DEFENSE INFORMATION SYSTEMS AGENCY
P. O. BOX 549
FORT MEADE, MARYLAND 20755-0549

IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **29 May 14**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Joint Interoperability Certification of the Forum Communications Consortium II with Software Release 9.1

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.** Forum Communications Consortium II with Software Release 9.1; 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 an Audio Only (AO) Unified Capabilities Conference System (UCCS) with the conditions described in Table 1. The SUT was tested with and is only certified for use with Assured Services Session Initiation Protocol (AS-SIP) directly connected to the following Local Session Controllers (LSCs): REDCOM High Density Exchange (HDX) and SLICE and the Avaya AS5300. The SUT is also certified with AS-SIP routed via the Avaya AS5300 Softswitch (SS) to all other LSCs listed on the UC Approved Products List (APL). The SUT is also certified for use with any Session Controller (SC), SS, Multifunction Switch (MFS), End Office (EO), Small End Office (SMEO), Private Branch Exchange (PBX)1 or PBX2 that is or was previously on the UC APL certified with a Primary Rate Interface (PRI) Digital Transmission Link Level 1 (T1) American National Standards Institute (ANSI) T1.619a or National Integrated Services Digital Network (ISDN) 2 (NI2) (ANSI T1.619a OR ANSI T1.607) interface. This certification expires upon changes that affect interoperability, but no later than three years from the date of this memorandum.

Table 1. Conditions

Condition	Operational Impact	Remarks
UCR Waivers		
None.		

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Conditions of Fielding		
None.		
Open Test Discrepancies		
The SUT does not support G.722, G.722.1, G.723.1 or G.728 audio standards.	Minor	See note 1.
The SUT does not meet Network Equipment Building Systems-3 (NEBS-3) Requirements. Per the vendor's Letter of Compliance, the SUT chassis does not have NEBS-3 Certification but is classified as a carrier grade industrial PC chassis that has some hardening to provide robustness to vibration, thermal overload, and electrical failover.	None	See note 2.
The SUT does not fully support the call transfer feature. The SUT does not support call transfer interaction with EIs that require the REFER method for transfer.	Minor	See note 1.
The SUT does not support Simple Network Transport Protocol (SNMP) v3.	Minor	See note 1.
The SUT does not reroute a conference call when a fault occurs	None	See note 2.
The SUT does not fully support automated scheduling in accordance with requirement AUX-004670.	Minor	See note 1.

NOTES:
1. DISA has accepted the vendor's POA&M and adjudicated this as minor.
2. DISA adjudicated this as minor and stated the intent to remove this requirement.

LEGEND:

DISA	Defense Information Systems Agency	POA&M	Plan of Action and Milestones
EI	End Instrument	SUT	System Under Test
PC	Personal Computer	UCR	Unified Capabilities Requirements

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 the UC APL product summary.

Table 2. SUT Interface Status

Interface	Threshold CR/FR Requirements (See note 1.)	Status	Remarks
Interfaces			
Ethernet 10 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 100 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 1000 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 10 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
Ethernet 100 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
Ethernet 1000 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
PRI T1 NI2 (ANSI T1.619a/T1 6.07) (C)	1	Met	
PRI T1 (H.320) (C)	1	Not Tested	This interface is only for a Video UCCS.
PRI E1 (H.320) (C)	1	Not Tested	This interface is only for a Video UCCS.

NOTES:
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.

Table 2. SUT Interface Status (continued)

LEGEND:			
ANSI	American National Standards Institute	MLPP	Multi-Level Precedence and Preemption
AS-SIP	Assured Services Session Initiation Protocol	PRI	Primary Rate Interface
C	Conditional	R	Required
CR	Capability Requirement	SS7	Signaling System 7
DSS1	Digital Subscriber Signaling 1	SUT	System Under Test
E1	European Basic Multiplex Rate (2.048 Mbps)	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FR	Functional Requirement	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
ID	Identification		
ISDN	Integrated Services Digital Network	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
Mbps	Megabits per second	UCR	Unified Capabilities Requirements

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	Met (See notes 2 and 3.)

NOTES:
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 Audio Only (AO) Unified Capabilities Conferencing System (UCCS) with the exceptions noted in Table 1. DISA adjudicated these exceptions as minor.

LEGEND:

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

Product Identification			
Product Name	Forum Communications Consortium II		
Software Release	9.1		
UC Product Type(s)	Unified Capabilities Conference System (UCCS) (Audio Only)		
Product Description	Unified Capabilities Conference System (UCCS) (Audio Only)		
Product Components (See note.)	**Component Name**	**Version**	**Remarks**
Audio Conferencing Server	Consortium II Chassis	9.1 Windows Server 2008 Standard Edition, SP2	
Local management items	Monitor, keyboard, mouse	Not Applicable	
Management Laptop (site-provided)		Microsoft Windows 7 SP1	

NOTE: The detailed component and subcomponent list is provided in Enclosure 3.

LEGEND:

APL	Approved Product List	UC	Unified Capabilities
SP	Service Pack		

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 2 through 13 December 2013 using test procedures derived from Reference (d). Patches were applied and regression testing was conducted from 3 through 7 February 2014. Patches

were applied and regression testing was conducted from 25 through 27 March 2014. Review of the vendor's LoC was completed on 5 April 2014. DISA adjudication of outstanding TDRs was completed on 8 April 2014. Information Assurance (IA) testing was conducted by DISA-led IA 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. 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. Cary Hogan, commercial telephone (520) 538-2589, DSN telephone 879-2589, FAX DSN 879-4347; e-mail address cary.v.hogan.civ@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Cary Hogan) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1320401.

FOR THE COMMANDER:

3 Enclosures a/s

for RIC HARRISON
Chief
Networks/Communications and UC Portfolio

Distribution (electronic mail):
DoD CIO
Joint Staff J-6, JCS
USD(AT&L)
ISG Secretariat, DISA, JTA
U.S. Strategic Command, J665
US Navy, OPNAV N2/N6FP12
US Army, DA-OSA, CIO/G-6 ASA(ALT), SAIS-IOQ
US Air Force, A3CNN/A6CNN
US Marine Corps, MARCORSYSCOM, SIAT, A&CE Division
US Coast Guard, CG-64
DISA/TEMC
DIA, Office of the Acquisition Executive
NSG Interoperability Assessment Team
DOT&E, Netcentric Systems and Naval Warfare
Medical Health Systems, JMIS IV&V
HQUSAISEC, AMSEL-IE-IS
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) Findings Summary For Forum Communications Consortium II Unified Capabilities Conference System (UCCS) Release (Rel.) 9.1 (Tracking Number 1320401)," Draft

CERTIFICATION SUMMARY

1. SYSTEM AND REQUIREMENTS IDENTIFICATION. The Forum Communications Consortium II with Software Release 9.1 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

