



## DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND

P.O. BOX 12798

FORT HUACHUCA, ARIZONA 85670-2798

IN REPLY  
REFER TO:

Networks and Transport Division (JTE)

14 September 2006

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**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 Nortel DSN CS1000M-SG Digital Switching System with Software Release 4.5w and Product Enhancements (including VoIP) is hereinafter referred to as the System Under Test (SUT). The SUT met all of its critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The SUT is certified for VoIP specifically with certified Assured Services Voice Application Local Area Networks (ASVALANs) posted on the JITC Telecom Switched Services Interoperability (TSSI) program web page ([http://jite.fhu.disa.mil/tssi/cert/cert\\_asvalan.html](http://jite.fhu.disa.mil/tssi/cert/cert_asvalan.html)) approved product list. The JITC also determined, through analysis, that the Nortel DSN CS1000M-MG with VoIP, is also certified for joint use within the DSN. The analysis determined the DSN CS1000M-MG employs the same software and trunk/line card hardware as the Nortel DSN CS1000M-SG, and therefore is functionally identical to the Nortel DSN CS1000M-SG. The difference between the two switches is scalability. The DSN CS1000M-SG supports up to a maximum of 2000 ports and the DSN CS1000M-MG supports a maximum of 16,000 ports. When the SUT is fielded without VoIP, it is certified for Joint use within the DSN as well. The SUT without VoIP product is referred to and marketed within DoD as the Nortel DSN M1 Option 61C. Additionally, the DSN CS1000M-MG without VoIP is also certified for joint use within the DSN via the same analysis done on the CS1000M-MG with VoIP. The DSN CS1000M-MG without VoIP is referred to and marketed within DoD as the Nortel DSN M1 Option 81C. The listed test discrepancies shown in the Certification Testing Summary (enclosure 2), have an overall minor operational impact. One of the discrepancies, was with the Call Forwarding Variable (a conditional requirement), which does not properly interact with precedence calls above

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

ROUTINE and is therefore not certified for joint use within the DSN. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: SMEO, Private Branch Exchange (PBX) 1, and PBX 2. 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 a review of the vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 21 November 2005 through 13 January 2006. Regression testing was conducted from 1 through 27 May 2006, and 19 through 30 June 2006. Patches were applied and additional regression testing was conducted 11 through 20 July 2006. Final regression testing was conducted 25 August 2006. Review of the LoC was completed on 30 July 2006. Review of Product Enhancement Packages was completed on 7 September 2006. 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 interoperability test summary of the SUT is contained in table 1. The SMEO required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in table 2. This interoperability test status is based on the SUT's ability to meet:

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

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 1. SUT Interoperability Test Summary**

<b>DSN Trunk Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration. <sup>1</sup> The SUT recognizes a wink start signal greater than the specified maximum limit. <sup>2</sup> The SUT does not support glare hold resolution for their CAS trunks. <sup>3</sup>
T1 CAS (MFR1)	No	Not Tested	This interface is not supported. <sup>4</sup>
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration. <sup>1</sup> The SUT does not support glare hold resolution for their CAS trunks. <sup>3</sup> The on/off hook pulse that frames the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). <sup>5</sup>
E1 CAS (MFR1)	No	Not Tested	This interface is not supported. <sup>4</sup>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT fails to automatically return trunks to a maintenance busy condition after the span is broken then restored. <sup>6</sup>
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
T1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. <sup>4</sup>
E1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. <sup>4</sup>
<b>DSN Line Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs.
ISDN BRI NI 1/2	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support NI2 BRI. <sup>7</sup> The only supported and certified interface is NI1 BRI with a single appearance of a single directory number. <sup>8</sup> The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications. <sup>9</sup> The BRI instruments do not support precedence call waiting. <sup>10</sup>
2-Wire Proprietary Digital	No	Certified	Met all critical CRs and FRs.
VoIP (ITU-T H.323 Proprietary)	No	Certified	Met all critical CRs and FRs. Precedence call waiting indication is unique on VoIP phones. <sup>11</sup>
<b>DSN Features and Capabilities</b>			
<b>Features and Capabilities</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
Common Features	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not correctly support the call forwarding variable feature. <sup>12</sup> The conference disconnect tone that is provided by the SUT does not meet the specifications. <sup>13</sup>
Attendant	No	Certified	Met all critical CRs and FRs with the following minor exceptions: Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call. <sup>14</sup>
Public Safety	Yes	Certified	Met all critical CRs and FRs with the following exception: The SUT cannot perform a tandem call trace of a specified distant office directory number. <sup>15</sup>
Preset Conferencing	No	Not Tested	This feature is not supported. <sup>16</sup>

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 1. SUT Interoperability Test Summary (continued)**

DSN Features and Capabilities					
Features and Capabilities	Critical	Status	Remarks		
Nailed-up Connections	No	Not Tested	This feature is not supported. <sup>16</sup>		
PAT	No	Not Tested	This feature is not supported. <sup>16</sup>		
DSN Hotline Services	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a protected hotline specified list. <sup>17</sup>		
Network Management	Yes	Certified	Met all critical CRs and FRs.		
Multiline Hunt Service	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT will not permit a BRI station to be a member of a multiline hunt group. <sup>18</sup>		
ISDN Services (EKTS)	No	Not Tested	This feature is not supported. <sup>16</sup>		
Synchronization	Yes	Certified	Met all critical CRs and FRs.		
Reliability	Yes	Certified	Met all critical CRs and FRs.		
Security	Yes	See note 19	See note 19		
VoIP System	No	Certified	The SUT is certified for VoIP specifically with certified ASVALAN posted on the JITC TSSI program web page ( <a href="http://jitic.fhu.disa.mil/tssi/cert_nortel">http://jitic.fhu.disa.mil/tssi/cert_nortel</a> ) approved product list. See note 20.		
Network Gateways					
Gateway	Interface & Signaling	Critical	Status	Remarks	
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.	
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all critical CRs and FRs.	
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs.	
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.	
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.	
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>21</sup>	Met all critical CRs and FRs.	
<b>LEGEND:</b>					
ANSI	- American National Standards Institute	GSCR	- Generic Switching Center Requirements	PM	- Program Manager
ASVALAN	- Assured Services Voice Application Local Area Network	H.323	- Standard for multi-media communications on packet-based networks	PRI	- Primary Rate Interface
BRI	- Basic Rate Interface	IPv4	- Internet Protocol version 4	PSTN	- Public Switched Telephone Network
CAS	- Channel Associated Signaling	IPv6	- Internet Protocol version 6	Q.931	- Signaling Standard for ISDN
CRs	- Capability Requirements	ISDN	- Integrated Services Digital Network	Q.955.3	- ISDN signaling standard for E1 MLPP
DISA	- Defense Information Systems Agency	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	SMEO	- Small End Office
DoD	- Department of Defense	JITC	- Joint Interoperability Test Command	SS7	- Signaling System 7
DP	- Dial Pulse	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements	SUT	- System Under Test
DRSN	- Defense Red Switch Network	Mbps	- Megabits per second	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DSN	- Defense Switched Network	MFR1	- Multifrequency Recommendation 1	T1.607	- ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DSS1	- Digital Subscriber Signaling 1	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
DTMF	- Dual Tone Multi-Frequency	ms	- millisecond	TSSI	- Telecom Switched Services Interoperability
E1	- European Basic Multiplex Rate (2.048 Mbps)	NI 1/2	- National ISDN 1 or 2	TPC	- Twisted Pair Copper
EKTS	- Electronic Key Telephone System	PAT	- Precedence Access Threshold	VoIP	- Voice over Internet Protocol
FRs	- Feature Requirements	PBX	- Private Branch Exchange		
GR	- Generic Requirement				
GR-506	- LSSGR: Signaling for Analog Interfaces				

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 1. SUT Interoperability Test Summary (continued)**

