



DEFENSE INFORMATION SYSTEMS AGENCY

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

30 Dec 09

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (g), 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 CallManager Version 4.3(2) SR1b with IOS Software Release 12.4(15) T7 is hereinafter referred to as the system under test (SUT). The SUT meets all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the Voice over Internet Protocol critical interoperability requirements with any certified Assured Services Local Area Network (ASLAN) on the Unified Capabilities (UC) Approved Products List (APL). The identified test discrepancies shown in the SUT Interoperability Test Summary 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 the original memorandum (28 April 2009).

3. The extension of this certification is based upon Desktop Review (DTR) 3. 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 Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 3 November through 19 December 2008 and documented in reference (c). DISA adjudication of outstanding test discrepancy reports and review of the vendor's LoC was completed on 17 December 2008. DSAWG grants accreditation based on the security testing completed by DISA-led Information Assurance test teams and published in a separate report, reference (d). DSAWG accreditation was granted on

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21 April 2009. This DTR was requested to include the following Media Convergence Server (MCS) servers as part of the SUT: MCS7835I2, MCS7845I2, MCS7835I3, MCS7845I3, and MCS7825I4. The desktop review request was approved on 9 December 2009.

4. The SUT certified hardware and software components are listed in Table 1. The interoperability test summary of the SUT is indicated in Table 2. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 3. This interoperability test status is based on the PBX 1's ability to meet:

- a. DSN services for Network and Applications specified in reference (e).
- b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (f) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in reference (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).

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Table 1. SUT Hardware and Software Components

Cisco Unified CallManager Version 4.3(2) SR1b , with IOS Software Release 12.4(15) T7			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
CallManagers <u>MCS7835H2</u> , <u>MCS7825H3</u> , <u>MCS7825H2</u> , <u>MCS7835H1</u> , MCS7835H, MCS7835I1, MCS7845H2, MCS7825H, MCS7835I, MCS7845H, MCS7845I, MCS7825- H1, MCS7825I1, MCS7845H1, MCS7845I1, MCS7835I2, MCS7845I2, MCS7835I3, MCS7845I3, and MCS7825I4	4.3(2) SR1b	Not Applicable	Processing/Signaling
Cisco 3745/3725 Multiservice Access Router (Gateway) (See note 2.)	IOS 12.4(15) T7	<u>NM HD 2V</u>	TDM Interface NM HD Voice, 2-slot IP communications voice/fax
		<u>NM HD 2VE</u>	TDM Interface NM HD Voice, 2-slot IP communications enhanced voice/fax
		<u>VIC 4FXS/DID</u>	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial
		<u>VIC2 2FXS</u>	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		VWIC 1MFT T1	Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1
		<u>VWIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		<u>VWIC 2MFT T1 DI</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1, Drop and Insert
Cisco 3845/3825 Integrated Services Router (Gateway)	IOS 12.4(15) T7	<u>NM HDV2</u>	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		<u>VWIC2 2MFT T1/E1</u>	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		<u>NM HDV2 2T1/E1</u>	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		<u>VIC 4FXS/DID</u>	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		<u>EM HDA 8FXS</u>	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 3.)

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Table 1. SUT Hardware and Software Components (continued)

Cisco Unified CallManager Version 4.3(2) SR1b with IOS Software Release 12.4(15) T7 (continued)			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
Cisco 2851 Integrated Services Router (Gateway)	IOS 12.4(15) T7	NM HDV2	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VVIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		EYM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VVIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G (See note 2.)	Load: P00308000900	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	Load: SCCP70.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7911G and CP-7906G	Load: SCCP11.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	Load: SCCP41.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	Load: SCCP42.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	Load: SCCP45.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	Load: SCCP75.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000300	Not Applicable	Expansion module
7915	B015-1-0-2	Not Applicable	Expansion module
7916	B016-1-0-2SR1	Not Applicable	Expansion module
CIS 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	CP-7961G IP phone, TEMPEST version
CIS 7975G (See note 5.)	SCCP75.8-4-1SR1S	Not Applicable	7975G IP phone, TEMPEST version
CRYPTEK 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	7961G IP phone, TEMPEST version with no PC interface and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones

