



## DEFENSE INFORMATION SYSTEMS AGENCY

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

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

### MEMORANDUM FOR DISTRIBUTION

**1 Jul 11**

**SUBJECT:** Extension of the Special Interoperability Test Certification of Cisco Unified Communications Manager Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T

**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 (h), see Enclosure

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 Unified Communications Manager Version 8.0(2) with IOS Software Release 15.1(1)T 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 identified test discrepancies shown in the Certification Testing Summary (Enclosure 2) have an overall minor operational impact. 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 of DISA Field Security Operations (FSO) accreditation, 23 September 2010.

3. The extension of this certification is based upon Desktop Review (DTR) 1. The original certification is based on interoperability testing conducted by JITC, DISA adjudication of open test discrepancy reports, review of the vendor's Letters of Compliance (LoC), and DSAWG accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 28 June through 13 August 2010 and documented in Reference (c). DISA adjudication of outstanding test discrepancy reports was completed on 25 August 2010. Review of the vendor's LoC was completed on 8 December 2010. The FSO granted accreditation on 31 March 2011 based on the security testing completed by DISA-led IA test teams and published in a separate report, Reference (d).

This DTR was requested to include Survivable Remote Site Telephony (SRST) functionality on the all certified Integrated Services Routers (ISRs). This DTR was reviewed by JITC and determined that Verification and Validation (V&V) testing would be needed to validate the SRST functionality and verify there was no impact on interoperability. The V&V testing was conducted from 25 through 29 April 2011 and 6 May 2011. The SRST functionality is dormant during normal SUT operation and after loss of primary DISN connectivity provides ROUTINE only intra-enclave voice features for Internet Protocol (IP) and analog end instruments registered to the ISR as well as access to the Public Switched Telephone Network (if available) for ROUTINE and emergency 911 calls. Therefore, JITC approves this DTR. SRST does not change the IA posture of this SUT, therefore the original DISA Network Systems Directorate approval (12 May 2011) applies to this DTR.

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. Defense Switched Network (DSN) services for Network and Applications specified in Reference (e).
- b. PBX 1 interface and signaling requirements for trunks/lines specified in References (f) and (g) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in References (f) and (g) verified through JITC testing and/or vendor submission of LoC.
- d. The overall system interoperability performance derived from test procedures listed in Reference (h).
- e. The Internet Protocol version 6 (IPv6) requirements were met through testing and review of the vendor LoC.

**Table 1. SUT Interoperability Test Summary**

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1. <sup>1</sup>
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and 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>2</sup>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.

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

<b>DSN Line Interfaces</b>				
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>	
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>3</sup>	
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.	
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 1.	
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The Cisco IP phones do not support Call Waiting. <sup>4</sup> The Cisco CP-7940G and CP-7960G are legacy end instruments that did not meet dual stack IPv6 requirements. <sup>5</sup>	
<b>DSN Features and Capabilities</b>				
<b>Features and Capabilities</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>	
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Full compliance of DSN Common Call Features was not met. <sup>3</sup>	
Attendant	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.	
Public Safety	Yes	Certified	The SUT met all critical CRs and FRs for Basic 911. <sup>6</sup>	
Conferencing	No	Not Tested (See note 7.)	The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing with the optional Cisco MeetingPlace® Express. The SUT does not support Preset Conferencing or Progressive Conferencing and these features are not required for a PBX 1.	
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.	
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT and is not required for a PBX 1.	
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a global diversion number. <sup>8</sup> The SUT does not support the Loss of Command and Control announcement. <sup>9</sup>	
Call Processing	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services	Yes	Certified	Met all critical CRs and FRs on the T1 PRI interface with the following minor exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS. <sup>2</sup>	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
Reliability	Yes	Certified	Met all critical CRs and FRs.	
Security	Yes	Certified	See note 10.	
VoIP System	No	Certified	The SUT is certified for VoIP with any certified ASLAN or ASLAN components posted on the UC APL. The SUT also met IPv4/IPv6 requirements. (See notes 5 and 11.)	
VoIP Softphone	No	Certified	Met all critical CRs and FRs.	
SRST	No	Certified		
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	The SUT T1 CAS interface did not meet all critical CRs and FRs. The SUT T1 CAS interface is not certified by JITC and is not required for a PBX 1. <sup>1</sup>
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	Although this interface is offered by the SUT, it was not tested. The SUT E1 CAS interface is not certified by JITC and 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>2</sup>
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs. <sup>12</sup>

