



## DEFENSE INFORMATION SYSTEMS AGENCY

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

6 Sep 12

### MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of the Cisco Unified Communications Manager (CUCM) Version 8.6.1

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 Defense Information Systems Agency (DISA) Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The CUCM Version 8.6.1 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 a Local Session Controller (LSC). The fielding of the SUT is limited to IP version 4 (IPv4) across the DISN. Although the SUT supports Internet Protocol version 6 (IPv6), it was not tested inter-enclave because of test limitations within the network infrastructure. Therefore, inter-enclave IPv6 is not covered under this certification, but intra-enclave use of IPv6 is authorized for 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 DISA via a vendor Plan of Action and Milestones (POA&M), which will address all new critical Test Discrepancy Reports (TDRs) within 120 days of identification. Testing was conducted using LSC product requirements derived from the Unified Capabilities Requirements (UCR), Reference (c), and LSC test procedures, 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 (UC) Approved Products List (APL) memorandum.

3. This finding is based on interoperability testing conducted by JITC, review of the vendor's Letters of Compliance (LoC), DISA adjudication of open TDRs, and DISA Information Assurance (IA) Certification Authority (CA) approval of the IA configuration. Interoperability testing was conducted by JITC, Fort Huachuca, Arizona, from 11 July through 5 August 2011. Verification and Validation testing was conducted by JITC, Fort Huachuca, Arizona, from 5 through 23 December 2011. Review of the vendor's LoC was completed on 19 June 2012. Adjudication of open TDRs was completed by DISA on 31 July 2012. The DISA CA provided a positive recommendation on 14 June 2012 based on the security testing completed by DISA-led

IA test teams and published in separate report, Reference (e). The acquiring agency or site will be responsible for the 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 Requirements <sup>1</sup>	Status	Remarks
<b>Line Interfaces</b>					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs and softphones.
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs and softphones.
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3ab. Applies to PEIs and softphones.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, and 13	Certified	Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments.
BRI	No	5.3.2.6.1.8	2, 4, 10, and 13	Not Tested	This interface is offered by the SUT; however, it was not tested because it does not support Assured Services.
<b>External Interfaces</b>					
10Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs. Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 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, and 13	Not Tested	This interface is not offered by the SUT.
T1 CAS	No	5.3.2.12.11	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs for DTMF.
E1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Not Tested	This interface is offered by the SUT; however, it was not tested and is not covered under this certification.
E1 PRI ITU-T Q.955.3	No <sup>3</sup>	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Not Certified	This interface is offered by the SUT; however, it was not tested and is not covered under this certification.
E1 PRI ITU-T Q.931	No <sup>3</sup>	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs for European PSTN connectivity.
<b>NM</b>					
10Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.

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

<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. The SUT must provide a minimum of one of the listed interfaces.			
3. This interface is conditionally required for deployment in Europe.			
<b>LEGEND:</b>			
10Base-X	10 Mbps Ethernet	IEEE	Institute of Electrical and Electronics Engineers
100Base-X	100 Mbps Ethernet	ISDN	Integrated Services Digital Network
1000Base-X	1000 Mbps Ethernet	ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
802.3ab	1000 Mbps Ethernet over Twisted Pair		
802.3i	10 Mbps twisted pair media for 10Base-X networks	LoC	Letter of Compliance
802.3j	10 Mbps fiber media for 10Base-X networks	Mbps	Megabits per second
802.3u	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbps with auto negotiation	MFR1	Multi-Frequency Recommendation 1
802.3z	Standard for Gigabit Ethernet	MG	Media Gateway
ANSI	American National Standards Institute	MLPP	Multi-Level Precedence and Preemption
AS-SIP	Assured Services Session Initiation Protocol	NI-2	National ISDN Standard 2
BRI	Basic Rate Interface	NM	Network Management
CAS	Channel Associated Signaling	PEI	Proprietary End Instrument
CCS7	Common Channel Signaling	PRI	Primary Rate Interface
CR	Capability Requirement	PSTN	Public Switched Telephone Network
DP	Dial Pulse	Q.931	Signaling Standard for ISDN
DSN	Defense Switched Network	Q.955.3	ISDN Signaling Standard for E1 MLPP
DTMF	Dual Tone Multi-Frequency	SS7	Signaling System 7
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
		UCR	Unified Capabilities Requirements

**Table 2. SUT Capability Requirements and Functional Requirements Status**

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
<b>1</b>	<b>Assured Services Product Features and Capabilities</b>			
	DSCP Packet Marking	Required	5.3.2.2.1.4	Met
	Voice Features and Capabilities	Required	5.3.2.2.2.1	Met <sup>2</sup>
	Public Safety Features	Required	5.3.2.2.2.2	Met
	ASAC – Open Loop	Required	5.3.2.2.2.3	Met
	Signaling Protocols	Required	5.3.2.2.3	Met
	Signaling Performance	Conditional	5.3.2.2.4	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 <sup>3</sup>
<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 Capability Requirements and Functional Requirements Status (continued)**

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
4	<b>Voice End Instruments</b>			
	Tones and Announcements	Required	5.3.2.6.1.1	Met <sup>4</sup>
	Audio Codecs	Required	5.3.2.6.1.2	Met <sup>4,5</sup>
	VoIP PEI or AEI Audio Performance	Required	5.3.2.6.1.3	Met <sup>4</sup>
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Met <sup>4</sup>
	Authentication to LSC	Required	5.3.2.6.1.5	Met <sup>4</sup>
	Analog Telephone Support	Required	5.3.2.6.1.6	Met
	Softphones	Conditional	5.3.2.6.1.7	Partially Met <sup>6</sup>
	ISDN BRI	Conditional	5.3.2.6.1.8	Not Tested
5	<b>Video End Instruments</b>			
	Video End Instrument	Required	5.3.2.6.2	Not Tested <sup>7</sup>
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Tested <sup>7</sup>
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Tested <sup>7</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>4,8</sup>
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	Met
	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	Partially Met <sup>9,10</sup>
	CCA MGC Component	Required	5.3.2.9.2.2	Met
	SG Component	Conditional	5.3.2.9.2.3	Not Tested <sup>9</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>9</sup>
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Partially Met <sup>10</sup>
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested <sup>9</sup>
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Partially Met <sup>11</sup>
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met <sup>9</sup>
	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 <sup>12</sup>
	CCA Interaction with EIs	Required	5.3.2.10.10	Partially Met <sup>4</sup>
CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met <sup>7</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>9</sup>	

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

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
8	<b>MG Requirements</b>			
	Role of MG In LSC	Required	5.3.2.12.3.1	Met
	MG Support for ASAC	Required	5.3.2.12.4.1	Met
	MG and IA Functions	Required	5.3.2.12.4.2	Met <sup>12</sup>
	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
	MG Interaction with VoIP EIs	Required	5.3.2.12.4.8	Met <sup>4</sup>
	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	Partially Met <sup>9, 10, 13</sup>
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Partially Met <sup>10, 13</sup>
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Partially Met <sup>13</sup>
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested <sup>9</sup>
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Partially Met <sup>10</sup>
	MG Support for CAS Trunks	Required	5.3.2.12.11	Met
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met
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	Not Met <sup>14</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
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested
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>15</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
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	Not Tested <sup>8</sup>
	Call Display	Required	5.3.2.26.2	Not Tested <sup>8</sup>
	Class of Service Override	Required	5.3.2.26.3	Not Tested <sup>8</sup>
	Busy Override and Busy Verification	Required	5.3.2.26.4	Not Tested <sup>8, 16</sup>
	Night service	Required	5.3.2.26.5	Not Tested <sup>8</sup>
	Automatic Recall of Attendant	Required	5.3.2.26.6	Not Tested <sup>8</sup>
	Calls in Queue to the Attendant	Required	5.3.2.26.7	Not Tested <sup>8, 17</sup>

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

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
15	<b>AS-SIP Requirements</b>			
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required	5.3.4.7	Not Tested <sup>4</sup>
	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>7</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 <sup>18</sup>
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Not Tested <sup>19</sup>
	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	Met	
16	<b>IPv6 Requirements</b>			
	Product Requirements	Required	5.3.5.4	Partially Met <sup>20</sup>
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	Met
Accounting Management	Required	5.3.2.19	Partially Met <sup>21</sup>	
<b>NOTES:</b>				
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.				
2. The SUT does not support a "ping ring" notification. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).				
3. The SUT does not support OPTIONS requests required to meet the failover to a secondary SS in accordance with UCR 2008, Change 2, section 5.3.2.3.2.1. DISA adjudicated this discrepancy and determined that the UCR failover requirements are immature and require a rewrite. DISA NS2 has agreed to a Condition of Fielding that the initial UC APL certification will not provide for failover capability on the condition the vendor will participate in an NS2-scheduled multi-vendor test event to refine failover requirements, modify software to support the new failover requirements, and demonstrate failover compliance.				
4. The SUT only supports voice PEIs. The vendor does not support AEIs (voice or video). DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.				
5. The SUT gateways equipped with the PVD3 modules do not support the ITU-T G.723 codec. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).				
6. The SUT softphone with Microsoft Windows Vista and Windows 7 OSs does not allow DSCP tagging per precedence level in accordance with UCR 2008, Change 2, Section 5.3.3.3.2. Microsoft Windows XP is the only OS that supports the five precedence levels. DISA adjudicated this as minor since all voice is queued together in the four-queue model currently used in deployed ASLANs.				
7. The SUT did not offer a video PEI. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.				
8. The SUT Operator Console/Attendant Station was not tested; however the vendor submitted an LoC for the requirements.				
9. The SUT met T1 ISDN PRI (ANSI T1.619a and ANSI T1.607), E1 PRI (ITU-T Q.931), and T1 CAS DTMF IWF requirements, which is all of the certified TDM interfaces.				
10. The SUT does not support NFAS on the T1 ISDN PRI interface. Although this is conditional for DSN connectivity, it is required for PSTN connectivity. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).				
11. The SUT met PEI CCA-IWF requirements. The SUT does not support AEIs. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.				

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

**NOTES (continued):**

12. The security requirements are tested by a DISA-led IA test team and published in a separate report, Reference (e).
13. The SUT must meet T1 PRI (T1.619a and NI2) IWF. The T1 CAS and T1 CCS7 IWF requirements are conditional. The SUT met T1 ISDN PRI (ANSI T1.619a and ANSI T1.607), E1 PRI (ITU-T Q.931), and T1 CAS DTMF IWF requirements.
14. The SUT does not properly handle ITU-T V.150 calls with the Avaya Communication Manager 6.0 and both vendors are working on the problem. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
15. The SUT does not support Commercial Cost Avoidance. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
16. The SUT does not fully comply with Busy Override and Busy Line Verification requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
17. The SUT does not fully comply with attendant console queuing requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
18. The SUT met this requirement with ANSI T1.619a ISDN PRI NI2 DSN and ISDN PRI NI2 PSTN TDM interfaces interworking with AS-SIP. This requirement was met with both testing and the vendor's LoC. The SUT does not support CCS7 TDM interface which is conditional for an LSC.
19. This requirement applies to gateways between AS-SIP and CCS7 links. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.
20. The vendor submitted an IPv6 LoC with noted discrepancies. The SUT does not support RFCs 4861 and 4862. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
21. The vendor submitted an NM LoC with noted discrepancies. The SUT does not comply with the requirement for the equipment impairment factor to be in accordance with ITU-T G.107. DISA adjudicated this as minor with a vendor POA&M to provide a MOS score. DISA also stated the intent to change this to conditional in the next version of the UCR (UCR 2013). The SUT does not have the ability to transfer records to a removable physical storage media. DISA adjudicated this as minor and stated the intent to delete this requirement in the next version of the UCR (UCR 2013).

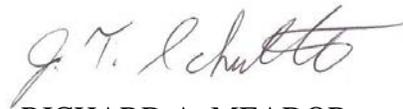
**LEGEND:**

AEI	AS-SIP End Instrument	MLPP	Multi-Level Precedence and Preemption
ANSI	American National Standards Institute	MOS	Mean Opinion Score
APL	Approved Products List	NFAS	Non Facility Associated Signaling
AS	Assured Services	NI2	National ISDN Standard 2
ASAC	Assured Services Admission Control	NM	Network Management
ASLAN	Assured Services Local Area Network	NMS	Network Management System
AS-SIP	Assured Services Session Initiation Protocol	OCONUS	Outside the Continental United States
BRI	Basic Rate Interface	OS	Operating System
CAS	Channel Associated Signaling	PBAS	Precedence Based Assured Services
CCA	Call Connection Agent	PEI	Proprietary End Instrument
CR	Capability Requirement	POA&M	Plan of Action and Milestones
CCS7	Common Channel Signaling	PRI	Primary Rate Interface
DISA	Defense Information Systems Agency	PSTN	Public Switched Telephone Network
DSCP	Differentiated Services Code Point	PVDM3	Packet Voice Digital Signal Processor Module 3
DSN	Defense Switched Network	Q.931	Signaling Standard for ISDN
DSS1	Digital Subscriber Signaling 1	RFCs	Request for Comments
DTMF	Dual Tone Multi-Frequency	SG	Signaling Gateway
E1	European Basic Multiplex Rate (2.048 Mbps)	SIP	Session Initiation Protocol
EBC	Edge Boundary Controller	SS	Softswitch
EI	End Instrument	SS7	Signaling System 7
FCAPS	Fault, Configuration, Accounting, Performance and Security	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IA	Information Assurance	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
ISDN	Integrated Services Digital Network	TDM	Time Division Multiplexing
IP	Internet Protocol	UC	Unified Capabilities
IPv6	Internet Protocol version 6	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UFS	User Features and Services
IWF	Interworking Function	US	United States
LoC	Letter of Compliance	V.150	Modem over Internet Protocol Networks
LSC	Local Session Controller	VoIP	Voice over Internet Protocol
Mbps	Megabits per second	VVoIP	Voice and Video over Internet Protocol
MG	Media Gateway	WAN	Wide Area Network
MGC	Media Gateway Controller	WWNDP	Worldwide 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>. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: [disa.meade.ns.list.unified-capabilities-certification-office@mail.mil](mailto:disa.meade.ns.list.unified-capabilities-certification-office@mail.mil).

