



## DEFENSE INFORMATION SYSTEMS AGENCY

P. O. BOX 549  
FORT MEADE, MARYLAND 20755-0549

IN REPLY  
REFER  
TO:

Joint Interoperability Test Command (JTE)

**30 Mar 12**

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

**References:** (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004  
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008  
(c) through (e), see Enclosure 1

1. References (a) and (b) establish the Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.
2. The REDCOM HDX LSC, Version Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7) is hereinafter referred to as the System Under Test (SUT). The SUT is certified for joint use in the Defense Information System Network (DISN) as an LSC. The fielding of the SUT is limited to IP version 4 (IPv4) across the DISN based on the fielding environment and a Plan of Action and Milestones (POA&M) addressing IPv6 discrepancies by 1 July 2012. Intra-enclave use of IPv4 and IPv6 is authorized for joint use. 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 Defense Information Systems Agency (DISA) via a vendor POA&M, which will address all new critical Test Discrepancy Reports (TDR) within 120 days of identification. Testing was conducted using LSC product requirements derived from, Reference (c) and LSC test procedures derived from Reference (d). No other configurations, features, or functions, except those cited within this memorandum, are certified by JITC. This certification expires upon changes that affect interoperability, but no later than three years from the date of the Unified Capabilities Approved Products List memorandum.
3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), and DISA Information Assurance (IA) Certification Authority (CA) approval of the IA configuration. Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 8 March through 21 May 2010. Adjudication of open TDRs was completed by DISA on 27 March 2012. Verification and Validation (V&V) testing was conducted from 21 February through 4 March and 7 through 18 November 2011. Review of the vendor's LoC was completed on 10 November 2011. The DISA CA has reviewed the IA Assessment Report for the SUT, Reference (e), and based on the findings in the report has

JITC Memo, JTE, Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

provided a positive recommendation on 20 March 2012. The acquiring agency or site will be responsible for the Department of Defense (DoD) Information Assurance Certification and Accreditation Process (DIACAP) accreditation. Enclosure 2 documents the test results and describes the tested network and system configurations including specified patch releases.

4. The interface, Capability Requirements (CR) and Functional Requirements (FR), and component status of the SUT is listed in Tables 1 and 2. The threshold CR/FRs for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c) and were used to evaluate the interoperability of the SUT. Enclosure 3 provides a detailed list of LSC requirements.

**Table 1. SUT Interface Interoperability Status**

Interface	Critical	UCR Reference	Threshold CR/FR <sup>1</sup>	Status	Remarks <sup>2</sup>
<b>Line Interfaces</b>					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs (voice only).
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs (voice only).
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, 16	Not Tested	This interface is not offered by the SUT.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, 13,	Certified	Met threshold CRs/FRs for 2-wire analog instruments. Applies to 2-wire secure and non-secure analog instruments, fax, and modem.
BRI	No	5.3.2.6.1.8	2, 4, 10, 13	Certified	Met threshold CRs/FRs for BRI instruments. Applies to voice and clearmode data.
<b>External Interfaces</b>					
10Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Not Tested	This interface is not offered by the SUT.
ISDN T1 PRI NI-2 (ANSI T1.619a)	Yes	5.3.2.4.3	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs. Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2 (ANSI T1.607)	Yes	5.3.2.4.3	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs. Provides PSTN Connectivity
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	2, 3, 7, 8, 10, 13	Not Tested	This interface is offered by the SUT; however, it was not tested.
T1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for T1 CAS.
E1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for T1 CAS.
E1 PRI ITU-T Q.955.3	No <sup>4</sup>	5.3.2.12.10	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for E1 PRI ITU-T Q.955.3. Provides Legacy DSN connectivity in Europe.
E1 PRI ITU-T Q.931	No <sup>4</sup>	5.3.2.12.10	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for E1 PRI ITU-T Q.931. Provides PSTN connectivity in Europe.

JITC Memo, JTE, Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

**Table 1. SUT Interface Interoperability Status (continued)**

Interface	Critical	UCR Reference	Threshold CR/FR <sup>1</sup>	Status	Remarks <sup>2</sup>
<b>NM</b>					
10Base-X	No <sup>3</sup>	5.3.2.4.4 5.3.2.7.2.8	16, 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No <sup>3</sup>	5.3.2.4.4 5.3.2.7.2.8	16, 17	Certified	Met threshold CRs/FRs. Verified via LoC.
<b>NOTES:</b>					
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.					
2. Detailed information pertaining to open TDRs and associated operational impacts is in Enclosure 2, paragraph 11.					
3. The SUT must provide a minimum of one of the listed interfaces.					
4. This interface is conditionally required for deployment in Europe.					
<b>LEGEND:</b>					
ANSI	American National Standards Institute		Mbps	Megabits per second	
AS-SIP	Assured Services Session Initiation Protocol		MFR1	Multi-Frequency Recommendation 1	
BRI	Basic Rate Interface		MLPP	Multi-Level Precedence and Preemption	
CAS	Channel Associated Signaling		NI-2	National ISDN-2	
CCS7	Common Channel Signaling 7		NM	Network Management	
CR	Capability Requirement		PEI	Proprietary End Instrument	
DP	Dial Pulse		PRI	Primary Rate Interface	
DSN	Defense Switched Network		PSTN	Public Switched Telephone Network	
DSS1	Digital Subscriber Signaling 1		Q.931	Signaling Standard for ISDN	
DTMF	Dual Tone Multi-Frequency		Q.955.3	ISDN Signaling Standard for E1 MLPP	
E1	European Basic Multiplex Rate (2.048 Mbps)		SS7	Signaling System 7	
FR	Functional Requirement		SUT	System Under Test	
ID	Identification		T1	Digital Transmission Link Level 1 (1.544 Mbps)	
IEEE	Institute of Electrical and Electronics Engineers		T1.607	ISDN – Layer 3 Signaling Specification For Circuit Switched Bearer Service For DSS1	
ISDN	Integrated Services Digital Network		T1.619a	SS7 and ISDN MLPP Signaling Standard For T1	
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector		TDR	Test Discrepancy Report	
LoC	Letter of Compliance		UCR	Unified Capabilities Requirements	

**Table 2. SUT CR and FR Status**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
<b>1</b>	<b>Assured Services Product Features and Capabilities</b>				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Partially Met <sup>2</sup>	
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met <sup>3</sup>	
	Public Safety Features	Required	5.3.2.2.2.2	Met	
	ASAC – Open Loop	Required	5.3.2.2.2.3	Partially Met <sup>4</sup>	
	Signaling Protocols	Required	5.3.2.2.3	Met	
<b>2</b>	<b>Registration, Authentication, and Failover</b>				
	Registration	Required	5.3.2.3.1	Met	
	Failover	Required	5.3.2.3.2	Met	
<b>3</b>	<b>Product Physical, Quality, and Environmental Factors</b>				
	Availability	Required	5.3.2.5.2.1	Met	
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	
	Loss of Packets	Required	5.3.2.5.4	Met	

**Table 2. SUT CR and FR Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
4	<b>Voice End Instruments</b>				
	Tones and Announcements	Required	5.3.2.6.1.1	Met	
	Audio Codecs	Required	5.3.2.6.1.2	Partially Met <sup>5</sup>	
	VoIP PEI or AEI Audio Performance	Required	5.3.2.6.1.3	Partially Met <sup>6</sup>	
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Met	
	Authentication to LSC	Required	5.3.2.6.1.5	Met	
	Analog Telephone Support	Required	5.3.2.6.1.6	Met	
	Softphones	Conditional	5.3.2.6.1.7	Not Tested <sup>7</sup>	
ISDN BRI	Conditional	5.3.2.6.1.8	Met		
5	<b>Video End Instruments</b>				
	Video End Instrument	Required	5.3.2.6.2	Not Met <sup>8</sup>	
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Met <sup>8</sup>	
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Met <sup>8</sup>	
6	<b>LSC Requirements</b>				
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	Met	
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	Met	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	Met	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	Met	
	Local Location Server and Directory	Required	5.3.2.7.2.5	Met	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	Met	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	Partially Met <sup>9</sup>	
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	Not Tested <sup>10</sup>	
Loop Avoidance	Required	5.3.2.7.3	Met		
7	<b>Call Connection Agent Requirements</b>				
	CCA IWF Component	Required	5.3.2.9.2.1	Met	
	CCA MGC Component	Required	5.3.2.9.2.2	Met	
	SG Component	Conditional	5.3.2.9.2.3	Not Tested <sup>10</sup>	
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested <sup>10</sup>	
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested <sup>10</sup>	
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Met	
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met	
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	Met	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	
CCA Interactions with the EBC	Required	5.3.2.10.4	Met		

**Table 2. SUT CR and FR Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
7	<b>Call Connection Agent Requirements (continued)</b>				
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	
	CCA Support for UFS	Required	5.3.2.10.6	Met	
	CCA Support for IA	Required	5.3.2.10.7	Met	
	CCA Interaction with EIs	Required	5.3.2.10.10	Met	
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met <sup>8</sup>	
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested <sup>10</sup>	
8	<b>MG Requirements</b>				
	Role of MG In LSC	Required	5.3.2.12.3.1	Partially Met <sup>11</sup>	
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	
	MG and IA Functions	Required	5.3.2.12.4.2	Met	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested <sup>10</sup>	
	MG Interaction with EIs	Required	5.3.2.12.4.8	Met	
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	
	MG Interface to TDM	Required	5.3.2.12.5	Met	
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested <sup>10</sup>	
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Met	
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Met	
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested <sup>10</sup>	
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	
	MG Support for CAS Trunks	Conditional	5.3.2.12.11	Met	
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met <sup>11</sup>	
MG Echo Cancellation	Required	5.3.2.12.13	Met		
MG Clock Timing	Required	5.3.2.12.14	Met		
MGC-MG CCA Functions	Required	5.3.2.12.15	Met		
MG ITU-T V.150.1	Required	5.3.2.12.16	Partially Met <sup>12</sup>		
MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17	Met		
9	<b>SG Requirements</b>				
	SG and CCS7 network Interactions	Conditional	5.3.2.13.5.1	Not Tested <sup>10</sup>	
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested <sup>10</sup>	
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested <sup>10</sup>	

**Table 2. SUT CR and FR Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
10	<b>WWNDP Requirements</b>				
	WWNDP	Required	5.3.2.16	Met	
	DSN WWNDP	Required	5.3.2.16.1	Met	
11	<b>Commercial Cost Avoidance</b>				
	Commercial Cost Avoidance	Required	5.3.2.23	Not Tested <sup>13</sup>	
12	<b>AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)</b>				
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	Not Tested <sup>10</sup>	
13	<b>Precedence Call Diversion</b>				
	Precedence Call Diversion	Required	5.3.2.25	Met	
14	<b>Attendant Station Features</b>				
	Precedence and Preemption	Required	5.3.2.26.1	Met	
	Call Display	Required	5.3.2.26.2	Met	
	Class of Service Override	Required	5.3.2.26.3	Met	
	Busy Override and Busy Verification	Required	5.3.2.26.4	Met	
	Night service	Required	5.3.2.26.5	Met	
	Automatic Recall of Attendant	Required	5.3.2.26.6	Met	
	Calls in Queue to the Attendant	Required	5.3.2.26.7	Met	
15	<b>AS-SIP Requirements</b>				
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required	5.3.4.7	Met	
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	
	Session Description Protocol	Required	5.3.4.9	Met	
	Precedence and Preemption	Required	5.3.4.10	Met	
	Video Telephony – General Rules	Required	5.3.4.12	Not Met <sup>8</sup>	
	Calling Services	Required	5.3.4.13	Met	
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	Met	
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	
	SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	Met	
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	
Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met		
Supplementary Services	Required	5.3.4.19	Partially Met <sup>3</sup>		
16	<b>IPv6 Requirements</b>				
	Product Requirements	Required	5.3.5.4	Partially Met <sup>14</sup>	

JITC Memo, JTE, Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

**Table 2. SUT CR and FR Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
17	<b>NM</b>				
	LSC Management Function	Required	5.3.2.7.2.6	Met	
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Met	
	General Management requirements	Required	5.3.2.17.2	Met	
	Requirement for FCAPS Management	Required	5.3.2.17.3	Met	
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met <sup>15</sup>	
	Accounting Management	Required	5.3.2.19	Partially Met <sup>15</sup>	

**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. The SUT correctly class marks calls based on the precedence level dialed; however, the DSCP values are not configurable from 0-63 as required by UCR 2008, Change 2. This was adjudicated by DISA as having a minor operational impact based on the vendor's submitted POA&M to fix by 1 July 2012.
3. The SUT met all critical CRs & FRs with one minor exception: In a 3-way conference, the SUT does not classmark the conference at the highest precedence level of each leg of the conference. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
4. The SUT does not support separate ASAC counts for video and voice. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
5. The SUT does not support ITU-T G.722.1 voice codec. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by December 2012.
6. The SUT only supports voice PEIs. The vendor did not support AEI voice capability. This was adjudicated by DISA to have a minor operational impact since there were no certified AEI video end instruments on the UC APL.
7. The vendor did not provide soft phones with the SUT for interoperability testing; therefore, they were not tested. Soft phones are a conditional requirement.
8. The SUT offers video; however, it is not covered under this certification. JITC only conducted limited video testing on the SUT with Polycom video EIs. Due to the fact that their video EI is not IPv6 capable and did not offer a GA load it did not meet this requirement. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.
9. The SUT does not comply with the requirement to be configurable to set interval times to verify status of its IP EIs with the default set at five minutes. This was adjudicated by DISA as having a minor operational impact with the intent to change the requirement to conditional in next UCR update.
10. This is a conditional requirement and was not tested.
11. The SUT met this requirement with VoIP PEIs only.
12. The SUT does not fully comply with the requirements for ITU-T V.150.1. For FAX (ITU-T T.38 transitions) the LSC maintains the session protocol (UDP). However, the port is changed when moving from VoIP to FoIP. The LSC sends a REINVITE (per MGC transitioning rules) with the new port number; therefore, SIP compliant devices (including the EBC) should follow the port change. The port selection is currently configurable for MoIP because MoIP does not use the REINVITE method for transition. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
13. The SUT does not currently support LDAP V3 for CCA. Transition to the LDAP format is currently in progress and is expected to be functional by late June 2011. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
14. The vendor submitted an IPv6 LoC with noted discrepancies which include: The SUT is not fully compliant with IPv6 neighbor discovery IAW RFC 4861. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
15. The vendor submitted an NM LoC with the following noted discrepancies: The SUT does not fully comply with the CDR and QoS requirements per UCR 2008, Change 2, section 5.3.2.19.2.1.x. The SUT does not comply with the Management Requirements for ASAC per UCR 2008, Change 2, section 5.3.2.18.2. These were adjudicated by DISA as having a minor operational impact with vendor POA&M to fix by 1 July 2012.

JITC Memo, JTE, Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

**Table 2. SUT CR and FR Status (continued)**

<b>LEGEND:</b>			
AEI	AS-SIP End Instrument	JITC	Joint Interoperability Test Command
Apl	Approved Products List	LDAP	Lightweight Directory Access Protocol
AS	Assured Services	LoC	Letter of Compliance
ASAC	Assured Services Admission Control	LSC	Local Session Controller
AS-SIP	Assured Services Session Initiation Protocol	MG	Media Gateway
BRI	Basic Rate Interface	MGC	Media Gateway Controller
CAS	Channel Associated Signaling	MoIP	Modem over Internet Protocol
CCA	Call Connection Agent	NM	Network Management
CCS7	Common Channel Signaling 7	NMS	Network Management System
CDR	Call Detail Records	OCONUS	Outside the Continental United States
CR	Capabilities Requirement	PBAS	Precedence-Based Assured Service
DISA	Defense Information Systems Agency	PEI	Proprietary End Instrument
DSCP	Differentiated Services Code Point	POA&M	Plan of Actions and Milestones
DSN	Defense Switched Network	PRI	Primary Rate Interface
EBC	Edge Boundary Controller	PSTN	Public Switched Telephone Network
EI	End Instrument	QoS	Quality of Service
FCAPS	Fault, Configuration, Accounting, Performance, and Security	RFC	Request for Comments
FoIP	Fax over Internet Protocol	SG	Signaling Gateway
FR	Functional Requirement	SIP	Session Initiation Protocol
GA	General Available	SS7	Signaling System Number 7
HDX	High Density Exchange	SUT	System Under Test
IA	Information Assurance	TDM	Time Division Multiplexing
IAW	in accordance with	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements
IP	Internet Protocol	UDP	User Datagram Protocol
IPv6	Internet Protocol version 6	UFS	User Features and Services
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	VVoIP	Voice and Video over Internet Protocol
IWF	Interworking Function	WAN	Wide Area Network
		WWNDP	World Wide Numbering and Dialing Plan

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet 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 <http://jit.fhu.disa.mil> (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at <http://www.disa.mil/ucco/>.

JITC Memo, JTE, Special Interoperability Test Certification of the REDCOM High Density Exchange (HDX)<sup>TM</sup> Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7)

6. The JITC point of contact is Anita Mananquil, JITC, commercial (520) 538-5164 or DSN 312-879-5164; e-mail address is anita.mananquil@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 0932802.

FOR THE COMMANDER:

3 Enclosures a/s

  
for BRADLEY A. CLARK  
Chief  
Battlespace Communications Portfolio

Distribution (electronic mail):

Joint Staff J-6

Joint Interoperability Test Command, Liaison, TE3/JT1

Office of Chief of Naval Operations, CNO N6F2

Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)

Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ

U.S. Marine Corps MARCORSSYSCOM, SIAT, MJI Division I

DOT&E, Net-Centric Systems and Naval Warfare

U.S. Coast Guard, CG-64

Defense Intelligence Agency

National Security Agency, DT

Defense Information Systems Agency, TEMC

Office of Assistant Secretary of Defense (NII)/DOD CIO

U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68

Defense Information Systems Agency, GS23

## **ADDITIONAL REFERENCES**

- (c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008, Change 2," 31 December 2010
- (d) Joint Interoperability Test Command, "Unified Capabilities Test Plan (UCTP)," Draft
- (e) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of REDCOM HDX™ Local Session Controllers, Version 4.0AR3 (TN# 0932802)," Draft

## CERTIFICATION TESTING SUMMARY

**1. SYSTEM TITLE.** The REDCOM® High Density Exchange (HDX)™ Local Session Controller (LSC), Software Release 4.0 Revision 3 with Specified Patch Group 7 (4.0AR3P7), hereinafter referred to as the System Under Test (SUT).

**2. SPONSOR.** United States Marine Corps, Attention: CWO4 Jeffrey Gardner, MARCORSYSCOM, 2200 Lester Street, Quantico, VA, 22134, Phone 703-432-4349, e-mail: jeffrey.a.gardner@usmc.mil.

**3. SYSTEM POC.** REDCOM Laboratories, Inc., Attention: Michael Gates, Government Technical Integration Engineer, Address: One Redcom Center, Victor, New York, 14564, Phone: 585-924-6500, e-mail: mgates@redcom.com.

**4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

**5. SYSTEM DESCRIPTION.** The SUT is a communications system based on the patented modular switching unit (MSU). Each MSU is a standard 19 inch rack mount platform that stands five rack units high. In addition to the common control circuit card slots, a single HDX MSU has 16 slots for various circuit cards based on the customer's specific needs. Multiple HDX MSUs can be stacked to allow for expansion while creating a single logical system. The HDX platform utilizes REDCOM's real time operating system (RTOS).

The SUT is a secure Unified Communications (UC) system using Transport Layer Security (TLS) and Secure Real-time Transport Protocol (SRTP). The system provides integrated IP telephony, conferencing, and voice mail that meets, Department of Defense (DoD) security and service assurance requirements. The SUT is used for creating, modifying, and terminating two-party (unicast) or multiparty (multicast) media streams.

The SUT is a session manager designed to increase productivity and collaboration by allowing users to collaborate in an integrated solution. The SUT supports the Session Initiation Protocol (SIP), Assured Services–SIP (AS-SIP), SRTP, Session Description Protocol, Security Descriptions for Media Streams, TLS, Multilevel Precedence and Preemption (MLPP) using American National Standards Institute (ANSI) T1.619a Primary Rate Interface (PRI). Services that are included are: voice, video calling, internal voicemail, audio ad hoc conferencing, and audio meet me conferencing.

**6. OPERATIONAL ARCHITECTURE.** Figure 2-1 depicts the LSC functional model and Figure 2-2 the notional operational architecture that the SUT may be used in.