System Identification	
Sponsor	United States Air Force
Sponsor Point of Contact	Joseph Gibbons, e-mail: joseph.gibbons@peterson.af.mil
Vendor Point of Contact	Bob Hartwig, Address: 1223 North Glenville Drive, Richardson TX 75081, e-mail: bob@bobjectsinc.com
System Name	Forum Communications Consortium II UCCS
Increment and/or Version	9.1
Product Category	Audio Only (AO) Unified Capabilities Conference System (UCCS)
System Background	
Previous certifications	None.
Tracking	
UCCO ID	1320401
System Tracking Program ID	4849
Requirements Source	
Unified Capabilities Requirements	Unified Capabilities Requirements 2013, Errata 1
Remarks	
Test Organization(s)	Joint Interoperability Test Command, Fort Huachuca, Arizona
LEGEND: ID Identification 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 an Audio Only UCCS that provides real-time audio conferencing capabilities for the Department of Defense (DoD). The SUT can support 2000 concurrent SIP calls or 1920 TDM ports over multiple conference types including Meet Me, Preset, and Blast Dial Conferences. The SUT connects to the Defense Information Systems Network (DISN) via the Wide Area Network (WAN) Softswitch (SS) or Local Session Controller (LSC) on the Assured Services Local Area Network (ASLAN) using Assured Services Session Initiation Protocol (AS-SIP) with Encrypted Signaling (Transport Layer Signaling [TLS]) and media (Secure Real-Time Transport Protocol [SRTP]). The SUT was tested with and is only certified for use with AS-SIP directly connected to the following LSCs: REDCOM High Density Exchange (HDX) and SLICE and the

Avaya AS5300. The SUT is also certified with AS-SIP routed via the Avaya AS5300 SS to all other LSCs listed on the Unified Capabilities (UC) Approved Products List (APL). The SUT is also certified for use with any Session Controller (SC), SS, Multifunction Switch (MFS), End Office (EO), Small End Office (SMEO), Private Branch Exchange (PBX)1 or PBX2 that is or was previously on the UC APL certified with a Primary Rate Interface (PRI) Digital Transmission Link Level 1 (T1) American National Standards Institute (ANSI) T1.619a or National Integrated Services Digital Network (ISDN) 2 (NI2) (ANSI T1.619a OR ANSI T1.607) interface.

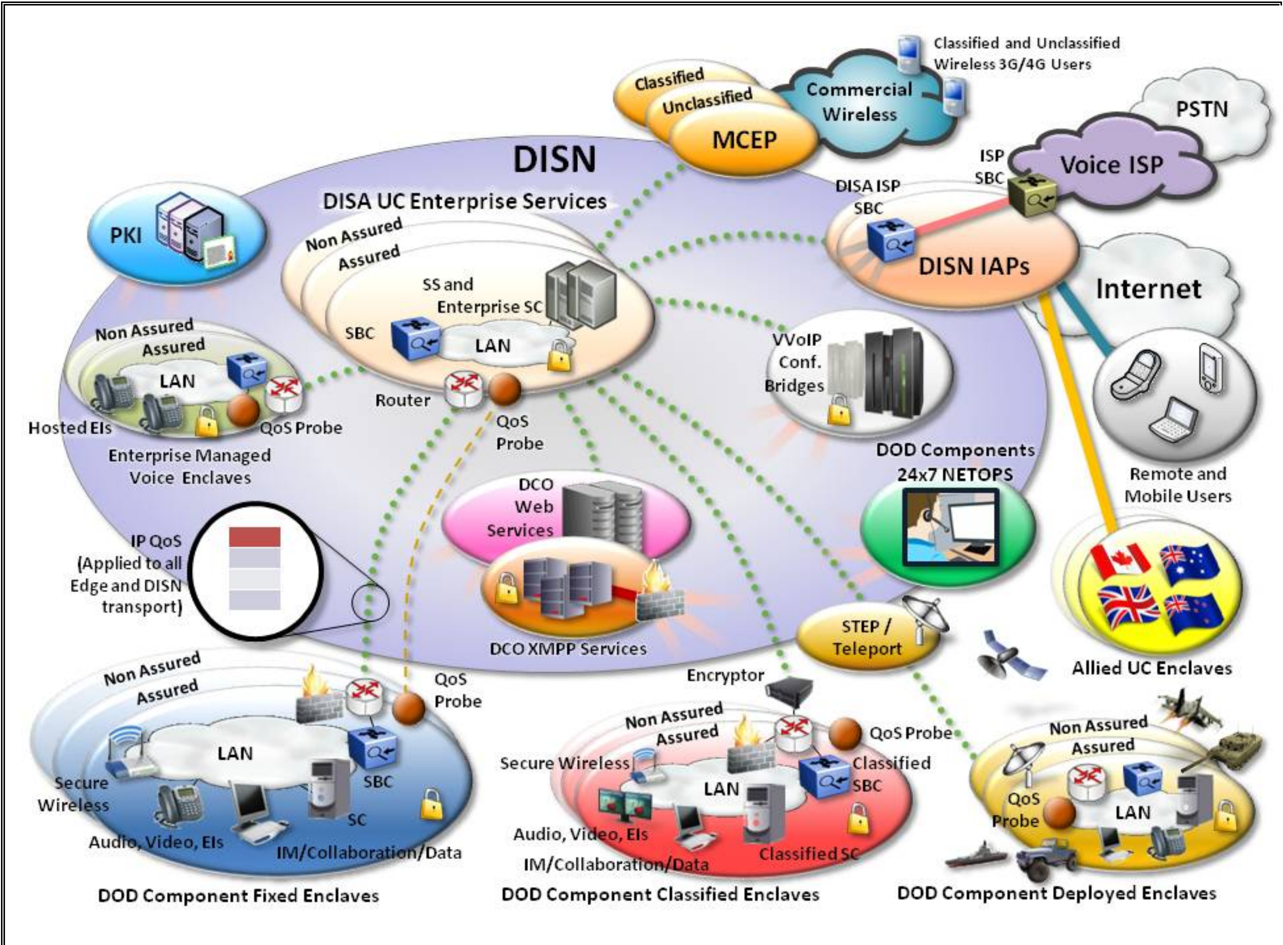
The SUT is composed of two components: The Consortium II chassis including the CPU with integrated network controller and multi-function cards. The Consortium II chassis runs on Microsoft Windows Server 2008 Standard Edition, SP2. The SUT includes a monitor, keyboard, and mouse for local administration. The SUT has multiple functions, which include configurable security levels, internal recording, and programmable Blast Dial conferences. The server application includes WebView, a secure, browser user interface which provides remote System Administration and User feature access to schedule, create, modify, and monitor conferences and participants using a standard web browser and a Public Key Infrastructure (PKI)-enabled Personal Computer (PC) over a LAN.