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

<b>NOTES:</b>	
1	The SUT T1 CAS interface does not recognize Remove from Service (Busy Out) or Restore to Service (Make Idle) condition from the distant end switch. These are critical requirements for this interface; therefore, this interface is not certified by JITC.
2	The SUT does not support NFAS on their T1 ISDN PRI NI2. DISA previously adjudicated this anomaly as having a minor operational impact and stated the intent to change this from required to conditional for a PBX 1.
3	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.
4	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.
5	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 Reference (c), Enclosure 2 and a dual stack call control agent in accordance with Reference (h). This was adjudicated by DISA as having a minor operational impact.
6	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.
7	The SUT does not support conferencing. However, the SUT can support Meet-Me Conferencing through the use of an optional adjunct conferencing system called the Cisco MeetingPlace® Express, which is covered under a separate certification.
8	The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified Communications Manager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.
9	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.
10	Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (e).
11	The SUT met all IPv4 and IPv6 requirements with the following discrepancies noted with the SUT, which were adjudicated by DISA as having a minor operational impact: <ol style="list-style-type: none"><li>The VoIP SUT session control components and end instruments can only have the signaling service Traffic Class configured for 21 different DSCP values and not the full range required of 0-63.</li><li>The MCS 7835 and the MCS 7825 call managers OAM traffic is tagged at zero and is not configurable.</li><li>The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.</li><li>When the CP-7940G and CP-7960G phones are powered up, some of the UDP/TFTP traffic has a DSCP value of 4 and 802.1Q value of 5 and cannot be changed.</li><li>The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.</li><li>The IP phones are incorrectly tagging IPv6 TCP traffic during power up.</li><li>The Soft Client is incorrectly tagging all traffic during power up.</li><li>The 802.1Q CoS tag values are not independently configurable from the DSCP values.</li><li>End Instruments do not support the manual configuration of the IPv6 default gateway.</li><li>Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.</li></ol>
12	This interface requirement was met by the vendor's LoC.