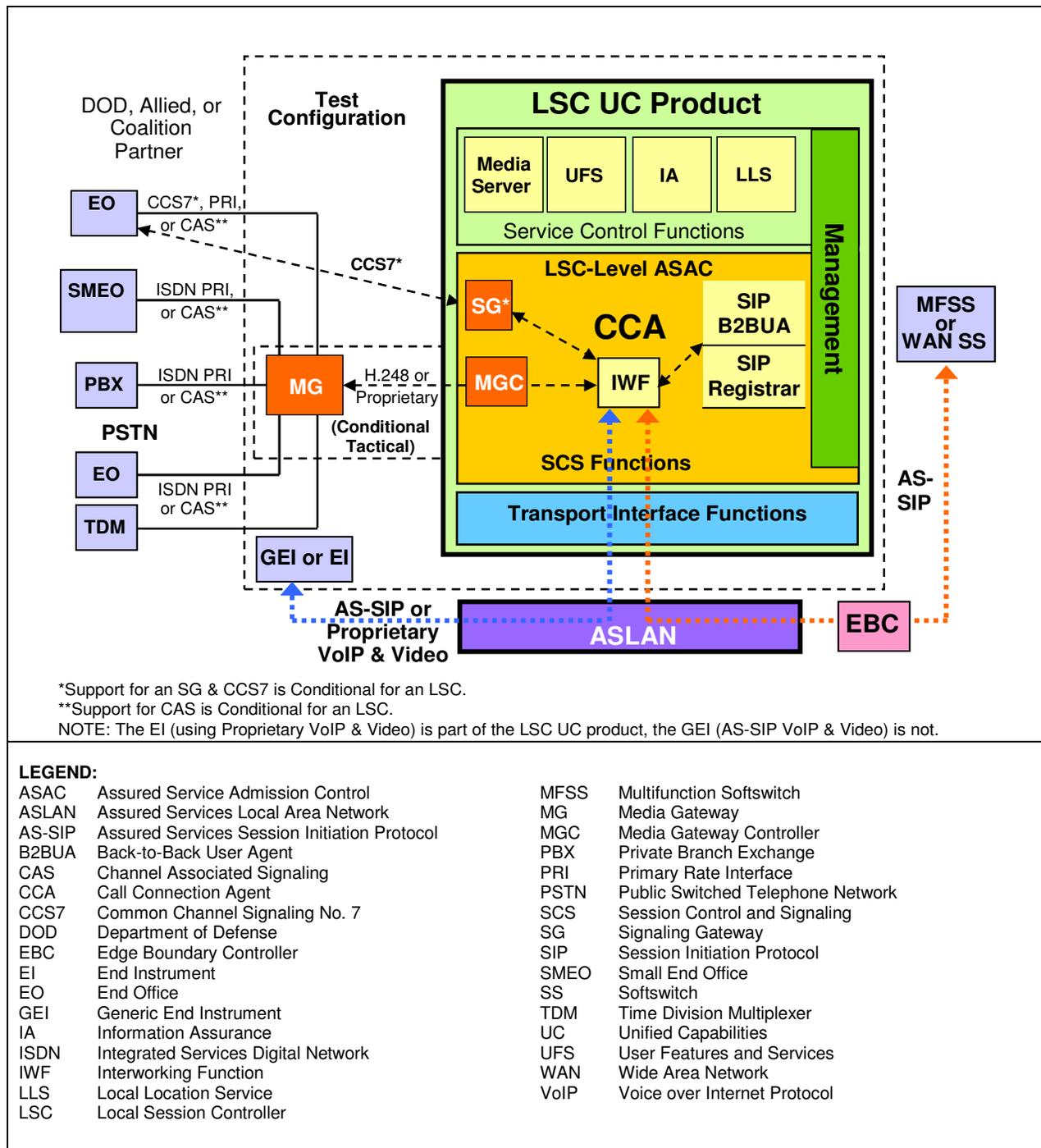
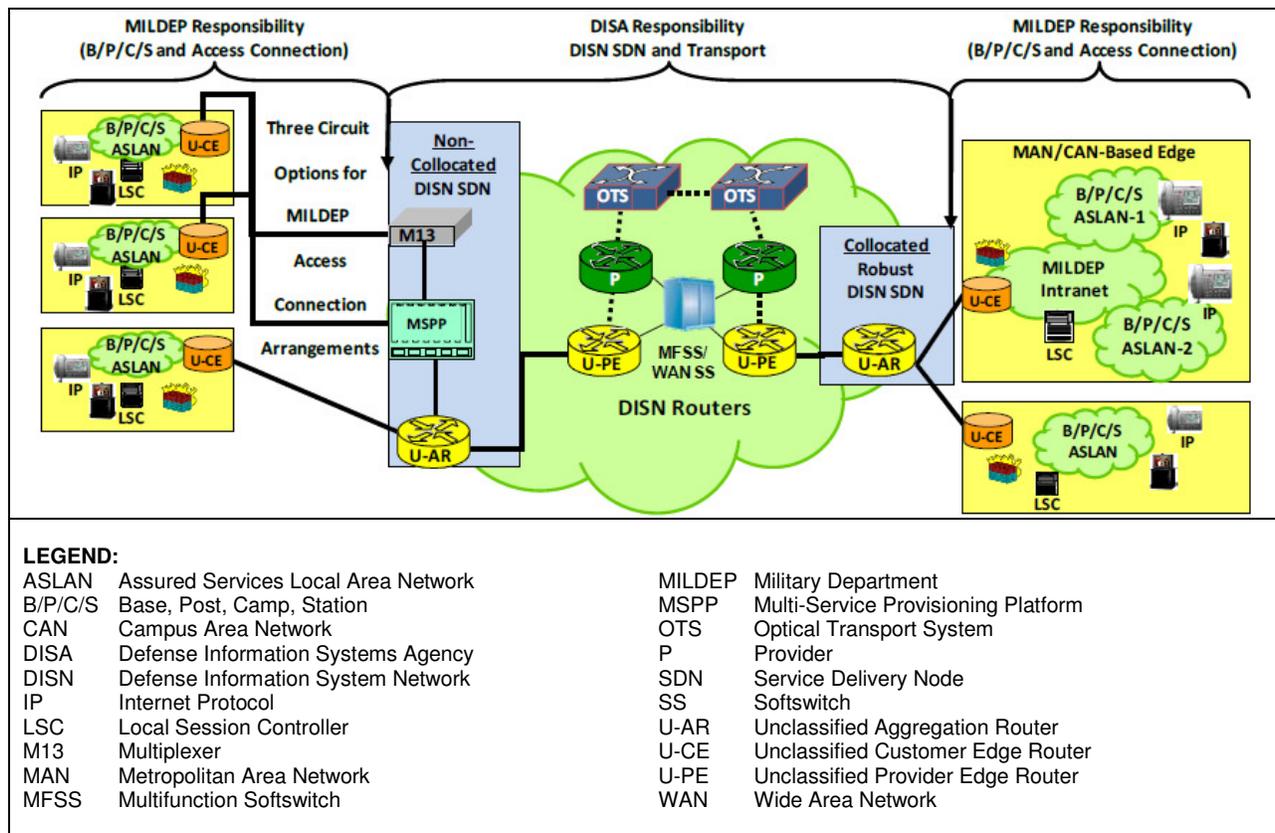


Figure 2-1. LSC Functional Reference Model



**Figure 2-2. UC Notional Operational Network Architecture**

**7. INTEROPERABILITY REQUIREMENTS.** The interface, Capability Requirements (CR) and Functional Requirements (FR), Information Assurance (IA), and other requirements for LSCs are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of Reference (c).

**7.1 Interfaces.** The SUT uses the external interfaces to connect to the Global Information Grid network and other Unified Capabilities Requirements (UCR) products. Table 2-1, shows the physical interfaces supported by the SUT. The table documents the physical interfaces and the associated standards.

**Table 2-1. SUT Interface Requirements**

Interface	Critical	UCR Reference	Criteria	Remarks
<b>Line Interfaces</b>				
10Base-X	Yes	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3i and 802.3j	
100Base-X	Yes	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4,10, 13, and 16) and meet interface criteria for 802.3.u	

**Table 2-1. SUT Interface Requirements (continued)**

Interface	Critical	UCR Reference	Criteria <sup>1</sup>	Remarks
<b>Line Interfaces (continued)</b>				
1000Base-X	No	5.3.2.6.3	Support minimum threshold CRs/FRs (2, 4, 10, 13, and 16) and meet interface criteria for 802.3z.	
2-wire analog	Yes	5.3.2.6.1.6	Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for analog.	Applies to 2-wire secure and non-secure analog instruments, fax, and modem.
BRI	No	5.3.2.6.1.8	Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for BRI	Applies to voice and clearmode data.
<b>External Interfaces</b>				
10Base-X	No <sup>2</sup>	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3i and 802.3j	
100Base-X	No <sup>2</sup>	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3u	
1000Base-X	No <sup>2</sup>	5.3.2.4.2	Support minimum threshold CRs/FRs (1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16) and meet interface criteria for 802.3z	
ISDN T1 PRI NI-2 (ANSI T1.619a)	Yes	5.3.2.4.3	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for ISDN T1 PRI (T1.619a)	Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2 (ANSI T1.607)	Yes	5.3.2.4.3	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for ISDN T1 PRI (NI-2)	Provides PSTN Connectivity.
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CCS7 (ANSI T1.619a)	
T1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CAS	T1 CAS with MLPP.
E1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for T1 CAS	E1 CAS with MLPP.
E1 PRI ITU-T Q.955.3	No	5.3.2.12.10	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for E1 PRI (Q.955.3)	Provides Legacy DSN European connectivity.
E1 PRI ITU-T Q.931	No	5.3.2.12.10	Support minimum threshold CRs/FRs (2, 3, 7, 8, 10, and 13) and meet interface criteria for E1 PRI (ITU-T Q.931)	Provides PSTN European connectivity.
<b>NM</b>				
10Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3i and 802.3j	
100Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	Support minimum threshold CR/FRs (16 and 17) and meet interface criteria for 802.3u	

**Table 2-1. SUT Interface Requirements (continued)**

<b>NOTES:</b>			
1. The CR/FR requirements are contained in Table 2-2. The CR/FR numbers represent a roll-up of UCR requirements. Enclosure 3 provides a list of more detailed requirements for security device products.			
2. Must provide a minimum of one of the listed interfaces.			
<b>LEGEND:</b>			
ANSI	American National Standards Institute	MFR1	Multi-Frequency Recommendation 1
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
CAS	Channel Associated Signaling	NI-2	National ISDN Standard 2
CR	Capability Requirement	NM	Network Management
CCS7	Common Channel Signaling	PRI	Primary Rate Interface
DP	Dial Pulse	PSTN	Public Switched Telephone Network
DSN	Defense Switched Network	Q.931	Signaling Standard for ISDN
DTMF	Dual Tone Multi-Frequency	Q.955.3	ISDN Signaling standard for E1 MLPP
E1	European Basic Multiplex Rate (2.048 Mbps)	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FR	Functional Requirement	T1.619a	SS7 and ISDN MLPP Signaling Standard For T1
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector		

**7.2 CR and FR.** The LSCs have required and conditional features and capabilities that are established by Sections 5.3.2, 5.3.4, 5.3.5, and 5.4 of the UCR. The SUT does not need to provide non-critical (conditional) requirements. If they are provided, they must function according to the specified requirements. The SUTs features and capabilities and its aggregated requirements are listed in Table 2-2. Detailed CR/FR requirements are provided in Table 3-1 of Enclosure 3.

**Table 2-2. LSC CRs and FRs**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR <sup>2</sup> Reference	Criteria
1	<b>Assured Services Product Features and Capabilities</b>			
	DSCP Packet Marking	Required	5.3.2.2.1.4	
	Voice Features and Capabilities	Required	5.3.2.2.2.1	
	Public Safety Features	Required	5.3.2.2.2.2	
	ASAC – Open Loop	Required	5.3.2.2.3	
	Signaling Protocols	Required	5.3.2.2.2.3	
Signaling Performance	Required	5.3.2.2.4		
2	<b>Registration, Authentication, and Failover</b>			
	Registration	Required	5.3.2.3.1	
	Failover	Required	5.3.2.3.2	
<b>Product Physical, Quality, and Environmental Factors</b>				
3	Availability	Required	5.3.2.5.2.1	
	Maximum Downtimes	Required	5.3.2.5.2.2	
	Loss of Packets	Required	5.3.2.5.4	

**Table 2-2. LSC CRs and FRs (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR <sup>2</sup> Reference	Criteria
4	<b>Voice EIs</b>			
	Tones and Announcements	Required	5.3.2.6.1.1	
	Audio Codecs	Required	5.3.2.6.1.2	
	VoIP PEI or AEI Audio Performance Requirements	Required	5.3.2.6.1.3	
	VoIP Sampling Standard	Required	5.3.2.6.1.4	
	Authentication To LSC	Required	5.3.2.6.1.5	
	Analog Telephone Support	Required	5.3.2.6.1.6	
	Softphones	Conditional	5.3.2.6.1.7	
	ISDN BRI	Conditional	5.3.2.6.1.8	
5	<b>Video EIs</b>			
	Video End Instrument	Required	5.3.2.6.2	
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	
6	<b>LSC Requirements</b>			
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	
	Local Location Server and Directory	Required	5.3.2.7.2.5	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	
	Loop Avoidance	Required	5.3.2.7.3	
7	<b>CCA Requirements</b>			
	CCA IWF Component	Required <sup>3</sup>	5.3.2.9.2.1	
	CCA MGC Component	Required <sup>3</sup>	5.3.2.9.2.2	
	SG Component	Conditional	5.3.2.9.2.3	
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required <sup>3</sup>	5.3.2.9.5.6	
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	
	CCA Interactions with the EBC	Required	5.3.2.10.4	
	CCA Support for Admission Control	Required	5.3.2.10.5	
	CCA Support for UFS	Required	5.3.2.10.6	
	CCA Support for IA	Required	5.3.2.10.7	
	CCA Interaction with EIs	Required	5.3.2.10.10	
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	
CCA Interactions with Service control Functions	Required	5.3.2.10.12		
	CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	

**Table 2-2. LSC CRs and FRs (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR <sup>2</sup> Reference	Criteria
8	<b>MG Requirements</b>			
	Role of MG In LSC	Required	5.3.2.12.3.1	
	MG Support for ASAC	Required	5.3.2.12.4.1	
	MG and IA Functions	Required	5.3.2.12.4.2	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	
	MG-EBC interactions	Required	5.3.2.12.4.5	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	
	MG Interaction with EIs	Required	5.3.2.12.4.8	
	MG support for User Features and Services	Required	5.3.2.12.4.9	
	MG Interface to TDM	Required <sup>4</sup>	5.3.2.12.5	
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	
	MG Interface to TDM PSTN in US	Required <sup>4</sup>	5.3.2.12.7	
	MG Interfaces to TDM PSTN OCONUS	Required <sup>4,5</sup>	5.3.2.12.8	
	MG Support for CCS7	Conditional	5.3.2.12.9	
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	
	MG Support for CAS Trunks	Required	5.3.2.12.11	
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	
	MG Echo Cancellation	Required	5.3.2.12.13	
MG Clock Timing	Required	5.3.2.12.14		
MGC-MG CCA Functions	Required	5.3.2.12.15		
MG V.150.1	Required	5.3.2.12.16		
MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17		
9	<b>SG Requirements</b>			
	SG and CCS7 network Interactions	Conditional	5.3.2.13.5.1	
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	
	SG Interworking Functions	Conditional	5.3.2.13.5.3	
10	<b>WWNDP Requirements</b>			
	WWNDP	Required	5.3.2.16	
	DSN WWNDP	Required	5.3.2.16.1	
11	<b>Commercial Cost Avoidance</b>			
	Commercial Cost Avoidance	Required	5.3.2.23	
12	<b>AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)</b>			
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	
13	<b>Precedence Call Diversion</b>			
	Precedence call Diversion	Required	5.3.2.25	

**Table 2-2. LSC CRs and FRs (continued)**

<b>CR/FR ID</b>	<b>Capability/Function</b>	<b>Applicability<sup>1</sup></b>	<b>UCR<sup>2</sup> Reference</b>	<b>Criteria</b>
<b>14</b>	<b>Attendant Station Features<sup>6</sup></b>			
	Precedence and Preemption	Required	5.3.2.26.1	
	Call Display	Required	5.3.2.26.2	
	Class of Service Override	Required	5.3.2.26.3	
	Busy Override and Busy Verification	Required	5.3.2.26.4	
	Night service	Required	5.3.2.26.5	
	Automatic Recall of Attendant	Required	5.3.2.26.6	
	Calls in Queue to the Attendant	Required	5.3.2.26.7	
<b>15</b>	<b>AS-SIP Requirements</b>			
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP Els	Required	5.3.4.7	
	SIP Session Keep-Alive Timer	Required	5.3.4.8	
	Session Description Protocol	Required	5.3.4.9	
	Precedence and Preemption	Required	5.3.4.10	
	Video Telephony – General Rules	Required	5.3.4.12	
	Calling Services	Required	5.3.4.13	
	SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances	Required	5.3.4.14	
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	
	SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	
	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	
Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18		
	Supplementary Services	Required	5.3.4.19	
<b>16</b>	<b>IPv6 Requirements</b>			
	Product Requirements	Required	5.3.5.4	
<b>17</b>	<b>NM</b>			
	LSC Management Function	Required	5.3.2.7.2.6	
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	
	General Management requirements	Required	5.3.2.17.2	
	Requirement for FCAPS Management	Required	5.3.2.17.3	
	NM requirements of Appliance Functions	Required	5.3.2.18	
	Accounting Management	Required	5.3.2.19	

**Table 2-2. LSC CRs and FRs (continued)**

**NOTES:**

1. Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in Enclosure 3.
2. Detailed requirements and associated criteria for LSCs are listed in Table 3-1 of Enclosure 3.
3. The LSC must meet T1 PRI (ANSI T1.619a and NI-2) CCA IWF. The T1 CAS and T1 CCS7 CCA IWF are conditional.
4. The LSC must meet TDM requirements for T1 PRI (ANSI T1.619a and NI-2). The TDM requirements for T1 CAS and T1 CCS7 are conditional.
5. The E1 requirements for OCOUNUS are conditionally required for deployments in Europe.

**LEGEND:**

AEI	AS-SIP End Instrument	LoC	Letter of Compliance
AS	Assured Services	LSC	Local Session Controller
ASAC	Assured Services Admission Control	MG	Media Gateway
AS-SIP	Assured Services Session Initiation Protocol	MGC	Media Gateway Controller
BRI	Basic Rate Interface	NM	Network Management
CAS	Channel Associated Signaling	NMS	Network Management System
CCA	Call Connection Agent	OCOUNUS	Outside the Continental United States
CCS7	Common Channel Signaling 7	PBAS	Precedence-Based Assured Service
CR	Capabilities Requirement	PEI	Proprietary End Instrument
DSCP	Differentiated Services Code Point	PRI	Primary Rate Increase
DSN	Defense Switched Network	PSTN	Public Switch Telephone Network
EBC	Edge Boundary Controller	SG	Signaling Gateway
EI	End Instrument	SIP	Session Initiation Protocol
FCAPS	Fault, Configuration, Accounting, Performance, and Security	SS7	Signaling System Number 7
FR	Functional Requirement	SUT	System Under Test
IA	Information Assurance	T1	1.544 Mbps North American trunk standard
IAD	Integrated Access Device	TDM	Time Division Multiplexing
ID	Identification	UCR	Unified Capabilities Requirements
IP	Internet Protocol	UFS	User Features and Services
IPv6	Internet Protocol version 6	VoIP	Voice over Internet Protocol
ISDN	Integrated Services Digital Network	VVoIP	Voice and Video over Internet Protocol
IWF	Interworking Function	WAN	Wide Area Network
		WWNDP	World Wide Numbering and Dialing Plan

**7.3 IA.** Table 2-3 details the IA requirements applicable to an LSC.

**Table 2-3. LSC IA Requirements**

Requirement	Applicability (See note.)	UCR Reference	Criteria
General Requirements	Required	5.4.6.2	Detailed requirements and associated criteria for LSCs are listed in the IA Assessment, Reference (e).
Authentication	Required	5.4.6.2.1	
Integrity	Required	5.4.6.2.2	
Confidentiality	Required	5.4.6.2.3	
Non-Repudiation	Required	5.4.6.2.4	
Availability	Required	5.4.6.2.5	

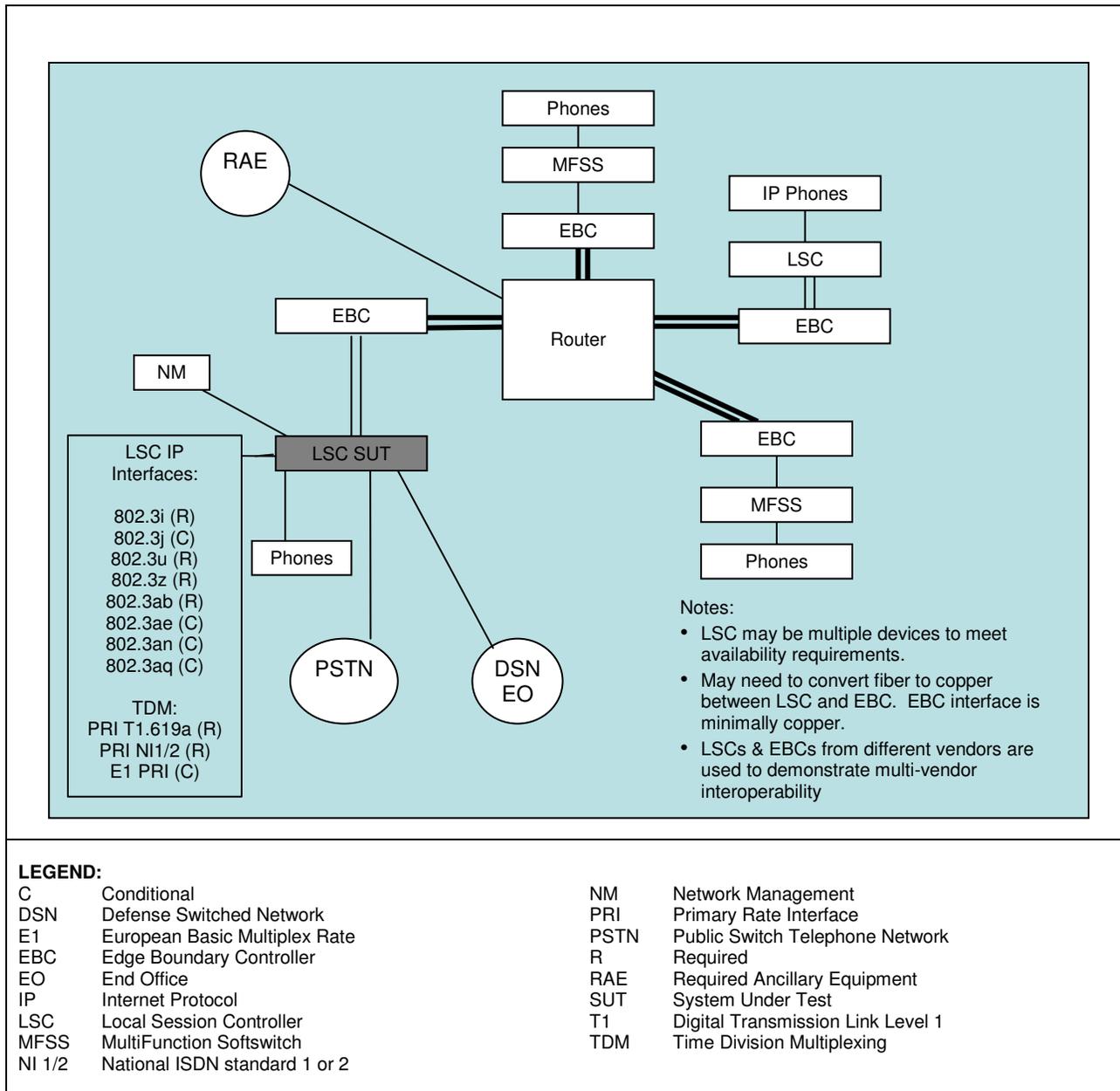
**NOTE:** Annotation of 'required' refers to high level requirement category. Applicability of each sub-requirement is provided in Reference (e).

**LEGEND:**

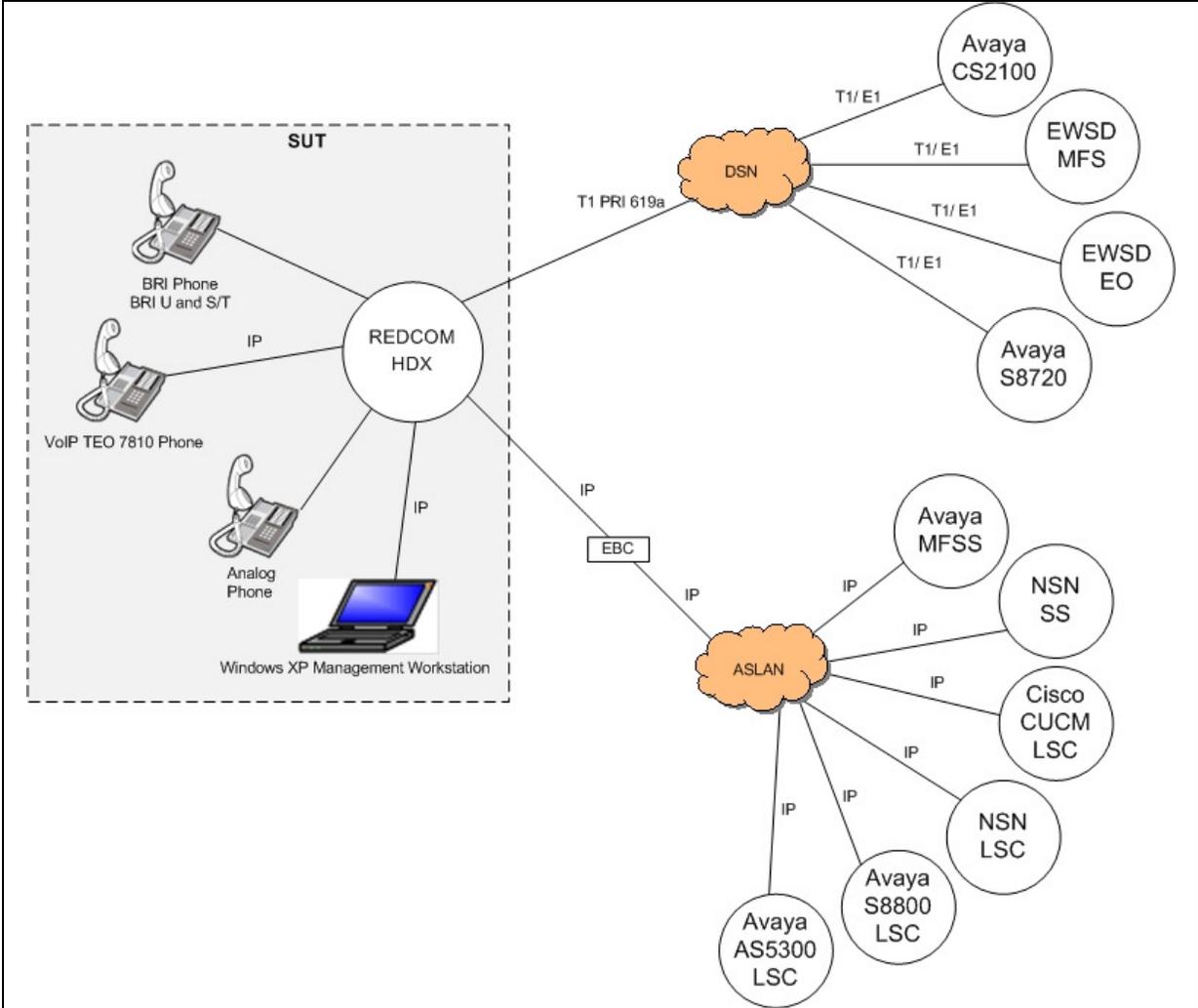
IA	Information Assurance	LSC	Local Session Controller
IATP	IA Test Plan	UCR	Unified capabilities Requirements

**7.4 Other.** None

**8. TEST NETWORK DESCRIPTION.** The SUT was tested at JITC in a manner and configuration similar to that of a notional operational environment. Figure 2-3 depicts the minimum test architecture for testing LSCs. Figure 2-4 depicts the SUT test configuration and the system’s required functions and features that were conducted using the test configurations.



