



## DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND

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IN REPLY  
REFER TO:

Battlespace Communications Portfolio (JTE)

7 March 2007

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M Cabinet and CS1000M Chassis (including Voice over Internet Protocol [VoIP]) and DSN Option 11C 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 Cabinet 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 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://jitc.fhu.disa.mil/tssi/cert/cert\\_asvalan.html](http://jitc.fhu.disa.mil/tssi/cert/cert_asvalan.html)) approved product list. One of the optional requirements, Call Forwarding Variable processing for precedence calls, did not meet the specifications and therefore is not certified for use in the DSN. The interoperability test summary and the rest of the exceptions that were identified during testing are listed in table 1. The DSN CS1000M Chassis employs the same software and trunk/line card hardware as the Nortel DSN CS1000M Cabinet. Analysis by JITC determined that the DSN CS1000M Chassis is functionally identical to the DSN CS1000M Cabinet for interoperability certification purposes, and it is also certified for joint use within the DSN. The DSN CS1000M Cabinet and DSN CS1000M Chassis without VoIP are referred to and marketed within Department of Defense (DoD) as the Nortel DSN 11C Cabinet and DSN 11C Chassis, respectively. Except for the absence of VoIP capability, these are functionally identical to CS1000M models for interoperability certification purposes, and are also certified for joint use within the DSN. The listed test discrepancies shown in the Certification Testing Summary (enclosure 2), have an overall minor operational impact. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Private Branch Exchange (PBX) 1, and

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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 25 July through 1 September 2006. Patches were applied and regression testing was conducted from 27 November through 18 December 2006. Review of the vendor's LoC was completed on 29 January 2007. 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 PBX 1 required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in table 2. PBX 2 requirements are a subset of the requirements 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. PBX 1 interface and signaling requirements for trunks/lines specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 CRs/FRs specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- d. 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).

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**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 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 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 100 ms (+/-) 5 ms. <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 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 CRs and FRs.
<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 CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not support NI 2 BRI. <sup>7</sup> The only supported and certified interface is NI 1 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 CRs and FRs.
VoIP (ITU-T H.323 with Proprietary Signaling Interface)	No	Certified	Met all CRs and FRs with the following minor exception: 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 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 CRs and FRs with the following minor exception: 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 CRs and FRs with the following minor 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>
Precedence Access Threshold	No	Not Tested	This feature is not supported. <sup>16</sup>
DSN Hotline Services	No	Certified	Met all CRs and FRs with the following minor exception: The SUT does not support a protected hotline specified list. <sup>17</sup>
Network Management	No	Certified	Met all CRs and FRs.
Multiline Hunt Service	No	Certified	Met all CRs and FRs with the following minor exception: 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>

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

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all 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/apl.html">http://jitc.fhu.disa.mil/tssi/apl.html</a> ) approved product list. See note 20.	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all CRs and FRs.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>21</sup>	Met all CRs and FRs.
<b>LEGEND:</b> ANSI - American National Standards Institute ASVALAN - Assured Services Voice Application Local Area Network BRI - Basic Rate Interface CAS - Channel Associated Signaling CFV - Call Forwarding Variable CRs - Capability Requirements DISA - Defense Information Systems Agency DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DSS1 - Digital Subscriber Signaling 1 DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements H.323 - Standard for multi-media communications on packet-based networks IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union - Telecommunication Standardization Sector JITC - Joint Interoperability Test Command LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption ms - millisecond NI 1 - National ISDN Standard 1 NI 2 - National ISDN Standard 2 NI 1/2 - National ISDN Standard 1 or 2 PBX 1 - Private Branch Exchange 1 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.931 - Signaling Standard for ISDN Q.955.3 - ISDN signaling standard for E1 MLPP SS7 - Signaling System 7 SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.607 - ISDN – Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1 T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TSSI - Telecom Switched Services Interoperability TPC - Twisted Pair Copper VoIP - Voice over Internet Protocol				

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

NOTES:	
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 interface 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 without manual intervention 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 paragraph 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 ms to 350 ms. The SUT recognizes wink start signals from 100 ms to 925 ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290 ms, this anomaly has no operational impact.
3	The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since glare resolution is conditional for a PBX 1, 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 masks the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100 ms (+/-) 5 ms. The pulse width was measured to be greater than 100 ms about 20 percent of the time with the highest at 128 ms. This 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, and the trunks can be returned to maintenance busy condition manually.
7	The SUT does not support an NI 2 BRI interface. The only supported and certified BRI interface is NI 1. The NI 2 BRI interface is not required for a PBX 1 as specified by GSCR paragraph 2.3.3. The primary differences between NI 1 and NI 2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future. Also, BRI is not required for a PBX 1. This anomaly has a minor operational impact.
8	The SUT will only support a BRI NI 1 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. Since BRI is not required for a PBX 1, 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 GSCR 5.5.1 paragraph. 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. Since this is a conditional requirement, there is 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 CFV is assigned to any station on the SUT (except BRI, which does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or PSTN. Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. In accordance with 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 paragraph 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 paragraph 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 GSCR paragraph 2.4.4. Since this is not required for a PBX 1, 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 GSCR paragraph 2.12. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. Since this feature is not required by a PBX 1 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. This anomaly has a minor operational impact.
19	Security is tested 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 the company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008: a. Conformance with IPv6 standards profile contained in the DoD IT Standards Registry (DISR). b. Maintaining interoperability in heterogeneous environments and with IPv4. c. Commitment to upgrade as the IPv6 standard evolves. d. Availability of contractor/vendor IPv6 technical support.
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.