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to [edward.a.mellon.civ@mail.mil](mailto:edward.a.mellon.civ@mail.mil). The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number is 1108301.

FOR THE COMMANDER:



for RICHARD A. MEADOR  
Chief  
Battlespace Communications Portfolio

3 Enclosures a/s

Distribution (electronic mail):

DoD CIO

Joint Staff J-6, JCS

USD(AT&L)

ISG Secretariat, DISA, JTA

U.S. Strategic Command, J665

US Navy, OPNAV N2/N6FP12

US Army, DA-OSA, CIO/G-6 ASA(ALT), SAIS-IOQ

US Air Force, A3CNN/A6CNN

US Marine Corps, MARCORSYSCOM, SIAT, A&CE Division

US Coast Guard, CG-64

DISA/TEMC

DIA, Office of the Acquisition Executive

NSG Interoperability Assessment Team

DOT&E, Netcentric Systems and Naval Warfare

Medical Health Systems, JMIS IV&V

HQUSAISEC, AMSEL-IE-IS

UCCO

## **ADDITIONAL REFERENCES**

- (c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008, Change 2," December 2010
- (d) Joint Interoperability Test Command, "Unified Capabilities Test Plan (UCTP)," Draft
- (e) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM), Version 8.6.1, (TN 1108301)," Draft

## CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** The Cisco Unified Communications Manager (CUCM) Version 8.6.1 with specified releases; hereinafter referred to as the System Under Test (SUT).
- 2. SPONSOR.** United States Air Force, Attention: Clint Rasic, HQ USAFE/A6CE, Address: Unit 3050 Box 125, APO AE 09094, e-mail: [clint.rasic@ramstein.af.mil](mailto:clint.rasic@ramstein.af.mil).
- 3. SYSTEM POC.** Cisco Systems Global Certification Team (GCT), 7025-2 Kit Creek Road, Research Triangle Park, North Carolina 27709, e-mail: [certteam@cisco.com](mailto:certteam@cisco.com).
- 4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.
- 5. SYSTEM DESCRIPTION.** The SUT is an enterprise Internet Protocol (IP) telephony call-processing solution that is scalable, distributable, and highly available. Multiple CUCM servers are clustered and managed as a single entity on an IP network, a capability which yields scalability of 1 to 30,000 IP phones per cluster, load balancing, and call-processing service redundancy. The SUT software is installed on the Cisco Unified Computing System (UCS) Server platforms or equivalent hardware. The tested UCS platforms were the UCS 5108 chassis containing the B200 M2 blade and the UCS C210 and C200 Unified Computing Servers. JITC analysis determined that all the server hardware platforms listed at the following URL, "[www.cisco.com/go/swonly](http://www.cisco.com/go/swonly)" under the link "Cisco Unified Communications on the Cisco Unified Computing System" are functional identically to the SUT are also certified for joint use, with the exception of the MCS servers. Although there are multiple MCS servers listed on the website, only the following MCS servers are supported: MCS 7825 models H3, H4, I3, I4, and the MCS 7835 & 7845 models H2, I2, I3. The SUT supports two types of user End Instrument (EI) connection types; IP phones (hardphones, softphones, and wireless) and analog devices connected to an Integrated Services Router (ISR) or to the Voice Gateway (VG) 224. The Cisco 2800/3800 series and 2900/3900 series ISRs are all also used as Voice over IP (VoIP) gateways with traditional Time Division Multiplexing (TDM) circuits such as Digital Transmission Link Level 1 (T1)/European Basic Multiplex Rate (E1) trunks. The Gateway routers provide connectivity from VoIP networks to traditional Private Branch Exchange (PBX) products.

The SUT is a distributed solution that connects to an Assured Services Local Area Network (ASLAN). It is a highly redundant solution consisting of multiple servers, voice gateways, and IP phones. The Local Session Controller (LSC) tested components, including a brief description, are listed in the following paragraphs.

**UCS 5108 Chassis with B200-M2 Blade.** The B200-M2 Blade is part of the Communications Manager Cluster, providing VoIP capabilities, database replication, messaging, identification, authentication, security, and management features to the IP phones. The B200-M2 is deployed as the Primary Communications Manager that is part of the Communications Manager Cluster.

**UCS C210-M2 and C200-M2 Servers.** The UCS C-Series servers are rack mount servers that are part of the Communications Manager Cluster. Configured in the SUT as subscriber nodes, these boxes offer scalability, load balancing, and redundancy in case the primary Communications Manager fails.

**SM-SRE-910-K9 Module.** The SM-SRE-910-K9 module is installed in an ISR-G2. The SM-SRE-910-K9 is a branch-office infrastructure platform that combines computing, networking, storage access, virtualization, and unified management into a cohesive system that is part of the Communications Manager Cluster. Configured in the SUT as subscriber nodes, these boxes offer scalability, load balancing, and redundancy in case the primary Communications Manager fails. It supports up to 300 users and 400 phones.

**Cisco 2811/2821/2851, 3825/3845, 2911/2921/2951, and 3925/3925E/3945/3945E ISRs.** These voice gateways provide connectivity for traditional TDM components to the VoIP network. They host both analog interfaces and T1/E1 trunk circuits. In addition, the SUT with Survivable Remote Site Telephony (SRST) feature configured will function as expected during normal operations but during a failover provides ROUTINE only intra-enclave dialing for IP and analog users registered to the ISR with access to the Public Switched Telephone Network (PSTN) for ROUTINE and emergency 911 calls. The SRST functionality is certified for use with the certified ISRs listed in Table 2-4.

**Cisco Inter-Working Gateway (IWG).** The Cisco IWG is an integrated Cisco Internetwork Operating System (IOS®) software application that runs on the Cisco 3800/3900/3900E Series ISRs. The IWG is an intelligent unified communications network border element. The IWG, in addition to other Cisco IOS® software features, includes session border controller (SBC) functions that help enable end-to-end IP-based transport of voice, video, and data between independent unified communications networks. IWG deployment as part of the LSC provides critical SIP message normalization functionality that allows for interoperability between components within the Unified Capabilities (UC) architecture.

**VG224 – Analog Voice Gateway.** The VG224 is a Voice Gateway that services only analog end devices (analog phone or fax). The VG224 does not support analog Department of Defense Secure Communications Devices (DSCDs). It uses a Registered Jack (RJ)-21 cable with Amphenol 50-pin connector to attach to a breakout panel for up to 24 individual RJ-11 connections to end devices such as analog phone and fax.

**Cisco CP-79XX IP phones.** The CP-7906G, CP-7911G, CP-7925G, CP-7931G, CP-7940G, CP-7941G, CP-7941G-GE, CP-7942G, CP-7945G, CP-7960G, CP-7961G, CP-7961G-GE, CP-7962G, CP-7965, CP-7970G, CP-7971G, and CP-7975G are programmed from the Cisco Communications Manager Cluster, and the features are

selected for the individual user from the Communications Manager Application. The CP-7925G is a wireless IP Phone.

**7914, 7915, 7916 Expansion Modules.** The 7914, 7915, 7916 are expansion modules for the CP-7962, CP-7965, CP-7975 series IP phones, maximum of two per phone.

**Cisco CP-69XX IP phones.** The CP-6901, CP-6911, CP-6921, CP-6941, CP-6945, and CP-6961 are programmed from the Cisco Communications Manager Cluster, and the features are selected for the individual user from the Communications Manager Application.

**Cisco IP Communicator.** The Cisco IP Communicator is a Microsoft Windows-based application that endows computers with the functionality of a Cisco CP-7975 IP Phone. This softphone is configured and managed the same as a Cisco CP-7975 IP Phone.

**CIS Secure DTD-7965-TSGB** is a Committee on National Security Systems (CNSS)-approved (Telecommunications Security Group [TSG]6) Cisco CP-7965G Unified IP Phone. It is approved for use in Sensitive Compartmented Information Facility (SCIF) and Special Access Program Facility (SAPF) environments. It also allows the on-hook security features to be engaged while in a call providing enhanced Hold and Mute security.

**CIS Secure DTD-7962-TSG-01** is a CNSS-approved (TSG6) CP-7962G Unified IP Phone. It is approved for use in SCIF and SAPF environments. It also allows the on-hook security features to be engaged while in a call providing enhanced Hold and Mute security.

**CIS Secure DTD-7962-T2** is a TEMPEST-certified and CNSS-approved (TSG6) CP-7962G Unified IP Phone. It is a full-featured, Internet Protocol version 6 (IPv6), Skinny Call Control Protocol (SCCP), and Session Initiation Protocol (SIP)-capable IP phone. It also includes CNSS-approved (TSG6) security features for Positive Disconnect On-Hook Security.

**Cryptek CT915-VIP1-0003** is a TEMPEST version of the CP-7961G with 100 Megabits per second (Mbps) fiber Local Area Network (LAN) interface and shared access.

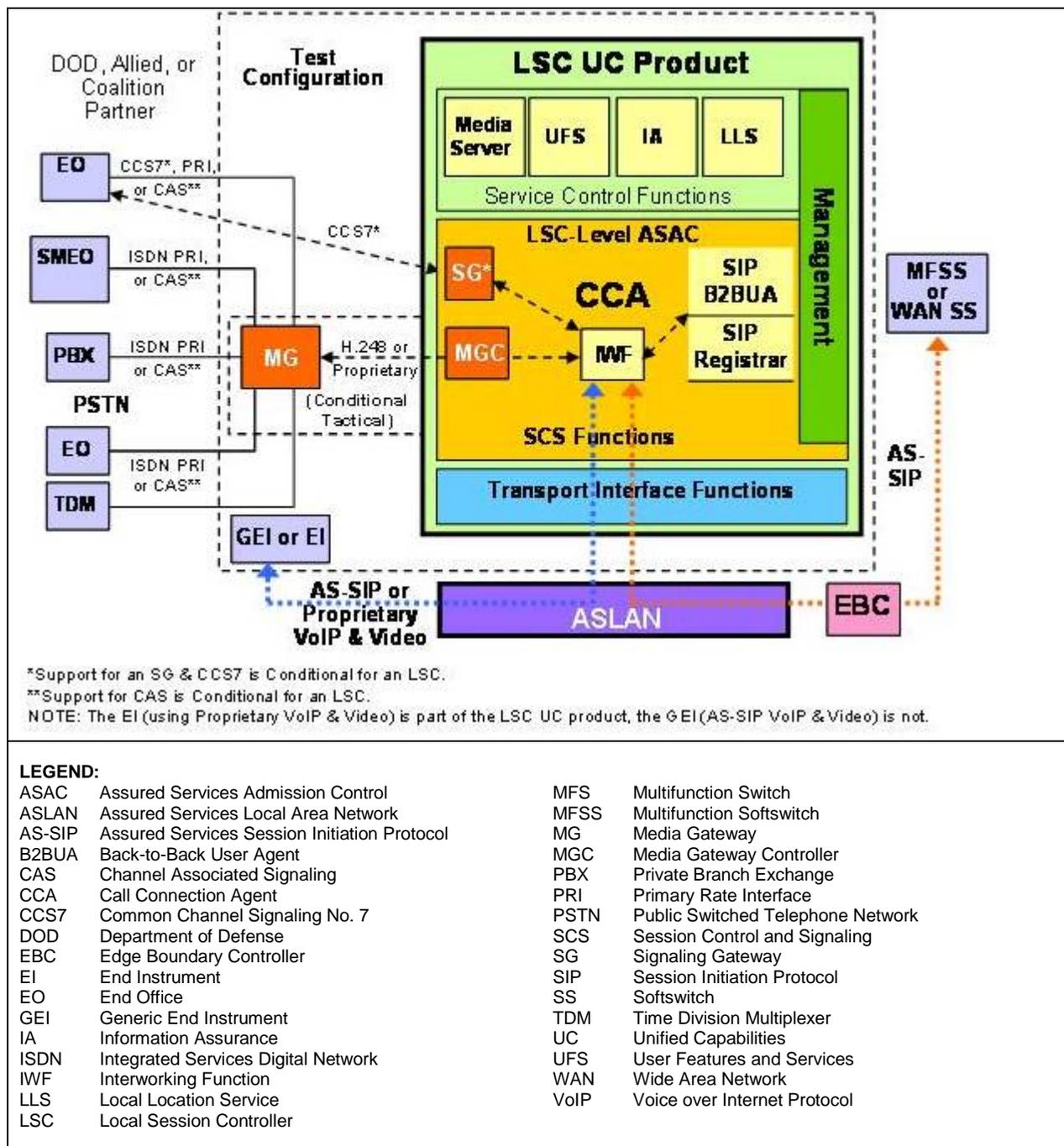
**Telecore 2151** is a TEMPEST certified, National Telecommunications Security Working Group (NTSWG, formerly known as TSG)-qualified IP Phone with dual 1000Mbps Small Form Factor Pluggable (SFP) fiber and 10/100/1000Mbps Shielded Twisted Pair (STP) Network connections, as well as 10/100/1000MB STP PC connections for shared access. The phone is capable of voice and voice communications across two IP networks from a single handset.

