



DEFENSE INFORMATION SYSTEMS AGENCY

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IN REPLY

REFER TO: Joint Interoperability Test Command (JTE)

9 June 2008

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Extension of the Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.3(1) Service Release (SR) 1A with Internetwork Operating System (IOS) Software Release 12.4(9) T1

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

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. Additional references are provided in the enclosure.
2. The CCM Version 4.3(1) SR 1A with IOS Software Release 12.4(9) T1 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 identified test discrepancies shown in reference (c) have an overall minor operational impact. The SUT components which are bolded and underlined in the tables throughout this certification letter are components that were tested in the JITC laboratory for this certification. The SUT components which are not bolded and not underlined, but also listed throughout the tables in this letter, are certified for joint use in the DSN as well. The JITC analysis determined these components contain the same hardware and software and are functionally identical to the tested components for interoperability certification purposes. The SUT was tested and is certified for voicemail with the Cisco Unity Voice Mail System found on the DSN APL. The SUT Digital Transmission Link Level 1 (T1) Channel Associated Signaling (CAS) interface was not tested and is not covered under this certification. The SUT T1 CAS interface is not certified by the JITC or authorized by the Program Management Office for use within the DSN. No other configurations, features, or functions, except those cited within this report, are certified by the JITC or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that could affect interoperability, but no later than three years from the date of the original memorandum (2 November 2007).
3. This is a certification based on a desktop review of the SUT. The original certification was granted based on interoperability testing conducted by JITC, review of vendor's Letters of Compliance (LoC), and review of patches applied to the SUT. Testing of the CCM Version

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4.3(1) with IOS Software Release 12.4(9) T1 was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 11 June through 13 July 2007. Review of vendor's LoC was completed on 17 July 2007. Regression testing of SR 1A was completed on 14 September 2007 and is documented in reference (c). A desktop review was requested to include the Cisco Unified IP Phone Expansion Module 7914. The desktop review was completed on 22 May 2008. JITC analysis determined the Cisco Unified IP Phone Expansion Module 7914 is certified for joint use within the DSN.

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 (d).
- b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.
- d. The overall system interoperability performance derived from test procedures listed in reference (f).
- e. Internet Protocol version 6 requirements specified in reference (e), paragraph 1.7, table 1-4, verified through vendor submission of LoC.

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Table 1. SUT Hardware and Software Components

CCM Version 4.3(1) SR 1A with IOS Software Release 12.4(9) T1			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
CallManagers <u>MCS7835H1, MCS7835H2,</u> <u>MCS7825H2, MCS7825H3,</u> MCS7825H, MCS7825-H1, MCS7825I1, MCS7835H, MCS7835I, MCS7835I1, MCS7845H, MCS7845H1, MCS7845I, MCS7845I1, MCS7845H2	CCM 4.3(1) SR 1A	Not Applicable	Processing/Signaling
Cisco 3745/3725 Multiservice Access Router (Gateway) (See note 2.)	IOS 12.4(9) T1	<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
		VVIC 1MFT T1	Voice/WAN Interface Card, 1-port, RJ-48, Multiflex Trunk T1
		<u>VVIC 2MFT T1</u>	Voice/WAN Interface Card, 2-port, RJ-48, Multiflex Trunk T1
		<u>VVIC 2MFT T1 DI</u>	Voice/WAN Interface Card, 1-port, RJ-48, Multiflex Trunk T1, Drop and Insert
Cisco 3845/3825 Integrated Services Router (Gateway)	IOS 12.4(9) T1	<u>NM HDV2</u>	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		<u>VVIC2 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 E1 (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
Cisco 2851 Integrated Services Router (Gateway)	IOS 12.4(9) T1	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
		<u>VVIC2 2MFT T1/E1</u>	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		<u>EM HDA 8FXS</u>	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 controller (See note 3.)
		VVIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		<u>VIC2 2FXS</u>	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station

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Table 1. SUT Hardware and Software Components (continued)

CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 (continued)			
Component (See note 1.)	Release	Sub-component (See note 1.)	Function
<u>CP-7940G</u> (See note 2.)	Ver: 8.0(4.0) App: P003080004000 Boot Load: CP-7940G PC0303010200	Not Applicable	IP Phone
<u>CP-7960G</u> (See note 2.)	Ver: 8.0(4.0) App: P003080004000 Boot Load: CP-7960G PC0303010001	Not Applicable	IP Phone
<u>CP-7970G</u>	SCCP70.8-2-1S Boot Load: 7970-64054100.BIN	Not Applicable	IP Phone
<u>CP-7971G-GE</u>	SCCP70.8-2-1S Boot Load: 7970-020706.BIN	Not Applicable	IP Phone
<u>CP-7911G,</u>	SCCP11.8-2-1S Boot Load: TnP11.3.0.1.20.BIN	Not Applicable	IP Phone
<u>CP-7941G</u>	SCCP41.8-2-1S Boot Load: 7941G_64- 020704128	Not Applicable	IP Phone
<u>CP-7941-GE</u>	SCCP41.8-2-1S Boot Load: 7941G-GE_64-02	Not Applicable	IP Phone
<u>CP-7961G</u>	SCCP41.8-2-1S Boot Load: 7961G- 64_020704128	Not Applicable	IP Phone
<u>CP-7961G-GE</u>	SCCP41.8-2-1S Boot Load: 7961G-GE_64- 020704128	Not Applicable	IP Phone
7914	Load: S000105000300	Not Applicable	Expansion module

LEGEND:

10/100BaseT - 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet	GE - Gigabit Ethernet	RJ - Registered Jack
App - application	GSCR - Generic Switching Center Requirements	SCCP - Skinny Client Control Protocol
CCM - Cisco CallManager	HD - High Density	SR - Service Release
CP - Cisco Phone	HDA - High Density Analog	SUT - System Under Test
DI - Drop and Insert	IOS - Internetwork Operating System	T1 - Digital Transmission Link Level 1 (1.544 Mbps)
DID - Direct Inward Dialing	IP - Internet Protocol	TDM - Time Division Multiplexing
DSN - Defense Switched Network	IPv6 - Internet Protocol version 6	V - Voice
E1 - European Basic Multiplex Rate (2.048 Mbps)	JITC - Joint Interoperability Test Command	VE - Voice/Fax Enhanced
EVM - Extension Voice Module	Mbps - Megabits per second	Ver - Version
Fax - facsimile	MCS - Media Convergence Server	VIC - Voice Interface Card
FXS - Foreign Exchange Station	MFT - Multiflex Trunk	VWIC - Voice WAN Interface Card
G - 10/100BaseT Ethernet	NM - Network Module	WAN - Wide Area Network
	PBX 1 - Private Branch Exchange 1	

NOTES:

- 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.
- 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 GSCR, 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.
 - There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 Integrated Services Routers 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.
- These components support both T1 and E1; however, the E1 interface was not tested. Therefore, the E1 interface is not authorized or approved for use within the DSN by the Program Management Office. Since E1 interfaces are not required for a PBX 1, there is no operational impact.
- 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.

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT offers a T1 CAS interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a PBX 1, there is no operational impact.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface. Since this is not a required interface for a PBX 1, there is no operational impact.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exception: Calls that are attempted over a trunk that is broken do not receive an ICA. ¹ The operational impact is minor.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The SUT does not support this interface. Since this is not a required interface for a PBX 1, there is no operational impact.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	This interface requires a minor configuration change to support secure devices. ² Met all CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number. ³ The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported. There is no operational impact because ISDN BRI NI 1/2 is not a required interface for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported. There is no operational impact because 2-Wire Proprietary Digital is not a required interface for a PBX 1.
VoIP	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number. ³ The operational impact is minor.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	No	Certified	Met all CRs and FRs with the following minor exceptions: Full compliance of DSN Common Call Features was not met. ^{4,5,6,7,8,9} The operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. There is no operational impact because Attendant features are not required for a PBX 1.
Public Safety	No	Certified	All public safety features are conditional. The SUT met all CRs and FRs for E911. The SUT does not support the other public safety features. There is no operational impact because public safety is not required for a PBX 1. ¹⁰
Preset Conferencing	No	Not Tested	This feature is not supported. There is no operational impact because Preset Conferencing is not required for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported. There is no operational impact because Nailed-up Connections are not required for a PBX 1.
Synchronization ¹¹	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 12.	See note 12.
VoIP			
Features and Capabilities	Critical	Status	Remarks
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASVALAN posted on the DSN APL. ^{13,14}