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**Table 2. PBX 1 Requirements**

<b>DSN Trunk Interfaces</b>				
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
T1 CAS (MFRI, DTMF, DP)	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> </ul>	<ul style="list-style-type: none"> <li>GSCR Section 7</li> <li>GSCR Section 7</li> <li>GSCR Section 5</li> <li>GSCR Section 2.5.7, 7.1.4 &amp; 7.2.2</li> </ul>
E1 CAS (MFRI, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> <li>WWNDP (R)</li> <li>Outpulsing digit formats (C: CAS only)</li> <li>Routing (C)</li> <li>Trunk Groups (C)</li> <li>Call Processing (C)</li> <li>CAS to CCS trunk interworking (C)</li> <li>PCM-24/PCM-30 Interoperation (C)</li> <li>Direct Inward Dialing (C)</li> </ul>	<ul style="list-style-type: none"> <li>GSCR Section 4.5.1</li> <li>GSCR Section 4.5.2</li> <li>GSCR Section 4.2</li> <li>GSCR Section 2.5.5 &amp; 2.5.6</li> <li>GSCR Section 4</li> <li>GSCR Section 3.10</li> <li>GSCR Section 7.3</li> <li>GSCR Section 2.3.2</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	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 Section 3</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>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	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 Section 3.10</li> <li>GSCR Section 3.10</li> <li>GSCR Section 3.10</li> <li>GSCR Section 3.10</li> <li>CJCSI 6215.01B</li> </ul>
		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>
<b>DSN Line Interfaces</b>				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>Directory Number Identification (R)</li> <li>Line signaling (R)</li> <li>Loop Start Line (R: 2-Wire Analog only)</li> <li>Alerting Signals and Tones (R)</li> <li>WWNDP (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 Section 2.1.1</li> <li>GSCR Section 5.2</li> <li>GSCR Section 5.2.1</li> <li>GSCR Section 5.5</li> <li>GSCR Section 4.5</li> <li>GSCR Section 4.1</li> <li>GSCR Section 4.3.3</li> <li>GSCR Section 4.3.4.1</li> </ul>
ISDN BRI NI 1/2 (ANSI T1.619a)	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>
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>
		Data	<ul style="list-style-type: none"> <li>Modem (VBD) (R)</li> <li>56 kbps switched data (R: BRI only)</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 Section 5.7</li> <li>GSCR Section 5.7</li> <li>GSCR Section 5.7</li> <li>GSCR Section 5.7</li> <li>CJCSI 6215.01B</li> </ul>
VoIP	No	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>

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**Table 2. PBX 1 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	No	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (C)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer, 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 Section 2.1.2</li> <li>• GSCR Section 2.1.3</li> <li>• GSCR Section 2.1.4</li> <li>• GSCR Section 2.1.5</li> <li>• GSCR Section 2.1.6</li> <li>• GSCR Section 2.1.7</li> <li>• GSCR Section 2.1.8</li> <li>• GSCR Section 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 Section 2.2.1</li> <li>• GSCR Section 2.2.2</li> <li>• GSCR Section 2.2.3</li> <li>• GSCR Section 2.2.4</li> <li>• GSCR Section 2.2.5</li> <li>• GSCR Section 2.2.6</li> <li>• GSCR Section 2.2.7</li> </ul>
Public Safety	No	<ul style="list-style-type: none"> <li>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing call trace (C)</li> <li>• Tandem call trace (C)</li> <li>• Trace of a call in progress (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 2.4.1</li> <li>• GSCR Section 2.4.2</li> <li>• GSCR Section 2.4.3</li> <li>• GSCR Section 2.4.4</li> <li>• GSCR Section 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 Section 2.6</li> <li>• GSCR Section 2.6</li> <li>• GSCR Section 2.6</li> <li>• GSCR Section 2.6.1</li> <li>• GSCR Section 2.6.2</li> <li>• GSCR Section 2.6.3</li> <li>• GSCR Section 2.6.4</li> <li>• GSCR Section 2.6.5</li> <li>• GSCR Section 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 Section 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 Section 2.11.1</li> <li>• GSCR Section 2.11.1</li> <li>• GSCR Section 2.11.1.1</li> <li>• GSCR Section 2.11.1.2</li> <li>• GSCR Section 2.11.1.3</li> <li>• GSCR Section 2.11.1.4</li> <li>• GSCR Section 2.11.1.5</li> <li>• GSCR Section 2.11.1.6</li> <li>• GSCR Section 2.11.1.7</li> <li>• GSCR Section 2.11.1.8</li> <li>• GSCR Section 2.11.1.9</li> <li>• GSCR Section 2.11.1.10</li> </ul>
DSN Hotline Services	No	<ul style="list-style-type: none"> <li>• Hotline restrictions (C)</li> <li>• Auto initiate (C)</li> <li>• Analog and digital (C)</li> <li>• Subscription basis (C)</li> <li>• Protected hotline calling (C)</li> <li>• WWNDP interoperable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12.1-4</li> <li>• GSCR Section 2.12.5</li> </ul>

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**Table 2. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities (continued)</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Network Management	No	<ul style="list-style-type: none"> <li>• Interfaces (C)</li> <li>• Measurements and data generation (C)</li> <li>• Fault management (C)</li> <li>• Configuration management (C)</li> <li>• Accounting management (C)</li> <li>• Performance management (C)</li> <li>• NM controls (C)</li> <li>• Remote access (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 9.1</li> <li>• GSCR Section 9.2</li> <li>• GSCR Section 9.3</li> <li>• GSCR Section 9.4</li> <li>• GSCR Section 9.5</li> <li>• GSCR Section 9.6</li> <li>• GSCR Section 9.7</li> <li>• GSCR Section 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• Electronic Key Telephone Systems (EKTS) (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 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 Section 11.1.1.2</li> <li>• GSCR Section 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 Section 12</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 13</li> </ul>
<b>VoIP</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 MOS 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</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency <math>\leq</math> 60 milliseconds</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Appendix 3</li> <li>• GSCR Appendix 3, paragraph 1.7</li> </ul>

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**Table 2. PBX 1 Requirements (continued)**

Network Gateways					
Gateway	Critical	Requirements Required or Conditional			References
PSTN <sup>1</sup>	No	Trunking	<ul style="list-style-type: none"> <li>Positive Identification Control (C)</li> <li>On-Netting (C)</li> <li>Off-Netting (C)</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>2</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 Section 5.5</li> <li>GSCR Section 4.4</li> <li>GSCR Section 4.1</li> <li>GSCR Section 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 Section 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 BER - Bit Error Ratio BRI - Basic Rate Interface C - Conditional CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCS - Chairman of the Joint Chiefs of Staff CJCSI - CJCS Instruction D/A - Digital to Analog Conversion DIACAP - DoD Information Assurance Certification and Accreditation Process DISA - Defense Information Systems Agency DISR - DoD IT Standards Registry DITSCAP - DoD IT Security Certification and Accreditation Process DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EIA - Electronic Industries Alliance G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement GR-512 - LSSGR: Reliability, Section 12 GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union-Telecommunication Standardization Sector kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 LAN - Local Area Network Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score NI 1/2 - National ISDN Standard 1 or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required SMEO - Small End Office SS7 - Signaling System 7 STE - Secure Terminal Equipment STIGs - Security Technical Implementation Guides STU-III - Secure Telephone Unit - 3rd generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-465-A - Group 3 Facsimile Apparatus for Document Transmission VBD - Variable bit data VoIP - Voice over Internet Protocol VTC - Video Teleconferencing WWNDP - Worldwide Numbering and Dialing Plan					
<b>NOTES:</b> 1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP. 2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.					

