



## DEFENSE INFORMATION SYSTEMS AGENCY

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

### MEMORANDUM FOR DISTRIBUTION

**3 Jun 11**

**SUBJECT:** Special Interoperability Test Certification of Cisco Unified Communications Manager Express (CUCME) Version 8.0 with Internetwork Operating System (IOS) Software Release 15.1(1) T3

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 (f), 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 Cisco CUCME Version 8.0 with IOS Software Release 15.1(1) T3 is hereinafter referred to as the system under test (SUT). The SUT meets all of its critical interoperability requirements and is certified for joint use within the Defense Information System Network (DISN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the Voice over Internet Protocol (VoIP) critical interoperability requirements with any certified Assured Services Local Area Network (ASLAN) or ASLAN components on the Unified Capabilities (UC) Approved Products List (APL). The SUT meets the critical interoperability requirements set forth in References (c) and (d), using test procedures derived from Reference (e). No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date the DISA Field Security Operations (FSO) provided a positive Certification and Accreditation (CA) Recommendation.

3. This finding is based on interoperability testing conducted by JITC, DISA adjudication of open test discrepancy reports (TDRs), review of the vendor's Letters of Compliance (LoC), and FSO CA Recommendation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 20 September through 29 October 2010. Review of vendor's LoC was completed on 15 January 2011. DISA adjudication of outstanding test discrepancy reports was completed on 25 February 2011. Verification and Validation testing was conducted from 4 through 8 April 2011 to validate TDR fixes. The FSO provided a positive CA Recommendation on 5 May 2011 based on the security testing completed by DISA-led IA test teams and published in a separate report, Reference (f).

Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The interoperability test summary of the SUT is indicated in Table 1. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT’s ability to meet:

- a. DSN services for Network and Applications specified in Reference (g).
- b. PBX 1 interface and signaling requirements for trunks/lines specified in Reference (c) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in Reference (c) verified through JITC testing and/or vendor submission of LoC.
- d. The voice quality, IPv6, and softphone requirements specified in Reference (d).
- e. The overall system interoperability performance derived from test procedures listed in Reference (e).

**Table 1. SUT Interoperability Test Summary**

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS. <sup>1</sup>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not provide a ping ring when the phone is configured with the Call Forward Variable feature. <sup>2</sup> The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. <sup>3</sup> All IP end instruments failed to register with IPv6. <sup>4</sup> CP-7940G, CIS-7961 and CP-7960G end instruments did not meet the Voice Quality E-model MOS; however, they did meet the SAGE MOS of 4.0 or better. <sup>5</sup>

**Table 1. SUT Interoperability Test Summary (continued)**

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco IP phones do not support Call Waiting. <sup>6</sup>	
Attendant	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Public Safety	Yes	Certified	All public safety features are conditional. The SUT met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features. <sup>7</sup>	
Conferencing	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Nailed-up Connections	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
DSN Hotline Services	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the Loss of Command and Control announcement. <sup>8</sup>	
Call Processing	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services	Yes	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
Reliability	Yes	Certified	Met all critical CRs and FRs.	
Security	Yes	Certified	See note 9.	
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL. (See note 3.)	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS. <sup>1</sup>
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs. <sup>10</sup>
<b>NOTES:</b>				
1 The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA adjudicated this anomaly as having a minor operational impact.				
2 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.				
3 The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception: 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 Enclosure 2 and a dual stack call control agent in accordance with Reference (d). This was adjudicated by DISA as having a minor operational impact.				
4 All dual stack IP end instruments can only be configured to register with IPv4. When configured to register with IPv6, the IP end instruments fail to register. This discrepancy was adjudicated by DISA as having a minor operational impact. All other dual stack IPv6 requirements were met.				
5 The CP-7940G, CIS-7961, and CP-7960G end instruments did not meet the Voice Quality E-model MOS; however, they did meet the SAGE MOS of 4.0 or better. This discrepancy was adjudicated by DISA as having a minor operational impact.				
6 All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.				

**Table 1. SUT Interoperability Test Summary (continued)**

<b>NOTES (continued):</b>			
7	The SUT only supports emergency basic 911 service. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.		
8	The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA previously adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1.		
9	Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).		
10	This interface requirement was met by the vendor's LoC.		
<b>LEGEND:</b>			
802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
ANSI	American National Standards Institute	MCS	Media Convergence Servers
APL	Approved Products List	MFR1	Multi-Frequency Recommendation 1
ASLAN	Assured Services Local Area Network	MLPP	Multi-Level Precedence and Preemption
BRI	Basic Rate Interface	ms	milliseconds
C2	Command and Control	NI 1/2	National ISDN Standard 1 or 2
CAS	Channel Associated Signaling	NI2	National ISDN Standard 2
CoS	Class of Service	NIC	Network Interface Card
CP	Cisco Phone	NFAS	Non Facility Associated Signaling
CRs	Capability Requirements	OAM	Operational Administration and Maintenance
DISA	Defense Information Systems Agency	MOS	Mean Opinion Score
DP	Dial Pulse	PBX 1	Private Branch Exchange 1
DSCP	Differentiated Services Code Point	PMO	Program Management Office
DSN	Defense Switched Network	PNT	Preemption Notification Tone
DSS1	Digital Subscriber Signaling 1	PRI	Primary Rate Interface
DTMF	Dual Tone Multi-Frequency	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.931	Signaling Standard for ISDN
EI	End Instrument	Q.955.3	ISDN Signaling standard for E1 MLPP
FRs	Feature Requirements	RTCP	RTP Control Protocol
GR	Generic Requirement	RTP	Real-time Transport Protocol
GR-506	LSSGR: Signaling for Analog Interfaces	SS7	Signaling System 7
ICA	Isolated Code Announcement	SUT	System Under Test
IEEE	Institute of Electrical and Electronics Engineers	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IP	Internet Protocol	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IPv4	Internet Protocol version 4	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IPv6	Internet Protocol version 6	TCP	Transmission Control Protocol
ISDN	Integrated Services Digital Network	TFTP	Trivial File Transfer Protocol
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UC	Unified Capabilities
JITC	Joint Interoperability Test Command	UCR	Unified Capabilities Requirements
LoC	Letters of Compliance	UDP	User Datagram Protocol
		VoIP	Voice over Internet Protocol

**Table 2. PBX 1 Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional	References	
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.3.1</li> <li>• UCR Section 5.2.1.3.2</li> <li>• UCR Section 5.2.1.3.4.1</li> <li>• UCR Section 5.2.1.3.4.1.1</li> <li>• UCR Section 5.2.1.3.4.2</li> <li>• UCR Section 5.2.1.3.4.2.1</li> </ul>	
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> <li>• PBX Line (C)</li> <li>• Direct Inward Dialing (C)</li> <li>• National ISDN 1/2 Primary Access (R)</li> <li>• ISDN ANSI MLPP Service Capability (R)</li> <li>• ITU-T ISDN Primary Access (Europe only) (C)</li> <li>• ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C)</li> <li>• Normal Wink Start Operations (C)</li> <li>• Glare Operation (R)</li> <li>• Abnormal Wink Start (R)</li> <li>• Glare Resolution (R)</li> <li>• Call for Service Timing (C)</li> <li>• Guard Timing (R)</li> <li>• Satellite Interface (C)</li> <li>• Disconnect Control (C)</li> <li>• Reselect and Retrial (C)</li> <li>• Off-Hook Supervision Transition (C)</li> <li>• Dial-Pulse Signals (C)</li> <li>• DTMF Signaling (C)</li> <li>• Standard Digit Format for Precedence (C)</li> <li>• MFR1 2/6 Signaling (C)</li> <li>• Alerting Signals and Tones (R)</li> <li>• DSN ISDN User-to-Network Signaling (R)</li> <li>• Application (R)</li> <li>• Physical Layer (R)</li> <li>• Data Link Layer (R)</li> <li>• Data Link Connection (R)</li> <li>• Peer-to-Peer Procedures of Data-Link Layer (R)</li> <li>• Layer 3 DSN User-to-Network Signaling (R)</li> <li>• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)</li> <li>• Sequence of Messages for DSN Circuit-Switched Calls (R)</li> <li>• Message Functional Definition and Content (R)</li> <li>• General Message Format and Information Elements Coding (R)</li> <li>• Supplementary Services (C)</li> <li>• PCM-24 Digital Trunk Interface (R)</li> <li>• Interface Characteristics (R)</li> <li>• Supervisory Channel Associated Signaling (C)</li> <li>• Clear Channel Capability (R)</li> <li>• Alarm and Restoral Requirements (R)</li> <li>• PCM-30 Digital Trunk Interface (Europe only) (R)</li> <li>• Interoperation of PCM-24 and PCM-30 (C)</li> <li>• Analog Trunk Interface (C)</li> <li>• Integrated Digital Loop Carrier (C)</li> <li>• Trunk Group-Remove from Service (C)</li> <li>• Trunk Group-Restore to Service (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.3.3.1.1</li> <li>• UCR Section 5.2.4.3.3.1.2</li> <li>• UCR Section 5.2.4.3.3.2.1</li> <li>• UCR Section 5.2.4.3.3.2.2</li> <li>• UCR Section 5.2.4.3.5</li> <li>• UCR Section 5.2.4.3.6</li> <li>• UCR Section 5.2.3.4.7</li> <li>• UCR Section 5.2.3.4.8</li> <li>• UCR Section 5.2.3.4.9</li> <li>• UCR Section 5.2.3.4.10</li> <li>• UCR Section 5.2.4.4.1</li> <li>• UCR Section 5.2.4.4.2</li> <li>• UCR Section 5.2.4.4.2.1</li> <li>• UCR Section 5.2.4.4.3</li> <li>• UCR Section 5.2.4.5.1</li> <li>• UCR Section 5.2.4.7.1.4.2</li> <li>• UCR Section 5.2.4.7.1.1</li> <li>• UCR Section 5.2.4.7.1.2</li> <li>• UCR Section 5.2.4.7.1.3</li> <li>• UCR Section 5.2.4.7.1.3.1</li> <li>• UCR Section 5.2.4.7.1.3.2</li> <li>• UCR Section 5.2.4.7.1.4</li> <li>• UCR Section 5.2.4.7.1.4.2</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.4.3</li> <li>• UCR Section 5.2.4.7.1.4.4</li> <li>• UCR Section 5.2.4.7.1.4.5</li> </ul>	
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)		<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.4.6</li> <li>• UCR Section 5.2.6.1</li> <li>• UCR Section 5.2.6.1.1</li> <li>• UCR Section 5.2.6.1.2</li> <li>• UCR Section 5.2.6.1.3</li> <li>• UCR Section 5.2.6.1.4</li> <li>• UCR Section 5.2.6.2</li> <li>• UCR Section 5.2.6.3</li> <li>• UCR Section 5.2.6.4</li> <li>• UCR Section 5.2.6.5</li> <li>• UCR Section 5.2.1.5.5</li> <li>• UCR Section 5.2.1.5.5</li> </ul>	

**Table 2. PBX 1 Requirements (continued)**

<b>DSN Trunk Interfaces (continued)</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
T1 CAS (MFR1, DTMF, DP)	No	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>	
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>	
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> </ul>	
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• FTR 1080B-2002</li> </ul>	
<b>DSN Line Interfaces</b>					
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• Directory Number Identification (R)</li> <li>• Analog Line (R)</li> <li>• National ISDN 1/2 Basic Access (R: BRI Only)</li> <li>• Basic Line Test Capabilities (R)</li> <li>• Advanced Line Test Capabilities (C)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Reverse Battery (R: 2-Wire Analog only)</li> <li>• Alerting Signals and Tones (R)</li> <li>• S/T Reference Point (R: ISDN BRI only)</li> <li>• VoIP System Requirements (R: VoIP Phones only)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.1.1</li> <li>• UCR Section 5.2.1.3.5</li> <li>• UCR Section 5.2.1.3.3</li> <li>• UCR Section 5.2.1.5.4.1.1</li> <li>• UCR Section 5.2.1.5.4.1.1</li> <li>• UCR Section 5.2.4.2.1</li> <li>• UCR Section 5.2.4.3.1</li> <li>• UCR Section 5.2.4.5.1</li> <li>• UCR Section 5.2.4.7.1.2.1</li> <li>• UCR Section 5.2.12.8</li> </ul>	
ISDN BRI NI 1/2 (ANSI T1.619a)	No			<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure Calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>
2-Wire Proprietary Digital	No			<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
VoIP (Ethernet IEEE 802.3u)	No	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R: 2-Wire Analog only)</li> <li>• Secure data (STE/STU-III) (R: 2-Wire Analog only)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>	
		VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: BRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• FTR 1080B-2002</li> </ul>	
<b>DSN Features &amp; Capabilities</b>					
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
Common Features	Yes	<ul style="list-style-type: none"> <li>• Individual Lines (R)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (R)</li> <li>• Three-way calling (R)</li> <li>• Add-on transfer, conference calling, and call hold (C)</li> <li>• Call Transfer Individual – All calls (R)</li> <li>• Call Transfer - Internal Only (R)</li> <li>• Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)</li> <li>• Call Transfer – Outside (R)</li> <li>• Call Transfer – Add-On Restricted Station (C)</li> <li>• Call Transfer – Attendant (C)</li> <li>• Call Hold (R)</li> <li>• Conference Calling – Six Way Station Controlled (C)</li> <li>• Call Forwarding Variable (R)</li> <li>• Call Forward Busy Line (R)</li> <li>• Call Forwarding – Don't Answer – All Calls (R)</li> <li>• Selective Call Forwarding (C)</li> <li>• Call pick-up (C)</li> <li>• Address Translation (C)</li> <li>• Assured Dial Tone (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.1.1</li> <li>• UCR Section 5.2.1.1.3</li> <li>• UCR Section 5.2.1.1.4</li> <li>• UCR Section 5.2.1.1.5.1</li> <li>• UCR Section 5.2.1.1.6</li> <li>• UCR Section 5.2.1.1.7</li> <li>• UCR Section 5.2.1.1.7.1</li> <li>• UCR Section 5.2.1.1.7.2</li> <li>• UCR Section 5.2.1.1.7.3</li> <li>• UCR Section 5.2.1.1.7.4</li> <li>• UCR Section 5.2.1.1.7.5</li> <li>• UCR Section 5.2.1.1.7.6</li> <li>• UCR Section 5.2.1.1.7.7</li> <li>• UCR Section 5.2.1.1.7.8</li> <li>• UCR Section 5.2.1.1.8.1</li> <li>• UCR Section 5.2.1.1.8.2</li> <li>• UCR Section 5.2.1.1.8.3</li> <li>• UCR Section 5.2.1.1.8.4</li> <li>• UCR Section 5.2.1.1.9.1</li> <li>• UCR Section 5.2.1.7</li> <li>• UCR Section 5.2.1.9</li> </ul>	
Attendant	No	<ul style="list-style-type: none"> <li>• Attendant Features (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.2.2</li> </ul>	

