



## 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)

28 October 2005

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

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

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification. Additional references are provided in enclosure 1.

2. The Avaya G3CSI (ProLogix) Digital Switching System with Software Release CM 3.0 (R013i.00.0.340.5) including Voice over Internet Protocol (VoIP), hereinafter referred to as the system under test (SUT), met all of its critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The identified test discrepancies shown in the Certification Testing Summary (enclosure 2), which remained open after software patches were applied and regression testing was completed, 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 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 certification is based on interoperability testing conducted by the JITC at the Global Information Grid Network Test Facility, Fort Huachuca, Arizona, from 18 May through 05 August 2005, and review of vendor's letters of compliance on 22 August 2005. Testing was conducted in an environment that emulates the DSN. The Certification Testing Summary in enclosure 2 provides more details about the test, documents the test results, and describes the tested network and system configurations. System interoperability should be verified before deployment in an operational environment that varies significantly from the test environment.

4. The SUT was tested with multiple Voice Application (VA) Local Area Network (LAN)s. Refer to the Telecom Switched Services Interoperability (TSSI) website <http://jitc.fhu.disa.mil/tssi> for the certified DSN VA LAN hardware and software components. The interoperability summary of the SUT is indicated in table 1. The Capability Requirements (CRs) and Feature Requirements

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

(FRs) for the DSN are listed in table 2. The Avaya switch product line offers a Remote Switch Unit capability referred to as the Survivable Remote Processor Expansion Port Network. Testing was performed on this capability, however it did not meet the minimum critical requirements and it is not certified. This interoperability test status is based upon evaluation of:

- 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 Letter(s) of Compliance (LoC).
- c. PBX 1 FRs/CRs specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- d. The overall system interoperability performance derived from test procedures listed in reference (e).

**Table 1. SUT 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, MFR1, DP)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT acknowledges a wink start signal beyond the maximum 350 ms. <sup>1</sup> The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</sup>
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT acknowledges a wink start signal beyond the maximum 350 ms. <sup>1</sup> The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</sup>
T1 ISDN PRI NI 1/2 (ANSI T1. 619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</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 CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP	No	Certified	Met all CRs and FRs.
<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 ERs and FRs with the following minor exceptions: The SUT does not support priority Call Pickup with calls above ROUTINE <sup>4</sup> and does not allow the two legs of a three-way call to be established at different precedence levels. <sup>5</sup>
Attendant	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
Public Safety	No	Certified	Met all critical CRs and FRs.

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

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

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Preset Conferencing	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.	
Nailed-up Connections	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.	
PAT	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.	
DSN Hotline Services	No	Certified	Met all CRs and FRs.	
Network Management	No	Not Tested	This capability is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.	
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs.	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 6.		
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6. <sup>7</sup>	
VoIP LANs	No	Certified	Met all CRs and FRs.	
Network Gateways				
	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, DP)	No	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.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified <sup>8</sup>	Met all critical CRs and FRs.
<b>LEGEND:</b> ANSI - American National Standards Institute BRI - Basic Rate Interface CAS - Channel Associated Signaling CRs - Capability Requirements DISA - Defense Information Systems Agency DITSCAP - Department of Defense Information Technology Security Certification and Accreditation Process 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 GSCR - Generic Switching Center Requirements IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network LAN - Local Area Network Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption ms - milliseconds NI 1/2 - National ISDN 1 or 2 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PNT - Preemption Notification Tone PRI - Primary Rate Interface PSTN - Public Switched Telephone Network SS7 - Signaling System 7 SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.607 - ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1 T1.619a - SS7 and ISDN MLPP Signaling Standard For T1 TPC - Twisted Pair Copper VoIP - Voice over Internet Protocol				
<b>NOTES:</b> 1 The SUT acknowledges a wink start signal beyond the 350 ms maximum (approximately 390 ms). On an E1 CAS the SUT will correctly acknowledge a wink start signal. The operational impact is minor. 2 The SUT does not support the correct length of PNT when an active call is directly preempted. A three second PNT is sent to the party being preempted instead of sending PNT until the preempted party goes on hook. The operational impact is minor. 3 The SUT does not support the correct distinctive ring cadence in accordance with the GSCR for precedence above ROUTINE calls placed via a trunk. The operational impact is minor. 4 The SUT does not support priority Call Pickup with precedence calls above ROUTINE. When a precedence call above ROUTINE is ringing in a call pickup group and a ROUTINE call is also ringing in the same call pickup group, the SUT randomly picks which call to pickup when the feature is activated. Since the higher precedence call is diverted to an alternate directory number if unanswered, the operational impact is minor. 5 The SUT does not support the classmarking of the two legs of a three-way call at different precedence levels. This is due to the fact that the SUT connects all three parties to a single time slot. Instead, the SUT classmarks all the parties at the highest precedence. The operational impact is minor. 6 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 7 The VoIP components provided did not support IPv6. The risk of not testing is minor due to the fact that IPv6 is not currently implemented within the DSN and is not scheduled to be fully implemented until 2008 8 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.				

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

**Table 2. PBX 1 Requirements**

Interface	Critical	Requirements Required (R) or Conditional (C)		References
<b>DSN Line Interfaces</b>				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line signaling (R)</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 Sect. 2.1.1</li> <li>• GSCR Sect. 5.2</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.3</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
		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>
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Facsimile	<ul style="list-style-type: none"> <li>• Analog: EIA/TIA-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 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 Trunk Interfaces</b>				
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> <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 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 CAS (MFR1, DTMF, DP)	No (Europe only)		Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Facsimile	<ul style="list-style-type: none"> <li>• Analog: EIA/TIA-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 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: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>

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

Interface	Critical	Requirements Required (R) or Conditional (C)	References
<b>DSN Features &amp; Capabilities</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 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	No	<ul style="list-style-type: none"> <li>• E911 (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 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 (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</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>• Operation 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	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 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>

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

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

Interface	Critical	Requirements Required (R) or Conditional (C)	References
<b>DSN Features &amp; Capabilities (continued)</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 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>			
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>• MOS 4.0 or better</li> <li>• ITU-T G.711 PCM Codec</li> <li>• Security in accordance with DITSCAP</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 msec</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> </ul>
LANs	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>• LAN parameters</li> <li>• CoS/QoS</li> <li>• VLANs</li> <li>• IEEE Standards Conformance</li> <li>• .99999 availability</li> <li>• Modular devices</li> <li>• 2 second link restoral</li> <li>• LAN NM</li> <li>• Traffic Engineering</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> </ul>

