



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

10 Mar 11

SUBJECT: Extension of the Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

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 (i), 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 7.1(2) with IOS Software Release 12.4(22)T2 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 (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 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 Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation of the original certification (31 March 2010).

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 DSAWG accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 14 July through 22 August 2009. Additional testing was conducted from 9 through 20 November 2009 and is documented in Reference (c). DISA adjudication of outstanding test discrepancy reports was completed on 2 September 2009. Review of the vendor's LoC was completed on 29 September 2009. DSAWG granted accreditation on 10 March 2010 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 Engineering Special (ES) 8 to release 7.1(2). The JITC determined there was minor risk in approving this DTR because the addition of this software update provides enhancements for the Assured Services-Session Initiation Protocol (AS-SIP) Internet Protocol (IP) interface and does not change the interoperability results for their PBX solution. Therefore, JITC approves this DTR. DISA Network Systems Directorate has approved the Information Assurance posture of SUT in this DTR on 4 February 2011.

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 SUT'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).
- e. The IPv6 requirements specified in References (f) and (h).
- f. The softphone requirements specified in Reference (i).

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

Table 1. SUT Hardware and Software Components

Cisco Unified Communications Manager Version 7.1(2), with IOS Software Release 12.4(22)T2			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
Communication Managers <u>MCS7835I2, MCS7835H2, MCS7825H3, MCS7825H4,</u> MCS7835H1, MCS7845H1, MCS7845H2, MCS7825I4, MCS7835I1, MCS7845I1, MCS7845I2	7.1(2.30008.3)	Not Applicable	Processing/Signaling
Cisco 3845 , 3825 Integrated Services Router (Gateway)	IOS 12.4(22) T2	<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 2.)
		<u>NM HDV2 2T1/E1</u>	2-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)
		<u>NM HDV2 1T1/E1</u>	1-port T1/E1 IP Communications HD voice/fax NM 2 T1/E1 controllers (See note 2.)
		<u>VIC3 4FXS/DID</u>	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		<u>VIC3 2FXS</u>	Voice Interface Card, 2-port, Foreign Exchange Station
		<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 3.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		<u>EM3 HDA 8FXS/DID</u>	8-Port HD analog and digital extension module for voice and fax (See note 3.)

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

Table 1. SUT Hardware and Software Components (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
Cisco 2851, 2821, 2811 Integrated Services Router (Gateway)	IOS 12.4(22)T2	NM HD 2VE	2-slot IP communications enhanced voice/fax network module
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		EVM 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 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 2.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 2.)
		VIC3 4FXS/DID	Voice interface card, 4-port, RJ-11, foreign exchange station, DID
		EM3 HDA 8FXS/DID	8-port HD analog and digital extension module for voice and fax (See note 3.)
		VIC3 2FXS	Voice Interface card, 2-port, RJ-11, Foreign exchange station
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G (See note 4.)	P00308010100	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	SCCP70.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7931G	SCCP31.8-5-2S	Not Applicable	IP Phone (with push to talk handset or with standard handset)
CP-7911G and 7906G	SCCP11.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	SCCP41.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	SCCP42.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	SCCP45.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	SCCP75.8-5-2S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000400	Not Applicable	Expansion module
7915	B015-1-0-3	Not Applicable	Expansion module
7916	B015-1-0-3	Not Applicable	Expansion module
General Dynamics C4 Systems Sectéra® vIPer™ (See note 5.)	Release 1.0, Software ver.6.04	Not Applicable	IP Phone (with standard handset)
Telecore 2151	2AE-00056-0003	Not Applicable	IP Phone (with push-to-talk handset or with standard handset), 100 Mbps shared access ⁶
CIS Secure DTD-7961-T-SG-SC-SC-X-X (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G TEMPEST version with 100 Mbps SC Fiber LAN and PC interfaces, TSG Positive Disconnect, no speakerphone, shared access
CIS Secure DTD-7975-X-XSC-RJ-ME-SE (See note 7.)	SCCP75.8-5-2S	Not Applicable	7975G Standard with 1000 Mbps SC Fiber LAN and RJ45 PC interfaces, shared access
CRYPTEK CT915-V-P1-003 (See note 7.)	SCCP41.8-5-2S	Not Applicable	7961G IP phone, Fiber TEMPEST version with 100MB Fiber LAN and no shared access