**Table 2. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Public Safety	Yes	<ul style="list-style-type: none"> <li>• Emergency Service Basic (911) Caller (R)</li> <li>• Emergency Service (911) Public Safety Answering Service (C)</li> <li>• Enhanced Emergency Service (E911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing call trace (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.4.1.1</li> <li>• UCR Section 5.2.1.4.1.2</li> <li>• UCR Section 5.2.1.4.1.3</li> <li>• UCR Section 5.2.1.4.2</li> <li>• UCR Section 5.2.1.4.3</li> </ul>
Conferencing	No	<ul style="list-style-type: none"> <li>• Preset Conferencing (C)</li> <li>• Meet-Me Conferencing (C)</li> <li>• Progressive Conferencing (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.6.1</li> <li>• UCR Section 5.2.1.6.2</li> <li>• UCR Section 5.2.1.6.3</li> </ul>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Nailed-Up Connections (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.8</li> </ul>
DSN Hotline Services	No	<ul style="list-style-type: none"> <li>• DSN Analog Hotline Service (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.12</li> </ul>
MLPP	Yes	<ul style="list-style-type: none"> <li>• MLPP Overview (R)</li> <li>• Preemption in the Network (R)</li> <li>• Network Facility with Lower Precedence Calls (R)</li> <li>• Network Facility with Equal or Higher Precedence Calls (R)</li> <li>• Precedence Call Diversion (R)</li> <li>• Channel Associated Signaling (C)</li> <li>• Primary Rate Interface (R)</li> <li>• Analog Line MLPP (R)</li> <li>• ISDN MLPP Basic Rate Interface (C)</li> <li>• ISDN Primary Rate Interface (R)</li> <li>• Precedence Call Waiting (R)</li> <li>• Call Forwarding (R)</li> <li>• Call Transfer (R)</li> <li>• Call Hold (R)</li> <li>• Three-Way Calling (R)</li> <li>• Call Pickup (C)</li> <li>• Conferencing (C)</li> <li>• Multiline Hunt Group (C)</li> <li>• Community of Interest (C)</li> <li>• MLPP Interaction with EKTS features (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.2.1.1</li> <li>• UCR Section 5.2.2.2</li> <li>• UCR Section 5.2.2.2.1</li> <li>• UCR Section 5.2.2.2.2</li> <li>• UCR Section 5.2.2.3</li> <li>• UCR Section 5.2.2.4.1</li> <li>• UCR Section 5.2.2.4.2</li> <li>• UCR Section 5.2.2.5</li> <li>• UCR Section 5.2.2.6</li> <li>• UCR Section 5.2.2.7</li> <li>• UCR Section 5.2.2.8.1</li> <li>• UCR Section 5.2.2.8.2</li> <li>• UCR Section 5.2.2.8.3</li> <li>• UCR Section 5.2.2.8.4</li> <li>• UCR Section 5.2.2.8.5</li> <li>• UCR Section 5.2.2.8.6</li> <li>• UCR Section 5.2.2.8.7.1</li> <li>• UCR Section 5.2.2.8.8</li> <li>• UCR Section 5.2.2.8.9</li> <li>• UCR Section 5.2.2.10.1</li> </ul>

**Table 2. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities (continued)</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Call Processing	Yes	<ul style="list-style-type: none"> <li>• Call Treatments (R)</li> <li>• Primary and Alternate Routing (R)</li> <li>• E&amp;M Lead Signaling States (C)</li> <li>• 4-Wire Analog User Access Lines (C)</li> <li>• 2-Wire User Access Lines (R)</li> <li>• Termination of Analog Lines (R)</li> <li>• DSN User Dialing (R)</li> <li>• Interswitch and Intraswitch Dialing (R)</li> <li>• Seven-Digit Dialing (R)</li> <li>• Ten-Digit Dialing (R)</li> <li>• Access Code (R)</li> <li>• Access Digit (R)</li> <li>• Precedence Digit (R)</li> <li>• Service Digit (R)</li> <li>• Route Code (R)</li> <li>• Area Code (R)</li> <li>• Switch Code (R)</li> <li>• Line Number (R)</li> <li>• Calling Name Delivery (C)</li> <li>• Calling Number Delivery (R)</li> <li>• Emergency Service 911 Conflict Resolution (R)</li> <li>• DSN Switch Outpulsing Digit Formats (C)</li> <li>• Standard Directory Number (C)</li> <li>• Standard Test Numbers (C)</li> <li>• Base Services – Abbreviated Numbers (R)</li> <li>• Digit Reception Requirements (R)</li> <li>• Screening (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.3.1</li> <li>• UCR Section 5.2.3.2</li> <li>• UCR Section 5.2.3.3.1</li> <li>• UCR Section 5.2.3.3.2</li> <li>• UCR Section 5.2.3.3.3</li> <li>• UCR Section 5.2.3.3.4</li> <li>• UCR Section 5.2.3.5.1.1</li> <li>• UCR Section 5.2.3.5.1.1</li> <li>• UCR Section 5.3.3.5.2.1</li> <li>• UCR Section 5.2.3.5.2.2</li> <li>• UCR Section 5.2.3.5.1.3</li> <li>• UCR Section 5.2.3.5.1.3.1</li> <li>• UCR Section 5.2.3.5.1.3.2</li> <li>• UCR Section 5.2.3.5.1.3.3</li> <li>• UCR Section 5.2.3.5.1.4</li> <li>• UCR Section 5.2.3.5.1.5</li> <li>• UCR Section 5.2.3.5.1.6</li> <li>• UCR Section 5.2.3.5.1.7</li> <li>• UCR Section 5.2.3.5.1.8.1</li> <li>• UCR Section 5.2.3.5.1.8.2</li> <li>• UCR Section 5.2.3.5.1.9</li> <li>• UCR Section 5.2.3.5.2</li> <li>• UCR Section 5.2.3.5.3</li> <li>• UCR Section 5.2.3.5.4</li> <li>• UCR Section 5.2.3.5.5</li> <li>• UCR Section 5.2.3.5.6</li> <li>• UCR Section 5.2.3.5.8</li> </ul>
ISDN Services	Yes	<ul style="list-style-type: none"> <li>• BRI Access, Call Control and Signaling (R)</li> <li>• Uniform Interface Configuration for BRIs (R)</li> <li>• EKTS (C)</li> <li>• PRI Access, Call Control and Signaling (R)</li> <li>• PRI Features (R)</li> <li>• Packet Data Features and Capabilities (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.9.2, Table 5.2.9-1</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-2</li> <li>• UCR Section 5.2.9.3, Table 5.2.9-3</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-4</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-5</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-6</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> <li>• Synchronization Performance Monitoring Criteria (C)</li> <li>• DS1 Traffic Interfaces (C)</li> <li>• DS0 Traffic Interconnects (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.10.1.1.2</li> <li>• UCR Section 5.2.10.1.1.2.2</li> <li>• UCR Section 5.2.10.2</li> <li>• UCR Section 5.2.10.3</li> <li>• UCR Section 5.2.10.4</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• System Availability (R)</li> <li>• Backup Power (R)</li> <li>• Power Components (R)</li> <li>• UPS Requirements (R)</li> <li>• UPS PBX 1 Load Capacity (R)</li> <li>• Backup Power (Environmental) (R)</li> <li>• Alarms (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.11.2</li> <li>• UCR Section 5.2.11.3</li> <li>• UCR Section 5.2.11.3.1</li> <li>• UCR Section 5.2.11.3.2</li> <li>• UCR Section 5.2.11.3.2.1</li> <li>• UCR Section 5.2.11.3.3</li> <li>• UCR Section 5.2.11.3.4</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 3</li> </ul>

**Table 2. PBX 1 Requirements (continued)**

<b>VoIP</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
VoIP System (See note 1.)	No	VoIP function is conditional. If VoIP is provided, all of the following requirements must be met: <ul style="list-style-type: none"> <li>• Voice Quality with MOS of 4.0 or better (R)</li> <li>• ITU-T G.711 PCM CODEC (R)</li> <li>• MLPP (R)</li> <li>• Security (R)</li> <li>• Network management (C)</li> <li>• System timing (R)</li> <li>• Latency ≤ 60 milliseconds (R)</li> <li>• IPv6 capable (R)</li> <li>• Service Class Tagging (R)</li> <li>• Softphone Requirements (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR 2008, Change 2, section 5.2.12.8.2.1</li> <li>• UCR section 5.2.12.8.2.2</li> <li>• UCR section 5.2.12.8.2.3</li> <li>• UCR section 5.2.12.8.2.4</li> <li>• UCR section 5.2.12.8.2.5</li> <li>• UCR section 5.2.12.8.2.6</li> <li>• UCR section 5.2.12.8.2.7</li> <li>• UCR 2008, Change 2, section 5.3.5</li> <li>• UCR section 5.2.12.8.2.9</li> <li>• UCR 2008, Change 2, section 5.3.2.6.1.7</li> </ul>
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
PSTN (See note 2.)	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> <li>• Ground Start Line (R)</li> <li>• Immediate Start (C)</li> <li>• Delay Dial (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.4.2.2</li> <li>• UCR Section 5.2.4.3.2</li> <li>• UCR Section 5.2.4.3.4</li> </ul>
<b>NOTES:</b> 1 All requirements are derived from the UCR 2008, Reference (c) with the exception of the voice quality, IPv6, and softphone requirements, because TDM requirements were not included in the UCR 2008, Change 2. The voice quality, IPv6, and softphone requirements are derived from the UCR 2008, Change 2, Reference (d). 2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.				

**Table 2. PBX 1 Requirements (continued)**

<b>LEGEND:</b>					
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR	Federal Telecommunications Recommendation	PCM-24	Pulse Code Modulation - 24 Channels
ANSI	American National Standards Institute	FTR 1080B G.711	Video Teleconferencing Services PCM of voice frequencies	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio	GR	Generic Requirement	PRI	Primary Rate Interface
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS)	PSTN	Public Switched Telephone Network
C	Conditional		Security	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling	H.320	Standard for Narrowband VTC	R	Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IEEE	Institute of Electrical and Electronics Engineers	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	IP	Internet Protocol	SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	IPv6	Internet Protocol version 6	STE	Secure Terminal Equipment
DISA	Defense Information Systems Agency	ISDN	Integrated Services Digital Network	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry	ITU-T	International Telecommunication Union- Telecommunication Standardization Sector	STU-III	Secure Telephone Unit -3rd generation
DoD	Department of Defense	kbps	kilobits per second	T.4	Standardization of Group 3 facsimile terminals for document transmission
DoDI	Department of Defense Instruction	Mbps	Megabits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DP	Dial Pulse	MFR1	Multi-Frequency Recommendation 1	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption	TDM	Time Division Multiplexing
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MOS	Mean Opinion Score	UCR	Unified Capabilities Requirements
DSN	Defense Switched Network	NI 1/2	National ISDN Standard 1 or 2	UPS	Uninterruptible Power Supply
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps	VBD	Variable bit data
E&M	Ear and Mouth	NX64	Data format restricted to multiples of 64 kbps	VoIP	Voice over Internet Protocol
E1	European Basic Multiplex Rate (2.048 Mbps)	PBX	Private Branch Exchange	VTC	Video Teleconferencing
EKTS	Electronic Key Telephone System	PBX 1 PCM	Private Branch Exchange 1 Pulse Code Modulation		

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: [ucco@disa.mil](mailto:ucco@disa.mil).

JITC Memo, JTE, Special Interoperability Test Certification of Cisco Unified Communication Manager Express Version 8.0 with Internetwork Operating System (IOS) Software Release 15.1(1) T3

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.mellon@disa.mil](mailto:edward.mellon@disa.mil). The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 1008202.