**Figure 2-3. LSC Minimum Test Architecture**



**LEGEND:**

ASLAN	Assured Services Local Area Network	MFSS	Multifunction Soft Switch
BRI	Basic Rate Interface	NSN	Nokia Siemens Networks
CM	Communication Manager	OS	Operating System
DSN	Defense Switched Network	POTS	Plain Old Telephone Service
E1	European Basic Multiplex Rate	PRI	Primary Rate Interface
EBC	Edge Boundary Controller	SP	Service Pack
EO	End Office	SS	Soft Switch
EWSD	Elektronisches Wählsystem Digital	S/T	ISDN BRI four-wire interface
HDX	High Density Exchange	SUT	System Under Test
IP	Internet Protocol	T1	Digital Transmission Link Level 1 (1.544 Mbps)
JITC	Joint Interoperability Test Command	U	ISDN BRI two-wire interface
LSC	Local Session Controller	VoIP	Voice over IP
MFS	Multifunction Switch		

**Figure 2-4. REDCOM HDX SUT Test Configuration**

**9. SYSTEM CONFIGURATIONS.** Table 2-4 provides the system configurations and hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine its interoperability capability with associated network devices and network traffic.

**Table 2-4. Tested System Configurations**

System Name		Software Release
Avaya CS2100 MFS		XA-Core SE09.1
Avaya CS2100/Aura AS5300 MFSS		CS2100 XA-Core SE09.1; AS5300 2.0 Patch Bulletin 18
Avaya Aura S8800 LSC		6.0.1 (00.1.510.1) Service Pack 19211
Avaya S8720 SMEO		(CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)
Cisco CUCM LSC		Version 8.0(2)
Nokia Siemens Networks HiQ8000 WAN SS		Software Release 13.90.02.10 Patch Set (PS)12 Patch (P) 90
Nokia Siemens Networks HiQ8000 LSC		Software Release 13.90.02.10 Patch Set (PS) 14 Patch (P) 102
Nokia Siemens Networks EWSD		19d Patch Set 46
REDCOM SLICE 2100		V4.0A (R3P7)
SUT LSC		
REDCOM HDX		V4.0A (R3P7)
TEO Tone Commander PEI 7810		V05.04.02.80
Polycom VVX 1500 (See note.)		BootBlock 3.0.1 (17960-001) SIP Version 4.0.1.13681
REDCOM HDX Components		
Cards		
Part Number	Part Description	Firmware/Software
MA0656-003	Dual Processor Board	V4.0A (R3P7)
MA0656-003	Dual Processor Board	V4.0A (R3P7)
MA0648-002	TSI Card	N/A
MA0703-004	Universal Service Controller	Cdi40a37
MA0724-301	Expanded Dynamic Line	V2.6a
MA0724-301	Expanded Dynamic Line	V2.6a
MA0530-322	BRI Card	Bri100r3p7
MA0530-322	BRI Card	Bri100r3p7
MA0732-001	Media Server Controller	Msc40a37
MA0368-102	Attendant Card	N/A
MA0683-144	Multi-E1/T1 Module	Cdi40a37
MA0683-144	Multi-E1/T1 Module	Cdi40a37
MA0728-163	Universal Clock Synchronizer	V3.2a
MA0366-101	Attendant Card	N/A
MA0504-011	Ring Generator	N/A

**Table 2-4. Tested System Configurations (continued)**

<b>NOTE:</b> The SUT offers video; however, it is not covered under this certification. JITC was only able to conduct limited video testing on the SUT with Polycom video EIs during the original test event. Due to the fact that their video EI is not IPv6 capable and did not offer a GA load it did not meet this requirement. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.			
<b>LEGEND:</b>			
BRI	Basic Rate Interface	PEI	Proprietary End Instrument
CM	Communication Manager	R	Release
CUCM	Cisco Unified Communication Manager	SIP	Session Initiation Protocol
E1	European Basic Multiplex Rate (2.048 Mbps)	SMEO	Small End Office
EWSD	Elektronisches Wählsystem Digital	SUT	System Under Test
HDX	High Density Exchange	T1	Digital Transmission Link Level 1 (1.544Mbps)
LSC	Local Session Controller	TSI	Time Slot Interchange
MFS	Multifunction Switch	V	Version
MFSS	Multifunction Soft Switch	VVX	Voice and Video Experience
N/A	Not Applicable	WAN SS	Wide Area Network Soft Switch
P	Patch		

**10. TESTING LIMITATIONS.** The JITC test team noted the following testing limitations including the impact they may have on interpretation of the results and conclusions. Any untested requirements are also included in the testing limitations.

**a. Proprietary End Instruments (PEI).** The JITC did not test PEIs for video requirements during this test event. JITC was only able to conduct limited video testing on the SUT with the Polycom video EI's during the original test event. Also, the testing that was conducted with Polycom during the original test event was on a software load that will never be a General Available (GA) load that can be purchased and it does not support IPv6.

**b. IPv 6.** The JITC tested the SUT to verify its IPv6 capabilities intra-enclave. The vendor submitted an IPv6 letter of compliance (LoC) with noted discrepancies. Open Test Discrepancy Reports (TDR) were adjudicated by Defense Information Systems Agency (DISA) as having a minor operational impact based on vendor's submitted Plan of Action and Milestones (POA&M) to fix by 1 July 2012. Paragraph 11 below provides detailed information about IPv6 results.

**c. Network Management.** The JITC did not test the SUT's Network Management (NM) capabilities to meet UCR requirements. The vendor submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA as having a minor operational impact based on vendor's submitted POA&M to fix by 1 February 2011. The JITC's evaluation of the SUT's NM capabilities is provided in paragraph 11 below.

**11. INTEROPERABILITY EVALUATION RESULTS.** The SUT meets the critical interoperability requirements for an LSC in accordance with the UCR and is certified for joint use with other UC Products listed on the UC Approved Products List. Additional discussion regarding specific testing results is located in subsequent paragraphs.

**11.1 Interfaces.** The SUT met line interface requirements for 10/100 Base-X interfaces. These IP line interfaces were met through use of PEIs (voice only) and 2-wire analog phones. The 2-wire analog phones are supported via an Integrated Access Device. The SUT met the external interface requirements for 10/100Base-X (AS-SIP) and Integrated Services Digital Network (ISDN) PRI for both ANSI T1.619a MLPP and National ISDN-2 (NI-2) commercial. The JITC did not test the other conditional interfaces. The interface status of the SUT is provided in Table 2-5.

**Table 2-5. SUT Interface Interoperability Status**

Interface	Critical	UCR Reference	Threshold CR/FR <sup>1</sup>	Status	Remarks <sup>2</sup>
<b>Line Interfaces</b>					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs (voice only).
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs (voice only).
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, 16	Not Tested	This interface is not offered by the SUT.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, 13,	Certified	Met threshold CRs/FRs for 2-wire analog instruments. Applies to 2-wire secure and non-secure analog instruments, fax, and modem.
BRI	No	5.3.2.6.1.8	2, 4, 10, 13	Certified	Met threshold CRs/FRs for BRI instruments. Applies to voice and clearmode data.
<b>External Interfaces</b>					
10Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No <sup>3</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, 16	Not Tested	This interface is not offered by the SUT.
ISDN T1 PRI NI-2 (ANSI T1.619a)	Yes	5.3.2.4.3	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs. Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2 (ANSI T1.607)	Yes	5.3.2.4.3	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs. Provides PSTN Connectivity
T1 CCS7 ANSI T1.619a	No	5.3.2.12.9	2, 3, 7, 8, 10, 13	Not Tested	This interface is offered by the SUT; however, it was not tested.
T1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for T1 CAS.
E1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for T1 CAS.
E1 PRI ITU-T Q.955.3	No <sup>4</sup>	5.3.2.12.10	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for E1 PRI ITU-T Q.955.3. Provides Legacy DSN connectivity in Europe.
E1 PRI ITU-T Q.931	No <sup>4</sup>	5.3.2.12.10	2, 3, 7, 8, 10, 13	Certified	Met threshold CRs/FRs for E1 PRI ITU-T Q.931. Provides PSTN connectivity in Europe.

**Table 2-5. SUT Interface Interoperability Status (continued)**

Interface	Critical	UCR Reference	Threshold CR/FR <sup>1</sup>	Status	Remarks <sup>2</sup>
<b>NM</b>					
10Base-X	No <sup>3</sup>	5.3.2.4.4 5.3.2.7.2.8	16, 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No <sup>3</sup>	5.3.2.4.4 5.3.2.7.2.8	16, 17	Certified	Met threshold CRs/FRs. Verified via LoC.
<b>NOTES:</b>					
1. The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 2-6. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.					
2. Detailed information pertaining to open TDRs and associated operational impacts is in paragraph 11.					
3. The SUT must provide a minimum of one of the listed interfaces.					
4. This interface is conditionally required for deployment in Europe.					
<b>LEGEND:</b>					
ANSI	American National Standards Institute		Mbps	Megabits per second	
AS-SIP	Assured Services Session Initiation Protocol		MFR1	Multi-Frequency Recommendation 1	
BRI	Basic Rate Interface		MLPP	Multi-Level Precedence and Preemption	
CAS	Channel Associated Signaling		NI-2	National ISDN-2	
CCS7	Common Channel Signaling 7		NM	Network Management	
CR	Capability Requirement		PEI	Proprietary End Instrument	
DP	Dial Pulse		PRI	Primary Rate Interface	
DSN	Defense Switched Network		PSTN	Public Switched Telephone Network	
DSS1	Digital Subscriber Signaling 1		Q.931	Signaling Standard for ISDN	
DTMF	Dual Tone Multi-Frequency		Q.955.3	ISDN Signaling Standard for E1 MLPP	
E1	European Basic Multiplex Rate (2.048 Mbps)		SS7	Signaling System 7	
FR	Functional Requirement		SUT	System Under Test	
ID	Identification		T1	Digital Transmission Link Level 1 (1.544 Mbps)	
IEEE	Institute of Electrical and Electronics Engineers		T1.607	ISDN – Layer 3 Signaling Specification For Circuit Switched Bearer Service For DSS1	
ISDN	Integrated Services Digital Network		T1.619a	SS7 and ISDN MLPP Signaling Standard For T1	
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector		TDR	Test Discrepancy Report	
LoC	Letter of Compliance		UCR	Unified Capabilities Requirements	

**11.2 CR and FR.** The SUT CR and FR status is depicted in Table 2-6. Detailed CR/FR requirements are provided in Enclosure 3, Table 3-1. A summary of the SUT's ability to meet UCR requirements are provided in the sub-paragraphs below. All requirements and associated references were derived from UCR 2008, Change 2. Discrepancies discussed below were adjudicated as having a minor operational impact based on vendor submission of a POA&M.

**Table 2-6. SUT CRs and FRs Status**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
1	<b>Assured Services Product Features and Capabilities</b>				
	DSCP Packet Marking	Required	5.3.2.2.1.4	Partially Met <sup>2</sup>	
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Partially Met <sup>3</sup>	
	Public Safety Features	Required	5.3.2.2.2.2	Met	
	ASAC – Open Loop	Required	5.3.2.2.2.3	Partially Met <sup>4</sup>	
	Signaling Protocols	Required	5.3.2.2.3	Met	
2	<b>Registration, Authentication, and Failover</b>				
	Registration	Required	5.3.2.3.1	Met	
	Failover	Required	5.3.2.3.2	Met	
3	<b>Product Physical, Quality, and Environmental Factors</b>				
	Availability	Required	5.3.2.5.2.1	Met	
	Maximum Downtimes	Required	5.3.2.5.2.2	Met	
4	<b>Voice End Instruments</b>				
	Tones and Announcements	Required	5.3.2.6.1.1	Met	
	Audio Codecs	Required	5.3.2.6.1.2	Partially Met <sup>5</sup>	
	VoIP PEI or AEI Audio Performance	Required	5.3.2.6.1.3	Partially Met <sup>5</sup>	
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Met	
	Authentication to LSC	Required	5.3.2.6.1.5	Met	
	Analog Telephone Support	Required	5.3.2.6.1.6	Met	
	Softphones	Conditional	5.3.2.6.1.7	Not Tested <sup>7</sup>	
5	<b>Video End Instruments</b>				
	Video End Instrument	Required	5.3.2.6.2	Not Met <sup>8</sup>	
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Met <sup>8</sup>	
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Met <sup>8</sup>	
6	<b>LSC Requirements</b>				
	PBAS/ASAC Requirements	Required	5.3.2.7.2.1	Met	
	Calling Number Delivery Requirements	Required	5.3.2.7.2.2	Met	
	LSC Signaling Requirements	Required	5.3.2.7.2.3	Met	
	Service Requirements under Total Loss of WAN Transport	Required	5.3.2.7.2.4	Met	
	Local Location Server and Directory	Required	5.3.2.7.2.5	Met	
	LSC Transport Interface Functions	Required	5.3.2.7.2.7	Met	
	LSC to PEI, AEI, and Operator Console Status Verification	Required	5.3.2.7.2.10	Partially Met <sup>9</sup>	
7	<b>Call Connection Agent Requirements</b>				
	CCA IWF Component	Required	5.3.2.9.2.1	Met	
	CCA MGC Component	Required	5.3.2.9.2.2	Met	
	SG Component	Conditional	5.3.2.9.2.3	Not Tested <sup>10</sup>	
	CCA-IWF Support for AS-SIP	Required	5.3.2.9.5.1	Met	
	CCA-IWF Support for SS7	Conditional	5.3.2.9.5.2	Not Tested <sup>10</sup>	

**Table 2-6. SUT CRs and FRs Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
7	<b>Call Connection Agent Requirements (continued)</b>				
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Met	
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested <sup>10</sup>	
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Met	
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met	
	CCA Preservation of Call Ringing State during Failure Conditions	Required	5.3.2.9.6	Met	
	CCA Interactions with Transport Interface Functions	Required	5.3.2.10.3	Met	
	CCA Interactions with the EBC	Required	5.3.2.10.4	Met	
	CCA Support for Admission Control	Required	5.3.2.10.5	Met	
	CCA Support for UFS	Required	5.3.2.10.6	Met	
	CCA Support for IA	Required	5.3.2.10.7	Met	
	CCA Interaction with EIs	Required	5.3.2.10.10	Met	
	CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met <sup>8</sup>	
	CCA Interactions with Service control Functions	Required	5.3.2.10.12	Met	
CCA Interworking between AS-SIP and SS7	Conditional	5.3.2.11	Not Tested <sup>10</sup>		
8	<b>MG Requirements</b>				
	Role of MG In LSC	Required	5.3.2.12.3.1	Partially Met <sup>11</sup>	
	MG Support for ASAC	Required	5.3.2.12.4.1	Met	
	MG and IA Functions	Required	5.3.2.12.4.2	Met	
	MG Interaction with Service Control Function	Required	5.3.2.12.4.3	Met	
	MG Interactions with IP Transport Interface Functions	Required	5.3.2.12.4.4	Met	
	MG-EBC interactions	Required	5.3.2.12.4.5	Met	
	MG IP-Based PSTN Interface Requirements	Conditional	5.3.2.12.4.7	Not Tested <sup>11</sup>	
	MG Interaction with EIs	Required	5.3.2.12.4.8	Met	
	MG support for User Features and Services	Required	5.3.2.12.4.9	Met	
	MG Interface to TDM	Required	5.3.2.12.5	Met	
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested <sup>10</sup>	
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Met	
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Met	
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested <sup>10</sup>	
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Met	
	MG Support for CAS Trunks	Conditional	5.3.2.12.11	Met	
MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met <sup>11</sup>		
MG Echo Cancellation	Required	5.3.2.12.13	Met		
MG Clock Timing	Required	5.3.2.12.14	Met		

**Table 2-6. SUT CRs and FRs Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
8	<b>MG Requirements (continued)</b>				
	MGC-MG CCA Functions	Required	5.3.2.12.15	Met	
	MG ITU-T V.150.1	Required	5.3.2.12.16	Partially Met <sup>12</sup>	
	MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17	Met	
9	<b>SG Requirements</b>				
	SG and CCS7 network Interactions	Conditional	5.3.2.13.5.1	Not Tested <sup>10</sup>	
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested <sup>10</sup>	
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested <sup>10</sup>	
10	<b>WWNDP Requirements</b>				
	WWNDP	Required	5.3.2.16	Met	
	DSN WWNDP	Required	5.3.2.16.1	Met	
11	<b>Commercial Cost Avoidance</b>				
	Commercial Cost Avoidance	Required	5.3.2.23	Not Tested <sup>13</sup>	
12	<b>AS-SIP Based for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices)</b>				
	AS-SIP Requirements for External Interfaces	Conditional	5.3.2.24	Not Tested <sup>10</sup>	
13	<b>Precedence Call Diversion</b>				
	Precedence Call Diversion	Required	5.3.2.25	Met	
14	<b>Attendant Station Features</b>				
	Precedence and Preemption	Required	5.3.2.26.1	Met	
	Call Display	Required	5.3.2.26.2	Met	
	Class of Service Override	Required	5.3.2.26.3	Met	
	Busy Override and Busy Verification	Required	5.3.2.26.4	Met	
	Night service	Required	5.3.2.26.5	Met	
	Automatic Recall of Attendant	Required	5.3.2.26.6	Met	
	Calls in Queue to the Attendant	Required	5.3.2.26.7	Met	
15	<b>AS-SIP Requirements</b>				
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required	5.3.4.7	Met	
	SIP Session Keep-Alive Timer	Required	5.3.4.8	Met	
	Session Description Protocol	Required	5.3.4.9	Met	
	Precedence and Preemption	Required	5.3.4.10	Met	
	Video Telephony – General Rules	Required	5.3.4.12	Not Met <sup>8</sup>	
	Calling Services	Required	5.3.4.13	Met	
	SIP Translation Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.14	Met	
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Met	
SIP Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.16	Met		

**Table 2-6. SUT CRs and FRs Status (continued)**

CR/FR ID	Capability/Function	Applicability <sup>1</sup>	UCR Reference	Status	Remarks
<b>AS-SIP Requirements (continued)</b>					
15	Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances	Required	5.3.4.17	Met	
	Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances	Required	5.3.4.18	Met	
	Supplementary Services	Required	5.3.4.19	Partially Met <sup>3</sup>	
<b>IPv6 Requirements</b>					
16	Product Requirements	Required	5.3.5.4	Partially Met <sup>14</sup>	
<b>NM</b>					
17	LSC Management Function	Required	5.3.2.7.2.6	Met	
	VVoIP NMS Interface Requirements	Required	5.3.2.4.4	Met	
	General Management requirements	Required	5.3.2.17.2	Met	
	Requirement for FCAPS Management	Required	5.3.2.17.3	Met	
	NM requirements of Appliance Functions	Required	5.3.2.18	Partially Met <sup>15</sup>	
	Accounting Management	Required	5.3.2.19	Partially Met <sup>15</sup>	

**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. The SUT correctly class marks calls based on the precedence level dialed; however, the DSCP values are not configurable from 0-63 as required by UCR 2008, Change 2. This was adjudicated by DISA as having a minor operational impact based on the vendor's submitted POA&M to fix by 1 July 2012.
3. The SUT met all critical CRs & FRs with one minor exception: In a 3-way conference, the SUT does not classmark the conference at the highest precedence level of each leg of the conference. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
4. The SUT does not support separate ASAC counts for video and voice. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.
5. The SUT does not support ITU-T G.722.1 voice codec. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by December 2012.
6. The SUT only supports voice PEIs. The vendor did not support AEI voice capability. This was adjudicated by DISA to have a minor operational impact since there were no certified AEI video end instruments on the UC APL.
7. The vendor did not provide soft phones with the SUT for interoperability testing; therefore, they were not tested. Soft phones are a conditional requirement.
8. The SUT offers video; however, it is not covered under this certification. JITC only conducted limited video testing on the SUT with Polycom video EIs. Due to the fact that their video EI is not IPv6 capable and did not offer a GA load it did not meet this requirement. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.
9. The SUT does not comply with the requirement to be configurable to set interval times to verify status of its IP EI's with the default set at five minutes. This was adjudicated by DISA as having a minor operational impact with the intent to change the requirement to conditional in next UCR update.
10. This is a conditional requirement and was not tested.
11. The SUT met this requirement with VoIP PEIs only.
12. The SUT does not fully comply with the requirements for ITU-T V.150.1. For FAX (ITU-T T.38 transitions) the LSC maintains the session protocol (UDP). However, the port is changed when moving from VoIP to FoIP. The LSC sends a REINVITE (per MGC transitioning rules) with the new port number; therefore, SIP compliant devices (including the EBC) should follow the port change. The port selection is currently configurable for MoIP because MoIP does not use the REINVITE method for transition. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

**Table 2-6. SUT CRs and FRs Status (continued)**

**NOTES continued:**

13. The SUT does not currently support LDAP V3 for CCA. Transition to the LDAP format is currently in progress and is expected to be functional by late June 2011. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

14. The vendor submitted an IPv6 LoC with noted discrepancies which include: The SUT is not fully compliant with IPv6 neighbor discovery IAW RFC 4861. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

15. The vendor submitted an NM LoC with the following noted discrepancies: The SUT does not fully comply with the CDR and QoS requirements per UCR 2008, Change 2, section 5.3.2.19.2.1.x. The SUT does not comply with the Management Requirements for ASAC per UCR 2008, Change 2, section 5.3.2.18.2. These were adjudicated by DISA as having a minor operational impact with vendor POA&M to fix by 1 July 2012.