<b>NOTES:</b>	
1	When any active trunk interface is physically broken and repaired, the SUT does not restore the span to service and remove the yellow alarm condition within the required time duration. In accordance with the GSCR paragraphs 7.1.4 and 7.2.2, the time required for the removal of the alarm condition after the physical restoration of a broken trunk is 15 (+/-5) seconds. The E1 CAS can take up to 90 seconds to restore, and all the other interfaces require 30 seconds to be restored. The operational impact is minor since the alarm clears automatically when the span is returned to service.
2	T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 milliseconds (ms) to 350ms. The SUT recognizes wink start signals from 100ms to 925ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290ms this anomaly has no operational impact.
3	The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since the SUT is a subtending switch off of a Multifunction Switch (MFS) and all MFS support glare hold, complementing the SUTs capability to support glare release. Therefore, the operational impact is minor.
4	This interface is not supported. There is no operational impact because it is not a critical requirement.
5	The on/off hook pulse that initiates the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). The pulse width was measured to be greater than 100 ms (highest @ 128 ms) about 20% of the time, but never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.
6	If a T1 ISDN PRI interface is broken then restored when all channels are in a maintenance busy condition, the SUT fails to automatically return the channels to the previous busy condition. This anomaly has no operational impact because it only occurs when the SUT is in a maintenance condition.
7	The SUT does not support an National ISDN (NI)2 BRI interface. The only supported and certified BRI interface is NI1. The NI2 BRI interface is required for SMEO operation as specified by GSCR paragraph 2.3.3. Since the primary differences between NI1 and NI2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future, this anomaly has a minor operational impact.
8	The SUT will only support a BRI NI1 voice line with a single directory number and a single appearance of a directory number. However, multiple appearances with different directory numbers can be supported with the digital proprietary instruments which account for the majority of digital instruments fielded within the DSN. Therefore, the operational impact is minor.
9	The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in the GSCR 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.
10	The SUT does not support precedence call waiting for their BRI instruments; however the SUT does support precedence call waiting for all other phone types. Also, this requirement is conditional and therefore, has no operational impact.
11	The SUT supports the "call waiting" indication on VoIP telephones with visual indicators in lieu of audible tones as specified by the GSCR. When call waiting is invoked on a VoIP phone, the phone displays call waiting text along with a flashing symbol. The call waiting symbol flashes twice for a ROUTINE call and three times for precedence above ROUTINE call. Since the requirement for audible tone is conditional, and there are two visual indicators to alert the VoIP user of a waiting call, the operational impact of not supporting audible tones is minor.
12	When call forwarding variable (CFV) is assigned to any station on the SUT (except BRI; does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or Public Switched Telephone Network (PSTN). Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. Per the GSCR only ROUTINE precedence calls will be forwarded and precedence calls above are diverted to the attendant console, night service or alternate directory number, therefore this feature is not certified for use within the DSN. This feature is a conditional requirement and will have a minor operational impact.
13	The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.
14	Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR 2.2.4. The proper override tone, however, is given to a station active with a call prior to the attendant's bridging into the active call. Since attendants rarely bridge into calls and active calls remain connected when an attendant does bridge into a call, the operational impact is minor.
15	The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in the GSCR. Since the SUT is predominately fielded within the DSN as a SMEO with no tandeming (e.g. subtending PBX1 or PBX2), this anomaly has a minor operational impact.
16	This feature is not supported. There is no operational impact because it is not a critical requirement.
17	The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by the GSCR. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. The operational impact is minor.
18	The SUT will not permit an ISDN BRI station to be a member of a multiline hunt group. All other phone types can be configured as members of a multiline hunt group. Since ISDN BRI voice users are rarely used within the DSN and this feature can be accomplished on the SUT with analog and digital proprietary stations, this anomaly has a minor operational impact.
19	Security is tested by DISA-led Information Assurance test teams and published in a separate report.
20	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 their respective company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008: <ul style="list-style-type: none"> <li>a. Conformance with IPv6 standards profile contained in the DoD IT Standards Registry (DISR).</li> <li>b. Maintaining interoperability in heterogeneous environments and with IPv4.</li> <li>c. Commitment to upgrade as the IPv6 standard evolves.</li> <li>d. Availability of contractor/vendor IPv6 technical support.</li> </ul>
21	Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 2. SMEO Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 SS7 (ANSI T1.619a)	No	Trunking	<ul style="list-style-type: none"> <li>• Framing (R)</li> <li>• Line Code (R)</li> <li>• Signaling (R)</li> <li>• Alarms (R)</li> <li>• WWNDP (R)</li> <li>• Outpulsing digit formats (R: CAS only)</li> <li>• Routing (R)</li> <li>• Trunk Groups (R)</li> <li>• Call Processing (R)</li> <li>• CAS to CCS trunk interworking (C)</li> <li>• PCM-24/PCM-30 Interoperation (R)</li> <li>• Direct Inward Dialing (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 5</li> <li>• GSCR Sect. 2.5.7, 7.1.4 &amp; 7.2.2</li> <li>• GSCR Sect. 4.5.1</li> <li>• GSCR Sect. 4.5.2</li> <li>• GSCR Sect. 4.2</li> <li>• GSCR Sect. 2.5.5 &amp; 2.5.6</li> <li>• GSCR Sect. 4</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 7.3</li> <li>• GSCR Sect. 2.3.2</li> </ul>
E1 SS7 (ITU-T Q.735.3)	No (Europe only)			
T1 CAS (MFR1)	No			
T1 CAS (DTMF, DP)	Yes			
E1 CAS (DTMF, DP)	Yes (Europe only)	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
E1 CAS (MFR1)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
DSN Line Interfaces				
Interface	Critical	Requirements Required or Conditional		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line Signaling (R)</li> <li>• Loop Start Line (R)</li> <li>• Alerting Signals and Tones (R)</li> <li>• WWNDP (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• 2W user access (R: 2-Wire Analog only)</li> <li>• Analog busy/idle (R: 2-Wire Analog only)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.1</li> <li>• GSCR Sect. 5.2</li> <li>• GSCR Sect. 5.2.1</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.3</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes			
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Announcements (R)</li> <li>• MLPP (R)</li> <li>• Secure Calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.1.3</li> <li>• GSCR Sect. 3.4.3/3.9</li> <li>• CJCSI 6215.01B</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
VoIP	No	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R)</li> <li>• 64 kbps switched data (R: BRI only)</li> <li>• NX56 synchronous BER (R: BRI only)</li> <li>• NX64 synchronous BER (R: BRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
		VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: BRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 2. SMEO Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Common Features	Yes	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer and conference calling and call hold (C)</li> <li>• Call forwarding (C)</li> <li>• Call pick-up (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.2</li> <li>• GSCR Sect. 2.1.3</li> <li>• GSCR Sect. 2.1.4</li> <li>• GSCR Sect. 2.1.5</li> <li>• GSCR Sect. 2.1.6</li> <li>• GSCR Sect. 2.1.7</li> <li>• GSCR Sect. 2.1.8</li> <li>• GSCR Sect. 2.1.9</li> </ul>
Attendant	No	<ul style="list-style-type: none"> <li>• Initiate all precedence levels (C)</li> <li>• Visual display (C)</li> <li>• Override class of service (C)</li> <li>• Override busy line (C)</li> <li>• Call deflection (C)</li> <li>• Auto recall (C)</li> <li>• Waiting queue (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.2.1</li> <li>• GSCR Sect. 2.2.2</li> <li>• GSCR Sect. 2.2.3</li> <li>• GSCR Sect. 2.2.4</li> <li>• GSCR Sect. 2.2.5</li> <li>• GSCR Sect. 2.2.6</li> <li>• GSCR Sect. 2.2.7</li> </ul>
Public Safety	Yes	<ul style="list-style-type: none"> <li>• E911 (C)</li> <li>• Trace of terminating calls (R)</li> <li>• Outgoing call trace (R)</li> <li>• Tandem call trace (R)</li> <li>• Trace of a call in progress (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.4.1</li> <li>• GSCR Sect. 2.4.2</li> <li>• GSCR Sect. 2.4.3</li> <li>• GSCR Sect. 2.4.4</li> <li>• GSCR Sect. 2.4.5</li> </ul>
Preset Conferencing	No	<ul style="list-style-type: none"> <li>• Support 10 bridges; 1 originator and 20 conferees per bridge (C)</li> <li>• Assign up to 20 address numbers per bridge (C)</li> <li>• Use KXX codes for bridge access (C)</li> <li>• Conference notification recorded announcement (C)</li> <li>• Auto retrial and alternate address (C)</li> <li>• Bridge release (C)</li> <li>• Lost connection (C)</li> <li>• Secondary conferencing (C)</li> <li>• Address translation (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6.1</li> <li>• GSCR Sect. 2.6.2</li> <li>• GSCR Sect. 2.6.3</li> <li>• GSCR Sect. 2.6.4</li> <li>• GSCR Sect. 2.6.5</li> <li>• GSCR Sect. 2.7</li> </ul>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Between any two like terminations (C)</li> <li>• PCM-24 and PCM-30, both CAS and CCS (C)</li> <li>• Supervision passed end-to-end for A/D or D/A (C)</li> <li>• Monitored and auto reconfigure (C)</li> <li>• Support at least 10% of circuits as nailed-up (C)</li> <li>• Non-preemptable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.8</li> </ul>
PAT	No	<ul style="list-style-type: none"> <li>• Classmark for/not for PAT screening (C)</li> <li>• 7 PAT mechanisms (C)</li> <li>• Outgoing call screening (C)</li> <li>• Functional structure (C)</li> <li>• Simultaneous calls limitation (C)</li> <li>• Overflow process (C)</li> <li>• Decrementing call-in-progress count (C)</li> <li>• Call treatment (C)</li> <li>• Queuing (C)</li> <li>• Attendant calls (C)</li> <li>• Operations measurement registers (C)</li> <li>• Maintenance and Administration of thresholds (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1.1</li> <li>• GSCR Sect. 2.11.1.2</li> <li>• GSCR Sect. 2.11.1.3</li> <li>• GSCR Sect. 2.11.1.4</li> <li>• GSCR Sect. 2.11.1.5</li> <li>• GSCR Sect. 2.11.1.6</li> <li>• GSCR Sect. 2.11.1.7</li> <li>• GSCR Sect. 2.11.1.8</li> <li>• GSCR Sect. 2.11.1.9</li> <li>• GSCR Sect. 2.11.1.10</li> </ul>

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 2. SMEO Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
DSN Hotline Services	Yes	<ul style="list-style-type: none"> <li>• Hotline restrictions (R)</li> <li>• Auto initiate (R)</li> <li>• Analog and digital (R)</li> <li>• Subscription basis (R)</li> <li>• Protected hotline calling (R)</li> <li>• WWNDP interoperable (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12.1-4</li> <li>• GSCR Sect. 2.12.5</li> </ul>
Network Management	Yes	<ul style="list-style-type: none"> <li>• Interfaces (R)</li> <li>• Measurements and data generation (R)</li> <li>• Fault management (R)</li> <li>• Configuration management (R)</li> <li>• Accounting management (R)</li> <li>• Performance management (R)</li> <li>• NM controls (R)</li> <li>• Remote access (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 9.1</li> <li>• GSCR Sect. 9.2</li> <li>• GSCR Sect. 9.3</li> <li>• GSCR Sect. 9.4</li> <li>• GSCR Sect. 9.5</li> <li>• GSCR Sect. 9.6</li> <li>• GSCR Sect. 9.7</li> <li>• GSCR Sect. 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• EKTS (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 10, table 10-3</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 11.1.1.2</li> <li>• GSCR Sect. 11.1.2.2</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• GR-512-CORE (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 12</li> </ul>
Security <sup>1</sup>	Yes	<ul style="list-style-type: none"> <li>• DITSCAP (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 13</li> </ul>
<b>VoIP</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
VoIP System	No	<p>VoIP function is conditional. If VoIP is provided, <b>all</b> of the following requirements must be met:</p> <ul style="list-style-type: none"> <li>• Voice Quality with Mean Opinion Score of 4.0 or better</li> <li>• Class of Service (CoS) and Quality of Service (QoS)</li> <li>• ITU-T G.711 PCM Codec</li> <li>• Traffic Engineering</li> <li>• Security in accordance with DITSCAP</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 ms</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> <li>• GSCR App. 3, paragraph 1.7</li> </ul>
<b>Network Gateways</b>			
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
PSTN <sup>2</sup>	Yes	<p>Trunking</p> <ul style="list-style-type: none"> <li>• Positive Identification Control (R)</li> <li>• On-Netting (R)</li> <li>• Off-Netting (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> </ul>
DRSN <sup>3</sup>	Yes	<p>Access</p> <ul style="list-style-type: none"> <li>• Alerting Signals and Tones (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• Analog busy/idle (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
		<p>Voice</p> <ul style="list-style-type: none"> <li>• MOS (C)</li> <li>• MLPP (C)</li> <li>• Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

