



## DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND

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IN REPLY  
REFER TO:

Networks and Transport Division (JTE)

October 16, 2006

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of the Avaya S8700 with Software Release Communication Manager (CM) 3.0 (R013x.00.0.340.3: Super Patch 11815) Digital Switching System including Voice over Internet Protocol (VoIP)

**References:** (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004  
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification. Additional references are provided in enclosure 1.
2. The Avaya S8700 with Software Release CM 3.0 (R013x.00.0.340.3: Super Patch 11815) Digital Switching System including VoIP is hereinafter referred to as the System Under Test (SUT). The SUT met all of its critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The identified test discrepancies shown in the Certification Testing Summary (enclosure 2), which remained open after Super Patch 11815 was applied and regression tested, have an overall minor operational impact. The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Small End Office (SMEO), Private Branch Exchange (PBX) 1, and PBX 2. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.
3. This finding is based on interoperability testing conducted by JITC and a review of the vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 24 October through 14 November 2005. Regression testing was conducted from 6 March through 21 April 2006. Review of the LoC was completed on 6 September 2006. Enclosure 2 documents the test results and describes the tested network and system configurations. System interoperability should be verified before deployment in an operational environment that varies significantly from the test environment.
4. The interoperability test summary of the SUT is contained in table 1. The SMEO required and conditional Capability Requirements (CRs) and Feature Requirements (FRs) are listed in table 2. This interoperability test status is based on the SUT's ability to meet:

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

**Table 1. SUT Interoperability Test Summary**

<b>DSN Trunk Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.
T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs.
E1 CAS (DTMF, MFR1, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs.
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 send an alerting message after preemption until party goes on hook. <sup>1</sup> The SUT, when connected to the Nortel MSL-100, fails to send a follow on service message (out of service). <sup>2</sup>
E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since it is not a critical requirement for a SMEO, the operational impact is minor.
<b>DSN Line Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
2-Wire Proprietary Digital	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
VoIP	No	Certified	The SUT met all critical CRs and FRs. <sup>6</sup>

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**Table 1. 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 critical CRs and FRs.	
Attendant	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT attendant console does not support the automatic recall feature. However, a subscriber through the use of a feature access code can initiate this feature. <sup>7</sup> The SUT does not support periodic Busy Override Tone when the Attendant Console is overriding a busy line condition. <sup>8</sup>	
Public Safety	Yes	Certified	Met all CRs and FRs.	
Preset Conferencing	No	Not Tested	See note 9.	
Nailed-up Connections	No	Not Tested	See note 9.	
PAT	No	Not Tested	See note 9.	
DSN Hotline Services	Yes	Certified	Met all critical CRs and FRs.	
Network Management	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services (EKTS)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: When an EKTS member is assigned to a MLHG a call to that EKTS member fails to ring the other EKTS members. <sup>10</sup>	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
Reliability	Yes	Certified	Met all critical CRs and FRs.	
Security	Yes	Not Tested	See note 11.	
<b>VoIP</b>				
<b>Features and Capabilities</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>	
VoIP System	Yes	Certified	Met all critical CRs and FRs. <sup>6</sup> The SUT is certified with any ASVALAN listed on the TSSI website at <a href="http://jitc.fhu.disa.mil/tssi">http://jitc.fhu.disa.mil/tssi</a> .	
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.
	T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all critical CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs.
	E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since it is not a critical requirement for a SMEO, the operational impact is minor.
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>12</sup>	Met all critical CRs and FRs.