**LEGEND:**

AEI	AS-SIP End Instrument	JITC	Joint Interoperability Test Command
Apl	Approved Products List	LDAP	Lightweight Directory Access Protocol
AS	Assured Services	LoC	Letter of Compliance
ASAC	Assured Services Admission Control	LSC	Local Session Controller
AS-SIP	Assured Services Session Initiation Protocol	MG	Media Gateway
BRI	Basic Rate Interface	MGC	Media Gateway Controller
CAS	Channel Associated Signaling	MoIP	Modem over Internet Protocol
CCA	Call Connection Agent	NM	Network Management
CCS7	Common Channel Signaling 7	NMS	Network Management System
CDR	Call Detail Records	OCONUS	Outside the Continental United States
CR	Capabilities Requirement	PBAS	Precedence-Based Assured Service
DISA	Defense Information Systems Agency	PEI	Proprietary End Instrument
DSCP	Differentiated Services Code Point	POA&M	Plan of Actions and Milestones
DSN	Defense Switched Network	PRI	Primary Rate Interface
EBC	Edge Boundary Controller	PSTN	Public Switched Telephone Network
EI	End Instrument	QoS	Quality of Service
FCAPS	Fault, Configuration, Accounting, Performance, and Security	RFC	Request for Comments
FoIP	Fax over Internet Protocol	SG	Signaling Gateway
FR	Functional Requirement	SIP	Session Initiation Protocol
GA	General Available	SS7	Signaling System Number 7
HDX	High Density Exchange	SUT	System Under Test
IA	Information Assurance	TDM	Time Division Multiplexing
IAW	in accordance with	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements
IP	Internet Protocol	UDP	User Datagram Protocol
IPv6	Internet Protocol version 6	UFS	User Features and Services
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	VVoIP	Voice and Video over Internet Protocol
IWF	Interworking Function	WAN	Wide Area Network
		WWNDP	World Wide Numbering and Dialing Plan

**a. Assured Services Product Features and Capabilities.**

(1) Differentiated Services Code Point Packet (DSCP) Marking. As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. The exact DSCP method used shall comply with UCR 2008, Change 2, paragraph 5.3.3.3.2. The SUT met all DSCP Packet Marking requirements for IP version 4 (IPv4) and IPv6 with one exception: The SUT correctly class marks calls based on the precedence level dialed; however, the DSCP values are not configurable from 0-63 as required by UCR 2008, Change 2. This was adjudicated by DISA as having a minor operational impact based on vendor's submitted POA&M to fix by 1 July 2012.

(2) Voice Features and Capabilities. The LSC must provide all of the features listed in Table 5.3.2.2-1 of the UCR. The SUT met all critical CRs and FRs with one exception: In a 3-way conference, the SUT does not classmark the conference at the highest precedence level of each leg of the conference. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

(3) Public Safety Features. The LSC must provide basic emergency service (911), tracing of terminating calls, outgoing call tracing, and tracing of a call in progress. The SUT met all Public Safety Features requirements.

(4) Assured Services Admission Control (ASAC) – Open Loop. The LSC must meet the ASAC requirements for the LSC and the MFSS. In the execution of ASAC, certain procedures need to be followed, such as (a) actions to be taken if a precedence session request cannot be completed because existing sessions are at equal or higher precedence, or (b) tones to be generated when a session is preempted. The SUT met all ASAC requirements with one exception: The SUT does not support separate ASAC counts for video and voice. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

(5) Signaling Protocols. The LSC must use appropriate signaling for specific trunk types. The control/management protocol between the Proprietary EI (PEI) and the LSC is, in general, proprietary. The signaling protocol used on UC IP trunks is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements. The LSC and this media gateway (MG) are required to support ANSI T1.619a PRI signaling to the Defense Information System Network (DISN). The SUT met all Signaling Protocol requirements. The SUT met this required interface and the following conditional DSN interfaces via its MG: ITU-T Q.931 and ITU-T Q.955.3 European Basic Multiplex Rate (E1) PRI, Digital Transmission Link Level 1 (T1), and E1 CAS.

(6) Signaling Performance. The LSC has conditional requirements for call setup and tear-down times. The SUT met all signaling performance requirements.

#### **b. Registration, Authentication, and Failover.**

(1) Registration. Registration and authentication between the LSC and EIs shall follow the requirements set forth in UCR 2008, Change 2, Section 5.4, of IA Requirements. This feature was tested by IA and is covered in the IA report (see paragraph 11.3).

(2) Failover. The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS. The LSC does not have dual power supplies, it does not properly switch to the backup processor when the Ethernet connection fails, and it has a single point of failure that could cause a loss of voice and/ or video service. This was adjudicated as minor based on the vendor's reliability calculations.

**c. Product Physical, Quality, and Environmental Factors.**

(1) Availability. The Assured Services subsystem shall have a hardware/software availability of 0.99999 (non-availability of no more than five minutes per year). This requirement was met via vendor LoC.

(2) Maximum Downtimes. The performance parameters associated with the Assured Service Local Area network (ASLAN), MFSS, and LSC, when combined, shall meet the following maximum downtime requirements:

- IP (10/100 Ethernet) network links – 35 minutes/year
- IP subscriber – 12 minutes/year

This requirement was met via vendor LoC.

(3) Loss of Packets. For Voice over IP (VoIP) devices, the voice quality shall have a Mean Opinion Score of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period. The SUT met all Packet Loss requirements for PEIs.

**d. Voice EI.** The SUT met PEI requirements except for the noted discrepancies in the following sub-paragraphs.

(1) Audio codecs. The LSC shall support the origination and termination of a voice session using the following codecs: ITU-T G.711 (a-law and  $\mu$ -law), ITU-T G.723.1, ITU-T G.729 or ITU-T G.729A, and ITU-T G.722.1. The SUT does not support ITU-T G.722.1. This discrepancy was adjudicated by DISA as having a minor operational impact based on vendor's submitted POA&M to fix by December 2012.

(2) Softphones. The softphone shall be conceptually identical to a traditional IP "hard" telephone and is required to provide voice features and functionality provided by a traditional IP hard telephone. The SUT was not tested against the softphone requirements because softphones are not supported by the SUT; however, this is a conditional requirement.

**e. Video EIs.** The SUT's video PEIs and AEIs must support both voice and video. The vendor did not support AEI video or voice capability. This was adjudicated by DISA to have a minor operational impact since there were no certified AEI video end instruments on the UC APL. The vendor's video PEI did not meet the requirements as listed in the following paragraphs and is not certified. The SUT is not certified for use with video EIs.

(1) Video EI. Video EIs are considered associated with the LSC and must have been designed in conjunction with the LSC design. An IP video instrument shall be designed in accordance with the acquiring activity requirements. The SUT provided Polycom Video EIs. JITC only conducted limited video testing on the SUT with the Polycom video EI's during the original test event. The testing that was conducted with the Polycom was on a software load that will never be a GA load that can be purchased and does not support IPv6. JITC further tested this video capability during the Multivendor video test at Fort Huachuca from 6 through 24 February 2012. The following capabilities were not tested: ASAC, MLPP and IPv6. Further testing of these capabilities is needed in order to verify that the SUT is fully capable of handling video EIs. This discrepancy was adjudicated as having a minor operational impact with a vendor submitted POA&M to fix by 1 February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.

(2) Display Messages, Tones, and Announcements. Tones and announcements, as appropriate for voice and video over IP, and as required, in UCR 2008, Change 2, paragraph 5.3.2.6.2.1. This was not met by the SUT. The SUT offers video; however, it is not covered under this certification. JITC only conducted limited video testing on the SUT with Polycom video EIs. Due to the fact that their video EI is not IPv6 capable and did not offer a GA load it did not meet this requirement. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.

(3) Video Codecs (Including Associated Audio Codecs). The product shall support the origination, maintenance, and termination of a video session using the following codecs: one G.xxx and one H.xxx must be used to create and sustain a video session. The SUT does not support ITU-T G.722.1 voice codec. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by December 2012.

#### **f. LSC Requirements.**

(1) Policy Based Assured Services (PBAS)/ASAC Requirements. The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Change 2, Section 5.2.2, MLPP. The SUT met all PBAS/ASAC requirements with one exception: The SUT does not support separate ASAC counts for video and voice. This discrepancy was adjudicated by DISA as having a minor operational impact based on vendor's submitted POA&M to fix by 1 July 2012.

(2) Calling Number Delivery Requirements. The calling number provided to the called party shall be determined by the dial plan serving the calling instrument in accordance with Telcordia Technologies General Requirements -31-CORE "CLASS<sup>SM</sup>

*Feature: Calling Number Delivery,*” Issue 1, June 2000. The SUT met all calling number delivery requirements.

(3) LSC Signaling Requirements. The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber. The SUT met all LSC Signaling Requirements.

(4) Service Requirements under Total Loss of Wide Area Network (WAN) Transport. In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions:

- Completion of local (intra-enclave) calls.
- Routing of calls to the Public Switched Telephone Network (PSTN) using a local MG (PRI or CAS as required by the local interface).
- User look-up of local directory information.

The SUT met all Service Requirements under Total Loss of WAN Transport.

(5) Local Location Server (LLS) and Directory. The purpose of the LLS is to provide information on call routing and called address translation (where a called address is contained within the called SIP Uniform Resource Identifier in the form of the called number). The SUT met all LLS and directory requirements.

(6) LSC Transport Interface Functions. The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all transport interface function requirements.

(7) LSC to PEI, AEI, and Operator Console Status Verification. Periodically, the LSC shall verify the status of its registered and authenticated IP EIs. The SUT does not meet all status verification requirements for AEIs. The SUT does not periodically check the status of its registered PEIs and is not configurable to set the interval for these checks to be conducted. The SUT is not configurable to set interval times to verify status of its IP EI's with the default set at five minutes. This was adjudicated by DISA as having a minor operational impact with the intent to change as conditional in next UCR update.

(8) Line-Side Custom Features Interference. Vendors may implement unique custom features applicable to the line-side of the LSC. Line-side custom features must not interfere with the Assured Services requirements. However, JITC did not test any of those features; therefore, they are not certified for use.

(9) Loop Avoidance. During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy

switch (e.g., End Office) and an LSC. The SUT met all Loop Avoidance requirements for all certified MGs, PSTN and DSN interfaces.

**g. Commercial Cost Avoidance (CCA) Requirements.**

(1) CCA Interworking Function (IWF) Component. The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the LSC supports for EIs, MGs, and Edge Boundary Controllers (EBC), and to interwork all these various signaling protocols with one another. The SUT met all CCA IWF requirements for T1 PRI (ANSI T1.619a and NI-2, ITU-T Q931 E1 PRI, ITU-T Q955.3 E1 PRI, T1 CAS, and E1 CAS). Other conditional IWFs (E1 PRI, CAS and Common Channel Signaling 7 (CCS7)) were not tested.

(2) CCA MG Controller (MGC) Component. The MGC within the CCA must control all MGs within the LSC or MFSS, control all trunks within each MG, control all signaling and media streams on each trunk within each MG, accept IP-encapsulated signaling streams from an Signaling Gateway (SG) or MG, and to use either ITU-T recommendation H.248 or a supplier-proprietary protocol to accomplish these controls. The SUT met all CCA MGC requirements.

(3) SG Component. The role of the CCA with respect to the SG is to control all SGs within the network appliance, and to control all signaling links (DoD CCS7) within each SG. The SG is conditional for an LSC and was not tested on the SUT.

(4) CCA-IWF Support for AS-SIP. The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements. The SUT met all requirements for CCA-IWF support for AS-SIP for required T1 PRI (ANSI T1.619a and NI-2, ITU-T Q931 E1 PRI, ITU-T Q955.3 E1 PRI, T1 CAS, and E1 CAS) interfaces.

(5) CCA-IWF Support for Signaling System 7 (SS7). CCA-IWF support for SS7 (aka CCS7) is a conditional requirement for LSCs and was not tested on the SUT.

(6) CCA-IWF Support for PRI, via MG. The CCA IWF shall support the United States (US)/National ISDN version of the ISDN PRI protocol. The SUT met all requirements for CCA-IWF support for T1 PRI (ANSI T1.619a and NI-2).

(7) CCA-IWF Support for CAS Trunks via MG. Support for CAS is a conditional requirement for LSCs. CCA-IWF Support for CAS trunk via MG is a conditional requirement for LSCs and was not tested on the SUT.

(8) CCA-IWF Support for PEI and AEI Signaling Protocols. The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols. The SUT met all requirements for CCA-IWF Support for PEI Voice Signaling Protocols only. No AEIs were tested.

(9) CCA-IWF Support for VoIP and Time Division Multiplexing (TDM) Protocol Interworking. The role of the IWF within the CCA is to support all the VoIP and TDM signaling protocols that the appliance supports for PEIs, AEIs, MGs, and EBCs, and interwork all these various signaling protocols with one another. The SUT met all requirements for CCA-IWF support for VoIP and TDM Protocol Interworking required T1 PRI (ANSI T1.619a and NI-2, ITU-T Q931 E1 PRI, ITU-T Q955.3 E1 PRI, T1 CAS, and E1 CAS) interfaces.

(10) CCA Preservation of Call Ringing State during Failure Conditions. The CCA in the LSC, MFSS, and WAN Softswitch (SS) shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA. The SUT met all requirements for CCA Preservation of Call Ringing State during Failure Conditions.

(11) CCA Interactions with Transport Interface Functions. The CCA interacts with Transport Interface functions by using them to communicate with PEIs, AEIs, the EBC, the MGs, and the SG over the ASLAN. The SUT met all requirements for CCA interactions with Transport Interface Functions with PEIs only.

(12) CCA Interactions with the EBC. The CCA interacts with the EBC by directing AS-SIP signaling packets to it (for signaling messages destined for an MFSS) and by accepting AS-SIP signaling packets from it (for signaling messages directed to the LSC from an MFSS). The SUT met all requirements for CCA interactions with the EBC.

(13) CCA Support for Admission Control. The CCA interacts with the ASAC component of the LSC and MFSS to perform specific functions related to ASAC, such as counting internal, outgoing, and incoming calls; managing separate call budgets for VoIP and Video over IP calls; and providing preemption. The SUT met all requirements for CCA support for Admission Control with one exception: The SUT does not support separate ASAC counts for video and voice. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

(14) CCA Support for User Features and Services (UFS). The UFS Server is responsible for providing features and services to VoIP and Video PEIs/AEIs on an LSC or MFSS, where the CCA alone cannot provide the feature or service. The SUT met all requirements for CCA Support for UFS for PEIs.

(15) CCA Support for IA. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated and authorized by the appliance. The IA function ensures that Voice and Video signaling streams that traverse the appliance and its ASLAN are encrypted properly SIP/Transport Layer Security (TLS). The IA requirements are tested separately; see paragraph 11.3.

(16) CCA Interaction with EIs. The LLS provides information on called address translation in response to call routing queries from the CCA. The CCA sends call routing queries to the LLS for both outgoing calls from appliance PEIs or AEIs (i.e., LSC and MFSS) and incoming calls to appliance PEIs or AEIs (i.e., LSC and MFSS). The SUT met all requirements for CCA interaction with PEIs.

(17) CCA Support for AS Voice and Video. The CCA in the MFSS or LSC needs to interact with VoIP PEIs and AEIs served by that MFSS or LSC. The VoIP interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The SUT offers video; however, due to the fact that their video EI is not IPv6 capable and did not offer a GA load, it did not meet this requirement. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 February 2013.

(18) CCA Interactions with Service Control Functions. The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions. The SUT met all requirements for CCA Interactions with Service Control Functions.

(19) CCA Interworking between AS-SIP and CCS7. Interworking is performed at a node with CCA (SIP/CCS7 IWF) functionality that processes/interworks incoming CCS7 messages to outgoing AS-SIP messages, and similarly, incoming AS-SIP messages to outgoing CCS7 messages. This is a conditional requirement for LSCs and was not tested.

#### **h. MG Requirements.**

(1) Role of MG in LSC. The MG supports interconnection of VoIP, Fax over IP (FoIP), and Modem over IP (MoIP) media streams with the LSC media server, which provides tones and announcements for LSC calls and LSC features. To support inter-enclave MoIP and FoIP, the LSC must meet ITU-T V.150.1 requirements. The SUT partially met the requirements for ITU-T V.150.1. For FAX (ITU-T T.38 transitions) the LSC maintains the session protocol (UDP). However, the port is changed when moving from VoIP to FoIP. The LSC sends a REINVITE (per MGC transitioning rules) with the new port number; therefore, SIP compliant devices (including the EBC) should follow the port change. The port selection is currently configurable for MoIP because MoIP does not use the REINVITE method for transition. This discrepancy was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 July 2012.

(2) MG Support for ASAC. The MG assists the CCA in performing ASAC (i.e., call preemption based on per-call precedence levels) for outgoing TDM calls at MGs and for incoming TDM calls at MGs. The SUT met all requirements for MG Support for ASAC with one exception: The SUT does not support separate ASAC

counts for video and voice. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

(3) MG and IA Functions. The Information Assurance function within the appliance ensures that end users, PEIs, AEIs, MGs, SGs, and EBCs that use the appliance are all properly authenticated by the appliance. The IA function also ensures that VoIP signaling streams and media streams that traverse the appliance and its ASLAN are properly encrypted, using SIP/TLS and SRTP, respectively. The IA requirements are tested separately; see IA report, Reference (e).

(4) MG Interaction with Service Control Function. The MG is responsible for routing individual VoIP, FoIP, and MoIP media streams to the media server when instructed to do so by the CCA/MGC. When instructed to do so by the CCA/MGC, the MG is responsible for removing individual VoIP, FoIP, and MoIP media streams from the media server, and for either disconnecting them entirely, or routing them on to other LSC end users (e.g., VoIP or video EIs). The SUT met all requirements for MG Interaction with Service Control Function.

(5) MG Interactions with IP Transport Interface Functions. The Transport Interface functions in the LSC provide interface and connectivity functions with the ASLAN and its IP packet transport network. The SUT met all requirements for MG Interactions with IP Transport Interface Functions.

(6) MG-EBC interactions. The MG interacts with the EBC by sending SRTP media streams to it (for call media destined for a PEI, AEI, or MG that is served by another appliance outside the LSC), or by accepting SRTP media streams from it (for call media arriving from a PEI, AEI, or MG that is served by another appliance outside the LSC). The SUT met all requirements for MG-EBC PEI interactions.

(7) MG IP-Based PSTN Interface Requirements. Voice and Video over IP (VVoIP) interfaces from the UC network to the PSTN have not been defined. Interfaces from an LSC or MFSS to the PSTN will be via an MG with TDM interfaces as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. This requirement is conditional, therefore was not tested.

(8) MG Interaction with VoIP EIs. The MG in the MFSS or LSC needs to interact with VoIP EIs served by that MFSS or LSC, and with VoIP EIs served by other MFSSs or LSCs. The VoIP signaling interface between the PEI and the MFSS or LSC is left up to the network appliance supplier. The VoIP signaling interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all MG Interaction with VoIP PEI requirements.

(9) MG Support for User Features and Services. The MG shall support the operation of features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today. The SUT met all requirements for MG Support for User Features and Services for VoIP PEIs only.

(10) MG Interface to TDM network elements in DoD Networks. Each appliance MG shall support TDM trunk groups that can interconnect with the following devices in DoD networks, in the US and worldwide: Private Branch Exchanges (PBXs), Small End Offices, EOs, and MFSS. The SUT met all requirements for MG Interface to TDM devices in DoD Networks.

(11) MG Interface to TDM Allied and Coalition. The appliance suppliers should support TDM trunk groups on their MG product that can interconnect with devices in US allied and coalition partner networks worldwide. This requirement is conditional and was not tested on the SUT.

(12) MG Interface to TDM PSTN in the United States (US). Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the US. The SUT met all requirements for MG Interface to TDM PSTN in the US using T1 PRI (NI-2) and T1 CAS.

(13) MG Interfaces to TDM PSTN outside the continental US (OCONUS). The appliance supplier (i.e., LSC or MFSS supplier) should support TDM trunk groups on its MG product that can interconnect with devices in foreign country telephone networks (OCONUS) worldwide. The SUT met the interface to TDM PSTN OCONUS with ITU-T Q.931 and E1 CAS interfaces.

(14) MG Support for CCS7. The MG shall support TDM trunk groups that are controlled by a separate CCA-to-SG signaling link that carries DoD CCS7 protocol. The MG shall support these TDM trunk groups, and the SG shall support DoD CCS7 signaling. This conditional requirement was not tested on the SUT.

(15) MG Support for ISDN PRI Trunks. The MG shall support ISDN PRI trunk groups that carry the US/National ISDN version of the ISDN PRI protocol. The SUT met all requirements for MG Support for ISDN T1 PRI (ANSI T1.619a and NI-2) Trunks.

(16) MG Support for CAS Trunks. The MG shall support CAS trunk groups that carry the US version of the CAS protocol. The CAS is a conditional requirement for LSCs. The SUT supports both T1 and E1 CAS interfaces.

(17) MG requirements for VoIP Internal Interfaces. The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN. The SUT met all requirements for VoIP Internal Interfaces for PEIs only.

(18) MG Echo Cancellation (EC). The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms. The MG shall provide echo cancellation for voice, G3 Fax, and variable bit data (VBD) modem fax calls. Each

MG EC shall be equipped with an “echo canceller disabling signal” tone detector. The SUT met all requirements for MG Echo Cancellation.

(19) MG Clock Timing. The MG shall derive its clock timing from a designated T1 or PRI interface. The SUT met all MG Clock Timing requirements.

(20) MGC-MC CCA Functions. The MGC within the CCA must control all MGs within the LSC or MFSS, control all trunks within each MG, control all signaling and media streams on each trunk within each MG, accept IP-encapsulated signaling streams from an SG or MG, and to use either ITU-T H.248 or a supplier-proprietary protocol to accomplish these controls. The SUT met all requirements for MGC-MC CCA functions.

(21) MG ITU-T V.150.1. When the MG uses ITU-T V.150.1 inband signaling to transition between audio, FoIP, modem relay, or VBD states or modes, the MG shall continue to use the established session’s protocol (e.g., decimal 17 for User Datagram Protocol (UDP) and port numbers so that the transition is transparent to the EBC. The SUT partially met the requirements for ITU-T V.150.1. For FAX (T.38 transitions) the LSC maintains the session protocol (UDP). However, the port is changed when moving from VoIP to FoIP. The LSC sends a REINVITE (per MGC transitioning rules) with the new port number; therefore, SIP compliant devices (including the EBC) should follow the port change. The port selection is currently configurable for MoIP because MoIP does not use the REINVITE method for transition. This discrepancy was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 July 2012.

(22) MG Preservation of Call Ringing during Failure. The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG. The SUT met all the requirements for MG Preservation of Call Ringing during Failure.

#### **i. SG Requirements.**

(1) SG and CCS7 network Interactions. The SG shall support signaling connectivity to the DoD CCS7 network based on UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features, specifications for CCS7. This is a conditional requirement for LSCs and was not tested on the SUT.

(2) SG Interactions with CCA. The SG shall support a supplier-specific interface to the CCA for interactions between the SG and CCA. This is a conditional requirement for LSCs and was not tested on the SUT.

(3) SG Interworking Functions. The SG will terminate CCS7 links on its CCS7 side and transport the CCS7 call control and service control protocols (i.e., ISDN

User part (ISUP) (ISUP) and Transaction Capabilities Application Part) to the CCA. Similarly, the SG will receive CCS7 call control and service control messages from the CCA. The SG is responsible for the appropriate formatting of the messages for transmission on the CCS7 links. This is a conditional requirement for LSCs and was not tested on the SUT.

**j. Worldwide Numbering and Dialing Plan (WWNDP) Requirements.**

(1) WWNDP. The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Change 2, paragraph 5.2.3.5.1.2, Interswitch and Intraswitch Dialing. The SUT met all requirements for WWNDP for PEIs.

(2) DSN WWNDP. The LSCs must support DSN WWNDP and must support mapping of DSN telephone numbers to SIP URIs, provides examples of DSN numbers using SIP URIs that use the syntax defined in Request for Comment (RFC) 3966. The SUT met all DSN WWNDP requirements.

**k. CCA.** The LSC must use a CCA functionality to route calls from an IP EI to a DISN E.164 number in a manner which will minimize commercial costs associated with PSTN calls. The SUT does not currently support Lightweight Directory Access Protocol (LDAP) V3 for CCA. This discrepancy was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to be available for testing 1 July 2012.

**l. AS-SIP Based Support for External Devices (Voicemail, Unified Messaging, and Automated Receiving Devices).** The LSC shall support all mandatory requirements in RFC 3842. The LSC shall support all mandatory requirements in IETF Internet Draft draft-levy-sip-diversion-08.txt, Diversion Indication in SIP. This is a conditional requirement. The JITC did not test any AS-SIP external devices.

