



DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND
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IN REPLY REFER TO: Networks and Transport Division (JTE)

22 May 2006

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.1(2) Service Release 1 with Internetworking Operating System (IOS) Software Release 12.3(8) YC2 (Includes Voice over Internet Protocol)

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01C, "Interoperability and Supportability of Information Technology and National Security Systems," 20 November 2003

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 Cisco CallManager Version 4.1(2) Service Release 1 with IOS Software Release 12.3(8) YC2, hereinafter referred to as the system under test (SUT), meets all of its critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the Voice over Internet Protocol (VoIP) critical interoperability requirements with 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>. **While all of the common call features are conditional for PBX 1 certification, the SUT supports only three common features: Call Forwarding (with analog telephones only), Call Waiting, and Call Hold (with VoIP telephones with two line appearances). The following common call features are disabled when Multi-Level Precedence and Preemption is provisioned and therefore are not covered in this certification: Call Pick-up, Call Transfer, Conference Calling, Three-way Call, and Multi-line Hunt Service.** This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.
3. This finding is based on interoperability testing conducted by JITC and review of vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 11 July through 5 August 2005. Review of vendor's LoC was completed on 23 August 2005. 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.

JITC Memo, JTE, Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.1(2) Service Release 1 with Internetworking Operating System (IOS) Software Release 12.3(8) YC2 (Includes Voice over Internet Protocol)

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 required and conditional 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).

Table 1. SUT Hardware and Software Components

Cisco CallManager (CCM) Version 4.1(2) SR1 with IOS Software Release 12.3(8) YC2 (includes Voice over Internet Protocol)			
Component (see note)	Release	Sub-component	Function
CallManagers <u>MCS7835H, MCS7835L, MCS7825</u> MCS7845H, MCS7845I	4.1(2)SR1		Processing/Signaling
Cisco 3745 Multiservice Access Routers (Gateway)	IOS 12.3(8) YC2	<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>VWIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		<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
Cisco 2691 Multiservice Access Router (Gateway)	IOS 12.3(8) YC2	<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, DID
		<u>VWIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
<u>CP-7940 and CP-7960</u>	7.2(2) App P00307020200 Boot PC0303010001		IP Phone
<u>CP-7970</u>	6.0(2) SR1 App Jar-70.62-1-0-2.sbn Boot 7970_64054100.bin		IP Phone

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Table 1. SUT Hardware and Software Components (continued)

LEGEND:	
App	- application
CP	- Cisco Phone
DID	- Direct Inward Dialing
Fax	- facsimile
FXS	- Foreign Exchange Station
HD	- High Density
IOS	- Internetworking Operating System
IP	- Internet Protocol
JITC	- Joint Interoperability Test Command
Mbps	- Megabits per second
MCS	- Media Call Service
MFT	- Multiflex Trunk
NM	- Network Module
RJ	- Registered Jack
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
VIC	- Voice Interface Card
VWIC	- Voice WAN Interface Card
WAN	- Wide Area Network

NOTE: 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.

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	Although the SUT supports T1 CAS, due to critical interoperability anomalies discovered during previous testing, it was not tested this time. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
VoIP	No	Certified	Met all CRs and FRs with certified DSN Voice Application Local Area Network.
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 were not met. ¹ Operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Public Safety	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Preset Conferencing	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
PAT	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Network Management	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Synchronization ²	Yes	Certified	Met all CRs and FRs.

Table 2. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 3.	See note 3.	
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6 capability. ⁴	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Tested	Although the SUT supports T1 CAS, due to critical interoperability anomalies discovered during previous testing, it was not tested this time. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ⁵	Met all critical CRs and FRs.
LEGEND: ANSI - American National Standards Institute BRI - Basic Rate Interface CAS - Channel Associated Signaling CRs - Capability Requirements DISA - Defense Information Systems Agency DoD - Department of Defense 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) EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network JITC - Joint Interoperability Test Command Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption NI 1/2 - National ISDN Standard 1 or 2 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network 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 VoIP - Voice over Internet Protocol				
NOTES: 1 Meets Call-Forwarding busy and no-answer call feature requirements on analog phones. Call Hold and Call Waiting Features were met on VoIP instruments with two line appearances. The following common call features are disabled when MLPP is provisioned: Call Pick-up, Call Transfer, Conference Calling, Precedence Call Waiting, Multi-line Hunt Service, or Three-way Calling. 2 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways. 3 Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 4 Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades. 5 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 Line Interfaces				
Interface	Critical	Requirements Required (R) or Conditional (C)		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> • DN Identification (R) • Line signaling (R) • 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 Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		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 Sect. 3.1.3 • GSCR Sect. 3.4.3/3.9 • CJCSI 6215.01B
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
		Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56 kbps switched data (R: 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 Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 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 Trunk Interfaces				
Interface	Critical	Requirements Required (R) or Conditional (C)		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> • Framing (R) • Line Code (R) • Signaling (R) • Alarms (R) • WWNDP (R) • Outpulsing digit formats (C: CAS only) • Routing (C) 	<ul style="list-style-type: none"> • GSCR Sect. 7 • GSCR Sect. 7 • GSCR Sect. 5 • GSCR Sect. 2.5.7, 7.1.4 & 7.2.2 • GSCR Sect. 4.5.1 • GSCR Sect. 4.5.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> • Trunk Groups (C) • Call Processing (C) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (C) • Direct Inward Dialing (C) 	<ul style="list-style-type: none"> • GSCR Sect. 4.2 • GSCR Sect. 2.5.5 & 2.5.6 • GSCR Sect. 4 • GSCR Sect. 3.10 • GSCR Sect. 7.3 • GSCR Sect.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 Sect. 3 • CJCSI 6215.01B
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
		Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 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 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	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 and conference calling and call hold (C) • Call forwarding (C) • Call pick-up (C) 	<ul style="list-style-type: none"> • GSCR Sect. 2.1.2 • GSCR Sect. 2.1.3 • GSCR Sect. 2.1.4 • GSCR Sect. 2.1.5 • GSCR Sect. 2.1.6 • GSCR Sect. 2.1.7 • GSCR Sect. 2.1.8 • GSCR Sect. 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 Sect. 2.2.1 • GSCR Sect. 2.2.2 • GSCR Sect. 2.2.3 • GSCR Sect. 2.2.4 • GSCR Sect. 2.2.5 • GSCR Sect. 2.2.6 • GSCR Sect. 2.2.7
Public Safety	No	<ul style="list-style-type: none"> • E911 (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 Sect. 2.4.1 • GSCR Sect. 2.4.2 • GSCR Sect. 2.4.3 • GSCR Sect. 2.4.4 • GSCR Sect. 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 retrial and alternate address (C) • Bridge release (C) • Lost connection (C) • Secondary conferencing (C) • Address translation (C) 	<ul style="list-style-type: none"> • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6.1 • GSCR Sect. 2.6.2 • GSCR Sect. 2.6.3 • GSCR Sect. 2.6.4 • GSCR Sect. 2.6.5 • GSCR Sect. 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 Sect. 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 Sect. 2.11.1 • GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1.1 • GSCR Sect. 2.11.1.2 • GSCR Sect. 2.11.1.3 • GSCR Sect. 2.11.1.4 • GSCR Sect. 2.11.1.5 • GSCR Sect. 2.11.1.6 • GSCR Sect. 2.11.1.7 • GSCR Sect. 2.11.1.8 • GSCR Sect. 2.11.1.9 • GSCR Sect. 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 Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12.1-4 • GSCR Sect. 2.12.5

