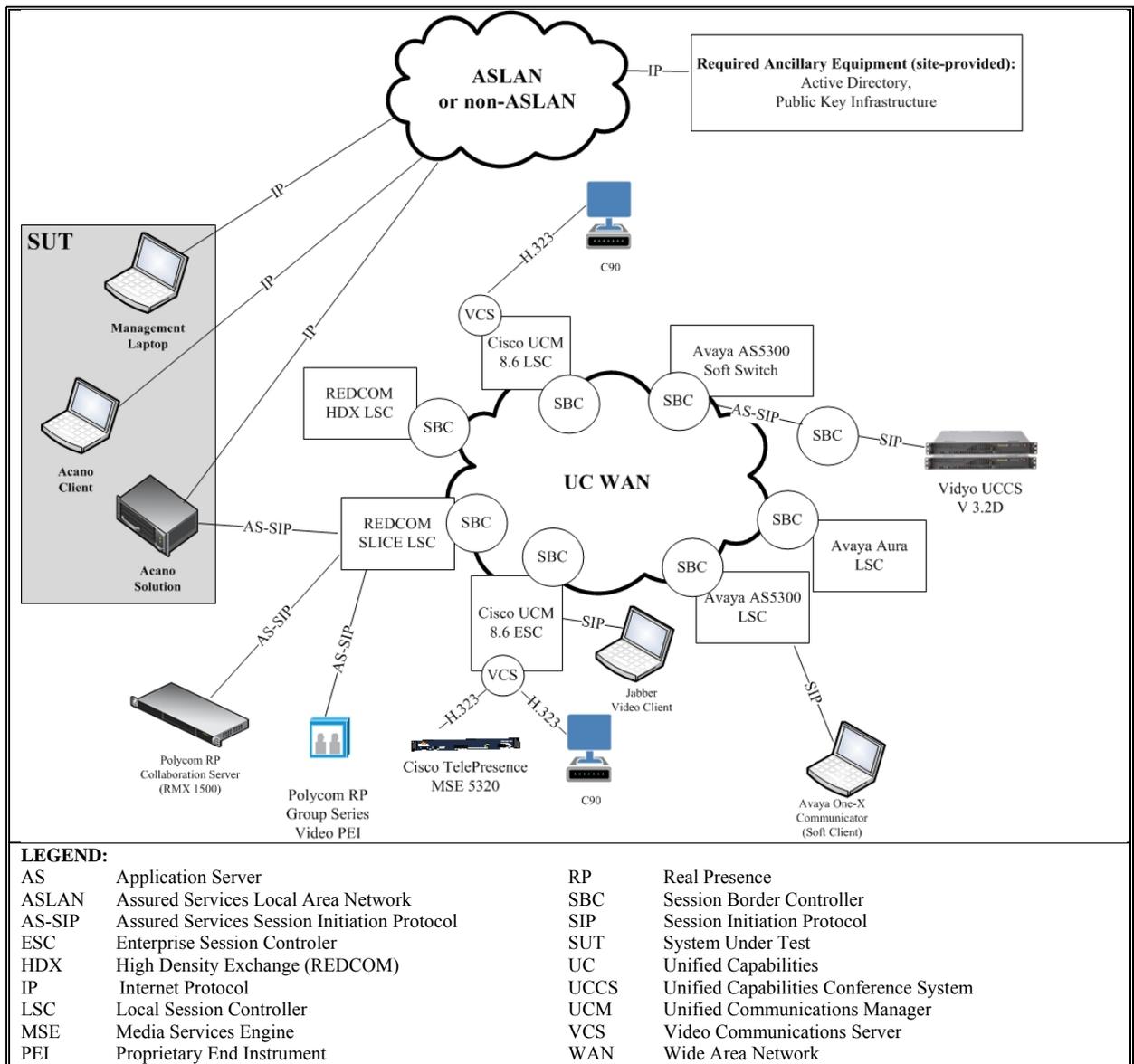


Figure 2-1. Notional UC Network Architecture



**Figure 2-2. UCCS Functional Reference Model**



**Figure 2-3. SUT Test Configuration**

**6. INTEROPERABILITY REQUIREMENTS, RESULTS, AND ANALYSIS.** The interface, Capability Requirements (CR) and Functional Requirements (FR), and other requirements for UCCS devices are established by UCR 2013, sections 3.4 and 5.

**a. Interface Status.** The status of JITC interface testing on the SUT is provided in Table 3-1. The conference system shall provide, as a minimum, an Ethernet-based interface to the network. When implemented as a standalone SUT, AS-SIP is the required interface between the conference system and the Session Controller (SC). The SUT can include a proprietary interface on the “line side” to a video EI if the video EI is from the same vendor as the SUT. The SUT met the interface requirements with testing and the vendor’s Letters of Compliance (LoC). The

SUT met all requirements for 10/100/1000 Megabits per Second (Mbps) Ethernet. All components of the SUT support 10/100/1000 Mbps. The SUT MCU supports AS-SIP natively to the LSC and proprietary XMPP to the Acano client through the SUT Ethernet interfaces.

## **b. Capability and Functional Requirements and Status**

(1) The UCR 2013, section 3.4.2 includes the System Description requirements in the subparagraphs below.

(a) System Architecture. The conference system shall provide, as a minimum, an Ethernet-based interface to the network. The SUT met this requirement. All components of the SUT support 10/100/1000 Mbps.

(b) Information Assurance. The conference system shall meet the Information Assurance requirements of all applicable DISA STIGs. Security testing is accomplished by DISA-led IA test teams and the results published in a separate report, Reference (e).

(2) The UCR 2013, section 3.4.3 includes the Service requirements in the subparagraphs below.

(a) Registration. All subscriber EIs and devices directly utilizing the conferencing system shall be required to register with the system in order to use conferencing services. The SUT does not provide a registration system. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

(b) Point-to-Point Conferencing. The point-to-point conferencing requirements are in the subparagraphs below.

1. The conference system shall provide IP-based point-to-point conferencing. Point-to-point conferencing consists of two participants with fully interactive audio and/or video capabilities. The system shall support EIs that are registered with the system to initiate point-to-point, fully interactive audio and video capability communications. This capability shall be supported through conferencing services that enable the resolution of resource conflicts by calendaring and scheduling system application programming interfaces (APIs) with enterprise scheduling systems. It is desired that the UCCS shall provide real-time conferencing status capability; e.g., busy, online, offline. The SUT met this requirement with testing.

2. The conference system shall support IP-based solutions using AS-SIP. The conference system may optionally support H.323 or dial-up Integrated Services Digital Network (ISDN) H.320 endpoint support. The SUT met the minimum requirement with native AS-SIP. The SUT met this requirement with testing.

3. The conference system shall support interactive conferences using IP. The conference system may optionally support ISDN transport. These conferences shall consist of proprietary and AS-SIP EIs. The conferences may optionally include H.323 EIs only, H.320 EIs

only, or a combination of H.323 EIs, Proprietary and AS-SIP EIs, and H.320 EIs. The SUT met the minimum requirement with native AS-SIP. The SUT met this requirement with testing.

(c) Multipoint Conferencing. The conference system shall provide multipoint conferencing. A multipoint conference consists of three or more EIs in a conference call and shall include the functions and features in the following subparagraphs.

1. The conference system shall distribute fully interactive video and/or audio streams among multiple participants according to the channel bandwidth of each participant. The SUT met this requirement with testing.

2. The conference system shall accommodate users on the same conference at different video rates, resolutions, and frame rates according to EI capability and not at the lowest common denominator level. The SUT met this requirement with testing by establishing a multipoint conference with EIs at different video and audio codecs.

3. The conference system shall provide interactive, multipoint conferences using IP transport and optionally ISDN transport. These conferences shall consist of proprietary and AS-SIP EIs. The conferences may optionally include H.323 EIs only, H.320 EIs only, or a combination of H.323 EIs, Proprietary and AS-SIP EIs, and H.320 EIs. The SUT met the minimum requirement with native AS-SIP. The SUT met this requirement with testing.

(d) In-Conference Control. The conference system shall provide in-conference control. In-conference control shall include the following functions and features.

1. The conference system shall provide a banner for each conference. The SUT does provide a banner for each conference; however, the banner consists of the Vendor's name.

2. The conference system shall provide notification of participants joining and leaving a conference, and provide an end-of-conference warning to all participants. The SUT partially meet this requirement with testing. SUT is capable of playing a tone every time a participant joins or leaves the conference. However, the SUT is not capable of providing an end-of-conference warning because the conferences do not have an end time. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

3. The conference system shall provide the ability to extend conferences, without disruption, to conferences in progress. The SUT does not meet this requirement. The SUT is not capable of extending conferences because the conferences do not have an end time. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

4. The conference system shall support presentation capability for different screen layouts locally, manageable by each individual host. The SUT met this requirement with testing.

5. The conference system shall provide the following conferencing chair control functionality: voice activated switching; broadcast mode; lecture mode; video switching with H.243 control; continuous presence; add/delete, accept/reject, connect/disconnect, mute/unmute, audio/video. The SUT partially meets this requirement with testing and the vendor's LoC. The SUT is capable of voice activating switching and continuous presence. SUT does not support, Lecture Mode, Broadcast Mode or H.243 chair control standards. Also the SUT can add / delete, connect / disconnect participants but is not capable of accept / reject, mute /unmute, audio / video. DISA adjudicated all of these discrepancies as minor and stated the intent to change these requirements to optional except the Mute/Unmute Audio/Video. The Mute/Unmute Audio/Video capability still required and DISA adjudicated that requirement as Minor with Vendor PoAM.

(e) Transcoding. The transcoding requirements are in the subparagraphs below.

1. The conference system shall provide transcoding availability regardless of the data speeds; the number of concurrent video calls; and the number of concurrent conferences, video sizes, frame rates, and conference modes (voice switching or continuous presence) without downgrading the conference to a lowest common denominator protocol. The SUT met this requirement with testing.

2. The conference system shall provide automatic video transcoding without downgrading the conference to a lowest common denominator protocol. The SUT met this requirement with testing with multipoint conferences with EIs configured with various codecs.

3. The conference system shall provide automatic audio transcoding without downgrading the conference to a lowest common denominator protocol. This requirement does not apply to the SUT, which is a VO UCCS.

(f) Variable Data Rates. The conference system shall provide support for variable data rates, the rate at which data (bits) is transmitted, usually expressed in bits per second (bps) per Conferencing Terminal Unit (CTU). The system shall support data rates of at least 64 kilobits per second (kbps) per CTU. The system shall provide speed matching and down speeding to facilitate adjustable data rates. The system shall provide bandwidth management of IP services in order to restrict the bandwidth used by active call connections to the bandwidth installed and available in the network access connections. The SUT met this requirement with testing.

(g) Audio Add-On. The conference system shall provide audio add-on features for audio-only participants in video conferences and support an external VoIP audio conference connecting to the video conference session. This requirement is not required for a VO UCCS; however, the SUT does support external audio add on calls via the SUT interfaces. Audio add-on calls were placed from the SUT interfaces to Audio only EI during an established video session. The SUT met the requirement with testing.

(h) Interactive Graphics Exchange. The UCCS shall optionally provide content sharing capability for participants to interact during VTC sessions that allow participants to view and display the same presentation material at the same time. The system shall provide a dedicated live video stream and a presentation video stream and still-frame graphics as specified in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols. The system shall provide Interactive Graphics Exchange with the following functions and features.

1. The conference system shall provide a means to allow participants to interactively view images from external sources with all or any of the participants in the conference. The SUT met this optional requirement with testing.

2. The conference system shall provide real-time participation of any combination of EIs. The system shall provide still image exchange as specified in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols. The SUT does not support this optional requirement.

3. Interactive graphics exchange capabilities shall include the following: point-to-point and multipoint conferencing services, interoperability with different vendor EIs and variable graphic resolutions, and connections among participants using any video EIs, connection types, and at any rates. The SUT met this optional requirement with testing to the different video EIs within the network. The SUT was tested with the other video EIs using variable graphics resolutions, connection types, and rates.

(i) Audio Conferencing. The conference system shall be able to support audio conferencing and provide the following functions and features.

1. The system shall be capable of accepting audio-only participants into a conference call for both scheduled and ad hoc video conferences. This requirement does not apply to the SUT, which is a VO UCCS.

2. The system shall provide an interface to an external audio conferencing system to allow cascading between a multi-point VTC call with a multi-point audio call through the use of the interface to an external audio conferencing system. The SUT met this requirement with testing and vendor's LoC.

3. The system shall provide internal conferencing capabilities for the support of audio-only participants. This requirement does not apply to the SUT, which is a VO UCCS.

(j) Integrated Services. This subsection describes services that are to be integrated in a means that provides the customer a user-friendly presentation, request, and access.

1. Web Access to Conference System. The conference system shall provide a Web-based portal for customer access to the UC conferencing services, features, and capabilities. As the conferencing services change, the Web-based portal shall reflect those changes. Additionally, the conferencing services Web portal shall provide the features in the subparagraphs below.

a. An enterprise-wide service for the identification and other pertinent information about users, conferencing services, and resources, and makes it accessible from any place at any time. The SUT does not provide web access to conference system. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

b. Awareness of relevant, accurate information about the conferencing service to users at all levels (strategic, operational, and tactical). The SUT does not provide web access to conference system. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

c. An integrated scheduling system that provides users the ability to schedule one or a combination of video and audio conference services in one Web interface. The SUT does not provide an integrated scheduling system. DISA adjudicated this as minor and stated the intent to change this requirement to optional.

2. Video Conferencing Recorded Content Retrieval and Management. The following subparagraphs describe the Video Conferencing Recorded Content Retrieval and Management services to be provided by the system. The SUT does not support these optional requirements.

a. The system shall provide the user the ability to view and listen to the recorded video conferences using a Web browser to retrieve streaming video.

b. The system shall provide a Web-based system for the meeting moderator to access the recorded video conference call.

c. The system shall inform the meeting moderator of the recorded video conference access information immediately after the completion of the video conference.

d. The system shall provide Web-based interfaces for users to search recorded content based on the combination of the following information: meeting topic, meeting date/time, keywords provided by meeting moderators, meeting leader name, meeting language, and meeting leader organization.

e. The Web interface shall provide a link to the content retrieval launch page.

f. The content retrieval launch page shall authenticate the users using PKI and prompt users to enter passwords defined by the meeting moderators during the meeting scheduling phase.

g. The content retrieval launch page shall maintain a record of every content request. The record of each request shall include the name, email address, and E.164

number or IP address of the requester; the identification of the recording; and the date and time of the request.

h. The content retrieval launch page shall allow users to choose which format of the supported streaming media formats to use when playing back the retrieved content.

i. The system shall provide the end user controls during streaming to pause/resume and select segments to play.

j. The content retrieval launch page shall allow users to download the stored content of a conference meeting, if this option is permitted by the meeting's moderator.

k. The streaming shall comply with the Streaming Service Protocol Requirements described in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols.

l. The system shall ensure the compatibility of stored content with the latest versions of media player clients.

3. Audio Conferencing Recorded Content Retrieval and Management. The following subparagraphs describe the Audio Conferencing Recorded Content Retrieval and Management services to be provided by the system. These requirements do not apply to the SUT, which is a VO UCCS.

a. The content management system shall comply with DoD 5200.1R with clear marking and labeling.

b. The content management system shall maintain recorded audio conferencing content ready for users to retrieve anytime for a period of 30 days after the completion of the conferences.

c. The content management system shall archive recorded audio conferencing content into permanent storage after 30 days.

d. The content management system shall allow users to request stored content from archive. The wait time to retrieve archived material shall be less than one working day.

e. The archive material shall be kept at the same fidelity levels of the original recordings. Compression techniques that cause a loss of fidelity are not acceptable encoding schemes for archive material.

(k) Interoperability. This subsection describes the system's interoperability requirements. The system shall maximize the use of standards-based interfaces. The system shall use functions, protocols, and formats that are publicly available. The SUT met this requirement with testing and the vendor's LoC.

1. For video equipment, the conference system shall adhere to Federal Telecommunications Recommendation 1080B-2002 (FTR-1080B). The SUT does not support this optional requirement.