FOR THE COMMANDER:

2 Enclosures a/s

  
for BRADLEY A. CLARK  
Chief  
Battlespace Communications Portfolio

Distribution (electronic mail):

Joint Staff J-6

Joint Interoperability Test Command, Liaison, TE3/JT1

Office of Chief of Naval Operations, CNO N6F2

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

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

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

DOT&E, Net-Centric Systems and Naval Warfare

U.S. Coast Guard, CG-64

Defense Intelligence Agency

National Security Agency, DT

Defense Information Systems Agency, TEMC

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

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

Defense Information Systems Agency, GS23

## ADDITIONAL REFERENCES

- (c) Office of the Assistant Secretary of Defense, “Department of Defense Unified Capabilities Requirements 2008 Change 1,” 22 January 2010
- (d) Office of the Assistant Secretary of Defense, “Department of Defense Unified Capabilities Requirements 2008 Change 2,” 31 December 2010
- (e) Joint Interoperability Test Command, “Defense Switched Network Generic Switch Test Plan (GSTP), Change 2,” 2 October 2006
- (f) Joint Interoperability Test Command, “Information Assurance (IA) Assessment of Cisco Unified Communications Manager Express (CUCME) on Integrated Service Routers (ISRs) with Internetwork Operating System (IOS) 15.1(1)T3 (Tracking Number 1008202),” **Draft**
- (g) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, “Policy for Department of Defense Voice Services with Real Time Services (RTS),” 9 November 2007

## CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** Cisco Unified Communications Manager Express Version 8.0 with Internetwork Operating System (IOS) Software Release 15.1(1) T3; hereinafter referred to as the System Under Test (SUT).
- 2. PROPONENT.** United States Army Information Systems Engineering Command.
- 3. PROGRAM MANAGER.** Frank Cobb, 1435 Porter St, Frederick, Maryland, 21702  
e-mail: frank.cobb@us.army.mil.
- 4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.
- 5. SYSTEM UNDER TEST DESCRIPTION.** The SUT is a Private Branch Exchange (PBX) 1 solution that allows call processing and signaling to take place on Integrated Service Router (ISR) and Integrated Service Router Generation 2 (ISR G2) platform. The SUT supports American National Standards Institute (ANSI) T1.619a Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2). The SUT supports two types of user end instrument connection types, IP and analog devices. In addition to performing call processing, this solution is also capable of functioning as a voice gateway by hosting T1 ISDN PRI NI 1/2 connections to the PSTN, other PBX products, and to an ASLAN on a single platform. The SUT consists of one ISR or ISR G2s in addition to phones. This solution can connect to an ASLAN as well as PSTN network using modules and interfaces available for the ISR and ISR G2 products. The SUT was tested with the 3845 and 3945 ISRs.

The 3845 supports up to 24 T1 trunks or 88 FXS ports and up to 250 IP end instruments. The 3825, 2801, 2811, 2821, and 2851 ISRs were not tested; however, they utilize the same software and similar hardware as the 3845. JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use. The 3825 supports up to 14 T1 trunks or 52 FXS ports and up to 175 IP end instruments. The 2801 supports 1 T1 trunk or 16 FXS ports and up to 25 IP end instruments. The 2811 supports up to 3 T1 trunks or 28 FXS ports and up to 35 IP end instruments. The 2821 supports up to 4 T1 trunks or 52 FXS ports and up to 50 IP end instruments. The 2851 supports up to 7 T1 trunks or 52 FXS ports and up to 100 IP end instruments.

The 3945 supports up to 24 T1 trunks or 112 FXS ports and up to 350 IP end instruments. The 3925, 3925E, 3945E, 2901, 2911, 2921, and 2951 ISRs were not tested; however, they utilize the same software and similar hardware as the 3945. JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use. The 3925 supports up to 20 T1 trunks or 64 FXS ports and up to 250 IP end instruments. The 3925E supports up to 20 T1 trunks or 64 FXS ports and up to 400 IP end instruments. The 3945E supports up to 24 T1 trunks or 112 FXS ports and up to 450 IP end instruments. The 2901 supports up to

4 T1 trunks or 16 FXS ports and up to 35 IP end instruments. The 2911 supports up to 6 T1 trunks or 40 FXS ports and up to 50 IP end instruments. The 2921 supports up to 10 T1 trunks or 40 FXS ports and up to 100 IP end instruments. The 2951 supports up to 16 T1 trunks or 64 FXS ports and up to 150 IP end instruments.

The subcomponents listed below, which are bolded and underlined, were tested by JITC in both the 3845 and 3945 ISRs. The other subcomponents 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.

**NM-HDV2-2T1/E1**, NM-HDV2-1T1/E1, NM-HDV2. These are High Density Voice Network Modules, 2nd generation with 2 T1 RJ-48 ports. These components are certified in the DISN with the T1 ISDN PRI ANSI T1.619a interface. These components are certified in the PSTN with the T1 ISDN PRI (ANSI T1.607) interface. The E1 interfaces were not tested and are not covered under this certification.

**VWIC2-2MFT-T1/E1**, VWIC2-1MFT-T1/E1. These are voice/WAN Interface Cards, 2nd generation with two T1 RJ-48 Multiflex Trunk ports. These components are certified in the DISN with the T1 ISDN PRI ANSI T1.619a interface. These components are certified in the PSTN with the T1 ISDN PRI (ANSI T1.607) interface. The E1 interfaces were not tested and are not covered under this certification.

**VIC3 4FXS/DID**, VIC3 2FXS/DID. These are Voice Interface Cards, 3<sup>rd</sup> generation with 4 RJ-11 Foreign Exchange Station/Direct Inward Dial ports.

**EVM-HD-8FXS/DID**. This is a High Density voice/fax extension module with 8 Foreign Exchange Station/Direct Inward Dial ports.

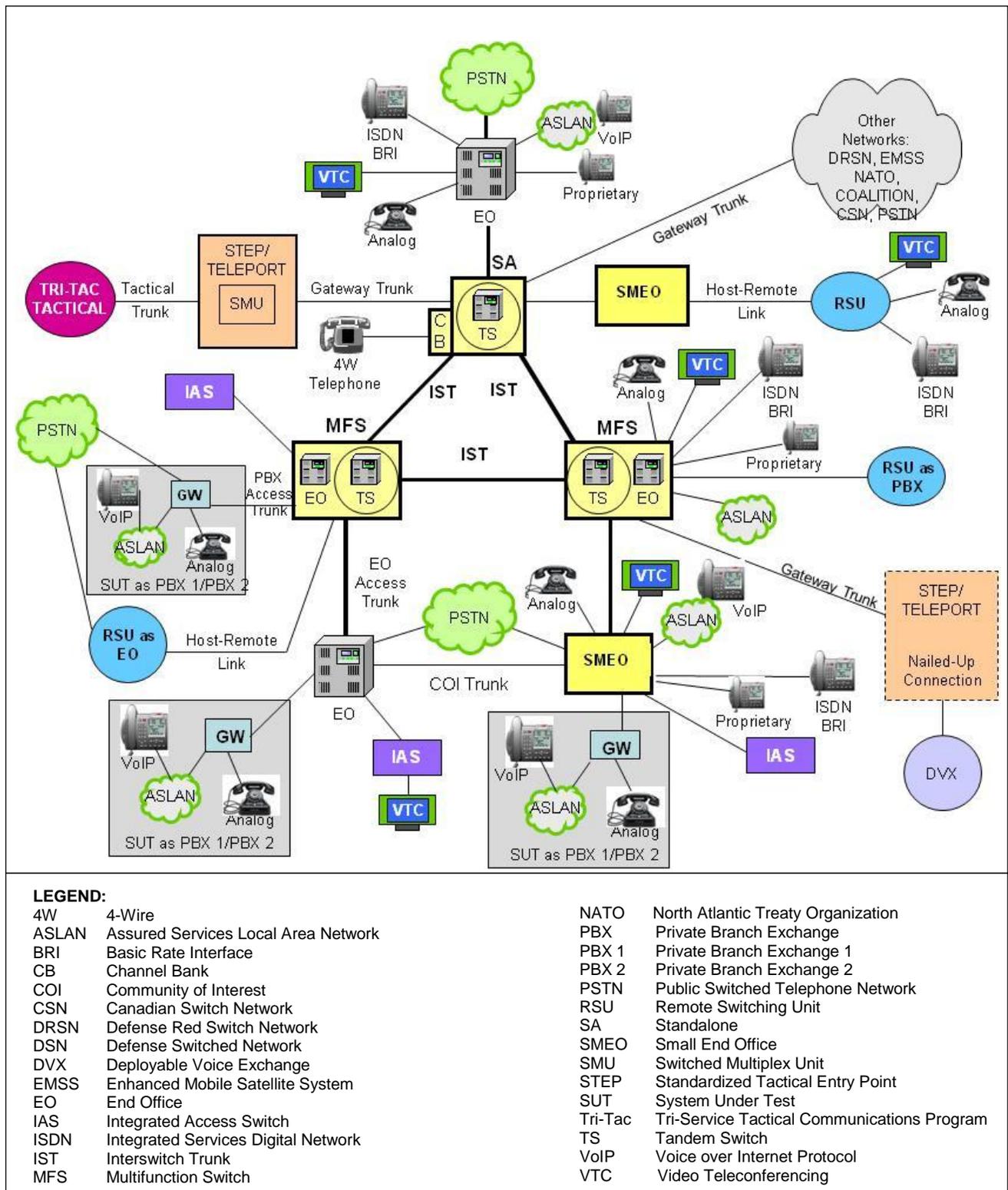
**EM3-HDA-8FXS/DID**. This is a voice/fax expansion module with 8 Foreign Exchange Station/Direct Inward Dial ports.

**PVDM2-64**, PVDM2-32, PVDM2-16, PVD2-8. This is a 64-channel packet fax and voice DSP module, 2<sup>nd</sup> generation. The 3800 and 3900 series of Integrated Service Routers (Gateways) are certified with the Packet Voice Digital Signal Processor Module 2 (PVDM2). Initial testing of the Packet Voice Digital Signal Processor Module 3 (PVDM3) in the 3945 Gateways showed excessive one-way latency. Testing on all gateways was completed with the PVDM2. The PVDM3 is not certified for use with the SUT.

Management of the SUT is through a site-provided, Secure Technical Implementation Guide (STIG)-compliant Personal Computer (PC), with Windows Experience (XP) Service Pack (SP)3 installed.

The Cisco IP Communicator (CIPC) is a softphone application installed on a site-provided PC with the Windows XP operating system platform. The Cisco IP Communicator is not certified on other operating system platforms.

**6. OPERATIONAL ARCHITECTURE.** The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.



**Figure 2-1. DSN Architecture**

**7. REQUIRED SYSTEM INTERFACES.** Requirements specific to PBX 1s are listed in Table 2-1. These requirements are derived from:

a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)", Reference (g).

b. UCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC), Reference (c).

c. UCR PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC, Reference (c).

d. The IPv6 and softphone requirements specified in Reference (d).

**Table 2-1. PBX 1 Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional	References	
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> <li>• PBX Line (C)</li> <li>• Direct Inward Dialing (C)</li> <li>• National ISDN 1/2 Primary Access (R)</li> <li>• ISDN ANSI MLPP Service Capability (R)</li> <li>• ITU-T ISDN Primary Access (Europe only) (C)</li> <li>• ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C)</li> <li>• Normal Wink Start Operations (C)</li> <li>• Glare Operation (R)</li> <li>• Abnormal Wink Start (R)</li> <li>• Glare Resolution (R)</li> <li>• Call for Service Timing (C)</li> <li>• Guard Timing (R)</li> <li>• Satellite Interface (C)</li> <li>• Disconnect Control (C)</li> <li>• Reselect and Retrial (C)</li> <li>• Off-Hook Supervision Transition (C)</li> <li>• Dial-Pulse Signals (C)</li> <li>• DTMF Signaling (C)</li> <li>• Standard Digit Format for Precedence (C)</li> <li>• MFR1 2/6 Signaling (C)</li> <li>• Alerting Signals and Tones (R)</li> <li>• DSN ISDN User-to-Network Signaling (R)</li> <li>• Application (R)</li> <li>• Physical Layer (R)</li> <li>• Data Link Layer (R)</li> <li>• Data Link Connection (R)</li> <li>• Peer-to-Peer Procedures of Data-Link Layer (R)</li> <li>• Layer 3 DSN User-to-Network Signaling (R)</li> <li>• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)</li> <li>• Sequence of Messages for DSN Circuit-Switched Calls (R)</li> <li>• Message Functional Definition and Content (R)</li> <li>• General Message Format and Information Elements Coding (R)</li> <li>• Supplementary Services (C)</li> <li>• PCM-24 Digital Trunk Interface (R)</li> <li>• Interface Characteristics (R)</li> <li>• Supervisory Channel Associated Signaling (C)</li> <li>• Clear Channel Capability (R)</li> <li>• Alarm and Restoral Requirements (R)</li> <li>• PCM-30 Digital Trunk Interface (Europe only) (R)</li> <li>• Interoperation of PCM-24 and PCM-30 (C)</li> <li>• Analog Trunk Interface (C)</li> <li>• Integrated Digital Loop Carrier (C)</li> <li>• Trunk Group-Remove from Service (C)</li> <li>• Trunk Group-Restore to Service (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.3.1</li> <li>• UCR Section 5.2.1.3.2</li> <li>• UCR Section 5.2.1.3.4.1</li> <li>• UCR Section 5.2.1.3.4.1.1</li> <li>• UCR Section 5.2.1.3.4.2</li> <li>• UCR Section 5.2.1.3.4.2.1</li> <li>• UCR Section 5.2.4.3.3.1.1</li> <li>• UCR Section 5.2.4.3.3.1.2</li> <li>• UCR Section 5.2.4.3.3.2.1</li> <li>• UCR Section 5.2.4.3.3.2.2</li> <li>• UCR Section 5.2.4.3.5</li> <li>• UCR Section 5.2.4.3.6</li> <li>• UCR Section 5.2.3.4.7</li> <li>• UCR Section 5.2.3.4.8</li> <li>• UCR Section 5.2.3.4.9</li> <li>• UCR Section 5.2.3.4.10</li> <li>• UCR Section 5.2.4.4.1</li> <li>• UCR Section 5.2.4.4.2</li> <li>• UCR Section 5.2.4.4.2.1</li> <li>• UCR Section 5.2.4.4.3</li> <li>• UCR Section 5.2.4.5.1</li> <li>• UCR Section 5.2.4.7.1.4.2</li> <li>• UCR Section 5.2.4.7.1.1</li> <li>• UCR Section 5.2.4.7.1.2</li> <li>• UCR Section 5.2.4.7.1.3</li> <li>• UCR Section 5.2.4.7.1.3.1</li> <li>• UCR Section 5.2.4.7.1.3.2</li> <li>• UCR Section 5.2.4.7.1.4</li> <li>• UCR Section 5.2.4.7.1.4.2</li> <li>• UCR Section 5.2.4.7.1.4.3</li> <li>• UCR Section 5.2.4.7.1.4.4</li> <li>• UCR Section 5.2.4.7.1.4.5</li> <li>• UCR Section 5.2.4.7.1.4.6</li> <li>• UCR Section 5.2.6.1</li> <li>• UCR Section 5.2.6.1.1</li> <li>• UCR Section 5.2.6.1.2</li> <li>• UCR Section 5.2.6.1.3</li> <li>• UCR Section 5.2.6.1.4</li> <li>• UCR Section 5.2.6.2</li> <li>• UCR Section 5.2.6.3</li> <li>• UCR Section 5.2.6.4</li> <li>• UCR Section 5.2.6.5</li> <li>• UCR Section 5.2.1.5.5</li> <li>• UCR Section 5.2.1.5.5</li> </ul>
E1 CAS (MFR1, DTMF, DP)	No (Europe only)			
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes			
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)			