**Table 2. SMEO Requirements (continued)**

LEGEND:					
2W	- 2-Wire	EIA	- Electronic Industries Alliance	PAT	- Precedence Access Threshold
A/D	- Analog to Digital Conversion	G.711	- Standard for PCM of Voice Frequencies	PCM	- Pulse Code Modulation
ANSI	- American National Standards Institute	GR	- Generic Requirement (Telcordia)	PCM-24	- Pulse Code Modulation - 24 Channels
App.	- Appendix	GSCR	- Generic Switching Center Requirements	PCM-30	- Pulse Code Modulation - 30 Channels
BER	- Bit Error Ratio	H.320	- Standard for Narrowband VTC	PRI	- Primary Rate Interface
BRI	- Basic Rate Interface	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PSTN	- Public Switched Telephone Network
C	- Conditional	IPV6	- Internet Protocol version 6	Q.735.3	- SS7 Signaling Standard for E1 MLPP
CAS	- Channel Associated Signaling	ISDN	- Integrated Services Digital Network	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CCS	- Common Channel Signaling	IT	- Information Technology	R	- Required
CJCS I	- Chairman of the Joint Chiefs of Staff Instruction	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	Sect.	- Section
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	SMEO	- Small End Office
DISA	- Defense Information Systems Agency	kbps	- kilobits per second	SS7	- Signaling System 7
DISR	- DoD IT Standards Registry	KXX	- K= any number 2-8; X= any number 1-9	STE	- Secure Terminal Equipment
DITSCAP	- DoD IT Security and Accreditation Process	Mbps	- Megabits per second	STU-III	- Secure Telephone Unit - 3 <sup>rd</sup> Generation
DN	- Directory Number	MFR1	- Multi-Frequency Recommendation 1	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DoD	- Department of Defense	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN Signaling Standard for T1
DP	- Dial Pulse	MOS	- Mean Opinion Score	TIA	- Telecommunications Industry Association
DSN	- Defense Switched Network	ms	- milliseconds	TIA/EIA-465A	- Group 3 Facsimile Apparatus for Document Transmission
DRSN	- Defense Red Switch Network	NI 1/2	- National ISDN Standard 1 or 2	VBD	- Variable bit data
DTMF	- Dual Tone Multi-Frequency	NM	- Network Management	VLAN	- Virtual LAN
E1	- European Basic Multiplex Rate (2.048 Mbps)	NX56	- Data format restricted to multiples of 56 kbps	VoIP	- Voice over Internet Protocol
E911	- Emergency 911 Service	NX64	- Data format restricted to multiples of 64 kbps	VTC	- Video Teleconferencing
EKTS	- Electronic Key Telephone System			WWNDP	- Worldwide Numbering and Dialing Plan

**NOTES:**

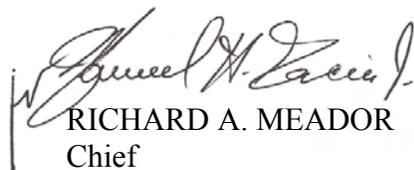
- Information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.
- Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.
- Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

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>.

6. The JITC point of contact is John Hooper, DSN 312-879-5041, commercial (520) 538-5041, FAX DSN 879-4347, or e-mail to [john.hooper@disa.mil](mailto:john.hooper@disa.mil). The tracking number for the SUT is 51824.

FOR THE COMMANDER:

2 Enclosures a/s



RICHARD A. MEADOR  
Chief  
Networks and Transport Division

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG) and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

Distribution:

Joint Staff J6I, Room 1E596, Pentagon, Washington, DC 20318-6000

Joint Interoperability Test Command, Liaison, ATTN: TED/JT1, 2W24-8C, P.O. Box 4502, Falls Church, VA 22204-4502

Defense Information Systems Agency, Net-Centricity Requirements and Assessment Branch, ATTN: GE333, Room 244, P.O. Box 4502, Falls Church, VA 22204-4502

Office of Chief of Naval Operations (N71CC2), CNO N6/N7, 2000 Navy Pentagon, Washington, DC 20350

Headquarters U.S. Air Force, AF/XICF, 1800 Pentagon, Washington, DC 20330-1800

Department of the Army, Office of the Secretary of the Army, CIO/G6, ATTN: SAIS-IOQ, 107 Army Pentagon, Washington, DC 20310-0107

U.S. Marine Corps (C4ISR), MARCORSSYSCOM, 2200 Lester St., Quantico, VA 22134-5010

DOT&E, Net-Centric Systems and Naval Warfare, 1700 Defense Pentagon, Washington, DC 20301-1700

U.S. Coast Guard, CG-64, 2100 2nd St. SW, Washington, DC 20593

Defense Intelligence Agency, 2000 MacDill Blvd., Bldg 6000, Bolling AFB, Washington, DC 20340-3342

National Security Agency, ATTN: DT, Suite 6496, 9800 Savage Road, Fort Meade, MD 20755-6496

Director, Defense Information Systems Agency, ATTN: GS235, Room 5W24-8A, P.O. Box 4502, Falls Church, VA 22204-4502

Office of Assistant Secretary of Defense (NII)/DoD CIO, Crystal Mall 3, 7th Floor, Suite 7000, 1851 S. Bell St., Arlington, VA 22202

Office of Under Secretary of Defense, AT&L, Room 3E144, 3070 Defense Pentagon, Washington, DC 20301

U.S. Joint Forces Command, J68, Net-Centric Integration, Communications, and Capabilities Division, 1562 Mitscher Ave., Norfolk, VA 23551-2488

Defense Information Systems Agency (DISA), ATTN: GS23 (Mr. Osman), Room 5w23, 5275 Leesburg Pike (RTE 7), Falls Church, VA 22041

## **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, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005
- (e) Executive Office of the President, "Transition Planning for Internet Protocol version 6 (IPv6)," 2 August 2005
- (f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 1, Revision 1," 1 June 2005

## CERTIFICATION TESTING SUMMARY

**1. SYSTEM TITLE.** Nortel Defense Switched Network (DSN) Communications Server (CS)1000M-Single Group (SG), hereinafter referred to as the System Under Test (SUT), and DSN CS1000M-Multi Group (MG) (including Voice over Internet Protocol [VoIP]) and DSN Meridian 1 (M1) Option 61C and DSN M1 Option 81C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages.

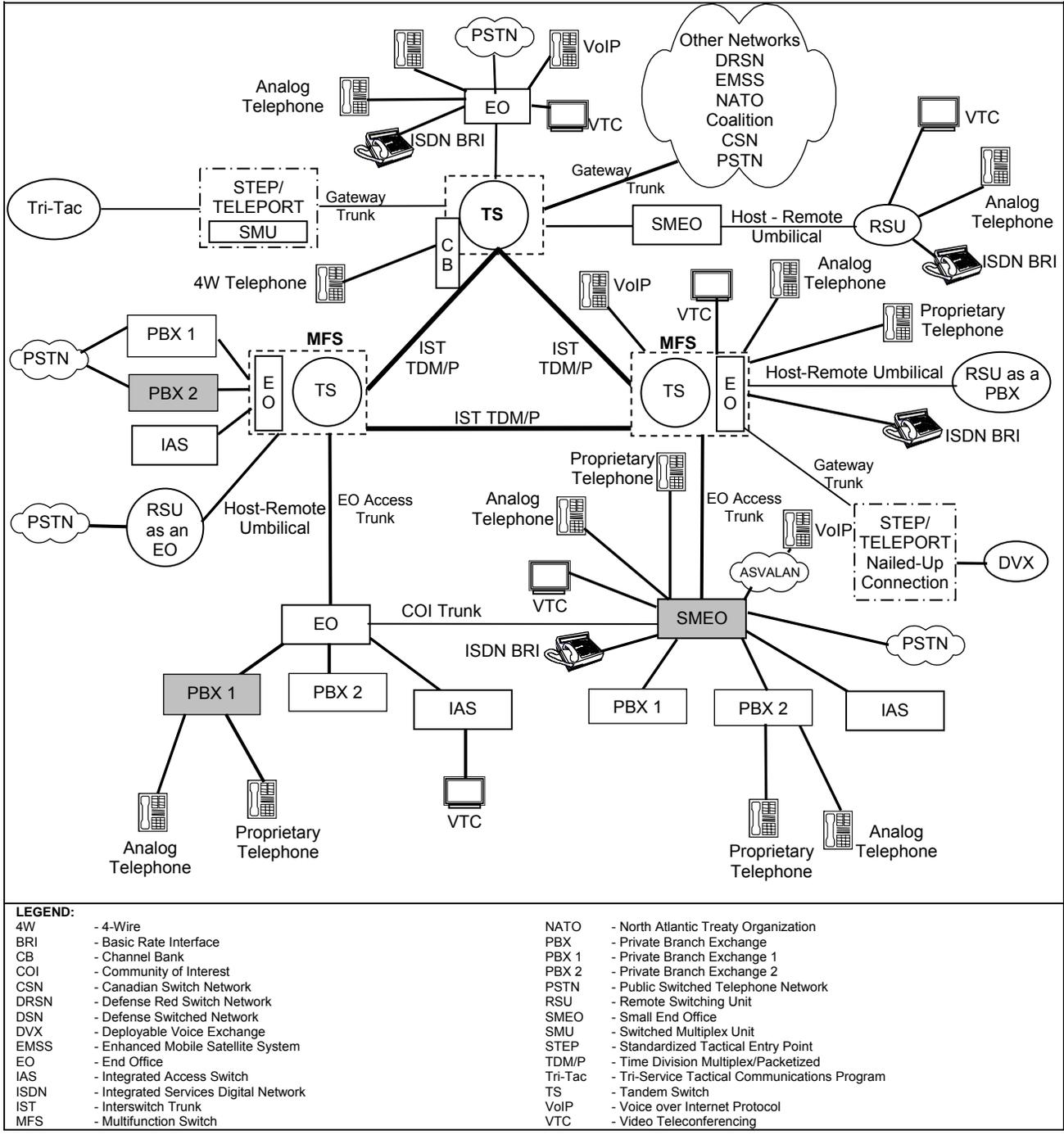
**2. PROPONENT.** Defense Information Systems Agency (DISA).