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	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) • NM controls (C) • Remote access (C) 	<ul style="list-style-type: none"> • GSCR Sect. 9.1 • GSCR Sect. 9.2 • GSCR Sect. 9.3 • GSCR Sect. 9.4 • GSCR Sect. 9.5 • GSCR Sect. 9.6 • GSCR Sect. 9.7 • GSCR Sect. 9.8
ISDN Services	No	<ul style="list-style-type: none"> • EKTS (C) 	<ul style="list-style-type: none"> • GSCR Sect. 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 Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	<ul style="list-style-type: none"> • GR-512-CORE (R) 	<ul style="list-style-type: none"> • GSCR Sect. 12
Security ¹	Yes	<ul style="list-style-type: none"> • DITSCAP (R) 	<ul style="list-style-type: none"> • GSCR Sect. 13
VoIP			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	References
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"> • MOS 4.0 or better • ITU-T G.711 PCM Codec • Security in accordance with DITSCAP • NM • Line timing • Internal Clock • Latency ≤ 60 ms • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3
LANs	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"> • LAN parameters • CoS/QoS • VLANs • IEEE Standards Conformance • .99999 availability • Modular devices • 2 second link restoral • LAN NM • Traffic Engineering 	<ul style="list-style-type: none"> • GSCR App. 3

Table 3. PBX 1 Requirements (continued)

Network Gateways				
Gateway	Critical	Requirements Required (R) or Conditional (C)		References
PSTN ²	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 ³	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 Sect. 5.5 GSCR Sect. 4.4 GSCR Sect. 4.1 GSCR Sect. 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 Sect. 3 CJCSI 6215.01B
EMSS	No	CJCS approved requirements not defined.		
NGCS	No	CJCS approved requirements not defined.		

LEGEND:					
2W	- 2-Wire	EKTS	- Electronic Key Telephone System	NX64	- Data format restricted to multiples of 64 kbps
A/D	- Analog to Digital Conversion	EMSS	- Enhanced Mobile Satellite System	PAT	- Precedence Access Threshold
ANSI	- American National Standards Institute	G.711	- PCM of Voice Frequencies	PBX 1	- Private Branch Exchange 1
App.	- Appendix	GR	- Generic Requirement	PCM	- Pulse Code Modulation
BER	- Bit Error Ratio	GSCR	- Generic Switching Center Requirements	PCM-24	- Pulse Code Modulation - 24 Channels
BRI	- Basic Rate Interface	H.320	- Standard for Narrowband VTC	PCM-30	- Pulse Code Modulation - 30 Channels
C	- Conditional	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PRI	- Primary Rate Interface
CAS	- Channel Associated Signaling	IPv6	- Internet Protocol version 6	PSTN	- Public Switched Telephone Network
CCS	- Common Channel Signaling	ISDN	- Integrated Services Digital Network	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCS	- Chairman of the Joint Chiefs of Staff	IT	- Information Technology	QoS	- Quality of Service
CJCSI	- CJCS Instruction	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	R	- Required
Codec	- coder/decoder	kbps	- kilobits per second	Sect.	- Section
CoS	- Class of Service	KXX	- K= any number 2-8; X= any number 1-9	SS7	- Signaling System 7
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	STE	- Secure Terminal Equipment
DISR	- DoD IT Standards Registry	Mbps	- Megabits per second	STU-III	- Secure Telephone Unit-3 rd generation
DITSCAP	- DoD IT Security and Accreditation Process	MLR1	- Multi-Frequency Recommendation 1	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DN	- Directory Number	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN Signaling Standard for T1
DoD	- Department of Defense	MOS	- Mean Opinion Score	TIA	- Telecommunications Industry Association
DP	- Dial Pulse	ms	- Milliseconds	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DSN	- Defense Switched Network	NATO	- North Atlantic Treaty Organization	VBD	- Variable bit data
DRSN	- Defense Red Switch Network	NGCS	- NATO Gateway Communication Switch	VLAN	- Virtual LAN
DTMF	- Dual Tone Multi-Frequency	NI 1/2	- National ISDN 1 or 2	VoIP	- Voice over Internet Protocol
E1	- European Basic Multiplex Rate (2.048 Mbps)	NM	- Network Management	VTC	- Video Teleconferencing
E911	- Enhanced 911	NX56	- Data format restricted to multiples of 56 kbps	WWNDP	- Worldwide Numbering and Dialing Plan
EIA	- Electronic Industries Alliance				

NOTES:	
1	Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.
2	Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
3	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://jitic.fhu.disa.mil/tssi>.

JITC Memo, JTE, Special Interoperability Test Certification of Cisco CallManager (CCM) Version 4.1(2) Service Release 1 with Internetworking Operating System (IOS) Software Release 12.3(8) YC2 (Includes Voice over Internet Protocol)

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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)," 23 April 2004

CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** Cisco CallManager (CCM) Version 4.1(2) Service Release 1 with Internetworking Operating System (IOS) Software Release 12.3(8) YC2 (Includes Voice over Internet Protocol (VoIP)), hereinafter referred to as the system under test (SUT).
- 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 VoIP configuration consists of the MCS7845 CallManagers, which are an integral part of a complete, scalable architecture, which runs the CCM software. 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 runs Windows 98, ME, NT 4.0 (Service Pack 4 or greater), 2000, or XP. The SUT was tested with Windows 2000 Server. The CCM software provides telephony features and capabilities to packet telephony network devices such as IP phones. While all the common call features are conditional for PBX 1 certification, the SUT supports only three common features: Call Forwarding (analog telephones only), Call Waiting, and Call Hold (VoIP telephones with two line appearances). The following common call features are disabled when Multi-Level Precedence and Preemption (MLPP) is provisioned and therefore are not covered in this certification: Call Pick-up, Call Transfer, Conference Calling, Three-way Call, and Multi-line Hunt Service.

The SUT is designed to be installed along with the VoIP components listed in table 2-1. The 3745 and 2691 gateway routers are included in this tested architecture.

The SUT gateway routers are scalable. The 2691 has one network module slot, and the 3745 has four network module slots. These network module slots can be populated with either a Network Module High Density two-slot voice (NM-HD-2V), or a NM-HD-2V extended (NM-HD-2VE) module. The NM-HD-2V supports up to eight analog lines, and does not support voice wide area network T1 cards. The NM-HD-2VE supports both voice interface analog and wide area network T1 cards as shown in table 2-1.

Table 2-1. SUT Hardware and Software Components

Cisco CallManager (CCM) Version 4.1(2) SR1 with IOS Software Release 12.3(8) YC2 (includes Voice over Internet Protocol)			
Component (see note)	Release	Sub-component	Function
CallManagers <u>MCS7835H, MCS7835I, MCS7825</u> MCS7845H, MCS7845I	4.1(2)SR1		Processing/Signaling
Cisco 3745 Multiservice Access Routers (Gateway)	IOS 12.3(8) YC2	<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>VVIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		<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
Cisco 2691 Multiservice Access Router (Gateway)	IOS 12.3(8) YC2	<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, DID
		<u>VVIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
<u>CP-7940 and CP-7960</u>	7.2(2) App P00307020200 Boot PC0303010001		IP Phone
<u>CP-7970</u>	6.0(2) SR1 App Jar-70.62-1-0-2.sbn Boot 7970_64054100.bin		IP Phone
LEGEND: App - application CP - Cisco Phone DID - Direct Inward Dialing Fax - facsimile FXS - Foreign Exchange Station HD - High Density IOS - Internetworking Operating System IP - Internet Protocol JITC - Joint Interoperability Test Command Mbps - Megabits per second MCS - Media Call Service MFT - Multiflex Trunk NM - Network Module RJ - Registered Jack 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 VIC - Voice Interface Card VVIC - Voice WAN Interface Card WAN - Wide Area Network			
NOTE: 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.			

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.