**General Dynamics vIPer** is a certified IP Phone manufactured by General Dynamics and tested for compatibility with the CUCM.

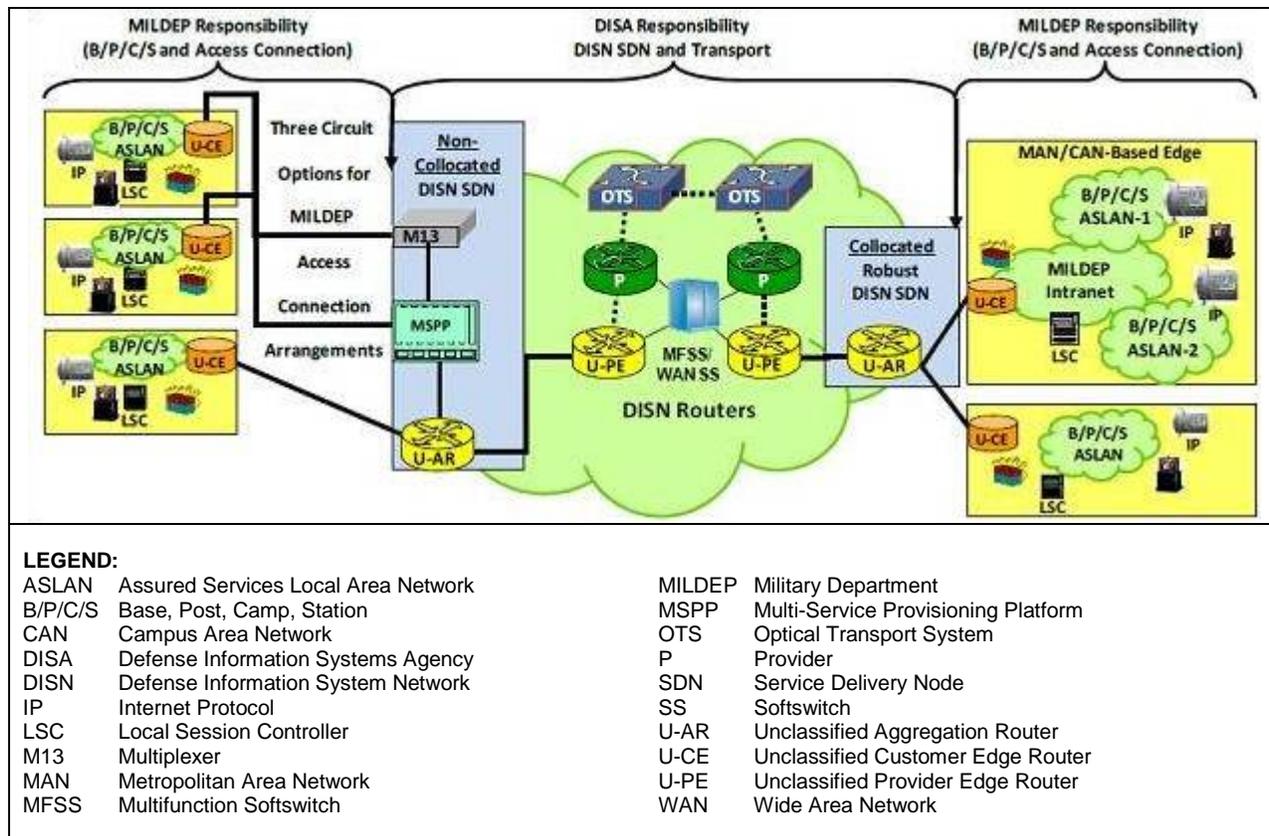
**Management Workstations.** The CUCM is managed with a site-provided, Security Technical Implementation Guide (STIG)-compliant workstation running Windows XP Service Pack (SP)3, Windows Vista SP 2, or Windows 7 SP1. The web interface allows the System Administrator (SA) to maintain the CUCM application, add and delete users, configure the system and gateway parameters, configure EIs, configure the dial plan, and to review call details. The monitoring utility, Real-Time Monitoring Tool (RTMT), must be installed on the management workstation. The RTMT, which runs as a client-side application, is used for monitoring performance, device status, device discovery, Computer Telephony Interface (CTI) applications, voice messaging ports alerts, and access to the system logs.

**Cisco Adaptive Security Appliance (ASA) 55xx series.** A Cisco ASA 5510, 5505, 5520, 5540, 5550, 5585-SSP10, 5585-SSP20, 5585-SSP30, 5585-SSP40 component is included in the solution to provide CAC support via VPN access.

**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 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 (GIG) network and other UC 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. LSC Interface Requirements**

Interface	Critical	UCR Reference	Criteria <sup>1</sup>
<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
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.

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

Interface	Critical	UCR Reference	Criteria <sup>1</sup>
<b>Line Interfaces (continued)</b>			
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.
BRI	No	5.3.2.6.1.8	Support minimum threshold CRs/FRs (2, 4, 10, and 13) and meet interface criteria for BRI.
<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 ANSI T1.619a	Yes <sup>3</sup>	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).
ISDN T1 PRI NI2	Yes <sup>4</sup>	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 (NI2).
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	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 with MLPP.
E1 PRI ITU-T Q.955.3	No <sup>5</sup>	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.955.3).
E1 PRI ITU-T Q.931	No <sup>5</sup>	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).
<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.
<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 2-2. These high-level CR/FR requirements refer to a detailed list of requirements provided in Enclosure 3.			
2. The SUT must provide a minimum of one of the listed interfaces.			
3. Provides legacy DSN and TELEPORT connectivity.			
4. Provides PSTN connectivity.			
5. Conditionally required for connectivity in Europe (ITU-T Q.955.3 DSN, ITU-T Q.931 commercial)			
<b>LEGEND:</b>			
ANSI	American National Standards Institute	Mbps	Megabits per second
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
CAS	Channel Associated Signaling	NI2	National ISDN Standard 2
CCS7	Common Channel Signaling 7	NM	Network Management
CR	Capability Requirement	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DSS1	Digital Subscriber Signaling 1	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.955.3	ISDN Signaling standard for E1 MLPP
FR	Functional Requirement	SS7	Signaling System 7
ID	Identification	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ISDN	Integrated Services Digital Network	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
ITU-T	International Telecommunication Union – Telecommunication Standardization Sector	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
LSC	Local Session Controller	UCR	Unified Capabilities Requirements

**7.2 Capability Requirements (CR) and Functional Requirements (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 Capability Requirements and Functional Requirements**

CR/FR ID	Capability/ Function	Applicability (See note 1.)	UCR Reference
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	5.3.2.2.2.3
	Signaling Protocols	Required	5.3.2.2.3
	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
3	<b>Product Physical, Quality, and Environmental Factors</b>		
	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
4	<b>Voice End Instruments</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 End Instruments</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	

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

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference
7	<b>Call Connection Agent Requirements</b>		
	CCA IWF Component	Required <sup>2</sup>	5.3.2.9.2.1
	CCA MGC Component	Required <sup>2</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>2</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	
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>3</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>3</sup>	5.3.2.12.7
	MG Interfaces to TDM PSTN OCONUS	Required <sup>3,4</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 ITU-T V.150.1	Required	5.3.2.12.16	
MG Preservation of Call Ringing during Failure	Required	5.3.2.12.17	

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

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference
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
14	<b>Attendant Station Features</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
15	<b>AS-SIP Requirements</b>		
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	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
15	<b>AS-SIP Requirements (continued)</b>		
	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
16	<b>IPv6 Requirements</b>		
	Product Requirements	Required	5.3.5.4
17	<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 Capability Requirements and Functional Requirements  
(continued)**

<b>NOTES:</b>			
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3, which includes the detailed requirements and associated criteria.			
2. The LSC must meet T1 PRI (ANSI T1.619a and NI-2) CCA IWF. The T1 CAS and T1 CCS7 CCA IWF requirements are conditional.			
3. 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.			
4. The E1 requirements for OCONUS are conditionally required for deployments in Europe.			
<b>LEGEND:</b>			
AEI	AS-SIP End Instrument	Mbps	Megabits per second
ANSI	American National Standards Institute	MG	Media Gateway
AS	Assured Services	MGC	Media Gateway Controller
ASAC	Assured Services Admission Control	MLPP	Multi-Level Precedence and Preemption
AS-SIP	Assured Services Session Initiation Protocol	NI-2	National ISDN Standard 2
BRI	Basic Rate Interface	NM	Network Management
CAS	Channel Associated Signaling	NMS	Network Management System
CCA	Call Connection Agent	OCONUS	Outside the Continental United States
CR	Capability Requirement	PBAS	Precedence Based Assured Services
CCS7	Common Channel Signaling 7	PEI	Proprietary End Instrument
DSCP	Differentiated Services Code Point	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate (2.048 Mbps)	SG	Signaling Gateway
EBC	Edge Boundary Controller	SIP	Session Initiation Protocol
EI	End Instrument	SS7	Signaling System 7
FCAPS	Fault, Configuration, Accounting, Performance and Security	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FR	Functional Requirement	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IA	Information Assurance	TDM	Time Division Multiplexing
ID	Identification	UCR	Unified Capabilities Requirements
ISDN	Integrated Services Digital Network	UFS	User Features and Services
IP	Internet Protocol	US	United States
IPv6	Internet Protocol version 6	V.150	Modem over Internet Protocol Networks
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	VoIP	Voice over Internet Protocol
IWF	Interworking Function	VVoIP	Voice and Video over Internet Protocol
LSC	Local Session Controller	WAN	Wide Area Network
		WWNDP	Worldwide Numbering and Dialing Plan

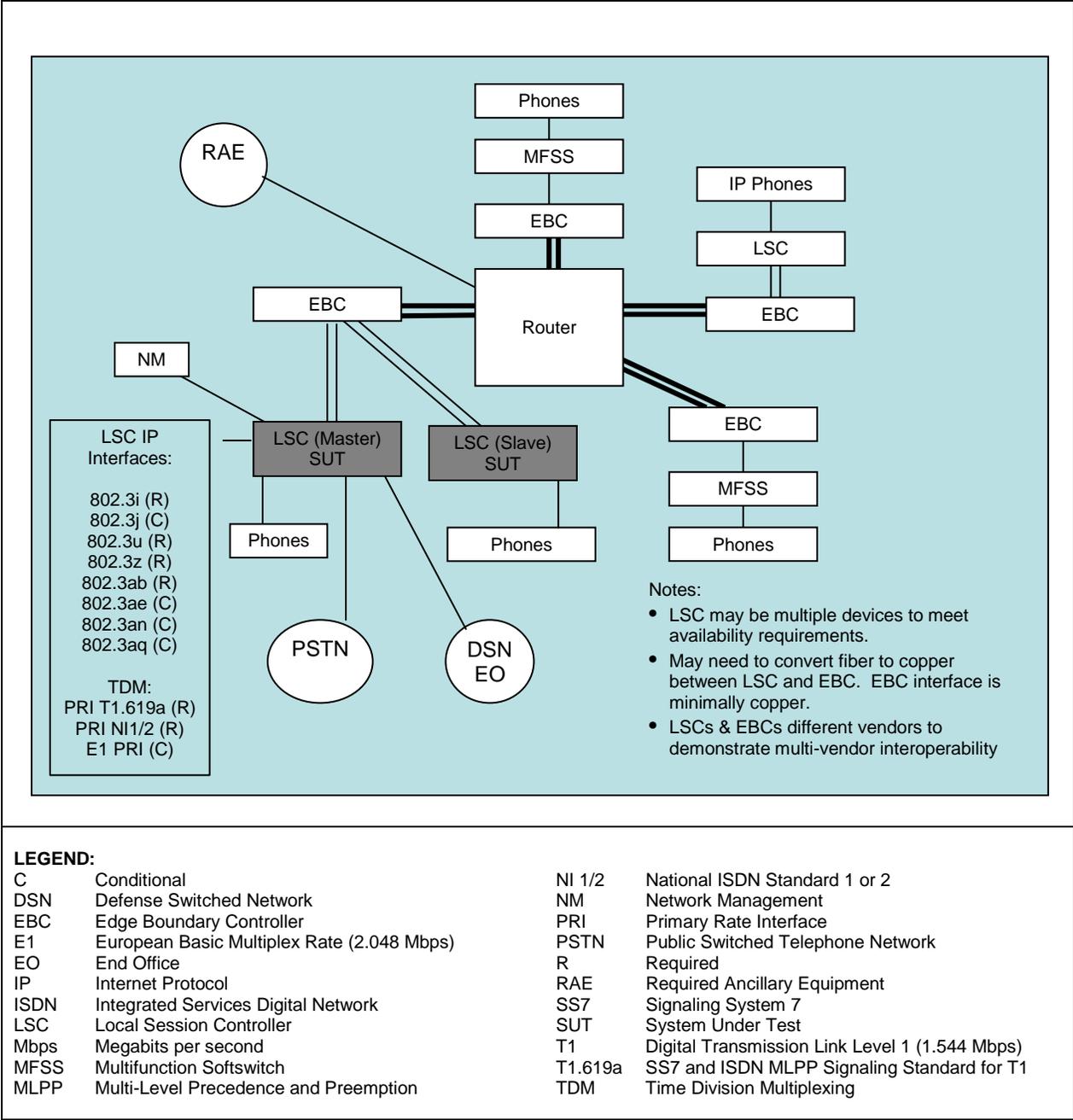
**7.3 Information Assurance.** Table 2-3 details the Information Assurance (IA) requirements applicable to an LSC.