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

<b>LEGEND:</b>	
ANSI	- American National Standards Institute
ASVALAN	- Assured Services Voice Application LAN
BRI	- Basic Rate Interface
CAS	- Channel Associated Signaling
CRs	- Capability Requirements
DISA	- Defense Information Systems Agency
DISR	- DoD IT Standards Registry
DN	- Directory Number
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	- LSSGR: Signaling for Analog Interfaces
GSCR	- Generic Switching Center Requirements
IPv4	- Internet Protocol version 4
IPv6	- Internet Protocol version 6
IT	- Information Technology
ISDN	- Integrated Services Digital Network
LAN	- Local Area Network
LSSGR	- Local Access and Transport Area (LATA) Switching System Generic Requirements
Mbps	- Megabits per second
MFR1	- Multifrequency Recommendation 1
MLHG	- Multi-Line Hunt Group
MLPP	- Multi-Level Precedence and Preemption
MSL	- Meridian Switching Load
NI 1/2	- National ISDN Standard 1 or 2
PAT	- Precedence Access Threshold
PM	- Program Manager
PRI	- Primary Rate Interface
PSTN	- Public Switched Telephone Network
SMEO	- Small End Office
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 does not send an alerting message over ANSI T1.619a ISDN T1 PRI until the preempted SUT subscriber goes on hook. The failure of the SUT to send an alerting message can result in the DN switch timing out and disconnecting the call due to no response (no alerting message) from the SUT. The timeout is approximately 8-10 seconds and can be adjusted with longer timeout. The SUT does send an alerting message when the preempted SUT subscriber hangs up promptly. Since the user hangs up on average within three seconds of being preempted and the timeout can be adjusted, the operational impact is minor.
2	The SUT fails to send a follow on service message (out of service) when connected to the Nortel MSL-100 via an ANSI T1.619a ISDN T1 PRI. In the case where the SUT places 18 or more channels on this particular interface in a maintenance busy state, the MSL-100 will send a Restart message after 2 minutes to check the status of the trunk(s) and the SUT responds with a Restart Acknowledgement message and no follow on service message (out of service) message. When the Nortel MSL-100 fails to receive the service message (out of service) from the SUT, the MSL-100 marks the channel(s) back in service. This anomaly only occurs with the MSL-100 and can be mitigated by placing channels in a maintenance busy condition at the MSL-100 vice the SUT. The operational impact is minor.
3	Met all DSN Announcement requirements except for Isolation Code Announcement. The SUT provides this announcement only for precedence calls above ROUTINE. ROUTINE precedence calls receive a T60 slow busy treatment. The operational impact is minor.
4	The SUT does not allow classmarking of each separate connection of a three-way call at different precedence levels. When a three-way call is established, the GSCR requires that "each connection shall maintain its assigned precedence level." The SUT, however, connects a three-way call in a single time slot and classmarks all parties at the highest precedence level of the two connections. Since all members of the conference are classmarked at the highest precedence of each connection there is no operational impact.
5	The SUT call pickup feature doesn't retrieve the call with the highest precedence first when more than one party is ringing within the pickup group. The calls are answered in a random sequence. Since unanswered calls above ROUTINE are diverted to the attendant, night service, or an alternate DN, the operational impact is minor.
6	An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008: (a) Conformance with IPv6 standards profile contained in the 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.
7	The SUT attendant console does not support the automatic recall feature. However, a subscriber can initiate this feature through the use of a feature access code. The operational impact is minor.
8	The SUT does not fully support Busy Override Tone when the Attendant Console is overriding a busy line condition. The attendant provides an initial tone before entering a call but does not periodically continue to provide the tone. The operational impact is minor.
9	This feature is not supported. There is no operational impact because it is not a critical requirement.
10	When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members. When a call is sent to a MLHG pilot number that causes an EKTS member to ring, all members of the EKTS group should have an incoming call appearance. The EKTS feature is certified as standalone and not when assigned as a member of a MLHG. MLHG interaction with EKTS is a conditional requirement; therefore, the operational impact is minor.
11	Security is tested by DISA-led Information Assurance test teams and published in a separate report.
12	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. SMEO Requirements**