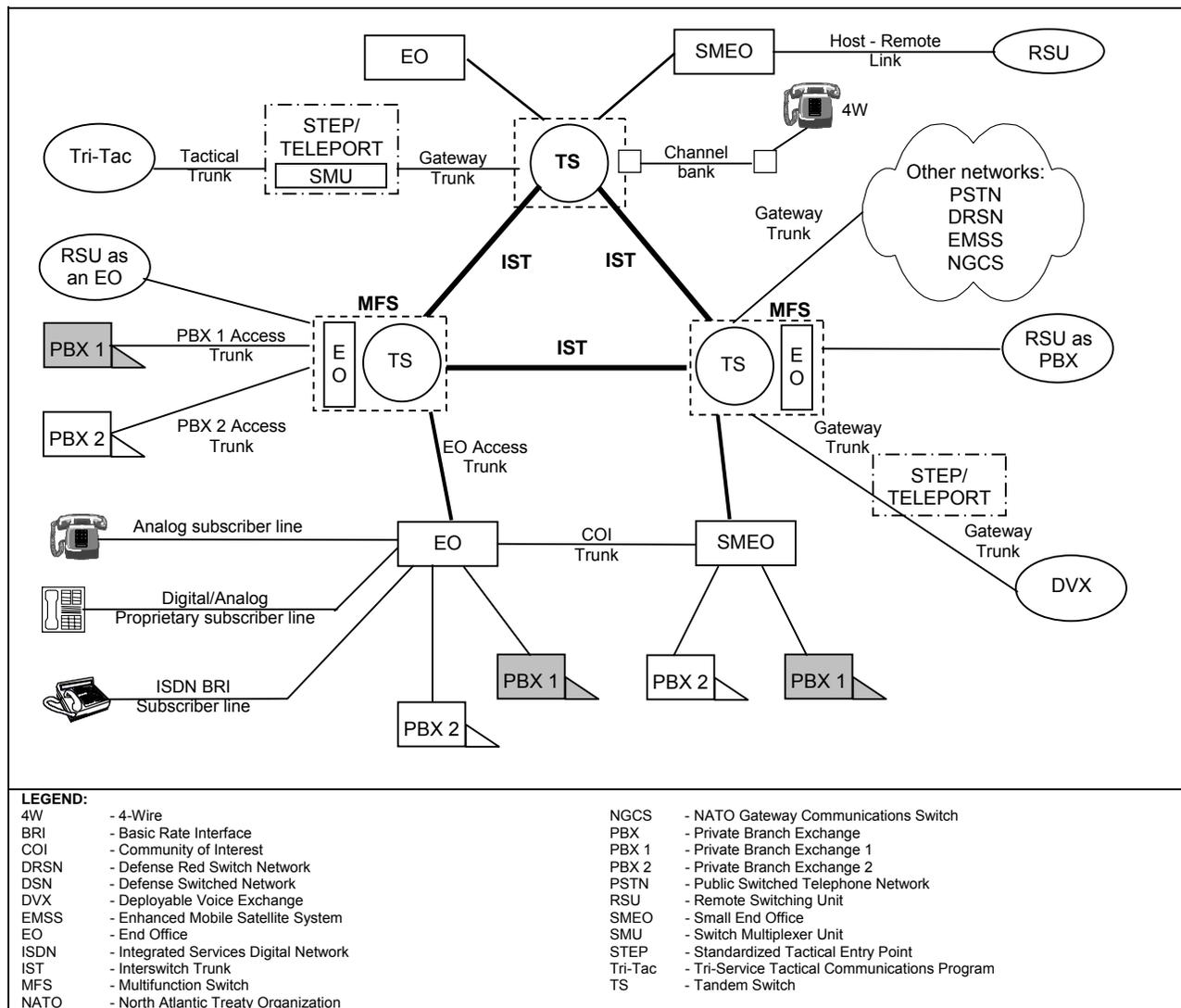


Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in table 2-2. 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 Letter(s) 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 2-2. PBX 1 Requirements

DSN Line Interfaces				
Interface	Critical	Requirements Required (R) or Conditional (C)		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> • DN Identification (R) • Line signaling (R) • 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 Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		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 Sect. 3.1.3 • GSCR Sect. 3.4.3/3.9 • CJCSI 6215.01B
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
		Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56 kbps switched data (R: 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 Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 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 Trunk Interfaces				
Interface	Critical	Requirements Required (R) or Conditional (C)		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> • Framing (R) • Line Code (R) • Signaling (R) • Alarms (R) • WWNDP (R) • Outpulsing digit formats (C: CAS only) • Routing (C) • Trunk Groups (C) • Call Processing (C) • CAS to CCS trunk interworking (C) • PCM-24/PCM-30 Interoperation (C) • Direct Inward Dialing (C) 	<ul style="list-style-type: none"> • GSCR Sect. 7 • GSCR Sect. 7 • GSCR Sect. 5 • GSCR Sect. 2.5.7, 7.1.4 & 7.2.2 • GSCR Sect. 4.5.1 • GSCR Sect. 4.5.2 • GSCR Sect. 4.2 • GSCR Sect. 2.5.5 & 2.5.6 • GSCR Sect. 4 • GSCR Sect. 3.10 • GSCR Sect. 7.3 • GSCR Sect.2.3.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		Voice	<ul style="list-style-type: none"> • MOS (R) • MLPP (R) • Secure calls (R)
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	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 Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 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 2-2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	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 and conference calling and call hold (C) • Call forwarding (C) • Call pick-up (C) 	<ul style="list-style-type: none"> • GSCR Sect. 2.1.2 • GSCR Sect. 2.1.3 • GSCR Sect. 2.1.4 • GSCR Sect. 2.1.5 • GSCR Sect. 2.1.6 • GSCR Sect. 2.1.7 • GSCR Sect. 2.1.8 • GSCR Sect. 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 Sect. 2.2.1 • GSCR Sect. 2.2.2 • GSCR Sect. 2.2.3 • GSCR Sect. 2.2.4 • GSCR Sect. 2.2.5 • GSCR Sect. 2.2.6 • GSCR Sect. 2.2.7
Public Safety	No	<ul style="list-style-type: none"> • E911 (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 Sect. 2.4.1 • GSCR Sect. 2.4.2 • GSCR Sect. 2.4.3 • GSCR Sect. 2.4.4 • GSCR Sect. 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 Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6 • GSCR Sect. 2.6.1 • GSCR Sect. 2.6.2 • GSCR Sect. 2.6.3 • GSCR Sect. 2.6.4 • GSCR Sect. 2.6.5 • GSCR Sect. 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 Sect. 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 Sect. 2.11.1 • GSCR Sect. 2.11.1 • GSCR Sect. 2.11.1.1 • GSCR Sect. 2.11.1.2 • GSCR Sect. 2.11.1.3 • GSCR Sect. 2.11.1.4 • GSCR Sect. 2.11.1.5 • GSCR Sect. 2.11.1.6 • GSCR Sect. 2.11.1.7 • GSCR Sect. 2.11.1.8 • GSCR Sect. 2.11.1.9 • GSCR Sect. 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 Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12 • GSCR Sect. 2.12.1-4 • GSCR Sect. 2.12.5

Table 2-2. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	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) • NM controls (C) • Remote access (C) 	<ul style="list-style-type: none"> • GSCR Sect. 9.1 • GSCR Sect. 9.2 • GSCR Sect. 9.3 • GSCR Sect. 9.4 • GSCR Sect. 9.5 • GSCR Sect. 9.6 • GSCR Sect. 9.7 • GSCR Sect. 9.8
ISDN Services	No	<ul style="list-style-type: none"> • EKTS (C) 	<ul style="list-style-type: none"> • GSCR Sect. 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 Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	<ul style="list-style-type: none"> • GR-512-CORE (R) 	<ul style="list-style-type: none"> • GSCR Sect. 12
Security ¹	Yes	<ul style="list-style-type: none"> • DITSCAP (R) 	<ul style="list-style-type: none"> • GSCR Sect. 13
VoIP			
Feature/ Capability	Critical	Requirements Required (R) or Conditional (C)	References
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"> • MOS 4.0 or better • ITU-T G.711 PCM Codec • Security in accordance with DITSCAP • NM • Line timing • Internal Clock • Latency ≤ 60 ms • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3
LANs	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"> • LAN parameters • CoS/QoS • VLANs • IEEE Standards Conformance • .99999 availability • Modular devices • 2 second link restoral • LAN NM • Traffic Engineering 	<ul style="list-style-type: none"> • GSCR App. 3

Table 2-2. PBX 1 Requirements (continued)

Network Gateways				
Gateway	Critical	Requirements Required (R) or Conditional (C)		References
PSTN ²	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 ³	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 Sect. 5.5 GSCR Sect. 4.4 GSCR Sect. 4.1 GSCR Sect. 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 Sect. 3 CJCSI 6215.01B
EMSS	No	CJCS approved requirements not defined.		
NGCS	No	CJCS approved requirements not defined.		