2. Compression Algorithms and Audio/Video Protocols

a. The conference system shall support the following audio and video standards for video conferencing: (1) Audio Protocols (G.711, G.722, G.722.1, G.723.1, G.728, G.729/G.729A). (2) Video Protocols (H.263-2000, H.264). The SUT supports all required audio and video protocols except G.723.1 audio codec and H.263-2000 video codec. DISA adjudicated this as minor and stated the intent to change this requirement to reflect H.263 vice H.263-2000 as the required video codec and intend to make G.723.1 audio codec optional.

b. The conference system shall support the following video standards for video conferencing: H.261, H.264 (SVC). The SUT supports H.261. The SUT does not support H.264 SVC. This requirement is optional for VO UCCS.

c. The conference system shall support the following audio standards for audio-only conferencing: G.711, G.722, G.722.1, G.723.1, G.728, G.729/G.729A. This requirement does not apply to the SUT, which is a VO UCCS.

d. The conference system shall provide interoperability for all endpoint devices that support AS-SIP during call setup. The SUT meets this requirement through testing and the vendor's LoC.

e. The conference system shall provide interoperability for all endpoint devices that support H.320 during call setup. The SUT does not support this optional requirement.

f. The conference system shall provide interoperability for all end point devices that support H.323 during call setup. The SUT does not support this optional requirement.

g. The conference system, including any proprietary video EIs, shall support the following video formats: (1) Sub-Quarter Common Intermediate Format (SQCIF), (2) Quarter Common Intermediate Format (QCIF), (3) Common Intermediate Format (FCIF, also called CIF), (4) 4 Full Common Intermediate Format (4FCIF, also called 4CIF), (5) 16 Full Common Intermediate Format (16FCIF, also called 16 Common Intermediate Format (16CIF)), (6) SD and HD video resolution formats for H.261, H.263, and H.264 codecs. The SUT met these requirements with the vendor's LoC with the following minor exception: The SUT does not support (5) 16 Full Common Intermediate Format (16FCIF, also called 16 Common Intermediate Format (16CIF)), which is an optional requirement.

h. The conference system shall ensure that the freeze-frame image feature is compliant with ITU-T H.239 and with H.261 Annex D. The SUT does not meet this

requirement. The SUT does not support H.239, which is part of the H.323 standard or H.261 Annex D, which is part of the H.320 standard. The SUT only support AS-SIP and does not support either H.323 or H.320. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

i. The system's freeze-frame image size shall support 4FCIF (4CIF), Video Graphics Array (VGA), Super Video Graphics Array (SVGA), Extended Graphics Array (XGA), and Widescreen Super Extended Graphics Array Plus (WSXGA+) when using H.239. The SUT does not meet this requirement. The SUT does not support H.239, which is part of the H.323 standard. The SUT only support AS-SIP and does not support H.323. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

j. The system's freeze-frame image size shall support HD (720p) and HD (1080p) when using H.239. The SUT does not support this optional requirement.

k. The system's freeze-frame image size shall support 4FCIF (4CIF) when using H.261 Annex D. The SUT does not meet this requirement. The SUT does not support H.261 Annex D, which is part of the H.320 standard. The SUT only support AS-SIP and does not support H.320. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

3. H.320 and H.323 Protocols. The conference system that supports H.323/H.320 protocols shall meet the following:

a. ISDN/PRI, H.323 V4, chair control, serial interfaces, content sharing VTC endpoint protocol requirements. The SUT does not support this optional requirement.

b. European E1 ISDN standards. The SUT does not support this optional requirement.

c. ISDN bonding up to 1.5 Mbps on Digital Transmission Link Level 1 (T1) and 2 Mbps on European Basic Multiplex Rate (E1) per International Organization for Standardization (ISO) 13871. The SUT does not support this optional requirement.

d. H.323/H.320 V4. The SUT does not support this optional requirement.

e. Far end camera control (FECC) H.281 and H.323 Annex Q. The SUT does not support this optional requirement.

f. Resource Availability Indicator (RAI)/Resource Availability Confirmation (RAC) for load balancing. The SUT does not support this optional requirement.

g. Chair control messages per H.246, H.242/H.243. The SUT does not support this optional requirement.

h. Direct, H.225 routed, and H.225-H.245 routed modes of H.323 gatekeeper operations. The SUT does not support this optional requirement.

i. Quality of Service (QoS) support using Differentiated Services Code Point (DSCP) marking of IP packets. The SUT does not support this optional requirement.

j. Automatic downspeed to available ISDN/IP bandwidth. The SUT does not support this optional requirement.

k. Automatic rate detection to match incoming video calls. The SUT does not support this optional requirement.

l. V.35/RS-449/EIA-530 Data Terminating Equipment (DTE) and Data Circuit-Terminating Equipment (DCE) interfaces. The SUT does not support this optional requirement.

m. H.239 for additional video channels or still images. The SUT does not support this optional requirement.

4. AS-SIP. The system shall ensure that all conferencing equipment meets the protocol requirements in the following subparagraphs.

a. The AS-SIP audio and video signaling conferencing requirements are specified in AS-SIP Section 10, Audio and Video Conference Services. The SUT meets this requirement with testing and the vendor's LoC.

b. FECC H.281 and H.323 Annex Q. The SUT does not support this optional requirement.

c. Chair control messages per H.246, H.242/H.243. The SUT does not support this optional requirement.

d. QoS support using DSCP marking of IP packets. The SUT met this requirement with testing.

e. Automatic downspeed to available IP bandwidth. The SUT meets this requirement with testing.

f. Automatic rate detection to match incoming video calls. The SUT meets this requirement with testing.

g. Support T.140 for text messages. The SUT does not support this optional requirement.

h. H.261 Annex D for still images. The SUT does not support this optional requirement.

5. Video Mixing Modes. The conference system shall ensure that all video mixing modes meet the following standards:

a. Video Switching Mode. The system shall ensure that the system supports video switching according to H.243 and H.323. The design shall minimize the time to switch and the disruption of video when switching from one video source to another. The SUT met this requirement with testing and the vendor's LoC.

b. Video Mixing (Picture Composition or Continuous Presence Mode). The system shall ensure that the system supports video mixing functions according to H.243 and H.323. The system shall support enhanced continuous presence multiple video mixing to include the 7 plus 1 format. The SUT met this requirement with testing and the vendor's LoC.

6. In-Conference Chair Control. The conference system shall ensure that the system supports chair control standards as defined in H.230, H.246, H.242, H.243 and H.245, including standards supporting Broadcast and Lecture mode capabilities. The following shall be supported:

a. H.320 chair control messages and procedures as defined in H.230, H.242, H.243, and H.245. The SUT does not support this optional requirement.

b. H.323 multipoint conferencing units (MCUs) shall support chair control messages and procedures as defined in H.323 and as carried forward to H.323 from H.243. The SUT does not support this optional requirement.

c. H.323 – H.320 gateways shall follow the H.246 message translation tables related to chair control functions. The SUT does not support this optional requirement.

7. Audio Conferencing. Audio systems shall support in-dial and out-dial IAW the DISN World Wide Numbering Plan and the Public Switched Telephone Network (PSTN) North American dialing plans. This requirement does not apply to the SUT, which is a VO UCCS.

8. Audio Conferencing for Time Division Multiplexing (TDM)-specific requirements are in the subparagraphs below:

a. The audio system's PSTN interfaces shall support T1/E1 (AT&T TR62411 or Telcordia TR-NWT-000170). This requirement does not apply to the SUT, which is a VO UCCS.

b. The audio system's Channel Associated Signaling (CAS) interfaces shall support Alcatel-Lucent Class 5 Electronic Switching System (5ESS) and GENBAND

Digital Multiplex System (DMS) switches. This requirement does not apply to the SUT, which is a VO UCCS.

c. The audio system's T1 PRI interfaces shall support Non-Facility Associated Signaling (NFAS) and D-channel backup, if the audio system supports more than two PRIs. This requirement does not apply to the SUT, which is a VO UCCS.

d. The audio system's T1 interface shall support extended super frame (ESF) framing and with bipolar with eight-zero substitution (B8ZS)/ Alternate Mark Inversion (AMI) coding. This requirement does not apply to the SUT, which is a VO UCCS.

e. The audio system's PSTN signaling module shall support ISDN PRI (5ESS, DMS, and National ISDN), and the PRI flavors of foreign countries such as Germany, Japan, and Korea. This requirement does not apply to the SUT, which is a VO UCCS.

f. The audio system shall support the Dialed Number Identification Service (DNIS) feature where the original dialed numbers are presented as generic address parameters (GAPs). This requirement does not apply to the SUT, which is a VO UCCS.

g. The audio system shall support the automatic number identification (ANI) feature and use it to identify a calling party, if applicable. This requirement does not apply to the SUT, which is a VO UCCS.

9. Audio Conferencing for VoIP-specific requirements are in the subparagraphs below:

a. The audio system IP interfaces shall support static assignment of the IP address, mask, default router, and Domain Name Service (DNS) entries. This requirement does not apply to the SUT, which is a VO UCCS.

b. The audio system shall support multiple DNS entries. If the primary DNS server does not respond to a DNS request, then a secondary DNS server shall be queried. This requirement does not apply to the SUT, which is a VO UCCS.

c. The audio system shall support configurable transmission control protocol (TCP) ports for AS-SIP messaging. This requirement does not apply to the SUT, which is a VO UCCS.

d. The audio system shall be able to set the IPv4/IPv6 Precedence Field bits of the Type of Service (TOS) byte and DSCP bits for media streams and signaling streams. This requirement does not apply to the SUT, which is a VO UCCS.

e. The audio system shall support Network Time Protocol (NTP), version 3 [RFC 1305]. This requirement does not apply to the SUT, which is a VO UCCS.

f. The audio system shall support SNMPv3 [RFC 3414]. This requirement does not apply to the SUT, which is a VO UCCS.

g. The audio system shall support Secure Real-Time Transport Protocol (SRTP) and Secure Real-Time Transport Control Protocol (SRTCP) [RFC 3711]. This requirement does not apply to the SUT, which is a VO UCCS.

h. The audio system shall support SRTCP and accurately report jitter, delay, and packet loss information to the far end using Real-Time Transport Protocol (RTP) Control Protocol Extended Reports (RTCP XR) [RFC 3611]. This requirement does not apply to the SUT, which is a VO UCCS.

i. The audio system's AS-SIP signaling module shall support RFC 3261 including loose route. This requirement does not apply to the SUT, which is a VO UCCS.

j. The audio system's AS-SIP signaling module shall allow AS-SIP URLs for both incoming and outgoing calls. This includes all alphanumeric characters allowed in legal SIP URLs. This requirement does not apply to the SUT, which is a VO UCCS.

k. The audio system's AS-SIP signaling module shall support Session Description Protocol (SDP) as defined in RFC 4566. This requirement does not apply to the SUT, which is a VO UCCS.

l. The audio system's AS-SIP signaling module shall implement user "hold" feature by using a=inactive or a=sendonly, or by sending a mid-call INVITE that includes a session description that is the same as in the original request, but the "c" destination addresses for the media streams to be put on hold are set to zero:c=IN IP4 0.0.0.0. This requirement does not apply to the SUT, which is a VO UCCS.

m. The AS-SIP signaling module shall support AS-SIP Digest Authentication [RFCs 3261 and 3310]. This requirement does not apply to the SUT, which is a VO UCCS.

n. The AS-SIP signaling module shall be able to reject incoming INVITE messages when the message does not come from pre-provisioned proxies. This requirement does not apply to the SUT, which is a VO UCCS.

o. The AS-SIP signaling module shall support call transfer as specified in AS-SIP Section 9.6, Call Transfer. This requirement does not apply to the SUT, which is a VO UCCS.

p. Audio systems shall support electronic numbering (ENUM) service registration for SIP (AS-SIP) Addresses-of-Record [RFC 3764]. This requirement does not apply to the SUT, which is a VO UCCS.

10. Audio Conferencing for Voice Medium-specific requirements are in the subparagraphs below:

a. The audio system shall support Dual Tone Multi-Frequency (DTMF) Generation/Recognition per Telcordia GR-181-CORE. This requirement does not apply to the SUT, which is a VO UCCS.

b. The audio system shall support G.711  $\mu$ A law [pulse code modulation (PCM)] and G.729. This requirement does not apply to the SUT, which is a VO UCCS.

c. The audio system's total media processing time shall be less than 50 ms including delays from jitter buffer, transcoding, mixing, packetization, and algorithm look ahead. This requirement does not apply to the SUT, which is a VO UCCS.

d. The audio system shall support G.168 compliance echo canceller (EC) with 128 ms echo path. This requirement does not apply to the SUT, which is a VO UCCS.

e. The audio system shall support Audio and Video Transport (AVT) payload type 0 and 8. [G.711 a/mu law]. This requirement does not apply to the SUT, which is a VO UCCS.

f. The audio system shall support AVT payload 18 [G.729]. This requirement does not apply to the SUT, which is a VO UCCS.

g. The audio system shall be able to accept in-band DTMF tones. This requirement does not apply to the SUT, which is a VO UCCS.

h. The audio system shall be able to send DTMF specified by RFC 4733. This requirement does not apply to the SUT, which is a VO UCCS.

i. The audio system shall be able to conceal 1 percent of packet loss without appreciable quality degradation. This requirement does not apply to the SUT, which is a VO UCCS.

j. The audio system shall be able to tolerate 40 ms of jitter for audio without appreciable quality degradation. This requirement does not apply to the SUT, which is a VO UCCS.

k. The audio system shall implement adaptive jitter buffers instead of static fix jitter buffers. This requirement does not apply to the SUT, which is a VO UCCS.

l. All hardware shall meet Network Equipment Building System-3 (NEBS-3) requirements. This requirement does not apply to the SUT, which is a VO UCCS.

11. Reduced Maximum Transmission Unit IP Environment. This subsection addresses the Maximum Transmission Unit (MTU) requirements for an IP network environment.

An MTU is the maximum size of an IP packet that will be accepted for transmission without fragmenting it into a smaller datagram. The MTU size shall be configurable to optimize video traffic. As a result of devices such as encryption units, the typical MTU size shall be changed to minimize the effect of fragmentation because of the additional overhead of the encryption. The SUT met this requirement with testing and the vendor's LoC.

a. The conference system shall ensure that all conferencing services provide signaling and media streams, and are capable of configuring the MTU. The SUT met this requirement with testing and the vendor's LoC.

b. The conference system shall ensure that all supporting services to video and audio services, including, but not limited to, reservation, monitoring, billing, administration, operator interface, and meeting control, are capable of working in the configurable MTU IP environment. The SUT met this requirement with testing and the vendor's LoC with the following minor exception. The SUT does not support reservation and scheduling. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement to optional in the next version of the UCR.

12. IPv6 Support. The conference system shall comply with the IPv6 requirements contained in Section 5, IPv6. The SUT met this requirement with testing and the vendor's LoC.

13. AS-SIP Support. The conference system shall comply with the AS-SIP requirements contained in AS-SIP 2013, Section 4, SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs. The SUT meets this requirement with testing and the vendor's LoC.

14. Assured Delivery. This subsection describes the set of capabilities, which ensures that mission-critical calls are set up and remain connected.

a. Quality of Service. The system or network device shall be able to set DSCPs on both signaling packets and media streams for both IPv4 and IPv6 as specified in Section 6.2.2, Differentiated Services Code Point. The SUT met this requirement with testing and vendor's LoC with the following minor exceptions. The SUT client who is host in Windows 7 laptop tags all DSCP packets at best effort while login as an user as required by IA. This is known Windows limitation that can be corrected by implementing Group Policies using Active Directory. DISA adjudicated this as Closed with a COF that the SUT client is deploy in accordance with the vendor DoD deployment guide and in accordance with the Windows Group Policies in Active Directory.

b. Congestion Response. The system shall provide measures to monitor bandwidth resource usage and to activate congestion management as needed within a timely fashion. The SUT met this requirement with the vendor's LoC.

c. Accounting. The system shall maintain a call summary of conference sessions. This will include the conference attendee identification, access methods, IP address,

E.164 numbers, time and date of the call, call duration, and total number of participants. The summaries shall be maintained for 30 days or IAW Information Assurance security requirements. The SUT met this requirement with the vendor's LoC.

d. Multilevel Precedence and Preemption. The system shall operate IAW the MLPP rules and procedures specified in Section 2.26.2, Multilevel Precedence and Preemption (inclusive as applies). The SUT does not support this optional requirement.

e. Multilevel Precedence and Preemption - Preset Conferencing. Each conferee shall be dialed at its designated precedence level. Each conferee may have a different precedence level. The conference host must be dialed at the highest precedence level of any conferee. The SUT does not support this optional requirement.

f. Multilevel Precedence and Preemption - Meet-Me Conferencing. When a priority session requests connection to a conference that is at conference maximum, then one of the lowest precedence conferees shall be preempted, selected through any deterministic method. (Conference maximum is the maximum number of conferees authorized for the same conference.) When a priority session requests connection to a conference that is not at conference maximum, but the system is at system maximum, then one of the lowest precedence conferees on another conference will be preempted, selected through any deterministic method. Note that the selection method shall not consider any ad hoc conferees for preemption, if an allocation of system resources dedicated to ad hoc conferences has been configured per requirement AUX-003880 in Section 3.4.4.2.1, Video Conference Capacity. (System maximum occurs when the maximum number of ports or resources are provisioned on the system.) Preempted conferees shall receive a preemption notification tone and be preempted. All remaining conferees on the system shall receive a conference disconnect tone (see Table 2.9-2, UC Information Signals). The SUT does not support this optional requirement.

g. Multilevel Precedence and Preemption - Ad hoc Conferencing. When either party of a two-party session brings in a third party, an ad hoc conference is created at the highest precedence level of the dialed conferees. Thereafter, any party of an ad hoc conference can attempt to add an additional conferee at any time. If that party is dialed at a precedence level higher than any of the current conferees, then the conference precedence level shall be elevated to a higher precedence level. If an allocation of system resources dedicated to ad hoc conferences has been configured per requirement AUX-003880 in Section 3.4.4.2.1, and there are no ad hoc system resources available to add on a conferee, then a conferee from the lowest precedence ad hoc conference will be preempted so that the conferee can be brought into the higher precedence ad hoc conference. If an allocation of system resources dedicated to ad hoc conferences has not been configured, but the system is at system maximum, then one of the lowest precedence conferees on another conference will be preempted, selected through any deterministic method, so that the conferee can be brought into the higher precedence ad hoc conference. The SUT does not support this optional requirement.

h. Multilevel Precedence and Preemption - Ad hoc Conferencing. When a higher precedence session (i.e., higher than the conference precedence level) is placed to any of the conferees, that conferee receives a preemption notification tone (see Table 2.9-2, UC

Information Signals). The other remaining conferees shall receive a conference disconnect tone, as described in Table 2.9-2. This tone indicates to the other parties that one of the conference call participants is being preempted. The SUT does not support this optional requirement.

(3) The UCR 2013, section 3.4.4 includes the requirements in the subparagraphs below.

(a) Service Performance. This section provides the service performance criteria and metrics for the system. The service performance criteria shall apply during simultaneous operation of the respective EIs. The system shall design and implement redundancy, failover, and fault tolerance at the component, subsystem, and system levels to support achieving service availability requirements in the presence of failures at the component, subsystem, and system levels. The system shall meet the requirements specified in Section 2.8, Product Physical, Quality, and Environmental Factors. The SUT meets this requirement with the vendor's LoC.

(b) Video Conference Quality

1. The conference system shall ensure that all video equipment used in designs can operate in the presence of minimal packet loss without degrading video quality below acceptable levels. The SUT meets this requirement with the vendor's LoC.