**m. Precedence Call Diversion.** The AS-SIP signaling appliance shall divert all unanswered VoIP calls above the ROUTINE precedence level to a designated directory number for precedence call diversion. The SUT met all precedence call diversion requirements.

**n. Attendant Station Features.**

(1) Precedence and Preemption. The Real Time Services (RTS) Attendant Console shall interoperate with PBAS/ASAC. The SUT met all the requirements for attendant station precedence and preemption.

(2) Call Display. The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant. The SUT met all the requirements for attendant station call display.

(3) Class of Service (CoS) Override. If the LSC, MFSS, or WAN SS supports assignment of a CoS to an individual EI, then this appliance and the attendant console shall give the attendant the ability to override any incoming call's calling party CoS (based on calling area or precedence) on a call-by-call basis. The SUT met all the requirements for attendant station class of service override.

(4) Busy Override and Busy Verification. The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition. The SUT met all the requirements for attendant station busy override and busy verification.

(5) Night service. The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number. The SUT met all the requirements for attendant station night service.

(6) Automatic Recall of Attendant. When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console. The SUT met all the requirements for attendant station automatic recall of attendant.

(7) Calls in Queue to the Attendant. The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue. The SUT met all the requirements for calls in queue to the attendant

#### **o. AS-SIP Requirements.**

(1) SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs. The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs must comply with the differentiated set of requirements defined for ITU-T H.323 and/or vendor-proprietary EIs if they serve ITU-T H.323 and/or vendor-proprietary EIs. The SUT met all the SIP Requirements for AS-SIP Signaling Appliances and PEIs.

(2) SIP Session Keep-Alive Timer. The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions in accordance with RFC 4028. The SUT met all keep-alive timer requirements.

(3) Session Description Protocol (SDP). A session description consists of a session-level description (details that apply to the whole session and all media streams) and optionally several media-level descriptions (details that apply to a single media stream). The LSC must support SDP in accordance with RFC 2327. The SUT met all SDP requirements.

(4) Precedence and Preemption. The LSC must meet the detailed requirements for the execution of preemption and the handling of precedence information as defined in UCR 2008, Change 2, paragraph 5.3.4.2.10. The SUT met all critical CRs & FRs with one exception: In a 3-way conference the SUT does not classmark the conference at the highest precedence level of each leg of the conference. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

(5) Video Telephony – General Rules. Video calls must meet the detailed requirements for video telephony messaging as defined in UCR 2008, Change 2, paragraph 5.3.4.12. JITC only conducted limited video testing on the SUT with the Polycom video EI's during the original test event. The testing that was conducted with the Polycom was on a software load that will never be a GA load that can be purchased and does not support IPv6. JITC further tested this video capability during the Multivendor video test at Fort Huachuca from 6 through 24 February 2012. The following capabilities were not tested: ASAC, MLPP and IPv6. Further testing of these capabilities is needed in order to verify that the SUT is fully capable of handling video EIs. This discrepancy was adjudicated as having a minor operational impact with a vendor submitted POA&M to fix by 1 February 2013. The vendor has stated the initial firmware will be available in 2012 and the GA load will be available in 2013. The SUT is not certified for use with video EIs.

(6) Calling Services. The LSC must meet AS-SIP call flow requirements for calling services features as defined in UCR 2008, Change 2, paragraph 5.3.4.13. The SUT met all calling services requirements.

(7) SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances. This specification uses SIP translation for converting between ISUP signaling and AS-SIP signaling but does not use SIP encapsulation of ISUP. This requirement applies to translations between AS-SIP and CCS7. The SUT met all the SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances.

(8) Relevant Timers for the Terminating Gateway and the Originating Gateway. This requirement applies to gateways between AS-SIP and CCS7 links. The SUT met all the requirements for Relevant Timers for the Terminating Gateway and the Originating Gateway.

(9) SIP Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances must comply with UCR 2008, Change 2, paragraph 5.3.4.7.1, AS-SIP Signaling Appliances and AS-SIP EIs, as well as the additional general requirements in UCR 2008, Change 2, paragraph 5.3.4.16. The SUT met all requirements for interworking AS-SIP signaling appliances.

(10) Keep-Alive Timer Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances must comply with UCR 2008,

Change 2, paragraph 5.3.4.8, SIP Session Keep-Alive Timer, as well as the additional E1 requirements listed in UCR 2008, Change 2, paragraph 5.3.4.17. The SUT met all keep-alive timer requirements for interworking AS-SIP signaling appliances.

(11) Precedence and Preemption Extensions for Interworking AS-SIP Signaling Appliances. The LSC must meet all requirements for header fields as listed in UCR 2008, Change 2, paragraph 5.3.4.18. The SUT met all requirements for precedence and preemption extensions for interworking AS-SIP signaling appliances.

(12) Supplementary Services. The LSC must meet call flow requirements as described in UCR 2008, Change 2, paragraph 5.3.4.19 for supplementary services. The SUT met all critical CRs & FRs with one exception: In a 3-way conference the SUT does not classmark the conference at the highest precedence level of each leg of the conference. This was adjudicated by DISA as having a minor operational impact with a vendor POA&M to fix by 1 July 2012.

#### **p. IPv6 Requirements.**

(1) Product Requirements. This requirement was test intra-enclave, and the vendor has submitted a LoC for IPv6, stating the SUT is not compliant with the following:

- The SUT does not fully support IPv6 neighbor discovery. The vendor LoC states their products support the earlier version of neighbor discovery in RFC 2461. The vendor is currently working on an update to comply with RFC 4861. This discrepancy was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 July 2012. Furthermore, to mitigate this discrepancy in the interim the TEO PEIs must all be configured IPv6 dual stack preferred IPv4 on IPv6.

- An IP TEO EI setup for dual stack IPv4 preferred calling into dual stack IPv6 preferred EI has no audio. This discrepancy was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 July 2012.

- IPv6 inter-switch requirements have not been tested by the JITC.

#### **q. Network Management (NM).**

(1) LSC Management Function. The LSC Management function supports functions for LSC Fault, Configuration, Accounting, Performance, and Security (FCAPS) management and audit logs. The SUT met the requirements for the LSC Management Function via LoC.

(2) VVoIP NM System (NMS) Interface Requirements. The physical interface between the DISA Voice and VVoIP element management system and the network

components (i.e., LSC, MFSS, and EBC) is 10/100-Megabits per second Ethernet interface. The interface will work in either of the two following modes using auto-negotiation: Institute of Electrical and Electronics Engineers, Inc. (IEEE), Ethernet Standard 802.3, 1993; or IEEE, Fast Ethernet Standard 802.3u, 1995. The SUT partially met VVoIP NMS Interface Requirements. The SUT is certified only with VoIP PEIs. This was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 February 2012.

(3) General Management Requirements. The LSC components shall each have an individual pair of Ethernet interfaces for management purposes, even in cases where the MFSS or LSC component contains multiple physical devices. The SUT met all general management requirements via LoC.

(4) Requirement for FCAPS Management. The LSC must meet all general requirements for the FCAPS management functional areas as defined in UCR 2008, Change 2, paragraph 5.3.2.17. The SUT met the requirements for FCAPS Management with the LoC.

(5) NM requirements of Appliance Functions. The LSC must meet all management requirements for ASAC, CCA, SG, and MG functions as defined in UCR 2008, Change 2, paragraph 5.3.2.18. The SUT does not support management requirements for ASAC as defined in the UCR. This discrepancy was adjudicated by DISA as having a minor operational impact based on vendor's submitted POA&M to fix by 1 July 2012.

(6) Accounting Management. Accounting management identifies a set of events during which call detail information is collected. These events are call connect, call attempt, and call disconnect. When these events are detected, specific call data will be provided by the network appliances that were involved in the event. The SUT met all accounting management requirements via LoC with one exception: The SUT does not fully comply with the CDR and QoS. This was adjudicated by DISA as having a minor operational impact with a vendor submitted POA&M to fix by 1 July 2012.

**11.3 Information Assurance.** The IA report is published in a separate report, Reference (e).

**11.4 Other.** None

**12. TEST AND ANALYSIS REPORT.** No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet 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 <http://jit.fhu.disa.mil>

(NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>. All associated data is available on the Defense Information Systems Agency Unified Capability Coordination Office (UCCO) website located at <http://www.disa.mil/ucco/>.

(This page left intentionally blank.)

## SYSTEM FUNCTIONAL AND CAPABILITY REQUIREMENTS

The local session controllers have required and conditional features and capabilities that are established by the Unified Capabilities Requirements (UCR). The System Under Test (SUT) need not provide conditional requirements. If they are provided, they must function according to the specified requirements. The detailed Functional Requirements (FR) and Capability Requirements for Internet Protocol Call Control products (Multi-Function SoftSwitch (MFSS), Local Session Controller (LSC), and Wide Area Network SoftSwitch (WAN SS) are listed in Table 3-1. Detailed Information Assurance (IA) requirements are included in Reference (e) and are not listed below.