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

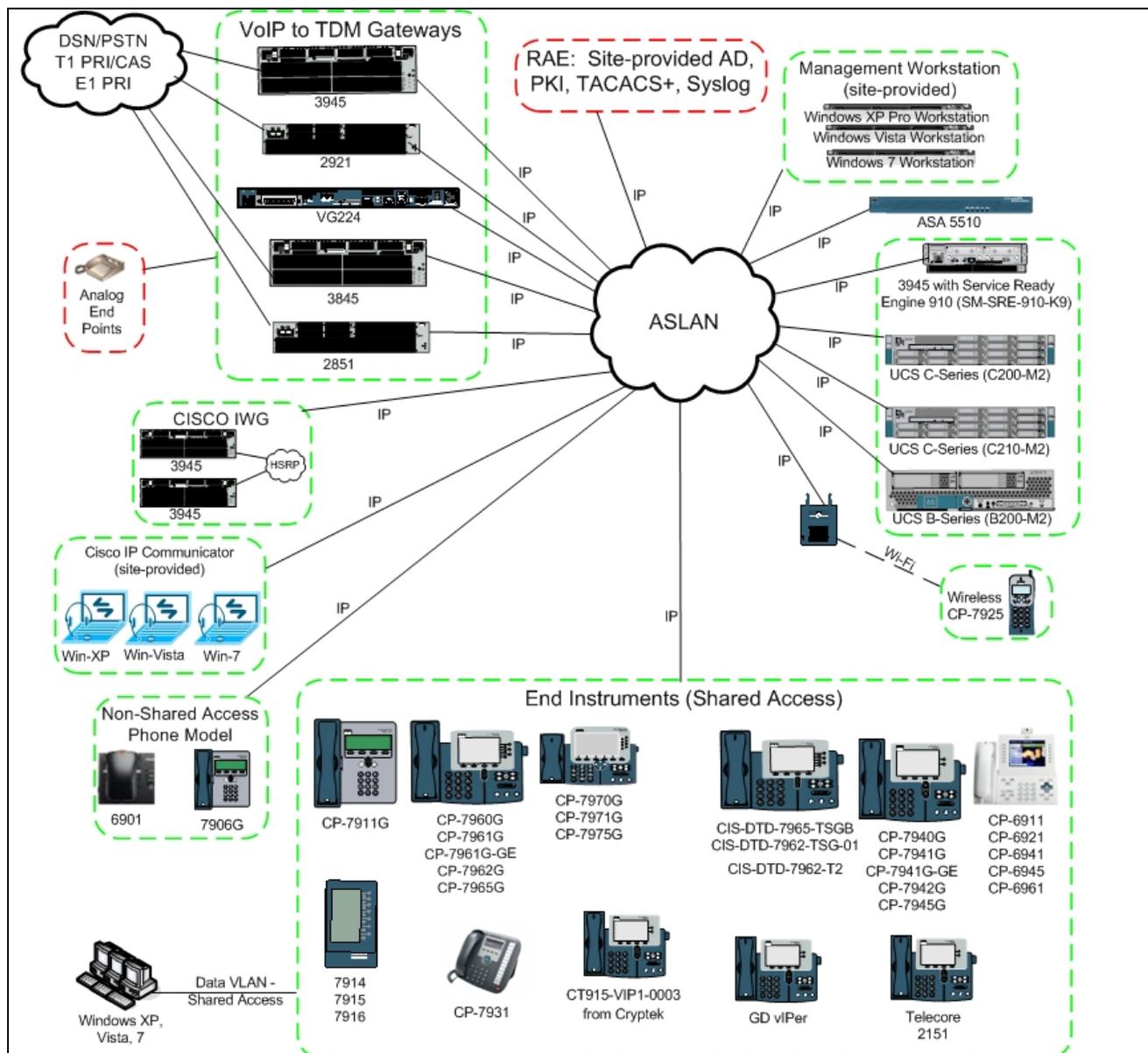
Requirement	Applicability (See note.)	UCR Reference	Criteria								
General Requirements	Required	5.4.6.2	Detailed requirements and associated criteria for LSC are listed in the IATP, Reference (e).								
Authentication	Required	5.4.6.2.1									
Integrity	Required	5.4.6.2.2									
Confidentiality	Required	5.4.6.2.3									
Non-Repudiation	Required	5.4.6.2.4									
Availability	Required	5.4.6.2.5									
<p><b>NOTE:</b> The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Reference (e), which includes the detailed requirements and associated criteria.</p> <p><b>LEGEND:</b></p> <table> <tr> <td>IA</td> <td>Information Assurance</td> <td>LSC</td> <td>Local Session Controller</td> </tr> <tr> <td>IATP</td> <td>IA Test Plan</td> <td>UCR</td> <td>Unified Capabilities Requirements</td> </tr> </table>				IA	Information Assurance	LSC	Local Session Controller	IATP	IA Test Plan	UCR	Unified Capabilities Requirements
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, Fort Huachuca, Arizona in a manner and configuration similar to that of a notional operational environment. Testing the system's required functions and features was conducted using the test configurations depicted in Figures 2-3 and 2-4. Figure 2-3 depicts the minimum test architecture for testing LSCs. Figure 2-4 depicts the SUT's test configuration.



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



**LEGEND:**

AD	Active Directory	PKI	Public Key Infrastructure
ASA	Adaptive Security Appliance	PRI	Primary Rate Interface
ASLAN	Assured Services Local Area Network	PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling	SUT	System Under Test
DSN	Defense Switched Network	T1	Digital Transmission Link Level 1(1.544 Mbps)
E1	European Basic Multiplex Rate (2.048 Mbps)	TACACS+	Terminal Access Controller Access Control System Plus
GE	Gigabit Ethernet	TDM	Time Division Multiplexing
GD	General Dynamics	UCS	Unified Computing System
HSRP	Hot Standby Router Protocol	VG	Voice Gateway
IP	Internet Protocol	VoIP	Voice over Internet Protocol
IWG	Inter-Working Gateway		
Mbps	Megabits per second		

**Figure 2-4. 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. SUT Tested System Configurations**

System Name		Software Release	
Acme Packet Net-Net 4500 SD EBC		6.2.0	
Acme Packet Net-Net 3820 SD EBC		6.2.0	
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		Communication Manager (CM) 6.0.1 (00.1.510.1) Service Pack 19211	
Avaya S8720 SMEO		Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)	
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 HDX		V4.0A (R3P7)	
Required Ancillary Equipment		Active Directory	
		Public Key Infrastructure	
		SysLog Server	
		Terminal Access Controller Access Control System Plus	
<b>Cisco Unified Communications Manager Version 8.6.1, with IOS Software Release 15.1.4M2</b>			
Component <sup>1</sup>	Release	Sub-component <sup>1</sup>	Function
<b>3945</b> , 2911, 2921, 2951, 3925 (Generation 2 Integrated Services Router)	IOS 15.1.4M2	<b>SM-SRE-910-K9</b>	Processing/Signaling
<b>UCS C210-M2 (with VMware)</b> , UCS-C210-M1, and <b>UCS-C200-M2</b>	8.6.1 (20003-2)	Not Applicable	Processing/Signaling
<b>UCS Server UCS5108 with B200-M2</b> and B200-M1 (with VMware)	8.6.1 (20003-2)	Not Applicable	Processing/Signaling
<b>ASA 5510</b> , 5505, 5520, 5540, 5550, 5585-SSP10, 5585-SSP20, 5585-SSP30, 5585-SSP40	ASA 8.4.3	Not Applicable	Provide CAC support via VPN access

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

<b>Cisco Unified Communications Manager Version 8.6.1, with IOS Software Release 15.1.4M2</b>			
<b>Component<sup>1</sup></b>	<b>Release</b>	<b>Sub-component<sup>1</sup></b>	<b>Function</b>
Cisco <b>3845</b> , 3825 Cisco 2811, 2821, <b>2851</b> (Generation 1 Integrated Service Router) (Gateway)	IOS 15.1.4M2	<b><u>NM HDV2</u></b>	IP Communications High-Density Digital Voice Network Module
		NM-HDA-4FXS	High density analog voice/fax network module with 4 FXS.
		NM-HD-2VE	2 Slot IP Communications Enhanced Voice/Fax Network Module
		NM-HD-1V	1-slot IP Communications Voice/Fax Network Module
		<b><u>VIC2 2FXS</u></b>	Two-Port Voice Interface Card - FXS and DID
		NM-HD-2V	2-slot IP communications enhanced voice/fax NM
		<b><u>VVIC2 2MFT T1/E1</u></b>	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		<b><u>VVIC2 1MFT T1/E1</u></b>	First Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EM3-HDA-8FXS/DID	HD analog and digital extension module for voice and fax
		<b><u>EVM HD 8FXS/DID</u></b>	HD analog and digital extension module for voice and fax
		<b><u>EM HDA 8FXS</u></b>	8-port analog FXS expansion module for voice and fax (See note 3.)
		<b><u>NM HDV2 2T1/E1</u></b>	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		<b><u>NM HDV2 1T1/E1</u></b>	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		<b><u>VIC 4FXS/DID</u></b>	Four-Port Voice Interface Card - FXS and DID
		<b><u>VIC3 4FXS/DID</u></b>	Four-Port Voice Interface Card - FXS and DID
		<b><u>VIC3 2FXS-E/DID</u></b>	Two-Port Voice Interface Card - FXS and enhanced DID
<b><u>VIC3 2FXS/DID</u></b>	Two-Port Voice Interface Card - FXS and DID		
PVDMII	Voice/Fax DSP Module		

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

<b>Cisco Unified Communications Manager Version 8.6.1, with IOS Software Release 15.1.4M2</b>			
<b>Component<sup>1</sup></b>	<b>Release</b>	<b>Sub-component<sup>1</sup></b>	<b>Function</b>
Cisco 2901, 2911, <b>2921</b> , 2951 Cisco 3925, 3925E, <b>3945</b> , 3945E (Generation 2 Integrated Service Router) (Gateway)	IOS 15.1.4M2	<b><u>SM-NM-ADAPTER</u></b>	SM (Service Module) to NM Adapter for 2 <sup>nd</sup> Generation ISRs (2951/3945) only
		NM-HD-1V	1-Slot IP Communications Voice/FAX NM
		<b><u>NM-HD-2V</u></b>	2-Slot IP Communications Voice/FAX NM
		<b><u>NM-HDV2</u></b>	IP Communications High-Density Digital Voice/FAX NM, with no onboard T1/E1 controllers and one VIC/VWIC slot
		<b><u>NM-HD-2VE</u></b>	2-Slot IP Communications Enhanced Voice/FAX NM
		NM-HDV2-1T1/E1	IP Communications High-Density Digital Voice NM with 1 T1/E1
		<b><u>NM-HDV2-2T1/E1</u></b>	IP Communications High-Density Digital Voice NM with 2 T1/E1
		<b><u>VIC3-2FXS/DID</u></b>	VIC, 2-port FXS
		<b><u>VIC3-2FXS-E/DID</u></b>	VIC, 2-port Enhanced FXS
		<b><u>VIC3-4FXS/DID</u></b>	VIC, 4-port FXS
		<b><u>VWIC2-1MFT-T1/E1</u></b>	2 <sup>nd</sup> Generation Voice/WAN Interface card 1-port RJ-48, Multiflex Trunk - T1/E1
		<b><u>VWIC2-2MFT-T1/E1</u></b>	2 <sup>nd</sup> Generation Voice/WAN Interface card 2-port RJ-48, Multiflex Trunk - T1/E1
		<b><u>VWIC3-1MFT-T1/E1</u></b>	3 <sup>rd</sup> Generation Voice/WAN Interface card 1-port RJ-48, Multiflex Trunk - T1/E1
		<b><u>VWIC3-2MFT-T1/E1</u></b>	3 <sup>rd</sup> Generation Voice/WAN Interface card 2-port RJ-48, Multiflex Trunk - T1/E1
		<b><u>VWIC3-4MFT-T1/E1</u></b>	3 <sup>rd</sup> Generation Voice/WAN Interface card 4-port RJ-48, Multiflex Trunk - T1/E1
		<b><u>EVM-HD-8FXS/DID</u></b>	High density voice/fax extension module - 8FXS/DID
		<b><u>EM3-HDA-8FXS</u></b>	8-port voice/fax expansion module - FXS
		PVDM2-ADPTR	PVDM2 Adapter
		<b><u>PVDM2</u></b> -8, 16, 32, 48, 64	Voice/Fax DSP Modules
<b><u>PVDM3</u></b> -16, 32, 64, 128, 192, <b>256</b>	Voice/Fax DSP Modules		