Management Workstation. The management workstation is a site-provided, Security Technical Implementation Guide (STIG)-compliant, Public Key (PK)-enabled workstation. The SUT can also be managed locally using the Monitor, Keyboard, and Mouse using the (WebView) browser via Localhost.

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.

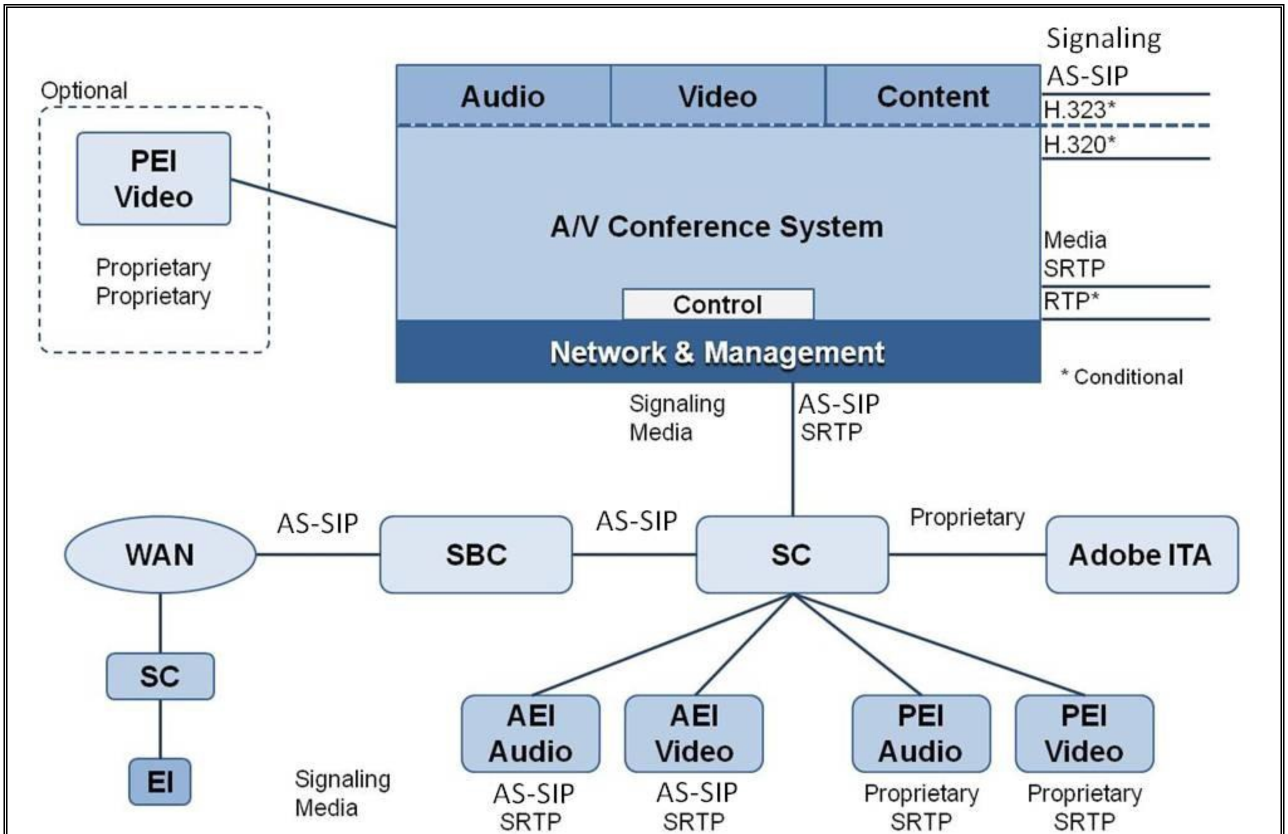
5. METHODOLOGY. Testing was conducted using UCCS requirements derived from the Unified Capabilities Requirements (UCR) 2013, Reference (c), and UCCS test procedures, Reference (d). Any discrepancies noted were written up in Test Discrepancy Reports (TDRs). The vendor submitted Plan of Action and Milestones (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.



LEGEND:

DCO	Defense Connection Online	NETOPS	Network Operations
DISA	Defense Information Systems Agency	PKI	Public Key Infrastructure
DISN	Defense Information Systems Network	PSTN	Public Switched Telephone Network
DoD	Department of Defense	QoS	Quality of Service
EI	End Instrument	SBC	Session Border Controller
IAP	Integrated Access Point	SC	Session Controller
IM	Instant Messaging	SS	Softswitch
IP	Internet Protocol	UC	Unified Capabilities
ISP	Internet Service Provider	VVoIP	Voice and Video over IP
LAN	Local Area Network	XMPP	Extensible Messaging and Presence Protocol
MCEP	Multi-Carrier Entry Point		