LEGEND:					
2W	- 2-Wire	EKTS	- Electronic Key Telephone System	NX64	- Data format restricted to multiples of 64
A/D	- Analog to Digital Conversion	EMSS	- Enhanced Mobile Satellite System	kbps	
ANSI	- American National Standards Institute	G.711	- PCM of Voice Frequencies	PAT	- Precedence Access Threshold
App.	- Appendix	GR	- Generic Requirement	PBX 1	- Private Branch Exchange 1
BER	- Bit Error Ratio	GSCR	- Generic Switching Center Requirements	PCM	- Pulse Code Modulation
BRI	- Basic Rate Interface	H.320	- Standard for Narrowband VTC	PCM-24	- Pulse Code Modulation - 24 Channels
C	- Conditional	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PCM-30	- Pulse Code Modulation - 30 Channels
CAS	- Channel Associated Signaling	IPv6	- Internet Protocol version 6	PRI	- Primary Rate Interface
CCS	- Common Channel Signaling	ISDN	- Integrated Services Digital Network	PSTN	- Public Switched Telephone Network
CJCS	- Chairman of the Joint Chiefs of Staff	IT	- Information Technology	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCSI	- CJCS Instruction	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	QoS	- Quality of Service
Codec	- coder/decoder	kbps	- kilobits per second	R	- Required
CoS	- Class of Service	KXX	- K= any number 2-8; X= any number 1-9	Sect.	- Section
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	SS7	- Signaling System 7
DISR	- DoD IT Standards Registry	Mbps	- Megabits per second	STE	- Secure Terminal Equipment
DITSCAP-Process	- DoD IT Security and Accreditation Process	MFR1	- Multi-Frequency Recommendation 1	STU-III	- Secure Telephone Unit-3 rd generation
DN	- Directory Number	MLPP	- Multi-Level Precedence and Preemption	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DoD	- Department of Defense	MOS	- Mean Opinion Score	T1.619a	- SS7 and ISDN Signaling Standard for T1
DP	- Dial Pulse	ms	- Milliseconds	TIA	- Telecommunications Industry Association
DSN	- Defense Switched Network	NATO	- North Atlantic Treaty Organization	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DRSN	- Defense Red Switch Network	NGCS	- NATO Gateway Communication Switch	VBD	- Variable bit data
DTMF	- Dual Tone Multi-Frequency	NI 1/2	- National ISDN 1 or 2	VLAN	- Virtual LAN
E1	- European Basic Multiplex Rate (2.048 Mbps)	NM	- Network Management	VoIP	- Voice over Internet Protocol
E911	- Enhanced 911	NX56	- Data format restricted to multiples of 56 kbps	VTC	- Video Teleconferencing
EIA	- Electronic Industries Alliance			WWNDP	- Worldwide Numbering and Dialing Plan

NOTES:

- Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.
- Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
- 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.

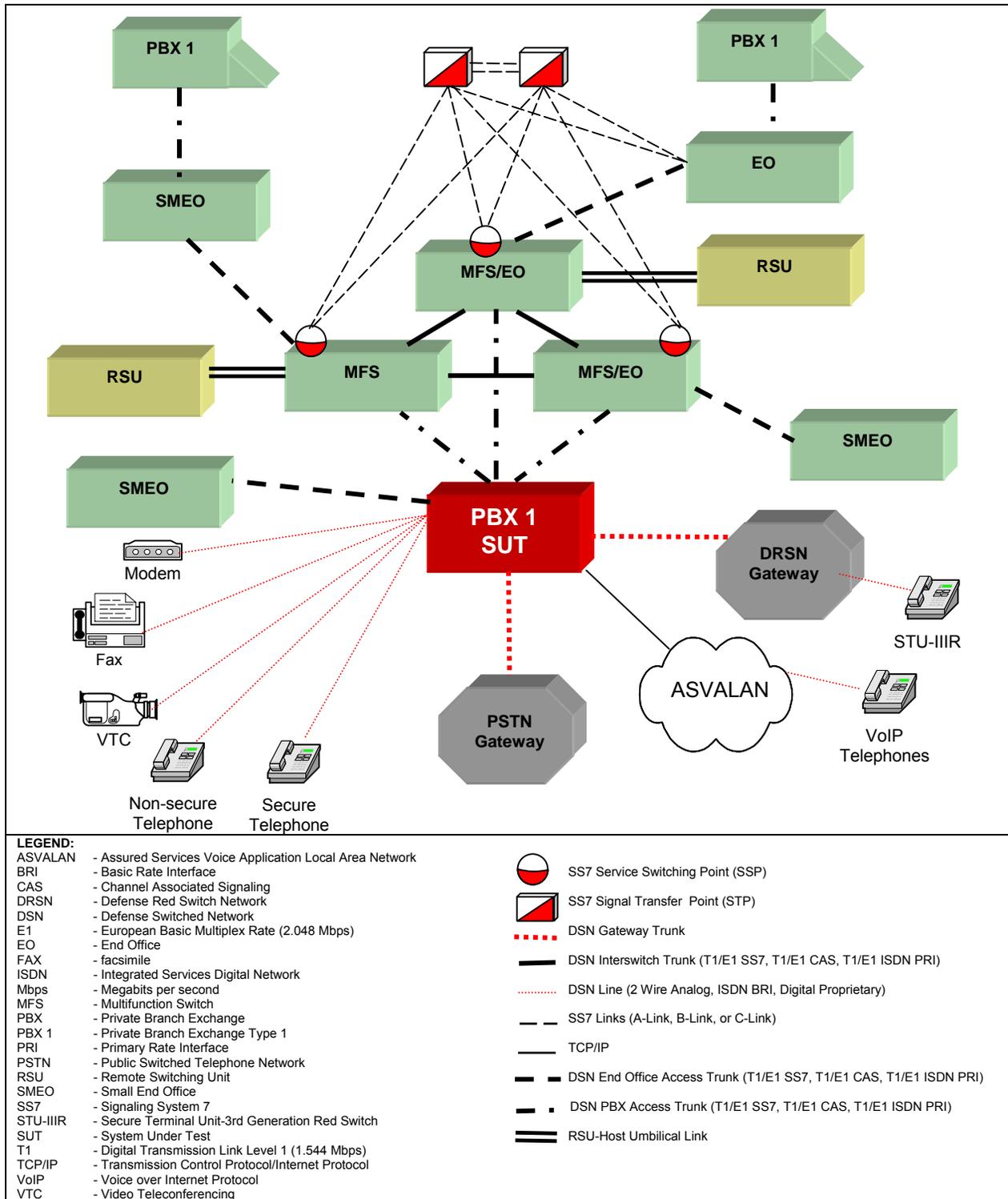
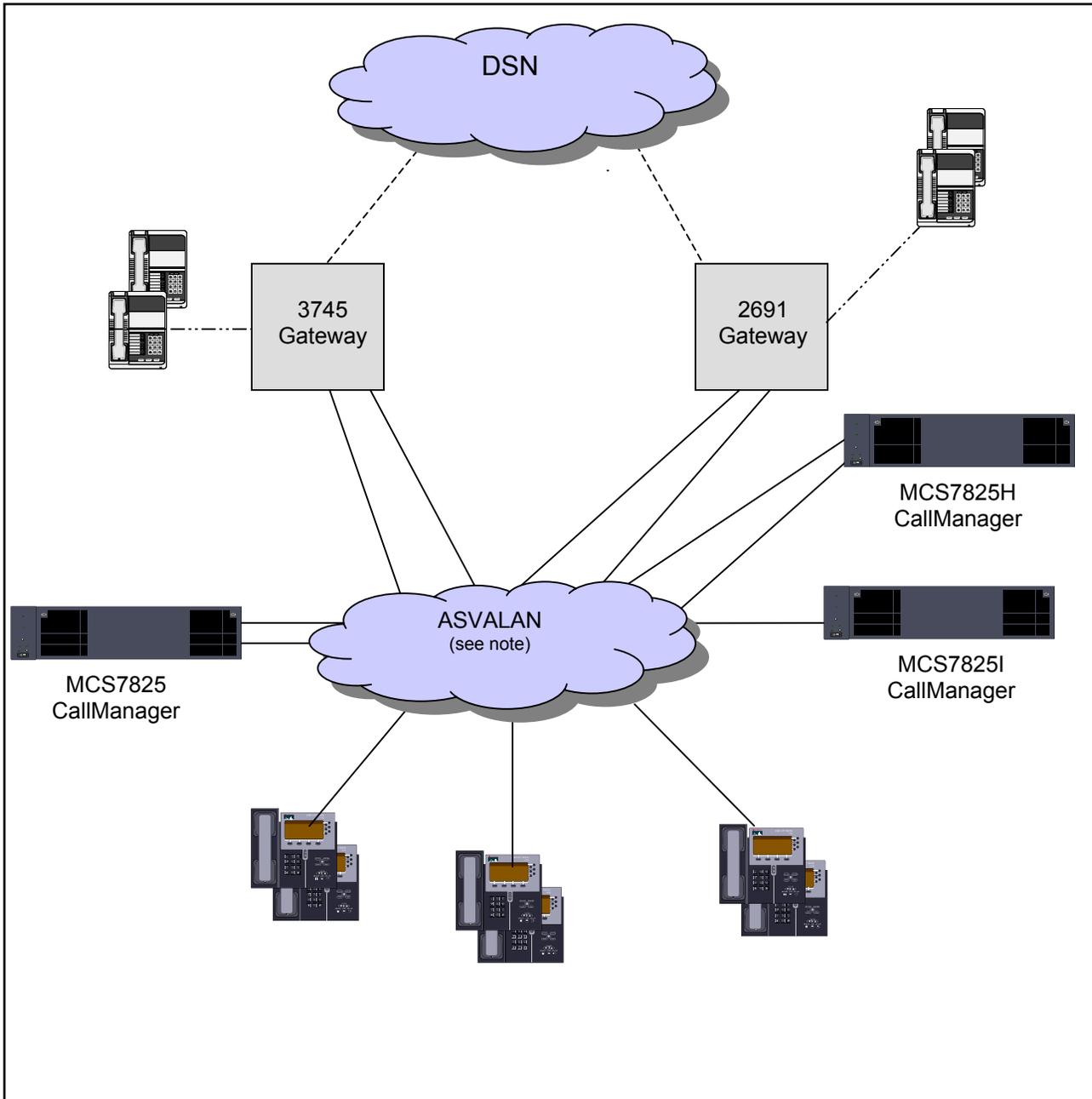