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

<b>Cisco Unified Communications Manager Version 8.6.1, with IOS Software Release 15.1.4M2</b>		
<b>Component<sup>1</sup></b>	<b>Release</b>	<b>Function</b>
<b><u>VG224</u></b>	IOS 15.1.4M2	Voice Gateway (VG) for support of up to 24 non-secure voice and facsimile analog endpoints only.
Cisco Inter-Working Gateway 3825/ <b><u>3845</u></b> , 3925/3925E/ <b><u>3945</u></b> /3945E	IOS 15.1.4M4	Session/Edge Border Controller
<b><u>CP-6901</u></b>	SCCP 9.2.1a	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, No Shared Access
CP-6911	SCCP 9.2.1a	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, Shared Access
CP-6921, CP-6941, CP- <b><u>6945</u></b> , CP- <b><u>6961</u></b>	SCCP 9.2.1s	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, Shared Access
<b><u>CP-7911G</u></b> , CP-7931G, CP-7941G, CP-7942G, CP-7961G, CP- <b><u>7962G</u></b>	SCCP 9.2.1s	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, Shared Access
CP-7906G	SCCP 9.2.1s	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, No Shared Access
<b><u>CP-7925G</u></b>	CP7925G-1.4.1	Wireless IP Phone, with max transmission of 802.11a/b/g of 54Mbps. No Shared Access.
CP-7940G, CP-7960G (See note 4.)	P00308010200	IP Phone (with push-to-talk handset or with standard handset), 10/100MB Ethernet, Shared Access
CP-7941G-GE, CP-7945G, CP- 7961G-GE, CP-7965G, CP- <b><u>7970G</u></b> , CP-7971G, CP-7975G	SCCP 9.2.1s	IP Phone (with push-to-talk handset or with standard handset), 10/100/1000MB Ethernet, Shared Access
7914	S00105000400	IP phone 14 line expansion for the 7962, 7965, and 7975 series. Maximum 2 per phone.
<b><u>7915</u></b>	B015-1-0-4-2	IP phone 12-24 line expansion for the 7962, 7965, and 7975 series. Maximum 2 per phone.
7916	B016-1-0-4-2	IP phone 12-24 line expansion for the 7962, 7965, and 7975 series. Maximum 2 per phone.
<b><u>Cisco IP Communicator</u></b>	8.6.1.0	Softphone. This Microsoft Windows-based applications emulates a Cisco CP-7975 IP Phone. Tested on a PC with Windows XP, Vista, and Windows 7; Cisco IPC 8.6; Cisco Unified Video Advantage 2.1(2); Tumbleweed 4.9.2.196; ActivClient 6.2.0.50
CIS Secure DTD-7965-TSGB	SCCP 9.2.1	CNSS (TSG6) Approved Cisco 7965G Unified IP Phone. Approved for use in SCIF and SAPF environments, also allows the on-hook security features to be engaged while in a call providing enhanced Hold and Mute security.
CIS Secure DTD-7962-TSG-01	SCCP 9.2.1	CNSS (TSG6) Approved Cisco 7962G Unified IP Phone. Approved for use in SCIF and SAPF environments.
CIS Secure DTD-7962-T2	SCCP 9.2.1	TEMPEST Certified and CNSS (TSG6) Approved Cisco 7962G Unified IP Phone. It is a full-featured, IPv6, SCCP and SIP capable IP phone. It also includes CNSS approved "TSG6" security features for Positive Disconnect On-Hook Security.
Cryptek CT915-VIP1-0003 (See note 5.)	SCCP 9.2.1	7961G TEMPEST Fiber version with 100MB Fiber LAN and shared access
General Dynamics vPer	P00308010200	Certified by General Dynamics and tested for compatibility.
<b>DoD Secure Communication Devices<sup>6</sup></b>		
General Dynamics Sectera vPer (PSTN)		2.14
General Dynamics Sectera vPer (IP)		1.0 Version 6.04
L-3 Communications STE		2.7
General Dynamics Sectera Wireline Terminal		12.05
L-3 Communications IP STE		1.2.5
L-3 Communications OMNI		6.01
<b><u>Telecore 2151</u></b>		V.2AE-00056-0200 FPGAV.2AE00020-0200

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

<b>NOTES:</b>					
1. Components <b>bolded and underlined&gt;</b> were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.					
2. These components are certified in the DISN with T1 ISDN PRI and T1 CAS (DTMF) interfaces. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.					
3. The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.					
4. The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (c).					
5. CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.					
6. DCSDs are certified for joint use with all gateways listed in Table 2-4 except for the VG-224.					
<b>LEGEND:</b>					
APL	Approved Product List	GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)	RJ	Registered Jack
AS-SIP	Assured Services Session Initiation Protocol	HDA	High Density Analog	SAPF	Special Access Program Facility
BIOS	Basic Input Output System	IOS	Internetwork Operating System	SC	fiber connector (square push-in)
CNSS	Committee on National Security Systems	IP	Internet Protocol	SCCP	Skinny Call Control Protocol
CP	Cisco Phone	IPv4	Internet Protocol version 4	SCIF	Sensitive Compartmented Information Facility
DID	Direct Inward Dialing	IPv6	Internet Protocol version 6	SD	Session Director
DISA	Defense Information Systems Agency	ISDN	Integrated Services Digital Network	SUT	System Under Test
DISN	Defense Information System Network	JITC	Joint Interoperability Test Command	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DSN	Defense Switched Network	LAN	Local Area Network	TDM	Time Division Multiplexing
E1	European Basic Multiplex Rate (2.048 Mbps)	Mbps	Megabits per second	UC	Unified Capabilities
EBC	Edge Boundary Controller	MCS	Media Convergence Server	UCR	Unified Capabilities Requirements
EM	Expansion Module	MFT	Multiflex Trunk	UCS	Unified Computing System
EVM	Extension Voice Module	MOS	Mean Opinion Score	V	Voice
Fax	facsimile	NM	Network Module	VE	Voice/Fax Enhanced
FXS	Foreign Exchange Station	PC	Personal Computer	VIC	Voice Interface Card
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)	PRI	Primary Rate Interface	VVIC	Voice WAN Interface Card
		PSTN	Public Switched Telephone Network	WAN	Wide Area Network

**10. TESTING LIMITATIONS.** JITC test teams 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. Call Loading.** JITC could not create a large volume of line calls because the line signaling protocol used by the SUT is proprietary. Also, JITC could not generate a large volume of Assured Services Session Initiation Protocol (AS-SIP) trunk calls because of limited lines which were provided during testing and lack of AS-SIP test equipment. These limitations pose a low risk to interoperability and should not impact overall results and conclusions. The use of operational data as the LSC is fielded will validate the SUT's ability to support its proposed number of subscribers (up to 30,000).

**b. Proprietary End Instruments.** JITC did not test PEIs for video requirements. Since the Defense Switched Network (DSN) has not deployed videophones under

legacy certifications, this poses a low operational impact. JITC will verify video capabilities of the SUT prior to amending the certification to include the capability.

**c. Internet Protocol version 6.** The IPv6 requirements were tested in the LSC configuration. JITC did not test IPv6 inter-enclave (i.e., between LSCs via an Edge Boundary Controller (EBC) because the EBC did not fully support IPv6 during the time of testing. JITC will verify inter-enclave IPv6 capabilities of the SUT prior to amending the certification to include the capability.

**d. Network Management (NM).** JITC did not test the SUT's ability to meet UCR NM requirements. The vendor did submit an NM LoC that was reviewed by JITC. JITC's evaluation of the SUT's NM capabilities is provided in paragraph 11.2.q.

**e. Master/Subtended.** JITC did not test the SUT to determine its ability to meet master/subtended requirements in UCR 2008, Change 2, section 5.3.2.29. Initial fielding of an LSC will not be used in this configuration. The operational impact was adjudicated to be minor.

**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 Approved Products List (APL). Additional discussion regarding specific testing results is located in subsequent paragraphs.

**11.1 Interfaces.** The SUT met line interface requirements for 10/100/1000 Base-X interfaces. These IP line interfaces were met through use of PEIs (voice only). The SUT supports 2-wire analog phones via a gateway. The SUT met the external interface requirements for 10/100/1000Base-X (AS-SIP), T1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) for both ANSI T1.619a MLPP and National ISDN-2 (NI-2) commercial, T1 Channel Associated Signaling (CAS) Dual Tone Multi-frequency (DTMF), and E1 ISDN PRI ITU-T Q.931. 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 Requirements Status**

Interface	Critical	UCR Reference	Threshold CR/FR Requirements <sup>1</sup>	Status	Remarks
<b>Line Interfaces</b>					
10Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to PEIs and softphones.
100Base-X	Yes	5.3.2.6.3	2, 4, 10, 13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to PEIs and softphones.
1000Base-X	No	5.3.2.6.3	2, 4, 10,13, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3ab. Applies to PEIs and softphones.
2-wire analog	Yes	5.3.2.6.1.6	2, 4, 10, and 13	Certified	Met threshold CRs/FRs for 2-wire instruments. Applies to 2-wire secure and non-secure analog instruments.
BRI	No	5.3.2.6.1.8	2, 4, 10, and 13	Not Tested	This interface is offered by the SUT; however, it was not tested because it does not support Assured Services.
<b>External Interfaces</b>					
10Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3i and 802.3j. Applies to AS-SIP trunk.
100Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3u. Applies to AS-SIP trunk.
1000Base-X	No <sup>2</sup>	5.3.2.4.2	1, 2, 3, 6, 7, 8, 10, 11, 13, 15, and 16	Certified	Met threshold CRs/FRs for IEEE 802.3z and 802.3ab. Applies to AS-SIP trunk.
ISDN T1 PRI ANSI T1.619a	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs. Provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2	Yes	5.3.2.4.3	2, 3, 7, 8, 10, and 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, and 13	Not Tested	This interface is not offered by the SUT.
T1 CAS	No	5.3.2.12.11	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs for DTMF.
E1 CAS (DP, DTMF, MFR1)	No	5.3.2.12.11	2, 3, 7, 8, 10, 13	Not Tested	This interface is offered by the SUT; however, it was not tested and is not covered under this certification.
E1 PRI ITU-T Q.955.3	No <sup>3</sup>	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Not Certified	This interface is offered by the SUT; however, it was not tested and is not covered under this certification.
E1 PRI ITU-T Q.931	No <sup>3</sup>	5.3.2.12.10	2, 3, 7, 8, 10, and 13	Certified	Met threshold CRs/FRs for European PSTN connectivity.
<b>NM</b>					
10Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	16 and 17	Certified	Met threshold CRs/FRs. Verified via LoC.
100Base-X	No <sup>2</sup>	5.3.2.4.4 5.3.2.7.2.8	16 and 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. The SUT must provide a minimum of one of the listed interfaces.					
3. This interface is conditionally required for deployment in Europe.					

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

<b>LEGEND:</b>			
10Base-X	10 Mbps Ethernet	IEEE	Institute of Electrical and Electronics Engineers
100Base-X	100 Mbps Ethernet	ISDN	Integrated Services Digital Network
1000Base-X	1000 Mbps Ethernet	ITU-T	International Telecommunication Union – Telecommunication Standardization Sector
802.3ab	1000 Mbps Ethernet over Twisted Pair		
802.3i	10 Mbps twisted pair media for 10Base-X networks	LoC	Letter of Compliance
802.3j	10 Mbps fiber media for 10Base-X networks	Mbps	Megabits per second
802.3u	100BASE-TX, 100BASE-T4, 100BASE-FX Fast Ethernet at 100 Mbps with auto negotiation	MFR1	Multi-Frequency Recommendation 1
802.3z	Standard for Gigabit Ethernet	MG	Media Gateway
ANSI	American National Standards Institute	MLPP	Multi-Level Precedence and Preemption
AS-SIP	Assured Services Session Initiation Protocol	NI-2	National ISDN Standard 2
BRI	Basic Rate Interface	NM	Network Management
CAS	Channel Associated Signaling	PEI	Proprietary End Instrument
CCS7	Common Channel Signaling 7	PRI	Primary Rate Interface
CR	Capability Requirement	PSTN	Public Switched Telephone Network
DP	Dial Pulse	Q.931	Signaling Standard for ISDN
DSN	Defense Switched Network	Q.955.3	ISDN Signaling Standard for E1 MLPP
DTMF	Dual Tone Multi-Frequency	SS7	Signaling System 7
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
		UCR	Unified Capabilities Requirements

**11.2 Capability Requirements (CR) and Functional Requirements (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 to be minor based on vendor submission and compliance to a Plan of Actions and Milestones (POA&M).