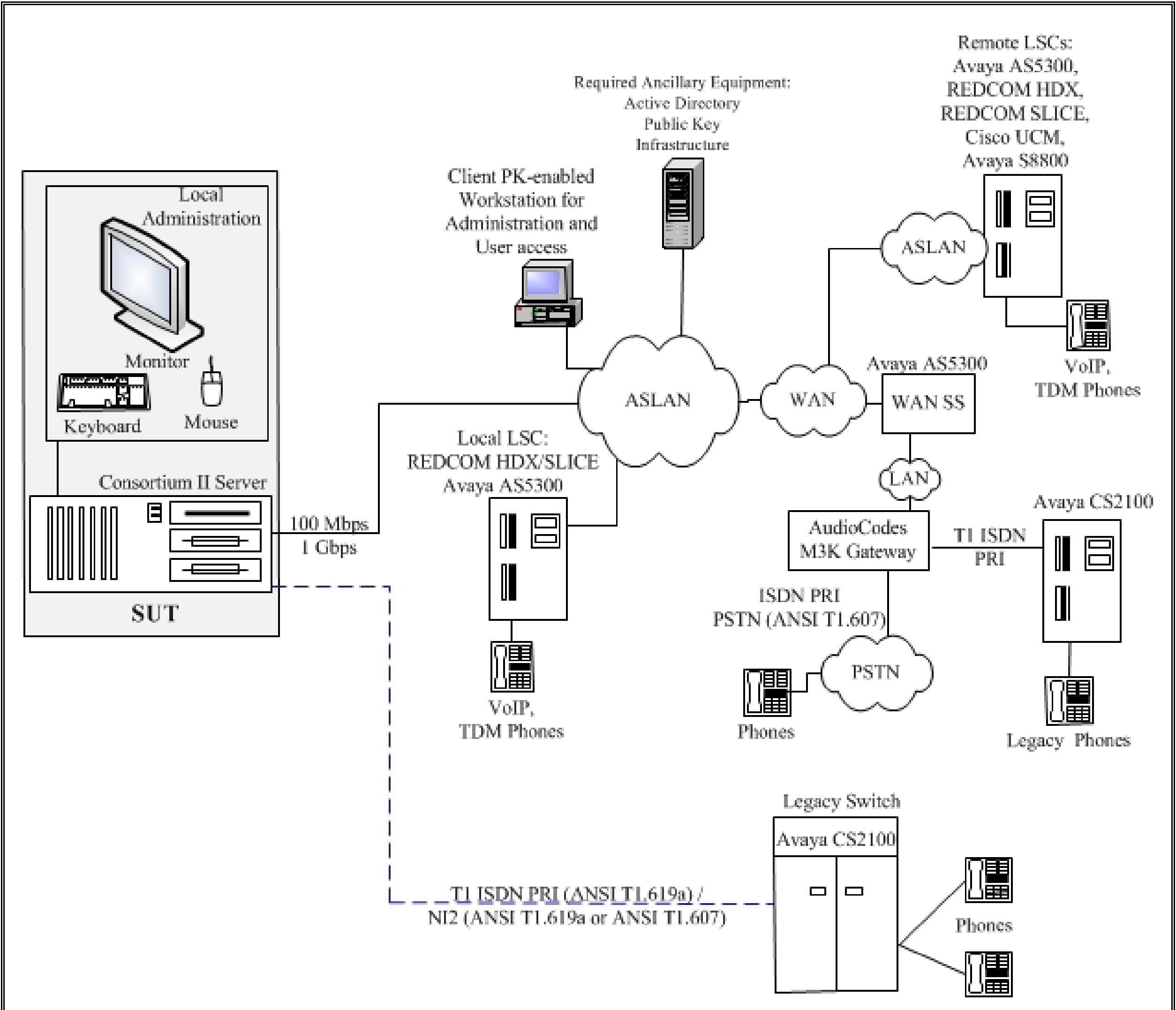
Figure 2-1. Notional UC Network Architecture



LEGEND:

AEI	AS-SIP End Instrument	PEI	Proprietary End Instrument
AS-SIP	Assured Services Session Initiation Protocol	RTP	Real-Time Transport Protocol
A/V	Audio/Video	SBC	Session Border Controller
EI	End Instrument	SC	Session Controller
H.320	ITU-T Standard for narrowband VTC	SRTP	Secure Real-Time Transport Protocol
H.323	ITU-T Standard for multi-media communications on packet-based networks	UCCS	Unified Capabilities Conference System
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	WAN	Wide Area Network

Figure 2-2. UCCS Functional Reference Model



LEGEND:

ANSI	American National Standards Institute	PSTN	Public Switch Telephone Network
ASLAN	Assured Services Local Area Network	SS	Softswitch
DSN	Defense Switched Network	SS7	Signaling System 7
ISDN	Integrated Services Digital Network	SUT	System Under Test
LAN	Local Area Network	T1	Digital Transmission Link Level 1 (1.544 Mbps)
LSC	Local Session Controller	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
M3K	Mediant 3000	TDM	Time Division Multiplexing
Mbps	Megabits per second	VoIP	Voice over Internet Protocol
MLPP	Multilevel Precedence and Preemption	WAN	Wide Area Network
PRI	Primary Rate Interface		

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 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 using AS-SIP. The SUT connects to the DISN via the WAN SS or LSC on the LAN using AS-SIP with Encrypted Signaling (TLS) and media (SRTP). The SUT was tested with and is only certified for use with AS-SIP directly connected to the following LSCs: REDCOM HDX and SLICE and the Avaya AS5300. The SUT is also certified with AS-SIP routed via the Avaya AS5300 SS to all other LSCs listed on the UC APL. The SUT is also certified for use with any SC, SS, MFS, EO, SMEO, PBX1, or PBX2 that is or was previously on the UC APL certified with a PRI T1 ANSI T1.619a or NI2 (ANSI T1.619a or ANSI T1.607) interface.

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 all requirements for 10/100/1000 Mbps Ethernet. In addition, the SUT met all requirements for a PRI T1 ANSI T1.619a or NI2 (ANSI T1.619a or ANSI T1.607) interface.

(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 met this requirement with the vendor's LoC.

(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 and the vendor's LoC. The SUT does not support the optional requirement for H.323 and H.320.

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. In addition, the SUT also meets the optional requirement for ISDN transport. The SUT met the requirements with a PRI T1 ANSI T1.619a or NI2 (ANSI T1.619a or ANSI T1.607) interface. 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 for audio streams 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. This requirement does not apply to the SUT, which is an AO UCCS.

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. In addition, the SUT also meets the optional requirement for ISDN transport. The SUT met the requirements with a PRI T1 ANSI T1.619a or NI2 (ANSI T1.619a or ANSI T1.607) interface. 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. This requirement does not apply to the SUT, which is an AO UCCS.

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 met this requirement with testing. The SUT provides entry and exit tones to participants when new participants join the conference or any participant leaves the conference.

3. The conference system shall provide the ability to extend conferences, without disruption, to conferences in progress. The SUT met this requirement with testing.

4. The conference system shall support presentation capability for different screen layouts locally, manageable by each individual host. This requirement does not apply to the SUT, which is an AO UCCS.

5. The conference system shall provide the conferencing chair control functionality in the subparagraphs below.

a. Voice activated switching. This requirement does not apply to the SUT, which is an AO UCCS.

b. Broadcast mode. The SUT met this requirement with the vendor's LoC.

c. Lecture mode. The SUT met this requirement with the vendor's LoC.

d. Video switching with H.243 control. This requirement does not apply to the SUT, which is an AO UCCS.

e. Continuous presence. This requirement does not apply to the SUT, which is an AO UCCS.

f. Add/delete, accept/reject, connect/disconnect, mute/unmute, audio/video. The SUT met this requirement with the vendor's LoC.

(a) 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. This requirement does not apply to the SUT, which is an AO UCCS.

2. The conference system shall provide automatic video transcoding without downgrading the conference to a lowest common denominator protocol. This requirement does not apply to the SUT, which is an AO UCCS.

3. The conference system shall provide automatic audio transcoding without downgrading the conference to a lowest common denominator protocol. The SUT met this requirement with testing and the vendor's LoC.

(b) 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. This requirement does not apply to the SUT, which is an AO UCCS.

(c) 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 does not apply to the SUT, which is an AO UCCS.

(d) 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. This requirement does not apply to the SUT, which is an AO UCCS.

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.

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.

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.

(e) 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 an AO 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. This requirement does not apply to the SUT, which is an AO UCCS.