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Table 1. SUT Hardware and Software Components (continued)

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 IOS software and hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.				
2	All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:				
	a. The component must already be JITC certified and currently fielded within the DSN.				
	b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.				
	c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP phones respectively as replacements for the CP-7940G and CP-7960G IP phones.				
3	These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.				
4	The EM HDA 8FXS expansion module requires the EVM HD module. Up to two EM HDA 8FXS expansion modules are supported for each EVM HD.				
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.				
LEGEND:					
10/100BaseT	10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet	GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)	PSTN	Public Switched Telephone Network
CP	Cisco Phone	HD	High Density	RJ	Registered Jack
DI	Drop and Insert	HDA	High Density Analog	SCCP	Skinny Client Control Protocol
DID	Direct Inward Dialing	IOS	Internetwork Operating System	SR	Service Release
DSN	Defense Switched Network	IP	Internet Protocol	SUT	System Under Test
E1	European Basic Multiplex Rate (2.048 Mbps)	IPv6	Internet Protocol version 6	T1	Digital Transmission Link Level 1 (1.544 Mbps)
EM	Expansion Module	ISDN	Integrated Services Digital Network	TDM	Time Division Multiplexing
EVM	Extension Voice Module	JITC	Joint Interoperability Test Command	UCR	Unified Capabilities Requirements
Fax	facsimile	Mbps	Megabits per second	V	Voice
FXS	Foreign Exchange Station	MCS	Media Convergence Server	VE	Voice/Fax Enhanced
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)	MFT	Multiflex Trunk	VIC	Voice Interface Card
		ms	milliseconds	VWIC	Voice WAN Interface Card
		NM	Network Module	WAN	Wide Area Network
		PC	Personal Computer		
		PRI	Primary Rate Interface		

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release and have not been fixed by the vendor. Wink start recognition is not within the required tolerance. ¹ An off-hook seizure below the minimum limit is treated as valid. ² A call fails to complete after trunk preemption. ³ Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA. ⁴ When channels on a T1 CAS trunk group are busied by the remote switching system, the SUT fails to acknowledge all of these busy outs. As a result calls originated by the SUT fail to complete and the proper treatment is not provided. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface 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 exceptions: Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA. ⁴ This interface does not support NFAS. ⁵ The SUT monitoring tool occasionally provides inaccurate reports when a remote trunk is busy. ⁶
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface 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 a minor configuration change ⁷ and the following minor exceptions: The SUT does not support an MLPP global diversion number. ⁸ Diversion to the alternate directory number is provided in lieu of the BNEA. ⁹ When an analog line is preempted at a precedence higher than the already established call, the analog interface will ring at ROUTINE. ¹⁰ The SUT gateway analog interface does not support required line features. ¹¹ The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
VoIP	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number. ⁸ Diversion to the alternate directory number is provided in lieu of the BNEA. ⁹ The operational impact is minor.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: Full compliance of DSN Common Call Features was not met. ^{11, 12, 13, 14, 15, 16, 17} The operational impact is minor.
Attendant	No	Not Tested	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.
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. ¹⁸

Table 2. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Conferencing	Yes	Certified	Meet-Me Conferencing is met through the use of the Cisco MeetingPlace. The SUT does not support 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.	
Nailed-up Connections	No	Not Tested	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.	
DSN Hotline Services	No	Not Tested	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.	
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. ¹⁹	
Call Processing	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services	Yes	Certified	Met all critical CRs and FRs. ⁵	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
Reliability	Yes	Certified	Met all critical CRs and FRs.	
Security	Yes	Certified	See note 20.	
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN posted on the UC APL. See notes 21 and 22.	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release. Wink start recognition is not within the required tolerance. ¹ An off-hook seizure below the minimum limit is treated as valid. ² A call fails to complete after trunk preemption. ³ The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface 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: This interface does not support NFAS. ⁵ The operational impact is minor.
	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.
NOTES:				
1 T1 CAS wink start recognition is not within the required tolerance of 100 ms to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.				
2 The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.				
3 During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.				
4 ROUTINE calls attempted over a trunk that is broken receive a T120 in lieu of an ICA. Calls above ROUTINE attempted over a trunk that is broken receive a BPA in lieu of an ICA. The operational impact is minor because they are treated with a BPA and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.				
5 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.				