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

Table 1. SUT Hardware and Software Components (continued)

Cisco Unified Communications Manager Version 7.1(2) with IOS Software Release 12.4(22)T2 (continued)			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
<u>Walker WS-2620</u>	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones
<u>Cisco IP Communicator</u> (See note 8.)	7.0.5	Not Applicable	Cisco Softphone Application

NOTES:

- Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.
- 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.
- The EM HDA 8FXS and EM3 HDA 8FXS/DID expansion modules require the EVM HD module. Up to two EM HDA 8FXS or EM3 HDA 8FXS/DID expansion modules are supported for each EVM HD.
- The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with the interim UCR IPv6 rules of engagement, Reference (h).
- This instrument is certified specifically with 2800 and 3800 series gateways with IOS 12.4(22) T2 or higher version listed on the UC APL.
- Although the Telecore 2151 supports both 100 Mbps and 1 Gbps shared access, due to MOS scores below the required 4.0 for 1 Gbps shared access, the Telecore 2151 is only certified for shared access at 100 Mbps.
- 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.
- Reference (i) is a DISA memo that stipulates interim softphone requirements that supersede the current UCR 2008 requirements until they are implemented in Change 1. The softphone shall be functionally identical to a traditional IP "Hard" telephone and will be required to provide voice features and functionality provided by a traditional IP "Hard" Telephone with following exceptions:
 - Audible and visual alerting to the end user of an incoming call, even if the application is running in the background.
 - Softphone application shall be exempt from reliability, availability and performance (packet loss, jitter, latency) requirements.
 - Microphone and speaker or headphone, or any other audio input/output device, Ethernet interface(s), and mouse (point and click) interaction.
 - IPv6 is not required.

LEGEND:

APL	Approved Product List	HD	High Density	PSTN	Public Switched Telephone Network
CP	Cisco Phone	HDA	High Density Analog	RJ	Registered Jack
DID	Direct Inward Dialing	IOS	Internetwork Operating System	SC	fiber connector (square push-in)
DISA	Defense Information Systems Agency	IP	Internet Protocol	SCCP	Skinny Call Control Protocol
DSN	Defense Switched Network	Ipv4	Internet Protocol version 4	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	UC	Unified Capabilities
Fax	facsimile	LAN	Local Area Network	UCR	Unified Capabilities Requirements
FXS	Foreign Exchange Station	Mbps	Megabits per second	V	Voice
Gbps	Gigabits per second	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
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)	MOS	Mean Opinion Score	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	The SUT T1 CAS interface was tested but did not meet all critical CRs and FRs. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1. 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. 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 exception: The SUT T1 ISDN PRI NI2 interface does not support NFAS. ²
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Certified	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 for use within the DSN. This interface is certified only for PSTN. This is not a required DSN 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 the following minor exception: The SUT gateway analog interface does not support required line features. ³ 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 (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. ⁴
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. The operational impact is minor. ³
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. ⁵
Conferencing	No	Not Certified	The SUT can support Meet-Me Conferencing through the optional MeetingPlace Express. ⁶ 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 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	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.
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 9.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN or ASLAN components posted on the UC APL. (See notes 4 and 10.)