**Table 2-1. PBX 1 Requirements (continued)**

<b>DSN Trunk Interfaces (continued)</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
T1 CAS (MFR1, DTMF, DP)	No	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>	
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>	
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• CJCSI 6215.01C</li> </ul>	
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• FTR 1080B-2002</li> </ul>	
<b>DSN Line Interfaces</b>					
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• Directory Number Identification (R)</li> <li>• Analog Line (R)</li> <li>• National ISDN 1/2 Basic Access (R: BRI Only)</li> <li>• Basic Line Test Capabilities (R)</li> <li>• Advanced Line Test Capabilities (C)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Reverse Battery (R: 2-Wire Analog only)</li> <li>• Alerting Signals and Tones (R)</li> <li>• S/T Reference Point (R: ISDN BRI only)</li> <li>• VoIP System Requirements (R: VoIP Phones only)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.1.1</li> <li>• UCR Section 5.2.1.3.5</li> <li>• UCR Section 5.2.1.3.3</li> <li>• UCR Section 5.2.1.5.4.1.1</li> <li>• UCR Section 5.2.1.5.4.1.1</li> <li>• UCR Section 5.2.4.2.1</li> <li>• UCR Section 5.2.4.3.1</li> <li>• UCR Section 5.2.4.5.1</li> <li>• UCR Section 5.2.4.7.1.2.1</li> <li>• UCR Section 5.2.12.8</li> </ul>	
ISDN BRI NI 1/2 (ANSI T1.619a)	No				
2-Wire Proprietary Digital	No				
VoIP (Ethernet IEEE 802.3u)	No		Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure Calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>
			Facsimile	<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
			Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R: 2-Wire Analog only)</li> <li>• Secure data (STE/STU-III) (R: 2-Wire Analog only)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> </ul>
		VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: BRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• FTR 1080B-2002</li> </ul>	
<b>DSN Features &amp; Capabilities</b>					
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
Common Features	Yes	<ul style="list-style-type: none"> <li>• Individual Lines (R)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (R)</li> <li>• Three-way calling (R)</li> <li>• Add-on transfer, conference calling, and call hold (C)</li> <li>• Call Transfer Individual – All calls (R)</li> <li>• Call Transfer - Internal Only (R)</li> <li>• Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)</li> <li>• Call Transfer – Outside (R)</li> <li>• Call Transfer – Add-On Restricted Station (C)</li> <li>• Call Transfer – Attendant (C)</li> <li>• Call Hold (R)</li> <li>• Conference Calling – Six Way Station Controlled (C)</li> <li>• Call Forwarding Variable (R)</li> <li>• Call Forward Busy Line (R)</li> <li>• Call Forwarding – Don't Answer – All Calls (R)</li> <li>• Selective Call Forwarding (C)</li> <li>• Call pick-up (C)</li> <li>• Address Translation (C)</li> <li>• Assured Dial Tone (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.1.1</li> <li>• UCR Section 5.2.1.1.3</li> <li>• UCR Section 5.2.1.1.4</li> <li>• UCR Section 5.2.1.1.5.1</li> <li>• UCR Section 5.2.1.1.6</li> <li>• UCR Section 5.2.1.1.7</li> <li>• UCR Section 5.2.1.1.7.1</li> <li>• UCR Section 5.2.1.1.7.2</li> <li>• UCR Section 5.2.1.1.7.3</li> <li>• UCR Section 5.2.1.1.7.4</li> <li>• UCR Section 5.2.1.1.7.5</li> <li>• UCR Section 5.2.1.1.7.6</li> <li>• UCR Section 5.2.1.1.7.7</li> <li>• UCR Section 5.2.1.1.7.8</li> <li>• UCR Section 5.2.1.1.8.1</li> <li>• UCR Section 5.2.1.1.8.2</li> <li>• UCR Section 5.2.1.1.8.3</li> <li>• UCR Section 5.2.1.1.8.4</li> <li>• UCR Section 5.2.1.1.9.1</li> <li>• UCR Section 5.2.1.7</li> <li>• UCR Section 5.2.1.9</li> </ul>	
Attendant	No	<ul style="list-style-type: none"> <li>• Attendant Features (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.2.2</li> </ul>	

**Table 2-1. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Public Safety	Yes	<ul style="list-style-type: none"> <li>• Emergency Service Basic (911) Caller (R)</li> <li>• Emergency Service (911) Public Safety Answering Service (C)</li> <li>• Enhanced Emergency Service (E911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing call trace (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.4.1.1</li> <li>• UCR Section 5.2.1.4.1.2</li> <li>• UCR Section 5.2.1.4.1.3</li> <li>• UCR Section 5.2.1.4.2</li> <li>• UCR Section 5.2.1.4.3</li> </ul>
Conferencing	No	<ul style="list-style-type: none"> <li>• Preset Conferencing (C)</li> <li>• Meet-Me Conferencing (C)</li> <li>• Progressive Conferencing (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.6.1</li> <li>• UCR Section 5.2.1.6.2</li> <li>• UCR Section 5.2.1.6.3</li> </ul>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Nailed-Up Connections (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.8</li> </ul>
DSN Hotline Services	No	<ul style="list-style-type: none"> <li>• DSN Analog Hotline Service (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.12</li> </ul>
MLPP	Yes	<ul style="list-style-type: none"> <li>• MLPP Overview (R)</li> <li>• Preemption in the Network (R)</li> <li>• Network Facility with Lower Precedence Calls (R)</li> <li>• Network Facility with Equal or Higher Precedence Calls (R)</li> <li>• Precedence Call Diversion (R)</li> <li>• Channel Associated Signaling (C)</li> <li>• Primary Rate Interface (R)</li> <li>• Analog Line MLPP (R)</li> <li>• ISDN MLPP Basic Rate Interface (C)</li> <li>• ISDN Primary Rate Interface (R)</li> <li>• Precedence Call Waiting (R)</li> <li>• Call Forwarding (R)</li> <li>• Call Transfer (R)</li> <li>• Call Hold (R)</li> <li>• Three-Way Calling (R)</li> <li>• Call Pickup (C)</li> <li>• Conferencing (C)</li> <li>• Multiline Hunt Group (C)</li> <li>• Community of Interest (C)</li> <li>• MLPP Interaction with EKTS features (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.2.1.1</li> <li>• UCR Section 5.2.2.2</li> <li>• UCR Section 5.2.2.2.1</li> <li>• UCR Section 5.2.2.2.2</li> <li>• UCR Section 5.2.2.3</li> <li>• UCR Section 5.2.2.4.1</li> <li>• UCR Section 5.2.2.4.2</li> <li>• UCR Section 5.2.2.5</li> <li>• UCR Section 5.2.2.6</li> <li>• UCR Section 5.2.2.7</li> <li>• UCR Section 5.2.2.8.1</li> <li>• UCR Section 5.2.2.8.2</li> <li>• UCR Section 5.2.2.8.3</li> <li>• UCR Section 5.2.2.8.4</li> <li>• UCR Section 5.2.2.8.5</li> <li>• UCR Section 5.2.2.8.6</li> <li>• UCR Section 5.2.2.8.7.1</li> <li>• UCR Section 5.2.2.8.8</li> <li>• UCR Section 5.2.2.8.9</li> <li>• UCR Section 5.2.2.10.1</li> </ul>

**Table 2-1. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities (continued)</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Call Processing	Yes	<ul style="list-style-type: none"> <li>• Call Treatments (R)</li> <li>• Primary and Alternate Routing (R)</li> <li>• E&amp;M Lead Signaling States (C)</li> <li>• 4-Wire Analog User Access Lines (C)</li> <li>• 2-Wire User Access Lines (R)</li> <li>• Termination of Analog Lines (R)</li> <li>• DSN User Dialing (R)</li> <li>• Interswitch and Intraswitch Dialing (R)</li> <li>• Seven-Digit Dialing (R)</li> <li>• Ten-Digit Dialing (R)</li> <li>• Access Code (R)</li> <li>• Access Digit (R)</li> <li>• Precedence Digit (R)</li> <li>• Service Digit (R)</li> <li>• Route Code (R)</li> <li>• Area Code (R)</li> <li>• Switch Code (R)</li> <li>• Line Number (R)</li> <li>• Calling Name Delivery (C)</li> <li>• Calling Number Delivery (R)</li> <li>• Emergency Service 911 Conflict Resolution (R)</li> <li>• DSN Switch Outpulsing Digit Formats (C)</li> <li>• Standard Directory Number (C)</li> <li>• Standard Test Numbers (C)</li> <li>• Base Services – Abbreviated Numbers (R)</li> <li>• Digit Reception Requirements (R)</li> <li>• Screening (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.3.1</li> <li>• UCR Section 5.2.3.2</li> <li>• UCR Section 5.2.3.3.1</li> <li>• UCR Section 5.2.3.3.2</li> <li>• UCR Section 5.2.3.3.3</li> <li>• UCR Section 5.2.3.3.4</li> <li>• UCR Section 5.2.3.5.1.1</li> <li>• UCR Section 5.2.3.5.1.1.1</li> <li>• UCR Section 5.3.3.5.2.1</li> <li>• UCR Section 5.2.3.5.2.2</li> <li>• UCR Section 5.2.3.5.1.3</li> <li>• UCR Section 5.2.3.5.1.3.1</li> <li>• UCR Section 5.2.3.5.1.3.2</li> <li>• UCR Section 5.2.3.5.1.3.3</li> <li>• UCR Section 5.2.3.5.1.4</li> <li>• UCR Section 5.2.3.5.1.5</li> <li>• UCR Section 5.2.3.5.1.6</li> <li>• UCR Section 5.2.3.5.1.7</li> <li>• UCR Section 5.2.3.5.1.8.1</li> <li>• UCR Section 5.2.3.5.1.8.2</li> <li>• UCR Section 5.2.3.5.1.9</li> <li>• UCR Section 5.2.3.5.2</li> <li>• UCR Section 5.2.3.5.3</li> <li>• UCR Section 5.2.3.5.4</li> <li>• UCR Section 5.2.3.5.5</li> <li>• UCR Section 5.2.3.5.6</li> <li>• UCR Section 5.2.3.5.8</li> </ul>
ISDN Services	Yes	<ul style="list-style-type: none"> <li>• BRI Access, Call Control and Signaling (R)</li> <li>• Uniform Interface Configuration for BRIs (R)</li> <li>• EKTS (C)</li> <li>• PRI Access, Call Control and Signaling (R)</li> <li>• PRI Features (R)</li> <li>• Packet Data Features and Capabilities (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.9.2, Table 5.2.9-1</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-2</li> <li>• UCR Section 5.2.9.3, Table 5.2.9-3</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-4</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-5</li> <li>• UCR Section 5.2.9.2, Table 5.2.9-6</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> <li>• Synchronization Performance Monitoring Criteria (C)</li> <li>• DS1 Traffic Interfaces (C)</li> <li>• DS0 Traffic Interconnects (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.10.1.1.2</li> <li>• UCR Section 5.2.10.1.1.2.2</li> <li>• UCR Section 5.2.10.2</li> <li>• UCR Section 5.2.10.3</li> <li>• UCR Section 5.2.10.4</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• System Availability (R)</li> <li>• Backup Power (R)</li> <li>• Power Components (R)</li> <li>• UPS Requirements (R)</li> <li>• UPS PBX 1 Load Capacity (R)</li> <li>• Backup Power (Environmental) (R)</li> <li>• Alarms (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.11.2</li> <li>• UCR Section 5.2.11.3</li> <li>• UCR Section 5.2.11.3.1</li> <li>• UCR Section 5.2.11.3.2</li> <li>• UCR Section 5.2.11.3.2.1</li> <li>• UCR Section 5.2.11.3.3</li> <li>• UCR Section 5.2.11.3.4</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 3</li> </ul>

