



DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND
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IN REPLY
REFER TO:

Battlespace Communications Portfolio (JTE)

23 March 2007

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.2(3) and CCM Version 4.2(3) Service Release (SR) 1, both with Internetworking Operating System (IOS) Software Release 12.4(9) T1 (Includes Voice over Internet Protocol [VoIP])

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 enclosure 1.

2. The CCM Version 4.2(3) 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 SUT meets the VoIP critical interoperability requirements with a certified Assured Services Voice Application Local Area Network (ASVALAN). Certified ASVALANs can be found on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>. After testing was complete and the certification letter for the SUT was generated, the vendor requested JITC to conduct a desktop review of CCM 4.2(3) SR1. JITC supported the vendor request and determined that the SR1 patches applied to the SUT had no impact on interoperability and that CCM 4.2(3) SR1 with IOS Software Release 12.4(9) T1 is functionally identical to the SUT; therefore it is also certified for joint use within the DSN as a PBX 1 and PBX 2. The identified test discrepancies shown in the Certification Testing Summary (enclosure 2) have an overall minor operational impact. The SUT offers a Digital Transmission Link Level 1 (T1) Channel Associated Signaling (CAS) trunk interface; however, due to critical interoperability discrepancies discovered during testing, it is not certified. This interface is not authorized or approved for connection to the DSN. There is no operational impact because T1 CAS is not a required interface for a PBX 1. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.

3. This certification is 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

JITC Memo, JTE, Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.2(3) and CCM Version 4.2(3) Service Release (SR) 1, both with Internetworking Operating System (IOS) Software Release 12.4(9) T1 (Includes Voice over Internet Protocol [VoIP])

Version 4.2(3) with IOS Software Release 12.4(9) T1 was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 30 October 2006 through 6 January 2007. Review of vendor's LoC was completed on 6 February 2007. After completion of the certification letter, on 23 March 2007, the vendor requested a desk top review of CCM 4.2(3)SR1 with IOS Software Release 12.4(9) T1, which JITC supported and completed on 19 April 2007. The date of this memorandum, 23 March 2007, reflects completion of testing and documentation review for the CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 and also applies to CCM Version 4.2(3) SR1 with IOS Software Release 12.4(9) T1. Enclosure 2 documents the test results and describes the tested network and system configurations. System interoperability should be verified before deployment in an operational environment that varies significantly from the test environment.

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 (c).
- b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- d. The overall system interoperability performance derived from test procedures listed in reference (e).
- e. Internet Protocol version 6 requirements specified in reference (d), paragraph 1.7, table 1-3, by 30 June 2008 in accordance with reference (f) verified through vendor submission of LoC.

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Table 1. SUT Hardware and Software Components

CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 (See note 1.)			
Component (See note 2.)	Release	Sub-component	Function
CallManagers <u>MCS7835H, MCS7835I,</u> <u>MCS7835H1,</u> MCS7825H, MCS 7835I, MCS7845H, MCS7845I, MCS7825-H1, MCS7825I1., MCS7845H1, MCS7845I1	CCM 4.2(3) (See note 1.)	Not Applicable	Processing/Signaling
<u>Cisco 3745/3725</u> Multiservice Access Router (Gateway) (See note 3.)	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 2-port RJ-48, Multiflex Trunk T1, Drop and Insert
<u>Cisco 3845/3825</u> 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 4.)
		<u>NM HDV2 2T1/E1</u>	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers E1 (See note 4.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 4.)
		<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 5.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		VVIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 4.)

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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) (See note 1.)			
Component (See note 2.)	Release	Sub-component	Function
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>VWIC2 2MFT T1/E1</u>	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 4.)
		<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 5.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 4.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 4.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 4.)
		<u>VIC2 2FXS</u>	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
<u>CP-7940G and CP-7960G</u> (See note 3.)	Ver: 8.0(4.0) App: P003080004000		IP Phone (with push-to-talk handset or with standard handset)
<u>CP-7970G and CP-7971G-GE</u>	Load: SCCP70.8-0-4SR1S		IP Phone (with push-to-talk handset or with standard handset)
<u>CP-7911G, CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE</u>	Load: SCCP11.8-0-4SR1S		IP Phone (with push-to-talk handset or with standard handset)
LEGEND:			
10/100BaseT	- 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet	G	- 10/100BaseT Ethernet
App	- application	GE	- Gigabit Ethernet
CCM	- Cisco CallManager	GSCR	- Generic Switching Center Requirements
CP	- Cisco Phone	HD	- High Density
DI	- Drop and Insert	HDA	- High Density Analog
DID	- Direct Inward Dialing	IOS	- Internetworking Operating System
DSN	- Defense Switched Network	IP	- Internet Protocol
DSP	- Digital Signal Processor	IPv6	- Internet Protocol version 6
E1	- European Basic Multiplex Rate (2.048 Mbps)	JITC	- Joint Interoperability Test Command
EVM	- Extension Voice Module	Mbps	- Megabits per second
Fax	- facsimile	MCS	- Media Call Service
FXS	- Foreign Exchange Station	MFT	- Multiflex Trunk
		NM	- Network Module
		RJ	- Registered Jack
		SCCP	- Skinny Client Control Protocol
		SR	- Service Release
		SUT	- System Under Test
		T1	- Digital Transmission Link Level 1 (1.544 Mbps)
		TDM	- Time Division Multiplexing
		V	- Voice
		VE	- Voice/Fax Enhanced
		Ver	- Version
		VIC	- Voice Interface Card
		VWIC	- Voice WAN Interface Card
		WAN	- Wide Area Network
NOTES:			
1 Testing was conducted on CCM Version 4.2(3). JITC conducted a desktop review which determined CCM Version 4.2(3) SR1 to be functionally identical and did not require further testing; therefore it is also certified for joint use.			
2 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.			
3 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 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:			
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 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.			
4 These components support both T1 and E1; however, the E1 interface was not tested is not authorized or approved for use within the DSN. Since E1 interfaces are conditional for a PBX 1, the operational impact is minor.			
5 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.			