Figure 2-2. Test Configuration



- LEGEND:**
- ASVALAN - Assured Services Voice Application Local Area Network
 - CP - Cisco Phone
 - DSN - Defense Switched Network
 - Mbps - Megabits per second
 - MCS - Media Call Service
 - SUT - System Under Test
 - TDM - Time Division Multiplexing
 - VoIP - Voice over Internet Protocol
- 4-Wire Digital TDM Interfaces
 - 100 Mbps Ethernet
 - 2-Wire Analog
-  CP79xx VoIP Telephones
 -  Analog Telephones

NOTE: The SUT is certified with any DSN VA LAN listed on the Approved Product list located on the Telecom Switched Services Interoperability website: <http://jitc.fhu.disa.mil/tssi>.

Figure 2-3. ASVALAN

9. SYSTEM CONFIGURATIONS. Table 2-3 provides the system configurations, hardware and software used in the test.

Table 2-3. Tested System Configurations

System Name		Software Release	
Nortel Networks MSL-100 (MFS, EO)		SE06	
Siemens EWSD (MFS, EO)		19d with Patch Set 44	
Lucent 5ESS (MFS, EO, SMEO, PBX 1, PBX 2)		5E16.2 SU 05-0005	
Avaya S8700 (SMEO, PBX 1, PBX 2)		Communication Manager (CM) 2.0.1 (R012x.00.1.221.1)	
Cisco CallManager (CCM) Version 4.1(2) SR1 with IOS Software Release 12.3(8) YC2 (includes Voice over Internet Protocol)			
Component (see note)	Release	Sub-component	Function
CallManagers <u>MCS7835H, MCS7835I, MCS7825</u> MCS7845H, MCS7845I	4.1(2)SR1		Processing/Signaling
Cisco 3745 Multiservice Access Routers (Gateway)	IOS 12.3(8) YC2	<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>VVIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		<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
Cisco 2691 Multiservice Access Router (Gateway)	IOS 12.3(8) YC2	<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, DID
		<u>VVIC 2MFT T1</u>	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
CP-7940 and CP-7960	7.2(2) App P00307020200 Boot PC0303010001		IP Phone
CP-7970	6.0(2) SR1 App Jar-70.62-1-0-2.sbn Boot 7970_64054100.bin		IP Phone
LEGEND:			
5ESS - Class 5 Electronic Switching System		NM - Network Module	
App - application		PBX 1 - Private Branch Exchange 1	
CP - Cisco Phone		PBX 2 - Private Branch Exchange 2	
DID - Direct Inward Dialing		RJ - Registered Jack	
EO - End Office		SE - Succession Enterprise	
EWSD - Elektronisches Wählsystem Digital		SMEO - Small End Office	
Fax - facsimile		SR - Service Release	
FXS - Foreign Exchange Station		SU - Software Update	
HD - High Density		T1 - Digital Transmission Link Level 1 (1.544 Mbps)	
IOS - Internetworking Operating System		TDM - Time Division Multiplexing	
IP - Internet Protocol		V - Voice	
JITC - Joint Interoperability Test Command		VE - Voice/Fax Enhanced	
Mbps - Megabits per second		VIC - Voice Interface Card	
MCS - Media Call Service		VoIP - Voice over IP	
MFS - Multifunction Switch		VVIC - Voice WAN Interface Card	
MFT - Multiflex Trunk		WAN - Wide Area Network	
MSL - Meridian Switching Load			
NOTE: 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 as described above.			

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs for T1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2) (American National Standards Institute (ANSI) T1.619a). This was the only trunk interface certified. The SUT supports T1 Channel Associated Signaling (CAS), however, due to a previously noted critical interoperability anomaly (interface does not correctly acknowledge a wink start signal) it was not tested. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1. Detailed trunk configurations and associated lessons learned can be found on the Telecom Switched Services Interoperability (TSSI) web page: <http://jitc.fhu.disa.mil/tssi>.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Analog (GR-506-CORE) and VoIP DSN Line Interfaces.

(3) Features and Capabilities. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions. **While all the common call features are conditional for PBX 1 certification, the SUT supports only three common features: Call Forwarding (with analog telephones only), Call Waiting, and Call Hold (with VoIP telephones with two line appearances).** The following common call features are disabled when MLPP is provisioned and therefore are not covered in this certification: **Call Pick-up, Call Transfer, Conference Calling, Three-way Call, and Multi-line Hunt Service.** The operational impact is minor. Refer to table 2-3 for specific instrument models covered under this certification. The following features and capabilities are covered under this certification letter: synchronization, reliability, security, and VoIP system. The following features and capabilities were not tested and therefore not covered in this certification letter: attendant console, public safety, preset conference, nailed up connections, precedence access threshold, DSN hotline, network management, and ISDN services (electronic key telephone systems).

(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 supports T1 CAS, however, due to previously noted critical interoperability anomalies it was not tested. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1.