Table 2. SUT Interoperability Test Summary (continued)

NOTES (continued):

- 6 A discrepancy exists that is associated with the monitoring tool that the SUT uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.
- 7 A configuration change was required on the analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.
- 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 CallManager. 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 When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the BNEA is not provided. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.
- 10 When an analog station is active with a call and is preempted by a higher precedence call, the analog station receives the proper PNT. However, after going on hook, the station rings at ROUTINE. This was found to be a minor impact because the station is still preempted correctly.
- 11 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX 1 for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a PBX 1.
- 12 The SUT does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.
- 13 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.
- 14 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.
- 15 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.
- 16 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. DISA adjudicated this anomaly as having a minor operational impact because the preempted user receives PNT and the other members remain connected.
- 17 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for Precedence above ROUTINE calls. There is no operational impact.
- 18 The SUT only supports emergency service 911 public safety features. 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.
- 19 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 adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.
- 20 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (d).
- 21 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria:
 - a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).
 - b. Maintaining interoperability in heterogeneous environments and with IPv4.
 - c. Commitment to upgrade as the IPv6 standard evolves.
 - d. Availability of contractor/vendor IPv6 technical support.

Table 2. SUT Interoperability Test Summary (continued)

NOTES (continued):

- 22 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP-7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:
- The component must already be JITC certified and currently fielded within the DSN.
 - This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.
 - There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

LEGEND:

ANSI	American National Standards Institute	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
APL	Approved Products List	Mbps	Megabits per second
ASLAN	Assured Services Local Area Network	MFR1	Multi-Frequency Recommendation 1
BNEA	Busy Not Equipped Announcement	MLPP	Multi-Level Precedence and Preemption
BPA	Block Precedence Announcement	ms	milliseconds
BRI	Basic Rate Interface	NI 1/2	National ISDN Standard 1 or 2
C2	Command and Control	NI2	National ISDN Standard 2
CAS	Channel Associated Signaling	NFAS	Non Facility Associated Signaling
CRs	Capability Requirements	PBX 1	Private Branch Exchange 1
DISA	Defense Information Systems Agency	PMO	Program Management Office
DP	Dial Pulse	PNT	Preemption Notification Tone
DSN	Defense Switched Network	PRI	Primary Rate Interface
DSS1	Digital Subscriber Signaling 1	PSTN	Public Switched Telephone Network
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.955.3	ISDN Signaling standard for E1 MLPP
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)
ICA	Isolated Code Announcement	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	UC	Unified Capabilities
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	VoIP	Voice over Internet Protocol
JITC	Joint Interoperability Test Command		