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

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

Interface	Critical	Requirements Required (R) or Conditional (C)	References		
<b>Network Gateways</b>					
PSTN <sup>2</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>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>		
EMSS	No	CJCS approved requirements not defined.			
NGCS	No	CJCS approved requirements not defined.			
<b>LEGEND:</b>					
2W	- 2-Wire	EKTS	- Electronic Key Telephone System	NX64	- Data format restricted to multiples of 64 kbps
A/D	- Analog to Digital Conversion	EMSS	- Enhanced Mobile Satellite System	PAT	- Precedence Access Threshold
AMA	- Automated Message Accounting	G.711	- PCM of voice frequencies	PBX 1	- Private Branch Exchange 1
ANSI	- American National Standards Institute	GR	- Generic Requirement	PCM	- Pulse Code Modulation
App.	- Appendix	GSCR	- Generic Switching Center Requirements	PCM-24	- Pulse Code Modulation - 24 Channels
BER	- Bit Error Ratio	H.320	- Standard for narrowband VTC	PCM-30	- Pulse Code Modulation - 30 Channels
BRI	- Basic Rate Interface	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PRI	- Primary Rate Interface
C	- conditional	IPv6	- Internet Protocol version 6	PSTN	- Public Switched Telephone Network
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	QoS	- Quality of Service
CJCS	- Chairman of the Joint Chiefs of Staff	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	R	- Required
CJCSI	- CJCS Instruction	kbps	- kilobits per second	Sect.	- Section
CoS	- Class of Service	KXX	- K= any number 2-8; X= any number 1-9	SS7	- Signaling System 7
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	STE	- Secure Terminal Equipment
DISA	- Defense Information Systems Agency	Mbps	- Megabits per second	STU-III	- Secure Telephone Unit - 3 <sup>rd</sup> generation
DISR	- DOD IT Standards Registry	MFR1	- Multi-Frequency Recommendation 1	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DITSCAP	- DOD IT Security and Accreditation Process	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN MLPP Signaling Standard For T1
DN	- Directory Number	MOS	- Mean Opinion Score	TIA	- Telecommunications Industry Association
DOD	- Department of Defense	msec	- milliseconds	TIA-465A	- Performance and Compatibility Requirements for Telephone Sets with Loop Signaling
DP	- Dial Pulse	NATO	- North Atlantic Treaty Organization	VBD	- Variable bit data
DRSN	- Defense Red Switch Network	NGCS	- NATO Gateway Communication Switch	VLAN	- Virtual LAN
DSN	- Defense Switched Network	NI 1/2	- National ISDN 1 or 2	VoIP	- Voice over Internet Protocol
DTMF	- Dual Tone Multi-Frequency	NM	- Network Management	VTC	- Video Teleconferencing
E1	- European Basic Multiplex Rate (2.048 Mbps)	NX56	- Data format restricted to multiples of 56 kbps	WWNDP	- Worldwide Numbering and Dialing Plan
E911	- Emergency 911				
EIA	- Electronic Industries Alliance				
<b>NOTES:</b>					
1 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.					
2 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.					
3 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.					

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

JITC Memo, JTE, Special Interoperability Test Certification of Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems (Includes Voice over Internet Protocol)

6. The JITC point of contact is Capt. Michel Roy, DSN 821-8575, commercial (520) 533-8575, FAX DSN 879-4347, or e-mail to michel.roy.ca@disa.mil.

FOR THE COMMANDER:



RICHARD A. MEADOR  
Chief  
Networks and Transport Division

2 Enclosures a/s

Distribution:

Joint Staff J6I, Room-1E565, Pentagon, Washington, DC 20318-6000

Joint Interoperability Test Command, Washington Operations Division, NSWC, ATTN: JT1, Building 900, 101 Strauss Avenue, Indian Head, MD 20640-5035

Defense Information Systems Agency, GIG Enterprise Services Engineering Directorate, NETCENTRICITY, REQUIREMENTS, ANALYSIS & ASSESSMENTS BRANCH, ATTN: GE333, Rm. 244, 5600 Columbia Pike, Falls Church, VA 22041-2770

Defense Information Systems Agency, GIG-Combat Support Directorate, DSN SYSTEMS MANAGEMENT BRANCH, ATTN: GS235, Rm. 5W248A, 5275 Leesburg Pike, Falls Church, VA 22041

Office of Chief of Naval Operations (N61C22), CNON6/7, 2000 Navy Pentagon, Washington, DC 20350

Headquarters US Air Force, AF/XICC, 1250 Pentagon, Washington, DC 20330-1250

Department of the Army, Office of the Secretary of the Army, G-6/ASA (ALT), ATTN: ASAALT (SAAL-SSI), 103 Army Pentagon, Washington, DC 20310-0103

US Marine Corps (C4ISR), MARCORSYSCOM, 2200 Lester Street, Quantico, VA 22134

DOT&E, Strategic and C3I Systems, 1700 Defense Pentagon, Washington, DC 20301-1700

US Coast Guard, COMDT/G-SCE (C4), 2100 2nd Street SW, Washington, DC 20593

Office of Assistant Secretary of Defense, OASD (NII)/DoD CIO, Crystal Mall 3, 7<sup>th</sup> Floor, Suite 700, 1931 Jefferson-Davis Hwy, Arlington, VA 22202

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

US Joint Forces Command, J6I, C4 Plans and Policy, 1562 Mitscher Ave, Norfolk, VA 23551-2488

Defense Intelligence Agency, ATTN: DS-CIO, Bldg 6000, Bolling AFB, Washington, DC 20340-3342

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

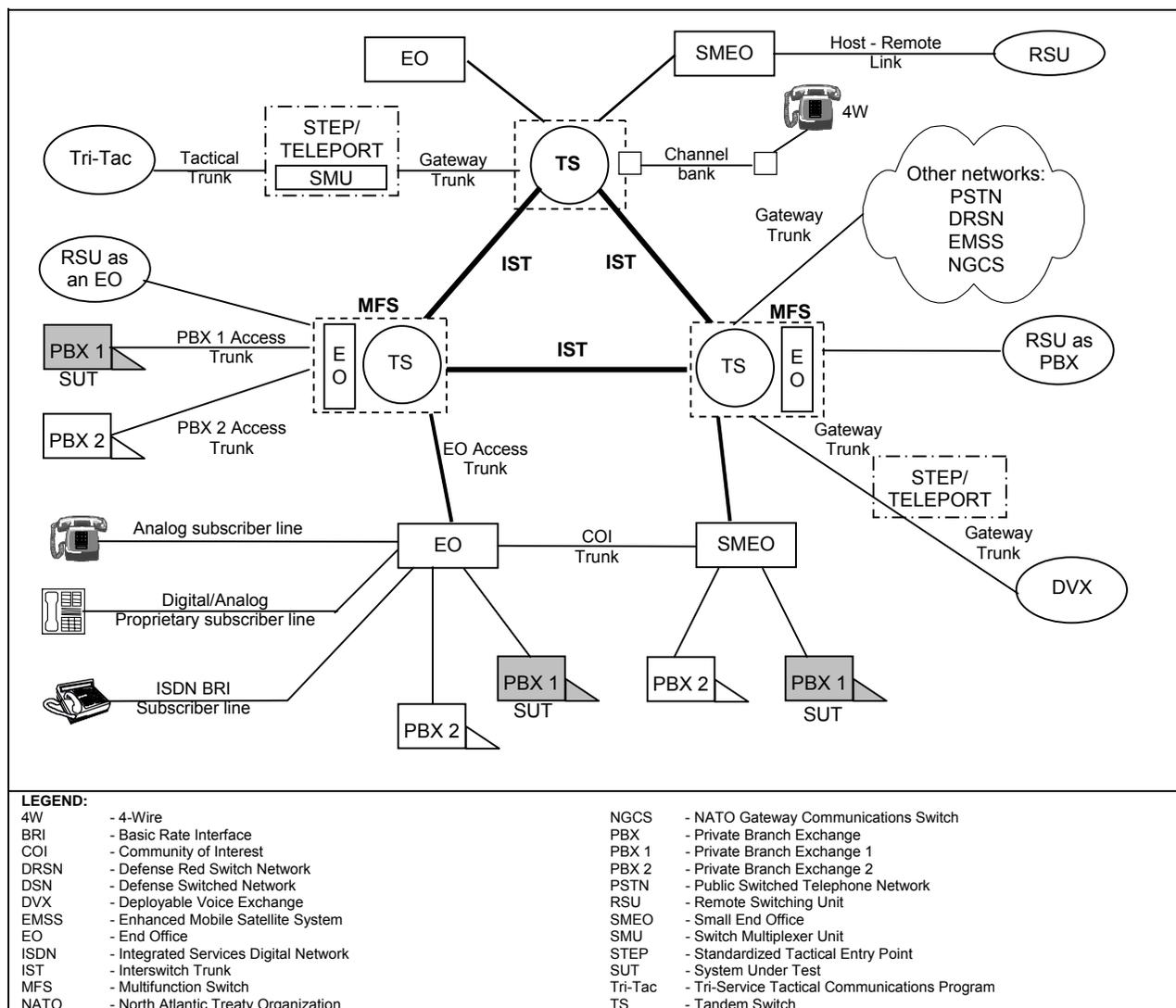
Defense Information Systems Agency (DISA), ATTN: GS23 (Mr. Osman), Room 5w23, 5275 Leesburg Pike (RTE 7), Falls Church, VA 22041