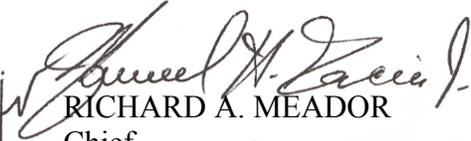
5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at <https://stp.fhu.disa.mil>. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at <http://jit.fhu.disa.mil> (NIPRNet), or <http://199.208.204.125> (SIPRNet). Information related to DSN testing is on the TSSI website at <http://jitc.fhu.disa.mil/tssi>.

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6. The JITC point of contact is Oskar Widecki, DSN 879-5269, commercial (520) 538-5269, FAX DSN 879-4347, or e-mail to oskar.widecki@disa.mil. The tracking number for the SUT is 51825.

FOR THE COMMANDER:

2 Enclosures a/s

  
RICHARD A. MEADOR  
Chief  
Battlespace Communications Portfolio

JITC Memo, JTE, Special Interoperability Test Certification of Nortel Defense Switched Network (DSN) Communications Server (CS) 1000M Cabinet and CS1000M Chassis (including Voice over Internet Protocol [VoIP]) and DSN Option 11C Digital Switching Systems with Software Release 4.5w and Product Enhancement Packages

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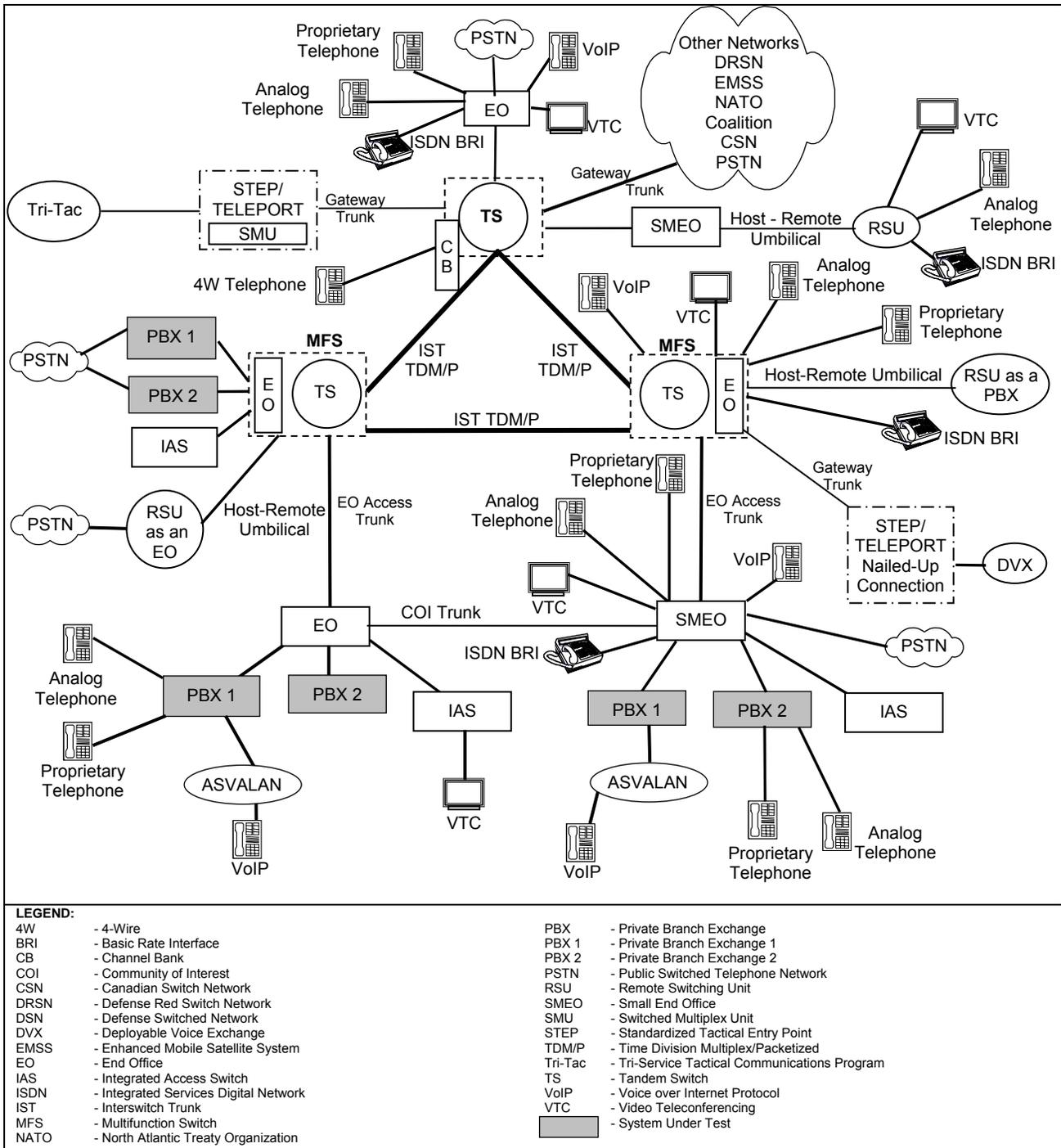
Defense Information Systems Agency (DISA), ATTN: GS23 (Mr. Osman), Room 5w23, 5275 Leesburg Pike (RTE 7), Falls Church, VA 22041

## **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 Cabinet, hereinafter referred to as the System Under Test (SUT), and CS1000M Chassis (including Voice over Internet Protocol [VoIP]) and DSN Option 11C 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 supports Digital Transmission Link Level 1 (T1) and European Basic Multiplex Rate (E1) interfaces. The SUT offers the following features: scalable, distributed platform for growth from 200 to 720 lines and up to 1000 Internet Protocol (IP) Phones, 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://jitic.fhu.disa.mil/tssi>) approved product list. The Nortel DSN CS1000M Chassis employs the same software and trunk/line card hardware as the SUT and was developed for scalability purposes. JITC analysis determined the DSN CS1000M Chassis including VoIP to be functionally identical to the SUT for interoperability certification purposes. The SUT is also certified without VoIP. When the SUT and the CS1000M Chassis are deployed without VoIP they are designated as the Nortel DSN Option 11C Cabinet and DSN Option 11C Chassis, respectively.
- 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 Private Branch Exchange (PBX) 1. The Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. PBX 2 requirements are a subset of the requirements listed in table 2-1. This architecture depicts the relationship of Military Department PBX 1s and PBX 2s to the other DSN switch types.