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Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it will not be certified. 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 that are attempted over a trunk that is broken or in a remote busy-out condition do not receive ICA. ⁴ This interface is not authorized or approved for connection to the DSN. There is no overall operational impact because T1 CAS is conditional for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 CAS is conditional for a PBX 1.
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 or in a remote busy-out condition do not receive an ICA. ⁴ The operational impact is minor.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 PRI is conditional for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs a minor configuration change ⁵ and the following minor exceptions: The SUT does not support an MLPP global diversion number. ⁶ The BNEA is not provided. ⁷ 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 conditional 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 conditional 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. ⁶ The BNEA is not provided. ⁷ 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. ^{8, 9, 10, 11, 12, 13} The operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. There is no operational impact because Attendant is conditional 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 a critical requirement for a PBX 1. ¹⁴
Preset Conferencing	No	Not Tested	This feature is not supported. There is no operational impact because Preset Conferencing is conditional 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 conditional for a PBX 1.
PAT	No	Not Tested	This feature is not supported. There is no operational impact because PAT is conditional for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported. There is no operational impact because DSN Hotline Services are conditional for a PBX 1.
Network Management	No	Not Tested	This feature is not supported. There is no operational impact because Network Management is conditional for a PBX 1.
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. There is no operational impact because ISDN Services are conditional for a PBX 1.
Synchronization ¹⁵	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 16.	See note 16.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASVALAN posted on the JITC TSSI program web page: http://jitc.fhu.disa.mil/tssi See notes 17 and 18.

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Table 2. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it will not be certified. 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 There is no overall operational impact because T1 CAS is conditional for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 CAS is conditional for a PBX 1.
	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	This interface is not supported. There is no operational impact because E1 PRI is conditional for a PBX 1.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ¹⁹	Met all critical CRs and FRs.
LEGEND: ANSI - American National Standards Institute ASVALAN - Assured Services Voice Application Local Area Network BNEA - Busy Not Equipped Announcement 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 EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements ICA - Isolated Code Announcement IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU-T - International Telecommunication Union - Telecommunication Standardization Sector JITC - Joint Interoperability Test Command 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 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PNT - Preempt Notification Tone 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 TDM - Time Division Multiplexing TPC - Twisted Pair Copper TSSI - Telecom Switched Services Interoperability VoIP - Voice over Internet Protocol				

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Table 2. SUT Interoperability Test Summary (continued)

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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.
4	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 they are treated with a Blocked Precedence Announcement (BPA) and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.
5	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.
6	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.
7	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.
8	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: Call-Waiting, Selective Call Rejection, and Denied Originating Service. All of these features are conditional for a PBX 1. There is no operational impact.
9	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.
10	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.
11	A short "ping" ring is not provided when calls are forwarded. There is a minor operational impact.
12	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.
13	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.
14	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.
15	To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways.
16	Security is tested by DISA-led Information Assurance test teams and published in a separate report.
17	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: 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.
18	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: 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 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.
19	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) Alarms (R) 	<ul style="list-style-type: none"> GSCR Section 7 GSCR Section 7 GSCR Section 5 GSCR Section 2.5.7, 7.1.4 & 7.2.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> WWNDP (R) Outpulsing digit formats (C: CAS only) Routing (C) Trunk Groups (C) Call Processing (R) CAS to CCS trunk interworking (C) PCM-24/PCM-30 Interoperation (C) Direct Inward Dialing (C) 	<ul style="list-style-type: none"> 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 GSCR Section 3.10 GSCR Section 7.3 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 CJCSI 6215.01B
		Facsimile	<ul style="list-style-type: none"> Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> DISR
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	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 GSCR Section 5.2.1 GSCR Section 5.5 GSCR Section 4.5 GSCR Section 4.1 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) Secure Calls (R) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Section 3.1.3 GSCR Section 3.4.3/3.9 CJCSI 6215.01B
		Facsimile	<ul style="list-style-type: none"> Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> DISR
2-Wire Proprietary Digital	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

JITC Memo, JTE, Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.2(3) and CCM Version 4.2(3) Service Release (SR) 1, both with Internetworking Operating System (IOS) Software Release 12.4(9) T1 (Includes Voice over Internet Protocol [VoIP])

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) • Tandem call trace (C) • Trace of a call in progress (C) 	<ul style="list-style-type: none"> • GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3 • GSCR Section 2.4.4 • GSCR Section 2.4.5
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 retrieval 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
PAT	No	<ul style="list-style-type: none"> • Classmark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C) 	<ul style="list-style-type: none"> • GSCR Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 2.11.1.10
DSN Hotline Services	No	<ul style="list-style-type: none"> • Hotline restrictions (C) • Auto initiate (C) • Analog and digital (C) • Subscription basis (C) • Protected hotline calling (C) • WWNDP interoperable (C) 	<ul style="list-style-type: none"> • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4 • GSCR Section 2.12.5

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Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Network Management	No	<ul style="list-style-type: none"> • Interfaces (C) • Measurements and data generation (C) • Fault management (C) • Configuration management (C) • Accounting management (C) • Performance management (C) • Network management controls (C) • Remote access (C) 	<ul style="list-style-type: none"> • GSCR Section 9.1 • GSCR Section 9.2 • GSCR Section 9.3 • GSCR Section 9.4 • GSCR Section 9.5 • GSCR Section 9.6 • GSCR Section 9.7 • GSCR Section 9.8
ISDN Services	No	<ul style="list-style-type: none"> • Electronic Key Telephone Systems (EKTS) (C) 	<ul style="list-style-type: none"> • GSCR Section 10, table 10-3
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
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 \leq 60 milliseconds • Packet Loss • IPv6 capable 	<ul style="list-style-type: none"> • GSCR Appendix 3 • GSCR Appendix 3, paragraph 1.7

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Table 3. PBX 1 Requirements (continued)

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

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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 tracking number for the SUT is 0609301.

FOR THE COMMANDER:



MANUEL H. GARCIA, JR.