## **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 (DISA), "Defense Switched Network (DSN) Generic Switching Center Requirements (GSCR), Change 1," 1 March 2005
- (e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP)," 23 April 2004

## CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** Avaya G3CSI (ProLogix) with Software Release CM 3.0 (R013i.00.0.340.5) Digital Switching Systems including Voice over Internet Protocol (VoIP), hereinafter referred to as the System Under Test (SUT).
- 2. PROPONENT.** Defense Information Systems Agency (DISA).
- 3. PROGRAM MANAGER.** Mr. Howard Osman, GS23, Room 5W23, 5275 Leesburg Pike, Falls Church, VA 22041, E-mail: Howard.Osman@disa.mil.
- 4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.
- 5. SYSTEM UNDER TEST DESCRIPTION.** The SUT has the maximum capacity of 900 ports. It supports a maximum of 500 lines and 400 trunks. The SUT provides call processing business applications such as voice messaging, shared voice mail, small call center-networking capabilities, and expert systems for remote diagnostics and self-healing. The Avaya switch product line offers a Remote Switch Unit capability referred to as the Survivable Remote Processor Expansion Port Network. Testing was performed on this capability, however it did not meet the minimum critical requirements and it is not certified. This product line also offers a VoIP capability. This capability was tested and is covered by this certification. Avaya's product line of digital switches is currently in use within the Defense Switched Network (DSN) providing Small End Office Switch and Private Branch Exchange (PBX) functionality.
- 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 PBXs. The Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.



**Figure 2-1. DSN Architecture**

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

- a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services."
- b. GSCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letter(s) of Compliance (LoC).
- c. GSCR PBX 1 Capability and Feature Requirements (CRs/FRs) verified through JITC testing and/or vendor submission of LoC.

**Table 2-1. PBX 1 Requirements**

Interface	Critical	Requirements Required (R) or Conditional (C)		References
<b>DSN Line Interfaces</b>				
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line signaling (R)</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 Sect. 2.1.1</li> <li>• GSCR Sect. 5.2</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.3</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
		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>
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Facsimile	<ul style="list-style-type: none"> <li>• Analog: EIA/TIA-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 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 Trunk Interfaces</b>				
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> <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 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 CAS (MFR1, DTMF, DP)	No (Europe only)		Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Facsimile	<ul style="list-style-type: none"> <li>• Analog: EIA/TIA-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 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: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>

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

Interface	Critical	Requirements Required (R) or Conditional (C)	References
<b>DSN Features &amp; Capabilities</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 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	No	<ul style="list-style-type: none"> <li>• E911 (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 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 (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</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>• Operation 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	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 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>

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

Interface	Critical	Requirements Required (R) or Conditional (C)	References
<b>DSN Features &amp; Capabilities (continued)</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 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>			
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>• MOS 4.0 or better</li> <li>• ITU-T G.711 PCM Codec</li> <li>• Security in accordance with DITSCAP</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 msec</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> </ul>
LANs	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>• LAN parameters</li> <li>• CoS/QoS</li> <li>• VLANs</li> <li>• IEEE Standards Conformance</li> <li>• .99999 availability</li> <li>• Modular devices</li> <li>• 2 second link restoral</li> <li>• LAN NM</li> <li>• Traffic Engineering</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> </ul>

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

Interface	Critical	Requirements Required (R) or Conditional (C)		References	
<b>Network Gateways</b>					
PSTN <sup>2</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>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>	
EMSS	No	CJCS approved requirements not defined.			
NGCS	No	CJCS approved requirements not defined.			
<b>LEGEND:</b>					
2W	- 2-Wire	EKTS	- Electronic Key Telephone System	NX64	- Data format restricted to multiples of 64 kbps
A/D	- Analog to Digital Conversion	EMSS	- Enhanced Mobile Satellite System	PAT	- Precedence Access Threshold
AMA	- Automated Message Accounting	G.711	- PCM of voice frequencies	PBX 1	- Private Branch Exchange 1
ANSI	- American National Standards Institute	GR	- Generic Requirement	PCM	- Pulse Code Modulation
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BRI	- Basic Rate Interface	IEEE	- Institute of Electrical and Electronics Engineers, Inc.	PRI	- Primary Rate Interface
C	- conditional	IPv6	- Internet Protocol version 6	PSTN	- Public Switched Telephone Network
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	QoS	- Quality of Service
CJCS	- Chairman of the Joint Chiefs of Staff	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	R	- Required
CJCSI	- CJCS Instruction	kbps	- kilobits per second	Sect.	- Section
CoS	- Class of Service	KXX	- K= any number 2-8; X= any number 1-9	SS7	- Signaling System 7
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	STE	- Secure Terminal Equipment
DISA	- Defense Information Systems Agency	Mbps	- Megabits per second	STU-III	- Secure Telephone Unit – 3 <sup>rd</sup> generation
DISR	- DOD IT Standards Registry	MFR1	- Multi-Frequency Recommendation 1	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DITSCAP	- DOD IT Security and Accreditation Process	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN MLPP Signaling Standard For T1
DN	- Directory Number	MOS	- Mean Opinion Score	TIA	- Telecommunications Industry Association
DOD	- Department of Defense	msec	- milliseconds	TIA-465A	- Performance and Compatibility Requirements for Telephone Sets with Loop Signaling
DP	- Dial Pulse	NATO	- North Atlantic Treaty Organization	VBD	- Variable bit data
DRSN	- Defense Red Switch Network	NGCS	- NATO Gateway Communication Switch	VLAN	- Virtual LAN
DSN	- Defense Switched Network	NI 1/2	- National ISDN 1 or 2	VoIP	- Voice over Internet Protocol
DTMF	- Dual Tone Multi-Frequency	NM	- Network Management	VTC	- Video Teleconferencing
E1	- European Basic Multiplex Rate (2.048 Mbps)	NX56	- Data format restricted to multiples of 56 kbps	WWNDP	- Worldwide Numbering and Dialing Plan
E911	- Emergency 911				
EIA	- Electronic Industries Alliance				
<b>NOTES:</b>					
1 DITSCAP 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 notional test configuration depicted in figure 2-2. Per this configuration, the SUT was tested as the end-point in relation to the other switches. Figure 2-3 depicts the VoIP notional diagram used to test the SUT interaction with a DSN Voice Application (VA) Local Area Network (LAN).