2. The conference system shall ensure that all video equipment used in the design provides adequate jitter buffer sizing to ensure an optimal end-to-end (E2E) video conferencing performance. The SUT meets this requirement with the vendor's LoC.

(c) Audio Conference Quality

1. The conference system shall ensure that the Mean Opinion Score (MOS) on the voice path meets the MOS requirements in Section 6, Network Infrastructure End-to-End Performance. This requirement does not apply to the SUT, which is a VO UCCS.

2. The conference system shall ensure that the implemented design possesses the adequate performance capacity and resources to support conference access processing requirements identified in Section 3.4.4.2, Capacity. This requirement does not apply to the SUT, which is a VO UCCS.

3. The conference system shall ensure the time needed to compile polling statistics to adequately support the audio conference capability. This requirement does not apply to the SUT, which is a VO UCCS.

(d) Capacity. The conference system vendor shall provide documentation stating the conference system's capacity and scalability. Statements regarding capacity and scalability will be validated at DISA's discretion. The SUT meets this requirement with the vendor's LoC.

(e) Video Conference Capacity.

1. The number of concurrent conferences supported shall be limited only by available ports and the number of licenses acquired, and shall not depend on the access methods, features, or number of participants in each conference. The SUT meets this requirement with the vendor's LoC.

2. Video Conference Capacity - The system shall have the capacity to support at least 1,000 concurrent 384 kbps video calls. The SUT meets this optional requirement with the vendor's LoC.

3. The system shall provide sufficient speed matching capacity to support that capability regardless of the access methods, algorithms, speeds, or the feature sets being used. The SUT meets this requirement with the vendor's LoC.

4. The system shall provide enough transcoding capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used. The SUT meets this requirement with the vendor's LoC.

5. The system shall provide sufficient H.323 gateway capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used. The SUT does not support this requirement. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

6. The system shall provide enough H.239 capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used. The SUT does not support this requirement. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

7. The system shall provide enough H.261 Annex D capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used. The SUT does not support this optional requirement.

8. The system shall support a minimum of 200 EIs for each multipoint conference. The SUT meet this optional requirement with vendor's LoC.

9. The system shall provide at least four audio added-on ports for each conference without the use of external audio systems or by cascading conference systems. This is not required for a VO UCCS. However, the SUT met this requirement with testing.

10. The system shall ensure that audio added-on does not compromise support for other capacity requirements. The SUT met this requirement with testing and the vendor's LoC.

11. The system shall support the ability to configure allocation (0 to 100 percent, default 20 percent) of system ports/resources to be dedicated to ad hoc video

conferences. Meet-me conferences shall not use resources allocated to ad hoc conferences, and vice versa. The SUT does not support this optional requirement.

(f) Audio Conference Capacity

1. The system shall be capable of scaling up to 2,500 concurrent audio calls. This requirement does not apply to the SUT, which is a VO UCCS.

2. The system shall support up to 200 participants in a single conference session. This requirement does not apply to the SUT, which is a VO UCCS.

3. The system shall be capable of scaling up to 500 concurrent conferences. This requirement does not apply to the SUT, which is a VO UCCS.

4. The system shall be capable of scaling up to 500 or more conference control web sessions. This requirement does not apply to the SUT, which is a VO UCCS.

5. The system shall be capable of scaling to support 10,000 or more reservations. This requirement does not apply to the SUT, which is a VO UCCS.

6. The audio conference system shall support more than 50 concurrent recordings. This requirement does not apply to the SUT, which is a VO UCCS.

7. The system shall support the ability to configure an allocation (0 to 100 percent; default 20 percent) of system ports and resources to be dedicated to ad hoc audio conferences. Meet-me conferences shall not use resources allocated to ad hoc conferences, and vice versa. This requirement does not apply to the SUT, which is a VO UCCS.

(g) Registration, Admission, Status, and Routing Function. The conference system shall have the capability to provide bandwidth management, EI registrations, admissions, status, and routing functions. Furthermore, the system shall be capable of scaling to support at a minimum of 1,000 concurrent conference calls and 10,000 concurrent registrations of EIs. The SUT does not support this optional requirement.

(h) Scalability. Scaling can be provided by deploying multiple systems and provisioning or load sharing between systems. The conference system should support a nominal growth of services without requiring major overhaul or major replacement of equipment.

1. The system architecture supporting the dedicated IP-based video services capability shall be able to scale to accommodate increased growth in dedicated IP-based video services EIs. The SUT meets this requirement with the vendor's LoC.

2. The system architecture supporting the dial-up video services capability shall be designed to support up to a 50 percent increase in dial-up video services. The SUT does not support this requirement. DISA has adjudicated this discrepancy as having a minor

operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

3. The system shall be able to scale to accommodate increased growth to support increases in connections to ISDN networks. Additional capacity in regards to this item shall be used only to support ISDN dial-up video traffic. The SUT does not support this optional requirement.

4. Audio add-on and audio conference service shall be able to scale to accommodate increased growth in call volume and EIs. This requirement does not apply to the SUT, which is a VO UCCS.

(4) The UCR 2013, section 3.4.5 includes the requirements in the subparagraphs below.

(a) Service Management. This section describes service management. The conference system shall provide reservation, scheduling, and registration services. The conferencing system service applications shall integrate or provide interfaces with Government network services and management applications. The conference system shall provide management functions to ensure continuous operations and accessibility of services with a data feed into the Government network services systems. The equipment and software applications shall be configurable to allow alarm and log file transmissions to be selective with the activation of a specific feature set. The SUT partially met this requirement vendor's LoC. The SUT does not support reservation, scheduling, and registration services. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

(b) General System Management

1. The conference system shall provide services management and monitoring for the Operation, Administration, Maintenance, and Provisioning (OAM&P) of conferencing services. The SUT met this requirement with the vendor's LoC.

2. The conference system shall provide a service monitor and management system to actively monitor elements and critical components within the system. The SUT met this requirement with the vendor's LoC.

3. Report data shall be in a form that is capable of being managed by the Government network services applications and network elements, which are based on commercial and industry standards. The data transmitted shall comply with industry standard management protocols and/or data formats. Such industry standard protocols for data exchange include, but are not limited to, Syslog, Common Object Request Broker Architecture (CORBA), SNMPv3, Transaction Language 1 (TL1), Java 2 Platform Enterprise Edition (J2EE), and Extensible Markup Language (XML). The SUT met this requirement with the vendor's LoC.

4. The conference system shall be responsible for managing and monitoring the following services and related resources. Furthermore, the system shall provide real-time,

read-write continuous Network Management capabilities. The level of monitoring shall be sufficient to be able to track the status, through standard interfaces and protocols, of individual discrete hardware and software components used to deliver the service to enable visibility of individual incidents affecting the service delivery:

a. Equipment and associated services. The SUT met this requirement with vendor's LoC.

b. Point-to-point and multipoint video services. The SUT met this requirement with vendor's LoC.

c. Audio services. The SUT met this requirement with vendor's LoC.

d. Reservation and scheduling. The SUT does not support this requirement. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

e. Gateways and interfaces. The SUT met this requirement with vendor's LoC.

f. Conferencing center Web site. The SUT does not support this requirement. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

g. Support systems. The SUT met this requirement with the vendor's LoC.

5. The conference system shall furnish and maintain a service monitor and management system with external interfaces or feeds into Government management application and monitoring systems. These interfaces shall provide the Government the capability to monitor the performance and status of video and audio services. These interfaces shall provide for the importing and exporting of video and audio management services and monitoring information. The Government shall have real-time access to all video and audio services management and monitoring data collected and stored by the system. These interfaces are further specified in Section 2.19.1, General Management. The SUT met this requirement with the vendor's LoC.

6. The conference system shall support the use of Simple Network Management Protocol version 3 (SNMPv3) for system management and monitoring. The SUT met this requirement with the vendor's LoC.

7. The conference system shall support a 10/100-Mbps Ethernet physical interface to the DISA VVoIP Element Management System (EMS). The interface shall work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. The SUT met this requirement with the vendor's LoC.

8. If the system will be deployed as a DISN asset, then local management traffic and VVoIP EMS management traffic shall use separate physical Ethernet interfaces. Redundant VVoIP EMS physical Ethernet interfaces may be used but are not required. Redundant local management physical Ethernet interfaces may be used but are not required. The SUT does not support this conditional requirement. SUT will not be deployed as a DISN asset.

(c) Fault Management

1. The conference system shall provide fault management for services and resources. The system shall provide the Government with all the fault information needed electronically to effectively manage all conferencing services. The SUT met this requirement with the vendor's LoC.

2. The fault management system shall include the following minimum requirements: TDR-19

a. Power status shall be provided for individual shelf, rack, or controller units. The SUT does not meet this requirement. DISA has adjudicated this discrepancy as having a minor operational impact.

b. Functional/Fault/Online/Offline status. The SUT met this requirement with the vendor's LoC.

c. Input/loss of input signal, or signal below working threshold. The SUT does not meet this requirement. DISA has adjudicated this discrepancy as having a minor operational impact.

d. Output/loss of output signal. The SUT does not meet this requirement. DISA has adjudicated this discrepancy as having a minor operational impact.

e. Input/output signal outside of specified range. The SUT does not meet this requirement. DISA has adjudicated this discrepancy as having a minor operational impact.

f. Intrusion detected. The SUT met this requirement with the vendor's LoC.

3. The fault management function shall perform the following:

a. Detect and identify faults. The fault management service provided shall monitor dedicated and dial-up video services resources, audio conferencing service status, support services status, and conduct alarm surveillance, maintain error logs, and analyze monitored or logged errors or events to anticipate faults. Faults shall be detected within 10 seconds of their occurrence. Faults shall be identified within 10 seconds of being detected. Faults shall be correlated within 10 seconds of identification. The Government shall be notified of service affecting faults within 10 seconds of fault correlation. The SUT partially complies

with this requirement with the vendor's LoC. The SUT does not support dedicated and / or dial up video services. DISA adjudicated this as minor and stated the intent to move this requirement to Section 15 and broaden by removing "dedicated and dial-up video services".

b. Isolate faults to include correlation of alarms. The fault management service provided shall initiate diagnostic testing and evaluate diagnostic results to determine the nature, severity, and specific cause(s) of the fault and isolate the fault to the video, audio, and support services at a component level. The SUT met this requirement with the vendor's LoC.

c. Temporary corrective action when a fault occurs. The fault management service provided shall reroute a conference call or service request to other hubs, circuits, or equipment in the case of a fault, including through the use of redundancy and failover capabilities. The SUT met this requirement with the vendor's LoC.

d. Correct faults. The fault management service provided shall implement corrective actions on faults to restore services to proper working order and complete resource/service restoration when the fault is with equipment or services provided by the conference system. This process shall incorporate backup and recovery capabilities to restore configurations and services to operational service. The SUT met this requirement with the vendor's LoC.

(d) Fault Management Information. The SUT complies with fault management information requirements in the subparagraphs below. The SUT met these requirements with vendor's LoC.

1. The conference system shall provide the Government with real-time monitoring of service-affecting events related to hardware and software components and subcomponents that compose the conference system. These service-affecting events impacting the scheduling and operation of system services include, but are not limited to, the following:

a. Outages for all conferencing services elements to be exported into the existing trouble tracking system.

b. Any hazardous condition, as specified in DISA Circular 310-55-1, that may cause loss of service.

2. The conference system shall be able to update all service management thresholds, as required.

3. The conference system shall maintain historical records of all fault alarm data and be able to export this data into the Government management application systems.

(e) Performance Management

1. The conference system shall provide a performance management system. The performance management system shall monitor and control all service performance and the

quality of the services and features supporting the conference system. Performance management shall perform the following functions. The SUT does not support this optional requirement.

2. Monitor, analyze, and characterize performance. The conference system shall monitor, analyze, and gather performance-related data to detect and characterize normal and degraded performance and be able to trend this data over time for metrics purposes. Section 3.4.4, Service Performance, defines normal performance requirements. The system shall provide notification if the service resources are being stressed with excess traffic loads. The SUT does not support this optional requirement.

3. Tune and control performance in areas of control: the conference system shall activate controls to tune all services performance to restore degraded resources/services to acceptable performance levels. If control actions will cause any user service disturbance, then these actions shall be approved by the Government before execution. The SUT does not support this optional requirement.

4. Maintain all services supporting the conference system through an operations database. The conference system shall maintain a database or be exportable to a Government network management tool supporting all conferencing services operational information, both real-time and historical, including, for example, traffic characterization data, performance data, and information on usage of resources/services. Historical records shall be kept of all performance data for a designated period of time. The SUT does not support this optional requirement.

5. Evaluate performance of services and features. The conference system shall continuously assess and monitor the performance of all conferencing services and features, according to the performance parameters identified in Section 3.4.4, Service Performance, to ensure that the performance levels of Government services and features meet the specification requirements of Section 3.4.3, Service, and Section 3.4.4, Service Performance. The SUT does not support this optional requirement.

(f) Government Performance Management Information. The conference system shall provide notification of events, exceptions, or measures related to the performance of services' resources, and associated service-affecting conditions to Government platforms as required. Performance degradation notification shall include, at a minimum, the following:

1. The conferencing services are composed of servers, applications and network services, appliances, and network devices responsible for supporting video services globally throughout the DoD community. The conferencing service is of a time-sensitive nature and one of the services that is being offered with the convergence of IP on the backbone. The SUT does not support this optional requirement.

2. There are two parts to the network management of this service: transport monitoring and video stream monitoring. First, the underlying network elements and servers need to be included in fault management and performance management activities at the physical layers and IP layers. Second, the video service needs to have instrumentation included that

would be able to monitor the conferencing user's experience, in order to isolate problems and reveal if the video service is meeting specific service-level requirements. The SUT does not support this optional requirement.

3. The performance management toolset should be able to collect information from the video device managers. The information it should collect would include, but not be limited to, the number of participants, duration of a session, video burst measurements, and capacity measurements. The SUT does not support this optional requirement.

(g) Security Management. Security testing is accomplished by DISA-led IA test teams and the results published in a separate report, Reference (e). Additional information can be obtained from the UC Approved Products List (APL). The SUT must be fielded in accordance with the IA guidelines and site-specific requirements. Security testing is accomplished by DISA-led IA test teams and the results published in a separate report, Reference (e)

(h) Online Directory. The SUT does not comply with online directory requirements in the subparagraphs below. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

1. The system shall provide an online directory service to support scheduling that shall include general information about all registered DoD video and audio users including, but not limited to, a user's point of contact, location, supported data rates, organization name, unit capabilities, and software versions.

2. The system shall update the online directory within 24 hours of learning of a new EI receiving service, a change in an existing EI's service status, or notification by DISA of any other change with regard to an EI.

3. The system shall provide a secure Web interface that implements a public key enablement application, allowing registered users with a valid DoD Public Key Infrastructure (PKI) or External Certification Authority (ECA) certificate and Internet connectivity to access the online directory.

4. The online directory shall support more than 1 million data entries and shall support more than 500 concurrent users at the same time.

5. The online directory shall be Web-based using modern and open technologies and provide interfaces, such as XML, Simple Object Access Protocol (SOAP), Web Service Description Language (WSDL), and Universal Discovery Description Interface (UDDI), allowing external data sources from others to perform directory lookups, queries, and updates as specified by "Horizontal Fusion Standards and Specifications," 3 November 2004, and DoD Joint Technical Architecture, Version 6, Volumes I and II, 3 October 2003.

6. The online directory shall support the discovery service that provides processes for discovery of information content or services that exploit metadata descriptions of information technology resources stored in directories, registries, and catalogs (to include search engines) as specified by “Horizontal Fusion Standards and Specifications,” 3 November 2004, and “DoD Joint Technical Architecture,” Version 6, Volumes I and II, 3 October 2003. Directory services shall be designed to meet Protected Personal Information/Personally Identifiable Information protection requirements and Privacy Act requirements.

7. The system shall provide an online directory service to allow authorized registered users to search system-wide for other authorized, registered service users in the directory and/or to update data entries.

(i) Registration System. The SUT does not comply with registration system requirements in the subparagraphs below. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

1. The conference system shall include an automated registration system that current and prospective subscribers can access online using a standard Web browser.

2. All data sent and received by the automated registration system shall be encrypted using Secure Socket Layer (SSL)/TLS technology, as a minimum, with DoD PKI certificate authentication and validation.

3. The automated registration system shall collect all data necessary to comply with provisioning requirements.

4. The automated registration system shall collect all data necessary for connection approval by the program office.

5. The automated registration system shall collect all data necessary to authorize users for access to operational systems including scheduling and reservations.

6. The automated registration system shall collect all data necessary to support the operational requirements of UC conferencing services. Upon connection approval and completion of verification and interoperability tests, all data shall be available to the operational systems for the scheduling and activation of conferences.

7. The automated registration system shall be the authoritative source of conference subscribers’ data. The system shall provide the necessary tools to allow authorized users to maintain and update user and endpoint information.

8. The automated registration system shall support Privacy Act statements as required by Information Assurance policy.

(j) Scheduling System. The SUT does not comply with scheduling system requirements in the subparagraphs below. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.

1. The primary component for requesting a conference shall be by an automated scheduling system that authorized users can access online using a standard Web browser. All data shall be encrypted using SSL/TLS technology, as a minimum, with DoD PKI certificate authentication and validation.

2. The automated scheduling system shall resolve the availability of all requested participants and conferencing resources. The customer shall be able to schedule a conference immediately or schedule it for some time in the future. Immediate start requests for conferences shall be activated in less than five minutes of confirmation of available resources and participants. The system shall support scheduling conferences with the conference size larger than the number of initially invited or scheduled endpoints. A unique identification number shall be assigned to each conference event to facilitate conference control and management. E-mail notifications shall be provided to all conference participants to facilitate event coordination. E-mail content shall be editable as a configuration feature. The system shall support encrypted e-mail for notifications. Scheduled conference information also shall be available through the scheduling Web interface.