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
1	As part of the session setup process, the LSC controls what DSCP to use in the subsequent session media stream packets. (two sub requirements)	5.3.2.2.1.4	Y	Y	NA
2	The SUT must provide the following features: Precedence Call Waiting, Call Forwarding, Call Transfer, Call Hold, Three-Way Calling, Hotline Service, and Calling Party and Called Party ID.	Table 5.3.2.2-1	Y	Y	Y
3	Calls to a DN that does not have any CF feature activated shall be delivered to the DN EI IAW the MLPP procedures specified in UCR 2008, Section 5.2.2 Multilevel Precedence and Preemption	5.3.2.2.2.1.1	Y	Y	Y
4	Call forwarding, when activated on a line DN, shall allow any terminating call at a ROUTINE DSN precedence level, to be completed to the designated destination (IAW the call forward options activated), and shall comply with the requirements as stated in Telcordia Technologies GR-217-CORE, GR-580-CORE, and GR-586-CORE.	5.3.2.2.2.1.1	Y	Y	Y
5	The Tracing of Terminating Calls feature identifies the calling number on intraoffice and interoffice calls terminating to a specified DN. When this feature is activated, the originating DN, the terminating DN, and the time and date are printed out for each call to the specified line.	5.3.2.2.2.2.2	Y	Y	NA
6	The Outgoing Call Tracing feature allows the tracing of nuisance calls to a specified DN suspected of originating from a given local office. The tracing is activated when the specified DN is entered. A printout of the originating DN, and the time and date, are generated for every call to the specified DN.	5.3.2.2.2.2.3	Y	Y	NA
7	The Tracing of a Call in Progress feature identifies the originating DN for a call in progress. Authorized personnel entering a request that includes the specific terminating DN involved in the call activate the feature.	5.3.2.2.2.2.4	Y	Y	NA
8	The Tandem Call Trace feature identifies the incoming trunk of a tandem call to a specified office DN. The feature is activated by entering the specified distant office DN for a tandem call trace. A printout of the incoming trunk number and terminating DN, and the time and date, is generated for every call to the specified DN.	5.3.2.2.2.2.5	Y	N	Y
9	One voice session budget unit shall be equivalent to 110 kilobits per second (kbps) of access circuit bandwidth independent of the PEI or AEI codec used. This includes ITU-T Recommendation G.711 encoding rate plus Internet Protocol Version 6 (IPv6) packet overhead plus ASLAN Ethernet overhead. IPv6 overhead, not IPv4 overhead, is used to determine bandwidth equivalents here.	5.3.2.2.2.3.1	Y	Y	NA
10	If the MFSS's count of an IPC is greater than or equal to the corresponding IPB, and it receives an INVITE request for a precedence session, the MFSS shall preempt a lower priority session (if such a session exists), and then proceed with processing the higher precedence session connect request.	5.3.2.2.2.3.1.2	Y	NA	NA
11	If the MFSS receives a CCA-ID for which there is no entry in ASAC budget table, the SS will reject the session and generate an alarm for the EMS.	5.3.2.2.2.3.1.2	Y	NA	NA
12	If necessary, the MFSS will preempt for a session request that is at precedence level FLASH OVERRIDE or FLASH and the counts equal the budgets.	5.3.2.2.2.3.2	Y	NA	NA
13	Registration and authentication between NEs shall follow the requirements set forth in UCR 2008, Section 5.4, Information Assurance Requirements.	5.3.2.3.1	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
14	The LSCs shall be registered to a primary and backup MFSS. In case of failure of the primary MFSS, the LSC will default to the backup MFSS.	5.3.2.3.2	Y	Y	NA
15	The LSC shall send an OPTIONS request with a Request-URI identifying the primary SS (the Request-URI does not have a userinfo part) on a configurable periodic time interval (default equals 45 seconds; minimum time interval equals 35 seconds). (4 sub requirements)	5.3.2.3.2.2a	NA	Y	NA
16	When a properly functioning primary SS receives the OPTIONS request from a served LSC, the primary SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.2a	NA	Y	NA
17	When the LSC sends a defined configurable number of successive OPTIONS requests (default equals 2) for which there either is no response or the response is a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response, then it must failover to the secondary SS. (3 sub requirements)	5.3.2.3.2.5a.1	NA	Y	NA
18	If the LSC receives a 200 OK response to an OPTIONS request from the primary SS before the configurable number of successive failures to the OPTIONS requests (default equals 2) has been reached, then no action is taken to failover to the secondary SS.	5.3.2.3.2.5b	NA	Y	NA
19	Whenever an originating SS sends an INVITE request to another SS and receives either a 408 (Request Time-Out), 503 (Service Unavailable), or 504 (Server Time-Out) response and the originating SS is not already awaiting a response to a pending OPTIONS request to the other SS, then the originating SS shall send an OPTIONS request with a Request-URI identifying the SS.	5.3.2.3.2.2b	Y	NA	Y
20	When a properly functioning SS receives the OPTIONS request, the SS shall respond with a 200 OK response that includes the Accept header and the Supported header.	5.3.2.3.2.2b	Y	NA	Y
21	Each MFSS (SS) shall be configured with knowledge of each pair of SSs that act as backups for each other. (7 sub requirements)	5.3.2.3.2.5b	Y	NA	Y
22	The Assured Services subsystem shall have a hardware/software availability of 0.99999 (nonavailability of no more than 5 minutes per year).	5.3.2.5.2.1	Y	Y	
23	The performance parameters associated with the ASLAN, MFSS, and LSC, when combined, shall meet the following maximum downtime requirements: • IP (10/100 Ethernet) network links – 35 minutes/year • IP subscriber – 12 minutes/year	5.3.2.5.2.2	Y	Y	NA
24	For these VoIP devices, the voice quality shall have a MOS of 4.0 (R-Factor equals 80) or better, as measured in accordance with the E-Model. Additionally, these devices shall not lose two or more consecutive packets in a minute and shall not lose more than seven voice packets (excluding signaling packets) in a 5-minute period.	5.3.2.5.4	Y	Y	NA
25	An IP voice instrument shall be designed in accordance with the acquiring activity requirements, but the following capabilities are specifically required as indicated: • <b>[Objective]</b> DoD Common Access Card (CAC) reader • <b>[Required]</b> Display calling number • <b>[Required]</b> Display precedence level of the session • <b>[Required]</b> Support for Dynamic Host Configuration Protocol (DHCP).	5.3.2.6.1	Y	Y	NA
26	Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement.	5.3.2.6.1.1	Y	Y	Y
27	The product shall support the origination and termination of a voice session using the following codecs: • ITU-T Recommendation G.711, to include both the $\mu$ -law and A-law algorithms • ITU-T Recommendation G.723.1 • ITU-T Recommendation G.729 or G.729A • ITU-T Recommendation G.722.1	5.3.2.6.1.2	Y	Y	NA
28	Voice over IP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006.	5.3.2.6.1.3	Y	Y	NA
29	For Fixed-to-Fixed calls, the product shall use 20 ms as the default voice sample length, and as the basis for the voice payload packet size.	5.3.2.6.1.4	Y	Y	NA
30	The PEI or AEI shall be capable of authenticating itself to its associated LSC and vice versa.	5.3.2.6.1.5 5.3.2.6.2.3	Y	Y	NA
31	Analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a TA or an Integrated Access Device (IAD) connected to an Ethernet port.	5.3.2.6.1.6	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
32	The LSC shall meet all the requirements for PBAS/ASAC, as appropriate for VoIP and Video over IP services, as specified in UCR 2008, Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.7.2.1	NA	Y	NA
33	The LSC shall support CND, as specified in UCR 2008, Section 5.2.3.2.2.1.8, Calling Number Delivery.	5.3.2.7.2.2	NA	Y	NA
34	The LSC must provide signaling on the line side for local intra-enclave subscriber-to-subscriber calls, and trunk-side signaling for calls between an external enclave and a local subscriber.	5.3.2.7.2.3	NA	Y	NA
35	In the event that a total loss of connectivity to the DISN WAN occurs, the LSC shall provide the following functions: <ul style="list-style-type: none"> <li>• Completion of local (intra-enclave) calls</li> <li>• Routing of calls to the PSTN using a local MG (PRI or CAS as required by the local interface)</li> <li>• User look-up of local directory information</li> </ul>	5.3.2.7.2.4	NA	Y	NA
36	The LSC Management function supports functions for LSC FCAPS management and audit logs. Collectively, these functions are called FCAPS Management and Audit Logs.	5.3.2.7.2.6	NA	Y	NA
37	The LSC Transport Interface functions provide interface and connectivity functions with the ASLAN and its IP packet transport network.	5.3.2.7.2.7	NA	Y	NA
38	The LSC shall provide an interface to the DISA NMS. The interface consists of a 10/100-Mbps Ethernet connection	5.3.2.7.2.8	NA	Y	NA
39	Periodically, the LSC shall verify the status of its registered and authenticated IP EIs, including operator (dial service attendant) consoles. The verification interval shall be configurable with the default set at 5 minutes.	5.3.2.7.2.10	NA	Y	NA
40	Line-side custom features must not interfere with the Assured Services requirements.	5.3.2.7.2.11	NA	Y	NA
41	During the call establishment process, the product shall be capable of preventing or detecting and stopping hair-pin routing loops over ANSI T1.619a and commercial PRI trunk groups (i.e., T1 PRI and E1 PRI) between a legacy switch (e.g., TDM EO) and an LSC	5.3.2.7.3	NA	Y	NA
42	When the AS-SIP TDM Gateway receives a call request over an ISDN MLPP PRI then the AS-SIP TDM Gateway MUST map the telephony numbers received from the Q.931 SETUP message to SIP URIs	5.3.2.7.4.3.3	Y	Y	NA
43	The AS-SIP TDM Gateway MG MUST support the ITU-T Recommendation G.711 ( $\mu$ -law and A-law) audio codec.	5.3.2.7.4.3.4	Y	Y	NA
44	The AS-SIP TDM Gateway MG MUST support RFC 4040 and the AS-SIP TDM Gateway MUST support the signaling for establishing the 64kbps unrestricted bearer per Section 5.3.4.7.7, 64 kbps Transparent Calls (Clear Channel).	5.3.2.7.4.3.4	Y	Y	NA
45	The AS-SIP TDM Gateway MG MUST support T.38 Fax Relay	5.3.2.7.4.3.4	Y	Y	NA
46	The AS-SIP TDM Gateway MG MUST support the SCIP-216 subset of V.150.1 Modem Relay (see Section 5.3.2.21.2, RTS SCIP Gateway Requirements) and the AS-SIP TDM Gateway MUST support the AS-SIP signaling requirements in support of modem relay	5.3.2.7.4.3.4	Y	Y	NA
47	The AS-SIP TDM Gateway MUST satisfy the Information Assurance requirements in Section 5.4 Information Assurance for a media gateway.	5.3.2.7.4.3.5	Y	Y	NA
48	The AS-SIP TDM Gateway MUST provide an interface to the DISA NMS. The interface MUST consist of a 10/100-Mbps Ethernet connection	5.3.2.7.4.3.9	Y	Y	NA
49	The AS-SIP IP Gateway MUST implement call count thresholds for voice sessions and for video sessions in order to perform Session Admission Control (SAC).	5.3.2.7.5.1.1	Y	Y	NA
50	The requirements for the TDM side of the MFSS are entirely the same as for the DSN MFS specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features. The TDM side of the MFSS must meet these requirements.	5.3.2.8.2.1	Y	NA	NA
51	MFSS shall support PRI signaling for TDM communication with other systems.	5.3.2.8.2.3	Y	NA	NA
52	The TDM side of the MFSS shall support CCS7 signaling for communication with other TDM systems.	5.3.2.8.2.3	Y	NA	NA
53	MFSS shall support AS-SIP signaling for IP communication with other MFSSs and LSCs.	5.3.2.8.2.3	Y	NA	NA
54	The MFSS shall provide internal signaling and media conversion for calls between the TDM side and SS side of the MFSS.	5.3.2.8.2.3	Y	NA	NA
55	The CCA/SG/MGC/MG complex in the SS side of the MFSS needs to interface and interact with the EO and Tandem functions in the TDM side of the MFSS.	5.3.2.8.2.4	Y	NA	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
56	The MFSS MG must support internal MG connections that interconnect the SS side of the MFSS with the EO and Tandem functions on the TDM side of the MFSS.	5.3.2.8.2.4	Y	NA	NA
57	The MFSS MG shall interact with the MFSS MGC so that Internal MG connections between the SS and TDM sides of the MFSS support (1) Intra-MFSS calls between TDM EIs connected to the TDM side, and PEIs/AEIs connected to the SS side of the MFSS (2) Incoming and outgoing calls to/from systems external to the MFSS that require conversion between TDM and IP	5.3.2.8.2.4	Y	NA	NA
58	When a U.S. ISDN PRI-based connection is used between the SS and TDM sides of the MFSS, the MFSS MG shall interact with the MFSS MGC so that U.S. ISDN PRI signaling (National ISDN PRI signaling with the Precedence Level IE and related MLPP IEs included) is used between the softswitch and TDM sides, and the T1.619/T1.619a version of the ISDN PRI MLPP feature operates correctly between the SS and TDM sides of the MFSS, for both VoIP-to-TDM calls and TDM-to-VoIP calls over this trunk group.	5.3.2.8.2.4	Y	NA	NA
59	The SS side of the MFSS shall meet all the requirements for MLPP, as appropriate for VoIP and Video over IP services, as specified in Section 5.2.2, Multilevel Precedence and Preemption.	5.3.2.8.2.6	Y	NA	NA
60	The SS side of the MFSS shall support CND as specified in UCR 2008, Section 5.3.2.2.2.1.8, Calling Number Delivery.	5.3.2.8.2.6	Y	NA	NA
61	The requirements for SCS functions (i.e., CCA, IWF, MG, MGC, and SG) and NM are provided in separate sections of this document. The MFSS must meet all these requirements.	5.3.2.8.2.6	Y	NA	NA
62	The CCA IWF must support AS-SIP and ISDN PRI protocols.	5.3.2.9.2.1	Y	Y	NA
63	The MGC within the CCA must control all MGs within the LSC or MFSS, support DoD ISDN trunks, control all signaling and media streams on each trunk within each MG, and accept IP-encapsulated signaling streams from an SG or MG.	5.3.2.9.2.2	Y	Y	NA
64	The CCA shall be responsible for controlling all the SGs within the MFSS and LSC.	5.3.2.9.2.3	Y	C	NA
65	The CCA shall be responsible for controlling each signaling link within each SG within the MFSS or LSC.	5.3.2.9.2.3	Y	C	NA
66	The CCA shall be responsible for controlling the DoD CCS7 signaling stream(s) within each signaling link within each SG.	5.3.2.9.2.3	Y	C	NA
67	Within the network appliance (i.e., MFSS and LSC), the CCA shall use either an IETF-standard set of CCS7-over-IP protocols, or a supplier-proprietary protocol to accomplish the above SG, signaling link, and signaling stream controls.	5.3.2.9.2.3	Y	C	NA
68	The CCA IWF shall support the AS-SIP protocol consistent with the detailed AS-SIP protocol requirements in Section 5.3.4, AS-SIP Requirements.	5.3.2.9.5.1	Y	Y	NA
69	The CCA IWF shall use the AS-SIP protocol on LSC-MFSS and MFSS-MFSS sessions.	5.3.2.9.5.1	Y	Y	NA
70	When the CCA IWF uses the AS-SIP protocol over the Access Segment between the EBC and the DISN WAN, or over the DISN WAN itself, the CCA IWF shall secure the AS-SIP protocol using TLS.	5.3.2.9.5.1	Y	Y	NA
71	The CCA IWF shall support the U.S./National ISDN version of the ISDN PRI protocol.	5.3.2.9.5.3	Y	Y	NA
72	The CCA IWF shall support reception of ISDN PRI messages from the MG and transmission of ISDN PRI messages to the MG.	5.3.2.9.5.3	Y	Y	NA
73	The CCA IWF shall be able to determine the ISDN PRI (and its D-Channel signaling link) that an incoming PRI message was received on, when processing an incoming PRI message from the MG.	5.3.2.9.5.3	Y	Y	NA
74	The CCA IWF shall be able to identify the ISDN PRI (and its D-Channel signaling link) that an outgoing PRI message will be sent on, when generating an outgoing PRI message to the MG.	5.3.2.9.5.3	Y	Y	NA
75	The CCA IWF shall be able to support multiple ISDN PRIs (and their D-Channel signaling links) at the MG, where each PRI is connected to a different PRI end point.	5.3.2.9.5.3	Y	Y	NA
76	The CCA IWF shall be able to differentiate between the individual ISDN PRIs (and their D-Channel signaling links) at the MG.	5.3.2.9.5.3	Y	Y	NA
77	The CCA IWF shall support the full set of ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a.	5.3.2.9.5.3	Y	Y	NA
78	The CCA IWF shall not support any of the ISDN MLPP requirements in ANSI T1.619 and ANSI T1.619a, on ISDN PRIs to TDM PBXs and switches in the U.S. PSTN.	5.3.2.9.5.3	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2008 CH2)</b>	<b>MFSS</b>	<b>LSC</b>	<b>WAN SS</b>
79	On ISDN PRIs from the CCA/MG to TDM PBXs and switches in allied and coalition partners (where those networks support U.S. "National ISDN" PRI), the CCA IWF shall support a DoD-user-configurable per-PRI option that allows the PRI to support or not support the ANSI T1.619/619a PRI MLPP feature on calls to and from that PRI.	5.3.2.9.5.3	Y	Y	NA
80	The CCA IWF shall be able to associate individual PRI configuration data with each individual PRI served by the MG and the CCA. The CCA IWF shall not require groups of PRIs served by the MG and the CCA to share "common" PRI configuration data.	5.3.2.9.5.3	Y	Y	NA
81	The CCA IWF shall support supplier-proprietary Voice and Video EIs and their associated proprietary EI signaling protocols.	5.3.2.9.5.5	Y	Y	NA
82	The CCA in the LSC, MFSS, and WAN SS shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within the CCA.	5.3.2.9.6	Y	Y	Y
83	The MFSS CCA shall be able to support MG connections between the SS side of the MFSS and the EO and Tandem functions on the TDM side of the MFSS.	5.3.2.10.1	Y	NA	NA
84	The CCA shall support assignment of the following items to itself: <ul style="list-style-type: none"> <li>• Only one CCA IP address (this one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address),</li> <li>• A CCA Fully Qualified Domain Name (FQDN) that maps to that IP address, and</li> <li>• A CCA SIP URI that uses that CCA FQDN as its domain name, and maps to the "SIP B2BUA" function within the CCA itself.</li> </ul>	5.3.2.10.3	Y	Y	NA
85	The CCA shall support assignment of the following items to each SIP and AS-SIP PEI and AEI on the Appliance LAN: <ul style="list-style-type: none"> <li>• Only one PEI or AEI IP address,</li> <li>• A PEI or AEI FQDN that maps to that IP address, and</li> <li>• A PEI or AEI SIP URI that uses that PEI or AEI FQDN as its domain name, and maps to the "SIP User Agent" function within the PEI or AEI.</li> </ul>	5.3.2.10.3	Y	Y	NA
86	The CCA shall support assignment of the following items to each MG on the Appliance LAN: <ul style="list-style-type: none"> <li>• Only one MG IP address (this one IP address may be implemented in the MG as either a single logical IP address or a single physical IP address),</li> <li>• An MG FQDN that maps to that IP address, and</li> <li>• An MG SIP URI that uses that MG FQDN as its domain name, and maps to the "UC Signaling and Media End Point" function within the MG.</li> </ul>	5.3.2.10.3	Y	Y	NA
87	The CCA shall support assignment of the following items to each SG on the Appliance LAN: <ul style="list-style-type: none"> <li>• Only one SG IP address (this one IP address may be implemented in the SG as either a single logical IP address or a single physical IP address),</li> <li>• An SG FQDN that maps to that IP address, and</li> <li>• An SG SIP URI that uses that SG FQDN as its domain name, and maps to the "UC Signaling End Point" function within the SG</li> </ul>	5.3.2.10.3	Y	C	NA
88	The CCA shall support assignment of the following items to the EBC: <ul style="list-style-type: none"> <li>• Only one EBC IP address (this one IP address may be implemented in the EBC as either a single logical IP address or a single physical IP address),</li> <li>• An EBC FQDN that maps to that IP address, and</li> <li>• An EBC SIP URI that uses that EBC FQDN as its domain name, and maps to the "SIP B2BUA" function within the EBC.</li> </ul>	5.3.2.10.3	Y	Y	NA
89	When directing VoIP sessions to other network appliances providing voice and video services across the DISN, the CCA shall direct these VoIP sessions to the EBC, so that the EBC can process them before directing them to the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	NA
90	When accepting VoIP sessions from other network appliances on the DISN, the CCA shall accept these VoIP sessions from the EBC, because the EBC relays them from the network appliances on the DISN WAN.	5.3.2.10.4	Y	Y	NA
91	The LSC and MFSS CCA shall meet all the requirements in Section 5.3.2.2.2.3, ASAC – Open Loop. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.10, Precedence and Preemption. The LSC and MFSS CCA shall meet all the requirements in Section 5.3.4.11, Policing of Call Count Thresholds.	5.3.2.10.5	Y	Y	NA
92	The CCA shall generate a redirecting number each time it forwards a VoIP or Video session request as part of a Call Forwarding feature.	5.3.2.10.6	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
93	It is expected that all Assured Services products, such as LSCs and MFSSs, will support vendor-proprietary VVoIP features and capabilities, in addition to supporting the required VVoIP features and capabilities that are listed.	5.3.2.2.2.1	Y	Y	Y
94	The CCA shall relay received SIP and TLS authentication credentials and encryption key information from sending end systems (i.e., users, PEIs, AEs, and EBCs) to the Information Assurance function to support the Information Assurance function's user, PEI, AEI, and EBC authentication capabilities, and its PEI, AEI, and EBC signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	NA
95	The CCA MGC shall relay received H.248 and IPSec (or proprietary-protocol-equivalent) authentication credentials and encryption key information from sending end systems (i.e., MGs and SGs) to the Information Assurance function to support the Information Assurance function's MG and SG authentication capabilities, and its MG and SG signaling stream encryption capabilities.	5.3.2.10.7	Y	Y	NA
96	The CCA shall relay authentication credentials received in a SIP or AS-SIP REGISTER message from an PEI, AEI, or EBC to the Information Assurance function.	5.3.2.10.7	Y	Y	NA
97	The CCA shall relay TLS encryption key information received from a PEI or AEI to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for Voice or Video sessions to/from that PEI or AEI.	5.3.2.10.7	Y	Y	NA
98	The CCA shall relay TLS encryption key information received from an EBC to the Information Assurance function so the Information Assurance function can verify that this encryption key information can be used on the signaling streams for the Voice or Video sessions to/from that EBC.	5.3.2.10.7	Y	Y	NA
99	The CCA within the appliance shall support all Information Assurance Appliance requirements in Section 5.4, Information Assurance Requirements, which involve the appliance's SCS functions and the appliance's MGC.	5.3.2.10.7	Y	Y	NA
100	The CCA shall support supplier-proprietary Voice and Video EIs, using EI-CCA protocols that are proprietary to the LSC or MFSS supplier.	5.3.2.10.10	Y	Y	NA
101	When the CCA IWF supports AS-SIP Voice and Video AEIs, the IWF shall support these AEIs using the set of AS-SIP protocol requirements in Section 5.3.2.22, Generic AS-SIP End Instrument and Video Codec Requirements, and Section 5.3.4, AS-SIP Requirements.	5.3.2.10.10	Y	Y	NA
102	The Appliance CCA (i.e., LSC or MFSS) shall support both assured Voice and Video services. The CCA shall support both assured Voice and assured Video sessions, and shall support these sessions from both VoIP EIs and Video EIs, as described in UCR 2008, Section 5.3.2.10.10, CCA Interactions with End Instrument(s).	5.3.2.10.11	Y	Y	NA
103	The Appliance CCA shall support common procedures and protocol for VoIP and Video session control.	5.3.2.10.11	Y	Y	NA
104	The Appliance CCA shall support common procedures and protocol for feature control, for the features and capabilities given in Table 5.3.2.2-1, Assured Services Product Features and Capabilities.	5.3.2.10.11	Y	Y	NA
105	On calls to and from Proprietary VoIP and Proprietary Video EIs, the CCA shall use the appropriate parameters within the appliance supplier's Proprietary protocol messages to differentiate Proprietary VoIP sessions from Proprietary Video sessions.	5.3.2.10.11	Y	Y	NA
106	When AS-SIP EIs are supported on calls to and from AS-SIP EIs, the CCA shall use the SDP message bodies in AS-SIP INVITE, UPDATE, REFER, and ACK messages, as well as the SDP message bodies in AS-SIP 200 OK responses and earlier 1xx provisional responses, to differentiate AS-SIP Voice sessions from AS-SIP Video sessions.	5.3.2.10.11	Y	Y	NA
107	The CCA shall track VoIP sessions against corresponding Appliance VoIP budgets, and shall separately track Video sessions against corresponding Video budgets. The CCA shall maintain the Appliance's VoIP budgets separate from the Appliance's Video budget.	5.3.2.10.11	Y	Y	NA
108	As part of LSC-Level ASAC and WAN-Level ASAC Policing, the CCA shall support PBAS/ASAC for both VoIP sessions and Video sessions.	5.3.2.10.11	Y	Y	NA
109	The CCA shall allow an individual EI to support both VoIP and Video sessions. The CCA shall allow an individual EI to have both VoIP and Video sessions active at the same time.	5.3.2.10.11	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
110	The CCA shall support the routing of both VoIP and Video session requests from LSCs to MFSSs, from MFSSs to LSCs, and from MFSSs to MFSSs, using AS-SIP. The CCA shall direct outgoing VoIP and Video session requests to EBCs, and shall accept incoming VoIP and Video session requests from EBCs, consistent with this LSC-to-MFSS routing, MFSS-to-LSC routing, and MFSS-to-MFSS routing.	5.3.2.10.11	Y	Y	NA
111	The CCA shall support the ability to remove VoIP and Video sessions and session requests from the media server so the CCA can continue with necessary session processing once the media server has completed its functions.	5.3.2.10.12	Y	Y	NA
112	The MG supports interconnection of VoIP, FoIP, and MoIP media streams with the following LSC functions and end-user devices: a. The LSC media server, which provides tones and announcements for LSC calls and LSC features b. AS-SIP VoIP, FoIP, and MoIP AElS on the LSC	5.3.2.12.3.1	NA	Y	NA
113	The MFSS MG shall be able to support MG trunk groups (referred to as internal MG connections) that either interconnect the SS (VoIP) side of the MFSS with the EO or Tandem functions on the TDM side of the MFSS.	5.3.2.12.3.2.1	Y		NA
114	On incoming call requests to a TDM trunk group, where the CCA/MGC applies a CAC Call Denial treatment to that call request, the MG shall connect the TDM called party on the incoming call request to the appropriate CAC Call Denial tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	NA
115	On incoming calls or call requests to a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM calling party on the incoming call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	NA
116	On outgoing calls or call requests from a TDM trunk group, where the CCA/MGC applies an ASAC Call Preemption treatment to that call or call request, the MG shall connect the TDM called party on the outgoing call or call request to the appropriate ASAC Call Preemption tone or announcement when instructed to do so by the MGC.	5.3.2.12.4.1.1	Y	Y	NA
117	Each MG within an appliance shall support all the appliance requirements in Section 5.4, Information Assurance Requirements, that involve an Appliance MG.	5.3.2.12.4.2	Y	Y	NA
118	When instructed to do so by the MGC, the MG shall direct TDM calls and call requests to the media server.	5.3.2.12.4.3	Y	Y	NA
119	Since each Appliance MG is an IP endpoint on the Appliance LAN, each MG shall support assignment of the following items to itself: • Only one MG IP address (This one IP address may be implemented in the CCA as either a single logical IP address or a single physical IP address.) • An MG FQDN that maps to that IP address • An MG SIP URI that uses that MG FQDN as its domain name, and maps to a "SIP User Agent" function within the MG.	5.3.2.12.4.4	Y	Y	NA
120	The MG shall interact with the Transport Interface functions in the appliances when the MG uses the native LAN protocols, IP, and UDP to exchange SRTP media streams with PEIs, AEIs, other MGs, and the EBC over the Appliance LAN	5.3.2.12.4.4	Y	Y	NA