3. The system shall provide internal conferencing capabilities for the support of audio-only participants. The SUT met this requirement with testing. All of the SUT participants are audio only.

(f) 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. The SUT met these requirements with the vendor's LoC.

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.

b. Awareness of relevant, accurate information about the conferencing service to users at all levels (strategic, operational, and tactical).

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.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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. The SUT met these requirements with the vendor's LoC.

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.

1. For video equipment, the conference system shall adhere to Federal Telecommunications Recommendation 1080B-2002 (FTR-1080B). This requirement does not apply to the SUT, which is an AO UCCS.

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). This requirement does not apply to the SUT, which is an AO UCCS.

b. The conference system shall support the following video standards for video conferencing: H.261, H.264 (SVC). The SUT supports H.261. This requirement does not apply to the SUT, which is an AO 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. The SUT supports G.729/G.729a. The SUT does not support G.722, G.722.1, G.723.1, G.728. DISA has accepted the vendor's POA&M and adjudicated this as minor.

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

e. The conference system shall provide interoperability for all endpoint devices that support H.320 during call setup. This requirement does not apply to the SUT, which is an AO UCCS.

f. The conference system shall provide interoperability for all end point devices that support H.323 during call setup. This requirement does not apply to the SUT, which is an AO UCCS.

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 called16 Common Intermediate Format (16CIF)), (6) SD and HD video resolution formats for H.261, H.263, and H.264 codecs. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

j. The system's freeze-frame image size shall support HD (720p) and HD (1080p) when using H.239. This requirement does not apply to the SUT, which is an AO UCCS.

k. The system's freeze-frame image size shall support 4FCIF (4CIF) when using H.261 Annex D. This requirement does not apply to the SUT, which is an AO UCCS.

3. H.320 and H.323 Protocols. The conference system that supports H.323/H.320 protocols shall meet the requirements in the following subparagraphs. This requirement does not apply to the SUT, which is an AO UCCS.

a. ISDN/PRI, H.323 V4, chair control, serial interfaces, content sharing VTC endpoint protocol requirements.

b. European E1 ISDN standards.

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.

d. H.323/H.320 V4.

e. Far end camera control (FECC) H.281 and H.323 Annex Q.

f. Resource Availability Indicator (RAI)/Resource Availability Confirmation (RAC) for load balancing.

g. Chair control messages per H.246, H.242/H.243.

h. Direct, H.225 routed, and H.225-H.245 routed modes of H.323 gatekeeper operations.

i. Quality of Service (QoS) support using Differentiated Services Code Point (DSCP) marking of IP packets.

j. Automatic downspeed to available ISDN/IP bandwidth.

k. Automatic rate detection to match incoming video calls.

l. V.35/RS-449/EIA-530 Data Terminating Equipment (DTE) and Data Circuit-Terminating Equipment (DCE) interfaces.

m. H.239 for additional video channels or still images.

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. This requirement does not apply to the SUT, which is an AO UCCS.

c. Chair control messages per H.246, H.242/H.243. This requirement does not apply to the SUT, which is an AO UCCS.

d. QoS support using DSCP marking of IP packets. This requirement does not apply to the SUT, which is an AO UCCS.

e. Automatic downspeed to available IP bandwidth. This requirement does not apply to the SUT, which is an AO UCCS.

f. Automatic rate detection to match incoming video calls. This requirement does not apply to the SUT, which is an AO UCCS.

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

h. H.261 Annex D for still images. This requirement does not apply to the SUT, which is an AO UCCS.

5. Video Mixing Modes. The conference system shall ensure that all video mixing modes meet the following standards in the subparagraphs below. This requirement does not apply to the SUT, which is an AO UCCS.

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.

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.

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 requirements in the following subparagraphs shall be supported. This requirement does not apply to the SUT, which is an AO UCCS.

a. H.320 chair control messages and procedures as defined in H.230, H.242, H.243, and H.245.

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.

c. H.323 – H.320 gateways shall follow the H.246 message translation tables related to chair control functions.

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.

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). The SUT met this requirement with the vendor's LoC.

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

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. The SUT does not support this optional requirement.

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

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

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). The SUT met this requirement with the vendor's LoC.

g. The audio system shall support the automatic number identification (ANI) feature and use it to identify a calling party, if applicable. The SUT met this requirement with the vendor's LoC.

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

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

c. The audio system shall support configurable transmission control protocol (TCP) ports for AS-SIP messaging. The SUT met this requirement with the vendor's LoC.

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

e. The audio system shall support Network Time Protocol (NTP), version 3 [RFC 1305]. The SUT met this requirement with the vendor's LoC.

f. The audio system shall support SNMPv3 [RFC 3414]. The SUT met this requirement with the vendor's LoC.

g. The audio system shall support Secure Real-Time Transport Protocol (SRTP) and Secure Real-Time Transport Control Protocol (SRTCP) [RFC 3711]. The SUT met this requirement with the vendor's LoC.

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]. The SUT met this requirement with the vendor's LoC.

i. The audio system's AS-SIP signaling module shall support RFC 3261including loose route. The SUT met this requirement with the vendor's LoC.

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

k. The audio system's AS-SIP signaling module shall support Session Description Protocol (SDP) as defined in RFC 4566. The SUT met this requirement with the vendor's LoC.

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

m. The AS-SIP signaling module shall support AS-SIP Digest Authentication [RFCs 3261 and 3310]. The SUT met this requirement with the vendor's LoC.

n. The AS-SIP signaling module shall be able to reject incoming INVITE messages when the message does not come from pre-provisioned proxies. The SUT met this requirement with testing.

o. The AS-SIP signaling module shall support call transfer as specified in AS-SIP Section 9.6, Call Transfer. The SUT met this requirement with testing with the following minor exception. The SUT does not fully support the call transfer feature. The SUT does not support call transfer interaction with EIs that require the REFER method for transfer. DISA has accepted the vendor's POA&M and adjudicated this as minor.

p. Audio systems shall support electronic numbering (ENUM) service registration for SIP (AS-SIP) Addresses-of-Record [RFC 3764]. The SUT met this requirement with the vendor's LoC.

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

b. The audio system shall support G.711 μ/A law [pulse code modulation (PCM)] and G.729. The SUT met this requirement with the vendor's LoC.