3. The original requester shall serve as the conference manager for all conferences. The system shall have the capability to assign other system users to act as surrogates for the original requester. This may include peers and/or workflow superiors. Conference management and control shall include the ability to make changes to a scheduled or active conference. Supported changes to active conferences shall include, but not be limited to, the addition or deletion of participants, early termination of a conference, and extension of the conference beyond the originally scheduled end time. Additions, deletions, and extensions of conferences shall occur without interruption to the existing conference, other than preemptive precedence calls. Additions and extensions to conferences shall be executed in less than five minutes of confirmation of available resources. Deletions and terminations of conferences shall be executed in less than 5 minutes of the request by the conference manager. The system shall support scheduling of recurring conferences.

(k) Accounting and Billing.

1. The conference system shall provide an Accounting Management function. The Accounting Management function shall do the following:

a. Provide accounting information to the Government regarding all conferencing services provided. The SUT met this requirement with the vendor's LoC.

b. Provide all conferencing services data to the Government at a level of detail that allows the Government to bill conferencing services customers based on usage. The SUT met this requirement with the vendor's LoC.

2. The Accounting Management function shall enable charges to be established for the use of dedicated and dial-up conferencing services resources. The SUT does not meet this requirement. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.

3. The conference system Accounting Management function shall provide for the collection, aggregation, storage, and reporting of all conferencing service usage data. The accounting management function shall consist of systems to activate and monitor customer accounts and to collect, aggregate, and report on usage data. The SUT met this requirement with the vendor's LoC.

4. The conference system shall provide the capability to perform the following Accounting Management functions:

a. Collect usage data from a Call Detailed Record (CDR) (i.e., at the level of the authorization code). The SUT met this requirement with the vendor's LoC.

b. Aggregate and combine data for generating reports as specified by the Government. The SUT met this requirement with the vendor's LoC.

c. Conference records per individual customer's account. The SUT met this requirement with the vendor's LoC.

d. Ensure continuous (24x7) monitoring, processing, and recording for all video services-related events and customer activity data. The SUT met this requirement with the vendor's LoC.

e. Maintain a database of various conference reports per individual customer account, including conference detail summary, completion summary, and exception reports. The SUT met this requirement with the vendor's LoC.

f. Archive data for possible later retrieval by the Government (e.g., in response to customer inquiries or to audit the data). The SUT met this requirement with the vendor's LoC.

5. The conference system shall provide the capability to transmit usage data to the EMS. The frequency of data transfer shall be determined by the Government based on volume of data collected. The system shall maintain all accounting data for at least one billing cycle. The SUT met this requirement with the vendor's LoC.

**c. Hardware/Software/Firmware Version Identification.** Table 3-3 provides the SUT components' hardware, software, and firmware tested. The JITC tested the SUT in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic. Table 3-4 provides the hardware, software, and firmware of the components used in the test infrastructure.

**7. TESTING LIMITATIONS.** None.

**8. CONCLUSION(S).** The SUT meets the critical interoperability requirements for a Video Only UCCS in accordance with the UCR and is certified for joint use with other UC Products listed on the APL. The SUT was tested and certified for use with AS-SIP interface when configured on the line side of the REDCOM HDX and SLICE LSCs. The SUT was tested with the REDCOM Release 4.0AR3P9; however, the SUT is certified with any version of the REDCOM HDX and SLICE LSCs currently or previously on the UC APL. The SUT meets the interoperability requirements for the interfaces listed in Table 3-1.

## DATA TABLES

**Table 3-1. Interface Status**

Interface (Protocol)	Threshold CR/FR Requirements (See note 1.)	Status	Remarks
Ethernet 10 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 100 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 1000 Mbps (AS-SIP) (R) (See note 2.)	1	Certified	
Ethernet 10000 Mbps (AS-SIP) (R) (See note 2.)	1	Not Tested	See note 3.
Ethernet 10 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
Ethernet 100 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
Ethernet 1000 Mbps (H.323) (C) (See note 2.)	1	Not Tested	See note 4.
T1 ISDN PRI (H.320) (C)	1	Not Tested	See note 4.
E1 ISDN PRI (H.320) (C)	1	Not Tested	See note 4.
<b>NOTES:</b>			
1. The UCR does not identify interface CR/FR applicability. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column are cross-referenced with Table 3-2.			
2. The UCR states that an Ethernet interface is required; however, it does not stipulate a specific rate. The SUT supports AS-SIP natively through the SUT Ethernet interfaces.			
3. Although the SUT supports this interface, it was not tested and is not covered under this certification			
4. The SUT does not support this conditional interface.			
<b>LEGEND:</b>			
AS-SIP	Assured Services Session Initiation Protocol	Mbps	Megabits per second
C	Conditional	PRI	Primary Rate Interface
CR	Capability Requirement	R	Required
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	UCR	Unified Capabilities Requirements
ISDN	Integrated Services Digital Network		

**Table 3-2. Capability and Functional Requirements and Status**

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	<b>UC Audio and Video Conference System (R)</b>		
	System Description (R)	3.4.2	Met (See note 2.)
	Service (R)	3.4.3	Partially Met (See notes 3, 4, 5, 6, 7, 8.)
	Service Performance (R)	3.4.4	Partially Met (See notes 9, 10, 11.)
	Service Management (R)	3.4.5	Partially Met (See notes 4, 12, 13, and 14.)
<b>NOTES:</b>			
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Table 3-5. The SUT met the requirements for a Video Only (VO) Unified Capabilities Conferencing System (UCCS) with the exceptions noted below.			
2. Security testing is accomplished by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).			
3. The SUT video client is unable to set DSCP tagging due to Windows 7 limitation. The SUT must be fielded in accordance with the deployment guide.			
4. The SUT does not provide a registration system or web access to the conference system. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.			
5. The SUT does not support all of the required audio and video codecs. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.			
6. The SUT does not fully comply with the chair control functionality requirements. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as having a minor operational impact.			

**Table 3-2. Capability and Functional Requirements and Status (continued)**

<b>NOTES (continued):</b>			
7. The SUT does not support H.239 or H.261 annex D standards. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
8. The SUT is not capable of extending a conference. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
9. The SUT does not support H.323 gateway capacity. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
10. The SUT does not provide H.239 capability. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
11. The SUT does not support dial-up video services. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR. The SUT does not provide an online directory. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.			
12. The SUT does not provide a scheduling system. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to optional in the next version of the UCR.			
13. The SUT is unable to manage or monitor reservation or scheduling services. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
14. The SUT partially complies with the Fault Management Systems requirement. DISA has adjudicated this discrepancy as having a minor operational impact.			
15. The SUT partially complies with Fault management functions requirement. DISA adjudicated this as minor and stated the intent to move this requirement to Section 15 and broaden by removing "dedicated and dial-up video services".			
16. The SUT does not support accounting management for dedicated or dial-up services. DISA has adjudicated this discrepancy as having a minor operational impact and stated the intent to change this requirement to conditional in the next version of the UCR.			
<b>LEGEND:</b>			
CR	Capability Requirement	ID	Identification
DISA	Defense Information Systems Agency	MTU	Maximum Transmission Unit
DMA	Distributed Media Application	POA&M	Plan of Action and Milestones
DSCP	Differentiated Services Code Point	R	Required
FR	Functional Requirement	SUT	System Under Test
IPv4	Internet Protocol version 4	UC	Unified Capabilities
IPv6	Internet Protocol version 6	UCR	Unified Capabilities Requirements

**Table 3-3. SUT Hardware/Software/Firmware Version Identification**

Component	Release	Sub-component	Function
Acano Solution UCCS	1.5	NA	MCU
Acano Solution video client	1.5.1.1	Window 7 Laptop	Video PEI
Management Laptop (site-provided)	Microsoft Windows 7 SP 1	NA	User interface to the SUT.
<b>LEGEND:</b>			
MCU	Multipoint Conferencing Unit	SP	Service Pack
NA	Not Applicable	SUT	System Under Test
PEI	Proprietary End Instrument	UCCS	Unified Capabilities Conference System

**Table 3-4. Test Infrastructure Hardware/Software/Firmware Version Identification**

System Name	Software Release	Function	
<b>Required Ancillary Equipment (site-provided)</b>			
Active Directory			
Public Key Infrastructure			
<b>Test Network Components</b>			
Cisco 2961	IOS 12.1(22)EA14(fc1)	switch	
Polycom Real Presence Group Series 300	4.1.0_J	AS-SIP Video PEI	
Polycom Real Presence Group Series 500	4.1.0_J	AS-SIP Video PEI	
Polycom Real Presence Group Series 700	4.1.0_J	AS-SIP Video PEI	
Polycom Real Presence Collaboration Server (RMX) 1500	8.3.0_J	AS-SIP UCCS	
Vidyo UCCS	3.2_D	AS-SIP UCCS	
Avaya AS5300	Release 3.0 Service Pack 8	Local Session Controller	
Avaya 1140E and Avaya 1120E	04.02.13.00	IP voice instrument	
AudioCodes MP112 and AudioCodes MP800	6.20A.055.001	Audio gateway	
Analog phones off of AudioCodes MP112 or MP800	Not Applicable	Analog voice instrument	
Avaya Media Server	7.5.0.919	Media server	
Avaya AS5300	Release 2.0 Patch Bundle 28	Local Session Controller	
Avaya AS5300 with CS2100	Release 2.0 Patch Bundle 28/ Succession Enterprise 09.1	Multifunction Softswitch	
REDCOM SLICE	4.0AR3P9	Local Session Controller	
REDCOM HDX	4.0AR3P9	Local Session Controller	
Cisco Unified Communications Manager (UCM)	8.6	Local Session Controller	
Cisco Unified Communications Manager (UCM)	8.6	Enterprise Session Controller	
Cisco MSE 5230	2.2 (1.17)	Proprietary MCU	
Cisco C90	TC 7.1.1	AS-SIP Video PEI	
Cisco SX 20	TC 7.1.1	AS-SIP Video PEI	
Cisco Jabber	9.2.6	XMPP client	
<b>LEGEND:</b>			
AS	Application Server	MSE	Media Services Engine
AS-SIP	Assured Services Session Initiation Protocol	PEI	Proprietary End Instrument
HDX	High Density Exchange (REDCOM)	UCCS	Unified Communications Conference System
IP	Internet Protocol	XMPP	Extensible Messaging and Presence Protocol
MCU	Multipoint Control Unit		

**Table 3-5. UCCS Capability/Functional Requirements**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
<b>1</b>	<b>3.4.2 – System Description</b>					
1-1	The conference system shall provide, as a minimum, an Ethernet-based interface to the network.	3.4.2.2 AUX-002720	T	R	R	R
1-2	The conference system shall meet the Information Assurance requirements of all applicable DISA Security Technical Implementation Guidelines (STIGs). Section 3.4.5.1.6, Security Management, contains conference system security management requirements.	3.4.2.3 AUX-002730	IA	R	R	R
<b>2</b>	<b>3.4.3 – Service</b>					
2-1	All subscriber EIs and devices directly utilizing the conferencing system shall be required to register with the system in order to use conferencing services. This registration requirement is not meant to exclude non-registered EIs and external parties from participating in a conference to which they have been invited by the registered subscriber that scheduled the conference. Section 3.4.5.3, Registration System, contains conference services registration system requirements.	3.4.3.1.1 AUX-002740	T	R	R	R
2-2	The conference system shall provide IP-based point-to-point conferencing. Point-to-point conferencing consists of two participants with fully interactive audio and/or video capabilities. The system shall support EIs that are registered with the system to initiate point-to-point, fully interactive audio and video capability communications. This capability shall be supported through conferencing services that enable the resolution of resource conflicts by calendaring and scheduling system application programming interfaces (APIs) with enterprise scheduling systems. It is desired that the UCCS shall provide real-time conferencing status capability; e.g., busy, online, offline.	3.4.3.1.2 AUX-002750	T	R	R	R
2-3	The conference system shall support IP-based solutions using AS-SIP, [Optional] H.323, and [Optional] dial-up ISDN H.320 endpoint support.	3.4.3.1.2 AUX-002760	T	R	R	R
2-4	The conference system shall support interactive conferences using IP and [Optional] ISDN transport. These conferences shall consist of Proprietary and AS-SIP EIs only, [Optional] H.323 EIs only, [Optional] H.320 EIs only, or [Optional] a combination of H.323 EIs, Proprietary and AS-SIP EIs, and H.320 EIs. ISDN transport applies only to conferences involving non-IP EIs.	3.4.3.1.2 AUX-002770	T	R	R	R
2-5	The conference system shall provide multipoint conferencing. A multipoint conference consists of three or more EIs in a conference call and shall include the following functions and features.	3.4.3.1.3 AUX-002780	T	R	R	R
2-6	The conference system shall distribute fully interactive video and/or audio streams among multiple participants according to the channel bandwidth of each participant.	3.4.3.1.3 AUX-002790	T	R	R	R
2-7	The conference system shall accommodate users on the same conference at different video rates, resolutions, and frame rates according to EI capability and not at the lowest common denominator level	3.4.3.1.3 AUX-002800	T		R	R
2-8	The conference system shall provide interactive, multipoint conferences using IP transport and [Optional] ISDN transport. These conferences shall consist of Proprietary and AS-SIP EIs only, [Optional] H.323 EIs only, [Optional] H.320 EIs only, or [Optional] a combination of H.323 EIs Proprietary and AS-SIP EIs, and H.320 EIs. ISDN transport applies only to conferences involving non-IP EIs.	3.4.3.1.3 AUX-002810	T	R	R	R
2-9	The conference system shall provide in-conference control. In-conference control shall include the following functions and features.	3.4.3.1.5 AUX-002820	T	R	R	R
2-10	The conference system shall provide a banner for each conference.	3.4.3.1.5 AUX-002830	T		R	R
2-11	The conference system shall provide notification of participants joining and leaving a conference, and provide an end-of-conference warning to all participants.	3.4.3.1.5 AUX-002840	T	R	R	R
2-12	The conference system shall provide the ability to extend conferences, without disruption, to conferences in progress.	3.4.3.1.5 AUX-002850	T	R	R	R
2-13	The conference system shall support presentation capability for different screen layouts locally, manageable by each individual host.	3.4.3.1.5 AUX-002860	T		R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
2-14	The conference system shall provide the following conferencing chair control functionality:	3.4.3.1.5 AUX-002870	N/A	R	R	R
2-15	Voice-activated switching.	3.4.3.1.5 AUX-002870.a	T		R	R
2-16	Broadcast mode.	3.4.3.1.5 AUX-002870.b	T	R	R	R
2-17	Lecture mode.	3.4.3.1.5 AUX-002870.c	T	R	R	R
2-18	Video switching with H.243 control.	3.4.3.1.5 AUX-002870.d	L/T		R	R
2-19	Continuous presence.	3.4.3.1.5 AUX-002870.e	T		R	R
2-20	Add/Delete, Accept/Reject, Connect/Disconnect, Mute/Unmute, audio/video.	3.4.3.1.5 AUX-002870.f	T	R	R	R
2-21	The conference system shall provide transcoding availability regardless of the data speeds; the number of concurrent video calls; and the number of concurrent conferences, video sizes, frame rates, and conference modes (voice switching or continuous presence) without downgrading the conference to a lowest common denominator protocol.	3.4.3.1.6 AUX-002880	T		R	R
2-22	The conference system shall provide automatic video transcoding without downgrading the conference to a lowest common denominator protocol.	3.4.3.1.6 AUX-002890	T		R	R
2-23	The conference system shall provide automatic audio transcoding without downgrading the conference to a lowest common denominator protocol.	3.4.3.1.6 AUX-002900	T	R		R
2-24	The conference system shall provide support for variable data rates, the rate at which data (bits) is transmitted, usually expressed in bits per second (bps) per Conferencing Terminal Unit (CTU). The system shall support data rates of at least 64 kilobits per second (kbps) per CTU. The system shall provide speed matching and down speeding to facilitate adjustable data rates. The system shall provide bandwidth management of IP services in order to restrict the bandwidth used by active call connections to the bandwidth installed and available in the network access connections.	3.4.3.1.7 AUX-002910	T		R	R
2-25	The conference system shall provide audio add-on features for audio-only participants in video conferences and support an external VoIP audio conference connecting to the video conference session.	3.4.3.1.8 AUX-002920	T			R
2-26	The UCCS shall provide content sharing capability for participants to interact during video teleconferencing (VTC) sessions that allow participants to view and display the same presentation material at the same time. The system shall provide a dedicated live video stream and a presentation video stream and still-frame graphics as specified in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols. The system shall provide Interactive Graphics Exchange with the following functions and features.	3.4.3.1.9 AUX-002930	T		O	O
2-27	The conference system shall provide a means to allow participants to interactively view images from external sources with all or any of the participants in the conference.	3.4.3.1.9 AUX-002940	T		O	O
2-28	The conference system shall provide real-time participation of any combination of EIs. The system shall provide still image exchange as specified in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols.	3.4.3.1.9 AUX-002950	T		O	O
2-29	Interactive graphics exchange capabilities shall include the following: <ul style="list-style-type: none"> <li>• Point-to-point and multipoint conferencing services</li> <li>• Interoperability with different vendor EIs and variable graphic resolutions.</li> <li>• Connections among participants using any video EIs, connection types, and at any rates.</li> </ul>	3.4.3.1.9 AUX-002960	T		O	O
2-30	The system shall be capable of accepting audio-only participants into a conference call for both scheduled and ad hoc video conferences.	3.4.3.1.10 AUX-002970	T			R
2-31	The system shall provide an interface to an external audio conferencing system to allow cascading between a multi-point VTC call with a multi-point audio call through the use of the interface to an external audio conferencing system.	3.4.3.1.10 AUX-002980	L/T		R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
2-32	The system shall provide internal conferencing capabilities for the support of audio-only participants.	3.4.3.1.10 AUX-002990	T	R		R
2-33	The conference system shall provide a Web-based portal for customer access to the UC conferencing services, features, and capabilities. As the conferencing services change, the Web-based portal shall reflect those changes. Additionally, the conferencing services Web portal shall provide the following:	3.4.3.2.1 AUX-003000	L/T	R	R	R
2-34	An enterprise-wide service for the identification and other pertinent information about users, conferencing services, and resources, and makes it accessible from any place at any time.	3.4.3.2.1 AUX-003000.a	L/T	R	R	R
2-35	Awareness of relevant, accurate information about the conferencing service to users at all levels (strategic, operational, and Tactical).	3.4.3.2.1 AUX-003000.b	L/T	R	R	R
2-36	An integrated scheduling system that provides users the ability to schedule one or a combination of video and audio conference services in one Web interface.	3.4.3.2.1 AUX-003000.c	L/T	R	R	R
2-37	The conference system shall provide a video conferencing recorded content request and retrieval system in accordance with (IAW) the following requirements.	3.4.3.2.2.1 AUX-003010	T		O	O
2-38	The system shall provide the user the ability to view and listen to the recorded video conferences using a Web browser to retrieve streaming video.	3.4.3.2.2.1 AUX-003020	T		O	O
2-39	The system shall provide a Web-based system for the meeting moderator to access the recorded video conference call.	3.4.3.2.2.1 AUX-003030	T		O	O
2-40	The system shall inform the meeting moderator of the recorded video conference access information immediately after the completion of the video conference.	3.4.3.2.2.1 AUX-003040	T		O	O
2-41	The system shall provide Web-based interfaces for users to search recorded content based on the combination of the following information: meeting topic, meeting date/time, keywords provided by meeting moderators, meeting leader name, meeting language, and meeting leader organization.	3.4.3.2.2.1 AUX-003050	T		O	O
2-42	The Web interface shall provide a link to the content retrieval launch page.	3.4.3.2.2.1 AUX-003060	T		O	O
2-43	The content retrieval launch page shall authenticate the users using PKI and prompt users to enter passwords defined by the meeting moderators during the meeting scheduling phase.	3.4.3.2.2.1 AUX-003070	T		O	O
2-44	The content retrieval launch page shall maintain a record of every content request. The record of each request shall include the name, email address, and E.164 number or IP address of the requester; the identification of the recording; and the date and time of the request.	3.4.3.2.2.1 AUX-003080	T		O	O
2-45	The content retrieval launch page shall allow users to choose which format of the supported streaming media formats to use when playing back the retrieved content.	3.4.3.2.2.1 AUX-003090	T		O	O
2-46	The system shall provide the end user controls during streaming to pause/resume and select segments to play.	3.4.3.2.2.1 AUX-003100	T		O	O
2-47	The content retrieval launch page shall allow users to download the stored content of a conference meeting, if this option is permitted by the meeting's moderator.	3.4.3.2.2.1 AUX-003110	T		O	O
2-48	The streaming shall comply with the Streaming Service Protocol Requirements described in Section 3.4.3.3.1, Compression Algorithms and Audio/Video Protocols.	3.4.3.2.2.1 AUX-003120	L/T		O	O
2-49	The system shall ensure the compatibility of stored content with the latest versions of media player clients.	3.4.3.2.2.1 AUX-003130	L/T		O	O
2-50	The conference system shall provide a content management system IAW the following requirements.	3.4.3.2.2.2 AUX-003140	L/T	O		O
2-51	The content management system shall comply with DoD 5200.1R with clear marking and labeling.	3.4.3.2.2.2 AUX-003150	L/T	O		O
2-52	The content management system shall maintain recorded audio conferencing content ready for users to retrieve anytime for a period of 30 days after the completion of the conferences.	3.4.3.2.2.2 AUX-003160	L/T	O		O
2-53	The content management system shall archive recorded audio conferencing content into permanent storage after 30 days.	3.4.3.2.2.2 AUX-003170	L	O		O