Chief

Battlespace Communications Portfolio

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ADDITIONAL REFERENCES

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (d) Defense Information Systems Agency (DISA), "Defense Switched Network (DSN) Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005
- (e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006
- (f) Executive Office of the President, "Transition Planning for Internet Protocol version 6 (IPv6)," 2 August 2005

CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** Cisco CallManager (CCM) Version 4.2(3), hereinafter referred to as the System Under Test (SUT), and CCM Version 4.2(3) Service Release (SR) 1 both with Internetworking Operating System (IOS) Software Release 12.4(9) T1, including Voice over Internet Protocol (VoIP).
- 2. PROPONENT.** Defense Information Systems Agency (DISA).
- 3. PROGRAM MANAGER.** Mr. Howard Osman, GS23, Room 5W23, 5275 Leesburg Pike, Falls Church, VA 22041, E-mail: Howard.Osman@disa.mil.
- 4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.
- 5. SYSTEM UNDER TEST DESCRIPTION.** The SUT is a Private Branch Exchange (PBX) 1. The SUT supports American National Standards Institute (ANSI) T1.619a Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2). The SUT offers a T1 Channel Associated Signaling (CAS) trunk interface; however, due to critical interoperability discrepancies discovered during testing, it will not be certified. This interface is not authorized or approved for connection to the DSN. There is no operational impact because T1 CAS is not a required interface for a PBX 1. The SUT VoIP configuration consists of CallManagers running the CCM software, gateways, and IP telephones. The CCM is the software-based call-processing component of the Cisco enterprise Internet Protocol (IP) telephone solution. The CCM software is a client-server application, loaded on a Personal Computer (PC) that is running a Cisco modified version of Windows 2000 Server. The CCM software provides telephony features and capabilities to packet telephony network devices such as IP phones. The CallManagers tested were the Media Call Service (MCS)7835H, MCS7835H1, and MCS7835I1. The following CallManagers were not tested; however, they utilize the same IOS software and hardware: MCS7835I, MCS7825H, MCS7825H1, MCS7825I, MCS7845H, MCS7845H1, MCS7845I, MCS7845I1. JITC analysis determined them to be functionally identical for interoperability certification purposes.

The 3745, 3845, and 2851 gateway routers are included in this tested architecture. The SUT gateway routers are scalable. The 2851 has one Network Module (NM) slot, one High-Density Extension Voice Module (EVM-HD) slot, and four High-Performance Wide Area Network (WAN) Interface Card (WIC) slots. These slots can be populated with up to 12 T1 trunks or 52 foreign-exchange-station (FXS) ports.

The 3845 has four NM slots and four HWIC slots. Each NM slot on the 3845 can accommodate a standard NM, an enhanced-network-module (NME) or an EVM-HD. The 3845 supports up to 24 T1 trunks or 88 FXS ports.

The 3745 has four NM slots and four WIC slots. The WIC slots are not capable of supporting Voice WICs (VWIC). The 3845 supports up to 18 T1 trunks or 48 FXS ports.

The 3725 and 3825 gateway routers were not tested; however, they utilize the same IOS software and hardware. JITC analysis determined them to be functionally identical for interoperability certification purposes.

6. OPERATIONAL ARCHITECTURE. The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including PBXs. The Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.

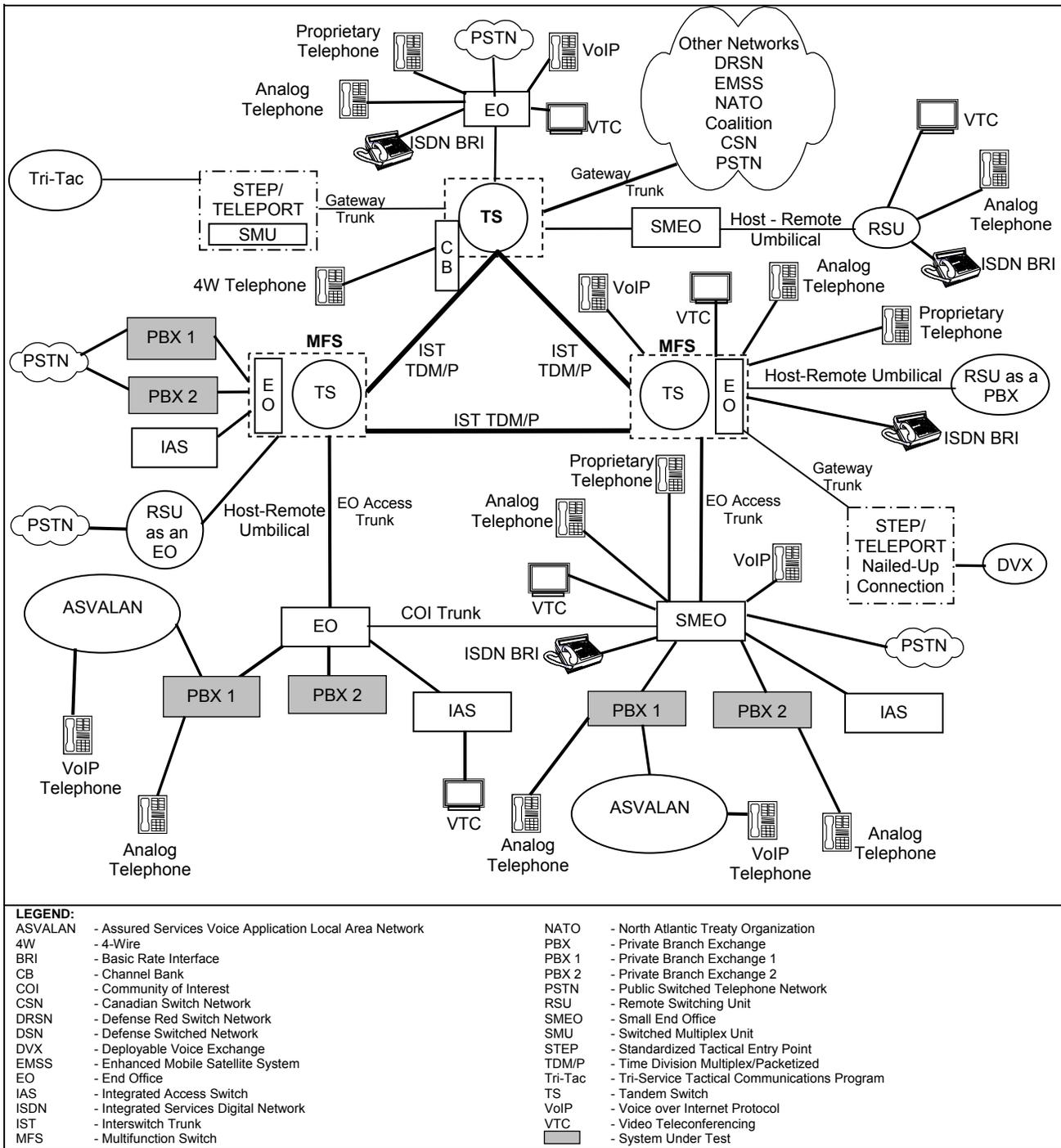


Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in table 2-1. These requirements are derived from:

a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services."

b. GSCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC).

c. GSCR PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC.