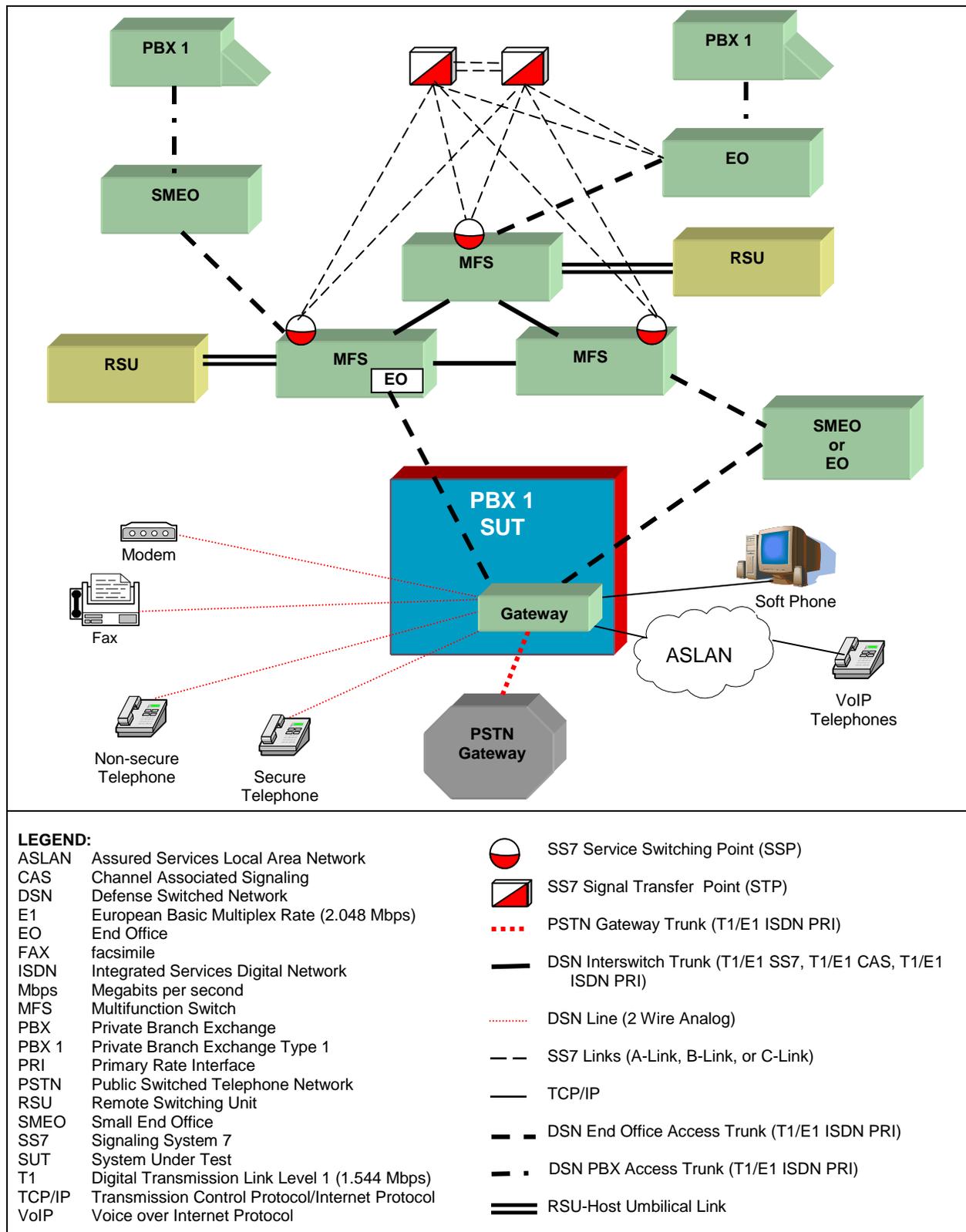
**Table 2-1. PBX 1 Requirements (continued)**

<b>VoIP</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
VoIP System (See note 1.)	No	VoIP function is conditional. If VoIP is provided, <b>all</b> of the following requirements must be met: <ul style="list-style-type: none"> <li>• Voice Quality with MOS of 4.0 or better (R)</li> <li>• ITU-T G.711 PCM CODEC (R)</li> <li>• MLPP (R)</li> <li>• Security (R)</li> <li>• Network management (C)</li> <li>• System timing (R)</li> <li>• Latency ≤ 60 milliseconds (R)</li> <li>• IPv6 capable (R)</li> <li>• Service Class Tagging (R)</li> <li>• Softphone Requirements (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR 2008, Change 2, section 5.2.12.8.2.1</li> <li>• UCR section 5.2.12.8.2.2</li> <li>• UCR section 5.2.12.8.2.3</li> <li>• UCR section 5.2.12.8.2.4</li> <li>• UCR section 5.2.12.8.2.5</li> <li>• UCR section 5.2.12.8.2.6</li> <li>• UCR section 5.2.12.8.2.7</li> <li>• UCR 2008, Change 2, section 5.3.5</li> <li>• UCR section 5.2.12.8.2.9</li> <li>• UCR 2008, Change 2, section 5.3.2.6.1.7</li> </ul>
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
PSTN (See note 2.)	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> <li>• Ground Start Line (R)</li> <li>• Immediate Start (C)</li> <li>• Delay Dial (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.4.2.2</li> <li>• UCR Section 5.2.4.3.2</li> <li>• UCR Section 5.2.4.3.4</li> </ul>
<b>NOTES:</b>				
1 All requirements are derived from the UCR 2008, Reference (c) with the exception of the voice quality, IPv6, and softphone requirements, because TDM requirements were not included in the UCR 2008, Change 2. The voice quality, IPv6, and softphone requirements are derived from the UCR 2008, Change 2, Reference (d).				
2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.				

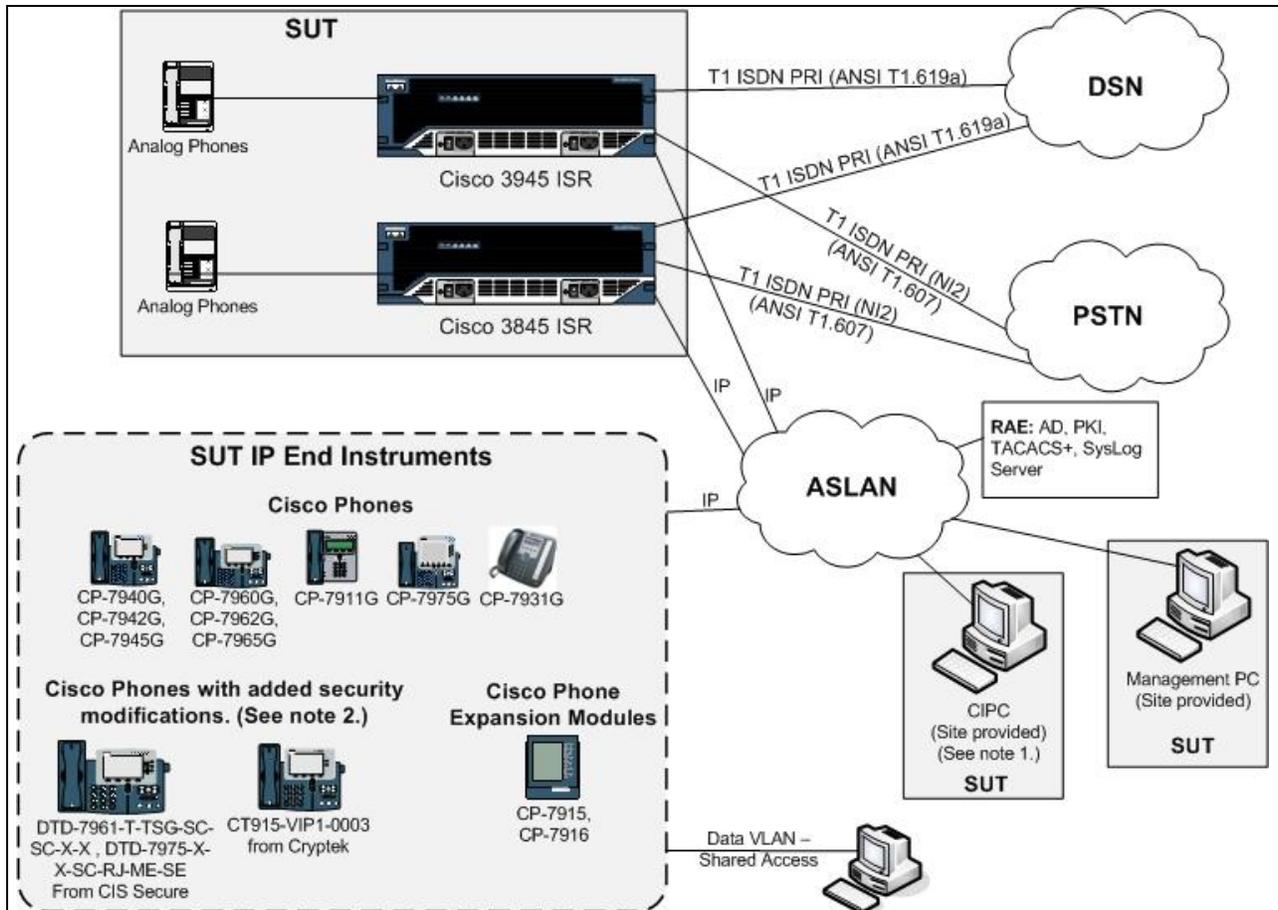
**Table 2-1. PBX 1 Requirements (continued)**

<b>LEGEND:</b>					
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR	Federal Telecommunications Recommendation	PCM-30	Pulse Code Modulation - 30 Channels
		FTR 1080B	Video Teleconferencing Services	PRI	Primary Rate Interface
		G.711	PCM of voice frequencies	PSTN	Public Switched Telephone Network
ANSI	American National Standards Institute	GR	Generic Requirement	Q.955.3	ISDN Signaling Standard for E1 MLPP
		GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	R	Required
BER	Bit Error Ratio			S/T	ISDN BRI four-wire interface
BRI	Basic Rate Interface	H.320	Standard for Narrowband VTC	SS7	Signaling System 7
C	Conditional	IEEE	Institute of Electrical and Electronics Engineers	STE	Secure Terminal Equipment
CAS	Channel Associated Signaling	IP	Internet Protocol	STIGs	Security Technical Implementation Guides
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IPv6	Internet Protocol version 6	STU-III	Secure Telephone Unit - 3rd generation
CODEC	Coder/Decoder	ISDN	Integrated Services Digital Network	T.4	Standardization of Group 3 facsimile terminals for document transmission
DIACAP	DoD Information Assurance Certification and Accreditation Process	IT	Information Technology	T1	Digital Transmission Link Level 1 (1.544 Mbps)
		ITU-T	International Telecommunication Union- Telecommunication Standardization Sector	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DISA	Defense Information Systems Agency	kbps	kilobits per second	TDM	Time Division Multiplexing
DISR	DoD IT Standards Registry	Mbps	Megabits per second	UCR	Unified Capabilities Requirements
DoD	Department of Defense	MFR1	Multi-Frequency Recommendation 1	UPS	Uninterruptible Power Supply
DoDI	Department of Defense Instruction	MLPP	Multi-Level Precedence and Preemption	VBD	Variable bit data
DP	Dial Pulse	MOS	Mean Opinion Score	VoIP	Voice over Internet Protocol
DS0	Digital Signal Level 0 (64 kbps)	NI 1/2	National ISDN Standard 1 or 2	VTC	Video Teleconferencing
		NX56	Data format restricted to multiples of 56 kbps		
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	NX64	Data format restricted to multiples of 64 kbps		
DSN	Defense Switched Network	PBX	Private Branch Exchange		
DTMF	Dual Tone Multi-Frequency	PBX 1	Private Branch Exchange 1		
E&M	Ear and Mouth	PCM	Pulse Code Modulation		
E1	European Basic Multiplex Rate (2.048 Mbps)	PCM-24	Pulse Code Modulation - 24 Channels		
EKTS	Electronic Key Telephone System				

**8. TEST NETWORK DESCRIPTION.** The SUT was tested at JITC's Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system's required functions and features was conducted using the notional test configuration depicted in Figure 2-2. The SUT test configuration with an Assured Services Local Area Network (ASLAN) is depicted in Figure 2-3. The SUT was tested as the end-point in relation to the other switches.



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



**NOTES:**

- 1 The Cisco IP Communicator was tested and certified on the Windows XP operating system platform. The Cisco IP Communicator is not certified on other operating system platforms.
- 2 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.