**Figure 2-1. DSN Architecture**

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

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

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

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

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. PBX 1 Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> <li>Framing (R)</li> <li>Line Code (R)</li> <li>Signaling (R)</li> <li>Alarms (R)</li> </ul>	<ul style="list-style-type: none"> <li>GSCR Section 7</li> <li>GSCR Section 7</li> <li>GSCR Section 5</li> <li>GSCR Section 2.5.7, 7.1.4 &amp; 7.2.2</li> </ul>
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> <li>WWNDP (R)</li> <li>Outpulsing digit formats (C: CAS only)</li> <li>Routing (C)</li> <li>Trunk Groups (C)</li> <li>Call Processing (C)</li> <li>CAS to CCS trunk interworking (C)</li> <li>PCM-24/PCM-30 Interoperation (C)</li> <li>Direct Inward Dialing (C)</li> </ul>	<ul style="list-style-type: none"> <li>GSCR Section 4.5.1</li> <li>GSCR Section 4.5.2</li> <li>GSCR Section 4.2</li> <li>GSCR Section 2.5.5 &amp; 2.5.6</li> <li>GSCR Section 4</li> <li>GSCR Section 3.10</li> <li>GSCR Section 7.3</li> <li>GSCR Section 2.3.2</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	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 Section 3</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>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	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 Section 3.10</li> <li>GSCR Section 3.10</li> <li>GSCR Section 3.10</li> <li>GSCR Section 3.10</li> <li>CJCSI 6215.01B</li> </ul>
		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. PBX 1 Requirements (continued)**

DSN Line Interfaces				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• Directory Number Identification (R)</li> <li>• Line signaling (R)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Alerting Signals and Tones (R)</li> <li>• WWNDP (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 Section 2.1.1</li> <li>• GSCR Section 5.2</li> <li>• GSCR Section 5.2.1</li> <li>• GSCR Section 5.5</li> <li>• GSCR Section 4.5</li> <li>• GSCR Section 4.1</li> <li>• GSCR Section 4.3.3</li> <li>• GSCR Section 4.3.4.1</li> </ul>
ISDN BRI NI 1/2 (ANSI T1.619a)	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 Section 3.1.3</li> <li>• GSCR Section 3</li> <li>• CJCSI 6215.01B</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: BRI only)</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 Section 5.7</li> <li>• GSCR Section 5.7</li> <li>• GSCR Section 5.7</li> <li>• GSCR Section 5.7</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>
DSN Features & Capabilities				
Feature/ Capability	Critical	Requirements Required or Conditional	References	
Common Features	No	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (C)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer, 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 Section 2.1.2</li> <li>• GSCR Section 2.1.3</li> <li>• GSCR Section 2.1.4</li> <li>• GSCR Section 2.1.5</li> <li>• GSCR Section 2.1.6</li> <li>• GSCR Section 2.1.7</li> <li>• GSCR Section 2.1.8</li> <li>• GSCR Section 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 Section 2.2.1</li> <li>• GSCR Section 2.2.2</li> <li>• GSCR Section 2.2.3</li> <li>• GSCR Section 2.2.4</li> <li>• GSCR Section 2.2.5</li> <li>• GSCR Section 2.2.6</li> <li>• GSCR Section 2.2.7</li> </ul>	
Public Safety	No	<ul style="list-style-type: none"> <li>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing call trace (C)</li> <li>• Tandem call trace (C)</li> <li>• Trace of a call in progress (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 2.4.1</li> <li>• GSCR Section 2.4.2</li> <li>• GSCR Section 2.4.3</li> <li>• GSCR Section 2.4.4</li> <li>• GSCR Section 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 Section 2.6</li> <li>• GSCR Section 2.6</li> <li>• GSCR Section 2.6</li> <li>• GSCR Section 2.6.1</li> <li>• GSCR Section 2.6.2</li> <li>• GSCR Section 2.6.3</li> <li>• GSCR Section 2.6.4</li> <li>• GSCR Section 2.6.5</li> <li>• GSCR Section 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 Section 2.8</li> </ul>	