**Table 2-6. SUT Capability Requirements and Functional Requirements Status**

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

**Table 2-6. SUT Capability Requirements and Functional Requirements Status  
(continued)**

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
4	<b>Voice End Instruments</b>			
	Tones and Announcements	Required	5.3.2.6.1.1	Met <sup>4</sup>
	Audio Codecs	Required	5.3.2.6.1.2	Met <sup>4,5</sup>
	VoIP PEI or AEI Audio Performance	Required	5.3.2.6.1.3	Partially Met <sup>4</sup>
	VoIP Sampling Standard	Required	5.3.2.6.1.4	Met <sup>4</sup>
	Authentication to LSC	Required	5.3.2.6.1.5	Met <sup>4</sup>
	Analog Telephone Support	Required	5.3.2.6.1.6	Met
	Softphones	Conditional	5.3.2.6.1.7	Partially Met <sup>6</sup>
	ISDN BRI	Conditional	5.3.2.6.1.8	Not Tested
5	<b>Video End Instruments</b>			
	Video End Instrument	Required	5.3.2.6.2	Not Met <sup>7</sup>
	Display Messages, Tones, and Announcements	Required	5.3.2.6.2.1	Not Met <sup>7</sup>
	Video Codecs (Including Associated Audio Codecs)	Required	5.3.2.6.2.2	Not Met <sup>7</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>4,8</sup>
	Line-Side Custom Features Interference	Conditional	5.3.2.7.2.11	Met
	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	Partially Met <sup>9,10</sup>
	CCA MGC Component	Required	5.3.2.9.2.2	Met
	SG Component	Conditional	5.3.2.9.2.3	Not Tested <sup>9</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>9</sup>
	CCA-IWF Support for PRI via MG	Required	5.3.2.9.5.3	Partially Met <sup>10</sup>
	CCA-IWF Support for CAS Trunks via MG	Conditional	5.3.2.9.5.4	Not Tested <sup>9</sup>
	CCA-IWF Support for PEI and AEI Signaling Protocols	Required	5.3.2.9.5.5	Partially Met <sup>11</sup>
	CCA-IWF Support for VoIP and TDM Protocol Interworking	Required	5.3.2.9.5.6	Met <sup>9</sup>
	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 <sup>12</sup>
	CCA Interaction with EIs	Required	5.3.2.10.10	Partially Met <sup>4</sup>
CCA Support for AS Voice and Video	Required	5.3.2.10.11	Partially Met <sup>7</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>9</sup>

**Table 2-6. SUT Capability Requirements and Functional Requirements Status  
(continued)**

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
8	<b>MG Requirements</b>			
	Role of MG In LSC	Required	5.3.2.12.3.1	Met
	MG Support for ASAC	Required	5.3.2.12.4.1	Met
	MG and IA Functions	Required	5.3.2.12.4.2	Met <sup>12</sup>
	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
	MG Interaction with VoIP Els	Required	5.3.2.12.4.8	Met <sup>4</sup>
	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	Partially Met <sup>9, 10, 13</sup>
	MG Interface to TDM Allied and Coalition	Conditional	5.3.2.12.6	Not Tested
	MG Interface to TDM PSTN in US	Required	5.3.2.12.7	Partially Met <sup>10, 13</sup>
	MG Interfaces to TDM PSTN OCONUS	Required	5.3.2.12.8	Partially Met <sup>13</sup>
	MG Support for CCS7	Conditional	5.3.2.12.9	Not Tested <sup>9</sup>
	MG Support for ISDN PRI Trunks	Required	5.3.2.12.10	Partially Met <sup>10</sup>
	MG Support for CAS Trunks	Required	5.3.2.12.11	Met
	MG requirements for VoIP Internal Interfaces	Required	5.3.2.12.12	Met
	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	Not Met <sup>14</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
	SG Interactions with CCA	Conditional	5.3.2.13.5.2	Not Tested
	SG Interworking Functions	Conditional	5.3.2.13.5.3	Not Tested
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>15</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
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	Not Tested <sup>8</sup>
	Call Display	Required	5.3.2.26.2	Not Tested <sup>8</sup>
	Class of Service Override	Required	5.3.2.26.3	Not Tested <sup>8</sup>
	Busy Override and Busy Verification	Required	5.3.2.26.4	Not Tested <sup>8, 16</sup>
	Night service	Required	5.3.2.26.5	Not Tested <sup>8</sup>
	Automatic Recall of Attendant	Required	5.3.2.26.6	Not Tested <sup>8</sup>
	Calls in Queue to the Attendant	Required	5.3.2.26.7	Not Tested <sup>8, 17</sup>

**Table 2-6. SUT Capability Requirements and Functional Requirements Status (continued)**

CR/FR ID	Capability/ Function	Applicability <sup>1</sup>	UCR Reference	Status
15	<b>AS-SIP Requirements</b>			
	SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs	Required	5.3.4.7	Not Tested <sup>4</sup>
	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>7</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 <sup>18</sup>
	Relevant Timers for the Terminating Gateway and the Originating Gateway	Required	5.3.4.15	Not Tested <sup>19</sup>
	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	Met	
16	<b>IPv6 Requirements</b>			
	Product Requirements	Required	5.3.5.4	Partially Met <sup>20</sup>
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	Met
Accounting Management	Required	5.3.2.19	Partially Met <sup>21</sup>	

**NOTES:**

- The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Enclosure 3.
- The SUT does not support a "ping ring" notification. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).
- The SUT does not support OPTIONS requests required to meet the failover to a secondary SS in accordance with UCR 2008, Change 2, section 5.3.2.3.2.1. DISA adjudicated this discrepancy and determined that the UCR failover requirements are immature and require a rewrite. DISA NS2 has agreed to a Condition of Fielding that the initial UC APL certification will not provide for failover capability on the condition the vendor will participate in an NS2-scheduled multi-vendor test event to refine failover requirements, modify software to support the new failover requirements, and demonstrate failover compliance.
- The SUT only supports voice PEIs. The vendor does not support AEIs (voice or video). DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
- The SUT gateways equipped with the PVDM3 modules do not support the ITU-T G.723 codec. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).
- The SUT softphone with Microsoft Windows Vista and Windows 7 OSs does not allow DSCP tagging per precedence level in accordance with UCR 2008, Change 2, Section 5.3.3.3.2. Microsoft Windows XP is the only OS that supports the five precedence levels. DISA adjudicated this as minor since all voice is queued together in the four-queue model currently used in deployed ASLANs.
- The SUT did not offer a video PEI. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
- The SUT Operator Console/Attendant Station was not tested; however the vendor submitted an LoC for the requirements.
- The SUT met T1 ISDN PRI (ANSI T1.619a and ANSI T1.607), E1 PRI (ITU-T Q.931), and T1 CAS DTMF IWF requirements, which is all of the certified TDM interfaces.
- The SUT does not support NFAS on the T1 ISDN PRI interface. Although this is conditional for DSN connectivity, it is required for PSTN connectivity. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).

**Table 2-6. SUT Capability Requirements and Functional Requirements Status  
(continued)**

**NOTES (continued):**

11. The SUT met PEI CCA-IWF requirements. The SUT does not support AEIs. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
12. The security requirements are tested by a DISA-led IA test team and published in a separate report, Reference (e).
13. The SUT must meet T1 PRI (T1.619a and NI2) IWF. The T1 CAS and T1 CCS7 IWF requirements are conditional. The SUT met T1 ISDN PRI (ANSI T1.619a and ANSI T1.607), E1 PRI (ITU-T Q.931), and T1 CAS DTMF IWF requirements.
14. The SUT does not properly handle ITU-T V.150 calls with the Avaya Communication Manager 6.0 and both vendors are working on the problem. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
15. The SUT does not support Commercial Cost Avoidance. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
16. The SUT does not fully comply with Busy Override and Busy Line Verification requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
17. The SUT does not fully comply with attendant console queuing requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
18. The SUT met this requirement with ANSI T1.619a ISDN PRI NI2 DSN and ISDN PRI NI2 PSTN TDM interfaces interworking with AS-SIP. This requirement was met with both testing and the vendor's LoC. The SUT does not support CCS7 TDM interface which is conditional for an LSC.
19. This requirement applies to gateways between AS-SIP and CCS7 links. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.
20. The vendor submitted an IPv6 LoC with noted discrepancies. The SUT does not support RFCs 4861 and 4862. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.
21. The vendor submitted an NM LoC with noted discrepancies. The SUT does not comply with the requirement for the equipment impairment factor to be in accordance with ITU-T G.107. DISA adjudicated this as minor with a vendor POA&M to provide a MOS score. DISA also stated the intent to change this to conditional in the next version of the UCR (UCR 2013). The SUT does not have the ability to transfer records to a removable physical storage media. DISA adjudicated this as minor and stated the intent to delete this requirement in the next version of the UCR (UCR 2013).

<b>LEGEND:</b>					
AEI	AS-SIP End Instrument	ID	Identification	PSTN	Public Switched Telephone Network
ANSI	American National Standards Institute	ISDN	Integrated Services Digital Network	PVDM3	Packet Voice Digital Signal Processor Module 3
APL	Approved Products List	IP	Internet Protocol	Q.931	Signaling Standard for ISDN
AS	Assured Services	IPv6	Internet Protocol version 6	RFCs	Request for Comments
ASAC	Assured Services Admission Control	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	SG	Signaling Gateway
ASLAN	Assured Services Local Area Network			SIP	Session Initiation Protocol
AS-SIP	Assured Services Session Initiation Protocol	IWF	Interworking Function	SS	Softswitch
BRI	Basic Rate Interface	LoC	Letter of Compliance	SS7	Signaling System 7
CAS	Channel Associated Signaling	LSC	Local Session Controller	SUT	System Under Test
CCA	Call Connection Agent	Mbps	Megabits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
CR	Capability Requirement	MG	Media Gateway	T1.607	ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
CCS7	Common Channel Signaling 7	MGC	Media Gateway Controller		
DISA	Defense Information Systems Agency	MLPP	Multi-Level Precedence and Preemption	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DSCP	Differentiated Services Code Point	MOS	Mean Opinion Score	TDM	Time Division Multiplexing
DSN	Defense Switched Network	NFAS	Non Facility Associated Signaling	UC	Unified Capabilities
DSS1	Digital Subscriber Signaling 1	NI2	National ISDN Standard 2	UCR	Unified Capabilities Requirements
DTMF	Dual Tone Multi-Frequency	NM	Network Management	UFS	User Features and Services
E1	European Basic Multiplex Rate (2.048 Mbps)	NMS	Network Management System	US	United States
EBC	Edge Boundary Controller	OCONUS	Outside the Continental United States	V.150	Modem over Internet Protocol Networks
EI	End Instrument	OS	Operating System	VoIP	Voice over Internet Protocol
FCAPS	Fault, Configuration, Accounting, Performance and Security	PBAS	Precedence Based Assured Services	VVoIP	Voice and Video over Internet Protocol
FR	Functional Requirement	PEI	Proprietary End Instrument	WAN	Wide Area Network
IA	Information Assurance	POA&M	Plan of Action and Milestones	WWNDP	Worldwide Numbering and Dialing Plan
		PRI	Primary Rate Interface		

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

(1) Differentiated Services Code Point (DSCP) Packet Marking. The UCR 2008, Change 2, section 5.3.2.2.1.4, states that 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 Section 5.3.3.3.2. The SUT met all DSCP Packet Marking requirements for IPv4 and IPv6 (intra-enclave only).

(2) Voice Features and Capabilities. The UCR 2008, Change 2, section 5.3.2.2.2.1, states that the LSC must provide all of the features listed in Table 5.3.2.2-1 of the UCR. The SUT met all Voice Features and Capabilities requirements, with the following minor exception: The SUT does not provide “Ping-Ring” on a call forward variable enabled phone. This was adjudicated by DISA as having a minor operational impact with the intent to change the requirement to conditional in the next version of the UCR (UCR 2013).

(3) Public Safety Features. The UCR 2008, Change 2, section 5.3.2.2.2.2, states 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 UCR 2008, Change 2, section 5.3.2.2.2, states 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.

(5) Signaling Protocols. The UCR 2008, Change 2, section 5.3.2.2.3, states the LSC must use appropriate signaling for specific trunk types. The control/management protocol between the PEI and the LSC is, in general, proprietary. The control/management protocol between the AEI and the LSC is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements, of this document. The signaling protocol used on UC IP trunks is AS-SIP as specified in Section 5.3.4, AS-SIP Requirements. The MG in the LSC uses ANSI T1.619a PRI signaling on DSN PRI trunks. The SUT met the Signaling Protocol requirements T1 PRI (ANSI T1.619a and NI-2), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931). The SUT met all Signaling Protocol requirements except for AEI.

(6) Signaling Performance. The UCR 2008, Change 2, section 5.3.2.2.4, states the conditional requirements for call setup and tear-down times. The SUT met all signaling performance requirements

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

(1) Registration. The UCR 2008, Change 2, section 5.3.2.3.1, states that registration and authentication between the LSC and EIs shall follow the requirements set forth in UCR 2008, Change 2, Section 5.4, Information Assurance Requirements. This feature is tested by a DISA-led IA test team and is covered in a separate report, Reference (e).

(2) Failover. The UCR 2008, Change 2, section 5.3.2.3.2, states that 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 SUT does not support OPTIONS requests required to meet the failover to a secondary SS in accordance with UCR 2008, Change 2, Section 5.3.2.3.2.1. DISA adjudicated this discrepancy and determined that the UCR failover requirements are immature and require a rewrite. DISA NS2 has agreed to a Condition of Fielding that the initial UC APL certification will not provide for failover capability on the condition the vendor will participate in an NS2-scheduled multi-vendor test event to refine failover requirements, modify software to support the new failover requirements, and demonstrate failover compliance.

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

(1) Availability. The UCR 2008, Change 2, section 5.3.2.5.2.1, states that the Assured Services subsystem shall have a hardware/software availability of 0.99999

(non-availability of no more than 5 minutes per year). This requirement was met via the vendor's LoC.

(2) Maximum Downtimes. The UCR 2008, Change 2, section 5.3.2.5.2.2, states that 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

This requirement was met via the vendor's LoC.