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

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

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Common Features	Yes	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer and conference calling and call hold (C)</li> <li>• Call forwarding (C)</li> <li>• Call pick-up (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.2</li> <li>• GSCR Sect. 2.1.3</li> <li>• GSCR Sect. 2.1.4</li> <li>• GSCR Sect. 2.1.5</li> <li>• GSCR Sect. 2.1.6</li> <li>• GSCR Sect. 2.1.7</li> <li>• GSCR Sect. 2.1.8</li> <li>• GSCR Sect. 2.1.9</li> </ul>
Attendant	No	<ul style="list-style-type: none"> <li>• Initiate all precedence levels (C)</li> <li>• Visual display (C)</li> <li>• Override class of service (C)</li> <li>• Override busy line (C)</li> <li>• Call deflection (C)</li> <li>• Auto recall (C)</li> <li>• Waiting queue (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.2.1</li> <li>• GSCR Sect. 2.2.2</li> <li>• GSCR Sect. 2.2.3</li> <li>• GSCR Sect. 2.2.4</li> <li>• GSCR Sect. 2.2.5</li> <li>• GSCR Sect. 2.2.6</li> <li>• GSCR Sect. 2.2.7</li> </ul>
Public Safety	Yes	<ul style="list-style-type: none"> <li>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (R)</li> <li>• Outgoing call trace (R)</li> <li>• Tandem call trace (R)</li> <li>• Trace of a call in progress (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.4.1</li> <li>• GSCR Sect. 2.4.2</li> <li>• GSCR Sect. 2.4.3</li> <li>• GSCR Sect. 2.4.4</li> <li>• GSCR Sect. 2.4.5</li> </ul>
Preset Conferencing	No	<ul style="list-style-type: none"> <li>• Support 10 bridges; 1 originator and 20 conferees per bridge (C)</li> <li>• Assign up to 20 address numbers per bridge (C)</li> <li>• Use KXX codes for bridge access (C)</li> <li>• Conference notification recorded announcement (C)</li> <li>• Auto retrieval and alternate address (C)</li> <li>• Bridge release (C)</li> <li>• Lost connection (C)</li> <li>• Secondary conferencing (C)</li> <li>• Address translation (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6.1</li> <li>• GSCR Sect. 2.6.2</li> <li>• GSCR Sect. 2.6.3</li> <li>• GSCR Sect. 2.6.4</li> <li>• GSCR Sect. 2.6.5</li> <li>• GSCR Sect. 2.7</li> </ul>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Between any two like terminations (C)</li> <li>• PCM-24 and PCM-30, both CAS and CCS (C)</li> <li>• Supervision passed end-to-end for A/D or D/A (C)</li> <li>• Monitored and auto reconfigure (C)</li> <li>• Support at least 10% of circuits as nailed-up (C)</li> <li>• Non-preemptable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.8</li> </ul>
PAT	No	<ul style="list-style-type: none"> <li>• Classmark for/not for PAT screening (C)</li> <li>• 7 PAT mechanisms (C)</li> <li>• Outgoing call screening (C)</li> <li>• Functional structure (C)</li> <li>• Simultaneous calls limitation (C)</li> <li>• Overflow process (C)</li> <li>• Decrementing call-in-progress count (C)</li> <li>• Call treatment (C)</li> <li>• Queuing (C)</li> <li>• Attendant calls (C)</li> <li>• Operations measurement registers (C)</li> <li>• Maintenance and Administration of thresholds (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1.1</li> <li>• GSCR Sect. 2.11.1.2</li> <li>• GSCR Sect. 2.11.1.3</li> <li>• GSCR Sect. 2.11.1.4</li> <li>• GSCR Sect. 2.11.1.5</li> <li>• GSCR Sect. 2.11.1.6</li> <li>• GSCR Sect. 2.11.1.7</li> <li>• GSCR Sect. 2.11.1.8</li> <li>• GSCR Sect. 2.11.1.9</li> <li>• GSCR Sect. 2.11.1.10</li> </ul>
DSN Hotline Services	Yes	<ul style="list-style-type: none"> <li>• Hotline restrictions (R)</li> <li>• Auto initiate (R)</li> <li>• Analog and digital (R)</li> <li>• Subscription basis (R)</li> <li>• Protected hotline calling (R)</li> <li>• WWNDP interoperable (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12.1-4</li> <li>• GSCR Sect. 2.12.5</li> </ul>

JITC Memo, JTE, Special Interoperability Test Certification of the Avaya S8700 with Software Release Communication Manager (CM) 3.0 (R013x.00.0.340.3: Super Patch 11815) Digital Switching System including Voice over Internet Protocol (VoIP)

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

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Network Management	Yes	<ul style="list-style-type: none"> <li>• Interfaces (R)</li> <li>• Measurements and data generation (R)</li> <li>• Fault management (R)</li> <li>• Configuration management (R)</li> <li>• Accounting management (R)</li> <li>• Performance management (R)</li> <li>• NM controls (R)</li> <li>• Remote access (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 9.1</li> <li>• GSCR Sect. 9.2</li> <li>• GSCR Sect. 9.3</li> <li>• GSCR Sect. 9.4</li> <li>• GSCR Sect. 9.5</li> <li>• GSCR Sect. 9.6</li> <li>• GSCR Sect. 9.7</li> <li>• GSCR Sect. 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• EKTS (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 10, table 10-3</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 11.1.1.2</li> <li>• GSCR Sect. 11.1.2.2</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• GR-512-CORE (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 12</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DIACAP (formerly known as DITSCAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 13</li> </ul>
<b>VoIP</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
VoIP System	No	<p>VoIP function is conditional. If VoIP is provided, <b>all</b> of the following requirements must be met:</p> <ul style="list-style-type: none"> <li>• Voice Quality with 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 ms</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR App. 3</li> <li>• GSCR App. 3, paragraph 1.7</li> </ul>