Table 2. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Tested	The SUT offers a T1 CAS interface; however, this interface was not tested and is not covered under this certification. Since this is not a required interface for a PBX 1, there is no operational impact.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	The SUT does not support this interface. Since this is not a required interface for a PBX 1, there is no operational impact.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	The SUT does not support this interface. Since this is not a required interface for a PBX 1, there is no operational impact.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ¹⁵	Met all critical CRs and FRs.
LEGEND: 10/100 BaseT - 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet ANSI - American National Standards Institute APL - Approved Products List ASVALAN - Assured Services Voice Application Local Area Network BRI - Basic Rate Interface CAS - Channel Associated Signaling CP - Cisco Phone CRs - Capability Requirements DISA - Defense Information Systems Agency DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DSS1 - Digital Subscriber Signaling 1 DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) E911 - Basic Emergency Service 911 FRs - Feature Requirements G - 10/100 BaseT Ethernet GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements ICA - Isolated Code Announcement IP - Internet Protocol IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU-T - International Telecommunication Union - Telecommunication Standardization Sector LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption ms - milliseconds NI 1/2 - National ISDN Standard 1 or 2 PBX 1 - Private Branch Exchange 1 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.931 - Signaling Standard for ISDN Q.955.3 - ISDN Signaling standard for E1 MLPP SS7 - Signaling System 7 SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.607 - ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1 T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TPC - Twisted Pair Copper VoIP - Voice over Internet Protocol				

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Table 2. SUT Interoperability Test Summary (continued)

<p>NOTES:</p> <ol style="list-style-type: none">1 Calls that are attempted over a trunk that is broken or in a remote busy-out condition do not receive an Isolated Code Announcement (ICA). The operational impact is minor because PBX 1 cannot support special command and control users and the calls are treated with a Blocked Precedence Announcement (PBA).2 To meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal, a configuration change was required on the analog gateways. 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.3 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number. The operational impact is minor because they can configure the diversion settings for all of the stations provisioned on the switch from a single location.4 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. The following common call features are not supported by the SUT and therefore are not covered in this certification: Precedence Call-Waiting, Selective Call Rejection, and Denied Originating Service. Since these features are not required for a PBX 1, there is no operational impact.5 All of the features on the IP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features.6 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. There is no operational impact.7 A short "ping" ring is not provided when calls are forwarded. There is a minor operational impact.8 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. The operational impact is minor because the preempted user receives Preempt Notification Tone (PNT) and the other members remain connected.9 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 GSCR requires this only for Precedence above ROUTINE calls. There is no operational impact.10 The SUT only supports E911 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 public safety features are not required for a PBX 1.11 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways.12 Security is tested by DISA-led Information Assurance test teams and published in a separate report.13 An IPv6 capable system or product, as defined in the GSCR, 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 by 30 June 2008:<ol style="list-style-type: none">a. Conformance 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.14 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 GSCR, 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:<ol style="list-style-type: none">a. The component must already be JITC certified and currently fielded within the DSN.b. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 Integrated Services Routers 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.15 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

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"> Framing (R) Line Code (R) Signaling (R: PRI only) Alarm and Restoral Requirements (R) Alarm and Restoral Requirements (C) WWNDP (R) Outpulsing digit formats (C: CAS only) Routing (C) Trunk Groups (C) Call Processing (R) 	<ul style="list-style-type: none"> GSCR Section 7.1.1 GSCR Section 7.1.4 GSCR Section 5 GSCR Section 7.1.4 GSCR Section 7.2.2 GSCR Section 4.5.1 GSCR Section 4.5.2 GSCR Section 4.2 GSCR Section 2.5.5 & 2.5.6 GSCR Section 4.1, 4.1.2, 4.1.3, 4.1.4, 4.1.5, 4.3.3, 4.5.1 GSCR Section 7.3 GSCR Section 2.3.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> PCM-24/PCM-30 Interoperation (C) Direct Inward Dialing (C) 	<ul style="list-style-type: none"> GSCR Section 2.3.2
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	<ul style="list-style-type: none"> MOS (R) MLPP (R) Secure calls (R) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Section 3.1, 3.2, 3.2.1, 3.2.2, 3.3, 3.4.2, 3.5.1 CJCSI 6215.01B
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> DISR
		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 6215.01B GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 CJCSI 6215.01B
		VTC	<ul style="list-style-type: none"> ITU-T H.320 (R: PRI only) 	<ul style="list-style-type: none"> DISR
DSN Line Interfaces				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> Directory Number Identification (R) Line signaling (R) Loop Start Line (R: 2-Wire Analog only) Alerting Signals and Tones (R) WWNDP (R) Call Treatments (R) 2W user access (R: 2-Wire Analog only) Analog busy/idle (R: 2-Wire Analog only) 	<ul style="list-style-type: none"> GSCR Section 2.1.1 GSCR Section 5.2.1 & 5.2.2 GSCR Section 5.2.1 GSCR Section 5.5 GSCR Section 4.5.1.2, 4.5.1.3, 4.5.1.4 & 4.5.1.5 GSCR Section 4.1.1 – 4.1.5 GSCR Section 4.3.3 GSCR Section 4.3.4.1
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Voice	<ul style="list-style-type: none"> MOS (R) Announcements (R) MLPP (R) Ground Start Line (R) Secure Calls (R) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Section 3.1.3 GSCR Section 3.4.3/3.9 GSCR Section 5.2.2 CJCSI 6215.01B
2-Wire Proprietary Digital	No	Facsimile	<ul style="list-style-type: none"> Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> DISR
VoIP IEEE 802.3u	No	Data	<ul style="list-style-type: none"> Modem (VBD) (R) 56 kbps switched data (R: BRI only) 64 kbps switched data (R: BRI only) NX56 synchronous BER (R: BRI only) NX64 synchronous BER (R: BRI only) Secure data (STE/STU-III) (R) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 CJCSI 6215.01B
		VTC	<ul style="list-style-type: none"> ITU-T H.320 (R: BRI only) 	<ul style="list-style-type: none"> DISR