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
2-54	The content management system shall allow users to request stored content from archive. The wait time to retrieve archived material shall be less than one working day.	3.4.3.2.2.2 AUX-003180	L	O		O
2-55	The archive material shall be kept at the same fidelity levels of the original recordings. Compression techniques that cause a loss of fidelity are not acceptable encoding schemes for archive material.	3.4.3.2.2.2 AUX-003190	L	O		O
2-56	This subsection describes the system's interoperability requirements. The system shall maximize the use of standards-based interfaces. The system shall use functions, protocols, and formats that are publicly available.	3.4.3.3 AUX-003200	L	R	R	R
2-57	For video equipment, the conference system shall adhere to Federal Telecommunications Recommendation 1080B-2002 (FTR-1080B).	3.4.3.3 AUX-003210	L/T		O	O
2-58	The conference system shall support the following audio and video standards for video conferencing: <u>Audio Protocols</u> G.711 G.722 G.722.1 G.723.1 G.728 G.729/G.729A <u>Video Protocols</u> H.363-200 H.264	3.4.3.3.1 AUX-003220	L/T		R	R
2-59	The conference system shall support the following video standards for video conferencing: H.261 H.264 (SVC)	3.4.3.3.1 AUX-003230	L/T		O	O
2-60	The conference system shall support the following audio standards for audio-only conferencing: G.711 G.722 G.722.1 G.723.1 G.728 G.729/G.729A	3.4.3.3.1 AUX-003240	L/T	R		R
2-61	The conference system shall provide interoperability for all end point devices that support AS-SIP during call setup.	3.4.3.3.1 AUX-003250	L/T	R	R	R
2-62	The conference system shall provide interoperability for all end point devices that support H.320 during call setup.	3.4.3.3.1 AUX-003260	L/T	O	O	O
2-63	The conference system shall provide interoperability for all end point devices that support H.323 during call setup.	3.4.3.3.1 AUX-003270	L/T	O	O	O
2-64	The conference system, including any proprietary Video EIs, shall support the following:	3.4.3.3.1 AUX-003280	L		R	R
2-65	Sub-Quarter Common Intermediate Format (SQCIF).	3.4.3.3.1 AUX-003280.a	L		R	R
2-66	Quarter Common Intermediate Format (QCIF).	3.4.3.3.1 AUX-003280.b	L		R	R
2-67	Common Intermediate Format (FCIF, also called CIF).	3.4.3.3.1 AUX-003280.c	L		R	R
2-68	4 Full Common Intermediate Format (4FCIF, also called 4CIF).	3.4.3.3.1 AUX-003280.d	L		R	R
2-69	16 Full Common Intermediate Format (16FCIF, also called 16 Common Intermediate Format (16CIF)).	3.4.3.3.1 AUX-003280.e	L		O	O

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS	
2-70	SD and HD video resolution formats for H.261, H.263, and H.264 codecs.	3.4.3.3.1 AUX-003280.f	L		O	O	
	<u>Video Format Standards</u>						<u>Video Resolution</u>
	SQCIF						128 x 96
	QCIF						176 x 144
	SIF(525)						352 x 240
	CIF/SIF(625)						352 x 288
	4SIF(525)						704 x 480
	4CIF/4SIF(625)						704 x 526
	16CIF						1408 x 1152
	DCIF						528 x 384
SD	720 x 480						
HD(720p)	1280 x 720						
HD (1080p)	1920 x 1080						
2-71	The conference system shall ensure that the freeze-frame image feature is compliant with ITU-T H.239 and with H.261 Annex D.	3.4.3.3.1 AUX-003290	L		R	R	
2-72	The system's freeze-frame image size shall support 4FCIF (4CIF), VGA, SVGA, XGA, and WSXGA+ when using H.239.	3.4.3.3.1 AUX-003300	L		R	R	
2-73	The system's freeze-frame image size shall support HD (720p) and HD (1080p) when using H.239.	3.4.3.3.1 AUX-003310	L		O	O	
2-74	The system's freeze-frame image size shall support 4FCIF (4CIF) when using H.261 Annex D.	3.4.3.3.1 AUX-003320	L		R	R	
2-75	The conference system that supports H.323/H.320 protocols shall meet the following ISDN/PRI, H.323 V4, chair control, serial interfaces, content sharing VTC endpoint protocol requirements:	3.4.3.3.2 AUX-003330	L/T		O	O	
2-76	ISDN PRI on ISDN interfaces (including Alcatel-Lucent 5ESS PRI, GENBAND DMS PRI, and National ISDN PRI).	3.4.3.3.2 AUX-003330.a	L/T		O	O	
2-77	European E1 ISDN standards.	3.4.3.3.2 AUX-003330.b	L/T		O	O	
2-78	ISDN bonding up to 1.5 Mbps on T1 and 2 Mbps on E1 per International Organization for Standardization (ISO) 13871.	3.4.3.3.2 AUX-003330.c	L/T		O	O	
2-79	H.323/320 V4.	3.4.3.3.2 AUX-003330.d	L/T		O	O	
2-80	Far end camera control (FECC) H.281 and H.323 Annex Q.	3.4.3.3.2 AUX-003330.e	L/T		O	O	
2-81	Resource Availability Indicator (RAI)/Resource Availability Confirmation (RAC) for load balancing.	3.4.3.3.2 AUX-003330.f	L/T		O	O	
2-82	Chair control messages per H.246, H.242/H.243.	3.4.3.3.2 AUX-003330.g	L/T		O	O	
2-83	Direct, H.225 routed, and H.225-H.245 routed modes of H.323 gatekeeper operations.	3.4.3.3.2 AUX-003330.h	L/T		O	O	
2-84	Quality of Service (QoS) support using DSCP marking of IP packets.	3.4.3.3.2 AUX-003330.i	L/T		O	O	
2-85	Automatic downspeed to available ISDN/IP bandwidth.	3.4.3.3.2 AUX-003330.j	L/T		O	O	
2-86	Automatic rate detection to match incoming video calls.	3.4.3.3.2 AUX-003330.k	L/T		O	O	
2-87	V.35/RS-449/EIA-530 Data Terminating Equipment (DTE) and Data Circuit-Terminating Equipment (DCE) interfaces (The implementation shall use EIA-530 interfaces, and the use of V.35 and RS-449 interfaces shall be phased out where multiple interfaces are supported in equipment. RS-366 interfaces shall also be supported for dial signaling bypass devices).	3.4.3.3.2 AUX-003330.l	L/T		O	O	
2-88	H.239 for additional video channels or still images.	3.4.3.3.2 AUX-003330.m	L/T		O	O	
2-89	The AS-SIP audio and video signaling conferencing requirements are specified in AS-SIP Section 10, Audio and Video Conference Services.	3.4.3.3.3 AUX-003340	L/T	R	R	R	
2-90	FECC H.281 and H.323 Annex Q.	3.4.3.3.3 AUX-003350	L/T		O	O	
2-91	Chair control messages per H.246, H.242/H.243.	3.4.3.3.3 AUX-003360	L/T		O	O	
2-92	QoS support using DSCP marking of IP packets.	3.4.3.3.3 AUX-003370	T	R	R	R	

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
2-93	Automatic downspeed to available IP bandwidth.	3.4.3.3.3 AUX-003380	L/T		R	R
2-94	Automatic rate detection to match incoming video calls.	3.4.3.3.3 AUX-003390	L/T		R	R
2-95	Support T.140 for text messages.	3.4.3.3.3 AUX-003400	L/T	O	O	O
2-96	H.261 Annex D for still images.	3.4.3.3.3 AUX-003410	L/T		O	O
2-97	Video Switching Mode. The system shall ensure that the system supports video switching according to H.243 and H.323. The design shall minimize the time to switch and the disruption of video when switching from one video source to another.	3.4.3.3.4 AUX-003420	L/T		R	R
2-98	Video Mixing (Picture Composition or Continuous Presence Mode). The system shall ensure that the system supports video mixing functions according to H.243 and H.323. The system shall support enhanced continuous presence multiple video mixing to include the 7 plus 1 format.	3.4.3.3.4 AUX-003430	L/T		R	R
2-99	H.320 chair control messages and procedures as defined in H.230, H.242, H.243 and H.245.	3.4.3.3.5 AUX-003440	L/T		O	O
2-100	H.323 multipoint conferencing units (MCUs) shall support chair control messages and procedures as defined in H.323 and as carried forward to H.323 from H.243.	3.4.3.3.5 AUX-003450	L/T		O	O
2-101	H.323 – H.320 gateways shall follow the H.246 message translation tables related to chair control functions.	3.4.3.3.5 AUX-003460	L/T		O	O
2-102	Audio systems shall support in-dial and out-dial IAW the DISN World Wide Numbering Plan and the PSTN North American dialing plans.	3.4.3.3.6 AUX-003470	T	R		R
2-103	The audio system's PSTN interfaces shall support T1/E1 (AT&T TR62411 or Telcordia TR-NWT-000170).	3.4.3.3.6 AUX-003480	L/T	O		O
2-104	The audio system's CAS interfaces shall support Alcatel-Lucent 5ESS and GENBAND DMS switches.	3.4.3.3.6 AUX-003490	L/T	O		O
2-105	The audio system's T1 PRI interfaces shall support Non-Facility Associated Signaling (NFAS) and D-channel backup, if the audio system supports more than two PRIs.	3.4.3.3.6 AUX-003500	L/T	O		O
2-106	The audio system's T1 interface shall support extended super frame (ESF) framing and with bipolar with eight-zero substitution (B8ZS)/ Alternate Mark Inversion (AMI) coding.	3.4.3.3.6 AUX-003510	L/T	O		O
2-107	The audio system's PSTN signaling module shall support ISDN PRI (5ESS, DMS, and National ISDN), and the PRI flavors of foreign countries such as Germany, Japan, and Korea.	3.4.3.3.6 AUX-003520	L/T	O		O
2-108	The audio system shall support the Dialed Number Identification Service (DNIS) feature where the original dialed numbers are presented as generic address parameters (GAPs).	3.4.3.3.6 AUX-003530	L/T	O		O
2-109	The audio system shall support the automatic number identification (ANI) feature and use it to identify a calling party, if applicable.	3.4.3.3.6 AUX-003540	L/T	O		O
2-110	The audio system IP interfaces shall support static assignment of the IP address, mask, default router, and Domain Name Service (DNS) entries.	3.4.3.3.6 AUX-003550	T	R		R
2-111	The audio system shall support multiple DNS entries. If the primary DNS server does not respond to a DNS request, then a secondary DNS server shall be queried.	3.4.3.3.6 AUX-003560	T	R		R
2-112	The audio system shall support configurable transmission control protocol (TCP) ports for AS-SIP messaging.	3.4.3.3.6 AUX-003570	L/T	R		R
2-113	The audio system shall be able to set the IPv4/IPv6 Precedence Field bits of the Type of Service (TOS) byte and DSCP bits for media streams and signaling streams.	3.4.3.3.6 AUX-003580	L/T	R		R
2-114	The audio system shall support Network Time Protocol (NTP), version 3 [RFC 1305].	3.4.3.3.6 AUX-003590	L/T	R		R
2-115	The audio system shall support SNMPv3 [RFC 3414].	3.4.3.3.6 AUX-003600	L/T	R		R
2-116	The audio system shall support Secure Real-Time Transport Protocol (SRTP) and Secure Real-Time Transport Control Protocol (SRTCP) [RFC 3711].	3.4.3.3.6 AUX-003610	L/T	R		R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
2-117	The audio system shall support SRTP and accurately report jitter, delay, and packet loss information to the far end using Real-Time Transport Protocol (RTP) Control Protocol Extended Reports (RTCP XR) [RFC 3611].	3.4.3.3.6 AUX-003620	L/T	R		R
2-118	The audio system's AS-SIP signaling module shall support RFC 3261 including loose route.	3.4.3.3.6 AUX-003630	L/T	R		R
2-119	The audio system's AS-SIP signaling module shall allow AS-SIP URLs for both incoming and outgoing calls. This includes all alphanumeric characters allowed in legal SIP URLs.	3.4.3.3.6 AUX-003640	L/T	R		R
2-120	The audio system's AS-SIP signaling module shall support Session Description Protocol (SDP) as defined in RFC 4566.	3.4.3.3.6 AUX-003650	L/T	R		R
2-121	The audio system's AS-SIP signaling module shall implement user "hold" feature by using a=inactive or a=sendonly, or by sending a mid-call INVITE that includes a session description that is the same as in the original request, but the "c" destination addresses for the media streams to be put on hold are set to zero:c=IN IP4 0.0.0.0.	3.4.3.3.6 AUX-003660	T	R		R
2-122	The AS-SIP signaling module shall support AS-SIP Digest Authentication [RFCs 3261 and 3310].	3.4.3.3.6 AUX-003670	L/T	R		R
2-123	The AS-SIP signaling module shall be able to reject incoming INVITE messages when the message does not come from pre-provisioned proxies.	3.4.3.3.6 AUX-003680	T	R		R
2-124	The AS-SIP signaling module shall support call transfer as specified in AS-SIP Section 9.6, Call Transfer.	3.4.3.3.6 AUX-003690	T	R		R
2-125	Audio systems shall support electronic numbering (ENUM) service registration for SIP (AS-SIP) Addresses-of-Record [RFC 3764].	3.4.3.3.6 AUX-003700	L/T	R		R
2-126	The audio system shall support DTMF Generation/Recognition per Telcordia GR-181-CORE.	3.4.3.3.6 AUX-003710	L/T	R		R
2-127	The audio system shall support G.711 $\mu$ /A law [pulse code modulation (PCM)] and G.729.	3.4.3.3.6 AUX-003720	L/T	R		R
2-128	The audio system's total media processing time shall be less than 50 ms including delays from jitter buffer, transcoding, mixing, packetization, and algorithm look ahead.	3.4.3.3.6 AUX-003730	L	R		R
2-129	The audio system shall support G.168 compliance echo canceller (EC) with 128 ms echo path.	3.4.3.3.6 AUX-003740	L/T	R		R
2-130	The audio system shall support Audio and Video Transport (AVT) payload type 0 and 8. [G.711 a/ $\mu$ law].	3.4.3.3.6 AUX-003750	L/T	R		R
2-131	The audio system shall support AVT payload 18 [G.729].	3.4.3.3.6 AUX-003760	L/T	R		R
2-132	The audio system shall be able to accept in-band DTMF tones.	3.4.3.3.6 AUX-003770	T	R		R
2-133	The audio system shall be able to send DTMF specified by RFC 4733.	3.4.3.3.6 AUX-003780	L/T	R		R
2-134	The audio system shall be able to conceal 1 percent of packet loss without appreciable quality degradation.	3.4.3.3.6 AUX-003790	L	R		R
2-135	The audio system shall be able to tolerate 40 ms of jitter for audio without appreciable quality degradation.	3.4.3.3.6 AUX-003800	L	R		R
2-136	The audio system shall implement adaptive jitter buffers instead of static fix jitter buffers.	3.4.3.3.6 AUX-003810	L/T	R		R
2-137	All hardware shall meet Network Equipment Building System-3 (NEBS-3) requirements.	3.4.3.3.6 AUX-003820	L	R		R
2-138	This subsection addresses the Maximum Transmission Unit (MTU) requirements for an IP network environment. An MTU is the maximum size of an IP packet that will be accepted for transmission without fragmenting it into a smaller datagram. The MTU size shall be configurable to optimize video traffic. As a result of devices such as encryption units, the typical MTU size shall be changed to minimize the effect of fragmentation because of the additional overhead of the encryption.	3.4.3.3.7 AUX-003830	L		R	R
2-139	The conference system shall ensure that all conferencing services provide signaling and media streams, and are capable of configuring the MTU.	3.4.3.3.7 AUX-003840	L	R	R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
2-140	The conference system shall ensure that all supporting services to video and audio services, including, but not limited to, reservation, monitoring, billing, administration, operator interface, and meeting control, are capable of working in the configurable MTU IP environment.	3.4.3.3.7 AUX-003850	L	R	R	R
2-141	The conference system shall comply with the IPv6 requirements contained in Section 5, IPv6.	3.4.3.3.8 AUX-003860	L	R	R	R
2-142	The conference system shall comply with the AS-SIP requirements contained in AS-SIP 2013, Section 4, SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs.	3.4.3.3.9 AUX-003870	L	R	R	R
2-143	The system or network device shall be able to set DSCPs on both signaling packets and media streams for both IPv4 and IPv6 as specified in Section 6.2.2, Differentiated Services Code Point.	3.4.3.4.1 AUX-003880	L/T	R	R	R
2-144	The system shall provide measures to monitor bandwidth resource usage and to activate congestion management as needed within a timely fashion.	3.4.3.4.2.1 AUX-003890	L	R	R	R
2-145	The system shall maintain a call summary of conference sessions. This will include the conference attendee identification, access methods, IP address, E.164 numbers, time and date of the call, call duration, and total number of participants. The summaries shall be maintained for 30 days or IAW Information Assurance security requirements.	3.4.3.4.2.2 AUX-003900	L	R	R	R
2-146	The system shall operate IAW the MLPP rules and procedures specified in Section 2.26.2, Multilevel Precedence and Preemption (inclusive as applies).	3.4.3.4.2.3 AUX-003910	L/T	O	O	O
2-147	Preset Conferencing. Each conferee shall be dialed at its designated precedence level. Each conferee may have a different precedence level. The conference host must be dialed at the highest precedence level of any conferee.	3.4.3.4.2.3 AUX-003920	T	O	O	O
2-148	Meet-Me Conferencing. When a priority session requests connection to a conference that is at conference maximum, then one of the lowest precedence conferees shall be preempted, selected through any deterministic method. (Conference maximum is the maximum number of conferees authorized for the same conference.) When a priority session requests connection to a conference that is not at conference maximum, but the system is at system maximum, then one of the lowest precedence conferees on another conference will be preempted, selected through any deterministic method. Note that the selection method shall not consider any ad hoc conferees for preemption, if an allocation of system resources dedicated to ad hoc conferences has been configured per requirement AUX-003880 in Section 3.4.4.2.1, Video Conference Capacity. (System maximum occurs when the maximum number of ports or resources are provisioned on the system.) Preempted conferees shall receive a preemption notification tone and be preempted. All remaining conferees on the system shall receive a conference disconnect tone (see Table 2.9-2, UC Information Signals).	3.4.3.4.2.3 AUX-003930	T	O	O	O
2-149	Ad hoc Conferencing. When either party of a two-party session brings in a third party, an ad hoc conference is created at the highest precedence level of the dialed conferees. Thereafter, any party of an ad hoc conference can attempt to add an additional conferee at any time. If that party is dialed at a precedence level higher than any of the current conferees, then the conference precedence level shall be elevated to a higher precedence level. If an allocation of system resources dedicated to ad hoc conferences has been configured per requirement AUX-003880 in Section 3.4.4.2.1, and there are no ad hoc system resources available to add on a conferee, then a conferee from the lowest precedence ad hoc conference will be preempted so that the conferee can be brought into the higher precedence ad hoc conference. If an allocation of system resources dedicated to ad hoc conferences has not been configured, but the system is at system maximum, then one of the lowest precedence conferees on another conference will be preempted, selected through any deterministic method, so that the conferee can be brought into the higher precedence ad hoc conference.	3.4.3.4.2.3 AUX-003940	T	O	O	O