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

d. The audio system shall support G.168 compliance echo canceller (EC) with 128 ms echo path. The SUT met this requirement with the vendor's LoC.

e. The audio system shall support Audio and Video Transport (AVT) payload type 0 and 8. [G.711 a/mu law]. The SUT met this requirement with the vendor's LoC.

f. The audio system shall support AVT payload 18 [G.729]. The SUT met this requirement with the vendor's LoC.

g. The audio system shall be able to accept in-band DTMF tones. The SUT met this requirement with testing.

h. The audio system shall be able to send DTMF specified by RFC 4733. The SUT met this requirement with the vendor's LoC.

i. The audio system shall be able to conceal 1 percent of packet loss without appreciable quality degradation. The SUT met this requirement with the vendor's LoC.

j. The audio system shall be able to tolerate 40 ms of jitter for audio without appreciable quality degradation. The SUT met this requirement with the vendor's LoC.

k. The audio system shall implement adaptive jitter buffers instead of static fix jitter buffers. The SUT met this requirement with the vendor's LoC.

l. All hardware shall meet Network Equipment Building System-3 (NEBS-3) requirements. Per the vendor's LoC, the SUT chassis does not have NEBS-3 Certification; however, it is classified as a carrier grade industrial PC chassis that has some hardening to provide robustness to vibration, thermal overload, and electrical failover. DISA adjudicated this as minor and stated the intent to remove this requirement.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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 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 the vendor's LoC.

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 the vendor's LoC.

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 met this requirement with testing and the vendor's LoC.

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 met this requirement with testing and the vendor's LoC.

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 met this requirement with testing.

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 met this requirement with testing.

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 met this requirement with testing.

(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 met this requirement for Medium Availability 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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

(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. The SUT met this requirement with the vendor's LoC.

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

3. The conference system shall ensure the time needed to compile polling statistics to adequately support the audio conference capability. The SUT met this requirement with the vendor's LoC.

(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 met 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. This requirement does not apply to the SUT, which is an AO UCCS.

2. Video Conference Capacity - The system shall have the capacity to support at least 1,000 concurrent 384 kbps video calls. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

4. The system shall provide enough transcoding capacity to support that capability regardless of the access methods, algorithms, speeds, or feature sets being used. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

8. The system shall support a minimum of 200 EIs for each multipoint conference. This requirement does not apply to the SUT, which is an AO UCCS.

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

10. The system shall ensure that audio added-on does not compromise support for other capacity requirements. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

(f) Audio Conference Capacity

1. The system shall be capable of scaling up to 2,500 concurrent audio calls. The SUT does not meet this optional requirement. The SUT can support 2000 concurrent SIP calls or 1920 TDM ports.

2. The system shall support up to 200 participants in a single conference session. The SUT met this optional requirement with the vendor's LoC.

3. The system shall be capable of scaling up to 500 concurrent conferences. The SUT met this optional requirement with the vendor's LoC.

4. The system shall be capable of scaling up to 500 or more conference control web sessions. The SUT met this optional requirement with the vendor's LoC.

5. The system shall be capable of scaling to support 10,000 or more reservations. The SUT met this optional requirement with the vendor's LoC.

6. The audio conference system shall support more than 50 concurrent recordings. The SUT met this optional requirement with the vendor's LoC.

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. The SUT does not support this optional requirement.

(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 met this requirement with the vendor's LoC. The SUT can support 2000 concurrent SIP calls or 1920 TDM ports.

(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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

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. This requirement does not apply to the SUT, which is an AO UCCS.

4. Audio add-on and audio conference service shall be able to scale to accommodate increased growth in call volume and EIs. The SUT met this requirement with the vendor's LoC. The SUT can support 2000 concurrent SIP calls or 1920 TDM ports in a single chassis and multiple chassis can support additional capacity through conference cascading.

(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 met this requirement with the vendor's LoC.

(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 testing and 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 testing and 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 testing and 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 the vendor's LoC.

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

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

d. Reservation and scheduling. The SUT met this requirement with the vendor's LoC.

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

f. Conferencing center Web site. The SUT met this requirement with the vendor's LoC.

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 does not support SNMPv3. DISA has accepted the vendor's POA&M and adjudicated this as minor.

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 testing and 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 met this requirement with the vendor's LoC.

(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:

a. Power status shall be provided for individual shelf, rack, or controller units. The SUT met this requirement with the vendor's LoC.

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 met this requirement with the vendor's LoC.

d. Output/loss of output signal. The SUT met this requirement with the vendor's LoC.

e. Input/output signal outside of specified range. The SUT met this requirement with the vendor's LoC.

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 met this requirement with the vendor's LoC.

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 testing and the vendor's LoC with the following minor exception. The SUT does not reroute a conference call when a fault occurs. DISA adjudicated this as minor and stated the intent to remove this requirement.

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 the 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. The SUT does not support the optional requirements in the subparagraphs below.

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.

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.

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.

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.

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.

(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. The SUT does not support the optional requirements in the subparagraphs below.

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.

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.

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.

(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.

(h) Online Directory. The SUT does not support the optional requirements in the subparagraphs below.

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 UCCS shall comply with registration system requirements in the subparagraphs below. The SUT met this requirement with the vendor's LoC.

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 UUCS shall comply with scheduling system requirements in the subparagraphs below.

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

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. The SUT met this requirement with the vendor's LoC with the following minor exception. The e-mail notifications are not encrypted. DISA has accepted the vendor's POA&M and adjudicated this as minor.

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

(k) Accounting and Billing. The UCCS shall comply with the Accounting and Billing Requirements in the subparagraphs below. The SUT met this requirement with the vendor's LoC.

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.

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.

2. The Accounting Management function shall enable charges to be established for the use of dedicated and dial-up conferencing services resources.

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.

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).

b. Aggregate and combine data for generating reports as specified by the Government.

c. Conference records per individual customer's account.

d. Ensure continuous (24x7) monitoring, processing, and recording for all video services-related events and customer activity data.

e. Maintain a database of various conference reports per individual customer account, including conference detail summary, completion summary, and exception reports.

f. Archive data for possible later retrieval by the Government (e.g., in response to customer inquiries or to audit the data).

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.

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 an Audio Only UCCS in accordance with the UCR and is certified for joint use with other UC Products listed on the APL. The SUT connects to the DISN via the WAN SS or LSC on the LAN using AS-SIP with Encrypted Signaling (TLS) and media (SRTP). The SUT was tested with and is only certified for use with AS-SIP directly connected to the following LSCs: REDCOM HDX and SLICE and the Avaya AS5300. The SUT is also certified with AS-SIP routed via the Avaya AS5300 SS to all other LSCs listed on the UC APL. The SUT is also certified for use with any SC, SS, MFS, EO, SMEO, PBX1, or PBX2 that is or was previously on the UC APL certified with a PRI T1 ANSI T1.619a or NI2 (ANSI T1.619a or ANSI T1.607) interface. The SUT meets the interoperability requirements for the interfaces listed in Table 3-1.