Table 2. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	SUT T1 CAS interface was tested but did not meet requirements. The SUT T1 CAS interface is therefore not certified by JITC. This is not a required interface for a PBX 1. 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. 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: The SUT T1 ISDN PRI NI/2 interface does not support NFAS. ²
	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
NOTES:				
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. In addition, when the Busy Out condition is invoked across the T1 CAS interface, it causes the SUT 3845 and 2851 gateway T1 CAS interface to deregister from its current subscriber and reregister to an alternate subscriber and then within 1 to 5 minutes repeat the process and go back to its original subscriber. During this transition period, calls are unable to process to the SUT.			
2	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 stated they intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.			
3	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. Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog phones. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP phones only. These features are required for a PBX 1 for all instruments, however since this requirement is a new UCR requirement and the vendor has 18 months to develop it (July 2010), the operational impact is minor. Since the SUT test window started before the 18 month development window expired, DISA stated this new feature requirement does not apply. 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. Although the SUT does not support Precedence Call Waiting, 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. 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. 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. There is no operational impact.			
4	The SUT met all IPv6 requirements through testing and LoC with the following exception: The Cisco CP-7940G and CP-7960G end instruments did not meet dual stack IPv6 requirements. These end instruments represent legacy end instruments which are IPv4 only; however, the SUT met the minimum requirement for dual stack IPv6 end instruments with the other IP end instruments listed in this table and a dual stack call control agent in accordance with Reference (h).			
5	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.			
6	Meet-Me Conferencing can be met through the use of an optional adjunct conferencing system called the Cisco Meeting Place 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 Communication 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 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.			
9	Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (d).			

Table 2. SUT Interoperability Test Summary (continued)

NOTES continued:			
10	The following discrepancies noted with the SUT were adjudicated by DISA on 2 September 2009 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 MCS7835 and the MCS7825 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 can not be changed.		
e.	The SUT management workstation provided during testing did not assign DSCP values for OAM IP traffic.		
f.	IP phones are incorrectly tagging IPv6 TCP traffic during power up.		
g.	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.	The MCS7825H4 Communication Manager server stopped transmitting IP Traffic. The NIC failover must be disabled to correct this problem. The NIC failover is offered on this server but is not required for a PBX1. NIC failover is not certified for any server platform and should not be enabled. This setting will be annotated in the deployment guide for this server.		
j.	End Instruments, except for the Telecore 2151, do not support the manual configuration of the IPv6 default gateway.		
k.	Communication 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.		
LEGEND:			
802.1Q	Standards for Local and Metropolitan Area Networks: Virtual Bridged Local Area Networks	LoC	Letters of Compliance
		LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	Mbps	Megabits per second
		MCS	Media Convergence Servers
ANSI	American National Standards Institute	MFR1	Multi-Frequency Recommendation 1
APL	Approved Products List	MLPP	Multi-Level Precedence and Preemption
ASLAN	Assured Services Local Area Network	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	NIC	Network Interface Card
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	PMO	Program Management Office
DP	Dial Pulse	PNT	Preemption Notification Tone
DSCP	Differentiated Services Code Point	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DSS1	Digital Subscriber Signaling 1	Q.931	Signaling Standard for ISDN
DTMF	Dual Tone Multi-Frequency	Q.955.3	ISDN Signaling standard for E1 MLPP
E1	European Basic Multiplex Rate (2.048 Mbps)	RTCP	RTP Control Protocol
EI	End Instrument	RTP	Real-time Transport Protocol
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
IEEE	Institute of Electrical and Electronics Engineers		
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
JITC	Joint Interoperability Test Command	VoIP	Voice over Internet Protocol

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 (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)
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> • 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)
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		<ul style="list-style-type: none"> • 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 5.2.4.4.3 • 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
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)		<ul style="list-style-type: none"> • 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 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 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
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) • 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-Wire Analog only) • Alerting Signals and Tones (R) • S/T Reference Point (R: ISDN BRI only) • VoIP System Requirements (R: VoIP Phones only) 	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • MOS (R) • Secure Calls (R)
		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: 2-Wire Analog only) • Secure data (STE/STU-III) (R: 2-Wire Analog only) 	<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 (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) 		<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • Attendant Features (C) 		<ul style="list-style-type: none"> • UCR Section 5.2.1.2.2