**LEGEND:**

AD	Active Directory	PRI	Primary Rate Interface
ANSI	American National Standards Institute	PSTN	Public Switched Telephone Network
ASLAN	Assured Services Local Area Network	SUT	System Under Test
CIPC	Cisco IP Communicator	SysLog	System Log
DSN	Defense Switch Network	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IP	Internet Protocol	TACACS+	Terminal Access Controller Access-Control System Plus
ISDN	Integrated Services Digital Network	VLAN	Virtual Local Area Network
ISR	Integrated Services Router	XP	Experience
Mbps	Megabits per second		
PKI	Public Key Infrastructure		

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

**9. SYSTEM CONFIGURATIONS.** Table 2-2 provides the system configurations, hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

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

System Name		Software Release	
Avaya CS2100		Succession Enterprise (SE) 09.1	
Siemens EWSD		19d with Patch Set 46	
Avaya S8720		Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)	
Required Ancillary Equipment		Active Directory	
		Public Key Infrastructure	
		SysLog Server	
		Terminal Access Controller Access Control System Plus	
Management PC		STIG-compliant PC with Windows Experience (XP) Service Pack (SP)3	
<b>Cisco Unified Communications Manager Version 8.0, with IOS Software Release 15.1(1) T3</b>			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
<b>Cisco 3845</b> , 3825, 2801, 2811, 2821, 2851 Integrated Services Router	IOS 15.1(1) T3	<b><u>NM-HDV2-2T1/E1</u></b> NM-HDV2-1T1/E1 NM-HDV2	High Density Voice Network Module, 2nd generation with 2 T1 RJ-48 ports (See note 2.)
		<b><u>VVIC2-2MFT-T1/E1</u></b> VVIC2-1MFT-T1/E1	Voice/WAN Interface Card, 2nd generation with two T1 RJ-48 Multiflex Trunk ports (See note 2.)
		<b><u>VIC3 4FXS/DID</u></b> VIC3 2FXS/DID	Voice Interface Card, 3 <sup>rd</sup> generation with 4 RJ-11 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>EVM-HD-8FXS/DID</u></b>	High Density voice/fax extension module with 8 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>EM3-HDA-8FXS/DID</u></b>	Voice/fax expansion module with 8 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>PVDM2-64</u></b> PVDM2-32 PVDM2-16 PVDM2-8	64-channel packet fax and voice DSP module, 2 <sup>nd</sup> generation (See note 3.)
<b>Cisco 3945</b> , 3925, 2901, 2911, 2921, 2951, 3925E, 3945E Integrated Services Router Generation 2	IOS 15.1(1) T3	<b><u>NM-HDV2-2T1/E1</u></b> NM-HDV2-1T1/E1 NM-HDV2	High Density Voice Network Module, 2 <sup>nd</sup> generation with 2 T1 RJ-48 ports (See note 2.)
		<b><u>VVIC2 2MFT T1/E1</u></b> VM-HDV2-1T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		<b><u>VIC3 4FXS/DID</u></b> VIC3 2FXS/DID	Voice Interface Card, 3rd generation with 4 RJ-11 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>EVM-HD-8FXS/DID</u></b>	High Density Voice/fax extension module with 8 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>EM3-HDA-8FXS/DID</u></b>	Voice/fax expansion module with 8 Foreign Exchange Station/Direct Inward Dial ports
		<b><u>PVDM2-64</u></b> PVDM2-32 PVDM2-16 PVDM2-8	64-channel packet fax and voice DSP module, 2 <sup>nd</sup> generation (See note 3.)

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

<b>Cisco Unified Communications Manager Express Version 8.0 with IOS Software Release 15.1(1) T3 (continued)</b>			
<b>Component (See note 1.)</b>	<b>Release</b>	<b>Sub-component (See note 1.)</b>	<b>Function</b>
<b><u>CP-7940G and CP-7960G</u></b> (See note 4.)	P00308000500	Not Applicable	IP Phone, 10/100MB Ethernet, No Shared Access
<b><u>CP-7911</u></b>	SCCP11.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7931G</u></b>	SCCP31.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7942G</u></b>	SCCP42.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7945G</u></b>	SCCP45.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7962G</u></b>	SCCP42.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7965G</u></b>	SCCP45.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>CP-7975G</u></b>	SCCP75.8-5-3S	Not Applicable	IP Phone, 10/100MB Ethernet, Shared Access
<b><u>7915</u></b>	B015-1-0-4	Not Applicable	Expansion module
<b><u>7916</u></b>	B015-1-0-4	Not Applicable	Expansion module
<b><u>CIS Secure DTD-7961-T-SG-SC-SC-X-X</u></b> (See note 5.)	SCCP41.8-5-3S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access
<b><u>CIS Secure DTD-7975-X-X-SC-RJ-ME-SE</u></b>	SCCP75.8-5-3S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
<b><u>CRYPTEK CT915-V-P1-003</u></b>	SCCP41.8-5-3S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and shared access
<b><u>Cisco IP Communicator</u></b>	7.0.5	Not Applicable	Cisco Softphone Application

**NOTES:**

- 1 Components bolded and underlined 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 the T1 ISDN PRI ANSI T1.619a interface. These components are certified in the PSTN with the T1 ISDN PRI (ANSI T1.607) interface. The E1 interfaces were not tested and are not covered under this certification.
- 3 The 3800 and 3900 series of Integrated Service Routers (Gateways) are certified with the Packet Voice Digital Signal Processor Module 2 (PVD2M2). Initial testing of the Packet Voice Digital Signal Processor Module 3 (PVD3M3) in the 3945 Gateways showed excessive one-way latency. Testing on all gateways was completed with the PVD2M2. The PVD3M3 is not certified for use with the SUT.
- 4 The SUT met all IPv6 requirements through testing and LoC with the following exception: 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 (d). This was adjudicated by DISA as having a minor operational impact.
- 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 The Cisco IP Communicator was tested and certified on the Windows XP operating system platform. The Cisco IP Communicator is not certified on other operating system platforms.

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

<b>LEGEND:</b>					
APL	Approved Product List	GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)	PRI	Primary Rate Interface
CP	Cisco Phone	HD	High Density	PSTN	Public Switched Telephone Network
CS	Communication Server	HDA	High Density Analog	RJ	Registered Jack
DID	Direct Inward Dialing	HDX	High Density Exchange	SC	fiber connector (square push-in)
DISA	Defense Information Systems Agency	IOS	Internetwork Operating System	SCCP	Skinny Call Control Protocol
DSN	Defense Switched Network	IP	Internet Protocol	SUT	System Under Test
E1	European Basic Multiplex Rate (2.048 Mbps)	IPv4	Internet Protocol version 4	T1	Digital Transmission Link Level 1 (1.544 Mbps)
EM	Expansion Module	IPv6	Internet Protocol version 6	TDM	Time Division Multiplexing
EVM	Extension Voice Module	ISDN	Integrated Services Digital Network	UC	Unified Capabilities
EWSD	Elektronisches Wählsystem Digital	JITC	Joint Interoperability Test Command	UCR	Unified Capabilities Requirements
Fax	facsimile	LAN	Local Area Network	V	Voice
FXS	Foreign Exchange Station	Mbps	Megabits per second	VE	Voice/Fax Enhanced
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)	MCS	Media Convergence Server	VIC	Voice Interface Card
Gbps	Gigabits per second	MFT	Multiflex Trunk	VWIC	Voice WAN Interface Card
		MOS	Mean Opinion Score	WAN	Wide Area Network
		NM	Network Module		
		PC	Personal Computer		

**10. TESTING LIMITATIONS.** None.

**11. TEST RESULTS**

**a. Discussion**

(1) DSN Trunk Interfaces

(a) The SUT met all critical CRs and FRs for T1 ISDN PRI NI 1/2 ANSI T1.619a interface with one minor exception. The SUT does not support Non Facility Associated Signaling (NFAS) on their T1 ISDN PRI NI2 interface in accordance with the UCR. DISA previously adjudicated this anomaly as having a minor operational impact.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE) and Voice over Internet Protocol (VoIP) DSN line interfaces with the minor exception listed in paragraph 11.a.(5)(a)8.

(3) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions: All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support standard or precedence Call Waiting on their VoIP phones, they do support multiple call appearances, which mitigates this discrepancy. This provides the ability for a user to receive additional calls while active with another call. A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. These discrepancies were adjudicated by DISA as minor on 25 February 2011.

(b) Attendant. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(c) Public Safety. The SUT meets the minimum critical interoperability requirements for Public Safety which is basic emergency service 911 service. This feature allows the user to dial 911 and the SUT then retranslates it to be routed to a Public Safety Answering Point via a trunk or line. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

(d) Conferencing. The SUT does not support Meet-me Conferencing-, Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.

(e) Nailed-up Connections. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(f) Multi-Level Precedence and Preemption (MLPP). Met all critical CRs and FRs with the following minor exception. The SUT does not support the Loss of Command and Control (C2) announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. This anomaly was adjudicated as minor because this announcement would rarely be invoked on a PBX 1.

(g) Call Processing. Met all critical CRs and FRs.

(h) ISDN Services. Met all critical CRs and FRs. The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA previously adjudicated this anomaly as having a minor operational impact.

(i) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

(j) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and met by vendor LoC.

(l) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateway.

The interfaces certified for the PSTN are T1 ISDN PRI NI 1/2 (ANSI T1.607, 2-Wire Analog Ground Start Line (GR-506 CORE)).

(5) VoIP. The SUT is certified with any ASLAN or any combination of certified ASLAN components listed on the UC APL.

(a) VoIP System. The UCR, section 5.2.12.8.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, section 5.2.12.8.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and for intra- and inter-switch calls end-to-end. The SUT meets MOS requirements of 4.0 or better with an average of 4.10 for 80 test calls conducted over 96 hours of active call time. The SUT met this requirement with all VoIP phones except the CP7940G, CP7960G and the CIS Secure DTD-7961-T-TSG-SC-X-X. These VoIP phones did meet the MOS measurement of 4.0 or better using the Sage MOS. This discrepancy was adjudicated by DISA on 25 February 2011 as having a minor operational impact.

2. Codec. In accordance with the UCR, section 5.2.12.8.2.2, the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill was required and was met by the SUT VoIP solution.

3. MLPP. In accordance with the UCR, section 5.2.12.8.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, section 3. All critical MLPP features and functions were met.

4. Security. Security requirements in accordance with the UCR, section 5.2.12.8.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, Reference (f).

5. Network Management (NM). In accordance with the UCR, section 5.2.12.8.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement was covered under a separate certification for the respective ASLANs listed on the UC APL. In accordance with the UCR, section 5.3.8, the switching system NM requirements are not required for a PBX 1 and were not tested.

6. Synchronization. In accordance with the UCR, section 5.2.10.1.1.2, the SUT is required to derive timing with line timing mode and an internal clock of stratum 4 or better. The SUT met this requirement.

7. Latency. The UCR, section 5.2.12.8.2.7, states that one-way system latency for the VoIP system must be 60 milliseconds (ms) or less as averaged over any five-minute period. The latency requirement is measured from IP or analog handset to the egress trunk. The SUT latency measurements were conducted from each phone type supported by the SUT for IPv4 and IPv6 traffic. There were 18, 20-minute interswitch phone calls with a measured latency between 46.11 ms to 59.9 ms, with an average of 56.48 ms which meets this requirement. These measurements do not include the Cisco IP Communicator softphone which is excluded from this requirement.

8. Internet Protocol version 6 (IPv6). In accordance with UCR, section 5.3.5, all systems submitted for testing must be IPv6 capable. Dual Stack solutions are preferred and tunneling solutions are unacceptable. IPv6-capable products, in accordance with UCR, section 5.3.5.3.1, can create or receive, process, and send or forward IPv6 packets in mixed IPv4/v6 environments. IPv6-capable networks can receive, process, and forward IPv6 packets from/to devices within the same network and from/to other networks and systems, where those networks and systems may be operating with only IPv4, only IPv6, or both IPv4 and IPv6. All of the SUT components covered under this certification met the IPv6 criteria through testing and the LoC with the following minor exceptions, which were adjudicated by DISA as having a minor operational impact:

a. 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 end instruments listed in Table 2-2 and a dual stack call control agent in accordance with Reference (d). This was adjudicated by DISA as having a minor operational impact.

b. The dual stack IP end instruments can only be configured to register with IPv4. When configured to register with IPv6 the IP end instruments fail to register. This discrepancy was adjudicated by DISA as having a minor operational impact. All other dual stack IPv6 requirements were met.

9. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system (i.e. Media Gateway and Session Control Agent) shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT meets the requirement. This was met by the SUT.

b. All session control components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. This was met by the SUT.

10. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system end user devices shall meet the following requirements:

a. All end instruments shall be capable of implementing Service Class tagging using the 8-bit Traffic Class in the IPv6 header and DSCP field in the IPv4 header. The SUT end instruments that support IPv6 dual stack used class tagging in the respective IP headers for IPv4 and IPv6, which meets the requirement.

b. All end instrument components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The DSCPs may be assigned by either having the end instrument itself assign the Traffic Class and DSCP tag or having the call control portion of the VoIP system tell the end instrument what value to assign. The SUT can assign any DSCP value from 0-63, which meets this requirement.

c. For VoIP, video, and data end products, any end system that supports convergence must preassign the VLAN using IEEE 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media, the LAN can assign the VLAN based on port-based VLAN assignment. The SUT end instruments have the capability of supporting shared access. Additionally the SUT end instruments have the capability to tag Real Time Traffic with the appropriate VLAN Identifier value. The Cisco VoIP phones that met the critical interoperability requirements for certification with 100 Mbps interface were the: CP7911G, CP7942G, CP7945G, CP7962G, CP7965G, CP7975G, Tempest phone Cryptek 7961G, CRYPTOTEK CT915 and Tempest phone CIS 7975G. The above phones have been tested and are certified for shared access (i.e., same switch port is shared by PC and IP phone) with the exception of the CP7906G. The CP7906G phone does not support shared access. The following phones are also certified for 1 Gbps shared access: CP7975G, CP7965G, CP7945G, and Tempest phone CIS 7975G. All VoIP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

11. The UCR 2008, Change 1, 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, unless explicitly stated here within this paragraph. The softphone application in conjunction with a general-purpose computer, including its mouse (point and click) interaction, shall support, as a minimum, the UCR 2008, Change 1, section 5.3.2 requirements. The softphone is exempt from the Network Infrastructure End-to-End requirements in the UCR 2008, Change 1, section 5.3.3. The softphone shall meet the Information Assurance requirements in the UCR 2008, Change 1, section 5.4. Information Assurance is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).