**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 digital telecommunications switching system that supports analog, VoIP, and digital Integrated Services Digital Network (ISDN) Basic Rate Interface (BRI) lines. The SUT supports analog and digital trunks including ISDN Primary Rate Interface (PRI) and Channel Associate Signaling (CAS). The SUT offers the following features: scalable, distributed platform for growth from 200 to 2000 lines, modular client/server architecture for flexibility, scalability, and a redundant call processing core for extra reliability in mission-critical enterprises. The SUT is certified for VoIP specifically with certified Assured Services Voice Application Local Area Networks (ASVALANs) posted on the JITC Telecom Switched Services Interoperability (TSSI) program web page (<http://jitc.fhu.disa.mil/tssi>) approved product list. The Nortel DSN CS1000M-MG employs the same software and trunk/line card hardware as the SUT and was developed for scalability purposes. JITC analysis determined the DSN CS1000M-MG including VoIP to be functionally identical to the SUT for interoperability certification purposes. The DSN CS1000M-SG supports up to a maximum of 2000 ports, 64 input-output (IO) ports, and 32 loops for a total of 36,000 hundred call seconds (CCS). The DSN CS1000M-MG supports a maximum of 16,000 ports, 255 IO ports, and 256 loops for a total of 288,000 CCS. The SUT is also certified without VoIP. When the SUT is deployed without VoIP, it is designated as the Nortel DSN M1 Option 61C. The DSN CS1000M-MG is also certified without VoIP. When the DSN CS1000M-MG is deployed without VoIP, it is designated as the Nortel DSN M1 Option 81C.

**6. OPERATIONAL ARCHITECTURE.** The 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 SMEOs. The Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. This architecture depicts the relationship of Military Department SMEOs to the other DSN switch types.



**Figure 2-1. DSN Architecture**

**7. REQUIRED SYSTEM INTERFACES.** Requirements specific to SMEOs 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 SMEO Capability and Feature Requirements (CRs/FRs) verified through JITC testing and/or vendor submission of LoC.

d. Internet Protocol version 6 (IPv6) requirements specified in the GSCR, paragraph 1.7, table 1-3, by 30 June 2008 in accordance with reference (e) verified through vendor submission of LoC signed by the Vice President of the company.

**Table 2-1. SMEO Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 SS7 (ANSI T1.619a)	No	Trunking	<ul style="list-style-type: none"> <li>• Framing (R)</li> <li>• Line Code (R)</li> <li>• Signaling (R)</li> <li>• Alarms (R)</li> <li>• WWNDP (R)</li> <li>• Outpulsing digit formats (R: CAS only)</li> <li>• Routing (R)</li> <li>• Trunk Groups (R)</li> <li>• Call Processing (R)</li> <li>• CAS to CCS trunk interworking (C)</li> <li>• PCM-24/PCM-30 Interoperation (R)</li> <li>• Direct Inward Dialing (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 5</li> <li>• GSCR Sect. 2.5.7, 7.1.4 &amp; 7.2.2</li> <li>• GSCR Sect. 4.5.1</li> <li>• GSCR Sect. 4.5.2</li> <li>• GSCR Sect. 4.2</li> <li>• GSCR Sect. 2.5.5 &amp; 2.5.6</li> <li>• GSCR Sect. 4</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 7.3</li> <li>• GSCR Sect. 2.3.2</li> </ul>
E1 SS7 (ITU-T Q.735.3)	No (Europe only)			
T1 CAS (MFR1)	No			
T1 CAS (DTMF, DP)	Yes			
E1 CAS (DTMF, DP)	Yes (Europe only)	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
E1 CAS (MFR1)	No (Europe only)	Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>

**Table 2-1. SMEO Requirements (continued)**

<b>DSN Line Interfaces</b>				
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line Signaling (R)</li> <li>• Loop Start Line (R)</li> <li>• Alerting Signals and Tones (R)</li> <li>• VVW NDP (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• 2W user access (R: 2-Wire Analog only)</li> <li>• Analog busy/idle (R: 2-Wire Analog only)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.1</li> <li>• GSCR Sect. 5.2</li> <li>• GSCR Sect 5.2.1</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.3</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes		Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Announcements (R)</li> <li>• MLPP (R)</li> <li>• Secure Calls (R)</li> </ul>
2-Wire Proprietary Digital	No	Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
VoIP	No	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R)</li> <li>• 64 kbps switched data (R: BRI only)</li> <li>• NX56 synchronous BER (R: BRI only)</li> <li>• NX64 synchronous BER (R: BRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
		VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: BRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
<b>DSN Features &amp; Capabilities</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
Common Features	Yes	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer and conference calling and call hold (C)</li> <li>• Call forwarding (C)</li> <li>• Call pick-up (C)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.2</li> <li>• GSCR Sect. 2.1.3</li> <li>• GSCR Sect. 2.1.4</li> <li>• GSCR Sect. 2.1.5</li> <li>• GSCR Sect. 2.1.6</li> <li>• GSCR Sect. 2.1.7</li> <li>• GSCR Sect. 2.1.8</li> <li>• GSCR Sect. 2.1.9</li> </ul>
Attendant	No	<ul style="list-style-type: none"> <li>• Initiate all precedence levels (C)</li> <li>• Visual display (C)</li> <li>• Override class of service (C)</li> <li>• Override busy line (C)</li> <li>• Call deflection (C)</li> <li>• Auto recall (C)</li> <li>• Waiting queue (C)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.2.1</li> <li>• GSCR Sect. 2.2.2</li> <li>• GSCR Sect. 2.2.3</li> <li>• GSCR Sect. 2.2.4</li> <li>• GSCR Sect. 2.2.5</li> <li>• GSCR Sect. 2.2.6</li> <li>• GSCR Sect. 2.2.7</li> </ul>
Public Safety	Yes	<ul style="list-style-type: none"> <li>• E911 (C)</li> <li>• Trace of terminating calls (R)</li> <li>• Outgoing call trace (R)</li> <li>• Tandem call trace (R)</li> <li>• Trace of a call in progress (R)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.4.1</li> <li>• GSCR Sect. 2.4.2</li> <li>• GSCR Sect. 2.4.3</li> <li>• GSCR Sect. 2.4.4</li> <li>• GSCR Sect. 2.4.5</li> </ul>
Preset Conferencing	No	<ul style="list-style-type: none"> <li>• Support 10 bridges; 1 originator and 20 conferees per bridge (C)</li> <li>• Assign up to 20 address numbers per bridge (C)</li> <li>• Use KXX codes for bridge access (C)</li> <li>• Conference notification recorded announcement (C)</li> <li>• Auto retrieval and alternate address (C)</li> <li>• Bridge release (C)</li> <li>• Lost connection (C)</li> <li>• Secondary conferencing (C)</li> <li>• Address translation (C)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6.1</li> <li>• GSCR Sect. 2.6.2</li> <li>• GSCR Sect. 2.6.3</li> <li>• GSCR Sect. 2.6.4</li> <li>• GSCR Sect. 2.6.5</li> <li>• GSCR Sect. 2.7</li> </ul>