Table 2-1. 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) • Alarms (R) 	<ul style="list-style-type: none"> • GSCR Section 7 • GSCR Section 7 • GSCR Section 5 • GSCR Section 2.5.7, 7.1.4 & 7.2.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> • WWNDP (R) • Outpulsing digit formats (C: CAS only) • Routing (C) • Trunk Groups (C) • Call Processing (R) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (C) • Direct Inward Dialing (C) 	<ul style="list-style-type: none"> • 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 • GSCR Section 3.10 • GSCR Section 7.3 • 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 • CJCSI 6215.01B
		Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	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

Table 2-1. PBX 1 Requirements (continued)

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 • GSCR Section 5.2.1 • GSCR Section 5.5 • GSCR Section 4.5 • GSCR Section 4.1 • 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) • Secure Calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Section 3.1.3 • GSCR Section 3.4.3/3.9 • CJCSI 6215.01B
		Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
2-Wire Proprietary Digital	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
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) • Tandem call trace (C) • Trace of a call in progress (C) 		<ul style="list-style-type: none"> • GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3 • GSCR Section 2.4.4 • GSCR Section 2.4.5
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 retrieval 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

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required or Conditional	References
PAT	No	<ul style="list-style-type: none"> • Classmark for/not for PAT screening (C) • 7 PAT mechanisms (C) • Outgoing call screening (C) • Functional structure (C) • Simultaneous calls limitation (C) • Overflow process (C) • Decrementing call-in-progress count (C) • Call treatment (C) • Queuing (C) • Attendant calls (C) • Operations measurement registers (C) • Maintenance and Administration of thresholds (C) 	<ul style="list-style-type: none"> • GSCR Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 2.11.1.10
DSN Hotline Services	No	<ul style="list-style-type: none"> • Hotline restrictions (C) • Auto initiate (C) • Analog and digital (C) • Subscription basis (C) • Protected hotline calling (C) • WWNDP interoperable (C) 	<ul style="list-style-type: none"> • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4 • GSCR Section 2.12.5
Network Management	No	<ul style="list-style-type: none"> • Interfaces (C) • Measurements and data generation (C) • Fault management (C) • Configuration management (C) • Accounting management (C) • Performance management (C) • Network management controls (C) • Remote access (C) 	<ul style="list-style-type: none"> • GSCR Section 9.1 • GSCR Section 9.2 • GSCR Section 9.3 • GSCR Section 9.4 • GSCR Section 9.5 • GSCR Section 9.6 • GSCR Section 9.7 • GSCR Section 9.8
ISDN Services	No	<ul style="list-style-type: none"> • Electronic Key Telephone Systems (EKTS) (C) 	<ul style="list-style-type: none"> • GSCR Section 10, table 10-3
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
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 Appendix 3 • GSCR Appendix 3, paragraph 1.7

Table 2-1. PBX 1 Requirements (continued)

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

8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC’s Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system’s required functions and features was conducted using the test configuration depicted in figure 2-2. The Assured Services Voice Application Local Area Network (ASVALAN) is depicted in figure 2-3. The SUT was tested as the end-point in relation to the other switches.

