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

IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **16 Nov 11**

MEMORANDUM FOR DISTRIBUTION

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 (g), see Enclosures included with the original certification memo (TN 1002901)

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 (CUCM) 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 Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation.

3. The extension of this certification is based upon Desktop Review (DTR) 4. 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 DISA Information Assurance (IA) approval of the IA configuration. 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. 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 DISA Certifying Authority (CA) 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 (c). This DTR was requested to include an upgrade of the 7900 series phones from firmware 9-0-2 to 9-2-1 to correct a vulnerability issue identified in IAVA 2011-B-0072. JITC approves this update based on Voice and Video test results of the 7900 series phones with firmware version 9-2-1, which was tested from 8 through 12 August 2011 with Cisco CUCM 8.0(2). DISA CA has approved the Information Assurance posture of the SUT in this DTR on 16 November 2011.

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 (d).

b. PBX 1 interface and signaling requirements for trunks/lines specified in References (e) and (f) verified through JITC testing and/or vendor submission of LoC.

c. PBX 1 CRs/FRs specified in References (e) and (f) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in Reference (g).

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)	No	Certified	Met all critical CRs and FRs.
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.[1]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
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.[2]
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.[3] The Cisco CP-7940G and CP-7960G are legacy end instruments that did not meet dual stack IPv6 requirements.[4]

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: Full compliance of DSN Common Call Features was not met.[2]
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.[5]
Conferencing	No	Not Tested[6]	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.[7] The SUT does not support the Loss of Command and Control announcement.[8]
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.[1]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified[9]	
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 [4] and [10].)
VoIP Softphone	No	Certified	Met all critical CRs and FRs.

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF)	No	Certified	Met all critical CRs and FRs.
	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.[1]
	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.[11]

Table 1. SUT Interoperability Test Summary (continued)

NOTES:

.

1 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.

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 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.

4 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 (g). This was adjudicated by DISA as having a minor operational impact.

5 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.

6 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.

7 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.

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 (c).

10 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:

a. 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.
b. The MCS 7835 and the MCS 7825 call managers OAM traffic is tagged at zero and is not configurable.
c. The 2851 and 3845 gateways are tagging IPv4 RTCP traffic at zero and it is not configurable.
d. 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.
e. The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.
f. The IP phones are incorrectly tagging IPv6 TCP traffic during power up.
g. The Soft Client is incorrectly tagging all traffic during power up.
h. The 802.1Q CoS tag values are not independently configurable from the DSCP values.
i. End Instruments do not support the manual configuration of the IPv6 default gateway.
j. Communications Managers are incorrectly tagging UDP/TFTP traffic to the end instrument during end instrument power up.

11 This interface requirement was met by the vendor's LoC.

Table 2. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Trunking	• PBX Line (C) • Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (R) • Glare Operation (R) • Abnormal Wink Start (R) • Glare Resolution (R) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (R) • Disconnect Control (R) • Reselect and Retrial (R) • Off-Hook Supervision Transition (R) • Dial-Pulse Signals (R) • DTMF Signaling (R) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (R) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (R) • Interoperation of PCM-24 and PCM-30 (R) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (R) • Trunk Group-Restore to Service (R)	• UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.2 • UCR Section 5.2.1.3.4.1 • UCR Section 5.2.1.3.4.1.1 • UCR Section 5.2.1.3.4.2 • UCR Section 5.2.1.3.4.2.1 • UCR Section 5.2.4.3.3.1.1 • UCR Section 5.2.4.3.3.1.2 • UCR Section 5.2.4.3.3.2.1 • UCR Section 5..2.4.3.3.2.2 • UCR Section 5.2.4.3.5 • UCR Section 5.2.4.3.6 • UCR Section 5.2.3.4.7 • UCR Section 5.2.3.4.8 • UCR Section 5..2.3.4.9 • UCR Section 5.2.3.4.10 • UCR Section 5..2.4.4.1 • UCR Section 5.2.4.4.2 • UCR Section 5..2.4.4.2.1 • UCR Section 5.2.4.4.3 • UCR Section 5..2.4.5.1 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5..2.4.7.1.1 • UCR Section 5.2.4.7.1.2 • UCR Section 5.2.4.7.1.3 • UCR Section 5..2.4.7.1.3.1 • UCR Section 5.2.4.7.1.3.2 • UCR Section 5.2.4.7.1.4 • UCR Section 5.2.4.7.1.4.2 • UCR Section 5.2.4.7.1.4.3 • UCR Section 5.2.4.7.1.4.4 • UCR Section 5.2.4.7.1.4.5 • UCR Section 5.2.4.7.1.4.6 • UCR Section 5.2.6.1 • UCR Section 5.2.6.1.1 • UCR Section 5.2.6.1.2 • UCR Section 5.2.6.1.3 • UCR Section 5.2.6.1.4 • UCR Section 5.2.6.2 • UCR Section 5.2.6.3 • UCR Section 5.2.6.4 • UCR Section 5.2.6.5 • UCR Section 5.2.1.5.5 • UCR Section 5.2.1.5.5