121	When sending VoIP media streams to PEIs or AEIs and MGs served by other network appliances, the MG shall direct these VoIP media streams to the EBC so the EBC can process them before sending them on to the remote PEIs or AEIs and MGs via the DISN WAN.	5.3.2.12.4.5	Y	Y	NA
122	When accepting VoIP media streams from PEIs or AEIs and MGs served by other network appliances, the MG shall accept these VoIP media streams from the appliance EBC, because the EBC relays them from the DISN WAN and the remote PEIs or AEIs and MGs on the DISN WAN. The MG shall recognize and act on the network-level IP addresses of the remote PEIs or AEIs and MGs, when accepting the VoIP sessions through the EBC from the DISN WAN and the remote PEIs or AEIs and MGs.	5.3.2.12.4.5	Y	Y	NA
123	The MG shall support the exchange of VoIP media streams with the following voice PEIs and AEIs both on the local appliance and on remote network appliances: a. Supplier-proprietary voice PEIs b. Voice SIP EIs, when the appliance supplier supports these EIs c. Voice H.323 EIs, when the appliance supplier supports these EIs d. Voice AS-SIP AEIs	5.3.2.12.4.8	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
124	<p>The MG shall support the operation of the following features for VoIP and Video end users, consistent with the operation of this feature on analog and ISDN lines in DoD TDM switches today:</p> <ul style="list-style-type: none"> <li>• Call Hold</li> <li>• Music on Hold</li> <li>• Call Waiting</li> <li>• Precedence Call Waiting</li> <li>• Call Forwarding Variable</li> <li>• Call Forwarding Busy Line</li> <li>• Call Forwarding No Answer</li> <li>• Call Transfer</li> <li>• Three-Way Calling</li> <li>• Hotline Service</li> <li>• Calling Party and Called Party ID (number only)</li> <li>• Call Pickup</li> </ul>	5.3.2.12.4.9	Y	Y	NA
125	<p>Each appliance MG shall support TDM trunk groups that can interconnect with the following NEs in DoD networks, in the United States and worldwide:</p> <ul style="list-style-type: none"> <li>• PBXs</li> <li>• SMEOs</li> <li>• EOs</li> <li>• MFSSs</li> </ul> <p>Media Gateway support for these TDM trunk groups shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem switches, and MFSSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.</p>	5.3.2.12.5	Y	Y	NA
126	<p>Each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States, including CONUS, Alaska, Hawaii, and U.S. Caribbean and Pacific Territories. Media Gateway support for these TDM trunk groups to the U.S. PSTN shall be identical to the support for these trunk groups in DoD TDM PBXs, EOs, Tandem Switches, and MFSSs today, as specified in UCR 2008, Section 5.2, Circuit-Switched Capabilities and Features.</p>	5.3.2.12.7	Y	Y	NA
127	<p>The MG shall support foreign country ISDN PRI, where the MG handles both the media channels and the signaling channel:</p> <ol style="list-style-type: none"> <li>1. For interconnection with a foreign country PSTN using foreign country ISDN PRI, from the country where the DoD user's B/P/C/S is located.</li> <li>2. Support for ETSI PRI is required on LSC trunk groups when the LSC is used in OCONUS ETSI-compliant countries.</li> <li>3. Support for ETSI PRI is required on MFSS trunk groups when the MFSS is used in OCONUS ETSI-compliant countries.</li> <li>4. Support for MLPP using ISDN PRI is not required on the above trunk groups.</li> </ol>	5.3.2.12.8	Y	Y	NA
128	<p>The MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The MG shall support these U.S. PRI trunk groups conformant with the detailed U.S. ISDN PRI requirements.</p>	5.3.2.12.10	Y	Y	NA
129	<p>The MG shall support multiple U.S. PRI trunk groups based on the needs of the DoD user deploying the appliance. The MG shall allow each U.S. PRI trunk group at the MG to connect to: TDM EO and tandem components of the local MFSS; a different U.S. PSTN TDM NE (e.g., PBX, TDM switch); a different DoD TDM NE (e.g., PBX, TDM switch); or a different DoD IP NE (e.g., LSC, MFSS), based on the interconnection needs of the DoD user.</p>	5.3.2.12.10	Y	Y	NA
130	<p>The MG shall support reception of ISDN PRI messages from the CCA MGC and transmission of ISDN PRI messages to the CCA MGC.</p>	5.3.2.12.10	Y	Y	NA
131	<p>The MG shall connect to the ASLAN of the appliance using the physical layer and data link layer protocols of the ASLAN. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as a physical layer and data link layer endpoint on a LAN switch in the ASLAN.</p>	5.3.2.12.12.1	Y	Y	NA
132	<p>The MG shall connect to the ASLAN of the appliance using the IP as a Network Layer Protocol. In this case, the MG shall appear to the MGC, EBC, and appliance PEIs/AEIs as an IP endpoint on an IP router on the ASLAN.</p>	5.3.2.12.12.2	Y	Y	NA
133	<p>The MG shall support IPv4 as a Network Layer Protocol. The MG shall also support IPv6 as a Network Layer Protocol.</p>	5.3.2.12.12.2	Y	Y	NA
134	<p>Conformant with Section 5.3.5, IPv6 Requirements, the MG shall support dual IPv4 and IPv6 stacks (i.e., support both IPv4 and IPv6 in the same IP end point) as described in RFC 4213.</p>	5.3.2.12.12.2	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
135	The MG shall support exchange of VoIP media streams with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the appliance EBC, with other PEIs/AEIs and MGs on other network appliances) using the following IETF-defined Media Transfer Protocols: • SRTP, conformant with RFC 3711 • SRTCP, conformant with RFC 3711	5.3.2.12.12.4	Y	Y	NA
136	The MG shall secure all VoIP media streams exchanged with appliance PEIs/AEIs, other appliance MGs, and the appliance EBC (and through the EBC, with PEIs/AEIs and MGs on other network appliances) using SRTP and SRTCP.	5.3.2.12.12.4	Y	Y	NA
137	The MG shall use UDP as the underlying Transport Layer Protocol, and IP as the underlying Network Layer Protocol, when SRTP is used for media stream exchange.	5.3.2.12.12.4	Y	Y	NA
138	When the VoIP signaling streams contain supplier-proprietary protocol messages instead of H.248 or ISDN PRI messages, the MG shall secure the proprietary protocol message exchange with the MGC using mechanisms that are as strong as, or stronger than, the use of IPSec to secure H.248 and PRI message exchange.	5.3.2.12.12.5	Y	Y	NA
139	The MG shall support TDM voice streams using the following: • ITU-T 64 kbps G.711 $\mu$ -law PCM over digital trunks • ITU-T 64 kbps G.711 A-law PCM over digital trunks • North American 56 kbps G.711 $\mu$ -law PCM over digital trunks • North American analog voice transmission over analog trunks on TDM trunk groups on the TDM side of the MG	5.3.2.12.12.8	Y	Y	NA
140	The MG shall convert between North American 56 kbps G.711 $\mu$ -law PCM and ITU-T 64 kbps G.711 $\mu$ -law PCM in cases where North American 56 kbps TDM voice trunks are used on the TDM side of the MG.	5.3.2.12.12.8	Y	Y	NA
141	The MG shall convert between North American analog voice transmission and ITU-T 64 kbps G.711 $\mu$ -law PCM in cases where North American analog voice trunks are used on the TDM side of the MG.	5.3.2.12.12.8	Y	Y	NA
142	The MG shall support uncompressed, packetized VoIP streams using ITU-T Recommendation G.711 $\mu$ -law PCM and ITU-T Recommendation G.711 A-law PCM (ITU-T Recommendation G.711, November 1998, plus Appendix I, September 1999, and Appendix II, September 2000) over the IP network on the VoIP side of the MG.	5.3.2.12.12.8.1	Y	Y	NA
143	The MG shall packetize/depacketize G.711 media streams received or sent between its TDM side and its VoIP side.	5.3.2.12.12.8.1	Y	Y	NA
144	The MG shall transport each packetized G.711 VoIP stream to and from the destination local PEI, local AEI, local MG, remote PEI (via an EBC), remote AEI (via an EBC), or remote MG (via an EBC) using SRTP, UDP, and IP protocol layers on the VoIP side of the MG.	5.3.2.12.12.8.1	Y	Y	NA
145	The MG shall support the use of uncompressed, packetized G.711 $\mu$ -law and A-law VoIP media streams for both Fixed and Deployable applications.	5.3.2.12.12.8.1	Y	Y	NA
146	The MG shall provide an EC capability with an echo path capacity (echo tail length) of at least 64 ms.	5.3.2.12.13.2.2	Y	Y	NA
147	The MG shall provide echo cancellation for voice, G3 Fax, and VBD modem fax calls.	5.3.2.12.13.2.2	Y	Y	NA
148	Each MG EC shall be equipped with an "echo canceller disabling signal" tone detector. This tone detector shall detect and respond to an in-band EC disabling signal from an end user's G3 Fax or VBD modem device. The EC disabling signal detected shall consist of a 2100-Hz tone with periodic phase reversals inserted in that tone.	5.3.2.12.13.2.2	Y	Y	NA
149	The MG tone detector/EC disabler shall detect the "echo canceller disabling signal" and disable the MG EC when, and only when, that signal is present for G3 Fax or VBD modem.	5.3.2.12.13.2.2	Y	Y	NA
150	The MG shall derive its clock timing from a designated T1 or PRI interface.	5.3.2.12.14	Y	Y	NA
151	The MGC within the CCA shall be responsible for controlling all the MGs within the LSC or MFSS.	5.3.2.12.15	Y	Y	Y
152	The MGC within the CCA shall be responsible for controlling all the trunks (i.e., DoD CCS7, PRI, or CAS) within each MG within the LSC or MFSS.	5.3.2.12.15	Y	Y	Y
153	The MGC within the CCA shall be responsible for controlling all media streams on each trunk within each MG.	5.3.2.12.15	Y	Y	NA
154	The MGC within the CCA shall accept IP signaling streams from an MG, conveying received PRI or CAS trunk signaling. The MGC shall return IP signaling streams to the MG accordingly, for conversion to transmitted PRI or CAS trunk signaling.	5.3.2.12.15	Y	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
155	Within the appliance (i.e., LSC or MFSS), the MGC shall use either ITU-T Recommendation H.248 (Gateway Control Protocol Version 3) or a supplier-proprietary protocol to accomplish the MG, trunk, and media stream controls described previously.	5.3.2.12.15	Y	Y	Y
156	Whenever the MG uses ITU-T Recommendation V.150.1, the following applies: ITU-T Recommendation V.150.1 provides for three states: audio, VBD, and modem relay. After call setup, inband signaling may be used to transition from one state to another. In addition, V.150.1 provides for the transition to FoIP using Fax Relay per ITU-T Recommendation T.38.	5.3.2.12.16	Y	Y	NA
157	The LSC MG, MFSS MG, and WAN SS MG shall not allow AS-SIP sessions that have reached the ringing state (i.e., an AS-SIP 180 (Ringing) message or 183 (Session Progress) has been sent from the called party to the calling party, and the calling party is receiving an audible ringing tone) to fail when an internal failure occurs within that MG.	5.3.2.12.17	Y	Y	Y
158	The precedence level and dialed number input to the PEI or AEI shall be as specified in UCR 2008, Section 5.2.3.5.1.2, Interswitch and Intraswitch Dialing.	5.3.2.16	Y	Y	NA
159	The DSN Worldwide Numbering and Dialing Plan will be used as the addressing schema within the current DSN and its migration into the SIP environment.	5.3.2.16.1	Y	Y	Y
160	The CCA shall allow session requests from LSC, MFSS Els, other appliances, and MFSS MGs to contain • Called addresses including DSN numbers from the DSN numbering plan • Called addresses including E.164 numbers from the E.164 numbering plan	5.3.2.16.1	Y	Y	Y
161	When a session request's called address includes a DSN number from the DSN numbering plan, the CCA shall determine whether the called DSN number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	Y
162	When a session request's called address includes an E.164 number from the E.164 numbering plan, the CCA shall determine whether the called E.164 number is local to the LSC or MFSS, or external to the LSC or MFSS.	5.3.2.16.1	Y	Y	Y
163	The access code shall include the access digit, followed by the precedence digit or the service digit.	5.3.2.16.1	Y	Y	Y
164	The access digit (e.g., 9) shall provide the indication to the LSC/MFSS that the following digits will indicate either UC call precedence, selected egress to the services of other systems or networks, or selected access to special UC features, such as individual trunk tests.	5.3.2.16.1	Y	Y	Y
165	The precedence digit (0, 1, 2, 3, or 4) shall permit a UC user to dial an authorized UC precedence level from properly classmarked 12-button telephone instruments. When the 7-digit intralSC dialing option is used, it is not necessary to dial or key the precedence access digit for ROUTINE precedence calls. The assignment of precedence digits is shown in Table 5.3.2.16-4, Precedence and Service Access.	5.3.2.16.1	Y	Y	Y
166	The service digits, 5 through 9, shall provide information to the LSC/MFSS to connect calls to Government or public telephone services or networks that are not part of the UC. The UC LSC/MFSS will collect the access code and all routing and address digits before attempting to route a call to prevent numbering ambiguities between the access codes and the 2-digit abbreviated dial codes. The assignment of service access codes is shown in Table 5.3.2.16-4, Precedence and Service Access.	5.3.2.16.1	Y	Y	Y
167	The CCA shall allow each VoIP and Video PEI and AEI served by an LSC or MFSS to have both a DSN number assigned and an E.164 number assigned.	5.3.2.16.1.1	Y	Y	Y
168	For VoIP and Video PEIs or AEIs that have both a DSN number and an E.164 number assigned, the CCA shall be able to match each PEI's or AEI's DSN number with its E.164 number, and to match each PEI's or AEI's E.164 number with its DSN number.	5.3.2.16.1.1	Y	Y	Y
169	The CCA shall be able to distinguish DSN called numbers from E.164 called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	Y
170	The CCA shall be able to distinguish local [DSN or E.164] called numbers from external [DSN or E.164] called numbers when processing VoIP and Video session requests from PEIs, AEIs, EBCs, MG line cards, and MG trunk groups.	5.3.2.16.1.2	Y	Y	Y
171	The MFSS or LSC is only required to support one network FQDN for use with SIP URI domain names: "uc.mil" if that appliance is used for SBU traffic, and "cuc.mil" if that appliance is used for classified traffic.	5.3.2.16.1.4.1	Y	Y	Y
172	The MFSS or LSC is required to ensure that all AS-SIP session requests entering or leaving that appliance use the network FQDN of that appliance (i.e., "uc.mil" for SBU traffic, or "cuc.mil" for Classified traffic) as the domain name in called SIP URIs.	5.3.2.16.1.4.1	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
173	All voice systems, TDM or IP technology-based, must contain subscriber assignment information.	5.3.2.16.1.5	Y	Y	Y
174	The LSC and the LRDB shall support the Commercial Cost Avoidance feature per the requirements in this section.	5.3.2.28.3	NA	Y	NA
175	The LSC shall support an interface to a LRDB to support DB queries and DB responses for the Commercial Cost Avoidance feature.	5.3.2.28.3	NA	Y	NA
176	The LRDB shall support an interface to the LSC to support DB queries and DB responses for the Commercial Cost Avoidance feature.	5.3.2.28.3	NA	Y	NA
177	The query-response interface between the LSC and the LRDB shall be LDAPv3 over TLS over IP. On the LSC, this LDAPv3 interface shall be compliant with IETF RFC 4511 and RFC 4510. On the LRDB, see the LDAPv3 interface requirement in Section 5.3.2.28.5.2.1, General Architecture, Protocols and Interfaces.	5.3.2.28.3	NA	Y	NA
178	The encoding of the LDAPv3 messages and data schema used on the DB query interface between the LSC and the LRDB shall follow the BER of ASN.1. On the LSC this encoding shall be consistent with Section 5.1, Protocol Encoding, of RFC 4511. On the LRDB, see the LDAPv3 interface requirement in Section 5.3.2.28.5.2.1, General Architecture, Protocols and Interfaces .	5.3.2.28.3	NA	Y	NA
179	The interface between the LSC and the LRDB shall be secured using TLS, consistent with the requirements for securing AS-SIP messages using TLS in Section 5.4, Information Assurance Requirements. This security shall provide mutual authentication between the LSC and the LRDB, message confidentiality for the DB query and DB response, and message integrity for the DB query and DB response.	5.3.2.28.3	NA	Y	NA
180	The interface between the LSC and the LRDB shall traverse the data firewalls (and not the EBC firewalls) at both the LSC and LRDB sites.	5.3.2.28.3	NA	Y	NA
181	The interface between the LSC and the LRDB shall traverse the CE Routers at both the LSC and LRDB sites, using the DSCP for User Signaling traffic, and the associated CE Router queues.	5.3.2.28.3	NA	Y	NA
182	The interface between the LSC and the LRDB shall terminate on the Ethernet interface used for VVoIP signaling traffic at the LSC, as described in Section 5.4, Information Assurance Requirements.	5.3.2.28.3	NA	Y	NA
183	The AS-SIP signaling appliance shall divert ALL unanswered UC VoIP calls above the ROUTINE precedence level to a designated UC DN for PCD.	5.3.2.25	Y	Y	C
184	Unanswered UC VoIP calls above the ROUTINE precedence level shall not be forwarded to voicemail, and shall not be forwarded to ACD systems. Instead, they should divert to the PCD DN when the PCD time period expires.	5.3.2.25	Y	Y	C
185	Unanswered UC VoIP calls at the ROUTINE precedence level shall still be forwarded to voicemail or to ACD systems (when CFDA is assigned to the called UC DN), even though PCD is enabled and configured for the ASSIP signaling appliance.	5.3.2.25	Y	Y	C
186	Calls above the ROUTINE precedence level that are destined to (directly dialed to) DN's assigned to voicemail or ACD systems shall only divert to the PCD DN as specified above (i.e., when they are unanswered at the voicemail or ACD system, and the PCD time period expires).	5.3.2.25	Y	Y	C
187	ROUTINE precedence level calls that are destined to (directly dialed to) DN's assigned to voicemail or ACD systems shall be allowed.	5.3.2.25	Y	Y	C
188	Incoming precedence calls to the attendant's listed DN, and incoming calls that are diverted to this attendant DN, shall be placed in a queue for the attendant console (or group of attendant consoles).	5.3.2.25	Y	Y	C
189	When a group of attendant consoles on the same LSC is used, and calls are either placed or diverted to the attendant console DN, call distribution across the Console Group shall be used to reduce excessive caller waiting times.	5.3.2.25	Y	Y	C
190	Incoming calls (placed and diverted) to the console DN shall be queued for attendant service by call precedence and time of arrival. The highest precedence call with the longest holding time in the queue shall be offered to an attendant first.	5.3.2.25	Y	Y	C
191	A recorded message of explanation (e.g., ATQA) shall be applied automatically to all the waiting calls in the Attendant Console queue (refer to Table 5.3.4-9, Announcements).	5.3.2.25	Y	Y	C
192	The RTS Attendant Console shall interoperate with PBAS/ASAC as described in <ul style="list-style-type: none"> <li>• Section 5.3.2.7.2.1, PBAS/ASAC Requirements</li> <li>• Section 5.3.2.2.2.3, ASAC – Open Loop</li> <li>• Section 5.3.4.10, Precedence and Preemption</li> </ul> The console shall be able to initiate all levels of RTS precedence calls (i.e., ROUTINE through FLASH-OVERRIDE).	5.3.2.26.1	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2008 CH2)</b>	<b>MFSS</b>	<b>LSC</b>	<b>WAN SS</b>
193	The RTS Attendant Console shall provide a visual display of each precedence level and the calling number, for incoming direct dialed calls to the attendant, and diverted calls to the attendant (e.g., calls that reach the attendant through PCD).	5.3.2.26.2	Y	Y	Y
194	The appliance and the attendant console shall give the attendant the ability to verify and override a busy line condition.	5.3.2.26.4	Y	Y	Y
195	If the attendant uses BLV on a called line, and that called line (called EI) is busy, the appliance and the attendant console shall give an audible and visual "called line busy" indication back to the attendant.	5.3.2.26.4	Y	Y	Y
196	The appliance and the attendant console shall prevent an attendant from activating BLV or Emergency Interrupt to called lines and called numbers that are located in the commercial network (the PSTN).	5.3.2.26.4	Y	Y	Y
197	The appliance and the attendant console shall give the attendant the ability to use Emergency Interrupt to interrupt an existing call on a busy line, and inform the busy user of a new incoming call.	5.3.2.26.4	Y	Y	Y
198	The appliance shall give selected destination EIs the ability to be exempt from Emergency Interrupt and attendant break-in.	5.3.2.26.4	Y	Y	Y
199	The appliance and the attendant console shall have the ability to route all calls that are normally directed to the console to a separate night service deflection number. The night service deflection number shall be a fixed (preconfigured) or manually-selected DN.	5.3.2.26.5	Y	Y	Y
200	When an attendant redirects an incoming call to a destination station, and that station is either busy or does not answer the call within a preset time, the appliance and the attendant console shall ensure that calling party on the redirected call is recalled automatically to the console.	5.3.2.26.6	Y	Y	Y
201	The appliance and the attendant console shall have the ability to place calls (both directed to the attendant and diverted to the attendant) into a waiting queue.	5.3.2.26.7	Y	Y	Y
202	The appliance and the attendant console shall ensure that calls in the attendant queue are not lost when a console is placed out of service or has its calls forwarded to a night service deflection number.	5.3.2.26.7	Y	Y	Y
203	The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for SIP EIs if they serve SIP EIs, and LSCs MUST comply with the differentiated set of requirements defined for H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs.	5.3.4.7.1	NA	Y	NA
204	All AS-SIP signaling appliances MUST comply with the SIP syntax and encoding rules set forth in RFC 3261. [(see RFC 3261, Section 25, Augmented BNF for the SIP Protocol) and with the corrections in RFC 5954 Essential Correction for IPv6 ABNF and URI Comparison in RFC 3261.]	5.3.4.7.1.1	Y	Y	Y
205	When an AS-SIP signaling appliance does not understand a header field in a request (and support for the header field is not a mandatory requirement under this specification), the AS-SIP signaling appliance MUST ignore that header field and continue processing the message. The AS-SIP signaling appliances MUST ignore any malformed header fields that are not necessary for processing requests.	5.3.4.7.1.3	Y	Y	Y
206	When an AS-SIP signaling appliance that is implemented as a SIP proxy receives a SIP Request message, 2xx response, or 18x response, then the AS-SIP signaling appliance MUST add a Record-Route header whereby the userinfo part of the SIP URI is a unique identifier for the AS-SIP signaling appliance and an IP address is used for the host name.	5.3.4.7.1.3c	Y	Y	Y
207	All AS-SIP signaling appliances MUST be call stateful.	5.3.4.7.1.4	Y	Y	Y
208	Upon receipt of a new request, AS-SIP signaling appliances MUST perform request validation, route information preprocessing, determine request targets, perform request forwarding, perform response processing, process timer C, handle transport error, handle CANCEL processing, and perform proxy route processing according to RFC 3261	5.3.4.7.1.5	Y	Y	Y
209	All AS-SIP signaling appliances MUST support generation of the long form of the SIP header fields along with the receipt and processing of the long form of the SIP header fields.	5.3.4.7.1.7	Y	Y	Y
210	All AS-SIP signaling appliances MUST support receiving and processing the compact form of the SIP header fields.	5.3.4.7.1.8	Y	Y	Y
211	All AS-SIP signaling appliances serving IP EIs MUST support the offer/answer model for the Session Description Protocol (SDP).	5.3.4.7.1.9	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
212	If an LSC receives a call request from a served IP EI and the LSC has been unable to establish a TLS connection with its EBC and is unable to do so upon receipt of the INVITE, then the AS-SIP signaling appliance MUST ensure that the IP EI plays the Isolated Code Announcement (ICA) and terminates the call request and MUST send an alarm to the NMS.	5.3.4.7.1.10	NA	Y	NA
213	When an SS receives an INVITE from either a served LSC or another SS where the Request-URI has a DSN telephone number for which the SS has no entry in its Location Server, then the SS MUST respond with a 404 (Not Found) response code.	5.3.4.7.1.12	Y	Y	Y
214	When an LSC receives an inbound INVITE from its primary (or secondary) SS whose Request-URI has a DSN telephone number for which the LSC has no entry in its Location Server, then the LSC MUST respond with a 404 (Not Found) response message.	5.3.4.7.1.13	Y	Y	Y
215	The LSCs serving IP EIs MUST ensure that all outbound INVITEs forwarded onto the UC WAN include a Supported header with the option tag "100rel."	5.3.4.7.1.14	NA	Y	NA
216	When an AS-SIP signaling appliance receives an INVITE (having an sdp offer) and will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST return an sdp answer in the first non-failure reliable provisional response.	5.3.4.7.1.15	Y	Y	Y
217	When an LSC receives an INVITE (having an sdp offer) intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT return an sdp answer in any provisional response and MUST only place the sdp answer in the 200 response.	5.3.4.7.1.16	NA	Y	NA
218	When an AS-SIP signaling appliance receives an Empty INVITE (i.e., an INVITE that does not include an sdp offer) and said AS-SIP signaling appliance will be interworking the signaling to the TDM network, then the AS-SIP signaling appliance MUST send an sdp offer in the first reliable non-failure provisional response (1xx response code greater than a 100 response code).	5.3.4.7.1.17	Y	Y	Y
219	When an AS-SIP signaling appliance receives an Empty INVITE intended for a served IP EI, then the AS-SIP signaling appliance MUST NOT send an sdp offer in any provisional response (1xx response code greater than a 100 response code) and MUST only send the sdp offer in the 200 response.	5.3.4.7.1.18	Y	Y	Y
220	When an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 180 (Ringing) response from the IP network, the AS-SIP signaling appliance MUST ensure that the appropriate ring back tone (e.g., ring back, precedence ring back) is generated on the TDM network.	5.3.4.7.1.19	Y	Y	Y
221	Announcements are not sent in-band on the DSN TDM network; therefore, when an AS-SIP signaling appliance that is interworking SIP signaling with the TDM network receives a 480 (Temporarily Unavailable), 486 (Busy Here), or 488 (Not Acceptable Here) response from the IP network with either no Reason header or a Reason header that does NOT have a preemption cause, the AS-SIP signaling appliance does NOT generate an announcement to be sent to the TDM network, rather it sends either a REL with Q.850 cause code 46 precedence call blocked (in the case of SS7 ISUP) or a Disconnect (in the case of ISDN) with the appropriate cause code message to the TDM network.	5.3.4.7.1.20	Y	Y	Y
222	An LSC that receives an outbound call request from a served IP EI MAY include an audio media feature tag and a video media feature tag, as appropriate, in the Contact header field of the INVITE message.	5.3.4.7.1.21	NA	Y	NA
223	The AS-SIP signaling appliances are NOT required to process and act on the audio media tag and the video media tag in the Contact header but all intermediary AS-SIP signaling appliances MUST preserve the audio media tag (if present) and the video tag (if present) when forwarding the INVITE. (i.e., intermediary AS-SIP signaling appliances MUST NOT strip off or modify the media feature tags).	5.3.4.7.1.22	Y	Y	Y
224	When an LSC receives a call request from a served IP EI intended for a destination outside the enclave, then the AS-SIP signaling appliance MUST generate the P-Asserted-Identity header.	5.3.4.7.1.23	NA	Y	NA
225	The LSC serving the AS-SIP EI MUST support authentication of the AS-SIP EIs. The user of the AS-SIP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.26	NA	Y	NA
226	The LSCs serving IP EIs (other than AS-SIP EIs) MUST support authentication of the IP EIs. The user of the IP EI is required to perform user authentication to the LSC when initiating precedence call requests.	5.3.4.7.1.31	NA	Y	NA