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
2-150	Ad hoc Conferencing. When a higher precedence session (i.e., higher than the conference precedence level) is placed to any of the conferees, that conferee receives a preemption notification tone (see Table 2.9-2, UC Information Signals). The other remaining conferees shall receive a conference disconnect tone, as described in Table 2.9-2. This tone indicates to the other parties that one of the conference call participants is being preempted.	3.4.3.4.2.3 AUX-003950	T	O	O	O
<b>3</b>	<b>3.4.4 – Service Performance</b>					
3-1	This section provides the service performance criteria and metrics for the system. The service performance criteria shall apply during simultaneous operation of the respective EIs. The system shall design and implement redundancy, failover, and fault tolerance at the component, subsystem, and system levels to support achieving service availability requirements in the presence of failures at the component, subsystem, and system levels. The system shall meet the requirements specified in Section 2.8, Product Physical, Quality, and Environmental Factors.	3.4.4 AUX-003960	L/T	R	R	R
3-2	The conference system shall ensure that all video equipment used in designs can operate in the presence of minimal packet loss without degrading video quality below acceptable levels.	3.4.4.1.1 AUX-003970	L		R	R
3-3	The conference system shall ensure that all video equipment used in the design provides adequate jitter buffer sizing to ensure an optimal end-to-end (E2E) video conferencing performance.	3.4.4.1.1 AUX-003980	L		R	R
3-4	The conference system shall ensure that the Mean Opinion Score (MOS) on the voice path meets the MOS requirements in Section 6, Network Infrastructure End-to-End Performance.	3.4.4.1.2 AUX-003990	L/T	R		R
3-5	The conference system shall ensure that the implemented design possesses the adequate performance capacity and resources to support conference access processing requirements identified in Section 3.4.4.2, Capacity.	3.4.4.1.2 AUX-004000	L/T	R		R
3-6	The conference system shall ensure the time needed to compile polling statistics to adequately support the audio conference capability.	3.4.4.1.2 AUX-004010	L	O		O
3-7	The conference system vendor shall provide documentation stating the conference system’s capacity and scalability. Statements regarding capacity and scalability will be validated at DISA’s discretion.	3.4.4.2 AUX-004020	L	R	R	R
3-8	The number of concurrent conferences supported shall be limited only by available ports and the number of licenses acquired, and shall not depend on the access methods, features, or number of participants in each conference.	3.4.4.2.1 AUX-004030	L		R	R
3-9	The system shall have the capacity to support at least 1,000 concurrent 384 kbps video calls.	3.4.4.2.1 AUX-004040	L		O	O
3-10	The system shall provide sufficient speed matching capacity to support that capability regardless of the access methods, algorithms, speeds, or the feature sets being used.	3.4.4.2.1 AUX-004050	L		R	R
3-11	The system shall provide enough transcoding capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used.	3.4.4.2.1 AUX-004060	L		R	R
3-12	The system shall provide sufficient H.323 gateway capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used.	3.4.4.2.1 AUX-004070	L		R	R
3-13	The system shall provide enough H.239 capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used.	3.4.4.2.1 AUX-004080	L		R	R
3-14	The system shall provide enough H.261 Annex D capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used.	3.4.4.2.1 AUX-004090	L		O	O
3-15	The system shall support a minimum of 200 EIs for each multipoint conference.	3.4.4.2.1 AUX-004100	L		O	O
3-16	The system shall provide at least four audio added-on ports for each conference without the use of external audio systems or by cascading conference systems.	3.4.4.2.1 AUX-004110	L/T			R
3-17	The system shall ensure that audio added-on does not compromise support for other capacity requirements.	3.4.4.2.1 AUX-004120	L/T		R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
3-18	The system shall support the ability to configure allocation (0 to 100 percent, default 20 percent) of system ports/resources to be dedicated to ad hoc video conferences. Meet-me conferences shall not use resources allocated to ad hoc conferences, and vice versa.	3.4.4.2.1 AUX-004130	L/T		O	O
3-19	The system shall be capable of scaling up to 2,500 concurrent audio calls.	3.4.4.2.2 AUX-004140	L	O		O
3-20	The system shall support up to 200 participants in a single conference session.	3.4.4.2.2 AUX-004150	L	O		O
3-21	The system shall be capable of scaling up to 500 concurrent conferences.	3.4.4.2.2 AUX-004160	L	O		O
3-22	The system shall be capable of scaling up to 500 or more conference control web sessions.	3.4.4.2.2 AUX-004170	L	O		O
3-23	The system shall be capable of scaling to support 10,000 or more reservations.	3.4.4.2.2 AUX-004180	L	O		O
3-24	The audio conference system shall support more than 50 concurrent recordings.	3.4.4.2.2 AUX-004190	L	O		O
3-25	The system shall support the ability to configure an allocation (0 to 100 percent; default 20 percent) of system ports and resources to be dedicated to ad hoc audio conferences. Meet-me conferences shall not use resources allocated to ad hoc conferences, and vice versa.	3.4.4.2.2 AUX-004200	L/T	O		O
3-26	The conference system shall have the capability to provide bandwidth management, EI registrations, admissions, status, and routing functions. Furthermore, the system shall be capable of scaling to support at a minimum of 1,000 concurrent conference calls and 10,000 concurrent registrations of EIs.	3.4.4.2.3 AUX-004210	L	O	O	O
3-27	The system architecture supporting the dedicated IP-based video services capability shall be able to scale to accommodate increased growth in dedicated IP-based video services EIs.	3.4.4.2.4 AUX-004220	L		R	R
3-28	The system architecture supporting the dial-up video services capability shall be designed to support up to a 50 percent increase in dial-up video services.	3.4.4.2.4 AUX-004230	L		R	R
3-29	The system shall be able to scale to accommodate increased growth to support increases in connections to ISDN networks. Additional capacity in regards to this item shall be used only to support ISDN dial-up video traffic.	3.4.4.2.4 AUX-004240	L		O	O
3-30	Audio add-on and audio conference service shall be able to scale to accommodate increased growth in call volume and EIs.	3.4.4.2.4 AUX-004250	L	R		R
<b>4</b>	<b>3.4.5 – Service Management</b>					
4-1	The conference system shall provide services management and monitoring for the Operation, Administration, Maintenance, and Provisioning (OAM&P) of conferencing services.	3.4.5.1.1 AUX-004260	L/T	R	R	R
4-2	The conference system shall provide a service monitor and management system to actively monitor elements and critical components within the system.	3.4.5.1.1 AUX-004270	L/T	R	R	R
4-3	Report data shall be in a form that is capable of being managed by the Government network services applications and network elements, which are based on commercial and industry standards. The data transmitted shall comply with industry standard management protocols and/or data formats. Such industry standard protocols for data exchange include, but are not limited to, Syslog, Common Object Request Broker Architecture (CORBA), SNMPv3, Transaction Language 1 (TL1), Java 2 Platform Enterprise Edition (J2EE), and Extensible Markup Language (XML).	3.4.5.1.1 AUX-004280	L	R	R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
4-4	The conference system shall be responsible for managing and monitoring the following services and related resources. Furthermore, the system shall provide real-time, read-write continuous Network Management capabilities. The level of monitoring shall be sufficient to be able to track the status, through standard interfaces and protocols, of individual discrete hardware and software components used to deliver the service to enable visibility of individual incidents affecting the service delivery: <ul style="list-style-type: none"> <li>• Equipment and associated services.</li> <li>• Point-to-point and multipoint video services.</li> <li>• Audio services.</li> <li>• Reservation and scheduling.</li> <li>• Gateways and interfaces.</li> <li>• Conferencing center Web site.</li> <li>• Support systems.</li> </ul>	3.4.5.1.1 AUX-004290	L/T	R	R	R
4-5	The conference system shall furnish and maintain a service monitor and management system with external interfaces or feeds into Government management application and monitoring systems. These interfaces shall provide the Government the capability to monitor the performance and status of video and audio services. These interfaces shall provide for the importing and exporting of video and audio management services and monitoring information. The Government shall have real-time access to all video and audio services management and monitoring data collected and stored by the system. These interfaces are further specified in Section 2.19.1, General Management.	3.4.5.1.1 AUX-004300	L	R	R	R
4-6	The conference system shall support the use of Simple Network Management Protocol version 3 (SNMPv3) for system management and monitoring.	3.4.5.1.1 AUX-004310	L	R	R	R
4-7	The conference system shall support a 10/100-Mbps Ethernet physical interface to the DISA VVoIP Element Management System (EMS). The interface shall work in either of the two following modes using auto-negotiation: IEEE, Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995.	3.4.5.1.1 AUX-004320	L	R	R	R
4-8	If the system will be deployed as a DISN asset, then local management traffic and VVoIP EMS management traffic shall use separate physical Ethernet interfaces. Redundant VVoIP EMS physical Ethernet interfaces may be used but are not required. Redundant local management physical Ethernet interfaces may be used but are not required.	3.4.5.1.1 AUX-004330	L	C	C	C
4-9	The conference system shall provide fault management for services and resources. The system shall provide the Government with all the fault information needed electronically to effectively manage all conferencing services.	3.4.5.1.2 AUX-004340	L	R	R	R
4-10	The fault management system shall include the following minimum requirements:	3.4.5.1.2 AUX-004350	L	R	R	R
4-11	Power status shall be provided for individual shelf, rack, or controller units.	3.4.5.1.2 AUX-004350.a	L	R	R	R
4-12	Functional/Fault/Online/Offline status.	3.4.5.1.2 AUX-004350.b	L	R	R	R
4-13	Input/loss of input signal, or signal below working threshold.	3.4.5.1.2 AUX-004350.c	L	R	R	R
4-14	Output/loss of output signal.	3.4.5.1.2 AUX-004350.d	L	R	R	R
4-15	Input/output signal outside of specified range.	3.4.5.1.2 AUX-004350.e	L	R	R	R
4-16	Intrusion detected.	3.4.5.1.2 AUX-004350.f	L	R	R	R
4-17	The fault management function shall perform the following:	3.4.5.1.2 AUX-004360	N/A	R	R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
4-18	Detect and identify faults. The fault management service provided shall monitor dedicated and dial-up video services resources, audio conferencing service status, support services status, and conduct alarm surveillance, maintain error logs, and analyze monitored or logged errors or events to anticipate faults: <ul style="list-style-type: none"> <li>• Faults shall be detected within 10 seconds of their occurrence.</li> <li>• Faults shall be identified within 10 seconds of being detected.</li> <li>• Faults shall be correlated within 10 seconds of identification.</li> <li>• The Government shall be notified of service affecting faults within 10 seconds of fault correlation.</li> </ul>	3.4.5.1.2 AUX-004360.a	L	R	R	R
4-19	Isolate faults to include correlation of alarms. The fault management service provided shall initiate diagnostic testing and evaluate diagnostic results to determine the nature, severity, and specific cause(s) of the fault and isolate the fault to the video, audio, and support services at a component level.	3.4.5.1.2 AUX-004360.b	L	R	R	R
4-20	Temporary corrective action when a fault occurs. The fault management service provided shall reroute a conference call or service request to other hubs, circuits, or equipment in the case of a fault, including through the use of redundancy and failover capabilities.	3.4.5.1.2 AUX-004360.c	L	R	R	R
4-21	Correct faults. The fault management service provided shall implement corrective actions on faults to restore services to proper working order and complete resource/service restoration when the fault is with equipment or services provided by the conference system. This process shall incorporate backup and recovery capabilities to restore configurations and services to operational service.	3.4.5.1.2 AUX-004360.d	L	R	R	R
4-22	The conference system shall provide the Government with real-time monitoring of service-affecting events related to hardware and software components and subcomponents that compose the conference system. These service-affecting events impacting the scheduling and operation of system services include, but are not limited to, the following: <ul style="list-style-type: none"> <li>• Outages for all conferencing services elements to be exported into the existing trouble tracking system.</li> <li>• Any hazardous condition, as specified in DISA Circular 310-55-1, that may cause loss of service.</li> </ul>	3.4.5.1.3 AUX-004370	L	R	R	R
4-23	The conference system shall be able to update all service management thresholds, as required.	3.4.5.1.3 AUX-004380	L	R	R	R
4-24	The conference system shall maintain historical records of all fault alarm data and be able to export this data into the Government management application systems.	3.4.5.1.3 AUX-004390	L	R	R	R
4-25	The conference system shall provide a performance management system. The performance management system shall monitor and control all service performance and the quality of the services and features supporting the conference system. Performance management shall perform the following functions.	3.4.5.1.4 AUX-004400	L	O	O	O
4-26	Monitor, analyze, and characterize performance. The conference system shall monitor, analyze, and gather performance-related data to detect and characterize normal and degraded performance and be able to trend this data over time for metrics purposes. Section 3.4.4, Service Performance, defines normal performance requirements. The system shall provide notification if the service resources are being stressed with excess traffic loads.	3.4.5.1.4 AUX-004410	L	O	O	O
4-27	Tune and control performance in areas of control: the conference system shall activate controls to tune all services performance to restore degraded resources/services to acceptable performance levels. If control actions will cause any user service disturbance, then these actions shall be approved by the Government before execution.	3.4.5.1.4 AUX-004420	L	O	O	O