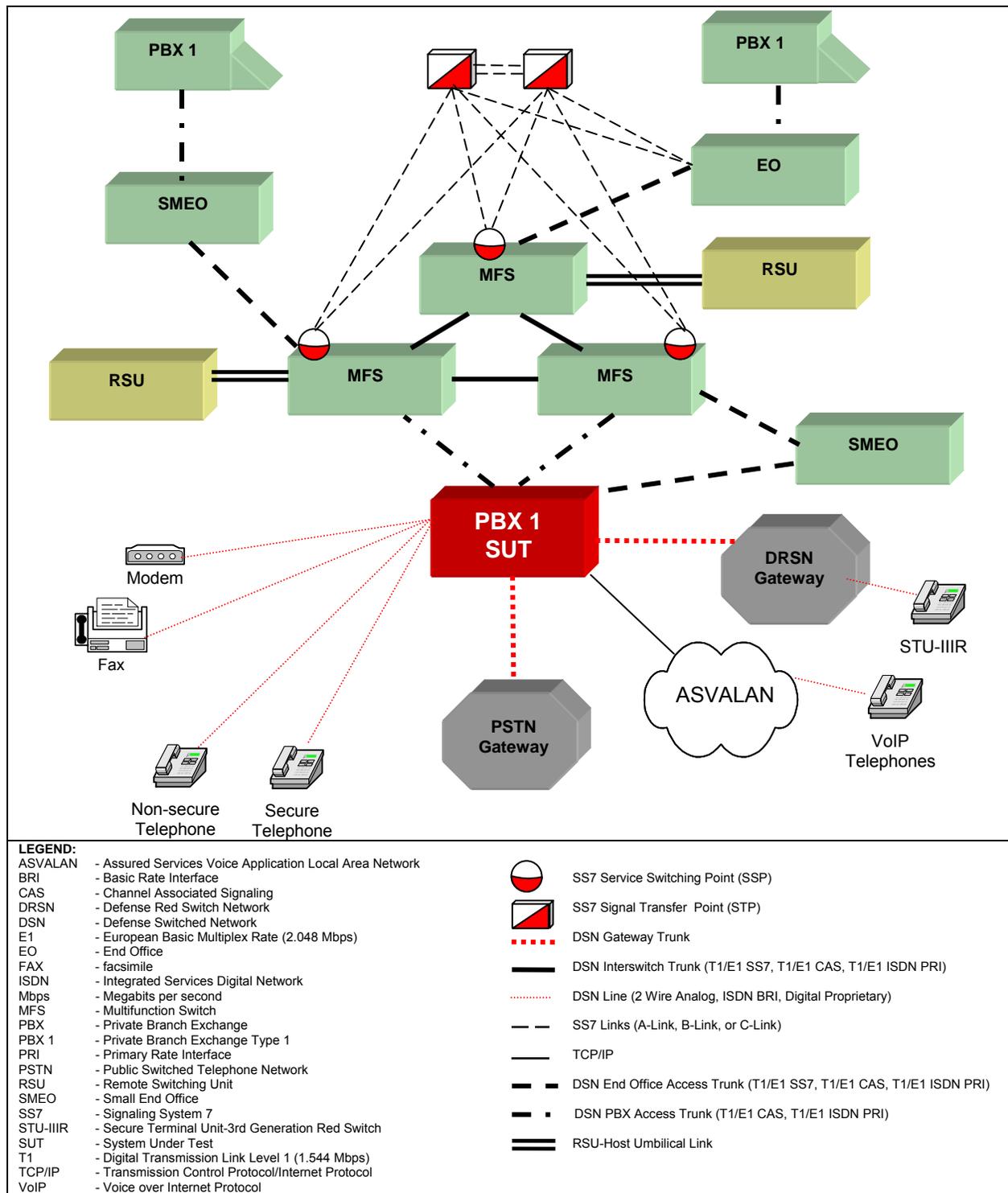


Figure 2-2. Test Configuration

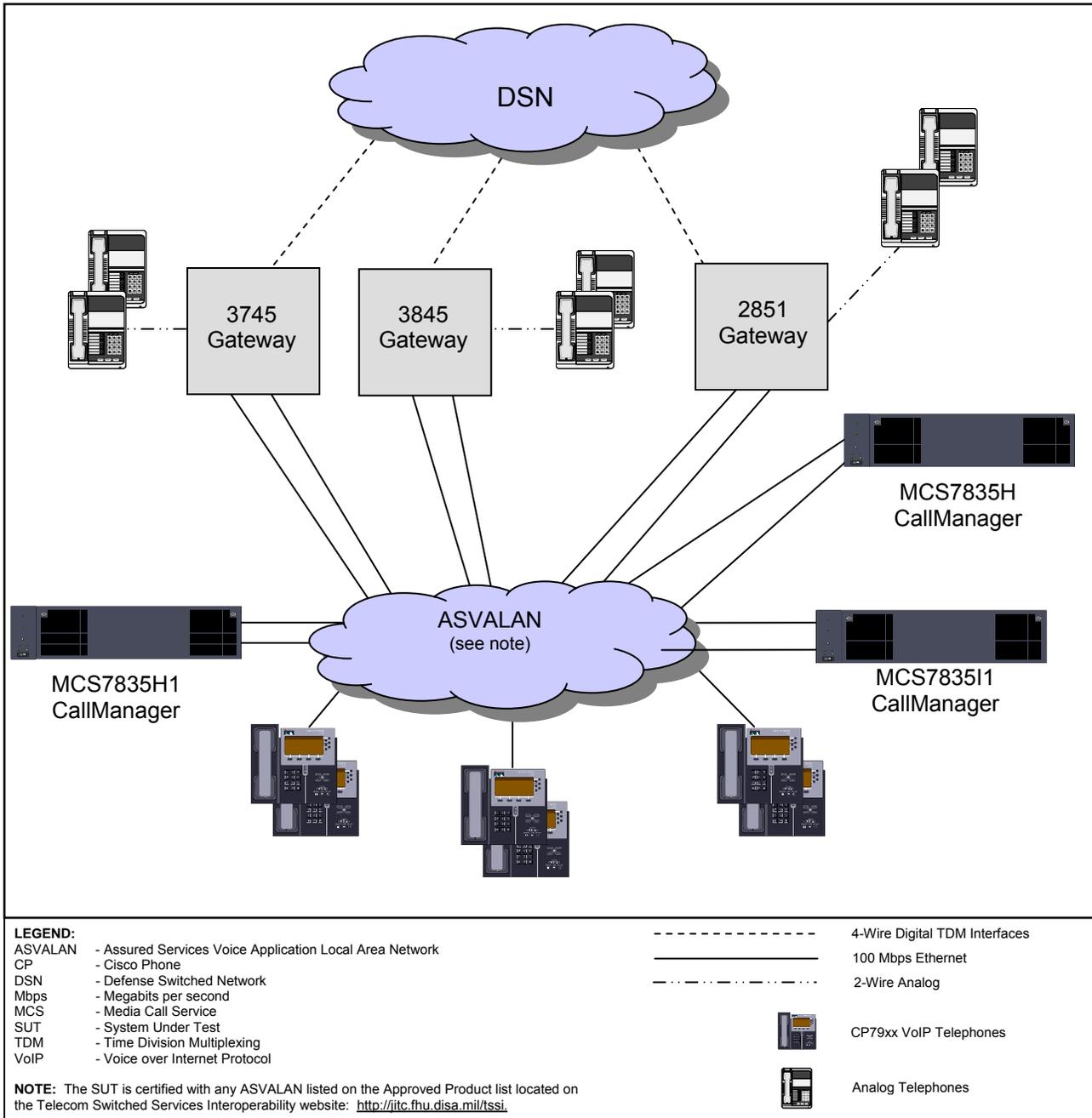


Figure 2-3. ASVALAN

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware and software used in the test.