(3) Loss of Packets. The UCR 2008, Change 2, section 5.3.2.5.4, states that for VoIP devices, the voice quality shall have a Mean Opinion Score (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 five-minute period. The SUT met all Packet Loss requirements for PEIs.

d. Voice End Instrument. The UCR 2008, Change 2, section 5.3.2.5.4, states that there are two types of IP voice instruments: PEIs and AS-SIP End Instruments (AEIs). The SUT met PEI requirements except for the noted softphone discrepancy in the following sub-paragraphs. The vendor does not support AEIs (voice or video). DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

(1) Tones and announcements. The UCR 2008, Change 2, section 5.3.2.5.4, states that tones and announcements, as required in UCR 2008, Change 2, section 5.3.2.4.1.1.1, UC Ringing Tones, Cadences, and Information Signals, and section 5.3.2.6.1.1.2, Announcements, shall be supported, except for the loss of C2 announcement. The SUT met all requirements for tones and announcements for PEIs with the following minor exception. The SUT does not support a "ping ring" notification. DISA adjudicated this as minor and stated the intent to change this to conditional in the next version of the UCR (UCR 2013).

(2) Audio codecs. The UCR 2008, Change 2, section 5.3.2.6.1.2, states that 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 met all audio codec requirements for PEIs.

(3) VoIP PEI or AEI Audio Performance Requirements. The UCR 2008, Change 2, section 5.3.2.6.1.3, states that VoIP PEIs or AEIs (i.e., handset, headset, and hands-free types) shall comply with TIA-810-B, November 3, 2006. The SUT met all audio performance requirements for PEIs.

(4) VoIP Sampling Standard. The UCR 2008, Change 2, section 5.3.2.6.1.4, states that 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. The SUT met the VoIP sampling standard requirements for PEIs.

(5) Authentication to LSC. The UCR 2008, Change 2, section 5.3.2.6.1.5, states that the PEI and AEI shall be capable of authenticating itself to its associated LSC and vice versa. The SUT met all PEI to LSC authentication requirements.

(6) Analog Telephone Support. The UCR 2008, Change 2, section 5.3.2.6.1.6, states that analog instruments, including secure analog EIs, analog facsimile EIs, and analog modem EIs, shall be supported by the LSC either by a Terminal Adapter or an Integrated Access Device (IAD) connected to an Ethernet port. The SUT met all analog telephone support requirements via FXS modules within their media gateways and a standalone VG224 analog gateway.

(7) Softphones. The UCR 2008, Change 2, section 5.3.2.6.1.7, states that 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 meet all softphones requirements with the following minor exception. The SUT softphone with Microsoft Windows Vista and Windows 7 OSs does not allow DSCP tagging per precedence level in accordance with UCR 2008, Change 2, Section 5.3.3.3.2. Microsoft Windows XP is the only OS that supports the five precedence levels. DISA adjudicated this as minor since all voice is queued together in the four-queue model currently used in deployed ASLANs.

(8) ISDN BRI. The UCR 2008, Change 2, section 5.3.2.6.1.8, states that the ISDN BRI EIs, including secure ISDN BRI EIs, may be supported by the LSC. This is a conditional requirement; no BRI EIs were provided on the SUT at the time of test.

e. Video End Instruments. The SUT must support both voice and video. PEIs and AEIs can support only voice, only video, or both voice and video. The SUT does not support AEIs. The SUT did not offer a video PEI. DISA has accepted and approved the vendor’s POA&M and adjudicated this discrepancy as having a minor operational impact. The SUT is certified without a video solution.

(1) Video End Instrument. The UCR 2008, Change 2, section 5.3.2.6.2, states that 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. This was not tested as the SUT did not provide any video end instruments.

(2) Display Messages, Tones, and Announcements. The UCR 2008, Change 2, section 5.3.2.6.2.1, states that tones and announcements, as appropriate for voice and video over IP, and as required, in UCR 2008, Change 2, sections 5.3.2.6.1.1.1 and

5.3.2.6.1.1.2, shall be supported by the PEI and AEI. This was not tested as the SUT did not provide any video end instruments.

(3) Video Codecs (Including Associated Audio Codecs). The UCR 2008, Change 2, section 5.3.2.6.2.2, states 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. This was not tested as the SUT did not provide any video end instruments.

#### f. LSC Requirements

(1) Precedence Based Assured Services (PBAS)/ASAC Requirements. The UCR 2008, Change 2, section 5.3.2.7.2.1, states 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 5.3.2.31.3, Multilevel Precedence and Preemption. The SUT met all PBAS/ASAC Requirements with the following exception. The SUT does not provide directionalization of its ASAC budget. This discrepancy was adjudicated to have a minor operational impact.

(2) Calling Number Delivery Requirements. The UCR 2008, Change 2, section 5.3.2.7.2.2, states that the calling number provided to the called party shall be determined by the dial plan serving the calling instrument in accordance with Telcordia Technologies GR-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 UCR 2008, Change 2, section 5.3.2.7.2.3, states that 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 WAN Transport. The UCR 2008, Change 2, section 5.3.2.7.2.4, states that 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 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. The SUT with SRST feature configured will function as expected during normal operations but during a failover provides ROUTINE only intra-enclave dialing for IP and analog users registered to the ISR with access to the PSTN for ROUTINE and emergency 911 calls. The SRST functionality is certified for use with the certified ISRs listed in Table 2-4.

(5) Local Location Server and Directory. The UCR 2008, Change 2, section 5.3.2.7.2.5, states that the purpose of the Local Location Server (LLS) is to provide

information on call routing and called address translation (where a called address is contained within the called SIP URI in the form of the called number). The SUT met all LLS and directory requirements.

(6) LSC Transport Interface Functions. The UCR 2008, Change 2, section 5.3.2.7.2.7, states that 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. The UCR 2008, Change 2, section 5.3.2.7.2.10, states that periodically, the LSC shall verify the status of its registered and authenticated IP EIs. The SUT met all status verification requirements for PEIs and attendant consoles with the vendor's LoC. The SUT does not support AEIs. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

(8) Line-Side Custom Features Interference. The UCR 2008, Change 2, section 5.3.2.7.2.11, states that 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. The SUT offers line-side custom features. However, JITC did not test any of those features; therefore, they are not certified for use.

(9) Loop Avoidance. The UCR 2008, Change 2, section 5.3.2.7.3, states that 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. The SUT met all Loop Avoidance requirements for T1 PRI (ANSI T1.619a and NI-2), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931).

#### g. Call Connection Agent (CCA) Requirements

(1) CCA Inter-Working Function (IWF) Component. The UCR 2008, Change 2, section 5.3.2.9.2.1, states that 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 EBCs, 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), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931).

(2) CCA Media Gateway Controller (MGC) Component. The UCR 2008, Change 2, section 5.3.2.9.2.2, states that 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 CCA MGC requirements.

(3) SG Component. The UCR 2008, Change 2, section 5.3.2.9.2.3, states that 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 (Common Channel Signaling 7 [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 UCR 2008, Change 2, section 5.3.2.9.5.1, states that 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 interfaces (T1 PRI (ANSI T1.619a and NI-2), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931)).

(5) CCA-IWF Support for Signaling System 7 (SS7). The UCR 2008, Change 2, section 5.3.2.9.5.2, states that CCA IWF may support the CCS7 protocol, consistent with the detailed CCS7 protocol requirements. CCA-IWF support for SS7 is a conditional requirement for LSCs and is not supported by the SUT.

(6) CCA-IWF Support for PRI, via MG. The UCR 2008, Change 2, section 5.3.2.9.5.3, states that the CCA IWF shall support the U.S./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), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931).

(7) CCA-IWF Support for CAS Trunks via MG. The UCR 2008, Change 2, section 5.3.2.9.5.4, states that support for CAS is a conditional requirement for LSCs. The CAS interface was tested and met all critical requirements for joint use.

(8) CCA-IWF Support for PEI and AEI Signaling Protocols. The UCR 2008, Change 2, section 5.3.2.9.5.5, states that 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 Signaling Protocols. No AEIs were tested.

(9) CCA-IWF Support for VoIP and TDM Protocol Interworking. The UCR 2008, Change 2, section 5.3.2.9.5.6, states that 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 interfaces: T1 PRI (ANSI T1.619a and NI-2), T1 CAS (DTMF), and E1 PRI (ITU-T Q.931).

(10) CCA Preservation of Call Ringing State during Failure Conditions. The UCR 2008, Change 2, section 5.3.2.9.6, states that 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. The SUT met T1 ISDN PRI (ANSI

T1.619a and ANSI T1.607), E1 PRI (ITU-T Q.931), and T1 CAS DTMF IWF requirements.

(11) CCA Interactions with Transport Interface Functions. The UCR 2008, Change 2, section 5.3.2.10.3, states that 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 the exception of AEIs. No AEIs were tested.

(12) CCA Interactions with the EBC. The UCR 2008, Change 2, section 5.3.2.10.4, states that 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 UCR 2008, Change 2, section 5.3.2.10.5, states that 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.

(14) CCA Support for User Features and Services (UFS). The UCR 2008, Change 2, section 5.3.2.10.6, states that 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 UCR 2008, Change 2, section 5.3.2.10.7, states that 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/TLS. The IA requirements are tested by a DISA-led IA test team and published in a separate report, Reference (e).

(16) CCA Interaction with EIs. The UCR 2008, Change 2, section 5.3.2.10.10, states that the CCA in the LSC needs to interact with VoIP PEIs and AEIs served by that LSC. 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 UCR 2008, Change 2, section 5.3.2.10.11, states that the CCA in the LSC shall support both assured Voice

and Video services. 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 VoIP interface between the AEI and the MFSS or LSC is AS-SIP. The SUT met all requirements for CCA support for AS Voice. The SUT did not provide Video. This was adjudicated to have a minor operational impact.

(18) CCA Interactions with Service Control Functions. The UCR 2008, Change 2, section 5.3.2.10.12, states that 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. The UCR 2008, Change 2, section 5.3.2.11, provides basic requirements for interworking call setup and release signaling between a DoD network using AS-SIP and a network using 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 on the SUT.

#### h. MG Requirements

(1) Role of MG In LSC. The UCR 2008, Change 2, section 5.3.2.12.3.1.7, states the MG supports interconnection of VoIP, Facsimile 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 vendor's LoC stipulates they meet ITU-T V.150.1 requirements for intra-enclave and PSTN calls. Based on multi-vendor interoperability testing the SUT does not properly handle ITU-T V.150 calls with the Avaya Communication Manager 6.0. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

(2) MG Support for ASAC. The UCR 2008, Change 2, section 5.3.2.12.4.1, states 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.

(3) MG and IA Functions. The UCR 2008, Change 2, section 5.3.2.12.4.2, states 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 Information Assurance 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 by an IA test team and the results published in a separate report, Reference (e).

(4) MG Interaction with Service Control Function. The UCR 2008, Change 2, section 5.3.2.12.4.3, states that 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 UCR 2008, Change 2, section 5.3.2.12.4.4, states that 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 UCR 2008, Change 2, section 5.3.2.12.4.5, states that 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 interactions for PEI interactions.

(7) MG IP-Based PSTN Interface Requirements. The UCR 2008, Change 2, section 5.3.2.12.4.7, states that the Voice and Video over IP 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. The SUT met this requirement,

(8) MG Interaction with VoIP EIs. The UCR 2008, Change 2, section 5.3.2.12.4.8, states that 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 requirements for MG Interaction with VoIP EIs.

(9) MG Support for User Features and Services. The UCR 2008, Change 2, section 5.3.2.12.4.9, states that 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.

(10) MG Interface to TDM network elements in DoD Networks. The UCR 2008, Change 2, section 5.3.2.12.5, states that each appliance MG shall support TDM

trunk groups that can interconnect with the following devices in DoD networks, in the United States and worldwide: PBXs, SMEOs, 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 UCR 2008, Change 2, section 5.3.2.12.6, states that the appliance suppliers should support TDM trunk groups on their MG product that can interconnect with devices in U.S. allied and coalition partner networks worldwide. This requirement is conditional and was not tested on the SUT.

(12) MG Interface to TDM PSTN in US. The UCR 2008, Change 2, section 5.3.2.12.7, states that each appliance MG shall support TDM trunk groups that can interconnect with NEs in the PSTN in the United States. The SUT met all requirements for MG Interface to TDM PSTN in the U.S. using T1 PRI and T1 CAS.

(13) MG Interfaces to TDM PSTN OCONUS. The UCR 2008, Change 2, section 5.3.2.12.8, states that 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 PTT networks (OCONUS) worldwide. This requirement was met for the E1 PRI (Q.931) interface.

(14) MG Support for CCS7. The UCR 2008, Change 2, section 5.3.2.12.9, states that the MG shall support TDM trunk groups that are controlled by a separate CCA-to-SG signaling link that carries CCS7 protocol. The MG shall support these TDM trunk groups, and the SG shall support CCS7 signaling. This conditional requirement was not tested on the SUT.

(15) MG Support for ISDN PRI Trunks. The UCR 2008, Change 2, section 5.3.2.12.10, states that the MG shall support ISDN PRI trunk groups that carry the U.S./National ISDN version of the ISDN PRI protocol. The SUT met all requirements for MG Support for ISDN T1 PRI Trunks.

(16) MG Support for CAS Trunks. The UCR 2008, Change 2, section 5.3.2.12.11, states that the MG shall support CAS trunk groups that carry the U.S. version of the CAS protocol. The SUT met all requirements for MG Support for CAS Trunks with DTMF.