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
4-28	Maintain all services supporting the conference system through an operations database. The conference system shall maintain a database or be exportable to a Government network management tool supporting all conferencing services operational information, both real-time and historical, including, for example, traffic characterization data, performance data, and information on usage of resources/services. Historical records shall be kept of all performance data for a designated period of time.	3.4.5.1.4 AUX-004430	L	O	O	O
4-29	Evaluate performance of services and features. The conference system shall continuously assess and monitor the performance of all conferencing services and features, according to the performance parameters identified in Section 3.4.4, Service Performance, to ensure that the performance levels of Government services and features meet the specification requirements of Section 3.4.3, Service, and Section 3.4.4, Service Performance.	3.4.5.1.4 AUX-004440	L	O	O	O
4-30	The conference system shall provide notification of events, exceptions, or measures related to the performance of services' resources, and associated service-affecting conditions to Government platforms as required. Performance degradation notification shall include, at a minimum, the following. The conferencing services are composed of servers, applications and network services, appliances, and network devices responsible for supporting video services globally throughout the DoD community. The conferencing service is of a time-sensitive nature and one of the services that is being offered with the convergence of IP on the backbone. There are two parts to the network management of this service: transport monitoring and video stream monitoring. First, the underlying network elements and servers need to be included in fault management and performance management activities at the physical layers and IP layers. Second, the video service needs to have instrumentation included that would be able to monitor the conferencing user's experience, in order to isolate problems and reveal if the video service is meeting specific service-level requirements.	3.4.5.1.5 AUX-004450	L	O	O	O
4-31	The performance management toolset should be able to collect information from the video device managers. The information it should collect would include, but not be limited to, the number of participants, duration of a session, video burst measurements, and capacity measurements.	3.4.5.1.5 AUX-004460	L		O	O
4-32	Certification and Accreditation. The conference system shall ensure that all systems and subsystems in UC conferencing undergo Certification and Accreditation (C&A) IAW the DoD Information Assurance Certification and Accreditation Process (DIACAP) and associated audits.	3.4.5.1.6 AUX-004470	IA	O	O	O
4-33	Best Security Practices. The conference system shall incorporate best security practices such as single sign-on, public key encryption (PKE), smart card, and biometrics in system security design of DoD information, but does not limit to certain security mechanisms.	3.4.5.1.6 AUX-004480	IA	O	O	O
4-34	Enterprise Security Management. The conference system shall implement enterprise management of security devices and applications such as the following: <ul style="list-style-type: none"> <li>• Firewalls and boundary protection.</li> <li>• Intrusion detection systems.</li> <li>• Operating systems, network devices, and applications security.</li> <li>• Vulnerability management.</li> </ul>	3.4.5.1.6 AUX-004490	IA	O	O	O
4-35	Security Configuration Specifications. The conference system shall comply with DoD reference documents such as STIGs or security recommendation guides from the DISA Facility Security Officer (FSO) that are pertinent to the UCCS or subcomponents.	3.4.5.1.6 AUX-004500	IA	O	O	O
4-36	The system shall provide an online directory service to support scheduling that shall include general information about all registered DoD video and audio users including, but not limited to, a user's point of contact, location, supported data rates, organization name, unit capabilities, and software versions.	3.4.5.2 AUX-004510	L/T	O	R	O

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/ TP ID</b>	<b>AO CS</b>	<b>VO CS</b>	<b>A/V CS</b>
4-37	The system shall update the online directory within 24 hours of learning of a new EI receiving service, a change in an existing EI's service status, or notification by DISA of any other change with regard to an EI.	3.4.5.2 AUX-004520	L	O	R	O
4-38	The system shall provide a secure Web interface that implements a public key enablement application, allowing registered users with a valid DoD PKI or External Certification Authority (ECA) certificate and Internet connectivity to access the online directory.	3.4.5.2 AUX-004530	L/T	O	R	O
4-39	The online directory shall support more than 1 million data entries and shall support more than 500 concurrent users at the same time.	3.4.5.2 AUX-004540	L	O	R	O
4-40	The online directory shall be Web-based using modern and open technologies and provide interfaces, such as XML, Simple Object Access Protocol (SOAP), Web Service Description Language (WSDL), and Universal Discovery Description Interface (UDDI), allowing external data sources from others to perform directory lookups, queries, and updates as specified by "Horizontal Fusion Standards and Specifications," 3 November 2004, and DoD Joint Technical Architecture, Version 6, Volumes I and II, 3 October 2003.	3.4.5.2 AUX-004550	L/T	O	R	O
4-41	The online directory shall support the discovery service that provides processes for discovery of information content or services that exploit metadata descriptions of information technology resources stored in directories, registries, and catalogs (to include search engines) as specified by "Horizontal Fusion Standards and Specifications," 3 November 2004, and "DoD Joint Technical Architecture," Version 6, Volumes I and II, 3 October 2003. Directory services shall be designed to meet Protected Personal Information/Personally Identifiable Information protection requirements and Privacy Act requirements.	3.4.5.2 AUX-004560	L	O	R	O
4-42	The system shall provide an online directory service to allow authorized registered users to search system-wide for other authorized, registered service users in the directory and/or to update data entries.	3.4.5.2 AUX-004570	T	O	R	O
4-43	The conference system shall include an automated registration system that current and prospective subscribers can access online using a standard Web browser.	3.4.5.3 AUX-004580	L/T	R	R	R
4-44	All data sent and received by the automated registration system shall be encrypted using Secure Socket Layer (SSL)/TLS technology, as a minimum, with DoD PKI certificate authentication and validation.	3.4.5.3 AUX-004590	IA	R	R	R
4-45	The automated registration system shall collect all data necessary to comply with provisioning requirements.	3.4.5.3 AUX-004600	L/T	R	R	R
4-46	The automated registration system shall collect all data necessary for connection approval by the program office.	3.4.5.3 AUX-004610	L/T	R	R	R
4-47	The automated registration system shall collect all data necessary to authorize users for access to operational systems including scheduling and reservations.	3.4.5.3 AUX-004620	L/T	R	R	R
4-48	The automated registration system shall collect all data necessary to support the operational requirements of UC conferencing services. Upon connection approval and completion of verification and interoperability tests, all data shall be available to the operational systems for the scheduling and activation of conferences.	3.4.5.3 AUX-004630	L/T	R	R	R
4-49	The automated registration system shall be the authoritative source of conference subscribers' data. The system shall provide the necessary tools to allow authorized users to maintain and update user and endpoint information.	3.4.5.3 AUX-004640	L/T	R	R	R
4-50	The automated registration system shall support Privacy Act statements as required by Information Assurance policy.	3.4.5.3 AUX-004650	IA	R	R	R
4-51	The primary component for requesting a conference shall be by an automated scheduling system that authorized users can access online using a standard Web browser. All data shall be encrypted using SSL/TLS technology, as a minimum, with DoD PKI certificate authentication and validation.	3.4.5.4 AUX-004660	L/T/IA	R	R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
4-52	The automated scheduling system shall resolve the availability of all requested participants and conferencing resources. The customer shall be able to schedule a conference immediately or schedule it for some time in the future. Immediate start requests for conferences shall be activated in less than 5 minutes of confirmation of available resources and participants. The system shall support scheduling conferences with the conference size larger than the number of initially invited or scheduled endpoints. A unique identification number shall be assigned to each conference event to facilitate conference control and management. E-mail notifications shall be provided to all conference participants to facilitate event coordination. E-mail content shall be editable as a configuration feature. The system shall support encrypted e-mail for notifications. Scheduled conference information also shall be available through the scheduling Web interface.	3.4.5.4 AUX-004670	L/T	R	R	R
4-53	The original requester shall serve as the conference manager for all conferences. The system shall have the capability to assign other system users to act as surrogates for the original requester. This may include peers and/or workflow superiors. Conference management and control shall include the ability to make changes to a scheduled or active conference. Supported changes to active conferences shall include, but not be limited to, the addition or deletion of participants, early termination of a conference, and extension of the conference beyond the originally scheduled end time. Additions, deletions, and extensions of conferences shall occur without interruption to the existing conference, other than preemptive precedence calls. Additions and extensions to conferences shall be executed in less than 5 minutes of confirmation of available resources. Deletions and terminations of conferences shall be executed in less than 5 minutes of the request by the conference manager. The system shall support scheduling of recurring conferences.	3.4.5.4 AUX-004680	L/T	R	R	R
4-54	The conference system shall provide an Accounting Management function. The Accounting Management function shall do the following:	3.4.5.5 AUX-004690	L/T	R	R	R
4-55	Provide accounting information to the Government regarding all conferencing services provided.	3.4.5.5 AUX-004690.a	L/T	R	R	R
4-56	Provide all conferencing services data to the Government at a level of detail that allows the Government to bill conferencing services customers based on usage.	3.4.5.5 AUX-004690.b	L	R	R	R
4-57	The Accounting Management function shall enable charges to be established for the use of dedicated and dial-up conferencing services resources.	3.4.5.5 AUX-004700	L	R	R	R
4-58	The conference system Accounting Management function shall provide for the collection, aggregation, storage, and reporting of all conferencing service usage data. The accounting management function shall consist of systems to activate and monitor customer accounts and to collect, aggregate, and report on usage data.	3.4.5.5 AUX-004710	L/T	R	R	R
4-59	The conference system shall provide the capability to perform the following Accounting Management functions:	3.4.5.5 AUX-004720	L/T	R	R	R
4-60	Collect usage data from a Call Detailed Record (CDR) (i.e., at the level of the authorization code).	3.4.5.5 AUX-004720.a	L/T	R	R	R
4-61	Aggregate and combine data for generating reports as specified by the Government.	3.4.5.5 AUX-004720.b	L	R	R	R
4-62	conference records per individual customer's account.	3.4.5.5 AUX-004720.c	L/T	R	R	R
4-63	Ensure continuous (24x7) monitoring, processing, and recording for all video services-related events and customer activity data.	3.4.5.5 AUX-004720.d	L	R	R	R
4-64	Maintain a database of various conference reports per individual customer account, including conference detail summary, completion summary, and exception reports.	3.4.5.5 AUX-004720.e	L	R	R	R
4-65	Archive data for possible later retrieval by the Government (e.g., in response to customer inquiries or to audit the data).	3.4.5.5 AUX-004720.f	L	R	R	R
4-66	The conference system shall provide the capability to transmit usage data to the EMS. The frequency of data transfer shall be determined by the Government based on volume of data collected. The system shall maintain all accounting data for at least one billing cycle.	3.4.5.5 AUX-004730	L	R	R	R

**Table 3-5. UCCS Capability/Functional Requirements (continued)**

<b>LEGEND:</b>			
4FCIF	4 Full Common Intermediate Format (also called 4CIF)	Mbps	Megabits per second
5ESS	Class 5 Electronic Switching System	MCU	Multipoint Conferencing Unit
16FCIF	16 Full Common Intermediate Format (also called 16 Common Intermediate Format (16CIF))	MLPP	Multi-level Precedence and Preemption
A/V	Audio/Visual	MOS	Mean Opinion Score
AO	Audio Only	ms	millisecond
AS-SIP	Assured Services Session Initiation Protocol	MTU	Maximum Transmission Unit
AVT	Audio and Video Transport	NA	Not Applicable
C	Conditional	O	Optional
CAS	Channel Associated Signaling	PKI	Public Key Infrastructure
CTU	Conferencing Terminal Unit	PSTN	Public Switched Telephone Network
CS	Conference System	PRI	Primary Rate Interface
DISA	Defense Information Systems Agency	QCIF	Quarter Common Intermediate Format
DISN	Defense Information Systems Network	QoS	Quality of Service
DMS	Digital Multiplex System	R	Required
DoD	Department of Defense	RFC	Request For Comments
DNS	Domain Name Service	SIP	Session Initiation Protocol
DSCP	Differentiated Services Code Point	SNMPv3	Simple Network Management Protocol version 3
DTMF	Dual Tone Multi-Frequency	SQCIF	Sub-Quarter Common Intermediate Format
E1	European Basic Multiplex Rate	SRTCP	Secure Real-Time Transport Control Protocol
E2E	End-to-End	SRTP	Secure Real-Time Transport Protocol
EI	End Instrument	SSL	Secure Sockets Layer
EMS	Element Management System	SVC	Scalable Video Codec
FCIF	Common Intermediate Format (also called CIF)	SVGA	Super Video Graphics Array
FECC	Far End Camera Control	T	Testable Item
HD	High Definition	T1	Digital Transmission Link Level 1
IA	Information Assurance	TLS	Transport Layer Security
IAW	in accordance with	TP	Test Plan
ID	Identification	UC	Unified Capabilities
IEEE	Institute of Electrical and Electronics Engineers	UCCS	Unified Capabilities Conference System
IP	Internet Protocol	UCR	Unified Capabilities Requirements
IPv4	Internet Protocol version 4	URL	Uniform Resource Locator
IPv6	Internet Protocol version 6	VGA	Video Graphics Array
ISDN	Integrated Services Digital Network	VO	Video Only
ITU	International Telecommunication Union	VoIP	Voice over Internet Protocol
kbps	kilobits per second	VTC	video teleconferencing
L	LoC Item	VVoIP	Voice and Video over Internet Protocol
LoC	Letter(s) of Compliance	WSXGA+	Widescreen Super Extended Graphics Array +
		XGA	Extended Graphics Array

**Table 3-6. IPv6 Requirements**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2013)</b>	<b>LoC/TP ID</b>	<b>UCCS</b>
<b>5.2 – IPv6 Requirements</b>				
v6-1	The product shall support dual IPv4 and IPv6 stacks as described in RFC 4213.	5.2.1 IP6-000010	L	R
v6-2	Dual-stack end points or Call Connection Agents (CCAs) shall be configured to choose IPv4 over IPv6.	5.2.1 IP6-000020	L	R
v6-3	All nodes and interfaces that are “IPv6-capable” must be carefully configured and verified that the IPv6 stack is disabled until it is deliberately enabled as part of a deliberate transition strategy. This includes the stateless autoconfiguration of link-local addresses. Nodes with multiple network interfaces may need to be separately configured per interface.	5.2.1 IP6-000030	L	R
v6-4	The system shall provide the same (or equivalent) functionality in IPv6 as in IPv4 consistent with the requirements in the UCR for its Approved Products List (APL) category. NOTE: This requirement applies only to products that are required to perform IPv6 functionality.	5.2.1 IP6-000050	L	R
v6-5	The product shall support the IPv6 format as described in RFC 2460 and updated by RFC 5095.	5.2.1 IP6-000060	L	R

**Table 3-6. IPv6 Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/TP ID	UCCS
<b>5.2 – IPv6 Requirements (continued)</b>				
v6-6	The product shall support the transmission of IPv6 packets over Ethernet networks using the frame format defined in RFC 2464. NOTE: This requirement does not mandate that the remaining sections of RFC 2464 have to be implemented.	5.2.1 IP6-000070	L	R
v6-7	The product shall support a minimum MTU of 1280 bytes as described in RFC 2460 and updated by RFC 5095.	5.2.1.1 IP6-000090	L	R
v6-8	If Path MTU Discovery is used and a “Packet Too Big” message is received requesting a next-hop MTU that is less than the IPv6 minimum link MTU, then the product shall ignore the request for the smaller MTU and shall include a fragment header in the packet.	5.2.1.1 IP6-000100	L	C
v6-9	The product shall not use the Flow Label field as described in RFC 2460.	5.2.1.2 IP6-000110	L	R
v6-10	The product shall be capable of setting the Flow Label field to zero when originating a packet.	5.2.1.2 IP6-000120	L	R
v6-11	The product shall be capable of ignoring the Flow Label field when receiving packets.	5.2.1.2 IP6-000140	L	R
v6-12	The product shall support the IPv6 Addressing Architecture as described in RFC 4291.	5.2.1.3 IP6-000150	L	R
v6-13	The product shall support the IPv6 Scoped Address Architecture as described in RFC 4007.	5.2.1.3 IP6-000160	L	R
v6-14	If a scoped address (RFC 4007) is used, then the product shall use a scope index value of zero when the default zone is intended.	5.2.1.3 IP6-000170	L	C
v6-15	If Dynamic Host Configuration Protocol (DHCP) is supported within an IPv6 environment, then it shall be implemented in accordance with the DHCP for IPv6 (DHCPv6) as described in RFC 3315.	5.2.1.4 IP6-000180	L	C
v6-16	If the product is a DHCPv6 client, then the product shall discard any messages that contain options that are not allowed to appear in the received message type (e.g., an Identity Association option in an Information-Request message).	5.2.1.4 IP6-000200	L	C
v6-17	If the product is a DHCPv6 client and the first retransmission timeout has elapsed since the client sent the Solicit message and the client has received an Advertise message(s), but the Advertise message(s) does not have a preference value of 255, then the client shall continue with a client-initiated message exchange by sending a Request message.	5.2.1.4 IP6-000220	L	C
v6-18	If the product is a DHCPv6 client and the DHCPv6 solicitation message exchange fails, then it shall restart the reconfiguration process after receiving user input, system restart, attachment to a new link, a system configurable timer, or a user defined external event occurs.	5.2.1.4 IP6-000230	L	C
v6-19	If the product is a DHCPv6 client and it sends an Information-Request message, then it shall include a Client Identifier option to allow it to be authenticated to the DHCPv6 server.	5.2.1.4 IP6-000240	L	C
v6-20	If the product is a DHCPv6 client, then it shall perform duplicate address detection upon receipt of an address from the DHCPv6 server before transmitting packets using that address for itself.	5.2.1.4 IP6-000250	L	C
v6-21	If the product is a DHCPv6 client, then it shall log all reconfigure events. NOTE: Some systems may not be able to log all this information (e.g., the system may not have access to this information).	5.2.1.4 IP6-000260	L	C
v6-22	If the product supports DHCPv6 and uses authentication, then it shall discard unauthenticated DHCPv6 messages from UC products and log the event.	5.2.1.4 IP6-000270	L	C
v6-23	The product shall support Neighbor Discovery for IPv6 as described in RFC 4861.	5.2.1.5 IP6-000280	L	R
v6-24	The product shall not set the override flag bit in the Neighbor Advertisement message for solicited advertisements for any cast addresses or solicited proxy advertisements.	5.2.1.5 IP6-000300	L	R
v6-25	When a valid “Neighbor Advertisement” message is received by the product and the product neighbor cache does not contain the target’s entry, the advertisement shall be silently discarded.	5.2.1.5 IP6-000310	L	R
v6-26	When a valid “Neighbor Advertisement” message is received by the product and the product neighbor cache entry is in the INCOMPLETE state when the advertisement is received and the link layer has addresses and no target link-layer option is included, the product shall silently discard the received advertisement.	5.2.1.5 IP6-000320	L	R