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 (R) • Outgoing call trace (R) 	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • Preset Conferencing (C) • Meet-Me Conferencing (C) • Progressive Conferencing (C) 	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • Nailed-Up Connections (C) 	<ul style="list-style-type: none"> • UCR Section 5.2.1.8
DSN Hotline Services	No	<ul style="list-style-type: none"> • DSN Analog Hotline Service (C) 	<ul style="list-style-type: none"> • UCR Section 5.2.1.12
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 (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) 	<ul style="list-style-type: none"> • 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 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 (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) 	<ul style="list-style-type: none"> • 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	<ul style="list-style-type: none"> • 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) 	<ul style="list-style-type: none"> • 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	<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 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	<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 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	<ul style="list-style-type: none"> • GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R) 	<ul style="list-style-type: none"> • UCR Section 3

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) • Softphone Requirements 		<ul style="list-style-type: none"> • 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 section 5.2.12.8.2.8 • UCR section 5.2.12.8.2.9 • UCR section 5.2.12.8.2.10 • DISA Memo Reference (i)
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 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4
<p>NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.</p>				

Table 3. PBX 1 Requirements (continued)

LEGEND:					
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	SS7	Signaling System 7
DIACAP	DoD Information Assurance Certification and Accreditation Process	IT ITU-T	Telecommunication Union-Telecommunication Standardization Sector	STE	Secure Terminal Equipment
DISA	Defense Information Systems Agency		Multi-Frequency Recommendation 1	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry		minute	STU-III	Secure Telephone Unit -3rd generation
DoD	Department of Defense	kbps	kilobits per second	T.4	Standardization of Group 3 facsimile terminals for document transmission
DoDI	Department of Defense Instruction	Mbps MFR1	Megabits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DP	Dial Pulse		Multi-Level Precedence and Preemption	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
DS0	Digital Signal Level 0 (64 kbps)	min	Mean Opinion Score	UCR	Unified Capabilities Requirements
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MLPP	National ISDN Standard 1 or 2	UPS	Uninterruptible Power Supply
DSN	Defense Switched Network	MOS	Data format restricted to multiples of 56 kbps	VBD	Variable bit data
DTMF	Dual Tone Multi-Frequency	NI 1/2	Data format restricted to multiples of 64 kbps	VoIP	Voice over Internet Protocol
E&M	Ear and Mouth	NX56	Private Branch Exchange	VTC	Video Teleconferencing
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Private Branch Exchange 1	yr	year
EKTS	Electronic Key Telephone System	PBX	Pulse Code Modulation		
FTR	Federal Telecommunications Recommendation	PBX 1 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 <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.

JITC Memo, JTE, Extension of the Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2

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 0901201. The tracking number for the Cisco Internet Protocol Communicator is 0911101.

FOR THE COMMANDER:

Enclosure a/s


for BRADLEY A. CLARK
Acting Chief
Battlespace Communications Portfolio

Distribution (electronic mail):

Joint Staff J-6

Joint Interoperability Test Command, Liaison, TE3/JT1

Office of Chief of Naval Operations, CNO N6F2

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

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

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

DOT&E, Net-Centric Systems and Naval Warfare

U.S. Coast Guard, CG-64

Defense Intelligence Agency

National Security Agency, DT

Defense Information Systems Agency, TEMC

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

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

Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

- (c) Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2," 31 March 2010
- (d) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified Communication Manager Version 7.1(2) with Internetwork Operating System (IOS) Software Release 12.4(22)T2 (TN0901201)," 10 March 2010
- (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," December 2008
- (g) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006
- (h) Office of the Secretary of Defense, "Interim Unified Capabilities (UC) IPv6 Rules of Engagement (ROE)," 31 July 2009
- (i) Defense Information Systems Agency NS3 Memorandum, "Softphone Certification" 20 April 2009