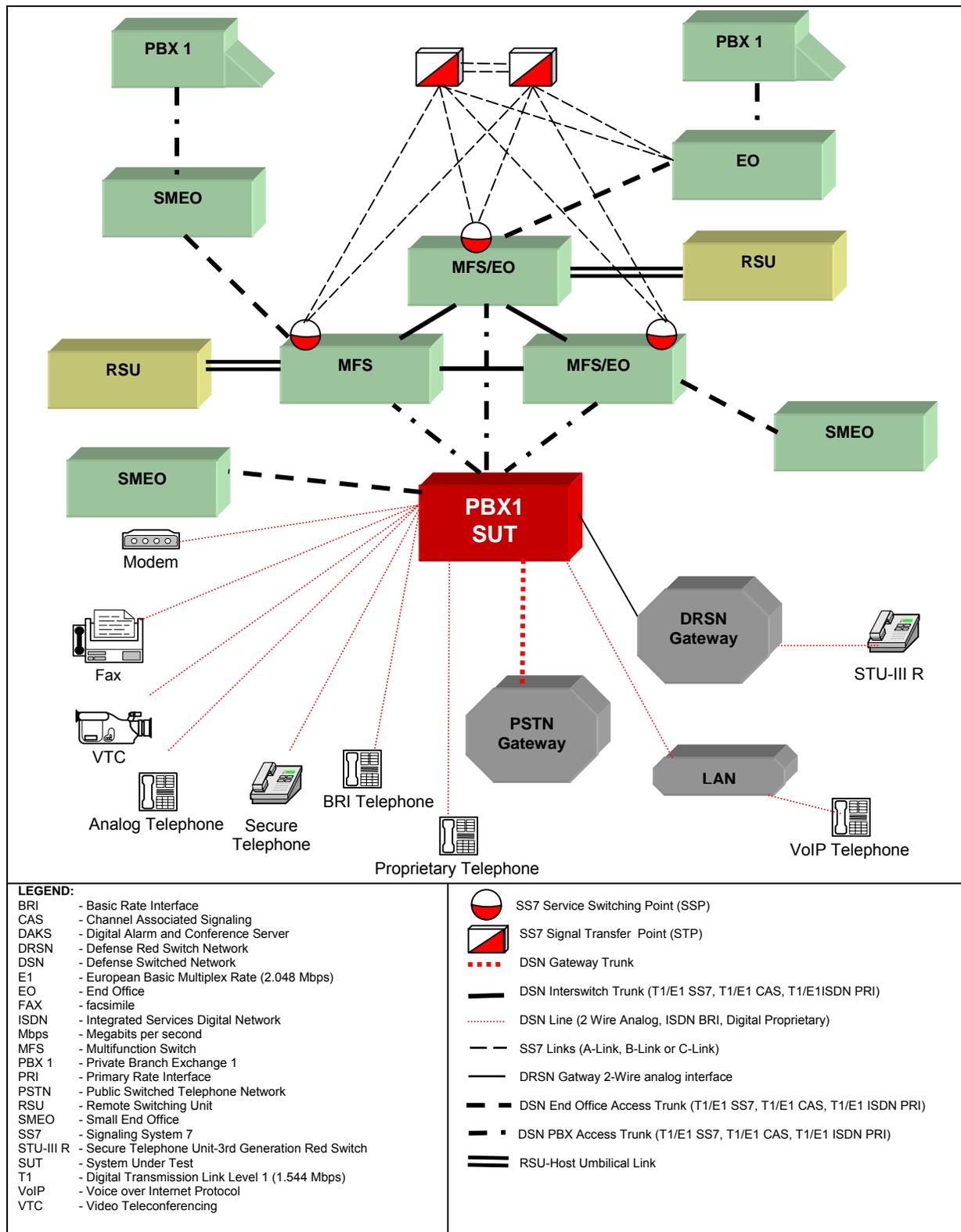
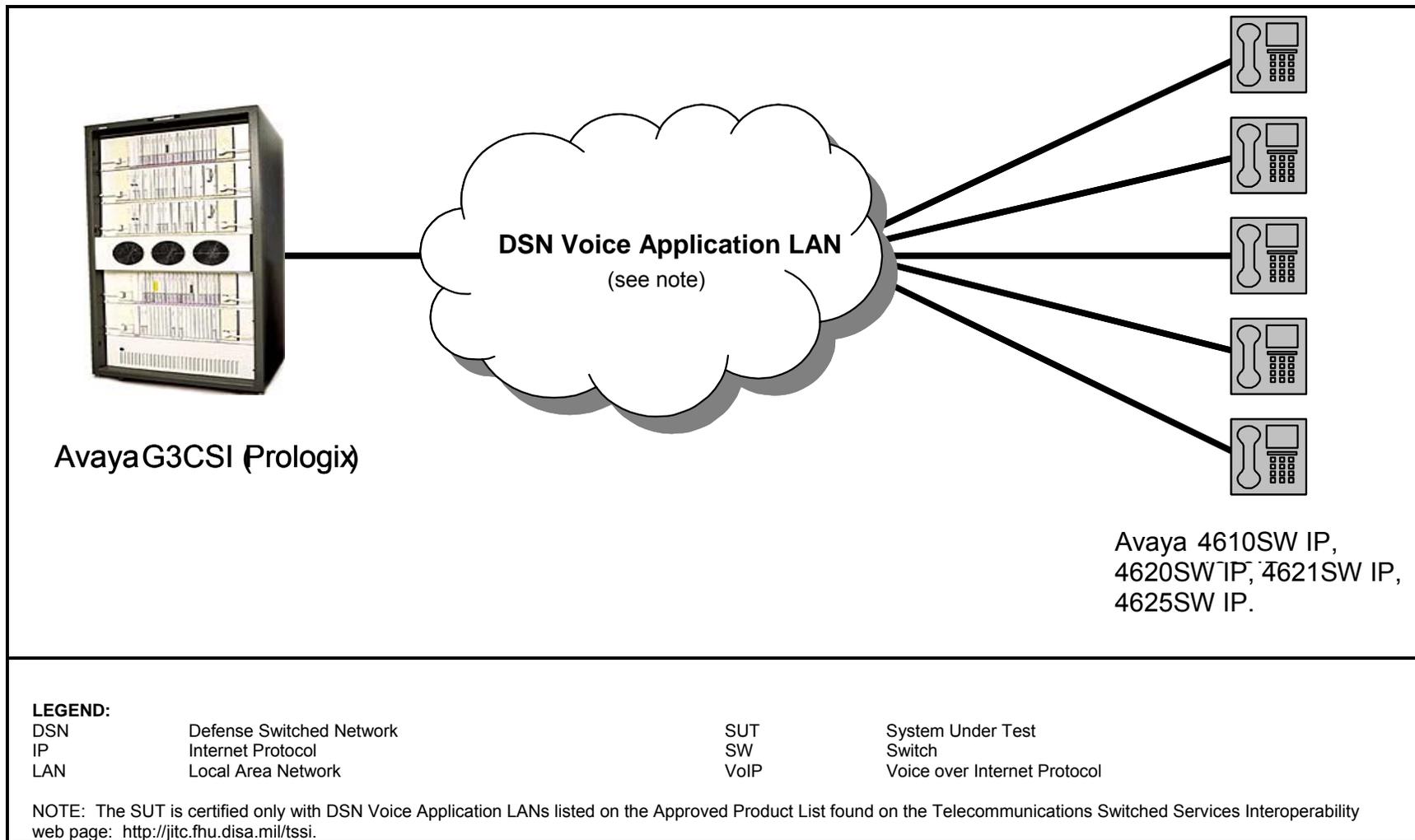