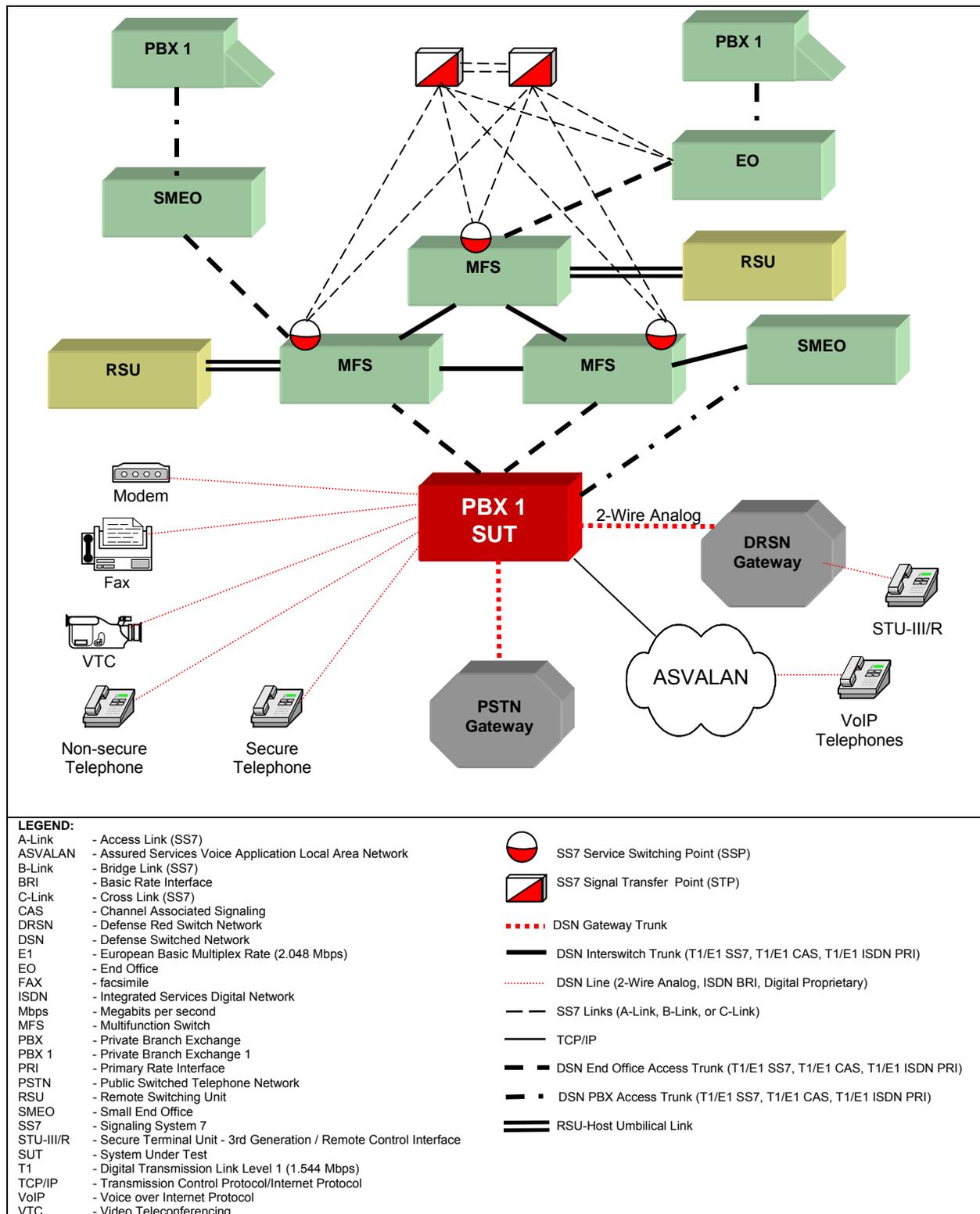
**Table 2-1. PBX 1 Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
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 Section 2.11.1</li> <li>• GSCR Section 2.11.1</li> <li>• GSCR Section 2.11.1.1</li> <li>• GSCR Section 2.11.1.2</li> <li>• GSCR Section 2.11.1.3</li> <li>• GSCR Section 2.11.1.4</li> <li>• GSCR Section 2.11.1.5</li> <li>• GSCR Section 2.11.1.6</li> <li>• GSCR Section 2.11.1.7</li> <li>• GSCR Section 2.11.1.8</li> <li>• GSCR Section 2.11.1.9</li> <li>• GSCR Section 2.11.1.10</li> </ul>
DSN Hotline Services	No	<ul style="list-style-type: none"> <li>• Hotline restrictions (C)</li> <li>• Auto initiate (C)</li> <li>• Analog and digital (C)</li> <li>• Subscription basis (C)</li> <li>• Protected hotline calling (C)</li> <li>• WWNDP interoperable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12</li> <li>• GSCR Section 2.12.1-4</li> <li>• GSCR Section 2.12.5</li> </ul>
Network Management	No	<ul style="list-style-type: none"> <li>• Interfaces (C)</li> <li>• Measurements and data generation (C)</li> <li>• Fault management (C)</li> <li>• Configuration management (C)</li> <li>• Accounting management (C)</li> <li>• Performance management (C)</li> <li>• NM controls (C)</li> <li>• Remote access (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 9.1</li> <li>• GSCR Section 9.2</li> <li>• GSCR Section 9.3</li> <li>• GSCR Section 9.4</li> <li>• GSCR Section 9.5</li> <li>• GSCR Section 9.6</li> <li>• GSCR Section 9.7</li> <li>• GSCR Section 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• Electronic Key Telephone Systems (EKTS) (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 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 Section 11.1.1.2</li> <li>• GSCR Section 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 Section 12</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Section 13</li> </ul>
<b>VoIP</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 MOS 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</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 milliseconds</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Appendix 3</li> <li>• GSCR Appendix 3, paragraph 1.7</li> </ul>

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

Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN <sup>1</sup>	No	Trunking	<ul style="list-style-type: none"> <li>Positive Identification Control (C)</li> <li>On-Netting (C)</li> <li>Off-Netting (C)</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>2</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 Section 5.5</li> <li>GSCR Section 4.4</li> <li>GSCR Section 4.1</li> <li>GSCR Section 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 Section 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 BER - Bit Error Ratio BRI - Basic Rate Interface C - Conditional CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCS - Chairman of the Joint Chiefs of Staff CJCSI - CJCS Instruction D/A - Digital to Analog Conversion DIACAP - DoD Information Assurance Certification and Accreditation Process DISA - Defense Information Systems Agency DISR - DoD IT Standards Registry DITSCAP - DoD IT Security Certification and Accreditation Process DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EIA - Electronic Industries Alliance G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement GR-512 - LSSGR: Reliability, Section 12 GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union-Telecommunication Standardization Sector kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 LAN - Local Area Network Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score NI 1/2 - National ISDN Standard 1 or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required SMEO - Small End Office SS7 - Signaling System 7 STE - Secure Terminal Equipment STIGs - Security Technical Implementation Guides STU-III - Secure Telephone Unit -3rd generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-465-A - Group 3 Facsimile Apparatus for Document Transmission VBD - Variable bit data VoIP - Voice over Internet Protocol VTC - Video Teleconferencing WWNDP - Worldwide Numbering and Dialing Plan				
<b>NOTES:</b> 1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP. 2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.				

**8. TEST NETWORK DESCRIPTION.** The SUT was tested at JITC’s Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system’s required functions and features was conducted using the test configuration depicted in figure 2-2. The SUT was tested as the end-point in relation to the other switches. Figure 2-3 depicts the VoIP configuration.