(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 at least 4.0 as measured over a 5-minute period. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.34 with all calls having an 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 50 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 802.1p/Q 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 802.1p/Q tag. 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 millisecond (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 7970, 7960, and 7940 phones. Although the phones are capable of shared access (i.e., same switch port is shared by PC and IP phone), dedicated access was tested (separate ports for phones and PCs). The IP phones' Ethernet ports were not tested and are not covered by this certification. During CCM failure testing, all completed calls remained active and were never lost. The 7970 phones are capable of web browsing; this feature was not tested and is not covered by this certification.

b. Scalability. The MCS7835H and MCS7835I can support a maximum of 5,000 IP subscribers and the MCS7825 can support 2,500 IP subscribers. However, with these call managers the manufacturer recommendation is not to exceed 20,000 users in a cluster of eight servers. 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 comprised 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. When MLPP is applied on the CCM, all call features listed in table 2-4 will automatically become disabled because they do not interact with the MLPP functionality. Table 2-4 shows which call features are affected when MLPP is activated on the CCM.

Table 2-4. MLPP and Call Feature Interaction

Call Feature (see note)	Phone Type	
	Analog	IP
Precedence Call Waiting	Disabled	OK (with two line appearances)
Call Hold	Disabled	OK (with two line appearances)
Call Forwarding No Answer	OK	Disabled
Call Forwarding Busy	OK	Disabled
Three-Way Calling	Disabled	Disabled
Call Transfer	Disabled	Disabled
Conference Calling	Disabled	Disabled
Multi-line Hunt Service	Disabled	Disabled
Call Pickup	Disabled	Disabled
LEGEND: IP - Internet Protocol MLPP - Multi-Level Precedence and Preemption VoIP - Voice over Internet Protocol NOTE: Meets Call-Forwarding busy and no-answer call feature requirements on analog phones. Call Hold and Call Waiting Features were met on VoIP instruments with two line appearances. The following common call features are disabled when Multi-Level Precedence and Preemption is provisioned: Call Pickup, Call Transfer, Conference Calling, Precedence Call Waiting, Multiline Hunt Service, or Three-way Calling.		

(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 trunk. The SUT average latency over 80 calls was measured to be 53.79 ms for 50 five-minute calls placed.

(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% for 50 five-minute calls placed.

(h) Internet Protocol version 6 (IPv6). Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades.

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. 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 and the detailed interoperability test status is shown table 2-6.

Table 2-5. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	Although the SUT supports T1 CAS, due to critical interoperability anomalies discovered during previous testing, it was not tested this time. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
VoIP	No	Certified	Met all CRs and FRs with certified DSN Voice Application Local Area Network.
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 were not met. ¹ Operational impact is minor.
Attendant	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Public Safety	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Preset Conferencing	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
PAT	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
DSN Hotline Services	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Network Management	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.
Synchronization ²	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 3.	See note 3.
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6 capability. ⁴

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

Network Gateways																																																																												
	Interface & Signaling	Critical	Status	Remarks																																																																								
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Tested	Although the SUT supports T1 CAS, due to critical interoperability anomalies discovered during previous testing, it was not tested this time. The risk of not testing is minor because T1 CAS is not a critical interface for a PBX 1.																																																																								
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported. The operational impact is minor because it is not a critical requirement for a PBX 1.																																																																								
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.																																																																								
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ⁵	Met all critical CRs and FRs.																																																																								
<p>LEGEND:</p> <table border="0"> <tr> <td>ANSI</td> <td>- American National Standards Institute</td> <td>Mbps</td> <td>- Megabits per second</td> </tr> <tr> <td>BRI</td> <td>- Basic Rate Interface</td> <td>MFR1</td> <td>- Multifrequency Recommendation 1</td> </tr> <tr> <td>CAS</td> <td>- Channel Associated Signaling</td> <td>MLPP</td> <td>- Multi-Level Precedence and Preemption</td> </tr> <tr> <td>CRs</td> <td>- Capability Requirements</td> <td>NI 1/2</td> <td>- National ISDN Standard 1 or 2</td> </tr> <tr> <td>DISA</td> <td>- Defense Information Systems Agency</td> <td>PAT</td> <td>- Precedence Access Threshold</td> </tr> <tr> <td>DoD</td> <td>- Department of Defense</td> <td>PBX 1</td> <td>- Private Branch Exchange 1</td> </tr> <tr> <td>DP</td> <td>- Dial Pulse</td> <td>PM</td> <td>- Program Manager</td> </tr> <tr> <td>DRSN</td> <td>- Defense Red Switch Network</td> <td>PRI</td> <td>- Primary Rate Interface</td> </tr> <tr> <td>DSN</td> <td>- Defense Switched Network</td> <td>PSTN</td> <td>- Public Switched Telephone Network</td> </tr> <tr> <td>DSS1</td> <td>- Digital Subscriber Signaling 1</td> <td>SS7</td> <td>- Signaling System 7</td> </tr> <tr> <td>DTMF</td> <td>- Dual Tone Multi-Frequency</td> <td>SUT</td> <td>- System Under Test</td> </tr> <tr> <td>E1</td> <td>- European Basic Multiplex Rate (2.048 Mbps)</td> <td>T1</td> <td>- Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>EKTS</td> <td>- Electronic Key Telephone System</td> <td>T1.607</td> <td>- ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1</td> </tr> <tr> <td>FRs</td> <td>- Feature Requirements</td> <td>T1.619a</td> <td>- SS7 and ISDN MLPP Signaling Standard for T1</td> </tr> <tr> <td>GR</td> <td>- Generic Requirement</td> <td>TDM</td> <td>- Time Division Multiplexing</td> </tr> <tr> <td>IPv6</td> <td>- Internet Protocol version 6</td> <td>TPC</td> <td>- Twisted Pair Copper</td> </tr> <tr> <td>ISDN</td> <td>- Integrated Services Digital Network</td> <td>VoIP</td> <td>- Voice over Internet Protocol</td> </tr> <tr> <td>JITC</td> <td>- Joint Interoperability Test Command</td> <td></td> <td></td> </tr> </table> <p>NOTES:</p> <ol style="list-style-type: none"> 1 Meets Call-Forwarding busy and no-answer call feature requirements on analog phones. Call Hold and Call Waiting Features were met on VoIP instruments with two line appearances. The following common call features are disabled when MLPP is provisioned: Call Pick-up, Call Transfer, Conference Calling, Precedence Call Waiting, Multi-line Hunt Service, or Three-way Calling. 2 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways. 3 Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 4 Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades. 5 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. 					ANSI	- American National Standards Institute	Mbps	- Megabits per second	BRI	- Basic Rate Interface	MFR1	- Multifrequency Recommendation 1	CAS	- Channel Associated Signaling	MLPP	- Multi-Level Precedence and Preemption	CRs	- Capability Requirements	NI 1/2	- National ISDN Standard 1 or 2	DISA	- Defense Information Systems Agency	PAT	- Precedence Access Threshold	DoD	- Department of Defense	PBX 1	- Private Branch Exchange 1	DP	- Dial Pulse	PM	- Program Manager	DRSN	- Defense Red Switch Network	PRI	- Primary Rate Interface	DSN	- Defense Switched Network	PSTN	- Public Switched Telephone Network	DSS1	- Digital Subscriber Signaling 1	SS7	- Signaling System 7	DTMF	- Dual Tone Multi-Frequency	SUT	- System Under Test	E1	- European Basic Multiplex Rate (2.048 Mbps)	T1	- Digital Transmission Link Level 1 (1.544 Mbps)	EKTS	- Electronic Key Telephone System	T1.607	- ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1	FRs	- Feature Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1	GR	- Generic Requirement	TDM	- Time Division Multiplexing	IPv6	- Internet Protocol version 6	TPC	- Twisted Pair Copper	ISDN	- Integrated Services Digital Network	VoIP	- Voice over Internet Protocol	JITC	- Joint Interoperability Test Command		
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DISA	- Defense Information Systems Agency	PAT	- Precedence Access Threshold																																																																									
DoD	- Department of Defense	PBX 1	- Private Branch Exchange 1																																																																									
DP	- Dial Pulse	PM	- Program Manager																																																																									
DRSN	- Defense Red Switch Network	PRI	- Primary Rate Interface																																																																									
DSN	- Defense Switched Network	PSTN	- Public Switched Telephone Network																																																																									
DSS1	- Digital Subscriber Signaling 1	SS7	- Signaling System 7																																																																									
DTMF	- Dual Tone Multi-Frequency	SUT	- System Under Test																																																																									
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JITC	- Joint Interoperability Test Command																																																																											

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>.