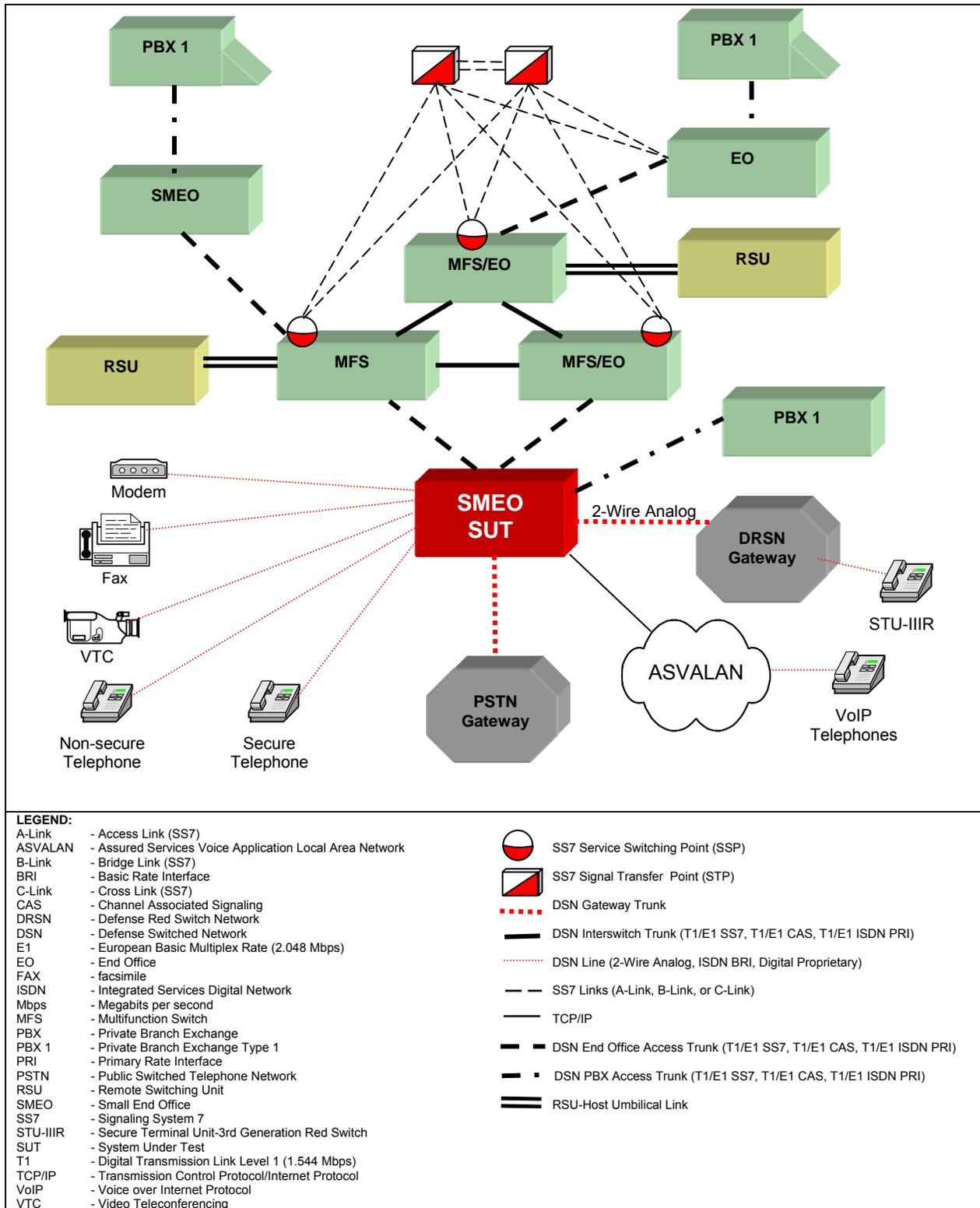
**Table 2-1. SMEO Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Between any two like terminations (C)</li> <li>• PCM-24 and PCM-30, both CAS and CCS (C)</li> <li>• Supervision passed end-to-end for A/D or D/A (C)</li> <li>• Monitored and auto reconfigure (C)</li> <li>• Support at least 10% of circuits as nailed-up (C)</li> <li>• Non-preemptable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.8</li> </ul>
PAT	No	<ul style="list-style-type: none"> <li>• Classmark for/not for PAT screening (C)</li> <li>• 7 PAT mechanisms (C)</li> <li>• Outgoing call screening (C)</li> <li>• Functional structure (C)</li> <li>• Simultaneous calls limitation (C)</li> <li>• Overflow process (C)</li> <li>• Decrementing call-in-progress count (C)</li> <li>• Call treatment (C)</li> <li>• Queuing (C)</li> <li>• Attendant calls (C)</li> <li>• Operations measurement registers (C)</li> <li>• Maintenance and Administration of thresholds (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1.1</li> <li>• GSCR Sect. 2.11.1.2</li> <li>• GSCR Sect. 2.11.1.3</li> <li>• GSCR Sect. 2.11.1.4</li> <li>• GSCR Sect. 2.11.1.5</li> <li>• GSCR Sect. 2.11.1.6</li> <li>• GSCR Sect. 2.11.1.7</li> <li>• GSCR Sect. 2.11.1.8</li> <li>• GSCR Sect. 2.11.1.9</li> <li>• GSCR Sect. 2.11.1.10</li> </ul>
DSN Hotline Services	Yes	<ul style="list-style-type: none"> <li>• Hotline restrictions (R)</li> <li>• Auto initiate (R)</li> <li>• Analog and digital (R)</li> <li>• Subscription basis (R)</li> <li>• Protected hotline calling (R)</li> <li>• WWNDP interoperable (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12.1-4</li> <li>• GSCR Sect. 2.12.5</li> </ul>
Network Management	Yes	<ul style="list-style-type: none"> <li>• Interfaces (R)</li> <li>• Measurements and data generation (R)</li> <li>• Fault management (R)</li> <li>• Configuration management (R)</li> <li>• Accounting management (R)</li> <li>• Performance management (R)</li> <li>• NM controls (R)</li> <li>• Remote access (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 9.1</li> <li>• GSCR Sect. 9.2</li> <li>• GSCR Sect. 9.3</li> <li>• GSCR Sect. 9.4</li> <li>• GSCR Sect. 9.5</li> <li>• GSCR Sect. 9.6</li> <li>• GSCR Sect. 9.7</li> <li>• GSCR Sect. 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• EKTS (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 10, table 10-3</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 11.1.1.2</li> <li>• GSCR Sect. 11.1.2.2</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• GR-512-CORE (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 12</li> </ul>
Security <sup>1</sup>	Yes	<ul style="list-style-type: none"> <li>• DITSCAP (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 13</li> </ul>
<b>VoIP</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
VoIP System	No	<p>VoIP function is conditional. If VoIP is provided, <b>all</b> of the following requirements must be met:</p> <ul style="list-style-type: none"> <li>• Voice Quality with Mean Opinion Score of 4.0 or better</li> <li>• Class of Service (CoS) and Quality of Service (QoS)</li> <li>• ITU-T G.711 PCM Codec</li> <li>• Traffic Engineering</li> <li>• Security in accordance with DITSCAP</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 ms</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> <li>• GSCR App. 3, paragraph 1.7</li> </ul>

**Table 2-1. SMEO Requirements (continued)**

Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN <sup>2</sup>	Yes	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (R)</li> <li>• On-Netting (R)</li> <li>• Off-Netting (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> </ul>
DRSN <sup>3</sup>	Yes	Access	<ul style="list-style-type: none"> <li>• Alerting Signals and Tones (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• Analog busy/idle (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
		Voice	<ul style="list-style-type: none"> <li>• MOS (C)</li> <li>• MLPP (C)</li> <li>• Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
<b>LEGEND:</b> 2W - 2-Wire 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 CJCS I - Chairman of the Joint Chiefs of Staff Instruction D/A - Digital to Analog Conversion DISA - Defense Information Systems Agency DISR - DoD IT Standards Registry DITSCAP - DoD IT Security and Accreditation Process DN - Directory Number DoD - Department of Defense DP - Dial Pulse DSN - Defense Switched Network DRSN - Defense Red Switch Network DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) E911 - Emergency 911 Service EKTS - Electronic Key Telephone System EIA - Electronic Industries Alliance G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement (Telcordia) GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IEEE - Institute of Electrical and Electronics Engineers, Inc. IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union - Telecommunication Standardization Sector LAN - Local Area Network kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score ms - milliseconds NI 1/2 - National ISDN Standard 1or 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 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.735.3 - SS7 Signaling Standard for E1 MLPP Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required Sect. - Section SMEO - Small End Office SS7 - Signaling System 7 STE - Secure Terminal Equipment STU-III - Secure Telephone Unit - 3 <sup>rd</sup> Generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-465A - Group 3 Facsimile Apparatus for Document Transmission VBD - Variable bit data VLAN - Virtual LAN VoIP - Voice over Internet Protocol VTC - Video Teleconferencing WWNDP - Worldwide Numbering and Dialing Plan				
<b>NOTES:</b> 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.				

**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 SUT was tested as the end-point in relation to the other switches. Figure 2-3 depicts the VoIP configuration.



**Figure 2-2. Notional Test Configuration**

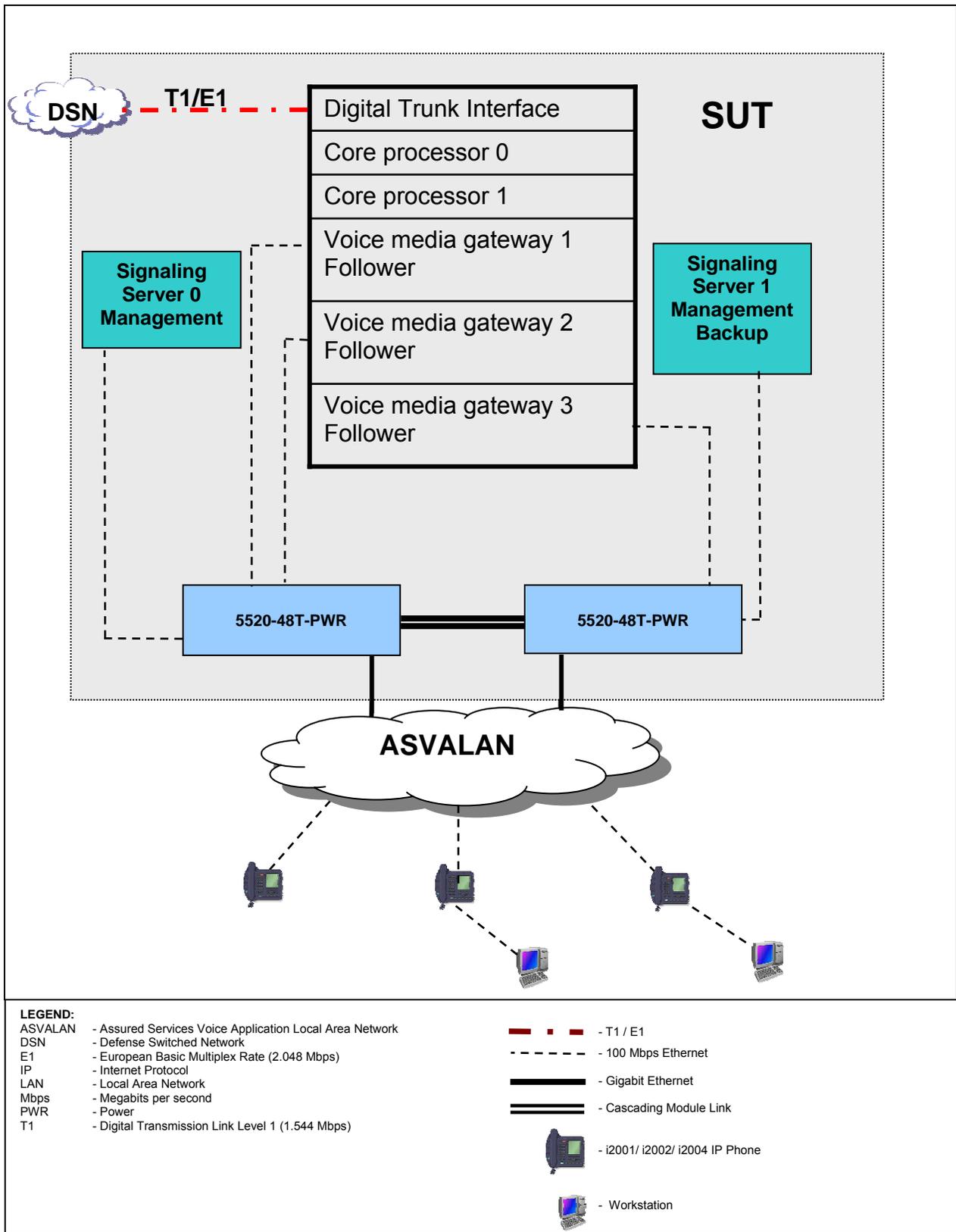


Figure 2-3. SUT IP Test Configuration with ASVALAN

**9. SYSTEM CONFIGURATIONS.** Table 2-2 provides the system configurations, hardware and software components tested with the SUT. Table 2-3 provides the Product Enhancement Packages that were installed.