The Cisco IP Communicator was tested on the Windows XP operating system platform only. The Cisco IP Communicator is only certified on the Windows XP operating system platform. The Cisco IP Communicator software is loaded on a site-provided Secure Technical Implementation Guide (STIG)-compliant PC with Windows XP SP 3, which must meet the following minimum Cisco requirements for the IP Communicator: Pentium P4 1.5 GHz or higher recommended, 100 MB free disk space, 1 GB RAM, a non-ISA full-duplex sound card (integrated or PCI-based) or USB sound Device, a 10/100 Mbps Ethernet network interface card, SVGA video card 800 x 600 x16-bit screen resolution (1024 x 768 x 16-bit or better recommended). If Cisco VPN Client software is installed, version 5.0 or later is required. If Cisco AnyConnect is installed, version 2.5 is required. For connectivity, a 128 kbps or faster network connection is recommended and adding Cisco Unified Video Advantage with a connection of 384 kbps or faster is required.

**b. System Interoperability Results.** The SUT is certified for joint use in the DISN as a PBX 1 and PBX 2 in accordance with the requirements set forth in the UCR. The SUT interoperability test summary is shown in Table 2-4. The SUT Interoperability Requirements/Status is shown in Table 2-5.

**Table 2-4. SUT Interoperability Test Summary**

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS. <sup>1</sup>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT gateway analog interface does not provide a ping ring when the phone is configured with the Call Forward Variable feature. <sup>2</sup> The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. <sup>3</sup> All IP end instruments failed to register with IPv6. <sup>4</sup> CP-7940G, CIS-7961 and CP-7960G end instruments did not meet the Voice Quality E-model MOS; however, they did meet the SAGE MOS of 4.0 or better. <sup>5</sup>

**Table 2-4. SUT Interoperability Test Summary (continued)**

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco IP phones do not support Call Waiting. <sup>6</sup>	
Attendant	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Public Safety	Yes	Certified	All public safety features are conditional. The SUT met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features. <sup>7</sup>	
Conferencing	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Nailed-up Connections	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
DSN Hotline Services	No	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the Loss of Command and Control announcement. <sup>8</sup>	
Call Processing	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services	Yes	Not Tested	The SUT does not support this feature and it is not required for a PBX 1.	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
Reliability	Yes	Certified	Met all critical CRs and FRs.	
Security	Yes	Certified	See note 9.	
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL. (See note 3.)	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface and it is not required for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT T1 ISDN PRI NI/2 interface does not support NFAS. <sup>1</sup>
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	The SUT supports this interface; however, it was not tested and is not required for a PBX 1.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs. <sup>10</sup>
<b>NOTES:</b>				
<p>1 The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA adjudicated this anomaly as having a minor operational impact.</p> <p>2 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.</p> <p>3 The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception: 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 Enclosure 2 and a dual stack call control agent in accordance with Reference (d). This was adjudicated by DISA as having a minor operational impact.</p> <p>4 All dual stack IP end instruments must be configured to register with IPv4 not IPv6. When configured to register with IPv6, the IP end instruments fail to register. This discrepancy was adjudicated by DISA as having a minor operational impact. All other dual stack IPv6 requirements were met.</p> <p>5 The CP-7940G, CIS-7961, and CP-7960G end instruments did not meet the Voice Quality E-model MOS, however they did meet the SAGE MOS of 4.0 or better. This discrepancy was adjudicated by DISA as having a minor operational impact.</p> <p>6 All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.</p>				

**Table 2-4. SUT Interoperability Test Summary (continued)**

**NOTES (continued):**

- 7 The SUT only supports emergency basic 911 service. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.
- 8 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA previously adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1.
- 9 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).
- 10 This interface requirement was met by the vendor's LoC.

**LEGEND:**

802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
ANSI	American National Standards Institute	MCS	Media Convergence Servers
APL	Approved Products List	MFR1	Multi-Frequency Recommendation 1
ASLAN	Assured Services Local Area Network	MLPP	Multi-Level Precedence and Preemption
BRI	Basic Rate Interface	ms	milliseconds
C2	Command and Control	NI 1/2	National ISDN Standard 1 or 2
CAS	Channel Associated Signaling	NI2	National ISDN Standard 2
CoS	Class of Service	NIC	Network Interface Card
CP	Cisco Phone	NFAS	Non Facility Associated Signaling
CRs	Capability Requirements	OAM	Operational Administration and Maintenance
DISA	Defense Information Systems Agency	MOS	Mean Opinion Score
DP	Dial Pulse	PBX 1	Private Branch Exchange 1
DSCP	Differentiated Services Code Point	PMO	Program Management Office
DSN	Defense Switched Network	PNT	Preemption Notification Tone
DSS1	Digital Subscriber Signaling 1	PRI	Primary Rate Interface
DTMF	Dual Tone Multi-Frequency	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.931	Signaling Standard for ISDN
EI	End Instrument	Q.955.3	ISDN Signaling standard for E1 MLPP
FRs	Feature Requirements	RTCP	RTP Control Protocol
GR	Generic Requirement	RTP	Real-time Transport Protocol
GR-506	LSSGR: Signaling for Analog Interfaces	SS7	Signaling System 7
ICA	Isolated Code Announcement	SUT	System Under Test
IEEE	Institute of Electrical and Electronics Engineers	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IP	Internet Protocol	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IPv4	Internet Protocol version 4	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IPv6	Internet Protocol version 6	TCP	Transmission Control Protocol
ISDN	Integrated Services Digital Network	TFTP	Trivial File Transfer Protocol
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UC	Unified Capabilities
JITC	Joint Interoperability Test Command	UCR	Unified Capabilities Requirements
LoC	Letters of Compliance	UDP	User Datagram Protocol
		VoIP	Voice over Internet Protocol

**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/tssj>. 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: [ucco@disa.mil](mailto:ucco@disa.mil).

**Table 2-5. SUT Interoperability Requirements/Status**

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Not Tested
T1 CAS (MFR1, DTMF, DP)	No	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (R)	UCR Section 5.2.4.3.3.1.2	Not Tested	
				Abnormal Wink Start (R)	UCR Section 5.2.4.3.3.2.1	Not Tested	
				Glare Resolution (R)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Not Tested	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Not Tested	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Not Tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Not Tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Not Tested	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Not Tested	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Not Tested	
				Supervisory Channel Associated Signaling (C)	UCR Section 5.2.6.1.2	Not Tested	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Not Tested	
			Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Not Tested		
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
	Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested				

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Not Tested (See note 2.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.1	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (R)	UCR Section 5.2.4.3.3.1.2	Not Tested	
				Abnormal Wink Start (R)	UCR Section 5.2.4.3.3.2.1	Not Tested	
				Glare Resolution (R)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Not Tested	
				Guard Timing (R)	UCR Section 5.2.4.3.6	Not Tested	
				Satellite Timing (C)	UCR Section 5.2.4.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Not Tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.2.4.4.2.1	Not Tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.2.4.4.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				DSN Transmission Interface (R)	UCR Section 5.2.5	Not Tested	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested	
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				National ISDN 1/2 Primary Access (R)	UCR Section 5.2.1.3.4.1	Partially Met	See note 3.
				ISDN ANSI MLPP Service Capability (R)	UCR Section 5.2.1.3.4.1.1	Met	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Met	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Met	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Met	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Application (R)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.2.4.7.1.4.6	Not Tested	See note 4.
				DSN Transmission Interface (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
				Clear Channel Capability (R)	UCR Section 5.2.6.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 5.2.6.1.4	Met	
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Met		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Met		
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
Facsimile	Analog: ITU-T T.4 (R)	DISR	Met				

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
T1 ISDN PRI NI 1/2 (ANSI T1.619a) (continued)	Yes	Certified	Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Met	
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Met	
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Not Tested	
				ITU-T ISDN Primary Access (C)	UCR Section 5.2.1.3.4.2	Not Tested	
				ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (C)	UCR Section 5.2.1.3.4.2.1	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 5.2.1.5.5	Not Tested	
				Call for Service Timing (R)	UCR Section 5.2.4.3.5	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.3.4.8	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.3.4.10	Not Tested	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4.2	Not Tested	
				Application (R)	UCR Section 5.2.4.7.1.1	Not Tested	
				Physical Layer (R)	UCR Section 5.2.4.7.1.2	Not Tested	
				Data Link Layer (R)	UCR Section 5.2.4.7.1.3	Not Tested	
				Data Link Connection (R)	UCR Section 5.2.4.7.1.3.1	Not Tested	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.2.4.7.1.3.2	Not Tested	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.2.4.7.1.4	Not Tested	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.2.4.7.1.4.2	Not Tested	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.2.4.7.1.4.3	Not Tested	
				Message Functional Definition and Content (R)	UCR Section 5.2.4.7.1.4.4	Not Tested	
				General Message Format and Information Elements Coding (R)	UCR Section 5.2.4.7.1.4.5	Not Tested	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested	
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				56 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				64 kbps switched data (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				NX56 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				NX64 synchronous BER (R: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested					
VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested				

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

DSN Line Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
2-Wire Loop Start Analog	Yes	Certified	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Met	
				PBX Line (C)	UCR Section 5.2.1.3.1	Met	
				Analog Line (R)	UCR Section 5.2.1.3.5	Met	
				Basic Line Test Capabilities (R)	UCR Section 5.2.1.5.4.1.1	Met	
				Advanced Line Test Capabilities (C)	UCR Section 5.2.1.5.4.1.1	Not Tested	See note 4.
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.4.2.1	Met	
				Reverse Battery (R)	UCR Section 5.2.4.3.1	Met	
			Voice	Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Met	
				MOS (R)	CJCSI 6215.01C	Met	
			Facsimile	Secure calls (R)	CJCSI 6215.01C	Met	
				Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met					
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested (See note 2.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				National ISDN 1/2 Basic Access (C)	UCR Section 5.2.1.3.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
				S/T Reference Point (R)	UCR Section 5.2.4.7.1.2.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	
			VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Not Tested	
2-Wire Proprietary Digital	No	Not Tested (See note 2.)	Access	Directory Number Identification (R)	UCR Section 5.2.1.1.1	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.2.4.5.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	

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

DSN Features and Capabilities						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 5.2.1.1.1	Met	
			Denied originating service (C)	UCR Section 5.2.1.1.3	Not Tested	
			Code restriction and diversion (C)	UCR Section 5.2.1.1.4	Met	
			Call waiting (R)	UCR Section 5.2.1.1.5.1	Partially Met	See note 5.
			Three-way calling (R)	UCR Section 5.2.1.1.6	Met	
			Add-on transfer, conference calling, and call hold (C)	UCR Section 5.2.1.1.7	Met	
			Call Transfer Individual – All calls (R)	UCR Section 5.2.1.1.7.1	Met	
			Call Transfer - Internal Only (R)	UCR Section 5.2.1.1.7.2	Met	
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)	UCR Section 5.2.1.1.7.3	Met	
			Call Transfer – Outside (R)	UCR Section 5.2.1.1.7.4	Met	
			Call Transfer – Add-On Restricted Station (C)	UCR Section 5.2.1.1.7.5	Not Tested	See note 4.
			Call Transfer – Attendant (C)	UCR Section 5.2.1.1.7.6	Not Tested	See note 4.
			Call Hold (R)	UCR Section 5.2.1.1.7.7	Met	
			Conference Calling – Six Way Station Controlled (C)	UCR Section 5.2.1.1.7.8	Not Tested	See note 4.
			Call Forwarding Variable (R)	UCR Section 5.2.1.1.8.1	Met	See note 6.
			Call Forward Busy Line (R)	UCR Section 5.2.1.1.8.2	Met	
			Call Forwarding – Don't Answer – All Calls (R)	UCR Section 5.2.1.1.8.3	Met	
			Selective Call Forwarding (C)	UCR Section 5.2.1.1.8.4	Met	
			Call pick-up (C)	UCR Section 5.2.1.1.9.1	Met	
			Address Translation (C)	UCR Section 5.2.1.7	Met	
Assured Dial Tone (C)	UCR Section 5.2.1.9	Met				
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 5.2.1.2.2	Not Tested	See note 4.
Public Safety	Yes	Certified	Emergency Service (911) Caller (R)	UCR Section 5.2.1.4.1.1	Met	See note 7.
			Emergency Service (911) Public Safety Answering Service (C)	UCR Section 5.2.1.4.1.2	Not Tested	See note 7.
			Enhanced Emergency Service (E911) (C)	UCR Section 5.2.1.4.1.3	Not Tested	See note 7.
			Trace of terminating calls (C)	UCR Section 5.2.1.4.2	Not Tested	See note 7.
			Outgoing call trace (C)	UCR Section 5.2.1.4.3	Not Tested	See note 7.