**Figure 2-2. Test Configuration**

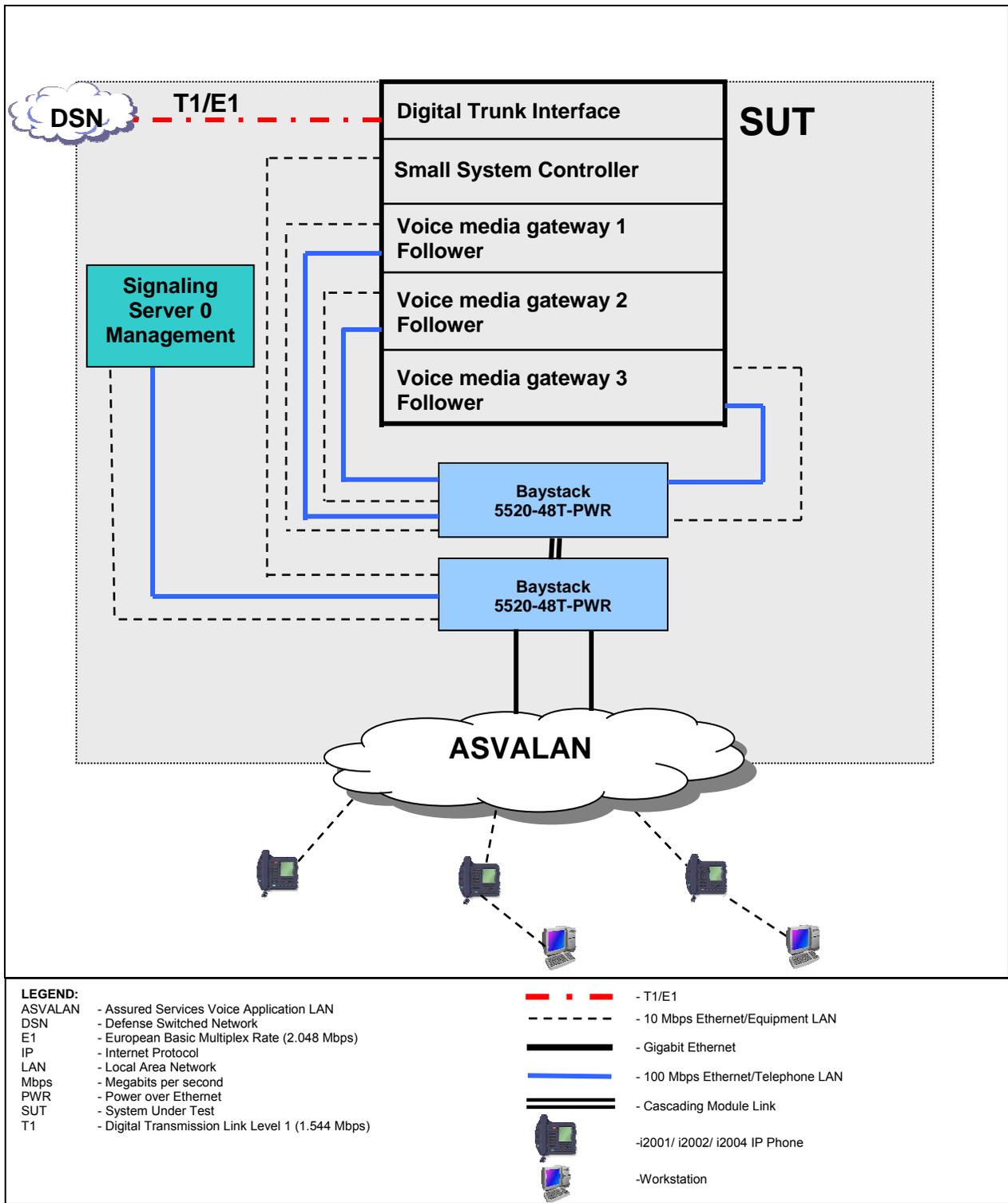


Figure 2-3. SUT IP Test Configuration with ASVALAN

**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 0</b>		
<b>Part Number</b>	<b>Part Description</b>	<b>Part Details</b>
NTDK78AA	Power Supply	Release 11
NTDK20EA	Small System Controller	Release 7
NTRB21AC	DTI or PRI T-1 card	Release 2
NTAK02BC	Serial Digital Interface/D-Channel	Release 1
NTRB21AB	DTI or PRI T-1 card	Release 1
NTRB21AC	DTI or PRI T-1 card	Release 1
NTRB21AA	DTI or PRI T-1 card	Release 9
NTBK50AA	E-1 PRI Card	Release 8
NTBK22AA	Multi-Purpose ISDN Signaling Processor	Release 10
NTAK02BC	Serial Digital Interface/D-Channel	Release 1
NTAG36AC	Recorded Announcement Board	Release 13
<b>Shelf 1</b>		
NTDK78AA	Power Supply	Release 11
NTDK20JA	Small System Controller	Release 1
NT8D02GA	Digital Line Card	Release 4
NT8D09AK	Analog Line Card	Release 4
NT8D15AK	Ear and Mouth Card	Release 10
NT6D71AA	U Interface Line Card	Release 9
NT6D71AA	U Interface Line Card	Release 9
NTAK10DC	E-1 CAS Card	Release 2
NTRH30AA	CallPilot	Release 13 (Firmware 4.0 Build 04.04.04)
<b>Shelf 2</b>		
NTDK78AA	Power Supply	Release 11
NTDK20JA	Small System Controller	Release 1
NT6D70AA	S Interface Line Card	SILC 13
NT8D02GA	Digital Line Card	Release 7
NTRB21AB	DTI or PRI T-1 Card	Release 1
NT8D09BA	Analog Line Card	Release 3
NTRB21AC	DTI or PRI T-1 Card	Release 3
NTVQ01BA	Media Card	Release 5 (Firmware 4.50.25 SMC)
NTVQ01BA	Media Card	Release 5 (Firmware 4.50.25 SMC)
NTVQ01BA	Media Card	Release 4 (Firmware 4.50.25 SMC)
<b>VoIP &amp; Secure Voice Zone</b>		
<b>Device Name</b>	<b>Software Release</b>	
Baystack 5520-48T-PWR	Firmware:4.2.0.11 / Software:4.2.0.005	
Signaling Server	IPL 4.50.25 SMC	
Voice Gateway Media Card	Load IPL 4.5.25 / Firmware SMFW 6.7	
<b>DSN Switches</b>		
<b>System Name</b>	<b>Software Release</b>	
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	