**Table 2-2. Tested System Configurations**

<b>SUT Components</b>		
<b>Shelf IPE 20 0 0</b>		
<b>Part Number</b>	<b>Part Description</b>	<b>Part Details</b>
NT6D40BA	Peripheral Equipment Power Supply	Release 2
NT6D42CD	Ring Generator	Release 2
NT8D14BB	Universal Trunk Card	Release 13
NT8D02GA	Digital Line Card	Release 7
NT8D09BA	Analog Line Card/Message Waiting	Release 3
NT6071AA	Universal Interface Line Card	Release 9
NT6071AA	Universal Interface Line Card	Release 9
NTRB18CA	Media Gateway Card	Release 4
NTVQ01BA	Voice Gateway Media Card	Release 5
NT8D01BC	Controller 4 Card	Release 12
NTRB18CA	Media Gateway Card	Release 4
NT8D16AB	Digital Tone Receivers	Release 4
NTVQ01BA	Voice Gateway Media Card	Release 5
<b>Core/Net Module 0 1 1</b>		
NT6D41AD	Common Equipment Power Supply	Release 4
NT8D17FA	Conference Tone And Digit Switch	Release 17
QPC414C	Enhanced Network Card	N/A
NT8D04BA	Network Card	Release 6
NT6D80AB	Multipurpose Serial Data Link	Release 17
NT5D12AH	Dual PRI Card	BOOT=GBOOT_CR.04 LOAD=DDPCR15+p
NT5D12AH	Dual PRI Card	BOOT=GBOOT_CR.04 LOAD=DDPCR15+p
NT5D97AD	Dual Port DTI/PRI Card	Release 7, Firmware: fddp2cr22+
QPC775F	Clock Controller	N/A
QPC43R	Peripheral Signaling	N/A
QPC441F	3-Port Extender	N/A
NTRB34AA	Core Network Interface	Release 3
NT5D03FB	Core Processor	CP 68060E/128 64Flash/64 DRAM
NT5D61AB	Input/Output Disk Unit	N/A
<b>Core/Net Module 0 0 0</b>		
NT6D41AD	Common Equipment Power Supply	Release 4
NT8D17FA	Conference Tone And Digit Switch	Release 17
NT8D04BA	Network Card	Release 6
NT5D12AH	Dual PRI Card	BOOT=GBOOT_CR.04 LOAD=DDPCR15+p
NT5D12AH	Dual PRI Card	BOOT=GBOOT_CR.04 LOAD=DDPCR15+p
NT6D73AA	Multipurpose ISDN Signaling Processor	N/A
NT5D12AH	Dual PRI Card	BOOT=GBOOT_CR.04 LOAD=DDPCR15+p
NT6D80AB	Multipurpose Serial Data Link	Release 1
QPC775F	Clock Controller	N/A
QPC441F	Peripheral Signaling	N/A
QPC441F	3-Port Extender	N/A
NTRB34AA	Core Network Interface	Release 3
NT5D03FB	Core Processor	CP 68060E/128 64 Flash/64 DRAM
NT5D61AB	Input/Output Disk Unit	N/A

**Table 2-2. Tested System Configurations (Continued)**

IPE Shelf 4 0 0		
NT6D40BA	Peripheral Equipment Power Supply	Release 2
NT6D42CD	Ring Generator	Release 2
NT8D02GA	Digital Line Card	Release 7
NT8D09BA	Analog Line Card/Message Waiting	Release 4
NTRB18CA	Media Gateway Card	Release 4
NT6D70AA	S/T Interface Line Card	Release 13
NT8D01BC	Controller 4 Card	Release 12
NT8D15AK	E&M Trunk Card	Release 10
NT8D09BA	Analog Line Card	Release 5
NT8D16AB	Digitone Receiver Card	Release 4
NT8D02GA	Digital Line Card	Release 7
NTVQ01BA	Voice Gateway Media Card	Release 5
VoIP & Secure Voice Zone		
Device Name	Software Release	
Baystack 5520-48T-PWR	Firmware:4.2.0.11 / Software:4.2.0.005	
Contivity	5.11	
Signaling Server	IPL 4.5.25	
Voice Gateway Media Card	Load IPL 4.5.25 / Firmware SMFW 6.7	
DSN Switches		
System Name	Software Release	
Nortel MSL-100 (MFS, EO)	SE08	
Avaya S8710 (SMEO, PBX 1, PBX 2)	CM 3.0 (R013x.00.0.340.3)	
Siemens EWSD (MFS, EO)	19d with Patch Set 46	
Lucent 5ESS (MFS, EO)	5E16.2 SU 06.0002	
Telephone Instruments		
Interface Type	Model (s)/ Release	
2-Wire Analog	Panasonic KX-TS15-W	
2-Wire Analog	Nortel 8314	
ISDN BRI	Nortel M5317T	
ISDN BRI	Tone Commander 6210U, 6210T, 6220U, 6220T, and 6220T TSG with Firmware 01.06.12. Includes 6030X Expansion Module with Firmware 01.01.03	
2-Wire Proprietary Digital	Nortel M2616, M3901, M3902, M3903, and M3904	
VoIP	i2001 NTDU90 / Firmware: 3.99 (0604D99)	
VoIP	i2002 NTDU91 / Firmware: 3.99 (0604D99)	
VoIP	i2004 NTDU92 / Firmware: 3.99 (0604D99)	
<b>LEGEND:</b>		
5ESS - Class 5 Electronic Switching System	EWSD - Elektronisches Wählsystem Digital	PRI - Primary Rate Interface
BRI - Basic Rate Interface	IPE - Intelligent Peripheral Equipment	PWR - Power over Ethernet
CM - Communication Manager	ISDN - Integrated Services Digital Network	SE - Succession Enterprise
CP - Core Processor	Mbps - Megabits per second	SMEO - Small End Office
DRAM - Dynamic Random Access Memory	MFS - Multifunction Switch	S/T - ISDN BRI 4-Wire interface
DSN - Defense Switched Network	MSL - Meridian Switching Load	SU - Software Update
E&M - Ear and Mouth	N/A - Not Applicable	SUT - System Under Test
E1 - European Basic Multiplex Rate (2.048 Mbps)	PBX 1 - Private Branch Exchange 1	T - ISDN BRI 4-Wire Interface
EO - End Office	PBX 2 - Private Branch Exchange 2	VoIP - Voice over Internet Protocol

**Table 2-3. Specified Product Enhancement Packages**

<b>CORE Software Patch Groups</b>					
<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>
p20702_1	5 August 2005	p20166_2	28 February 2006	p21302_1	9 January 2006
p20990_2	20 September 2005	p20335_1	2 November 2005	p21318_1	28 November 2005
p20906_1	6 September 2005	p20506_2	14 March 2006	p21319_2	12 December 2005
p18870_1	8 September 2005	p20556_1	24 January 2006	p21321_1	16 December 2005
p19899_1	1 September 2005	p20588_1	1 September 2005	p21345_1	1 February 2006
p20020_1	1 September 2005	p20589_1	8 November 2005	p21383_1	8 December 2005
p20620_1	1 September 2005	p20644_1	23 November 2005	p21394_1	20 January 2006
p19908_1	1 September 2005	p20676_1	1 September 2005	p21442_3	25 January 2006
p20418_1	8 September 2005	p20703_1	27 November 2005	p21452_1	16 March 2006
p20436_1	8 September 2005	p20720_1	1 September 2005	p21466_1	10 January 2006
p20616_2	15 November 2005	p20734_1	31 January 2006	p21469_1	17 January 2006
p20642_3	10 August 2005	p20758_1	2 November 2005	p21473_1	21 February 2006
p20659_1	12 August 2005	p20835_1	25 August 2005	p21478_2	2 January 2006
p20764_1	11 August 2005	p20839_1	26 October 2005	p21533_3	28 February 2006
p20708_1	11 August 2005	p20845_2	21 September 2005	p21618_2	15 March 2006
p20765_1	17 August 2005	p20880_1	18 January 2006	p21631_1	22 March 2006
p20811_1	28 September 2005	p20893_1	28 October 2005	p21676_2	11 April 2006
p20441_1	16 November 2005	p20911_1	20 October 2005	p21716_1	27 February 2006
p21451_1	15 December 2005	p20930_1	14 September 2005	p21755_1	9 March 2006
p21354_1	2 December 2005	p20931_1	15 November 2005	p21758_2	21 April 2006
p21440_2	21 March 2006	p20948_5	8 December 2005	p20223_2	10 May 2006
p20419_3	23 March 2006	p20981_3	18 October 2005	p21448_3	25 May 2006
p21450_1	9 November 2005	p20986_1	19 September 2005	p22095_1	26 May 2006
p19832_2	11 January 2006	p21011_2	17 January 2006	p19852_1	16 November 2005
p21794_1	20 March 2006	p21015_1	23 January 2006	pbri	1 June 2006
p21795_1	27 March 2006	p21029_1	24 January 2006	p21573_1	11 April 2006
p21802_1	17 March 2006	p21036_1	31 October 2005	p21449_2	23 June 2006
p21803_1	17 March 2006	p21038_1	28 September 2005	p22095	26 May 2006
p21873_1	3 April 2006	p21048_1	30 September 2005	p22237_1	30 June 2006
p18240_1	1 September 2005	p21052_2	27 January 2006	p22238_1	2 June 2006
p19237_2	16 March 2006	p21107_2	27 January 2006	p22239_1	2 June 2006
p19964_1	20 January 2006	p21115_3	11 November 2005	p22513_1	23 August 2006
p20025_1	2 November 2005	p21116_1	9 December 2005		
p20105_1	20 January 2006	p21150_1	25 October 2005		
<b>Signaling Server Software Patch Groups</b>					
<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>
p21474_1.ss1	25 July 2006	p21112_1.ss1	25 July 2006	p20990_2.ss1	25 July 2006
p21553_1.ss1	25 July 2006	p21090_1.ss1	25 July 2006	p20848_1.ss1	25 July 2006
p21883_1.ss1	25 July 2006	p21083_1.ss1	25 July 2006	p20805_1.ss1	25 July 2006
p21861_1.ss1	25 July 2006	p21075_1.ss1	25 July 2006	p20781_1.ss1	25 July 2006
p21652_1.ss1	25 July 2006	p21044_1.ss1	25 July 2006	p20754_2.ss1	25 July 2006
p21641_1.ss1	25 July 2006	p21017_1.ss1	25 July 2006	p20746_1.ss1	25 July 2006
p21571_1.ss1	25 July 2006	p20999_1.ss1	25 July 2006	p20737_1.ss1	25 July 2006
p21564_1.ss1	25 July 2006	p20987_1.ss1	25 July 2006	p20736_1.ss1	25 July 2006
p21562_1.ss1	25 July 2006	p20962_1.ss1	25 July 2006	p20704_1.ss1	25 July 2006
p21544_1.ss1	25 July 2006	p20938_3.ss1	25 July 2006	p20498_3.ss1	25 July 2006
p21315_1.ss1	25 July 2006	p20897_1.ss1	25 July 2006	p20257_1.ss1	25 July 2006
p21290_1.ss1	25 July 2006	p20884_1.ss1	25 July 2006	p19322_1.ss1	25 July 2006
p21178_1.ss1	25 July 2006	p20882_1.ss1	25 July 2006	p18859_1.ss1	25 July 2006
p21207_1.ss1	25 July 2006	p20876_1.ss1	25 July 2006	p20732_1.ss1	25 July 2006
p21172_1.ss1	25 July 2006	p20853_1.ss1	25 July 2006	p20618_1.ss1	25 July 2006
<b>Voice Media Gateway Card Software Patch Groups</b>					
<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>	<b>Patch Name</b>	<b>Date Created</b>
p21534_2.lsa	25 July 2006	p21083_1.lsa	25 July 2006	p20962_1.lsa	25 July 2006
p21512_1.lsa	25 July 2006	p21030_1.lsa	25 July 2006	p20889_1.lsa	25 July 2006
p21112_1.lsa	25 July 2006	p21017_1.lsa	25 July 2006	p20781_1.lsa	25 July 2006
p21090_1.lsa	25 July 2006	p20990_2.lsa	25 July 2006	p18859_1.lsa	25 July 2006