(17) MG requirements for VoIP Internal Interfaces. The UCR 2008, Change 2, section 5.3.2.12.12, states that 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.

(18) MG Echo Cancellation. The UCR 2008, Change 2, section 5.3.2.12.13, states that 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 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 UCR 2008, Change 2, section 5.3.2.12.14, states that the MG shall derive its clock timing from a designated T1 or PRI interface. The SUT met all MG Clock Timing requirements.

(20) MGC-MG CCA Functions. The UCR 2008, Change 2, section 5.3.2.12.15, states that 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 CCA MGC requirements.

(21) MG ITU-T V.150.1. The UCR 2008, Change 2, section 5.3.2.12.16, states that the 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 UDP) and port numbers so that the transition is transparent to the EBC. The vendor’s LoC stipulates they meet ITU-T V.150.1 requirements for intra-enclave and PSTN calls. Based on multi-vendor interoperability testing the SUT does not properly handle ITU-T V.150 calls with the Avaya Communication Manager 6.0. DISA has accepted and approved the vendor’s POA&M and adjudicated this discrepancy as having a minor operational impact.

(22) MG Preservation of Call Ringing during Failure. The UCR 2008, Change 2, section 5.3.2.12.17, states that 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 this requirement through testing.

#### i. SG Requirements

(1) SG and CCS7 Network Interactions. The UCR 2008, Change 2, section 5.3.2.13.5.1, states that the SG shall support signaling connectivity to the 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 is not supported by the SUT.

(2) SG Interactions with CCA. The UCR 2008, Change 2, section 5.3.2.13.5.2, states that 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 UCR 2008, Change 2, section 5.3.2.13.5.3, states that the SG will terminate CCS7 links on its CCS7 side and transport the CCS7 call control and service control protocols (i.e., ISUP and TCAP) 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 UCR 2008, Change 2, section 5.3.2.16, states that the precedence level and dialed number input to the PEI or AEI shall be as specified in the sub-paragraphs of this section. The SUT met all requirements for WWNDP for PEIs.

(2) DSN WWNDP. The UCR 2008, Change 2, section 5.3.2.16.1, states that 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 RFC 3966. The SUT met all DSN WWNDP requirements for PEIs.

k. Commercial Cost Avoidance. The UCR 2008, Change 2, section 5.3.2.23, states that the Commercial Cost Avoidance requirements are in UCR 2008, Change 2, sections 5.3.2.28.3 and 5.3.2.28.4. The LSC must use a Commercial Cost Avoidance functionality to route calls from an IP EI to a PSTN E.164 number in a manner which will minimize commercial costs associated with DSN calls. The SUT does not support Commercial Cost Avoidance. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

l. AS-SIP Based for External Devices (Voicemail, Unified Messaging and Automated Receiving Devices). The UCR 2008, Change 2, section 5.3.2.24, states that the LSC shall support all mandatory requirements in RFC 3842. No AS-SIP external devices were tested.

m. Precedence Call Diversion (PCD). The UCR 2008, Change 2, section 5.3.2.25, states that the AS-SIP signaling appliance shall divert ALL unanswered Real Time Services (RTS) VoIP calls above the ROUTINE precedence level to a designated RTS DN for PCD. The SUT met all precedence call diversion requirements.

n. Attendant Station Features. The UCR 2008, Change 2, section 5.3.2.26, states that the attendant features in this section apply to attendant consoles that are provided as part of the local LSC, or provided be an external CPE attendant console. No attendant station was provided on the SUT at the time of test; therefore, none of the following features were tested. The vendor provided an LoC, which stated they met the requirements in the sub-paragraphs below with two minor exceptions.

(1) Precedence and Preemption. The RTS Attendant Console shall interoperate with PBAS/ASAC.

(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.

(3) Class of Service 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.

(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 does not fully comply with Busy Override and Busy Line Verification requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

(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.

(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.

(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 does not fully comply with attendant console queuing requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

#### o. AS-SIP Requirements

(1) SIP Requirements for AS-SIP Signaling Appliances and AS-SIP EIs. The UCR 2008, Change 2, section 5.3.4.7, states that there are two categories of IP EIs; AS-SIP EIs are IP EIs that support AS-SIP, and Proprietary IP EIs, which are SIP, ITU-T H.323, or other vendor proprietary EIs. The LSCs must meet the requirements defined in this section for supporting the interoperable AS-SIP line side interface with AS-SIP EIs. The LSCs that support SIP EIs MUST comply with the differentiated set of requirements defined for ITU-T H.323 and/or vendor-proprietary EIs if they serve H.323 and/or vendor-proprietary EIs. The SUT only supports voice PEIs. The vendor does not support AEIs (voice or video). DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

(2) SIP Session Keep-Alive Timer. The UCR 2008, Change 2, section 5.3.4.8, states that 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). The UCR 2008, Change 2, section 5.3.4.9, states that 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, section 5.3.4.10. The SUT met all critical precedence and preemption requirements.

(5) Video Telephony – General Rules. Video calls must meet the detailed requirements for video telephony messaging as defined in UCR 2008, Change 2, section 5.3.4.12. Video telephony requirements were not tested on the SUT. The SUT is not fully compliant with the requirements. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact. The SUT is certified without a video solution.

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

(7) SIP Translation Requirements for Inter-working AS-SIP Signaling Appliances. The UCR 2008, Change 2, section 5.3.4.14 addresses the AS-SIP-TDM signaling interworking requirements for an LSC. The SUT met this requirement with ANSI T1.619a ISDN PRI NI2 DSN and ISDN PRI NI2 PSTN TDM interfaces interworking with AS-SIP. This requirement was met with both testing and the vendor's LoC. The SUT does not support CCS7 TDM interface which is conditional for an LSC.

(8) Relevant Timers for the Terminating Gateway and the Originating Gateway. The UCR 2008, Change 2, section 5.3.4.15, includes the relevant timers for terminating and originating gateways. This requirement applies to gateways between AS-SIP and CCS7 links. Because CCS7 is a conditional requirement for LSCs and not supported by the SUT, this requirement was not tested.

(9) SIP Requirements for Interworking AS-SIP Signaling Appliances. Interworking AS-SIP signaling appliances must comply with UCR 2008 Section 5.3.4.7.1, AS-SIP Signaling Appliances and AS-SIP EIs, as well as the additional general requirements in UCR 2008 Section 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,

Section 5.3.4.8, SIP Session Keep-Alive Timer, as well as the additional E1 requirements listed in UCR 2008 Section 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, section 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, section 5.3.4.19, for supplementary services. The SUT met this requirement with testing and the vendor's LoC.

p. IPv6 Requirements. The UCR 2008, Change 2, section 5.3.4.14, depicts the IPv6 requirements. These requirements were met by the SUT with both testing and the vendor's LoC. The vendor submitted an IPv6 LoC with the following noted discrepancies: The SUT does not support Request for Comments (RFCs) 4861 and 4862. The vendor states support for these RFCs will be available in a future release. DISA has accepted and approved the vendor's POA&M and adjudicated this discrepancy as having a minor operational impact.

q. NM. The Vendor submitted an NM LoC with noted discrepancies. Open TDRs were adjudicated by DISA to have a minor operational impact with vendor submitted PoAM.

(1) LSC Management Function. The UCR 2008, Change 2, section 5.3.2.7.2.6, states that the LSC Management function supports functions for LSC Fault, Configuration, Accounting, Performance and Security (FCAPS) management and audit logs. This was met by the SUT with a vendor submitted LoC.

(2) VVoIP Network Management System (NMS) Interface Requirements. The UCR 2008, Change 2, section 5.3.2.4.4, states that 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. The SUT met this requirement with the vendor's LoC.

(3) General Management Requirements. The UCR 2008, Change 2, section 5.3.2.17.2, states that management of DoD UC Voice and Video services requires each UC product have a minimum of two separate management domains. Typically, one domain will provide local support referred to as Operational Administration and Management (OA&M) and the other domain will provide a remote centralized management capability referred to as NM. LSCs must be capable of providing the following NM data to the end-to-end (E2E) RTS EMS: alarm/log, performance, accounting. The communication between the LSC and EMS shall be via IP. The LSC

shall issue notifications and respond to requests for resource information. The SUT met all general management requirements with the vendor's LoC.

(4) Requirement for FCAPS Management. The UCR 2008, Change 2, section 5.3.2.17.3, states that the LSC must meet all general requirements for the FCAPS management functional areas. This was met by the SUT with a vendor submitted LoC.

(5) NM requirements of Appliance Functions. The UCR 2008, Change 2, section 5.3.2.18.2, states that the LSC must meet all management requirements for ASAC, CCA, SG, and MG functions. This was met by the SUT with a vendor submitted LoC.

(6) Accounting Management. The UCR 2008, Change 2, section 5.3.2.19, provides a minimum set of requirements to capture basic call information for accounting purposes. 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 does not comply with the requirement for the equipment impairment factor to be in accordance with ITU-T G.107. DISA adjudicated this as minor with a vendor POA&M to provide a MOS score. DISA also stated the intent to change this to conditional in the next version of the UCR (UCR 2013). The SUT does not have the ability to transfer records to a removable physical storage media. DISA adjudicated this as minor and stated the intent to delete this requirement in the next version of the UCR (UCR 2013).

**11.3 Information Assurance.** The IA requirements are tested by an IA test team and the results 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>. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: [disa.meade.ns.list.unified-capabilities-certification-office@mail.mil](mailto:disa.meade.ns.list.unified-capabilities-certification-office@mail.mil).

## 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. LSC Products Capability/Functional Requirements**

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. LSC 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. LSC 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. LSC 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 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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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, AEIs, 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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
123	<p>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:</p> <ul style="list-style-type: none"> <li>a. Supplier-proprietary voice PEIs</li> <li>b. Voice SIP EIs, when the appliance supplier supports these EIs</li> <li>c. Voice H.323 EIs, when the appliance supplier supports these EIs</li> <li>d. Voice AS-SIP AEIs</li> </ul>	5.3.2.12.4.8	Y	Y	NA
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
133	The MG shall support IPv4 as a Network Layer Protocol. The MG shall also support IPv6 as a Network Layer Protocol.	5.3.2.12.12.2	Y	Y	NA
134	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.	5.3.2.12.12.2	Y	Y	NA
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., CCS7, PRI, or CAS) within each MG within the LSC or MFSS.	5.3.2.12.15	Y	Y	Y

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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 EIs, other appliances, and MFSS MGs to contain <ul style="list-style-type: none"> <li>• Called addresses including DSN numbers from the DSN numbering plan</li> <li>• Called addresses including E.164 numbers from the E.164 numbering plan</li> </ul>	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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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
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

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

ID	Requirement	UCR Ref (UCR 2008 CH2)	MFSS	LSC	WAN SS
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. LSC Products Capability/Functional Requirements Table (continued)**

LEGEND:		
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 Canceller	
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	
le	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	kilobits per second
	LAN	Local Area Network
	LDAP	Lightweight Directory Access Protocol
	LDAPv3	Lightweight Directory Access Protocol version 3
	LSC	Local Session Controller
	MAC	Media Access Control
	Mbps	Megabytes per second
	MFS	Multifunction Switch
	MFSS	Multifunction SoftSwitch
	MG	Media Gateway
	MGC	Media Gateway Controller
	MLD	Multicast Listener Discovery
	MLPP	Multilevel Precedence and Preemption
	Modem	Modulator/Demodulator
	MolP	Modem over Internet Protocol
	MOS	Mean Opinion Score
	MPLS	Multiprotocol Label Switching
	ms	milliseconds
	MTU	Maximum Transmission Unit
	NE	Network Element
	NM	Network Management
	NMS	Network Management System
	OA&M	Operations, Administration, and Maintenance
	OCONUS	Outside the Continental United States
	OSI	Open Systems Interconnect
	OTGR	Operations Technology Generic Requirements
	PBAS	Precedence Based Assured Services
	PBX	Private Branch Exchange
	PCD	Precedence Call Diversion
	PCM	Pulse Code Modulation
	PEI	Proprietary End Instrument
	PLCP	Physical Layer Convergence Protocol
	PRI	Primary Rate Interface
	PSTN	Public Switch Telephone Network
	REL	Release Message
	RFC	Request For Communication
	RTS	Real Time Services
	SAC	Session Admission Control
	SBU	Sensitive, but Unclassified
	SCIP	Secure Communications Interoperability Protocol
	SCS	Session Control and Signaling
	SDP	Session Description Protocol
	SG	Signaling Gateway
	SIP	Session Initiation Protocol
	SMEO	Small End Office
	SONET	Synchronous Optical Network
	SRTCP	Secure Real-Time Transport Control Protocol
	S RTP	Secure Real-Time Transport Protocol
	SS	Softswitch
	SS7	Signaling System number 7
	SUT	System Under Test
	TA	Terminal Adaptor
	TCA	Threshold Crossing Alert
	TCLw	Weighted Terminal Coupling Loss
	TDM	Time Division Multiplexing
	TIA	Telecommunications Industry Association
	TLS	Transport Layer Security
	UAC	User Agent Client
	UAS	User Agent Server
	UC	Unified Capabilities
	UCR	Unified Capabilities Requirements
	UDP	User Datagram Protocol
	URI	Uniform Resource Identifier
	U.S.	United States
	VBD	Voice Band Data
	VoIP	Voice over Internet Protocol
	VVoIP	Voice and Video over Internet Protocol
	WAN	Wide Area Network
	Y	Yes