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

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	Mbps - Megabits per second	SMEO - Small End Office
BRI - Basic Rate Interface	MFS - Multifunction Switch	SMFW - Succession Media Firmware
CAS - Channel Associated Signaling	MSL - Meridian Switching Load	S/T - ISDN BRI 4-Wire interface
CM - Communication Manager	PBX 1 - Private Branch Exchange 1	SU - Software Update
DSN - Defense Switched Network	PBX 2 - Private Branch Exchange 2	SUT - System Under Test
DTI - Digital Trunk Interface	PRI - Primary Rate Interface	T - ISDN BRI 4-Wire Interface
E-1 - European Basic Multiplex Rate (2.048 Mbps)	PWR - Power over Ethernet	T-1 - Digital Transmission Link Level 1
EO - End Office	SE - Succession Enterprise	TSG - Telephone Secure Group
EWSD - Elektronisches Wählsystem Digital	SILC - S Interface Line Card	U - ISDN BRI 2-Wire Interface
IPL - Internet Protocol Line	SMC - Succession Media Card	VoIP - Voice over Internet Protocol
ISDN - Integrated Services Digital Network		

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

Core Software Patch Groups					
Patch Name	Date Created	Patch Name	Date Created	Patch Name	Date Created
Q01404357	15 December 2006	Q01316548	15 December 2006	Q01216707	15 December 2006
Q01174670	15 December 2006	Q01259981	15 December 2006	Q01301795	15 December 2006
Q01381841	15 December 2006	Q01335294	15 December 2006	Q01305245	15 December 2006
Q01257721	15 December 2006	Q01287799	15 December 2006	Q01307901	15 December 2006
Q01173871	15 December 2006	Q01326429	15 December 2006	Q01289110	15 December 2006
Q01280834	15 December 2006	Q01267406	15 December 2006	Q01277756	15 December 2006
Q01279732	15 December 2006	Q01269458	15 December 2006	Q01100258	15 December 2006
Q01257418	15 December 2006	Q01278582	15 December 2006	Q01188689	15 December 2006
Q01267308	15 December 2006	Q01261568	15 December 2006	Q01237349	15 December 2006
Q01237867	15 December 2006	Q01205499	15 December 2006	Q01052805	15 December 2006
Q01249770	15 December 2006	Q01265224	15 December 2006	Q01222906	15 December 2006
Signaling Server Software Patch Groups					
Patch Name	Date Created	Patch Name	Date Created	Patch Name	Date Created
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
Voice Media Gateway Card Software Patch Groups					
Patch Name	Date Created	Patch Name	Date Created	Patch Name	Date Created
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/>.

**(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 interface 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 without manual intervention 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 paragraph 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 milliseconds (ms) to 350 ms. The SUT recognizes wink start signals from 100 ms to 925 ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290 ms, this anomaly has no operational impact.

**(c)** The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since glare resolution is conditional for a PBX 1, the operational impact is minor.

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

**(e)** 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.

**(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 NI 2 BRI interface. The only supported and certified BRI interface is NI 1. The NI 2 BRI interface is not required for a PBX 1 as specified by GSCR paragraph 2.3.3. The primary differences between NI 1 and NI 2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future. Also, BRI is not required for a PBX 1. This anomaly has a minor operational impact.

**(b)** The SUT will only support a BRI NI 1 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. Since BRI is not required for a PBX 1, 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 GSCR paragraph 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.

**(d)** 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. Since this is a conditional requirement, there is no operational impact.

**(e)** 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

**(3) Features and Capabilities.** 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)** When call Call Forwarding Variable (CFV) is assigned to any station on the SUT (except BRI, which does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or PSTN. Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. In accordance with 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.

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

**(c)** Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR paragraph 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.

**(d)** The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in GSCR paragraph 2.4.4. Since this is not required for a PBX 1, this anomaly has a minor operational impact.

**(e)** The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by GSCR paragraph 2.12. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. Since this feature is not required by a PBX 1, the operational impact is minor.

**(f)** 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. 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 (DRSN) Gateways with no exceptions. The PSTN certified interfaces are T1 and E1 CAS, T1 and E1 PRI, and Ground Start Line. The DRSN certified interface is Twisted Pair Copper 2-Wire analog.

**(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. Class of Service (CoS) and Quality of Service (QoS).** The GSCR, appendix 3, section A3.3.2, outlines several methodologies to implement CoS and QoS. The 802.1p/Q at the Data Link Layer (L2) and Differentiated Services Code Point (DSCP) at the Network Layer (L3) were two CoS mechanisms that the certified network products employed. The SUT provides CoS by assignment of an 802.1p/Q tag. 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. Coder/Decoder (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 CODEC with a 20 ms packet fill was required and was met by the SUT VoIP solution.

#### **4. Traffic Engineering**

**a. Phones.** The Nortel IP phones that met the critical interoperability requirements for certification were the i2001, i2002, and i2004 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 switch 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 Digital Signal Processors for IP to Time Division Multiplexing (TDM) calls. Table 2-4 shows the maximum number of telephony subscribers supported by each processor. 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 Only	Mixed TDM and IP Phones <sup>1</sup>
MSC <sup>2</sup>	CS1000M Chassis	720	800	144 TDM/300 IP
SCC88	CS1000M Chassis	720	1000	640 TDM/800 IP
SSC	CS1000M Cabinet	720	1000	600 TDM/1000 IP

**LEGEND:**  
CS - Communications Server  
IP - Internet Protocol  
MSC - Mini System Controller  
SSC - Small System Controller  
SUT - System Under Test  
TDM - Time Division Multiplexing

**NOTES:**  
1 Nortel Engineering Configurator (NNEC) must be used to determine the IP/TDM phone ratio.  
2 With MSC, systems are limited to a maximum of 2000 Corporate Directory entries.

**(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 percent averaged over any five-minute period. The SUT packet loss was measured at 0.00 percent.

**(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 Internet Protocol version 4 (IPv4). IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008:

- 1.** Conformant with IPv6 standards profile contained in the DISR.

with IPv4.

2. Maintaining interoperability in heterogeneous environments and
3. Commitment to upgrade as the IPv6 standard evolves.
4. Availability of contractor/vendor IPv6 technical support.