Table 2-6. SUT Interoperability Requirements/Status

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
T1 CAS	No	Not Tested ¹	Trunking	Framing (R)	GSCR Sect. 7	Not Tested	
				Line Code (R)	GSCR Sect. 7	Not Tested	
				Signaling (R)	GSCR Sect. 5	Not Tested	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Not Tested	
				WWNDP (R)	GSCR Sect. 4.5.1	Not Tested	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Not Tested	
				Routing (C)	GSCR Sect. 4.2	Not Tested	
				Trunk Groups (C)	GSCR Sect. 2.5.5 & 2.5.6	Not Tested	
				Call Processing (R)	GSCR Sect. 4	Not Tested	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Not Tested	
				PCM-24/PCM-30 Interoperation (C)	GSCR Sect. 7.3	Not Tested	
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01B	Not Tested	
				MLPP (R)	GSCR Sect. 3	Not Tested	
				Secure calls (R)	CJCSI 6215.01B	Not Tested	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Tested	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested					
Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Not Tested					
VTC	ITU-T H.320 (C: ISDN PRI only)	DISR	Not Tested				

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
E1 CAS	No	Not Tested ²	Trunking	Framing (R)	GSCR Sect. 7	Not Tested	
				Line Code (R)	GSCR Sect. 7	Not Tested	
				Signaling (R)	GSCR Sect. 5	Not Tested	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Not Tested	
				WWNDP (R)	GSCR Sect. 4.5.1	Not Tested	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Not Tested	
				Routing (C)	GSCR Sect. 4.2	Not Tested	
				Trunk Groups (C)	GSCR Sect. 2.5.5 & 2.5.6	Not Tested	
				Call Processing (R)	GSCR Sect. 4	Not Tested	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Not Tested	
				PCM-24/PCM-30 Interoperation (C)	GSCR Sect. 7.3	Not Tested	
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01B	Not Tested	
				MLPP (R)	GSCR Sect. 3	Not Tested	
				Secure calls (R)	CJCSI 6215.01B	Not Tested	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Tested	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
				NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	
Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Not Tested					
VTC	ITU-T H.320 (C: ISDN PRI only)	DISR	Not Tested				

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
T1 ISDN PRI (ANSI T1.619a)	Yes	Certified	Trunking	Framing (R)	GSCR Sect. 7	Met	
				Line Code (R)	GSCR Sect. 7	Met	
				Signaling (R)	GSCR Sect. 5	Met	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Met	
				Timing (R)	GSCR Sect. 11.1.1.2	Met	
				WWNDP (R)	GSCR Sect. 4.5.1	Met	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Met	
				Routing (C)	GSCR Sect. 4.2	Met	
				Trunk Groups (C)	GSCR Sect. 2.5.5 & 2.5.6	Met	
				Call Processing (R)	GSCR Sect. 4	Met	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Not Tested	Minor ²
				PCM-24/PCM-30 Interoperation (C)	GSCR Sect. 7.3	Not Tested	Minor ²
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Met		
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	Minor ³
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	Minor ³
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested	Minor ³
NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Tested		Minor ³			
Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met					
VTC	ITU-T H.320 (R: ISDN PRI only)	DISR	Not Tested				

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Access	DN Identification (R)	GSCR Sect 2.1.1	Met	
				Line signaling (R)	GSCR Sect 5.2	Met	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Met	
				WWNDP (R)	GSCR Sect. 4.5	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
				2W user access (R)	GSCR Sect 4.3.3	Met	
				Analog busy/idle (R)	GSCR Sect 4.3.4.1	Met	
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				Announcements (R)	GSCR Sect. 3.1.3	Not Tested ²	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met	
VTC	ITU-T H.320 (R: ISDN BRI only)	DISR	Not Tested				

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces (continued)							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
ISDN BRI NI 1/2	No	Not Tested ²	Access	DN Identification (R)	GSCR Sect 2.1.1	Not Tested	
				Line signaling (R)	GSCR Sect 5.2	Not Tested	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Not Tested	
				WWNDP (R)	GSCR Sect. 4.5	Not Tested	
				Call Treatments (R)	GSCR Sect. 4.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01B	Not Tested	
				Announcements (R)	GSCR Sect. 3.1.3	Not Tested	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Not Tested	
				Secure calls (R)	CJCSI 6215.01B	Not Tested	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Tested	
				56-kbps switched data (R)	GSCR Sect. 3.10	Not Tested	
				64-kbps switched data (R)	GSCR Sect. 3.10	Not Tested	
				NX56 synchronous BER (R)	GSCR Sect. 3.10	Not Tested	
				NX64 synchronous BER (R)	GSCR Sect. 3.10	Not Tested	
				Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Not Tested	
			VTC	ITU-T H.320 (R: ISDN BRI only)	DISR	Not Tested	

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces (continued)							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
Digital Proprietary	No	Not Tested ²	Access	DN Identification (R)	GSCR Sect 2.1.1	Not Tested	
				Line signaling (R)	GSCR Sect 5.2	Not Tested	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Not Tested	
				WWNDP (R)	GSCR Sect. 4.5	Not Tested	
			Voice	Call Treatments (R)	GSCR Sect. 4.1	Not Tested	
				MOS (R)	CJCSI 6215.01B	Not Tested	
VoIP	No	Certified	Access	DN Identification (R)	GSCR Sect 2.1.1	Met	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Met	
				WWNDP (R)	GSCR Sect. 4.5	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Met	
DSN Features & Capabilities							
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
Common Features	No	Certified	Selective call rejection (C)		GSCR Sect. 2.1.2	Not Tested	Minor ³
			Denied originating service (C)		GSCR Sect. 2.1.3	Not Tested	Minor ³
			Code restriction and diversion (C)		GSCR Sect. 2.1.4	Not Tested	Minor ³
			Three-way calling (C)		GSCR Sect. 2.1.5	Not Tested	Minor ³
			Add-on transfer, conference calling, and call hold (C)		GSCR Sect. 2.1.6	Not Tested	Minor ³
			Call forwarding (C)		GSCR Sect. 2.1.7	Met	See note 3.
			Call pick-up (C)		GSCR Sect. 2.1.8	Not Tested	Minor ³
Attendant	No	Not Tested ²	Call waiting (C)		GSCR Sect. 2.1.9	Met	See note 3.
			Initiate all precedence levels (C)		GSCR Sect. 2.2.1	Not Tested	
			Visual display (C)		GSCR Sect. 2.2.2	Not Tested	
			Override class of service (C)		GSCR Sect. 2.2.3	Not Tested	
			Override busy line (C)		GSCR Sect. 2.2.4	Not Tested	
			Call deflection (C)		GSCR Sect. 2.2.5	Not Tested	
			Auto recall (C)		GSCR Sect. 2.2.6	Not Tested	
Waiting queue (C)		GSCR Sect. 2.2.7	Not Tested				