Table 2. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP)	No	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Facsimile	• Analog: ITU-T T.4 (R)	• DISR
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • UCR Section 5.2.2.9.6 • CJCSI 6215.01C
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog	Yes	Access	• Directory Number Identification (R) • Analog Line (R) • National ISDN 1/2 Basic Access (R: BRI Only) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R: 2-WireAnalog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.5 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.1.5.4.1.1 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.5.1 • UCR Section 5.2.4.7.1.2.1 • UCR Section 5.2.12.8
ISDN BRI NI 1/2 (ANSI T1.619a)	No			
2-Wire Proprietary Digital	No			
VoIP (Ethernet IEEE 802.3u)	No	Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (R) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (R)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.3 • UCR Section 5.2.1.1.4 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1 • UCR Section 5.2.1.7 • UCR Section 5.2.1.9
Attendant	No	• Attendant Features (C)		• UCR Section 5.2.1.2.2

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service Basic (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (R) • Outgoing call trace (R)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3 • UCR Section 5.2.1.4.2 • UCR Section 5.2.1.4.3
Conferencing	No	• Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C)	• UCR Section 5.2.1.6.1 • UCR Section 5.2.1.6.2 • UCR Section 5.2.1.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 5.2.1.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 5.2.1.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (R) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (R) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 5.2.2.1.1 • UCR Section 5.2.2.2 • UCR Section 5.2.2.2.1 • UCR Section 5.2.2.2.2 • UCR Section 5.2.2.3 • UCR Section 5.2.2.4.1 • UCR Section 5.2.2.4.2 • UCR Section 5.2.2.5 • UCR Section 5.2.2.6 • UCR Section 5.2.2.7 • UCR Section 5.2.2.8.1 • UCR Section 5.2.2.8.2 • UCR Section 5.2.2.8.3 • UCR Section 5.2.2.8.4 • UCR Section 5.2.2.8.5 • UCR Section 5.2.2.8.6 • UCR Section 5.2.2.8.7.1 • UCR Section 5.2.2.8.8 • UCR Section 5.2.2.8.9 • UCR Section 5.2.2.10.1