Figure 2-2. Notional Test Configuration



**Figure 2-3. SUT VoIP Test Diagram**

**9. SYSTEM CONFIGURATIONS.** Table 2-2 provides the system configurations and the hardware and software components tested with the SUT. The certified DSN VA LAN hardware and software components are listed on the Telecom Switched Services Interoperability (TSSI) web page: <http://jitc.fhu.disa.mil/tssi>.

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

System Name		Software Release		
Nortel Networks MSL-100 (MFS, EO)		SE06		
Avaya S8700 (SMEO, PBX 1, PBX 2)		CM 2.1 (R012x.01.0.411.7)		
Siemens EWSD (MFS, EO)		19d with Patch Set 44		
Siemens HiPath 4000 (PBX 1, PBX 2)		V1.0 SMR12		
Lucent 5ESS (MFS, EO)		5E16.2 SU05-0005		
SMU 96 Tactical Gateway		RD302185		
Tekelec STP		23.1		
Nortel Networks Broad Band STP		3.0.3.18d		
DSS Red Switch		8.03		
MARCONI ATM switches		Versions 6.2 and 7.1		
Avaya G3CSI, ProLogix		Vintage	Software Release	
<b>SUT</b>	Processor TN2402	000004	CM 3.0 (R013i.00.0.340.5)	
	Maintenance TN771D	000007		
	Tone Clock Card TN2182C	000004		
	IP Media Processor Board TN2302AP	HW13 FW102		
	Announcement Card TN750C	000013		
	Control LAN Card TN799DP	HW01 FW131		
	Analog Card TN793B	000006		
	DS1 Interface Card TN464HP	HW00 FW117		
	DS1 Interface Card TN464GP	HW06 FW017		
	BRI Line Card TN556D	000001		
	DS1 Interface Card TN2464BP	HW05 FW017		
	Digital Line Card TN2224B	000003		
	<b>Telephones</b>			
	IP Phone – 4610SW IP			a10d01b2_2.bin
IP Phone – 4620 IP		a20d01a2_2.bin		
IP Phone – 4620SW IP		a20d01b2_2.bin		
IP Phone – 4621SW IP		a20d01b2_2.bin		
IP Phone – 4625SW IP		a25d01a2_5.bin		
<b>LEGEND:</b>				
5ESS	- Class 5 Electronic Switching System	Mbps	- Megabits per second	
ATM	- Asynchronous Transfer Mode	MFS	- Multifunction Switch	
BRI	- Basic Rate Interface	MSL	- Meridian Switching Load	
CM	- Communication Manager	PBX 1	- Private Branch Exchange 1	
DS1	- Digital Signal Level 1 (1.544 Mbps North American) (2.048 Mbps European)	PBX 2	- Private Branch Exchange 2	
DSS	- Digital Small Switch	SE	- Succession Enterprise	
EO	- End Office	SMEO	- Small End Office	
EWSD	- Elektronisches Wählsystem Digital	SMR	- System Maintenance Release	
FW	- Firmware	SMU	- Switch Multiplexer Unit	
HW	- Hardware	STP	- Signal Transfer Point	
IP	- Internet Protocol	SUT	- System Under Test	
LAN	- Local Area Network	SW	- Switch	

**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. Detailed trunk configurations and associated lessons learned can be found on the DISA web page: <http://jitc.fhu.disa.mil/>. The following minor exceptions are noted:

(a) The SUT acknowledges a wink start signal beyond the maximum 350 milliseconds (ms). On an E1 CAS the SUT correctly acknowledges a wink start. On the T1 CAS the SUT acknowledges a wink start signal up to 395ms. The operational impact is minor.

(b) The SUT does not support the correct length of preemption notification tone (PNT) when an active call is directly preempted. A three second PNT is sent to the party being preempted instead of sending PNT until the preempted party goes on hook in accordance with the GSCR. The operational impact is minor.

**(2) DSN Line Interfaces.** The SUT met all critical interoperability certification requirements for DSN Line Interfaces. Refer to table 2-2 for specific instrument models tested under this certification test.

**(3) Features and Functions.** The SUT met all critical interoperability certification requirements for Features and Functions with the following minor exceptions:

(a) The SUT does not support Priority Call Pickup of calls above ROUTINE. When precedence call above ROUTINE is ringing in a call pickup group and a ROUTINE call is also ringing in the same call pickup group, the SUT randomly picks which call to pickup when the feature is activated instead of the highest precedence call. Since the higher precedence call is diverted to an alternate directory number if unanswered, the operational impact is minor.

(b) The SUT does not support the classmarking of the two legs of a three-way call at different precedence levels. This is due to the fact that the SUT connects all three parties to a single time slot. Instead, the SUT classmarks all the parties at the highest precedence. The operational impact is minor.

(c) The SUT does not support the correct distinctive ring cadence in accordance with the GSCR for precedence above ROUTINE calls placed via a trunk. The operational impact is minor.

**(4) Network Gateways.** The SUT met all critical interoperability certification requirements for the following Network Gateways: Public Switched Telephone Network and Defense Red Switch Network.

**(5) VoIP.** The SUT VoIP solution comprises the G3CSI (Prologix) Time Division Multiplexing (TDM) circuit switch and the LAN as shown in figure 2-3. The LAN infrastructure consisted of a certified DSN VA LAN listed on the TSSI website: <http://jitc.fhu.disa.mil/tssi>. The results for the overall VoIP system and LAN, as defined by the GSCR, appendix 3, are presented below.

**(a) VoIP System.** GSCR, appendix 3, section A3.2, outlines the requirements for the VoIP system that encompass end-to-end VoIP requirements (i.e., encompassing both the circuit switch and LAN). The following paragraphs detail the results of the SUT VoIP solution.