Table 2-2. Tested System Configurations

System Name		Software Release	
Nortel CS 2100		Succession Enterprise 08 (SE08)	
Siemens EWSD		19d with Patch Set 46	
Lucent 5ESS		5E16.2 Software Update 06-0002	
Avaya S8700		Communication Manager (CM) 3.0 (R013x.00.0.340.3: Super Patch 11815)	
Nortel DSN CS1000M Single Group		4.5w	
CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 (See note 1.)			
Component (See note 2.)	Release	Sub-component	Function
CallManagers <u>MCS7835H</u> , <u>MCS7835I1</u> , <u>MCS7835H1</u> , MCS7825H, MCS 7835I, MCS7845H, MCS7845I, MCS7825-H1, MCS7825I1,, MCS7845H1, MCS7845I1	CCM 4.2(3) (See note 1.)	Not Applicable	Processing/Signaling
Cisco 3745/3725 Multiservice Access Router (Gateway) (See note 3.)	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 2-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 4.)
		<u>NM HDV2 2T1/E1</u>	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers E1 (See note 4.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 4.)
		<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 5.)
		<u>EVM HD 8FXS/DID</u>	HD analog and digital extension module for voice and fax
		VVIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 4.)

Table 2-2. Tested System Configurations (continued)

CCM Version 4.2(3) with IOS Software Release 12.4(9) T1 (continued) (See note 1.)																																													
Component (See note 2.)	Release	Sub-component	Function																																										
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																																										
		VVIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 4.)																																										
		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 5.)																																										
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 4.)																																										
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 4.)																																										
		VVIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 4.)																																										
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station																																										
CP-7940G and CP-7960G (See note 3.)	Ver: 8.0(4.0) App: P003080004000		IP Phone (with push-to-talk handset or with standard handset)																																										
CP-7970G and CP-7971G-GE	Load: SCCP70.8-0-4SR1S		IP Phone (with push-to-talk handset or with standard handset)																																										
CP-7911G, CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	Load: SCCP11.8-0-4SR1S		IP Phone (with push-to-talk handset or with standard handset)																																										
LEGEND:																																													
<table border="0"> <tr> <td>5ESS - Class 5 Electronic Switching System</td> <td>EWSD - Elektronisches Wählsystem Digital</td> <td>MFT - Multiflex Trunk</td> </tr> <tr> <td>10/100BaseT- 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet</td> <td>Fax - facsimile</td> <td>NM - Network Module</td> </tr> <tr> <td>App - application</td> <td>FXS - Foreign Exchange Station</td> <td>RJ - Registered Jack</td> </tr> <tr> <td>CCM - Cisco CallManager</td> <td>G - 10/100BaseT Ethernet</td> <td>SCCP - Skinny Client Control Protocol</td> </tr> <tr> <td>CP - Cisco Phone</td> <td>GE - Gigabit Ethernet</td> <td>SR - Service Release</td> </tr> <tr> <td>CS - Communication Server</td> <td>GSCR - Generic Switching Center Requirements</td> <td>SUT - System Under Test</td> </tr> <tr> <td>DI - Drop and Insert</td> <td>HD - High Density</td> <td>T1 - Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>DID - Direct Inward Dialing</td> <td>HDA - High Density Analog</td> <td>TDM - Time Division Multiplexing</td> </tr> <tr> <td>DSN - Defense Switched Network</td> <td>IOS - Internetworking Operating System</td> <td>V - Voice</td> </tr> <tr> <td>DSP - Digital Signal Processor</td> <td>IP - Internet Protocol</td> <td>VE - Voice/Fax Enhanced</td> </tr> <tr> <td>E1 - European Basic Multiplex Rate (2.048 Mbps)</td> <td>IPv6 - Internet Protocol version 6</td> <td>Ver - Version</td> </tr> <tr> <td>EVM - Extension Voice Module</td> <td>JITC - Joint Interoperability Test Command</td> <td>VIC - Voice Interface Card</td> </tr> <tr> <td></td> <td>Mbps - Megabits per second</td> <td>VVIC - Voice WAN Interface Card</td> </tr> <tr> <td></td> <td>MCS - Media Call Service</td> <td>WAN - Wide Area Network</td> </tr> </table>				5ESS - Class 5 Electronic Switching System	EWSD - Elektronisches Wählsystem Digital	MFT - Multiflex Trunk	10/100BaseT- 10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet	Fax - facsimile	NM - Network Module	App - application	FXS - Foreign Exchange Station	RJ - Registered Jack	CCM - Cisco CallManager	G - 10/100BaseT Ethernet	SCCP - Skinny Client Control Protocol	CP - Cisco Phone	GE - Gigabit Ethernet	SR - Service Release	CS - Communication Server	GSCR - Generic Switching Center Requirements	SUT - System Under Test	DI - Drop and Insert	HD - High Density	T1 - Digital Transmission Link Level 1 (1.544 Mbps)	DID - Direct Inward Dialing	HDA - High Density Analog	TDM - Time Division Multiplexing	DSN - Defense Switched Network	IOS - Internetworking Operating System	V - Voice	DSP - Digital Signal Processor	IP - Internet Protocol	VE - Voice/Fax Enhanced	E1 - European Basic Multiplex Rate (2.048 Mbps)	IPv6 - Internet Protocol version 6	Ver - Version	EVM - Extension Voice Module	JITC - Joint Interoperability Test Command	VIC - Voice Interface Card		Mbps - Megabits per second	VVIC - Voice WAN Interface Card		MCS - Media Call Service	WAN - Wide Area Network
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<ol style="list-style-type: none"> Testing was conducted on CCM Version 4.2(3). JITC conducted a desktop review which determined CCM Version 4.2(3) SR1 to be functionally identical and did not require further testing; therefore it is also certified for joint use. 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: <ol style="list-style-type: none"> 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 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 is not authorized or approved for use within the DSN. Since E1 interfaces are conditional for a PBX 1, the operational impact is minor. 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. 																																													

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces

(a) The SUT met all critical CRs and FRs for T1 ISDN PRI NI 1/2 ANSI T1.619a. This was the only DSN trunk interface certified.

(b) The SUT supports T1 CAS; however, due to critical interoperability discrepancies it will not be certified. This interface is not authorized or approved for connection to the DSN. There is no operational impact because T1 CAS is conditional for a PBX 1. The following paragraphs detail discrepancies that have a major operational impact for T1 CAS interoperability that were discovered during testing:

1. T1 CAS wink start recognition is not within the required tolerance of 100 milliseconds (ms) to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.