**10. TESTING LIMITATIONS.** None.

## **11. TEST RESULTS**

### **a. Discussion**

**(1) DSN Trunk Interfaces.** The SUT met all critical interoperability certification requirements for DSN Trunk Interfaces with the exceptions noted in the following subparagraphs. The overall operational impact of these discrepancies is minor. Detailed trunk configurations and associated lessons learned can be found on the TSSI web page: [http://jitc.fhu.disa.mil/tssi/cert\\_nortel](http://jitc.fhu.disa.mil/tssi/cert_nortel).

**(a)** When any active trunk interface is physically broken and repaired, the SUT does not restore the span to service and remove the yellow alarm condition within the required time duration. In accordance with the GSCR paragraphs 7.1.4 and 7.2.2, the time required for the removal of the alarm condition after the physical restoration of a broken trunk is 15 (+/-5) seconds. The E1 CAS can take up to 90 seconds to restore, and all the other interfaces require 30 seconds to be restored. The operational impact is minor since the alarm clears automatically when the span is returned to service.

**(b)** T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 milliseconds (ms) to 350ms. The SUT recognizes wink start signals from 100ms to 925ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290ms this anomaly has no operational impact.

**(c)** The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since the SUT is a subtending switch off of a Multifunction Switch (MFS) and all MFS support glare hold, complementing the SUTs capability to support glare release, the operational impact is minor.

**(d)** If a T1 ISDN PRI interface is broken then restored when all channels are in a maintenance busy condition, the SUT fails to automatically return the channels to the previous busy condition. This anomaly has no operational impact because it only occurs when the SUT is in a maintenance condition.

**(e)** The on/off hook pulse that initiates the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). The pulse width was measured to be greater than 100 ms (highest @ 128 ms) about 20% of the time, but never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.

**(2) DSN Line Interfaces.** The SUT met all critical interoperability certification requirements for DSN Line Interfaces with the exceptions noted in the following

subparagraphs. Refer to table 2-2 for specific instrument models tested under this certification test. The overall operational impact of these discrepancies is minor.

(a) The SUT does not support an National ISDN (NI)2 BRI interface. The only supported and certified BRI interface is NI1. The NI2 BRI interface is required for SMEO operation as specified by GSCR paragraph 2.3.3. Since the primary differences between NI1 and NI2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future, this anomaly has a minor operational impact.

(b) The SUT will only support a BRI NI1 voice line with a single directory number and a single appearance of a directory number. However, multiple appearances with different directory numbers can be supported with the digital proprietary instruments which account for the majority of digital instruments fielded within the DSN. Therefore, the operational impact is minor.

(c) The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in the GSCR 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.

**(3) Features and Functions.** The SUT met all critical interoperability certification requirements for Features and Capabilities with the exceptions noted in the following subparagraphs. The overall operational impact of these discrepancies is minor.

(a) The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.

(b) When call forwarding variable (CFV) is assigned to any station on the SUT (except BRI; does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or Public Switched Telephone Network (PSTN). Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. Per the GSCR only ROUTINE precedence calls will be forwarded and precedence calls above are diverted to the attendant console, night service or alternate directory number. Therefore, this feature is not certified for use within the DSN. This feature is a conditional requirement and will have a minor operational impact.

(c) The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by the GSCR. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. The operational impact is minor.

**(d)** The SUT supports the "call waiting" indication on VoIP telephones with visual indicators in lieu of audible tones as specified by the GSCR. When call waiting is invoked on a VoIP phone, the phone displays call waiting text along with a flashing symbol. The call waiting symbol flashes twice for a ROUTINE call and three times for precedence above ROUTINE call. Since the requirement for audible tone is conditional, and there are two visual indicators to alert the VoIP user of a waiting call, the operational impact of not supporting audible tones is minor.

**(e)** The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in the GSCR. Since the SUT is predominately fielded within the DSN as a SMEO with no tandeming (e.g. subtending PBX1 or PBX2), this anomaly has a minor operational impact.

**(f)** Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR 2.2.4. The proper override tone, however, is given to a station active with a call prior to the attendant's bridging into the active call. Since attendants rarely bridge into calls and active calls remain connected when an attendant does bridge into a call, the operational impact is minor.

**(g)** The SUT will not permit an ISDN BRI station to be a member of a multiline hunt group. All other phone types can be configured as members of a multiline hunt group. Since ISDN BRI voice users are rarely used within the DSN and this feature can be accomplished on the SUT with analog and digital proprietary stations, this anomaly has a minor operational impact.

**(4) Network Gateways.** The SUT met all critical interoperability certification requirements for the PSTN and Defense Red Switch Network Gateways with no exceptions.

**(5) VoIP.** The SUT is certified with any certified ASVALAN listed on the Approved Products List located on the TSSI web page: <http://jitic.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, section A3.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with Department of Defense Information Technology Standards Registry (DISR) voice quality standards. This applies from handset to handset and from handset to gateway trunk in the DSN. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.20 and never measured below 4.0. The average inter-switch MOS was 4.25 with a minimum measured MOS value of 4.17. This average was based on a total of 200 calls.

**2. 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. The 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice Virtual Local Area Network 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 injected on the certified ASVALAN to insure that all CoS and QoS settings were working properly. Packet captures indicated all tags were set properly.

**3. Codec.** In accordance with the GSCR, appendix 3, section A3.2.2, the International Telecommunication Union - Telecommunication Standardization Sector (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 Nortel Internet Protocol (IP) phones that met the critical interoperability requirements for certification were the i2001, i2002 and i2004 IP phones. Shared access (i.e., same switch port is shared by Personal Computer [PC] and IP phone), was tested with this configuration. The IP phones were connected to the 100 Megabits per second (Mbps) full duplex access switch via the 10/100 SW port. Data was connected to the 100 Mbps PC port on the back of the phones with Ethernet, which were configured for 100 Mbps full duplex. In this configuration there was no degradation of voice quality, therefore this system is certified for shared access with the data port configured for 100 Mbps.

**b. Scalability.** The SUT Voice Media Gateway Card only supports 32 DSP's for IP to Time Division Multiplexing (TDM) calls. Table 2-4 shows the maximum number of telephony subscribers supported by each processor. The SUT was tested with the Motorola CP4 call processor. The Motorola CP3, Pentium II and Pentium IV processors utilize the same software and product enhancement packages as the Motorola CP4. JITC analysis determined the Motorola CP3, Pentium II and Pentium IV processors to be functionally identical to the SUT for interoperability certification purposes. 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.

**Table 2-4. SUT Telephony Capacity**

Call Server	Platform Name	Total Number of Phone Connections Supported			
		TDM Only	IP phone with access to PSTN	IP Phones (no access to PSTN)	Mixed TDM and IP Phones <sup>1</sup>
CP3 <sup>2</sup> / CP4 <sup>2</sup>	CS1000M-SG	2000	2000	3000	1000 TDM/1000 IP
CP3 <sup>2</sup> / CP4 <sup>2</sup>	CS1000M-MG	4000	3000	3000	2500 TDM/1000 IP
CP PII, CP PIV	CS1000M-SG	2000	3000	5000	1000 TDM/2000 IP
CP PII, CP PIV	CS1000M-MG	16000	15000	15000	8000 TDM/5000 IP, 10000 TDM/4000 IP, 12000 TDM/3000 IP
<b>LEGEND:</b> CP - Call Processor CS - Communication Server IP - Internet Protocol MG - Multi Group P - Pentium PSTN - Public Switched Telephone Network SG - Single Group SUT - System Under Test TDM - Time Division Multiplexing					
<b>NOTES:</b> 1 Nortel Engineering Configurator (NNEC) must be used to determine the IP/TDM phone ratio. 2 CP3 and CP4 are Motorola Processors					

**5. Multi-Level Precedence and Preemption (MLPP).** The GSCR, section 3, details the requirements for MLPP. All critical MLPP features and functions were met by the SUT.

**(b) Security.** Security requirements in accordance with the GSCR, appendix 3, are 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 requires that the vendor provide a management system to monitor the performance of the ASVALAN portion of the VoIP system. This requirement was verified via a LoC because of the numerous third party systems and applications capable of performing this function.

**(d) Synchronization.** Synchronization is required for 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 TDM-based interfaces.

**(e) 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 140 calls, with a minimum duration of 5 minutes for each call, was measured to be 53.4 ms.

**(f) 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 0.00%.

**(g) 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 their respective company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008:

1. Conformant with IPv6 standards profile contained in the DISR.
2. Maintaining interoperability in heterogeneous environments and with IPv4.
3. Commitment to upgrade as the IPv6 standard evolves.
4. Availability of contractor/vendor IPv6 technical support.