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Common Features	No	<ul style="list-style-type: none"> • Selective call rejection (C) • Denied originating service (C) • Code restriction and diversion (C) • Call waiting (C) • Three-way calling (C) • Add-on transfer, conference calling, and call hold (C) • Call forwarding (C) • Call pick-up (C) 	<ul style="list-style-type: none"> • GSCR Section 2.1.2 • GSCR Section 2.1.3 • GSCR Section 2.1.4 • GSCR Section 2.1.5 • GSCR Section 2.1.6 • GSCR Section 2.1.7 • GSCR Section 2.1.8 • GSCR Section 2.1.9
Attendant	No	<ul style="list-style-type: none"> • Initiate all precedence levels (C) • Visual display (C) • Override class of service (C) • Override busy line (C) • Call deflection (C) • Auto recall (C) • Waiting queue (C) 	<ul style="list-style-type: none"> • GSCR Section 2.2.1 • GSCR Section 2.2.2 • GSCR Section 2.2.3 • GSCR Section 2.2.4 • GSCR Section 2.2.5 • GSCR Section 2.2.6 • GSCR Section 2.2.7
Public Safety	No	<ul style="list-style-type: none"> • Basic Emergency Service (911) (C) • Trace of terminating calls (C) • Outgoing call trace (C) 	<ul style="list-style-type: none"> • GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3
Preset Conferencing	No	<ul style="list-style-type: none"> • Support 10 bridges; 1 originator and 20 conferees per bridge (C) • Assign up to 20 address numbers per bridge (C) • Use KXX codes for bridge access (C) • Conference notification recorded announcement (C) • Auto retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C) 	<ul style="list-style-type: none"> • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6.1 • GSCR Section 2.6.2 • GSCR Section 2.6.3 • GSCR Section 2.6.4 • GSCR Section 2.6.5 • GSCR Section 2.7
Nailed-up Connections	No	<ul style="list-style-type: none"> • Between any two like terminations (C) • PCM-24 and PCM-30, both CAS and CCS (C) • Supervision passed end-to-end for A/D or D/A (C) • Monitored and auto reconfigure (C) • Support at least 10% of circuits as nailed-up (C) • Non-preemptable (C) 	<ul style="list-style-type: none"> • GSCR Section 2.8
Synchronization	Yes	<ul style="list-style-type: none"> • Line timing mode (R) • Internal Stratum 4 (R) 	<ul style="list-style-type: none"> • GSCR Section 11.1.1.2 • GSCR Section 11.1.2.2
Reliability	Yes	<ul style="list-style-type: none"> • GR-512-CORE (R) 	<ul style="list-style-type: none"> • GSCR Section 12
Security	Yes	<ul style="list-style-type: none"> • GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R) 	<ul style="list-style-type: none"> • GSCR Section 13