**1. Voice Quality.** In accordance with the GSCR, appendix 3, VoIP calls shall have an average Mean Opinion Score (MOS) score of at least 4.0 as measured in accordance with DISR voice quality standard. The measured MOS over 65 intra-switch calls was 4.18, and the MOS for 80 inter-switch calls was measured at 4.20.

**2. Class of Service (CoS) and Quality of Service (QoS).** The GSCR, appendix 3, outlines several methodologies to implement CoS and QoS. 802.1p/Q at the Data Link Layer (L2) and Differentiated Services Code Point (DSCP) at the Network Layer (L3) were two CoS mechanisms that the certified network products employed. The SUT provides CoS by assignment of an 802.1p/Q tag. Switches within the topology were configured with multiple Virtual VLANs to separate data from voice traffic. 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice VLAN traffic was assigned to a high priority queue, ensuring voice traffic took precedence over data traffic. For DSCP, L2 signaling was set for 6 and L3 signaling was set for 48 in the tested configuration. By using the Ixia test equipment, a data load of 1.2 times the total link aggregate was inserted on the DSN VA LAN to insure that all CoS and QoS settings were working properly. CoS/QoS were met in accordance with the GSCR, appendix 3.

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

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

**a. Phones.** The only SUT phones to meet all requirements for certification were the 4610SW, 4620SW, 4620IP, 4621SW, and 4625SW. Although the phones are capable of shared access (i.e., same switch port is shared by Personal Computer (PC) and IP phone), the shared access was not tested and is not covered under this certification.

**b. Scalability.** The SUT can support up to 8 MedPro cards in a configuration, not to exceed the guidelines in appendix E of the GSCR for a single point

of failure (for any combination of analog, ISDN, and IP subscribers). However, the manufacturer recommendation for release 3.0 is not to exceed 900 users for 8 MedPro cards. The SUT LAN solution tested consisted of one certified DSN VA LAN as shown in figure 2-3. The DSN VA LAN can be scaled to meet the 900 total concurrent ports (all types, e.g., analog, ISDN, IP, and trunks) 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.

(i) The SUT's IP MedPro cards have a limitation in that they can only support 64 IP subscribers and still meet DSN assured connectivity requirements. To determine the number of MedPro cards per switch to meet the minimum interoperability requirements, the following formula must be used:

$$n = \text{Total number of Media Processor cards} = \text{total VoIP users} / 64.$$

For redundancy purposes, the number of MedPro cards shall be implemented on an **n+1** card basis (i.e., 64 users require 2 MedPro cards, 128 users require 3 MedPro cards, etc.).

(ii) To determine the number of C-LAN cards needed to support IP subscribers use:

$$n = \text{Total number of C-LAN cards} = \text{number of VoIP users} / 250.$$

This is based on the manufacturer recommendation that no more than 250 users per C-LAN card be assigned. C-LAN cards shall also be implemented on an **n+1** card basis to meet redundancy requirements (i.e., 250 users require 2 C-LAN cards, 500 users require 3 C-LAN cards, etc.).

**5. MLPP.** The GSCR, appendix 3, section A3.2.3, details the requirements for Multi-Level Precedence and Preemption. Currently there are no mature standards for implementing MLPP over Internet Protocol (IP), requiring the vendor to implement proprietary IP signaling. All critical MLPP features and functions were met by the SUT.

**(b) Security.** Security requirements in accordance with the GSCR, appendix 3, section A3.2.4, were verified using the Information Assurance Test Plan (IATP). Results of the security testing were reported in a separate test report generated by the DISA Information Assurance test personnel.

**(c) Network Management (NM).** The GSCR, appendix 3, section A3.2.5, defines the overall Network Management (NM) requirements that VoIP systems must meet. The SUT VoIP system met these NM requirements. The switching system NM requirements per the GSCR, section 9, were also met by the SUT.

**(d) Internal Clock.** The switching system Internal Clock requirements in accordance with the GSCR, section 11 were met.

**(e) Synchronization.** Synchronization is also required for VoIP and circuit switched systems. Synchronization per the GSCR, section 11, was met by the SUT. The SUT derived synchronization with line timing mode via traditional TDM based interfaces (i.e., T1 or E1 digital).

**(f) Latency.** The GSCR, appendix 3, section A3.2.7, requires one-way system latency for the VoIP system is 60 milliseconds (ms) or less as averaged over any 5-minute period. The latency requirement is measured from IP handset to the egress trunk. The SUT average latency over 100 calls was measured to be 57.5 ms.

**(g) Packet Loss.** The GSCR, appendix 3, section A3.1.3, states packet loss shall not exceed 0.05% averaged over any five-minute period. The SUT packet loss was measured at 0.001%

**(h) Internet Protocol version 6 (IPv6).** The GSCR, appendix 3, section A3.2.8, states that the DSN LAN components must be IPv6 capable. The VoIP components provided did not support this requirement. The operational risk is minor due to the fact that IPv6 is not currently implemented within the DSN and is not scheduled to be fully implemented until 2008.

**b. System Interoperability Results.** The SUT with Software Release Version CM 3.0 (R013i.00.0.340.5) is certified for joint use in the DSN as a PBX 1 in accordance with the requirements set forth in the GSCR. The interoperability test summary is shown in table 2-3 and the detailed interoperability test status is shown table 2-4.

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

**Table 2-3. 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, MFR1, DP)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT acknowledges a wink start signal beyond the maximum 350 ms. <sup>1</sup> The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</sup>
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT acknowledges a wink start signal beyond the maximum 350 ms. <sup>1</sup> The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</sup>
T1 ISDN PRI NI 1/2 (ANSI T1. 619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support the correct length of PNT. <sup>2</sup> The SUT does not support distinctive ring cadence for ROUTINE and precedence above ROUTINE calls. <sup>3</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 CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP	No	Certified	Met all CRs and FRs.
<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 ERs and FRs with the following minor exceptions: The SUT does not support priority Call Pickup with calls above ROUTINE <sup>4</sup> and does not allow the two legs of a three-way call to be established at different precedence levels. <sup>5</sup>
Attendant	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
Public Safety	No	Certified	Met all critical CRs and FRs.
Preset Conferencing	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
Nailed-up Connections	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
PAT	No	Not Tested	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
DSN Hotline Services	No	Certified	Met all CRs and FRs.
Network Management	No	Not Tested	This capability is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs.
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 6.	
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6. <sup>7</sup>
VoIP LANs	No	Certified	Met all CRs and FRs.