DATA TABLES

Table 3-1. Interface Status

Interface	Threshold CR/FR Requirements (See note 1.)	Status	Remarks
Interfaces			
Ethernet 10 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 100 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 1000 Mbps (AS-SIP) (R) (See note 2.)	1	Met	
Ethernet 10 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
Ethernet 100 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
Ethernet 1000 Mbps (H.323) (C) (See note 2.)	1	Not Tested	This interface is only for a Video UCCS.
PRI T1 NI2 (ANSI T1.619a/T1 6.07) (C)	1	Met	
PRI T1 (H.320) (C)	1	Not Tested	This interface is only for a Video UCCS.
PRI E1 (H.320) (C)	1	Not Tested	This interface is only for a Video UCCS.

NOTES:
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.

LEGEND:

ANSI	American National Standards Institute	MLPP	Multi-Level Precedence and Preemption
AS-SIP	Assured Services Session Initiation Protocol	PRI	Primary Rate Interface
C	Conditional	R	Required
CR	Capability Requirement	SS7	Signaling System 7
DSS1	Digital Subscriber Signaling 1	SUT	System Under Test
E1	European Basic Multiplex Rate (2.048 Mbps)	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FR	Functional Requirement	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
ID	Identification		
ISDN	Integrated Services Digital Network	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
Mbps	Megabits per second	UCR	Unified Capabilities Requirements

Table 3-2. Capability and Functional Requirements and Status

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	**UC Audio and Video Conference System (R)**		
	System Description (R)	3.4.2	Met (See note 2.)
	Service (R)	3.4.3	Partially Met (See notes 3, 4, 5.)
	Service Performance (R)	3.4.4	Met
	Service Management (R)	3.4.5	Partially Met (See notes 6, 7, 8.)

NOTES:
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 an Audio 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 does not support G.722, G.722.1, G.723.1 or G.728 audio standards. DISA has accepted the vendor's POA&M and adjudicated this as minor.
4. The SUT does not meet Network Equipment Building Systems-3 (NEBS-3) Requirements. Per the vendor's Letter of Compliance, the SUT chassis does not have NEBS-3 Certification but is classified as a carrier grade industrial PC chassis that has some hardening to provide robustness to vibration, thermal overload, and electrical failover. DISA adjudicated this as minor and stated the intent to remove this requirement.
5. The SUT does not fully support the call transfer feature. The SUT does not support call transfer interaction with EIs that require the REFER method for transfer. DISA has accepted the vendor's POA&M and adjudicated this as minor.
6. The SUT does not support Simple Network Transport Protocol (SNMP) v3. DISA has accepted the vendor's POA&M and adjudicated this as minor.
7. The SUT does not reroute a conference call when a fault occurs. DISA adjudicated this as minor and stated the intent to remove this requirement.
8. The SUT does not fully support automated scheduling in accordance with requirement AUX-004670. DISA has accepted the vendor's POA&M and adjudicated this as minor.

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

LEGEND:	
DISA Defense Information Systems Agency	POA&M Plan of Action and Milestones
EI End Instrument	SUT System Under Test
PC Personal Computer	UCR Unified Capabilities Requirements

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

Component	Release	Sub-component	Function
Consortium II Chassis	9.1 Windows Server 2008 Standard Edition, SP2	Not Applicable	Audio Conferencing Server
Monitor, keyboard, mouse	Not Applicable	Not Applicable	Local management items
Management Laptop (site-provided)	Microsoft Windows 7 SP1	Not Applicable	Site-provided management laptop

LEGEND:
SP Service Pack
SUT System Under Test

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

System Name	Software Release	Function
Required Ancillary Equipment (site-provided)		
Active Directory		
Public Key Infrastructure		
Test Network Components		
Avaya AS5300	Release 3.0 Service Pack 7	Local Session Controller
Avaya AS5300	Release 3.0 with AudioCodes M3K Gateway	WAN Softswitch
REDCOM SLICE	4.0AR3P9	Local Session Controller
REDCOM HDX	4.0AR3P9	Local Session Controller
Cisco UCM	8.0	Local Session Controller
Cisco UCM	8.6	Local Session Controller
Avaya S8800	Communication Manager (CM) 6.3	Local Session Controller
Avaya CS2100	Success Enterprise (SE)09.1	Multifunction Switch
Cisco 4506		LAN switch

LEGEND:	
AS Application Server	LAN Local Area Network
CS Communication Server	M3K Mediant 3000
HDX High Density Exchange	UCM Unified Communications Manager
IOS Internetwork Operating System	WAN Wide Area Network

Table 3-5. UCCS Capability/Functional Requirements

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
1	**3.4.2 – System Description**					
1-1	The conference system shall provide, as a minimum, an Ethernet-based interface to the network.	3.4.2.2 AUX-002720	L	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
2	**3.4.3 – Service**					
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	L	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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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	L/T		R	R
2-20	Add/Delete, Accept/Reject, Connect/Disconnect, Mute/Unmute, audio/video.	3.4.3.1.5 AUX-002870.f	L/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: • Point-to-point and multipoint conferencing services • Interoperability with different vendor EIs and variable graphic resolutions. • Connections among participants using any video EIs, connection types, and at any rates.	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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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: Audio Protocols: G.711, G.722, G.722.1, G.723.1, G.728, G.729/G.729A Video Protocols: 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/T		R	R
2-65	Sub-Quarter Common Intermediate Format (SQCIF).	3.4.3.3.1 AUX-003280.a	L/T		R	R
2-66	Quarter Common Intermediate Format (QCIF).	3.4.3.3.1 AUX-003280.b	L/T		R	R
2-67	Common Intermediate Format (FCIF, also called CIF).	3.4.3.3.1 AUX-003280.c	L/T		R	R
2-68	4 Full Common Intermediate Format (4FCIF, also called 4CIF).	3.4.3.3.1 AUX-003280.d	L/T		R	R
2-69	16 Full Common Intermediate Format (16FCIF, also called16 Common Intermediate Format (16CIF)).	3.4.3.3.1 AUX-003280.e	L/T		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. Video Format Standards — Video Resolution 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	3.4.3.3.1 AUX-003280.f	L/T		O	O
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/T		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/T		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/T		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/T		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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
2-93	Automatic downspeed to available IP bandwidth.	3.4.3.3.3 AUX-003380	T		R	R
2-94	Automatic rate detection to match incoming video calls.	3.4.3.3.3 AUX-003390	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	L/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	L/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 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].	3.4.3.3.6 AUX-003620	L/T	R		R
2-118	The audio system's AS-SIP signaling module shall support RFC 3261including 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	L/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 µ/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/T	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/T	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/T	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/T	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/T		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/T	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
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/T	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/T	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/T	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/T	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/T	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	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
3	**3.4.4 – Service Performance**					
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/T		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/T		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/T	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/T		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/T		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/T		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/T		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/T		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/T		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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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/T	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/T	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/T	O		O
3-24	The audio conference system shall support more than 50 concurrent recordings.	3.4.4.2.2 AUX-004190	L/T	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/T		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/T		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/T		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/T	R		R
4	**3.4.5 – Service Management**					
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/T	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
4-4	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: • Equipment and associated services. • Point-to-point and multipoint video services. • Audio services. • Reservation and scheduling. • Gateways and interfaces. • Conferencing center Web site. • Support systems.	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/T	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/T	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/T	R	R	R
4-10	The fault management system shall include the following minimum requirements:	3.4.5.1.2 AUX-004350	L/T	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/T	R	R	R
4-12	Functional/Fault/Online/Offline status.	3.4.5.1.2 AUX-004350.b	L/T	R	R	R
4-13	Input/loss of input signal, or signal below working threshold.	3.4.5.1.2 AUX-004350.c	L/T	R	R	R
4-14	Output/loss of output signal.	3.4.5.1.2 AUX-004350.d	L/T	R	R	R
4-15	Input/output signal outside of specified range.	3.4.5.1.2 AUX-004350.e	L/T	R	R	R
4-16	Intrusion detected.	3.4.5.1.2 AUX-004350.f	L/T	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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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: • 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.	3.4.5.1.2 AUX-004360.a	L/T	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/T	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/T	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/T	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: • Outages for all conferencing services elements to be exported into the existing trouble tracking system. • Any hazardous condition, as specified in DISA Circular 310-55-1, that may cause loss of service.	3.4.5.1.3 AUX-004370	L/T	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/T	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/T	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/T	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/T	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/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
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/T	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/T	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/T	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/T		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	L/T	O	O	O
4-34	Enterprise Security Management. The conference system shall implement enterprise management of security devices and applications such as the following: • Firewalls and boundary protection. • Intrusion detection systems. • Operating systems, network devices, and applications security. • Vulnerability management.	3.4.5.1.6 AUX-004490	L/T	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)