(c) A discrepancy exists that is associated with the monitoring tool that SUT uses to check the status of the trunk interfaces on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks. During a remote busy-out condition, the SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. Multi-Level Precedence and Preemption (MLPP) interaction was tested during this condition for both of the trunk interfaces that are supported. The T1 PRI trunk interface interaction met interoperability requirements while in this condition. However, it was observed that calls could not be originated over a T1 CAS while this condition is present. This discrepancy has a minor operational impact for T1 PRI operation, and it has a major operational impact for T1 CAS operation.

(d) Calls that are attempted over a trunk that is broken or in a remote busy-out condition do not receive Isolated Code Announcement (ICA). The operational impact is minor because they are treated with a Blocked Precedence Announcement

(BPA) and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Analog (GR-506-CORE) and VoIP DSN Line Interfaces. The following paragraphs detail the discrepancies that have a minor operational impact for interoperability that were discovered during testing:

(a) 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.

(b) 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 Busy Not Equipped Announcement (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.

(c) 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) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions: 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: Call-Waiting, Selective Call Rejection, and Denied Originating Service. There is no operational impact because they are not critical requirements for a PBX 1. Refer to table 2-3 for a list of the Common Features supported for the phone types and associated testing observations.

Table 2-3. SUT Common Call Feature Availability

Call Feature	Phone Type	
	Analog	IP ¹
Precedence Call Waiting	Not Supported	Not Supported ²
Call Hold	Not Supported	Passed
Call Forwarding No Answer	Passed	Passed
Call Forwarding Busy	Passed	Passed
Call Forwarding Variable	Not Supported	Passed ³
Three-Way Calling	Not Supported	Passed ⁴
Call Transfer	Not Supported	Passed
Multi-line Hunt Service	Passed ⁵	Passed ⁵
Call Pickup	Not Supported	Passed
Selective Call Rejection	Not Supported	Not Supported
Denied Originating Service	Not Supported	Not Supported

LEGEND:
IP - Internet Protocol
MLPP - Multi-Level Precedence and Preemption
SUT - System Under Test
VoIP - Voice over Internet Protocol

NOTES:
1 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. There is no operational impact.
2 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.
3 A short "ping" ring is not provided when calls are forwarded. There is a minor operational impact.
4 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.
5 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.

(b) Attendant. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(c) Public Safety. Met all CRs and FRs with the following minor exception: Full compliance of DSN Public Safety Features was not met. The SUT only supports the Basic Emergency Service 911 public safety feature. 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, Trace of a call in progress. There is no operational impact because it is not a critical requirement for a PBX 1.

(d) Preset Conferencing. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(e) Nailed-up Connections. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(f) Precedence Access Threshold. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(g) DSN Hotline Services. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(h) Network Management. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(i) ISDN Services Electronic Key Telephone System. This feature is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.

(j) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization. To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways.

(k) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and verified by vendor LoC.

(l) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report.

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) and the Defense Red Switch Network (DRSN) Network Gateways. The only interface certified for the PSTN is T1 ISDN PRI NI 1/2 (ANSI T1.607) and the only interface certified for the DRSN is Twisted Pair Copper 2-wire analog (GR-506-CORE). The SUT offers a T1 CAS trunk interface; however, due to critical interoperability discrepancies discovered during testing, it will not be certified. This interface is not authorized or approved for connection to the DSN. There is no operational impact because it is not a critical requirement. Refer to paragraph 11.a.(1)(b) for specific information.

(5) VoIP. The SUT is certified with any certified ASVALAN, which can be found on the TSSI web page: <http://jitc.fhu.disa.mil/tssi>.

(a) VoIP System. The GSCR, appendix 3, section A3.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the GSCR, appendix 3, VoIP calls shall have a Mean Opinion Score (MOS) of 4.0 or better. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.34 with all calls having a MOS of at least 4.0. Inter-switch calls had an average MOS of 4.36 with all calls having an MOS of at least 4.0. These averages were based on a total of 545 intra-switch and inter-switch calls.

2. Class of Service (CoS) and Quality of Service (QoS). The GSCR, appendix 3, section A3.3.2, outlines several methodologies to implement CoS

and QoS. The 802.1Q at the Data Link Layer (L2) and Differentiated Services Code Point (DSCP) at the Network Layer (L3) were two CoS mechanisms that the certified network products employed. The SUT provides CoS by assignment of an 802.1Q tag and/or a DSCP value. Switches within the topology were configured with multiple Virtual LANs (VLANs) to separate data from voice traffic. The 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice VLAN traffic was assigned to a high priority queue, ensuring voice traffic took precedence over data traffic. Priority bits for L2 voice signaling was set for 6 and voice media was set for 5. The L3 DSCP value for voice signaling was set for 48 and voice media for 46, in the tested configuration. By using the Ixia test equipment, a data load of 1.2 times the total link aggregate, was put on the certified ASVALAN to insure that all CoS and QoS settings were working properly. Captures indicated all tags were set properly.

3. Codec. In accordance with the GSCR, appendix 3, section A3.2.2, the ITU-T G.711 Pulse-Code Modulation (PCM) codec with a 20 ms packet fill was required and was met by the SUT VoIP solution.

4. Traffic Engineering.

a. Phones. The Cisco IP phones that met the critical interoperability requirements for certification were the 7911G, 7940G, 7941G, 7941G-GE, 7960G, 7961G-GE, 7961GE, 7970G, 7971G-GE phones. The phones are capable of shared access (i.e., same switch port is shared by PC and IP phone). The 7970G and 7971G-GE phones are capable of web browsing; however, this feature was not tested, is not covered by this certification, and is not authorized for use within the DSN. All IP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

b. Scalability. The MCS7835s can support a maximum of 2,500 IP subscribers, the MCS7825 can support 1,000 IP subscribers, and the MCS7845 can support 7,500 IP subscribers. However; the configurations range from 1,000 IP subscribers with two MCS7825s to 30,000 subscribers with eight MCS7845s. The recommendation is to consult an engineer to determine the appropriate configurations. The SUT is certified with any certified ASVALAN, which can be found on the TSSI web page: <http://jitc.fhu.disa.mil/tssi>. The ASVALAN can be scaled to meet the maximum subscribers as long as it is composed of the equipment and software listed in this certification, and meets the traffic engineering constraints contained in the GSCR, appendix 3.