**b. System Interoperability Results.** The SUT is certified for joint use in the DSN as a SMEO with VoIP in accordance with the requirements set forth in the GSCR. The identified test discrepancies shown 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, DP)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration. <sup>1</sup> The SUT recognizes a wink start signal greater than the specified maximum limit. <sup>2</sup> The SUT does not support glare hold resolution for their CAS trunks. <sup>3</sup>
T1 CAS (MFR1)	No	Not Tested	This interface is not supported. <sup>4</sup>
E1 CAS (DTMF, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not restore the span to service within the required time duration. <sup>1</sup> The SUT does not support glare hold resolution for their CAS trunks. <sup>3</sup> The on/off hook pulse that frames the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). <sup>5</sup>
E1 CAS (MFR1)	No	Not Tested	This interface is not supported. <sup>4</sup>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT fails to automatically return trunks to a maintenance busy condition after the span is broken then restored. <sup>6</sup>
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Certified	Met all critical CRs and FRs.
T1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. <sup>4</sup>
E1 SS7 (ANSI T1.619a)	No	Not Tested	This interface is not supported. <sup>4</sup>

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

<b>DSN Line Interfaces</b>				
<b>Interface &amp; Signaling</b>		<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Analog (GR-506-CORE)		Yes	Certified	Met all critical CRs and FRs.
ISDN BRI NI 1/2		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support NI2 BRI. <sup>7</sup> The only supported and certified interface is NI1 BRI with a single appearance of a single directory number. <sup>8</sup> The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications. <sup>9</sup> The BRI instruments do not support precedence call waiting. <sup>10</sup>
2-Wire Proprietary Digital		No	Certified	Met all critical CRs and FRs.
VoIP (ITU-T H.323 Proprietary)		No	Certified	Met all critical CRs and FRs. Precedence call waiting indication is unique on VoIP phones. <sup>11</sup>
<b>DSN Features and Capabilities</b>				
<b>Features and Capabilities</b>		<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
Common Features		No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not correctly support the call forwarding variable feature. <sup>12</sup> The conference disconnect tone that is provided by the SUT does not meet the specifications. <sup>13</sup>
Attendant		No	Certified	Met all critical CRs and FRs with the following minor exceptions: Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call. <sup>14</sup>
Public Safety		Yes	Certified	Met all critical CRs and FRs with the following exception: The SUT cannot perform a tandem call trace of a specified distant office directory number. <sup>15</sup>
Preset Conferencing		No	Not Tested	This feature is not supported. <sup>16</sup>
Nailed-up Connections		No	Not Tested	This feature is not supported. <sup>16</sup>
PAT		No	Not Tested	This feature is not supported. <sup>16</sup>
DSN Hotline Services		Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support a protected hotline specified list. <sup>17</sup>
Network Management		Yes	Certified	Met all critical CRs and FRs.
Multiline Hunt Service		No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT will not permit a BRI station to be a member of a multiline hunt group. <sup>18</sup>
ISDN Services (EKTS)		No	Not Tested	This feature is not supported. <sup>16</sup>
Synchronization		Yes	Certified	Met all critical CRs and FRs.
Reliability		Yes	Certified	Met all critical CRs and FRs.
Security		Yes	See note 19	See note 19.
VoIP System		No	Certified	The SUT is certified for VoIP specifically with certified ASVALAN posted on the JITC TSSI program web page ( <a href="http://jitc.fhu.disa.mil/tssi/cert_nortel">http://jitc.fhu.disa.mil/tssi/cert_nortel</a> ) approved product list. See note 20.
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all critical CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.

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

Network Gateways (continued)				
Gateway	Interface & Signaling	Critical	Status	Remarks
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>21</sup>	Met all critical CRs and FRs.
<b>LEGEND:</b>				
ANSI	- American National Standards Institute	GSCR	- Generic Switching Center Requirements	PM - Program Manager
ASVALAN	- Assured Services Voice Application Local Area Network	H.323	- Standard for multi-media communications on packet-based networks	PRI - Primary Rate Interface PSTN - Public Switched Telephone Network
BRI	- Basic Rate Interface	IPv4	- Internet Protocol version 4	Q.931 - Signaling Standard for ISDN
CAS	- Channel Associated Signaling	IPv6	- Internet Protocol version 6	Q.955.3 - ISDN signaling standard for E1 MLPP
CRs	- Capability Requirements	ISDN	- Integrated Services Digital Network	SMEO - Small End Office
DISA	- Defense Information Systems Agency	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	SS7 - Signaling System 7
DoD	- Department of Defense	JITC	- Joint Interoperability Test Command	SUT - System Under Test
DP	- Dial Pulse	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements	T1 - Digital Transmission Link Level 1 (1.544 Mbps)
DRSN	- Defense Red Switch Network	Mbps	- Megabits per second	T1.607 - ISDN - Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DSN	- Defense Switched Network	MFR1	- Multifrequency Recommendation 1	T1.619a - SS7 and ISDN MLPP Signaling Standard for T1
DSS1	- Digital Subscriber Signaling 1	MLPP	- Multi-Level Precedence and Preemption	TSSI - Telecom Switched Services Interoperability
DTMF	- Dual Tone Multi-Frequency	ms	- millisecond	TPC - Twisted Pair Copper
E1	- European Basic Multiplex Rate (2.048 Mbps)	NI 1/2	- National ISDN 1 or 2	VoIP - Voice over Internet Protocol
EKTS	- Electronic Key Telephone System	PAT	- Precedence Access Threshold	
FRs	- Feature Requirements	PBX	- Private Branch Exchange	
GR	- Generic Requirement			
GR-506	- LSSGR: Signaling for Analog Interfaces			
<b>NOTES:</b>				
<p>1 When any active trunk interface is physically broken and repaired, the SUT does not restore the span to service and remove the yellow alarm condition within the required time duration. In accordance with the GSCR paragraphs 7.1.4 and 7.2.2, the time required for the removal of the alarm condition after the physical restoration of a broken trunk is 15 (+/-5) seconds. The E1 CAS can take up to 90 seconds to restore, and all the other interfaces require 30 seconds to be restored. The operational impact is minor since the alarm clears automatically when the span is returned to service.</p> <p>2 T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 milliseconds (ms) to 350ms. The SUT recognizes wink start signals from 100ms to 925ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290ms this anomaly has no operational impact.</p> <p>3 The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since the SUT is a subtending switch off of a Multifunction Switch (MFS) and all MFS support glare hold, complementing the SUTs capability to support glare release. Therefore, the operational impact is minor.</p> <p>4 This interface is not supported. There is no operational impact because it is not a critical requirement.</p> <p>5 The on/off hook pulse that initiates the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100ms (+/-5ms). The pulse width was measured to be greater than 100 ms (highest @ 128 ms) about 20% of the time, but never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.</p> <p>6 If a T1 ISDN PRI interface is broken then restored when all channels are in a maintenance busy condition, the SUT fails to automatically return the channels to the previous busy condition. This anomaly has no operational impact because it only occurs when the SUT is in a maintenance condition.</p> <p>7 The SUT does not support a National ISDN (NI)2 BRI interface. The only supported and certified BRI interface is NI1. The NI2 BRI interface is required for SMEO operation as specified by GSCR paragraph 2.3.3. Since the primary differences between NI1 and NI2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future, this anomaly has a minor operational impact.</p> <p>8 The SUT will only support a BRI NI1 voice line with a single directory number and a single appearance of a directory number. However, multiple appearances with different directory numbers can be supported with the digital proprietary instruments which account for the majority of digital instruments fielded within the DSN. Therefore, the operational impact is minor.</p> <p>9 The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in the GSCR 5.5.1. The precedence above ROUTINE cadence is distinct from the ROUTINE cadence when it is configured properly; therefore this anomaly has no operational impact.</p> <p>10 The SUT does not support precedence call waiting for their BRI instruments; however the SUT does support precedence call waiting for all other phone types. Also, this requirement is conditional and therefore, has no operational impact.</p> <p>11 The SUT supports the "call waiting" indication on VoIP telephones with visual indicators in lieu of audible tones as specified by the GSCR. When call waiting is invoked on a VoIP phone, the phone displays call waiting text along with a flashing symbol. The call waiting symbol flashes twice for a ROUTINE call and three times for precedence above ROUTINE call. Since the requirement for audible tone is conditional, and there are two visual indicators to alert the VoIP user of a waiting call, the operational impact of not supporting audible tones is minor.</p> <p>12 When call forwarding variable (CFV) is assigned to any station on the SUT (except BRI; does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or Public Switched Telephone Network (PSTN). Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. Per the GSCR only ROUTINE precedence calls will be forwarded and precedence calls above are diverted to the attendant console, night service or alternate directory number, therefore this feature is not certified for use within the DSN. This feature is a conditional requirement and will have a minor operational impact.</p> <p>13 The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.</p> <p>14 Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR 2.2.4. The proper override tone, however, is given to a station active with a call prior to the attendant's bridging into the active call. Since attendants rarely bridge into calls and active calls remain connected when an attendant does bridge into a call, the operational impact is minor.</p> <p>15 The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in the GSCR. Since the SUT is predominately fielded within the DSN as a SMEO with no tandeming (e.g. subtending PBX1 or PBX2), this anomaly has a minor operational impact.</p> <p>16 This feature is not supported. There is no operational impact because it is not a critical requirement.</p> <p>17 The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by the GSCR. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. The operational impact is minor.</p> <p>18 The SUT will not permit an ISDN BRI station to be a member of a multiline hunt group. All other phone types can be configured as members of a multiline hunt group. Since ISDN BRI voice users are rarely used within the DSN and this feature can be accomplished on the SUT with analog and digital proprietary stations, this anomaly has a minor operational impact.</p> <p>19 Security is tested by DISA-led Information Assurance test teams and published in a separate report.</p> <p>20 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 their respective company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008:</p> <ol style="list-style-type: none"> <li>Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DISR).</li> <li>Maintaining interoperability in heterogeneous environments and with IPv4.</li> <li>Commitment to upgrade as the IPv6 standard evolves.</li> <li>Availability of contractor/vendor IPv6 technical support.</li> </ol> <p>21 Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.</p>				