Table 3. 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"> • 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 (C) • Glare Operation (C) • Abnormal Wink Start (C) • Glare Resolution (C) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (C) • Disconnect Control (C) • Reselect and Retrial (C) • Off-Hook Supervision Transition (C) • Dial-Pulse Signals (C) • DTMF Signaling (C) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) 	<ul style="list-style-type: none"> • UCR Section 2.3.1 • UCR Section 2.3.2 • UCR Section 2.3.4.1 • UCR Section 2.3.4.1.1 • UCR Section 2.3.4.2 • UCR Section 2.3.4.2.1 • UCR Section 5.3.3.1.1 • UCR Section 5.3.3.1.2 • UCR Section 5.3.3.2.1 • UCR Section 5.3.3.2.2 • UCR Section 5.3.5 • UCR Section 5.3.6 • UCR Section 5.3.7 • UCR Section 5.3.8 • UCR Section 5.3.9 • UCR Section 5.3.10
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> • Dial-Pulse Signals (C) • DTMF Signaling (C) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) 	<ul style="list-style-type: none"> • UCR Section 5.4.1 • UCR Section 5.4.2 • UCR Section 5.4.2.1 • UCR Section 5.4.3 • UCR Section 5.5 • UCR Section 5.7.1 • UCR Section 5.7.1.1 • UCR Section 5.7.1.2
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		<ul style="list-style-type: none"> • 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) 	<ul style="list-style-type: none"> • UCR Section 5.7.1.3 • UCR Section 5.7.1.3.1 • UCR Section 5.7.1.3.2 • UCR Section 5.7.1.4 • UCR Section 5.7.1.4.2 • UCR Section 5.7.1.4.3 • UCR Section 5.7.1.4.4 • UCR Section 5.7.1.4.5
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)		<ul style="list-style-type: none"> • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (C) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (C) • Interoperation of PCM-24 and PCM-30 (C) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (C) • Trunk Group-Restore to Service (C) 	<ul style="list-style-type: none"> • UCR Section 5.7.1.4.6 • UCR Section 7.1 • UCR Section 7.1.1 • UCR Section 7.1.2 • UCR Section 7.1.3 • UCR Section 7.1.4 • UCR Section 7.2 • UCR Section 7.3 • UCR Section 7.4 • UCR Section 7.5 • UCR Section 2.5.5 • UCR Section 2.5.6

Table 3. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No	Voice	<ul style="list-style-type: none"> • MOS (R) • Secure calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • CJCSI 6215.01C
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> • Analog: ITU-T T.4 (R) 	<ul style="list-style-type: none"> • DISR
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	<ul style="list-style-type: none"> • 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) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: PRI only) 	<ul style="list-style-type: none"> • FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> • Directory Number Identification (R) • National ISDN 1/2 Basic Access (C) • Analog Line (R) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) 	<ul style="list-style-type: none"> • UCR Section 2.1.1 • UCR Section 2.3.3 • UCR Section 2.3.5 • UCR Section 2.5.4.1.1 • UCR Section 2.5.4.1.2 • UCR Section 5.2.1
ISDN BRI NI 1/2 (ANSI T1.619a)	No		<ul style="list-style-type: none"> • Reverse Battery (R) • Alerting Signals and Tones (R) • S/T Reference Point (ISDN BRI) (C) 	<ul style="list-style-type: none"> • UCR Section 5.3.1 • UCR Section 5.5 • UCR Section 5.7.1.2.1
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> • MOS (R) • Secure Calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	<ul style="list-style-type: none"> • Analog: ITU-T T.4 (R) 	<ul style="list-style-type: none"> • DISR
		Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • Secure data (STE/STU-III) (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • CJCSI 6215.01C
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: BRI only) 	<ul style="list-style-type: none"> • FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	Critical	Requirements Required or Conditional		References
Common Features	Yes	<ul style="list-style-type: none"> • Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (C) • 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 (C) 		<ul style="list-style-type: none"> • UCR Section 2.1 • UCR Section 2.1.3 • UCR Section 2.1.4 • UCR Section 2.1.5 • UCR Section 2.1.6 • UCR Section 2.1.7 • UCR Section 2.1.7.1 • UCR Section 2.1.7.2 • UCR Section 2.1.7.3 • UCR Section 2.1.7.4 • UCR Section 2.1.7.5 • UCR Section 2.1.7.6 • UCR Section 2.1.7.7 • UCR Section 2.1.7.8 • UCR Section 2.1.8.1 • UCR Section 2.1.8.2 • UCR Section 2.1.8.3 • UCR Section 2.1.8.4 • UCR Section 2.1.9 • UCR Section 2.7 • UCR Section 2.9
Attendant	No	<ul style="list-style-type: none"> • Attendant Features (C) 		<ul style="list-style-type: none"> • UCR Section 2.2