ID	Requirement	UCR Ref (UCR 2013)	LoC/ TP ID	AO CS	VO CS	A/V CS
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/T	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/T	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	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	L/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 recived 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	L/T	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	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/T	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/T	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/T	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/T	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/T	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/T	R	R	R

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

LEGEND:

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 called16 Common Intermediate Format (16CIF))	MLPP	Multi-level Precedence and Preemption
		MOS	Mean Opinion Score
A/V	Audio/Visual	ms	millisecond
AO	Audio Only	MTU	Maximum Transmission Unit
AS-SIP	Assured Services Session Initiation Protocol	NA	Not Applicable
AVT	Audio and Video Transport	O	Optional
C	Conditional	PKI	Public Key Infrastructure
CAS	Channel Associated Signaling	PSTN	Public Switched Telephone Network
CTU	Conferencing Terminal Unit	PRI	Primary Rate Interface
CS	Conference System	QCIF	Quarter Common Intermediate Format
DISA	Defense Information Systems Agency	QoS	Quality of Service
DISN	Defense Information Systems Network	R	Required
DMS	Digital Multiplex System	RFC	Request For Comments
DoD	Department of Defense	SIP	Session Initiation Protocol
DNS	Domain Name Service	SNMPv3	Simple Network Management Protocol version 3
DSCP	Differentiated Services Code Point	SQCIF	Sub-Quarter Common Intermediate Format
DTMF	Dual Tone Multi-Frequency	SRTCP	Secure Real-Time Transport Control Protocol
E1	European Basic Multiplex Rate	SRTP	Secure Real-Time Transport Protocol
E2E	End-to-End	SSL	Secure Sockets Layer
EI	End Instrument	SVC	Scalable Video Codec
EMS	Element Management System	SVGA	Super Video Graphics Array
FCIF	Common Intermediate Format (also called CIF)	T	Testable Item
FECC	Far End Camera Control	T1	Digital Transmission Link Level 1
HD	High Definition	TLS	Transport Layer Security
IA	Information Assurance	TP	Test Plan
IAW	in accordance with	UC	Unified Capabilities
ID	Identification	UCCS	Unified Capabilities Conference System
IEEE	Institute of Electrical and Electronics Engineers	UCR	Unified Capabilities Requirements
IP	Internet Protocol	URL	Uniform Resource Locator
IPv4	Internet Protocol version 4	VGA	Video Graphics Array
IPv6	Internet Protocol version 6	VO	Video Only
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
ITU	International Telecommunication Union	VTC	video teleconferencing
kbps	kilobits per second	VVoIP	Voice and Video over Internet Protocol
L	LoC Item	WSXGA+	Widescreen Super Extended Graphics Array +
LoC	Letter(s) of Compliance	XGA	Extended Graphics Array

Table 3-6. IPv6 Requirements

ID	Requirement	UCR Ref (UCR 2013)	LoC/TP ID	UCCS
	5.2 – IPv6 Requirements			
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
	5.2 – IPv6 Requirements (continued)			
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
	5.2 – IPv6 Requirements (continued)			
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
	5.2 – IPv6 Requirements (continued)			
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
	5.2 – IPv6 Requirements (continued)				
v6-70	If the product is using AS-SIP, and the <addrtype> is IPv6, and the <connection-address> is a unicast address, then the product shall support generation and processing of unicast IPv6 addresses having the following formats: • 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. • x: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.		5.2.1.13 IP6-001060	L	C
v6-71	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: • x:x:x:x:x:x:x:x format: 1080:0:0:0:8:800:200C:417A. • x:x:x:x:x:x:d.d.d.d format: 1080:0:0:0:8:800:116.23.135.22. • compressed zeros: 1080::8:800:200C:417A.		5.2.1.13 IP6-001070	L	C
v6-72	If the product is using AS-SIP, and the <addrtype> is IPv6, and the <connection-address> 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.		5.2.1.13 IP6-001080	L	C
v6-73	If the product is using AS-SIP, and the <addrtype> 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.		5.2.1.13 IP6-001090	L	C
v6-74	If the product is using AS-SIP, and the <addrtype> is IPv6, and the <connection-address> 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.		5.2.1.13 IP6-001100	L	C
v6-75	If the product is using AS-SIP, then the product shall support the processing of IPv6 multicast group addresses having the <number of address> field and may support generating the <number of address> field. This field has the identical format and operation as the IPv4 multicast group addresses.		5.2.1.13 IP6-001110	L	C
v6-76	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.		5.2.1.14 IP6-001150	L	R
v6-77	If the product acts as an IPv6 tunnel broker, then the product shall support the function as defined in RFC 3053.		5.2.1.14 IP6-001160	L	C
v6-78	The UCCS must be IPv6-capable. Use guidance in Table 5.2-4 for NA/SS.		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
	5.2 – IPv6 Requirements (continued)				
v6-78 (continued)	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

LEGEND:

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 Protocl
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