**Table 2-3. SUT Interoperability Test Summary (continued)**

Network Gateways																																																																																
	Interface & Signaling	Critical	Status	Remarks																																																																												
PSTN	T1 CAS (DTMF, DP)	No	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.																																																																												
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified <sup>8</sup>	Met all critical CRs and FRs.																																																																												
<p><b>LEGEND:</b></p> <table border="0"> <tr> <td>ANSI</td> <td>- American National Standards Institute</td> <td>Mbps</td> <td>- Megabits per second</td> </tr> <tr> <td>BRI</td> <td>- Basic Rate Interface</td> <td>MFR1</td> <td>- Multifrequency Recommendation 1</td> </tr> <tr> <td>CAS</td> <td>- Channel Associated Signaling</td> <td>MLPP</td> <td>- Multi-Level Precedence and Preemption</td> </tr> <tr> <td>CRs</td> <td>- Capability Requirements</td> <td>ms</td> <td>- milliseconds</td> </tr> <tr> <td>DISA</td> <td>- Defense Information Systems Agency</td> <td>NI 1/2</td> <td>- National ISDN 1 or 2</td> </tr> <tr> <td>DITSCAP</td> <td>- Department of Defense Information Technology Security Certification and Accreditation Process</td> <td>PAT</td> <td>- Precedence Access Threshold</td> </tr> <tr> <td>DP</td> <td>- Dial Pulse</td> <td>PBX 1</td> <td>- Private Branch Exchange 1</td> </tr> <tr> <td>DRSN</td> <td>- Defense Red Switch Network</td> <td>PM</td> <td>- Program Manager</td> </tr> <tr> <td>DSN</td> <td>- Defense Switched Network</td> <td>PNT</td> <td>- Preemption Notification Tone</td> </tr> <tr> <td>DSS1</td> <td>- Digital Subscriber Signaling 1</td> <td>PRI</td> <td>- Primary Rate Interface</td> </tr> <tr> <td>DTMF</td> <td>- Dual Tone Multi-Frequency</td> <td>PSTN</td> <td>- Public Switched Telephone Network</td> </tr> <tr> <td>E1</td> <td>- European Basic Multiplex Rate (2.048 Mbps)</td> <td>SS7</td> <td>- Signaling System 7</td> </tr> <tr> <td>EKTS</td> <td>- Electronic Key Telephone System</td> <td>SUT</td> <td>- System Under Test</td> </tr> <tr> <td>FRs</td> <td>- Feature Requirements</td> <td>T1</td> <td>- Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>GR</td> <td>- Generic Requirement</td> <td>T1.607</td> <td>- ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1</td> </tr> <tr> <td>GSCR</td> <td>- Generic Switching Center Requirements</td> <td>T1.619a</td> <td>- SS7 and ISDN MLPP Signaling Standard For T1</td> </tr> <tr> <td>IPv6</td> <td>- Internet Protocol version 6</td> <td>TPC</td> <td>- Twisted Pair Copper</td> </tr> <tr> <td>ISDN</td> <td>- Integrated Services Digital Network</td> <td>VoIP</td> <td>- Voice over Internet Protocol</td> </tr> <tr> <td>LAN</td> <td>- Local Area Network</td> <td></td> <td></td> </tr> </table> <p><b>NOTES:</b></p> <ol style="list-style-type: none"> <li>1 The SUT acknowledges a wink start signal beyond the 350 ms maximum (approximately 390 ms). On an E1 CAS the SUT will correctly acknowledge a wink start signal. The operational impact is minor.</li> <li>2 The SUT does not support the correct length of PNT when an active call is directly preempted. A three second PNT is sent to the party being preempted instead of sending PNT until the preempted party goes on hook. The operational impact is minor.</li> <li>3 The SUT does not support the correct distinctive ring cadence in accordance with the GSCR for precedence above ROUTINE calls placed via a trunk. The operational impact is minor.</li> <li>4 The SUT does not support priority Call Pickup with precedence calls above ROUTINE. When a precedence call above ROUTINE is ringing in a call pickup group and a ROUTINE call is also ringing in the same call pickup group, the SUT randomly picks which call to pickup when the feature is activated. Since the higher precedence call is diverted to an alternate directory number if unanswered, the operational impact is minor.</li> <li>5 The SUT does not support the classmarking of the two legs of a three-way call at different precedence levels. This is due to the fact that the SUT connects all three parties to a single time slot. Instead, the SUT classmarks all the parties at the highest precedence. The operational impact is minor.</li> <li>6 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.</li> <li>7 The VoIP components provided did not support IPv6. The risk of not testing is minor due to the fact that IPv6 is not currently implemented within the DSN and is not scheduled to be fully implemented until 2008.</li> <li>8 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.</li> </ol>					ANSI	- American National Standards Institute	Mbps	- Megabits per second	BRI	- Basic Rate Interface	MFR1	- Multifrequency Recommendation 1	CAS	- Channel Associated Signaling	MLPP	- Multi-Level Precedence and Preemption	CRs	- Capability Requirements	ms	- milliseconds	DISA	- Defense Information Systems Agency	NI 1/2	- National ISDN 1 or 2	DITSCAP	- Department of Defense Information Technology Security Certification and Accreditation Process	PAT	- Precedence Access Threshold	DP	- Dial Pulse	PBX 1	- Private Branch Exchange 1	DRSN	- Defense Red Switch Network	PM	- Program Manager	DSN	- Defense Switched Network	PNT	- Preemption Notification Tone	DSS1	- Digital Subscriber Signaling 1	PRI	- Primary Rate Interface	DTMF	- Dual Tone Multi-Frequency	PSTN	- Public Switched Telephone Network	E1	- European Basic Multiplex Rate (2.048 Mbps)	SS7	- Signaling System 7	EKTS	- Electronic Key Telephone System	SUT	- System Under Test	FRs	- Feature Requirements	T1	- Digital Transmission Link Level 1 (1.544 Mbps)	GR	- Generic Requirement	T1.607	- ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1	GSCR	- Generic Switching Center Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard For T1	IPv6	- Internet Protocol version 6	TPC	- Twisted Pair Copper	ISDN	- Integrated Services Digital Network	VoIP	- Voice over Internet Protocol	LAN	- Local Area Network		
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**Table 2-4. SUT Interoperability Requirements/Status**

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

**Table 2-4. SUT Interoperability Requirements/Status**

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)		Reference	Test Results	Remarks
E1 CAS	No	Certified	Trunking	Framing (R)	GSCR Sect. 7	Met	
				Line Code (R)	GSCR Sect. 7	Met	
				Signaling (R)	GSCR Sect. 5	Met	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Met	
				WWNDP (R)	GSCR Sect. 4.5.1	Met	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Met	
				Routing (C)	GSCR Sect. 4.2	Met	
				Trunk Groups (C)	GSCR Sect. 2.5.5 & 2.5.6	Met	
				Call Processing (R)	GSCR Sect. 4	Met	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Met	
				PCM-24/PCM-30 Interoperation (C)	GSCR Sect. 7.3	Met	
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Met		
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Facsimile	Analog: EIA/TIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
				NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met					
VTC	ITU-T H.320 (R: ISDN PRI only)	DISR	Met				

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)		Reference	Test Results	Remarks
T1 ISDN PRI (ANSI T1.619a)	Yes	Certified	Trunking	Framing (R)	GSCR Sect. 7	Met	
				Line Code (R)	GSCR Sect. 7	Met	
				Signaling (R)	GSCR Sect. 5	Met	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Met	
				Timing (R)	GSCR Sect. 11.1.1.2	Met	
				WWNDP (R)	GSCR Sect. 4.5.1	Met	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Met	
				Routing (C)	GSCR Sect. 4.2	Met	
				Trunk Groups (C)	GSCR Sect. 2.5.5 & 2.5.6	Met	
				Call Processing (R)	GSCR Sect. 4	Met	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Met	
				PCM-24/PCM-30 Interoperation (C)	GSCR Sect. 7.3	Met	
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Met		
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Facsimile	Analog: EIA/TIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Met	
NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Met					
	Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met				
VTC	ITU-T H.320 (R: ISDN PRI only)	DISR	Met				