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (R) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (R) • Digit Reception Requirements (R) • Screening (R)	• UCR Section 5.2.3.1 • UCR Section 5.2.3.2 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.3.4 • UCR Section 5.2.3.5.1.1 • UCR Section 5.2.3.5.1.1 • UCR Section 5.3.3.5.2.1 • UCR Section 5.2.3.5.2.2 • UCR Section 5.2.3.5.1.3 • UCR Section 5.2.3.5.1.3.1 • UCR Section 5.2.3.5.1.3.2 • UCR Section 5.2.3.5.1.3.3 • UCR Section 5.2.3.5.1..4 • UCR Section 5.2.3.5.1.5 • UCR Section 5.2.3.5.1.6 • UCR Section 5.2.3.5.1.7 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.1.9 • UCR Section 5.2.3.5.2 • UCR Section 5.2.3.5.3 • UCR Section 5.2.3.5.4 • UCR Section 5.2.3.5.5 • UCR Section 5.2.3.5.6 • UCR Section 5.2.3.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (R) • Uniform Interface Configuration for BRIs (R) • EKTS (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2, Table 5.2.9-1 • UCR Section 5.2.9.2, Table 5.2.9-2 • UCR Section 5.2.9.3, Table 5.2.9-3 • UCR Section 5.2.9.2, Table 5.2.9-4 • UCR Section 5.2.9.2, Table 5.2.9-5 • UCR Section 5.2.9.2, Table 5.2.9-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 5.2.11.2 • UCR Section 5.2.11.3 • UCR Section 5.2.11.3.1 • UCR Section 5.2.11.3.2 • UCR Section 5.2.11.3.2.1 • UCR Section 5.2.11.3.3 • UCR Section 5.2.11.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 3

Table 2. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
VoIP System[1]	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • Softphone Requirements (C)		• UCR section 5.2.12.8.2.1 • UCR section 5.2.12.8.2.2 • UCR section 5.2.12.8.2.3 • UCR section 5.2.12.8.2.4 • UCR section 5.2.12.8.2.5 • UCR section 5.2.12.8.2.6 • UCR section 5.2.12.8.2.7 • UCR 2008, Change 1, section 5.3.5 • UCR section 5.2.12.8.2.9 • UCR 2008, Change 1, section 5.3.2.6.1.7
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN[2]	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

NOTES:
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)

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	FTR 1080B-2002	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
		G.711	PCM of voice frequencies		
		GR	Generic Requirement	PCM-30	Pulse Code Modulation - 30 Channels
ANSI	American National Standards Institute	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PRI	Primary Rate Interface
BER	Bit Error Ratio			PSTN	Public Switched Telephone Network
BRI	Basic Rate Interface	H.320	Standard for Narrowband VTC		
C	Conditional	IEEE	Institute of Electrical and Electronics Engineers	Q.955.3	ISDN Signaling Standard for E1 MLPP
CAS	Channel Associated Signaling				
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP	Internet Protocol	R	Required
		IPv6	Internet Protocol version 6	S/T	ISDN BRI four-wire interface
CODEC	Coder/Decoder	ISDN	Integrated Services Digital Network		
DIACAP	DoD Information Assurance Certification and Accreditation Process			SS7	Signaling System 7
		IT	Information Technology	STE	Secure Terminal Equipment
		ITU-T	International Telecommunication Union-Telecommunication Standardization Sector	STIGs	Security Technical Implementation Guides
DISA	Defense Information Systems Agency			STU-III	Secure Telephone Unit -3rd generation
DISR	DoD IT Standards Registry				
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		
		MFR1	Multi-Frequency Recommendation 1		
DP	Dial Pulse			T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0 (64 kbps)	MLPP	Multi-Level Precedence and Preemption		
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)			T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
		MOS	Mean Opinion Score		
DSN	Defense Switched Network	NI 1/2	National ISDN Standard 1 or 2	TDM	Time Division Multiplexing
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps	UCR	Unified Capabilities Requirements
E&M	Ear and Mouth				
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps	UPS	Uninterruptible Power Supply
EKTS	Electronic Key Telephone System	PBX	Private Branch Exchange	VBD	Variable bit data
		PBX 1	Private Branch Exchange 1	VoIP	Voice over Internet Protocol
FTR	Federal Telecommunications Recommendation	PCM	Pulse Code Modulation	VTC	Video Teleconferencing

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.

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. 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:

Enclosure 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, "Information Assurance (IA) Assessment of Cisco Unified Communications Manager (CUCM) 8.0 (Tracking Number 1002901)," 31 March 2011

(d) 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