5. MLPP. The GSCR, section 3, details the requirements for MLPP. The SUT met all CRs and FRs for VoIP MLPP interaction.

(b) Security. Security requirements in accordance with the GSCR, appendix 3, were verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel.

(c) Network Management (NM). The GSCR, appendix 3, defines the overall NM requirements that VoIP systems must meet. The NM requirements for the SUT LAN were satisfied with vendor LoC. The switching system NM requirements in accordance with the GSCR, section 9, are not required for a PBX 1 and were not tested.

(d) Internal Clock. The switching system Internal Clock requirements in accordance with the GSCR, appendix 3, are not required for a PBX 1 and were not tested.

(e) Synchronization. Synchronization is required for overall voice platforms to include VoIP systems. For the SUT solution, synchronization in accordance with the GSCR, section 11, was met. The SUT derived synchronization with line timing mode via traditional T1 Time Division Multiplexing (TDM)-based interfaces. To meet this requirement, a direct T1 interface must be daisy-chained between multiple gateways to allow synchronize timing of all TDM-based gateway interfaces from a single TDM interface.

(f) Latency. The GSCR, appendix 3, section A3.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress of the trunk. The SUT average 5-minute latency over 545 calls with SRTP turned off was measured to be 53.79 ms without ever exceeding 60ms. However, with SRTP turned on the latency was measured above 60 ms with an average of 62 ms. The operational impact is minor.

(g) Packet Loss. The GSCR, appendix 3, section A3.3.1.3, states packet loss shall not exceed 0.05% averaged over any five-minute period. The SUT packet loss was measured at 0.003% over a thirty-minute period and never exceeded 0.05%.

(h) Internet Protocol version 6 (IPv6). 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:

1. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).

with IPv4.

2. Maintaining interoperability in heterogeneous environments and

3. Commitment to upgrade as the IPv6 standard evolves.

4. Availability of contractor/vendor IPv6 technical support.

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 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.

b. System Interoperability Results. The SUT including VoIP is certified for joint use in the DSN as a PBX 1 and PBX 2 in accordance with the requirements set forth in the GSCR. A desktop review determined CCM Version 4.2(3) SR1 with IOS Software Release 12.4(9) T1 to be functionally identical to the SUT. Therefore, this certification letter covers both the CCM Version 4.2(3) and CCM Version 4.2(3) SR1 with IOS Software Release 12.4(9) T1. The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The interoperability test summary is shown in table 2-5.

Table 2-5. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it will not be certified. 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 that are attempted over a trunk that is broken or in a remote busy-out condition do not receive an ICA. ⁴ This interface is not authorized or approved for connection to the DSN. There is no overall operational impact because T1 CAS is conditional for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 CAS is conditional for a PBX 1.
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 or in a remote busy-out condition do not receive an ICA. ⁴ The operational impact is minor.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 PRI is conditional for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs a minor configuration change ⁵ and the following minor exceptions: The SUT does not support an MLPP global diversion number. ⁶ The BNEA is not provided. ⁷ 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 conditional 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 conditional 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. ⁶ The BNEA is not provided. ⁷ 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. ^{8, 9, 10, 11, 12, 13} The operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. There is no operational impact because Attendant is conditional 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 a critical requirement for a PBX 1. ¹⁴
Preset Conferencing	No	Not Tested	This feature is not supported. There is no operational impact because Preset Conferencing is conditional 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 conditional for a PBX 1.
PAT	No	Not Tested	This feature is not supported. There is no operational impact because PAT is conditional for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported. There is no operational impact because DSN Hotline Services are conditional for a PBX 1.
Network Management	No	Not Tested	This feature is not supported. There is no operational impact because Network Management is conditional for a PBX 1.
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. There is no operational impact because ISDN Services are conditional for a PBX 1.

Table 2-5. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Synchronization ¹⁵	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 16.	See note 16.	
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASVALAN posted on the JITC TSSI program web page: http://jitc.fhu.disa.mil/tssi See notes 17 and 18.	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	Although the SUT supports T1 CAS, due to critical interoperability discrepancies discovered during testing, it will not be certified. 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 There is no overall operational impact because T1 CAS is conditional for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. There is no operational impact because E1 CAS is conditional for a PBX 1.
	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	This interface is not supported. There is no operational impact because E1 PRI is conditional for a PBX 1.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ¹⁹	Met all CRs and FRs.
LEGEND: ANSI - American National Standards Institute ASVALAN - Assured Services Voice Application Local Area Network BNEA - Busy Not Equipped Announcement 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 EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements ICA - Isolated Code Announcement IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU-T - International Telecommunication Union - Telecommunication Standardization Sector JITC - Joint Interoperability Test Command 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 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PNT - Preempt Notification Tone 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 TDM - Time Division Multiplexing TPC - Twisted Pair Copper TSSI - Telecom Switched Services Interoperability VoIP - Voice over Internet Protocol				

Table 2-5. SUT Interoperability Test Summary (continued)

<p>NOTES:</p> <ol style="list-style-type: none">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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required 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. Although this is a critical requirement for T1 CAS, there is no operational impact because T1 CAS is not required for a PBX 1.4 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 they are treated with a Blocked Precedence Announcement (BPA) and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.5 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.6 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.7 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.8 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: Call-Waiting, Selective Call Rejection, and Denied Originating Service. All of these features are conditional for a PBX 1. There is no operational impact.9 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.10 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.11 A short "ping" ring is not provided when calls are forwarded. There is a minor operational impact.12 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 PNT and the other members remain connected.13 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.14 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 1.15 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways.16 Security is tested by DISA-led Information Assurance test teams and published in a separate report.17 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.18 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 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.19 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.
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12. TEST AND ANALYSIS REPORT. 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 TSSI website at <http://jitc.fhu.disa.mil/tssi>.