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

DSN Line Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)		Reference	Test Results	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Access	DN Identification (R)	GSCR Sect 2.1.1	Met	
				Line signaling (R)	GSCR Sect 5.2	Met	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Met	
				WWNDP (R)	GSCR Sect. 4.5	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
				2W user access (R)	GSCR Sect 4.3.3	Met	
				Analog busy/idle (R)	GSCR Sect 4.3.4.1	Met	
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Facsimile	Analog: EIA/TIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met	
			VTC	ITU-T H.320 (R: ISDN BRI only)	DISR	Met	

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

DSN Line Interfaces (continued)							
Interface	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)		Reference	Test Results	Remarks
ISDN BRI NI 1/2	No	Certified	Access	DN Identification (R)	GSCR Sect 2.1.1	Met	
				Line signaling (R)	GSCR Sect 5.2	Met	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Met	
				WWNDP (R)	GSCR Sect. 4.5	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
			Voice	MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Met	
				Secure calls (R)	CJCSI 6215.01B	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
				56-kbps switched data (R)	GSCR Sect. 3.10	Met	
				64-kbps switched data (R)	GSCR Sect. 3.10	Met	
				NX56 synchronous BER (R)	GSCR Sect. 3.10	Met	
				NX64 synchronous BER (R)	GSCR Sect. 3.10	Met	
				Secure data (STE/STU-III) (R)	GSCR Sect. 3.10	Met	
			VTC	ITU-T H.320 (R: ISDN BRI only)	DISR	Met	

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

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

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

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

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

DSN Features & Capabilities (continued)						
Features/ Capabilities	Critical	Status	GSCR Requirement Required (R) Conditional (C)	Reference	Test Results	Remarks
Network Management	No	Not Tested	Interfaces (C)	GSCR Sect. 9.1	Not Tested	See note 3.
			Measurements and data generation (C)	GSCR Sect. 9.2	Not Tested	See note 3.
			Fault management (C)	GSCR Sect. 9.3	Not Tested	See note 3.
			Configuration management (C)	GSCR Sect. 9.4	Not Tested	See note 3.
			Accounting management (C)	GSCR Sect. 9.5	Not Tested	See note 3.
			Performance management (C)	GSCR Sect. 9.6	Not Tested	See note 3.
			NM controls (C)	GSCR Sect. 9.7	Not Tested	See note 3.
Remote access (C)	GSCR Sect. 9.8	Not Tested	See note 3.			
ISDN Services	No	Certified	EKTS (C)	GSCR Sect. 10, table 10-3	Met	
Synchronization	Yes	Certified	Line timing mode (R)	GSCR Sect. 11.1.1.2	Met	
			Internal Stratum 4 (R)	GSCR Sect. 11.1.2.2	Met	
Reliability	Yes	Certified	GR-512-CORE (R)	GSCR Sect. 12	Met	
Security	Yes	See note 4.	DITSCAP (R)	GSCR Sect. 13	See note 4.	
VoIP System	No	Certified	ITU-T G.711 PCM Codec (R)	GSCR App. 3	Met	
			Security in accordance with DITSCAP (R)	GSCR App. 3	Met	
			NM (R)	GSCR App. 3	Met	
			Latency @ 60 msec or less (R)	GSCR App. 3	Met	
			IPv6 capable (R)	GSCR App. 3	Not Tested	See note 5.
LANs	No	Certified	LAN parameters (R)	GSCR App. 3	Met	
			CoS/QoS (R)	GSCR App. 3	Met	
			VLANs (R)	GSCR App. 3	Met	
			IEEE Standards Conformance (R)	GSCR App. 3	Met	
			.99999 availability (R)	GSCR App. 3	Met	
			Modular devices (R)	GSCR App. 3	Met	
			2 second link restoral (R)	GSCR App. 3	Met	
			LAN NM (R)	GSCR App. 3	Met	
			Traffic Engineering (R)	GSCR App. 3	Met	
Network Gateway						
Gateway	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)	Reference	Test Results	Remarks
PSTN	Yes	Certified	Trunking	Positive Identification Control (R)	CJCSI 6215.01B	Met
				On-Netting (R)	CJCSI 6215.01B	Met
				Off-Netting (R)	CJCSI 6215.01B	Met

**Table 2-4. SUT Interoperability Requirements/Status (continued)**

Network Gateway							
Gateway	Critical	Interface Status	GSCR Requirement Required (R) Conditional (C)		Reference	Test Results	Remarks
DRSN <sup>6</sup>	Yes	Certified	Access	Alerting Signals and Tones (R)	GSCR Sect. 5.5	Met	
				Call Processing (R)	GSCR Sect. 4.4	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
				Analog busy/idle (R)	GSCR Sect. 4.3.4.1	Met	
			Voice	MOS (C)	CJCSI 6215.01B	Met	
				MLPP (C)	GSCR Sect. 3	Met	
				Secure Calls (C)	CJCSI 6215.01B	Met	
				Alerting Signals and Tones (R)	GSCR Sect. 5.5	Met	
<b>LEGEND:</b> 2W - 2-Wire 911 - 911 Emergency Service A/D - Analog to Digital Conversion ANSI - American National Standards Institute App. - Appendix BER - Bit Error Ratio BRI - Basic Rate Interface C - Conditional CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCSI - Chairman of the Joint Chiefs of Staff Instruction CoS - Class of Service D/A - Digital to Analog Conversion DISA - Defense Information Systems Agency DISR - Department of Defense Information Technology Standards Registry DITSCAP - Department of Defense Information Technology Security and Accreditation Process DN - Directory Number DRSN - Defense Red Switch Network DSN - Defense Switched Network E1 - European Basic Multiplex Rate EIA - Electronic Industries Alliance EKTS - Electronic Key Telephone System G.711 - PCM of voice frequencies GR - Generic Requirement GSCR - Generic Switching Center Requirements H.320 - Standard for narrowband VTC IEEE - Institute of Electrical and Electronics Engineers, Inc. IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network 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 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score msec - milliseconds NI 1/2 - National ISDN Standard 1 or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network QoS - Quality of Service R - Required Sect. - Section SS7 - Signaling System 7 STE - Secure Terminal Equipment STU-III - Secure Telephone Unit-3 <sup>rd</sup> generation SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard For T1 TIA - Telecommunications Industry Association TIA-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 This feature or capability is not applicable to this interface. 2 This feature is not supported by the SUT. Since this is a conditional requirement, the risk of not testing is minor. 3 This feature is offered by the SUT but was not tested. Since this is a conditional requirement, the risk of not testing is minor. 4 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 5 The VoIP components provided did not support this requirement. The operational risk is minor due to the fact that IPv6 is not currently implemented within the DSN and is not scheduled to be fully implemented until 2008. 6 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.							