**Table 3-6. IPv6 Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/TP ID	UCCS
<b>5.2 – IPv6 Requirements (continued)</b>				
v6-27	When address resolution fails on a neighboring address, the entry shall be deleted from the product’s neighbor cache.	5.2.1.5 IP6-000330	L	R
v6-28	The product shall support the ability to configure the product to ignore Redirect messages.	5.2.1.5.1 IP6-000340	L	R
v6-29	The product shall only accept Redirect messages from the same router as is currently being used for that destination.	5.2.1.5.1 IP6-000350	L	R
v6-30	If “Redirect” messages are allowed, then the product shall update its destination cache in accordance with the validated Redirect message.	5.2.1.5.1 IP6-000360	L	C
v6-31	If the valid “Redirect” message is allowed and no entry exists in the destination cache, then the product shall create an entry.	5.2.1.5.1 IP6-000370	L	C
v6-32	If redirects are supported, then the device shall support the ability to disable this functionality.	5.2.1.5.1 IP6-000380	L	C
v6-33	The product shall prefer routers that are reachable over routers whose reachability is suspect or unknown.	5.2.1.5.2 IP6-000400	L	R
v6-34	If the product supports stateless IP address autoconfiguration including those provided for the commercial market, then the product shall support IPv6 Stateless Address Autoconfiguration (SLAAC) for interfaces supporting UC functions in accordance with RFC 4862.	5.2.1.6 IP6-000420	L	C
v6-35	If the product supports IPv6 SLAAC, then the product shall have a configurable parameter that allows the function to be enabled and disabled. Specifically, the product shall have a configurable parameter that allows the “managed address configuration” flag and the “other stateful configuration” flag to always be set and not perform stateless autoconfiguration.	5.2.1.6 IP6-000430	L	C
v6-36	If the product supports IPv6 SLAAC, then the product shall have the configurable parameter set not to perform stateless autoconfiguration.	5.2.1.6 IP6-000440	L	C
v6-37	While nodes are not required to autoconfigure their addresses using SLAAC, all IPv6 Nodes shall support link-local address configuration and Duplicate Address Detection (DAD) as specified in RFC 4862. In accordance with RFC 4862, DAD shall be implemented and shall be on by default. Exceptions to the use of DAD are noted in the following text.	5.2.1.6 IP6-000450	L	R
v6-38	A node MUST allow for autoconfiguration-related variable to be configured by system management for each multicast-capable interface to include DupAddrDetectTransmits where a value of zero indicates that DAD is not performed on tentative addresses as specified in RFC 4862.	5.2.1.6 IP6-000460	L	R
v6-39	The product shall support manual assignment of IPv6 addresses.	5.2.1.6 IP6-000470	L	R
v6-40	The product shall support the Internet Control Message Protocol (ICMP) for IPv6 as described in RFC 4443.	5.2.1.7 IP6-000520	L	R
v6-41	The product shall support the capability to enable or disable the ability of the product to generate a Destination Unreachable message in response to a packet that cannot be delivered to its destination for reasons other than congestion.	5.2.1.7 IP6-000540	L	R
v6-42	The product shall support the enabling or disabling of the ability to send an Echo Reply message in response to an Echo Request message sent to an IPv6 multicast or anycast address.	5.2.1.7 IP6-000550	L	R
v6-43	The product shall validate ICMPv6 messages, using the information contained in the payload, before acting on them.	5.2.1.7 IP6-000560	L	R
v6-44	The product shall support MLD as described in RFC 2710.	5.2.1.8 IP6-000680	L	R
v6-45	If the product uses IPsec, then the product shall be compatible with the Security Architecture for the IPsec described in RFC 4301.	5.2.1.9 IP6-000690	L	C
v6-46	If RFC 4301 is supported, then the product shall not support the mixing of IPv4 and IPv6 in a SA.	5.2.1.9 IP6-000700	L	C
v6-47	If RFC 4301 is supported, then the product’s security association database (SAD) cache shall have a method to uniquely identify a SAD entry.	5.2.1.9 IP6-000710	L	C
v6-48	If RFC 4301 is supported, then the product shall implement IPsec to operate with both integrity and confidentiality.	5.2.1.9 IP6-000720	L	C
v6-49	If RFC 4301 is supported, then the product shall be capable of enabling and disabling the ability of the product to send an ICMP message informing the sender that an outbound packet was discarded.	5.2.1.9 IP6-000730	L	C

**Table 3-6. IPv6 Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/TP ID	UCCS
<b>5.2 – IPv6 Requirements (continued)</b>				
v6-50	If an ICMP outbound packet message is allowed, then the product shall be capable of rate limiting the transmission of ICMP responses.	5.2.1.9 IP6-000740	L	C
v6-51	If RFC 4301 is supported, then the system’s Security Policy Database (SPD) shall have a nominal, final entry that discards anything unmatched.	5.2.1.9 IP6-000750	L	C
v6-52	If RFC 4301 is supported, and the product receives a packet that does not match any SPD cache entries, and the product determines it should be discarded, then the product shall log the event and include the date/time, Security Parameter Index (SPI) if available, IPSec protocol if available, source and destination of the packet, and any other selector values of the packet.	5.2.1.9 IP6-000760	L	C
v6-53	If RFC 4301 is supported, then the product should include a management control to allow an administrator to enable or disable the ability of the product to send an IKE notification of an INVALID_SELECTORS.	5.2.1.9 IP6-000770	L	C
v6-54	If RFC 4301 is supported, then the product shall support the ESP Protocol in accordance with RFC 4303.	5.2.1.9 IP6-000780	L	C
v6-55	If RFC 4303 is supported, then the product shall be capable of enabling anti-replay.	5.2.1.9 IP6-000790	L	C
v6-56	If RFC 4303 is supported, then the product shall check, as its first check, after a packet has been matched to its SA whether the packet contains a sequence number that does not duplicate the sequence number of any other packet received during the life of the security association.	5.2.1.9 IP6-000800	L	C
v6-57	If RFC 4301 is supported, then the product shall support IKEv1 as defined in RFC 2409.	5.2.1.9 IP6-000810	L	C
v6-58	To prevent a Denial of Services (DoS) attack on the initiator of an IKE_SA, the initiator shall accept multiple responses to its first message, treat each as potentially legitimate, respond to it, and then discard all the invalid half-open connections when it receives a valid cryptographically protected response to any one of its requests. Once a cryptographically valid response is received, all subsequent responses shall be ignored whether or not they are cryptographically valid.	5.2.1.9 IP6-000820	L	C
v6-59	If RFC 4301 is supported, then the product shall support extensions to the Internet IP Security Domain of Interpretation for the Internet Security Association and Key Management Protocol (ISAKMP) as defined in RFC 2407.	5.2.1.9 IP6-000830	L	C
v6-60	If RFC 4301 is supported, then the product shall support the ISAKMP as defined in RFC 2408.	5.2.1.9 IP6-000840	L	C
v6-61	If the product supports the IPSec Authentication Header Mode, then the product shall support the IP Authentication Header (AH) as defined in RFC 4302.	5.2.1.9 IP6-000850	L	C
v6-62	If RFC 4301 is supported, then the product shall support manual keying of IPSec.	5.2.1.9 IP6-000860	L	C
v6-63	If RFC 4301 is supported, then the product shall support the ESP and AH cryptographic algorithm implementation requirements as defined RFC 4835.	5.2.1.9 IP6-000870	L	C
v6-64	If RFC 4301 is supported, then the product shall support the IKEv1 security algorithms as defined in RFC 4109.	5.2.1.9 IP6-000880	L	C
v6-65	If the product uses Uniform Resource Identifiers (URIs) in combination with IPv6, then the product shall use the URI syntax described in RFC 3986.	5.2.1.10 IP6-000990	L	C
v6-66	If the product uses the Domain Name Service (DNS) resolver for IPv6 based queries, then the product shall conform to RFC 3596 for DNS queries.	5.2.1.10 IP6-001000	L	C
v6-67	For traffic engineering purposes, the bandwidth required per voice subscriber is calculated to be 110.0 kbps (each direction) for each IPv6 call. This is based on G.711 (20 ms codec) with IP overhead (100 kbps) resulting in a 250-byte bearer packet plus 10 kbps for signaling, Ethernet Interframe Gap, and the Secure Real-Time Transport Control Protocol (SRTCP) overhead. Based on overhead bits included in the bandwidth calculations, vendor implementations may use different calculations and hence arrive at slightly different numbers.	5.2.1.11 IP6-001010	L	R
v6-68	The product shall forward packets using the same IP version as the version in the received packet.	5.2.1.12 IP6-001040	L	R
v6-69	When the product is establishing media streams from dual-stacked appliances for AS-SIP signaled sessions, the product shall use the Alternative Network Address Type (ANAT) semantics for the Session Description Protocol (SDP) in accordance with RFC 4091.	5.2.1.12 IP6-001050	L	R

**Table 3-6. IPv6 Requirements (continued)**

ID	Requirement	UCR Ref (UCR 2013)	LoC/TP ID	UCCS	
<b>5.2 – IPv6 Requirements (continued)</b>					
v6-70	<p>If the product is using AS-SIP, and the &lt;addrtype&gt; is IPv6, and the &lt;connection-address&gt; is a unicast address, then the product shall support generation and processing of unicast IPv6 addresses having the following formats:</p> <ul style="list-style-type: none"> <li>• x:x:x:x:x:x:x:x (where x is the hexadecimal values of the eight 16-bit pieces of the address). Example: 1080:0:0:0:8:800:200C:417A.</li> <li>• x:x:x:x:x:d.d.d.d (where x is the hexadecimal values of the six high-order 16-bit pieces of the address, and d is the decimal values of the four low-order 8-bit pieces of the address (standard IPv4 representation). For example, 1080:0:0:0:8:800:116.23.135.22.</li> </ul>	5.2.1.13 IP6-001060	L	C	
v6-71	<p>If the product is using AS-SIP, then the product shall support the generation and processing of IPv6 unicast addresses using compressed zeros consistent with one of the following formats:</p> <ul style="list-style-type: none"> <li>• x:x:x:x:x:x:x:x format: 1080:0:0:0:8:800:200C:417A.</li> <li>• x:x:x:x:x:d.d.d.d format: 1080:0:0:0:8:800:116.23.135.22.</li> <li>• compressed zeros: 1080::8:800:200C:417A.</li> </ul>	5.2.1.13 IP6-001070	L	C	
v6-72	<p>If the product is using AS-SIP, and the &lt;addrtype&gt; is IPv6, and the &lt;connection-address&gt; is a multicast group address (i.e., the two most significant hexadecimal digits are FF), then the product shall support the generation and processing of multicast IPv6 addresses having the same formats as the unicast IPv6 addresses.</p>	5.2.1.13 IP6-001080	L	C	
v6-73	<p>If the product is using AS-SIP, and the &lt;addrtype&gt; is IPv6, then the product shall support the use of RFC 4566 for IPv6 in SDP as described in AS-SIP 2013, Section 4, SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs.</p>	5.2.1.13 IP6-001090	L	C	
v6-74	<p>If the product is using AS-SIP, and the &lt;addrtype&gt; is IPv6, and the &lt;connection-address&gt; is an IPv6 multicast group address, then the multicast connection address shall not have a Time To Live (TTL) value appended to the address as IPv6 multicast does not use TTL scoping.</p>	5.2.1.13 IP6-001100	L	C	
v6-75	<p>If the product is using AS-SIP, then the product shall support the processing of IPv6 multicast group addresses having the &lt;number of address&gt; field and may support generating the &lt;number of address&gt; field. This field has the identical format and operation as the IPv4 multicast group addresses.</p>	5.2.1.13 IP6-001110	L	C	
v6-76	<p>The products shall support Differentiated Services as described in RFC 2474 for a voice and video stream in accordance with Section 2, Session Control Products, and Section 6, Network Infrastructure End-to-End Performance, plain text DSCP plan.</p>	5.2.1.14 IP6-001150	L	R	
v6-77	<p>If the product acts as an IPv6 tunnel broker, then the product shall support the function as defined in RFC 3053.</p>	5.2.1.14 IP6-001160	L	C	
v6-78	<p>The UCCS must be IPv6-capable. Use guidance in Table 5.2-4 for NA/SS.</p>	Table 5.2-1	L	R	
v6-78 (continued)	RFC 2407	The Internet IP Security Domain of Interpretation for ISAKMP	Table 5.2-4	L	C
	RFC 2408	Internet Security Association and Key Management Protocol (ISAKMP)	Table 5.2-4	L	C
	RFC 2409	The Internet Key Exchange (IKE)	Table 5.2-4	L	C
	RFC 2460	Internet Protocol, Version 6 (IPv6) Specification	Table 5.2-4	L	R-2
	RFC 2464	Transmission of IPv6 Packets over Ethernet Networks	Table 5.2-4	L	R-3
	RFC 2474	Definition of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers	Table 5.2-4	L	R-4
	RFC 2710	Multicast Listener Discovery (MLD) for IPv6	Table 5.2-4	L	R-8
	RFC 3053	IPv6 Tunnel Broker	Table 5.2-4	L	C
	RFC 3315	Dynamic Host Configuration Protocol for IPv6 (DHCPv6)	Table 5.2-4	L	C
	RFC 3596	DNS Extensions to Support IPv6	Table 5.2-4	L	C
	RFC 3986	Uniform Resource Identifier (URI): Generic Syntax	Table 5.2-4	L	C
	RFC 4007	IPv6 Scoped Address Architecture	Table 5.2-4	L	R
RFC 4091	The Alternative Network Address Types (ANAT) Semantics for the Session Description Protocol (SDP) Grouping Framework	Table 5.2-4	L	R	

**Table 3-6. IPv6 Requirements (continued)**

ID	Requirement		UCR Ref (UCR 2013)	LoC/TP ID	UCCS
<b>5.2 – IPv6 Requirements (continued)</b>					
v6-78 (contin- ued)	RFC 4092	Usage of the Session Description Protocol (SDP) Alternative Network Address Types (ANAT) Semantics in the Session Initiation Protocol (SIP)	Table 5.2-4	L	R
	RFC 4109	Algorithms for Internet Key Exchange Version 1 (IKEv1)	Table 5.2-4	L	C
	RFC 4213	Basic Transition Mechanisms for IPv6 Hosts and Routers	Table 5.2-4	L	R-1
	RFC 4291	IP Version 6 Addressing Architecture	Table 5.2-4	L	R
	RFC 4301	Security Architecture for the Internet Protocol	Table 5.2-4	L	C
	RFC 4302	IP Authentication Header	Table 5.2-4	L	C
	RFC 4303	IP Encapsulating Security Payload (ESP)	Table 5.2-4	L	C
	RFC 4443	Internet Control Message Protocol (ICMPv6) for the Internet Protocol Version 6 (IPv6) Specification	Table 5.2-4	L	R
	RFC 4566	SDP: Session Description Protocol	Table 5.2-4	L	C
	RFC 4835	Cryptographic Algorithm Implementation Requirements for Encapsulating Security Payload (ESP) and Authentication Header (AH)	Table 5.2-4	L	C
	RFC 4861	Neighbor Discovery for IP Version 6 (IPv6)	Table 5.2-4	L	R
	RFC 4862	IPv6 Stateless Address Autoconfiguration	Table 5.2-4	L	C
RFC 5095	Deprecation of Type 0 Routing Headers in IPv6	Table 5.2-4	L	R	
<b>LEGEND:</b>					
AH	Authentication Header	kbps	kilobits per second		
ANAT	Alternative Network Address Type	L	LoC Item		
AS-SIP	Assured Services Session Initiation Protocol	LoC	Letter(s) of Compliance		
C	Conditional	MLD	Multicast Listener Discovery		
DAD	Duplicate Address Detection	ms	millisecond		
DHCP	Dynamic Host Configuration Protocol	MTU	Maximum Transmission Unit		
DHCPv6	Dynamic Host Configuration Protocol for IPv6	NA/SS	Network Appliance/Simple Server		
DNS	Domain Name Service	R	Required		
DSCP	Differentiated Services Code Point	RFC	Request for Comments		
EI	End Instrument	SA	Security Architecture		
ESP	Encapsulating Security Protocol	SAD	Security Association Database		
ID	Identification	SDP	Session Description Protocol		
ICMP	Internet Control Message Protocol	SIP	Session Initiation Protocol		
ICMPv6	Internet Control Message Protocol for IPv6	SLAAC	Stateless Address Autoconfiguration		
IP	Internet Protocol	SPD	Security Policy Database		
IPSec	Internet Protocol Security	TTL	Time To Live		
IPv4	Internet Protocol version 4	UC	Unified Capabilities		
IPv6	Internet Protocol version 6	UCCS	Unified Capabilities Conferencing System		
ISAKMP	Internet Security Association and Key Management Protocol	UCR	Unified Capabilities Requirements		
		URI	Uniform Resource Identifiers		