**b. System Interoperability Results.** The SUT is certified for joint use in the DSN as a PBX 1 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. One of the optional requirements, Call Forwarding Variable processing for precedence calls, did not meet the specifications and therefore is not certified for use in the DSN. The interoperability test summary and the rest of the exceptions that were identified during testing are listed 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 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 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 100 ms (+/-) 5 ms. <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 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 CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT does not support NI 2 BRI. <sup>7</sup> The only supported and certified interface is NI 1 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 CRs and FRs.
VoIP (ITU-T H.323 with Proprietary Signaling Interface)	No	Certified	Met all CRs and FRs with the following minor exception: Precedence call waiting indication is unique on VoIP phones. <sup>11</sup>

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

<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 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 CRs and FRs with the following minor exception: 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 CRs and FRs with the following minor 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>	
Precedence Access Threshold	No	Not Tested	This feature is not supported. <sup>16</sup>	
DSN Hotline Services	No	Certified	Met all CRs and FRs with the following minor exception: The SUT does not support a protected hotline specified list. <sup>17</sup>	
Network Management	No	Certified	Met all CRs and FRs.	
Multiline Hunt Service	No	Certified	Met all CRs and FRs with the following minor exception: 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 CRs and FRs.	
Reliability	Yes	Certified	Met all 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/apl.html">http://jitc.fhu.disa.mil/tssi/apl.html</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 CRs and FRs.
	E1 CAS (DTMF, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all CRs and FRs.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>21</sup>	Met all CRs and FRs.

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

LEGEND:			
ANSI	- American National Standards Institute	GR-506-CORE	- LSSGR: Signaling for Analog Interfaces
ASVALAN	- Assured Services Voice Application Local Area Network	GSCR	- Generic Switching Center Requirements
BRI	- Basic Rate Interface	H.323	- Standard for multi-media communications on packet-based networks
CAS	- Channel Associated Signaling	IPv4	- Internet Protocol version 4
CFV	- Call Forwarding Variable	IPv6	- Internet Protocol version 6
CRs	- Capability Requirements	ISDN	- Integrated Services Digital Network
DISA	- Defense Information Systems Agency	IT	- Information Technology
DoD	- Department of Defense	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector
DP	- Dial Pulse	JITC	- Joint Interoperability Test Command
DRSN	- Defense Red Switch Network	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements
DSN	- Defense Switched Network	Mbps	- Megabits per second
DSS1	- Digital Subscriber Signaling 1	MFR1	- Multifrequency Recommendation 1
DTMF	- Dual Tone Multi-Frequency	MLPP	- Multi-Level Precedence and Preemption
E1	- European Basic Multiplex Rate (2.048 Mbps)	ms	- millisecond
EKTS	- Electronic Key Telephone System	NI 1	- National ISDN Standard 1
FRs	- Feature Requirements	NI 2	- National ISDN Standard 2
GR	- Generic Requirement	NI 1/2	- National ISDN Standard 1 or 2
		PBX 1	- Private Branch Exchange 1
		PM	- Program Manager
		PRI	- Primary Rate Interface
		PSTN	- Public Switched Telephone Network
		Q.931	- Signaling Standard for ISDN
		Q.955.3	- ISDN signaling standard for E1 MLPP
		SS7	- Signaling System 7
		SUT	- System Under Test
		T1	- Digital Transmission Link Level 1 (1.544 Mbps)
		T1.607	- ISDN - Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
		T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
		TSS1	- Telecom Switched Services Interoperability
		TPC	- Twisted Pair Copper
		VoIP	- Voice over Internet Protocol

**NOTES:**

- 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 interface 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 without manual intervention when the span is returned to service.
- T1 CAS wink start signals greater than the specified maximum limit are recognized as valid by the SUT. The GSCR paragraph 5.3.3.3.1 and GSCR figure 3-2 defines the wink start recognition limits between 100 ms to 350 ms. The SUT recognizes wink start signals from 100 ms to 925 ms in duration. Since all certified switches within the DSN must generate the wink start signal within 140-290 ms, this anomaly has no operational impact.
- The SUT does not support glare hold resolution on CAS trunks. It only supports glare release. Since glare resolution is conditional for a PBX 1, the operational impact is minor.
- This interface is not supported. There is no operational impact because it is not a critical requirement.
- The on/off hook pulse that masks the preemption signal on the E1 CAS is intermittently out of the required tolerance of 100 ms (+/-) 5 ms. The pulse width was measured to be greater than 100 ms about 20 percent of the time with the highest at 128 ms. This never had any impact on the ability of the SUT to support call preemption. Therefore, this anomaly has no operational impact.
- 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, and the trunks can be returned to maintenance busy condition manually.
- The SUT does not support an NI 2 BRI interface. The only supported and certified BRI interface is NI 1. The NI 2 BRI interface is not required for a PBX 1 as specified by GSCR paragraph 2.3.3. The primary differences between NI 1 and NI 2 are supplemental features which currently are not fielded within the DSN nor are there plans to field them in the future. Also, BRI is not required for a PBX 1. This anomaly has a minor operational impact.
- The SUT will only support a BRI NI 1 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. Since BRI is not required for a PBX 1, the operational impact is minor.
- The precedence above ROUTINE ringing cadence that the SUT applies to BRI phones does not meet the specifications as detailed in GSCR paragraph 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.
- 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. Since this is a conditional requirement, there is no operational impact.
- 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.
- When call CFV is assigned to any station on the SUT (except BRI, which does not support CFV) and CFV is invoked by the user all precedence calls placed to that instrument are forwarded to the DSN or PSTN. Additionally any station with CFV invoked does not receive a "ping" ring when calls are being forwarded. In accordance with 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.
- The conference disconnect tone that is provided by the SUT does not meet the specifications designated in GSCR paragraph 5.5.2. The SUT conference disconnect tone is distinguishable from other DSN tones and cadences; therefore this anomaly has a minor operational impact.
- Stations cannot be classmarked to prohibit the attendant console from performing a busy override to an active call, as specified in the GSCR paragraph 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.
- The SUT cannot perform a tandem call trace of a specified distant office directory number as specified in GSCR paragraph 2.4.4. Since this is not required for a PBX 1, this anomaly has a minor operational impact.
- This feature is not supported. There is no operational impact because it is not a critical requirement.
- The SUT will not allow the protection of a hotline call originator through the use of a hotline list as required by GSCR paragraph 2.12. However, this capability can be accomplished with the SUT by classmarking authorized hotline users for receiving only calls from other hotline callers. Since this feature is not required by a PBX 1 the operational impact is minor.
- 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. This anomaly has a minor operational impact.
- Security is tested DISA-led Information Assurance test teams and published in a separate report.
- An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor must state, in writing, compliance to the following criteria by 30 June 2008:
  - Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DISR).
  - Maintaining interoperability in heterogeneous environments and with IPv4.
  - Commitment to upgrade as the IPv6 standard evolves.
  - Availability of contractor/vendor IPv6 technical support.
- 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.