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

DSN Features and Capabilities						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
Conferencing	No	Not Tested	Preset Conferencing (C)	UCR Section 5.2.1.6	Not Tested	See note 4.
			Meet-Me Conferencing (R)	UCR Section 5.2.1.6.2	Not Tested	See note 4.
			Progressive Conferencing (C)	UCR Section 5.2.1.6.3	Not Tested	See note 4.
Nailed-up Connections	No	Not Tested	Nailed-Up Connections (C)	UCR Section 5.2.1.8	Not Tested	See note 4.
DSN Hotline Services	No	Not Tested	DSN Analog Hotline Service (C)	UCR Section 5.2.1.12	Not Tested	See note 4.
MLPP	Yes	Certified	MLPP Overview (R)	UCR Section 5.2.2.1.1	Met	See note 8.
			Preemption in the Network (R)	UCR Section 5.2.2.2	Met	
			Network Facility with Lower Precedence Calls (R)	UCR Section 5.2.2.2.1	Met	
			Network Facility with Equal or Higher Precedence Calls (R)	UCR Section 5.2.2.2.2	Met	
			Precedence Call Diversion (R)	UCR Section 5.2.2.3	Met	
			Channel Associated Signaling (C)	UCR Section 5.2.2.4.1	Not Tested	See note 2.
			Primary Rate Interface (R)	UCR Section 5.2.2.4.2	Met	
			Analog Line MLPP (R)	UCR Section 5.2.2.5	Met	
			ISDN MLPP Basic Rate Interface (C)	UCR Section 5.2.2.6	Not Tested	See note 2.
			ISDN Primary Rate Interface (R)	UCR Section 5.2.2.7	Met	
			Precedence Call Waiting (R)	UCR Section 5.2.2.8.1	Met	
			Call Forwarding (R)	UCR Section 5.2.2.8.2	Met	
			Call Transfer (R)	UCR Section 5.2.2.8.3	Met	
			Call Hold (R)	UCR Section 5.2.2.8.4	Met	
			Three-Way Calling (R)	UCR Section 5.2.2.8.5	Met	
			Call Pickup (C)	UCR Section 5.2.2.8.6	Met	
			Conferencing (C)	UCR Section 5.2.2.8.7.1	Not Met	
			Multiline Hunt Group (C)	UCR Section 5.2.2.8.8	Not Tested	See note 4.
Community of Interest (C)	UCR Section 5.2.2.8.9	Not Tested	See note 4.			
MLPP Interaction with EKTS features (C)	UCR Section 5.2.2.10.1	Not Tested	See note 4.			

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

DSN Features and Capabilities						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
Call Processing	Yes	Certified	Call Treatments (R)	UCR Section 5.2.3.1	Met	
			Primary and Alternate Routing (C)	UCR Section 5.2.3.2	Met	
			E&M Lead Signaling States (C)	UCR Section 5.2.3.3.1	Not Tested	See note 2.
			4-Wire Analog User Access Lines (C)	UCR Section 5.2.3.3.2	Not Tested	See note 2.
			2-Wire User Access Lines (R)	UCR Section 5.2.3.3.3	Met	
			Termination of Analog Lines (R)	UCR Section 5.2.3.3.4	Met	
			DSN User Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Interswitch and Intraswitch Dialing (R)	UCR Section 5.2.3.5.1.1	Met	
			Seven-Digit Dialing (R)	UCR Section 5.3.3.5.2.1	Met	
			Ten-Digit Dialing (R)	UCR Section 5.2.3.5.2.2	Met	
			Access Code (R)	UCR Section 5.2.3.5.1.3	Met	
			Access Digit (R)	UCR Section 5.2.3.5.1.3.1	Met	
			Precedence Digit (R)	UCR Section 5.2.3.5.1.3.2	Met	
			Service Digit (R)	UCR Section 5.2.3.5.1.3.3	Met	
			Route Code (R)	UCR Section 5.2.3.5.1.4	Met	
			Area Code (R)	UCR Section 5.2.3.5.1.5	Met	
			Switch Code (R)	UCR Section 5.2.3.5.1.6	Met	
			Line Number (R)	UCR Section 5.2.3.5.1.7	Met	
			Calling Name Delivery (C)	UCR Section 5.2.3.5.1.8.1	Not Tested	See note 4.
			Calling Number Delivery (R)	UCR Section 5.2.3.5.1.8.2	Met	
			Emergency Service 911 Conflict Resolution (R)	UCR Section 5.2.3.5.1.9	Met	
			DSN Switch Outpulsing Digit Formats (C)	UCR Section 5.2.3.5.2	Met	
			Standard Directory Number (R)	UCR Section 5.2.3.5.3	Met	
			Standard Test Numbers (C)	UCR Section 5.2.3.5.4	Not Tested	See note 4.
			Base Services – Abbreviated Numbers (C)	UCR Section 5.2.3.5.5	Not Tested	See note 4.
Digit Reception Requirements (R)	UCR Section 5.2.3.5.6	Met				
Screening (C)	UCR Section 5.2.3.5.8	Met				
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (C)	UCR Section 5.2.9.2, Table 5.2.9-1	Not Tested	See note 2.
			Uniform Interface Configuration for BRIs (C)	UCR Section 5.2.9.2, Table 5.2.9-2	Not Tested	See note 2.
			EKTS (C)	UCR Section 5.2.9.2, Table 5.2.9-3	Not Tested	See note 2.
			PRI Access, Call Control and Signaling (R)	UCR Section 5.2.9.2, Table 5.2.9-4	Met	See note 3.
			PRI Features (R)	UCR Section 5.2.9.2, Table 5.2.9-5	Met	See note 3.
			Packet Data Features and Capabilities (C)	UCR Section 5.2.9.2, Table 5.2.9-6	Not Tested	See note 2.

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

DSN Features and Capabilities						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
Synchroniza- tion	Yes	Certified	Line timing mode (R)	UCR Section 5.2.11.2	Met	
			Internal Stratum 4 (R)	UCR Section 5.2.10.1.1.2.2	Met	
			Synchronization Performance Monitoring Criteria (C)	UCR Section 5.2.10.2	Not Tested	See note 4.
			DS1 Traffic Interfaces (C)	UCR Section 5.2.10.3	Not Tested	See note 4.
			DS0 Traffic Interconnects (C)	UCR Section 5.2.10.4	Not Tested	See note 4.
Reliability	Yes	Certified	System Availability (R)	UCR Section 5.2.11.2	Met	See note 9.
			Backup Power (R)	UCR Section 5.2.11.3	Not Tested	See note 10.
			Power Components (R)	UCR Section 5.2.11.3.1	Not Tested	See note 10.
			UPS Requirements (R)	UCR Section 5.2.11.3.2	Not Tested	See note 10.
			UPS PBX 1 Load Capacity (R)	UCR Section 5.2.11.3.2.1	Not Tested	See note 10.
			Backup Power (Environmental) (R)	UCR Section 5.2.11.3.3	Not Tested	See note 10.
Security	Yes	Certified	Alarms (R)	UCR Section 5.2.11.3.4	Not Tested	See note 10.
			GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 3	Met	See note 11.
VoIP						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
VoIP System	No	Certified	Voice Service Quality at least 95% will have MOS of 4.0	UCR 2008, Change 2, section 5.2.12.8.2.1	Partially Met	See note 12.
			ITU-T G.711 PCM CODEC (R)	UCR Section 5.2.12.8.2.2	Met	
			MLPP (R)	UCR Section 5.2.12.8.2.3	Met	
			Security (R)	UCR Section 5.2.12.8.2.4	Met	
			Network management (C)	UCR Section 5.2.12.8.2.5	Met	
			System timing (R)	UCR Section 5.2.12.8.2.6	Met	
			Latency ≤ 60 milliseconds (R)	UCR Section 5.2.12.8.2.7	Met	
			IPv6 capable (R)	UCR 2008, Change 2, section 5.3.5	Met	See notes 13 and 14.
			Service Class Tagging (R)	UCR Section 5.2.12.8.2.9	Met	
			VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R)	UCR Section 5.2.12.8.2.10	Met	
Softphone Requirements (R)	UCR 2008, Change 2, section 5.3.2.6.1.7	Met	See note 15.			

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

Network Gateways							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
PSTN	No	Certified	Trunking	Positive Identification Control (C)	CJCSI 6215.01C	Met	
				On-Netting (C)	CJCSI 6215.01C	Met	
				Off-Netting (C)	CJCSI 6215.01C	Met	
				Ground Start Line (R)	UCR Section 5.2.2	Met	See note 9.
				Immediate Start (C)	UCR Section 5.3.2	Met	
				Delay Dial (C)	UCR Section 5.3.4	Met	

**NOTES:**

- 1 The SUT supports this interface; however, it was not tested and is not covered under this certification. This interface is not required for a PBX 1.
- 2 The SUT does not support this interface and it is not required for a PBX 1.
- 3 The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA adjudicated this anomaly as having a minor operational impact.
- 4 The SUT does not support this feature and it is not required for a PBX 1.
- 5 All of the features on the VoIP phones were tested using multiple line appearances. Although the SUT does not support Precedence Call Waiting on their VoIP phones, they do support multiple call appearances on their VoIP phones. This provides the ability for a user to receive additional calls while active with another call.
- 6 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.
- 7 The SUT only supports emergency basic 911 service. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. These public safety features are not required for a PBX 1.
- 8 The SUT does not support the Loss of Command and Control (C2) announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA previously adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1.
- 9 This was met with the vendor's LoC.
- 10 This requirement is a non-testable requirement. It is the responsibility of the respective base/post/camp/station communications agency to provide this with the SUT when installed.
- 11 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).
- 12 The CP-7940G, CIS-7961, and CP-7960G end instruments did not meet the Voice Quality E-model MOS, however they did meet the SAGE MOS of 4.0 or better. This discrepancy was adjudicated by DISA as having a minor operational impact.
- 13 The SUT met all IPv4 and IPv6 requirements through testing and LoC with the following exception: 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 Enclosure 2 and a dual stack call control agent in accordance with Reference (d). This was adjudicated by DISA as having a minor operational impact.
- 14 All dual stack IP end instruments can only be configured to register with IPv4. When configured to register with IPv6, the IP end instruments fail to register. This discrepancy was adjudicated by DISA as having a minor operational impact. All other dual stack IPv6 requirements were met.
- 15 The Cisco IP Communicator was tested and certified on the Windows XP operating system platform. The Cisco IP Communicator is not certified on other operating system platforms.

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

<b>LEGEND:</b>		
ANSI	American National Standards Institute	FTR 1080B Video Teleconferencing Services
BER	Bit Error Ratio	G.711 PCM of voice frequencies
BRI	Basic Rate Interface	GR Generic Requirement
C	Conditional	GR-815 Generic Requirements For Network Element/Network System (NE/NS) Security
CAS	Channel Associated Signaling	H.320 Standard for Narrowband VTC
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP Internet Protocol
CODEC	Coder/Decoder	IPv4 Internet Protocol version 4
CP	Cisco Phone	IPv6 Internet Protocol version 6
DIACAP	DoD Information Assurance Certification and Accreditation Process	ISDN Integrated Services Digital Network
DISA	Defense Information Systems Agency	IT Information Technology
DISR	DoD IT Standards Registry	ITU-T International Telecommunication Union - Telecommunication Standardization Sector
DoD	Department of Defense	kbps kilobits per second
DoDI	Department of Defense Instruction	LoC Letters of Compliance
DP	Dial Pulse	Mbps Megabits per second
DS0	Digital Signal Level 0 (64 kbps)	MFR1 Multi-Frequency Recommendation 1
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	min minute
DSN	Defense Switched Network	MLPP Multi-Level Precedence and Preemption
DTMF	Dual Tone Multi-Frequency	MOS Mean Opinion Score
E&M	Ear and Mouth	NFAS Non Facility Associated Signaling
E1	European Basic Multiplex Rate (2.048 Mbps)	NI 1/2 National ISDN Standard 1 or 2
EKTS	Electronic Key Telephone System	NI2 National ISDN Standard 2
FTR	Federal Telecommunications Recommendation	NX56 Data format restricted to multiples of 56 kbps
		NX64 Data format restricted to multiples of 64 kbps
		PBX Private Branch Exchange
		PBX 1 Private Branch Exchange 1
		PCM Pulse Code Modulation
		PCM-24 Pulse Code Modulation - 24 Channels
		PCM-30 Pulse Code Modulation - 30 Channels
		PRI Primary Rate Interface
		PSTN Public Switched Telephone Network
		Q.955.3 ISDN Signaling Standard for E1 MLPP
		R Required
		S/T ISDN BRI 4-wire interface
		SS7 Signaling System 7
		STE Secure Terminal Equipment
		STIGs Security Technical Implementation Guides
		STU-III Secure Telephone Unit -3rd generation
		SUT System Under Test
		T1 Digital Transmission Link Level 1 (1.544 Mbps)
		T1.619a SS7 and ISDN MLPP Signaling Standard for T1
		T.4 Standardization of Group 3 facsimile terminals for document transmission
		UCR Unified Capabilities Requirements
		UPS Uninterruptible Power Supply
		VBD Variable bit data
		VoIP Voice over Internet Protocol
		VTC Video Teleconferencing
		yr year