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Features & Capabilities (continued)						
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional	Reference	Test Results	Remarks
Public Safety	No	Not Tested ²	911 (C)	GSCR Sect. 2.4.1	Not Tested	
			Trace of terminating calls (C)	GSCR Sect. 2.4.2	Not Tested	
			Outgoing call trace (C)	GSCR Sect. 2.4.3	Not Tested	
			Tandem call trace (C)	GSCR Sect. 2.4.4	Not Tested	
			Trace of a call in progress (C)	GSCR Sect. 2.4.5	Not Tested	
Preset Conferencing	No	Not Tested ²	Support 10 bridges; 1 originator and 20 conferees per bridge (C)	GSCR Sect. 2.1.6	Not Tested	
			Assign up to 20 address numbers per bridge (C)	GSCR Sect. 2.6	Not Tested	
			Use KXX codes for bridge access (C)	GSCR Sect. 2.6	Not Tested	
			Conference notification recorded announcement (C)	GSCR Sect. 2.6.1	Not Tested	
			Auto retrieval and alternate address (C)	GSCR Sect. 2.6.2	Not Tested	
			Bridge release (C)	GSCR Sect. 2.6.3	Not Tested	
			Lost connection (C)	GSCR Sect. 2.6.4	Not Tested	
			Secondary conferencing (C)	GSCR Sect. 2.6.5	Not Tested	
Nailed-Up Connections	No	Not Tested ²	Address translation (C)	GSCR Sect. 2.7	Not Tested	
			Between any two like terminations (C)	GSCR Sect. 2.8	Not Tested	
			PCM-24 and PCM-30, both CAS and CCS (C)	GSCR Sect. 2.8	Not Tested	
			Supervision passed end-to-end for A/D or D/A (C)	GSCR Sect. 2.8	Not Tested	
			Monitored and auto reconfigure (C)	GSCR Sect. 2.8	Not Tested	
PAT	No	Not Tested ²	Support at least 10% of circuits as nailed-up (C)	GSCR Sect. 2.8	Not Tested	
			Non-preemptable (C)	GSCR Sect. 2.8	Not Tested	
			Classmark for/not for PAT screening (C)	GSCR Sect. 2.11.1	Not Tested	
			7 PAT mechanisms (C)	GSCR Sect. 2.11.1	Not Tested	
			Outgoing call screening (C)	GSCR Sect. 2.11.1.1	Not Tested	
			Functional structure (C)	GSCR Sect. 2.11.1.2	Not Tested	
			Overflow Process (C)	GSCR Sect. 2.11.1.3	Not Tested	
			Simultaneous calls limitation (C)	GSCR Sect. 2.11.1.4	Not Tested	
			Decrementing call-in-progress count (C)	GSCR Sect. 2.11.1.5	Not Tested	
			Call treatment (C)	GSCR Sect. 2.11.1.6	Not Tested	
			Queuing (C)	GSCR Sect. 2.11.1.7	Not Tested	
			Attendant calls (C)	GSCR Sect. 2.11.1.8	Not Tested	
DSN Hotline Services	No	Not Tested ²	Operations measurement registers (C)	GSCR Sect. 2.11.1.9	Not Tested	
			Maintenance and Administration of thresholds (C)	GSCR Sect. 2.11.1.10	Not Tested	
			Hotline restrictions (C)	GSCR Sect. 2.12	Not Tested	
			Auto initiate (C)	GSCR Sect. 2.12	Not Tested	
			Analog and digital (C)	GSCR Sect. 2.12	Not Tested	
DSN Hotline Services	No	Not Tested ²	Subscription basis (C)	GSCR Sect. 2.12	Not Tested	
			Protected hotline calling (C)	GSCR Sect. 2.12.1-4	Not Tested	
			WWNDP interoperable (C)	GSCR Sect. 2.12.5	Not Tested	

Table 2-6. SUT Interoperability Requirements/Status (continued)

DSN Features & Capabilities (continued)						
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional	Reference	Test Results	Remarks
Network Management	No	Not Tested ²	Interfaces (C)	GSCR Sect. 9.1	Not Tested	
			Measurements and data generation (C)	GSCR Sect. 9.2	Not Tested	
			Fault management (C)	GSCR Sect. 9.3	Not Tested	
			Configuration management (C)	GSCR Sect. 9.4	Not Tested	
			Accounting management (C)	GSCR Sect. 9.5	Not Tested	
			Performance management (C)	GSCR Sect. 9.6	Not Tested	
			NM controls (C)	GSCR Sect. 9.7	Not Tested	
Remote access (C)	GSCR Sect. 9.8	Not Tested				
ISDN Services	No	Not Tested ²	EKTS (C)	GSCR Sect. 10, table 10-3	Not Tested	
Synchronization ⁴	Yes	Certified	Line timing mode (R)	GSCR Sect. 11.1.1.2	Met	
			Internal Stratum 4 (R)	GSCR Sect. 11.1.2.2	Met	
Reliability	Yes	Certified	GR-512-CORE (R)	GSCR Sect. 12	Met	
Security	Yes	See note 5.	DITSCAP (R)	GSCR Sect. 13	See note 5.	
VoIP System	No	Certified	ITU-T G.711 PCM Codec (R)	GSCR App. 3	Met	
			Security in accordance with DITSCAP (R)	GSCR App. 3	Not Tested ⁵	
			NM (R)	GSCR App. 3	Met	
			Latency @ 60 ms or less (R)	GSCR App. 3	Met	
			IPv6 capable (R)	GSCR App. 3	Not Tested ⁶	
			CoS/QoS (R)	GSCR App. 3	Met	
			Traffic Engineering (R)	GSCR App. 3	Met	
Network Gateway						
Gateway	Critical	Interface Status	GSCR Requirement Required or Conditional	Reference	Test Results	Remarks
PSTN	Yes	Certified	Trunking	Positive Identification Control (R)	CJCSI 6215.01B	Met
				On-Netting (R)	CJCSI 6215.01B	Met
				Off-Netting (R)	CJCSI 6215.01B	Met

Table 2-6. SUT Interoperability Requirements/Status (continued)

Network Gateway							
Gateway	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
DRSN ⁷	Yes	Certified	Access	Alerting Signals and Tones (R)	GSCR Sect. 5.5	Met	
				Call Processing (R)	GSCR Sect. 4.4	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
			Voice	Analog busy/idle (R)	GSCR Sect. 4.3.4.1	Met	
				MOS (C)	CJCSI 6215.01B	Met	
				MLPP (C)	GSCR Sect. 3	Met	
				Secure Calls (C)	CJCSI 6215.01B	Met	
Alerting Signals and Tones (R)	GSCR Sect. 5.5	Met					
LEGEND: 2W - 2-Wire 911 - 911 Emergency Service A/D - Analog to Digital Conversion ANSI - American National Standards Institute App. - Appendix 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 CoS - Class of Service D/A - Digital to Analog Conversion DISA - Defense Information Systems Agency DISR - DoD Information Technology Standards Registry DITSCAP - DoD Information Technology Security and Accreditation Process DN - Directory Number DoD - Department of Defense DRSN - Defense Red Switch Network DSN - Defense Switched Network E1 - European Basic Multiplex Rate EIA - Electronic Industries Alliance EKTS - Electronic Key Telephone System G.711 - PCM of Voice Frequencies GR - Generic Requirement GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU - International Telecommunication Union ITU-T - ITU-Telecommunication Standardization Sector JITC - Joint Interoperability Test Command Kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 Mbps - Megabits per second MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score ms - milliseconds NI 1/2 - National ISDN Standard 1 or 2 NM - Network Management 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 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network QoS - Quality of Service R - Required Sect. - Section SS7 - Signaling System 7 STE - Secure Terminal Equipment STU-III - Secure Telephone Unit-3 rd generation SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TDM - Time Division Multiplexing 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 This interface is not a critical requirement for a PBX 1. The risk of not testing is minor. 2 This interface or feature is not supported by the SUT. The operational impact is minor because it is not a critical requirement for a PBX 1. 3 Meets Call-Forwarding busy and no-answer call feature requirements on analog phones. Call Hold and Call Waiting Features were met on VoIP instruments with two line appearances. The following common call features are disabled when Multi-Level Precedence and Preemption is provisioned: Call Pick-up, Call Transfer, Conference Calling, Precedence Call Waiting, Multi-line Hunt Service, or Three-way Calling. 4 To meet this requirement, a direct T1 interface must be connected between multiple gateways to synchronize timing of all TDM-based interfaces between gateways. 5 Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 6 Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades. 7 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.							