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

<b>LEGEND:</b>			
802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	JITC	Joint Interoperability Test Command
		LoC	Letters of Compliance
		LSSGR	Local Access and Transport Area (LATA) Switching Systems
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps		Generic Requirements
ANSI	American National Standards Institute	Mbps	Megabits per second
APL	Approved Products List	MCS	Media Convergence Servers
ASLAN	Assured Services Local Area Network	MFR1	Multi-Frequency Recommendation 1
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
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	NFAS	Non Facility Associated Signaling
CP	Cisco Phone	OAM	Operational Administration and Maintenance
CRs	Capability Requirements	PBX 1	Private Branch Exchange 1
DISA	Defense Information Systems Agency	PRI	Primary Rate Interface
DP	Dial Pulse	PSTN	Public Switched Telephone Network
DSCP	Differentiated Services Code Point	Q.931	Signaling Standard for ISDN
DSN	Defense Switched Network	Q.955.3	ISDN Signaling standard for E1 MLPP
DSS1	Digital Subscriber Signaling 1	RTCP	RTP Control Protocol
DTMF	Dual Tone Multi-Frequency	RTP	Real-time Transport Protocol
E1	European Basic Multiplex Rate (2.048 Mbps)	SRST	Survivable Remote Site Telephony
FRs	Feature Requirements	SS7	Signaling System 7
GR	Generic Requirement	SUT	System Under Test
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IEEE	Institute of Electrical and Electronics Engineers	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IP	Internet Protocol	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IPv4	Internet Protocol version 4	TCP	Transmission Control Protocol
IPv6	Internet Protocol version 6	TFTP	Trivial File Transfer Protocol
ISDN	Integrated Services Digital Network	UC	Unified Capabilities
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UCR	Unified Capabilities Requirements
		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>• 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 (R)</li> <li>• Glare Operation (R)</li> <li>• Abnormal Wink Start (R)</li> <li>• Glare Resolution (R)</li> <li>• Call for Service Timing (R)</li> <li>• Guard Timing (R)</li> <li>• Satellite Timing (R)</li> <li>• Disconnect Control (R)</li> <li>• Reselect and Retrial (R)</li> <li>• Off-Hook Supervision Transition (R)</li> <li>• Dial-Pulse Signals (R)</li> <li>• DTMF Signaling (R)</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 (R)</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 (R)</li> <li>• Analog Trunk Interface (C)</li> <li>• Integrated Digital Loop Carrier (C)</li> <li>• Trunk Group-Remove from Service (R)</li> <li>• Trunk Group-Restore to Service (R)</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. 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			
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>
		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> <li>• SRST (C)</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> <li>• UCR Section 5.3.2.7.2.11</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 (R)</li> <li>• Outgoing call trace (R)</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 (R)</li> <li>• Primary Rate Interface (R)</li> <li>• Analog Line MLPP (R)</li> <li>• ISDN MLPP Basic Rate Interface (R)</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 (R)</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 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 1, section 5.3.5</li> <li>• UCR section 5.2.12.8.2.9</li> <li>• UCR 2008, Change 1, 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 (e) with the exception of the IPv6 and softphone requirements, because TDM requirements were not included in the UCR 2008, Change 1. The IPv6 and softphone requirements are derived from the UCR 2008, Change 1, Reference (f).				
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 1080B-2002 G.711	Video Teleconferencing Services PCM of voice frequencies	PCM-24	Pulse Code Modulation - 24 Channels
ANSI	American National Standards Institute	GR GR-815	Generic Requirement Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-30	Pulse Code Modulation - 30 Channels
BER	Bit Error Ratio		Standard for Narrowband VTC	PRI	Primary Rate Interface
BRI	Basic Rate Interface	H.320	Institute of Electrical and Electronics Engineers	PSTN	Public Switched Telephone Network
C	Conditional	IEEE	Internet Protocol	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling		Internet Protocol version 6	R	Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP IPv6	Integrated Services Digital Network	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	ISDN	Information Technology International	SRST	Survivable Remote Site Telephony
DIACAP	DoD Information Assurance Certification and Accreditation Process	IT ITU-T	Telecommunication Union-Telecommunication Standardization Sector	SS7	Signaling System 7
DISA	Defense Information Systems Agency		kilobits per second	STE	Secure Terminal Equipment
DISR	DoD IT Standards Registry		Megabits per second	STIGs	Security Technical Implementation Guides
DoD	Department of Defense	kbps	Multi-Frequency Recommendation 1	STU-III	Secure Telephone Unit -3rd generation
DoDI	Department of Defense Instruction	Mbps MFR1	Multi-Level Precedence and Preemption	T.4	Standardization of Group 3 facsimile terminals for document transmission
DP	Dial Pulse		Mean Opinion Score	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0 (64 kbps)	MLPP	National ISDN Standard 1 or 2	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MOS	Data format restricted to multiples of 56 kbps	TDM	Time Division Multiplexing
DSN	Defense Switched Network	NI 1/2	Data format restricted to multiples of 64 kbps	UCR	Unified Capabilities Requirements
DTMF	Dual Tone Multi-Frequency	NX56	Private Branch Exchange	UPS	Uninterruptible Power Supply
E&M	Ear and Mouth		Private Branch Exchange 1	VBD	Variable bit data
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Pulse Code Modulation	VoIP	Voice over Internet Protocol
EKTS	Electronic Key Telephone System	PBX PBX 1		VTC	Video Teleconferencing
FTR	Federal Telecommunications Recommendation	PCM			

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 <https://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, Extension of the Special Interoperability Test Certification of Cisco Unified Communications Manager Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T

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 1002901.

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 MARCORSYSCOM, 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) Joint Interoperability Test Command, Memo, "Special Interoperability Test Certification of Cisco Unified Communications Manager Version 8.0(2) with Internetwork Operating System (IOS) Software Release 15.1(1)T," 31 March 2011
- (d) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM) 8.0 (Tracking Number 1002901)," 31 March 2011
- (e) 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
- (e) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008," 22 January 2009
- (f) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008 Change 1," 22 January 2010
- (g) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006