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Public Safety	Yes	<ul style="list-style-type: none"> • Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (C) • Outgoing call trace (C) 	<ul style="list-style-type: none"> • UCR Section 2.4.1.1 • UCR Section 2.4.1.2 • UCR Section 2.4.1.3 • UCR Section 2.4.2 • UCR Section 2.4.3
Conferencing	Yes	<ul style="list-style-type: none"> • Preset Conferencing (C) • Meet-Me Conferencing (R) • Progressive Conferencing (C) 	<ul style="list-style-type: none"> • UCR Section 2.6 • UCR Section 2.6.2 • UCR Section 2.6.3
Nailed-up Connections	No	<ul style="list-style-type: none"> • Nailed-Up Connections (C) 	<ul style="list-style-type: none"> • UCR Section 2.8
DSN Hotline Services	No	<ul style="list-style-type: none"> • DSN Analog Hotline Service (C) 	<ul style="list-style-type: none"> • UCR Section 2.12
MLPP	Yes	<ul style="list-style-type: none"> • 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 (C) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (C) • 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) 	<ul style="list-style-type: none"> • UCR Section 3.1 • UCR Section 3.2 • UCR Section 3.2.1 • UCR Section 3.2.2 • UCR Section 3.3 • UCR Section 3.4.1 • UCR Section 3.4.2 • UCR Section 3.5 • UCR Section 3.6 • UCR Section 3.7 • UCR Section 3.8.1 • UCR Section 3.8.2 • UCR Section 3.8.3 • UCR Section 3.8.4 • UCR Section 3.8.5 • UCR Section 3.8.6 • UCR Section 3.8.7 • UCR Section 3.8.8 • UCR Section 3.8.9 • UCR Section 3.11

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Call Processing	Yes	<ul style="list-style-type: none"> • Call Treatments (R) • Primary and Alternate Routing (C) • 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 (C) • Digit Reception Requirements (R) • Screening (C) 	<ul style="list-style-type: none"> • UCR Section 4.1 • UCR Section 4.2 • UCR Section 4.3.1 • UCR Section 4.3.2 • UCR Section 4.3.3 • UCR Section 4.3.4 • UCR Section 4.5.1.1 • UCR Section 4.5.1.2 • UCR Section 4.5.1.2.1 • UCR Section 4.5.1.2.2 • UCR Section 4.5.1.3 • UCR Section 4.5.1.3.1 • UCR Section 4.5.1.3.2 • UCR Section 4.5.1.3.3 • UCR Section 4.5.1.4 • UCR Section 4.5.1.5 • UCR Section 4.5.1.6 • UCR Section 4.5.1.7 • UCR Section 4.5.1.8.1 • UCR Section 4.5.1.8.2 • UCR Section 4.5.1.9 • UCR Section 4.5.2 • UCR Section 4.5.3 • UCR Section 4.5.4 • UCR Section 4.5.5 • UCR Section 4.5.6 • UCR Section 4.5.8
ISDN Services	Yes	<ul style="list-style-type: none"> • BRI Access, Call Control and Signaling (C) • Uniform Interface Configuration for BRIs (C) • Electronic Key Telephone Systems (EKTS) (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C) 	<ul style="list-style-type: none"> • UCR Section 10, Table 10-1 • UCR Section 10, Table 10-2 • UCR Section 10, Table 10-3 • UCR Section 10, Table 10-4 • UCR Section 10, Table 10-5 • UCR Section 10, Table 10-6
Synchronization	Yes	<ul style="list-style-type: none"> • Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C) 	<ul style="list-style-type: none"> • UCR Section 11.1.1.2 • UCR Section 11.1.2.2 • UCR Section 11.2 • UCR Section 11.3 • UCR Section 11.4
Reliability	Yes	<ul style="list-style-type: none"> • System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R) 	<ul style="list-style-type: none"> • UCR Section 12.2 • UCR Section 12.3 • UCR Section 12.3.1 • UCR Section 12.3.2 • UCR Section 12.3.2.2 • UCR Section 12.3.3 • UCR Section 12.3.4
Security	Yes	<ul style="list-style-type: none"> • GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R) 	<ul style="list-style-type: none"> • UCR Section 13