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

<b>ID</b>	<b>Requirement</b>	<b>UCR Ref (UCR 2008 CH2)</b>	<b>MFSS</b>	<b>LSC</b>	<b>WAN SS</b>
227	When an LSC serving H.323 and/or proprietary EIs receives a request that contains a Require header field with one or more option tags that it does not understand, then it MUST return a 420 (Bad Extension) response code. The response MUST include an Unsupported header field listing those option tags the element did not understand.	5.3.4.7.1.35	NA	Y	NA
228	The LSCs and AS-SIP EIs MUST support the generating, receiving, and processing of SIP CANCEL requests.	5.3.4.7.2.2	NA	Y	NA
229	The AS-SIP signaling appliances MUST support the keep-alive mechanism for SIP sessions.	5.3.4.8.1.1	Y	Y	Y
230	The AS-SIP signaling appliances MUST support the generating, receiving, and processing of the Session-Expires and Min-SE header fields.	5.3.4.8.1.3	Y	Y	Y
231	The AS-SIP signaling appliances MUST support the 422 (Session Interval Too Small) response code.	5.3.4.8.1.4	Y	Y	Y
232	The AS-SIP signaling appliances MUST support the option tag "timer" for use with the Supported and Require header fields; however, an AS-SIP signaling appliance acting as a UAC or a SIP EI acting as a UAC MUST NOT place the option tag "timer" in either a Require header or a Proxy-Require header.	5.3.4.8.1.5	Y	Y	Y
233	When an AS-SIP signaling appliance receives an outbound request from a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAC behavior (when responsible for performing the refresh).	5.3.4.8.1.8	Y	Y	Y
234	When an AS-SIP signaling appliance receives a call request from another AS-SIP signaling appliance, and the destination is a served H.323 or proprietary IP EI, then the AS-SIP signaling appliance MUST operate in accordance with the UAS behavior (when responsible for performing the refresh).	5.3.4.8.1.10	Y	Y	Y
235	When SDP information is present in a SIP message, the SIP message MUST have a content-type header having the MIME Content-Type "application/sdp".	5.3.4.9.1.2	Y	Y	Y
236	The SDP parser in the AS-SIP signaling appliance (and all AS-SIP EIs, including AS-SIP video conferencing EIs) MUST be able to accept and handle without error any of the SDP line types enumerated in RFC 2327 even if the application ignores the contents.	5.3.4.9.1.3	Y	Y	Y
237	The precedence level of the call request MUST be set forth in a SIP Resource-Priority header field whose syntax is in accordance with RFC 4412, as modified in UCR 2008, Section 5.3.4.10.2	5.3.4.10.2.1	Y	Y	Y
238	Video telephony EIs MUST, as the default configuration, require an end user wishing to place a call that includes video, to affirmatively signal the intention to include video to the EI every time the caller wishes to engage in a video telephony call.	5.3.4.12.1.1	Y	Y	Y
239	Every time a caller requests a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before, or upon successful completion of, session establishment.	5.3.4.12.1.2	Y	Y	Y
240	When an INVITE with an sdp offer that includes both audio and video capabilities is received by an LSC serving a destination EI that supports video telephony, then when the call request is received by the destination EI the destination EI MUST indicate to the callee that a telephony call requesting video connectivity has been received.	5.3.4.12.2.1	Y	Y	Y
241	Every time a callee accepts a video telephony call and the video portion of the telephony call is successfully established, then the video telephony EI MUST provide the user with an affirmative confirmation that the video is enabled either before or upon successful session establishment.	5.3.4.12.2.3	Y	Y	Y
242	AS-SIP Signaling appliances must follow call flows depicted in section 5.3.4.13 for all call features and calling services.	5.3.4.13	Y	Y	Y
243	AS-SIP Signaling appliances must follow requirements depicted in section 5.3.4.14 for all IP to TDM and TDM to IP translations.	5.3.4.14	Y	Y	Y
244	When an interworking AS-SIP signaling appliance receives a request that contains a Require header field with one or more option-tags that it does not understand, then the interworking AS-SIP signaling appliance MUST return a 420 (Bad Extension) response. The response MUST include an Unsupported header field listing those option-tags the element did not understand.	5.3.4.16.1.1	Y	Y	Y
245	All outbound INVITEs generated by an interworking AS-SIP signaling appliance MUST include a Supported header with the option tag "100rel."	5.3.4.16.1.2	Y	Y	Y
246	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP INVITE requests. Interworking AS-SIP signaling appliances MUST support generating and receiving SIP re-INVITEs.	5.3.4.16.2.1	Y	Y	Y
247	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP CANCEL requests.	5.3.4.16.2.2	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
248	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP OPTIONS requests.	5.3.4.16.2.4	Y	Y	Y
249	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP BYE requests	5.3.4.16.2.5	Y	Y	Y
250	Interworking AS-SIP signaling appliances MUST support generating and receiving SIP ACK requests	5.3.4.16.2.6	Y	Y	Y
251	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP PRACK method. Interworking AS-SIP signaling appliances MUST support use of the option tag "100rel" with the Require header and Supported header, and MUST support the use of header fields RACK and RSeq.	5.3.4.16.2.8	Y	Y	Y
252	Interworking AS-SIP signaling appliances MUST support generating and receiving the SIP UPDATE method	5.3.4.16.2.9	Y	Y	Y
253	Inter-working AS-SIP signaling appliances MUST be capable of receiving/processing REFER requests, the Refer-To header, and the REFER event package.	5.3.4.16.2.10	Y	Y	Y
254	Interworking AS-SIP signaling appliances MUST support the NOTIFY method for event notification.	5.3.4.16.2.12	Y	Y	Y
256	Interworking AS-SIP signaling appliances MUST, in adherence with the enumerated RFCs, be capable of generating, receiving, and processing the SIP headers listed in UCR 2008 Section 5.3.4.16.3.1	5.3.4.16.3.1	Y	Y	Y
257	The From header MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.3	Y	Y	Y
258	The To header of a request that is part of a dialog MUST include a tag field as specified in RFC 3261, Section 19.3.	5.3.4.16.3.4	Y	Y	Y
259	Interworking AS-SIP signaling appliances MUST support the use of option tags for the Require, Supported, and Unsupported headers.	5.3.4.16.3.5	Y	Y	Y
260	When the interworking LSC sends an initial AS-SIP INVITE to its local EBC intended for its SS, the interworking LSC MUST add two Route header field values, which either takes the form of a route set comprising two Route headers where the first Route header is the sip uri for the EBC at the enclave and the second Route header is the sip uri for the EBC serving the SS, or takes the form of one Route header with two comma-separated field values.	5.3.4.16.3.6.1	NA	Y	NA
261	When an interworking SS forwards an initial AS-SIP INVITE to a peer SS, then the interworking SS MUST add a route set comprising two Route headers where the first Route header is the SIP URI for the EBC that serves the interworking SS, and the second Route header is the SIP URI for the EBC serving the peer SS.	5.3.4.16.3.7	Y	NA	Y
262	When an interworking AS-SIP signaling appliance generates an outbound AS-SIP request, the interworking AS-SIP signaling appliance MUST add its own VIA header to the AS-SIP request.	5.3.4.16.3.8	Y	Y	Y
263	When an interworking AS-SIP signaling appliance receives a SIP response to be translated into TDM signaling, then the interworking AS-SIP signaling appliance operates as the UAC for SIP purposes.	5.3.4.16.3.9	Y	Y	Y
264	When an interworking AS-SIP signaling appliance receives an inbound SIP request to be translated into TDM signaling, then the AS-SIP signaling appliance operates as the UAS for SIP purposes.	5.3.4.16.3.10	Y	Y	Y
265	When an interworking AS-SIP signaling appliance generates a SIP response on behalf of a signaling message received from the TDM network, then before forwarding the SIP response the interworking AS-SIP signaling appliance MUST include the VIA headers received in the corresponding SIP request.	5.3.4.16.3.11	Y	Y	Y
266	When an interworking AS-SIP signaling appliance operating as an originating gateway receives an IAM from the TDM network and sends an INVITE to another AS-SIP signaling appliance (SS or LSC), then the interworking AS-SIP signaling appliance MUST add a CCA-ID parameter to the SIP URI of the Contact header populated with its unique identifier before forwarding the INVITE onward to the next AS-SIP signaling appliance.	5.3.4.16.3.12	Y	Y	Y
267	Interworking AS-SIP signaling appliances MUST support generating, receiving, and processing the provisional (1xx) response codes: 100 (Trying), 180 (Ringing), and 183 (Session Progress).	5.3.4.16.4.1	Y	Y	Y
268	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the successful (2xx) response codes: 200 (OK) [RFC 3261, Section 21.2, 200 OK] and 202 (Accepted)	5.3.4.16.4.2	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
269	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the request failure (4xx) response codes: 400 (Bad Request), 401 (Unauthorized), 403 (Forbidden), 404 (Not Found), 405 (Method Not Allowed), 406 (Not Acceptable), 407 (Proxy Authentication Required), 408 (Request Timeout), 410 (Gone), 413 (Request Entity Too Large), 414 (Request-URI Too Long), 415 (Unsupported Media Type), 416 (Unsupported URI Scheme), 417 (Unknown Resource-Priority), 420 (Bad Extension), 421 (Extension Required), 422 (Session Interval Too Small), 423 (Interval Too Brief), 480 (Temporarily Unavailable), 481 (Call/Transaction Does Not Exist), 482 (Loop Detected), 483 (Too Many Hops), 484 (Address Incomplete), 485 (Ambiguous), 486 (Busy Here), 487 (Request Terminated), 488 (Not Acceptable Here), and 491 (Request Pending).	5.3.4.16.4.4	Y	Y	Y
270	Interworking AS-SIP signaling appliances upon properly receiving a CANCEL request for an INVITE MUST first send a 200 (OK) response code to the CANCEL, and then follow up with a 487 (Request Terminated) response code to the INVITE.	5.3.4.16.4.5	Y	Y	Y
271	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the server failure (5xx) response codes: 500 (Server Internal Error), 501 (Not Implemented), 502 (Bad Gateway), 503 (Service Unavailable), 504 (Server Timeout), 505 (Version Not Supported), 513 (Message Too Large) [RFC 3261, Section 21.5, Server Failure 5xx], and 580 (Precondition Failure)	5.3.4.16.4.6	Y	Y	Y
272	Interworking AS-SIP signaling appliances MUST support generating and receiving/processing the global failures (6xx) response codes: 600 (Busy Everywhere), 603 (Decline), 604 (Does Not Exist Anywhere), and 606 (Not Acceptable).	5.3.4.16.4.7	Y	Y	Y
273	When an interworking AS-SIP signaling appliance receives an outbound request from the PSTN (i.e., the interworking AS-SIP signaling appliance is operating as an originating gateway) and the destination is NOT an IP EI directly served by the interworking AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAC behavior set forth in RFC 4028.	5.3.4.17.1.1	Y	Y	Y
274	When an interworking AS-SIP signaling appliance acting as a terminating gateway receives a call request from another AS-SIP signaling appliance, then the interworking AS-SIP signaling appliance MUST operate in accordance with the UAS behavior set forth in RFC 4028.	5.3.4.17.1.3	Y	Y	Y
275	Interworking AS-SIP signaling appliances MUST support the option tag "resource-priority" for use with the Require header.	5.3.4.18.3.1	Y	Y	Y
276	The interworking AS-SIP signaling appliance MUST receive and accept a Require header field with the option tag "resource-priority" in the INVITE, UPDATE, and REFER messages. Interworking AS-SIP signaling appliances MUST NOT reject the message with a 420 (Bad Extension) response code, but rather it MUST accept the request and translate it into the appropriate TDM signaling message as required.	5.3.4.18.3.2	Y	Y	Y
277	If an interworking AS-SIP signaling appliance receives an inbound ROUTINE call request over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, the interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth).	5.3.4.18.4.5	Y	Y	Y
278	If an interworking AS-SIP signaling appliance receives an inbound precedence call request (i.e., with precedence level PRIORITY or above) over the IP network for a destination on the TDM network and the interworking AS-SIP signaling appliance has insufficient bandwidth-related resources (e.g., lack of circuit-switched trunk capacity for bearer traffic) to handle the call request, and if there are insufficient existing calls (and/or call requests) of lower precedence whose removal would provide the necessary resources to support the pending call request, then: - The interworking AS-SIP signaling appliance MUST reply with a 488 (Not Acceptable Here) response code and SHOULD include a Warning header with warning code 370 (Insufficient Bandwidth), and - The AS-SIP signaling appliance serving the calling IP EI MUST arrange for a BPA to be played to the calling IP EI before terminating the call.	5.3.4.18.4.6	Y	Y	Y
279	AS-SIP signaling appliances must follow all call flows depicted in UCR 2008 Section 5.3.4.19 for all supplementary services.	5.3.4.19	Y	Y	Y
280	The product shall support dual IPv4 and IPv6 stacks as described in RFC 4213.	5.3.5.4	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
281	Dual stack end points or Call Control Agents shall be configured to choose IPv4 over IPv6.	5.3.5.4	Y	Y	Y
282	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.	5.3.5.4	Y	Y	Y
283	The product shall support the IPv6 format as described in RFC 2460 and updated by RFC 5095.	5.3.5.4	Y	Y	Y
284	The product shall support the transmission of IPv6 packets over Ethernet networks using the frame format defined in RFC 2464.	5.3.5.4	Y	Y	Y
285	The product shall support a minimum MTU of 1280 bytes.	5.3.5.4.1	Y	Y	Y
286	The product shall not use the Flow Label field as described in RFC 2460. The product shall be capable of setting the Flow Label field to zero when originating a packet. The product shall not modify the Flow Label field when forwarding packets. The product shall be capable of ignoring the Flow Label field when receiving packets.	5.3.5.4.2	Y	Y	Y
287	The product shall support the IPv6 Addressing Architecture as described in RFC 4291.	5.3.5.4.3	Y	Y	Y
288	The product shall support the IPv6 Scoped Address Architecture as described in RFC 4007.	5.3.5.4.3	Y	Y	Y
289	The product shall support Neighbor Discovery for IPv6 as described in RFC 2461 and RFC 4861.	5.3.5.4.5	Y	Y	Y
290	The product shall not set the override flag bit in the Neighbor Advertisement message for solicited advertisements for anycast addresses or solicited proxy advertisements.	5.3.5.4.5	Y	Y	Y
291	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.	5.3.5.4.5	Y	Y	Y
292	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.	5.3.5.4.5	Y	Y	Y
293	When address resolution fails on a neighboring address, the entry shall be deleted from the product's neighbor cache.	5.3.5.4.5	Y	Y	Y
294	The product shall support the ability to configure the product to ignore Redirect messages. The product shall only accept Redirect messages from the same router as is currently being used for that destination.	5.3.5.4.5.1	Y	Y	Y
295	If the product supports routing functions, the product shall inspect valid router advertisements sent by other routers and verify that the routers are advertising consistent information on a link and shall log any inconsistent router advertisements. The product shall prefer routers that are reachable over routers whose reachability is suspect or unknown.	5.3.5.4.5.2	NA	NA	NA
296	The product shall support manual assignment of IPv6 addresses.	5.3.5.4.6	Y	Y	Y
297	The product shall support the ICMPv6 as described in RFC 4443. The product shall have a configurable rate limiting parameter for rate limiting the forwarding of ICMP messages.	5.3.5.4.7	Y	Y	Y
298	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.	5.3.5.4.7	Y	Y	Y
299	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.	5.3.5.4.7	Y	Y	Y
300	The product shall validate ICMPv6 messages, using the information contained in the payload, before acting on them.	5.3.5.4.7	Y	Y	Y
301	If the product supports routing functions, the product shall support the Multicast Listener Discovery (MLD) process as described in RFC 2710 and extended in RFC 3810.	5.3.5.4.8	Y	Y	Y
302	For traffic engineering purposes, the bandwidth required per voice subscriber is calculated to be 110.0 kbps (each direction) for each IPv6 call.	5.3.5.4.11	Y	Y	Y
303	The product shall forward packets using the same IP Version as the Version in the received packet.	5.3.5.4.12	Y	Y	Y
304	The product shall use the Alternative Network Address Types (ANAT) semantics for the Session Description Protocol (SDP) in accordance with RFC 4091 when establishing media streams from dual-stacked appliances for AS-SIP signaled sessions.	5.3.5.4.12	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
305	The product shall prefer any IPv4 address to any IPv6 address when using ANAT semantics.	5.3.5.4.12	Y	Y	Y
306	The product shall place the option tag "SDP-ANAT" in a Required header field when using ANAT semantics in accordance with RFC 4092.	5.3.5.4.12	Y	Y	Y
307	The products shall support Differentiated Services as described in RFC 2474 for a voice and video stream in accordance with Section 5.3.2, Assured Services Requirements, and Section 5.3.3, Network Infrastructure E2E Performance Requirements, plain text DSCP plan.	5.3.5.4.14	Y	Y	Y
308	The LSC must meet all requirements for FCAPS Management and audit logs as listed in UCR 2008 section 5.3.2.7.2.6	5.3.2.7.2.6	NA	Y	NA
309	The physical interface between the DISA VVoIP EMS and the network components (i.e., LSC, MFSS, EBC, CE Router) is a 10/100-Mbps Ethernet interface. The interface will 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.	5.3.2.4.4	Y	Y	Y
310	Redundant physical Ethernet interfaces are required for signaling and bearer traffic. If the primary signaling and bearer Ethernet interface fails, then traffic shall be switched to the backup signaling and bearer Ethernet interface.	5.3.2.4.4	Y	Y	Y
311	The MFSS shall provide a single, common interface to the DISA NMS. The single interface shall provide access to MFSS features and functions for both the TDM and SS side of the MFSS.	5.3.2.8.3.1	Y	NA	Y
312	The MFSS-to-NMS interface shall be an Ethernet connection as specified in Section 5.3.2.4.4, VVoIP NMS Interface Requirements.	5.3.2.8.3.1	Y	NA	Y
313	LSCs and SSs must be capable of providing the following NM data to the E2E RTS EMS: • Alarm/log data • Performance data (e.g., traffic data) • Accounting data (e.g., call detail recording)	5.3.2.17.2	Y	Y	Y
314	LSCs and SSs must allow the E2E RTS EMS to have access to perform LSC/SS datafill administration and network controls.	5.3.2.17.2	Y	Y	Y
315	Communications between VVoIP EMS and the VVoIP network appliances shall be via IP.	5.3.2.17.2	Y	Y	Y
316	A network appliance shall issue state change notifications for changes in the states of replaceable components, including changes in operational state or service status, and detection of new components.	5.3.2.17.2	Y	Y	Y
317	A network appliance shall be provisioned by the VVoIP EMS with the address and Transport Layer port information associated with its Core Network interfaces.	5.3.2.17.2	Y	Y	Y
318	A network appliance shall be capable of maintaining and responding to VVoIP EMS requests for resource inventory, configuration, and status information concerning Core Network interface resources (e.g., IP or MAC addresses) that have been installed and placed into service.	5.3.2.17.2	Y	Y	Y
319	A network appliance shall be capable of setting the Administrative state and maintaining the Operational state of each Core Network interface, and maintaining the time of the last state change.	5.3.2.17.2	Y	Y	Y
320	Alarm messages must be distinguishable from administrative log messages.	5.3.2.17.3.1.1	Y	Y	Y
321	The NEs shall detect their own fault (alarm) conditions.	5.3.2.17.3.1.2	Y	Y	Y
322	The NEs shall generate alarm notifications.	5.3.2.17.3.1.3	Y	Y	Y
323	The network elements shall send the alarm messages in NRT. More than 99.95 percent of alarms shall be detected and reported in NRT. Near Real Time is defined as event detection and alarm reporting within 5 seconds of the event, excluding transport time.	5.3.2.17.3.1.4	Y	Y	Y
324	The network components shall send alarm messages in SNMPv3 format.	5.3.2.17.3.1.5	Y	Y	Y
325	Capability to access and modify configuration data by the VVoIP EMS shall be controllable by using an access privileges function within the network appliance.	5.3.2.17.3.2.1	Y	Y	Y
326	The VVoIP NEs shall be able to receive and respond to remote NM commands.	5.3.2.17.3.4.2	Y	Y	Y
327	When ASAC budgets are reduced, by NM action, below the current budget allocation, any previous sessions (regardless of precedence level) in excess of the new budget shall be allowed to terminate naturally. This assumes that the CE Router queue bandwidths would not be reduced until the LSC session count fell below or equal to the newly commanded reduced budget, to prevent the corruption of existing sessions.	5.3.2.17.3.4.2.2	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
328	The LSC, MFSS, and WAN SS shall have the capability of setting the percentage of calls to be blocked to the designated destination(s).	5.3.2.17.3.4.2.7	Y	Y	Y
329	FLASH and FLASH-OVERRIDE calls shall not be affected by NM controls.	5.3.2.17.3.4.2.7	Y	Y	Y
330	Within IP, directionalization is controlled by designating all or part of the call budget as inbound (i.e., local destination) and/or outbound (i.e., local origination). The default is no designation (i.e., calls up to the total budget can be inbound or outbound in any combination). It does not change the total budget, only the sourcing direction of the budget; therefore, there is no impact to the router queue bandwidths.	5.3.2.17.3.4.2.10	C	C	C
331	Within IP, the routing of all traffic (i.e., VVoIP and non-VVoIP) is handled via MPLS in the DISN core. The MPLS automatically finds the most effective route for the traffic.	5.3.2.17.3.4.2.11	Y	Y	Y
332	The WAN-level ASAC must be able to account for each subtended LSC under its control. Therefore, the MFSS and WAN SS ASAC must be able to set call budgets for multiple LSC locations via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13	Y	NA	Y
333	The LSC-level ASAC is required to only account for itself. Therefore, the LSC ASAC must be able to set call budgets for only the PEI/AEIs under its control via the VVoIP EMS and local EMS access points.	5.3.2.17.3.4.2.13	NA	Y	NA
334	The product shall have the capability of setting a PEI/AEI's maximum allowed precedence level for originating a call. This is a "subscriber class mark feature," which is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14	NA	Y	NA
335	The product shall have the capability of controlling the destination(s) that a PEI or AEI is restricted from calling. This is a subscriber class mark feature that is controlled by the LSC system administrator.	5.3.2.17.3.4.2.14	NA	Y	NA
336	The ASAC must provide the separate counts for voice and video, in 5-minute intervals. The MFSS and WAN SS ASAC must provide these counts for each of the subtended LSCs under its control, while the LSC is only to provide these counts for the PEIs/AEIs that it controls.	5.3.2.18.2	Y	Y	Y
337	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data: 1. Host Name of the CCA controlling the call processing. 2. Start Date of call (In Julian or Calendar). 3. Start Time of Call (Hour + Minute + Second). 4. Elapsed Time of Call and/or Stop Time of call. 5. Calling Number. 6. Called Number (included all dialed digits). 7. Precedence level of call. (NOTE: This may be accomplished either by a specific precedence level designation field in the call, or by providing the dialed precedence level access digits in the called number field.)	5.3.2.19.2.1	Y	Y	Y
338	For the selected recording format that is chosen, of all the call information that will be provided, the following call data shall be provided in the record data if it applies to the call: Conference Call Indicator.	5.3.2.19.2.1	Y	Y	Y
339	The product shall provide a voice quality record at the completion of each voice session. The voice quality record shall be included in the CDR that the LSC, MFSS, or WAN SS generates for that session, and shall conform to the E-Model, as described in TIA TSB-116-A, and ITU-T Recommendation G.107. The voice quality record shall contain the calculated R-Factor for the Voice session per TIA TSB-116-A.	5.3.2.19.2.1.1	Y	Y	Y
340	As part of the voice quality record, the product shall provide the raw voice session statistics that are used to make the R-Factor calculation to include, as a minimum, the latency, packet loss, Equipment Impairment Factor (Ie), and the TCLw. The product shall provide the jitter for the session.	5.3.2.19.2.1.1	Y	Y	Y
341	The product shall generate an alarm to the VVoIP EMS when the session R-Factor calculation in the CDR fails to meet a configurable threshold. By default, the threshold shall be an R-Factor value of 80, which is equivalent to an MOS value of 4.0.	5.3.2.19.2.1.1	Y	Y	Y
342	The mass storage in the BA must be non-volatile. The mass storage in the BA must be able to retain at least five average-busy-season business days of AMA data. (NOTE: This is needed to provide adequate capacity for high-volume storage of CDRs.)	5.3.2.19.2.3	Y	Y	Y
343	The BA should be able to output the records electronically over a secured connection. The BA should have the ability to transfer the records to a physical storage media that is also removable.	5.3.2.19.2.4	Y	Y	Y

**Table 3-1. Unified Capabilities Products Capability/Functional Requirements Table (continued)**

<b>LEGEND:</b>	
ACD	Automatic Call Distributor
AEI	AS-SIP End Instrument
AMA	Automatic Message Accounting
ANAT	Alternative Network Address Types
ANSI	American National Standards Institute
ASAC	Assured Services Admission Control
ASLAN	Assured Services Local Area Network
AS-SIP	Assured Services Session Initiation Protocol
ATQA	Attendant Queue Announcement
B2BUA	Back-to-back User Agent
BA	Billing Agent
BER	Bit Error Rate
BLV	Busy Line Verification
BNF	Backus-Naur Form
C	Conditional
C2	Command and Control
CAC	Common Access Card
CAS	Channel Associated Signaling
CCA	Call Control Agent
CCS7	Common Channel Signaling 7
CDR	Call Data Record
CE	Customer Edge
CF	Call Forward
CH1	Change 1
CID	Craft Input Device
CND	Calling Number Delivery
CONUS	Continental United States
D-Channel	Data Channel
DB	Database
DHCP	Dynamic Host Configuration Protocol
DISA	Defense Information Systems Agency
DISN	Defense Information System Network
DN	Directory Number
DoD	Department of Defense
DS1	Digital Signal Level 1
DS3	Digital Signal Level 3
DSCP	Differentiated Services Code Point
DSN	Defense Switched Network
E2E	End-to-end
EBC	Edge Boundary Controller
EC	Echo Cancellor
EI	End Instrument
EMS	Element Management System
EO	End Office
ETSI	European Telecommunications Standards Institute
FCAPS	Fault, Configuration, Accounting, Performance, and Security
FoIP	Fax over Internet Protocol
FQDN	Fully Qualified Domain Name
G3 Fax	Group 3 Facsimile
Hz	Hertz
IAD	Integrated Access Device
IAW	In Accordance With
ICA	Isolated Code Announcement
ID	Identification
ICMPv6	Internet Control Message Protocol for IPv6
Ie	Equipment Impairment Factor
IEEE	Institute of Electrical and Electronics Engineers, Inc.
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPB	IP ASAC Budget
IPC	IP ASAC Call Count
IPSec	Internet Protocol Security
IPv4	Internet Protocol Version 4
IPv6	Internet Protocol Version 6
ISDN	Integrated Services Digital Network
ISUP	ISN User Part
ITU-T	International Telecommunications Union – Telecommunication Standardization Sector
IWF	Interworking Function
	kbps
	LAN
	LDAP
	LDAPv3
	LSC
	MAC
	Mbps
	MFS
	MFSS
	MG
	MGC
	MLD
	MLPP
	Modem
	MolP
	MOS
	MPLS
	ms
	MTU
	NE
	NM
	NMS
	OA&M
	OCONUS
	OSI
	OTGR
	PBAS
	PBX
	PCD
	PCM
	PEI
	PLCP
	PRI
	PSTN
	REL
	RFC
	RTS
	SAC
	SBU
	SCIP
	SCS
	SDP
	SG
	SIP
	SMEO
	SONET
	SRTCP
	SRTP
	SS
	SS7
	SUT
	TA
	TCA
	TCLw
	TDM
	TIA
	TLS
	UAC
	UAS
	UC
	UCR
	UDP
	URI
	U.S.
	VBD
	VoIP
	VVoIP
	WAN
	Y
	kilobits per second
	Local Area Network
	Lightweight Directory Access Protocol
	Lightweight Directory Access Protocol version 3
	Local Session Controller
	Media Access Control
	Megabytes per second
	Multifunction Switch
	Multifunction SoftSwitch
	Media Gateway
	Media Gateway Controller
	Multicast Listener Discovery
	Multilevel Precedence and Preemption
	Modulator/Demodulator
	Modem over Internet Protocol
	Mean Opinion Score
	Multiprotocol Label Switching
	milliseconds
	Maximum Transmission Unit
	Network Element
	Network Management
	Network Management System
	Operations, Administration, and Maintenance
	Outside the Continental United States
	Open Systems Interconnect
	Operations Technology Generic Requirements
	Precedence Based Assured Services
	Private Branch Exchange
	Precedence Call Diversion
	Pulse Code Modulation
	Proprietary End Instrument
	Physical Layer Convergence Protocol
	Primary Rate Interface
	Public Switch Telephone Network
	Release Message
	Request For Communication
	Real Time Services
	Session Admission Control
	Sensitive, but Unclassified
	Secure Communications Interoperability Protocol
	Session Control and Signaling
	Session Description Protocol
	Signaling Gateway
	Session Initiation Protocol
	Small End Office
	Synchronous Optical Network
	Secure Real-Time Transport Control Protocol
	Secure Real-Time Transport Protocol
	Softswitch
	Signaling System number 7
	System Under Test
	Terminal Adaptor
	Threshold Crossing Alert
	Weighted Terminal Coupling Loss
	Time Division Multiplexing
	Telecommunications Industry Association
	Transport Layer Security
	User Agent Client
	User Agent Server
	Unified Capabilities
	Unified Capabilities Requirements
	User Datagram Protocol
	Uniform Resource Identifier
	United States
	Voice Band Data
	Voice over Internet Protocol
	Voice and Video over Internet Protocol
	Wide Area Network
	Yes