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)				
VoIP				
VoIP System	No	<p>VoIP function is conditional. If VoIP is provided, all of the following requirements must be met:</p> <ul style="list-style-type: none"> • Voice Quality with MOS of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security • Network management • Line timing • Internal Clock • Latency ≤ 60 milliseconds • Packet Loss • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3, para. A3.2.1 • GSCR App. 3, para. A3.3.2 & A3.3.3 • GSCR App. 3, para. A3.2.2 • GSCR App. 3, para. A3.3.4.4 • GSCR App. 3, para. A3.2.4 • GSCR App. 3, para. A3.2.5 • GSCR App. 3, para. A3.2.6 • GSCR App. 3, para. A3.2.6 • GSCR App. 3, para. A3.2.7 • GSCR App. 3, para. A3.3.1.3 • GSCR Section 1, para. 1.7 & App. 3, para. A3.2.8 	
Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN ¹	No	Trunking	<ul style="list-style-type: none"> • Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN ²	Yes	Access	<ul style="list-style-type: none"> • Alerting Signals and Tones (R) • Call Processing (C) • Call Treatments (R) • Analog busy/idle (R) 	<ul style="list-style-type: none"> • GSCR Section 5.5 • GSCR Section 4.4 • GSCR Section 4.1 • GSCR Section 4.3.4.1
		Voice	<ul style="list-style-type: none"> • MOS (C) • MLPP (C) • Secure calls (C) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Section 3 • CJCSI 6215.01B

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Table 3. PBX 1 Requirements (continued)

LEGEND:					
2W	- 2-Wire	E&M	- Ear and Mouth	NX64	- Data format restricted to multiples of 64 kbps
A/D	- Analog to Digital Conversion	G.711	- Standard for PCM of Voice Frequencies	para	- paragraph
ANSI	- American National Standards Institute	GR	- Generic Requirement	PBX 1	- Private Branch Exchange 1
App	- Appendix	GR-512-CORE	- LSSGR: Reliability, Section 12	PCM	- Pulse Code Modulation
BER	- Bit Error Ratio	GR-815	- Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-24	- Pulse Code Modulation - 24 Channels
BRI	- Basic Rate Interface			PCM-30	- Pulse Code Modulation - 30 Channels
C	- Conditional	GSCR	- Generic Switching Center Requirements	PRI	- Primary Rate Interface
CAS	- Channel Associated Signaling	H.320	- Standard for Narrowband VTC	PSTN	- Public Switched Telephone Network
CCS	- Common Channel Signaling	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCSI	- Chairman of the Joint Chiefs of Staff Instruction	IPv6	- Internet Protocol version 6	R	- Required
D/A	- Digital to Analog Conversion	ISDN	- Integrated Services Digital Network	SS7	- Signaling System 7
DIACAP	- DoD Information Assurance Certification and Accreditation Process	IT	- Information Technology	STE	- Secure Terminal Equipment
DISR	- DoD IT Standards Registry	ITU-T	- International Telecommunication Union-Telecommunication Standardization Sector	STIGs	- Security Technical Implementation Guides
DITSCAP	- DoD IT Security Certification and Accreditation Process	kbps	- kilobits per second	STU-III	- Secure Telephone Unit -3rd generation
DoD	- Department of Defense	KXX	- K= any number 2-8; X= any number 1-9	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DP	- Dial Pulse	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
DRSN	- Defense Red Switch Network	Mbps	- Megabits per second	TIA	- Telecommunications Industry Association
DSN	- Defense Switched Network	MFR1	- Multi-Frequency Recommendation 1	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DTMF	- Dual Tone Multi-Frequency	MLPP	- Multi-Level Precedence and Preemption	VBD	- Variable bit data
E1	- European Basic Multiplex Rate (2.048 Mbps)	MOS	- Mean Opinion Score	VoIP	- Voice over Internet Protocol
EIA	- Electronic Industries Alliance	NI 1/2	- National ISDN Standard 1 or 2	VTC	- Video Teleconferencing
		NX56	- Data format restricted to multiples of 56 kbps	WWNDP	- Worldwide Numbering and Dialing Plan

NOTES:
1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

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>.

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 0702902.

FOR THE COMMANDER:

Enclosure a/s



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ADDITIONAL REFERENCES

- (c) Joint Interoperability Test Command, Memo, JTE, "Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.3(1) Service Release (SR) 1A with Internetwork Operating System (IOS) Software Release 12.4(9) T1," 2 November 2007
- (d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (e) Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007
- (f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006