Table 3. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	Critical	Requirements Required or Conditional		References
VoIP System	No	VoIP function is conditional. If VoIP is provided, all of the following requirements must be met: <ul style="list-style-type: none"> • 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) • VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R) 		<ul style="list-style-type: none"> • UCR App. 3, para. A3.2.1 • UCR App. 3, para. A3.2.2 • UCR App. 3, para. A3.2.3 • UCR App. 3, para. A3.2.4 • UCR App. 3, para. A3.2.5 • UCR App. 3, para. A3.2.6 • UCR App. 3, para. A3.2.7 • UCR App. 3, para. A3.2.8 • UCR App. 3, para. A3.2.9 • UCR App. 3, para. A3.2.10
Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN (See note.)	No	Trunking	<ul style="list-style-type: none"> • Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.2 • UCR Section 5.3.2 • UCR Section 5.3.4
<p>NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.</p>				

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7

Table 3. PBX 1 Requirements (continued)

LEGEND:					
ANSI	American National Standards Institute	FTR 1080B-2002	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	G.711	PCM of voice frequencies	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	GR	Generic Requirement		
C	Conditional	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PRI	Primary Rate Interface
CAS	Channel Associated Signaling	H.320	Standard for Narrowband VTC	PSTN	Public Switched Telephone Network
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	IP	Internet Protocol	Q.955.3	ISDN Signaling Standard for E1 MLPP
CODEC	Coder/Decoder	IPv6	Internet Protocol version 6	R	Required
DIACAP	DoD Information Assurance Certification and Accreditation Process	ISDN	Integrated Services Digital Network	S/T	ISDN BRI four-wire interface
DISR	DoD IT Standards Registry	IT	Information Technology	SS7	Signaling System 7
DoD	Department of Defense	ITU-T	International Telecommunication Union- Telecommunication Standardization Sector	STE	Secure Terminal Equipment
DoDI	Department of Defense Instruction	kbps	kilobits per second	STIGs	Security Technical Implementation Guides
DP	Dial Pulse	Mbps	Megabits per second	STU-III	Secure Telephone Unit -3rd generation
DS0	Digital Signal Level 0 (64 kbps)	MFR1	Multi-Frequency Recommendation 1	T.4	Standardization of Group 3 facsimile terminals for document transmission
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	min	minute	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DSN	Defense Switched Network	MLPP	Multi-Level Precedence and Preemption	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DTMF	Dual Tone Multi-Frequency	MOS	Mean Opinion Score	UCR	Unified Capabilities Requirements
E&M	Ear and Mouth	NI 1/2	National ISDN Standard 1 or 2	UPS	Uninterruptible Power Supply
E1	European Basic Multiplex Rate (2.048 Mbps)	NX56	Data format restricted to multiples of 56 kbps	VBD	Variable bit data
EKTS	Electronic Key Telephone System	NX64	Data format restricted to multiples of 64 kbps	VoIP	Voice over Internet Protocol
FTR	Federal Telecommunications Recommendation	para. PBX	paragraph Private Branch Exchange	VTC	Video Teleconferencing
		PBX 1	Private Branch Exchange 1	yr	year
		PCM	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), or <http://199.208.204.125> (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7

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 0814401.

FOR THE COMMANDER:



for RICHARD A. MEADOR
Chief
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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 CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7," 28 April 2009
- (d) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7," 21 April 2009
- (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
- (f) Defense Information Systems Agency, "Department of Defense Networks Unified Capabilities Requirements," 21 December 2007
- (g) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006