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

Network Gateways					
Gateway	Critical	Requirements Required or Conditional		References	
PSTN <sup>1</sup>	Yes	Trunking	<ul style="list-style-type: none"> <li>Positive Identification Control (R)</li> <li>On-Netting (R)</li> <li>Off-Netting (R)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01B</li> <li>CJCSI 6215.01B</li> <li>CJCSI 6215.01B</li> </ul>	
DRSN <sup>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 Sect. 5.5</li> <li>GSCR Sect. 4.4</li> <li>GSCR Sect. 4.1</li> <li>GSCR Sect. 4.3.4.1</li> </ul>	
		Voice	<ul style="list-style-type: none"> <li>MOS (C)</li> <li>MLPP (C)</li> <li>Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01B</li> <li>GSCR Sect. 3</li> <li>CJCSI 6215.01B</li> </ul>	
<b>LEGEND:</b>					
2W	- 2-Wire	EIA	- Electronic Industries Alliance	NX64	- Data format restricted to multiples of 64 kbps
A/D	- Analog to Digital Conversion	G.711	- Standard for PCM of Voice Frequencies	PAT	- Precedence Access Threshold
ANSI	- American National Standards Institute	GR	- Generic Requirement (Telcordia)	PCM	- Pulse Code Modulation
App.	- Appendix	GR-512	- LSSGR: Reliability, Section 12	PCM-24	- Pulse Code Modulation - 24 Channels
BER	- Bit Error Ratio	GR-815	- Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-30	- Pulse Code Modulation - 30 Channels
BRI	- Basic Rate Interface		- Generic Switching Center Requirements	PRI	- Primary Rate Interface
C	- Conditional	GSCR	- Generic Switching Center Requirements	PSTN	- Public Switched Telephone Network
CAS	- Channel Associated Signaling	H.320	- Standard for Narrowband VTC	Q.735.3	- SS7 Signaling Standard for E1 MLPP
CCS	- Common Channel Signaling	IPv6	- Internet Protocol version 6	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCS	- Chairman of the Joint Chiefs of Staff	ISDN	- Integrated Services Digital Network	R	- Required
CJCSI	- CJCS Instruction	IT	- Information Technology	Sect.	- Section
D/A	- Digital to Analog Conversion	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	SMEO	- Small End Office
DIACAP	- DoD Information Assurance Certification and Accreditation Process	LAN	- Local Area Network	SS7	- Signaling System 7
DISR	- DoD IT Standards Registry	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements	STE	- Secure Terminal Equipment
DITSCAP	- DoD IT Security Certification and Accreditation Process	kbps	- kilobits per second	STIGs	- Security Technical Implementation Guides
DN	- Directory Number	KXX	- K= any number 2-8; X= any number 1-9	STU-III	- Secure Telephone Unit - 3 <sup>rd</sup> Generation
DoD	- Department of Defense	Mbps	- Megabits per second	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DP	- Dial Pulse	MFR1	- Multi-Frequency Recommendation 1	T1.619a	- SS7 and ISDN Signaling Standard for T1
DSN	- Defense Switched Network	MLPP	- Multi-Level Precedence and Preemption	TIA	- Telecommunications Industry Association
DRSN	- Defense Red Switch Network	MOS	- Mean Opinion Score	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DTMF	- Dual Tone Multi-Frequency	ms	- milliseconds	VBD	- Variable bit data
E1	- European Basic Multiplex Rate (2.048 Mbps)	NI 1/2	- National ISDN Standard 1or 2	VoIP	- Voice over Internet Protocol
EKTS	- Electronic Key Telephone System	NM	- Network Management	VTC	- Video Teleconferencing
		NX56	- Data format restricted to multiples of 56 kbps	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.					

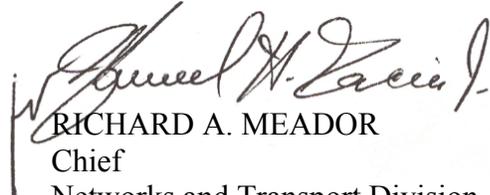
5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive 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 Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>.

JITC Memo, JTE, Special Interoperability Test Certification of the Avaya S8700 with Software Release Communication Manager (CM) 3.0 (R013x.00.0.340.3: Super Patch 11815) Digital Switching System including Voice over Internet Protocol (VoIP)

6. The JITC point of contact is Mr. Joseph Schulte, DSN 879-5164, commercial (520) 538-5164, FAX DSN 879-4347, or e-mail to joseph.schulte@disa.mil. The tracking number for the SUT is 42191.

FOR THE COMMANDER:

2 Enclosures a/s



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## **ADDITIONAL REFERENCES**

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (d) Defense Information Systems Agency (DISA), "Defense Switched Network (DSN) New Generic Switching Center Requirements (GSCR), 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.** Avaya S8700 with software release Communication Manager (CM) 3.0 (R013x.00.0.340.3: Super Patch 11815) Digital Switching System including Voice over Internet Protocol (VoIP) is hereinafter referred to as the System Under Test (SUT).

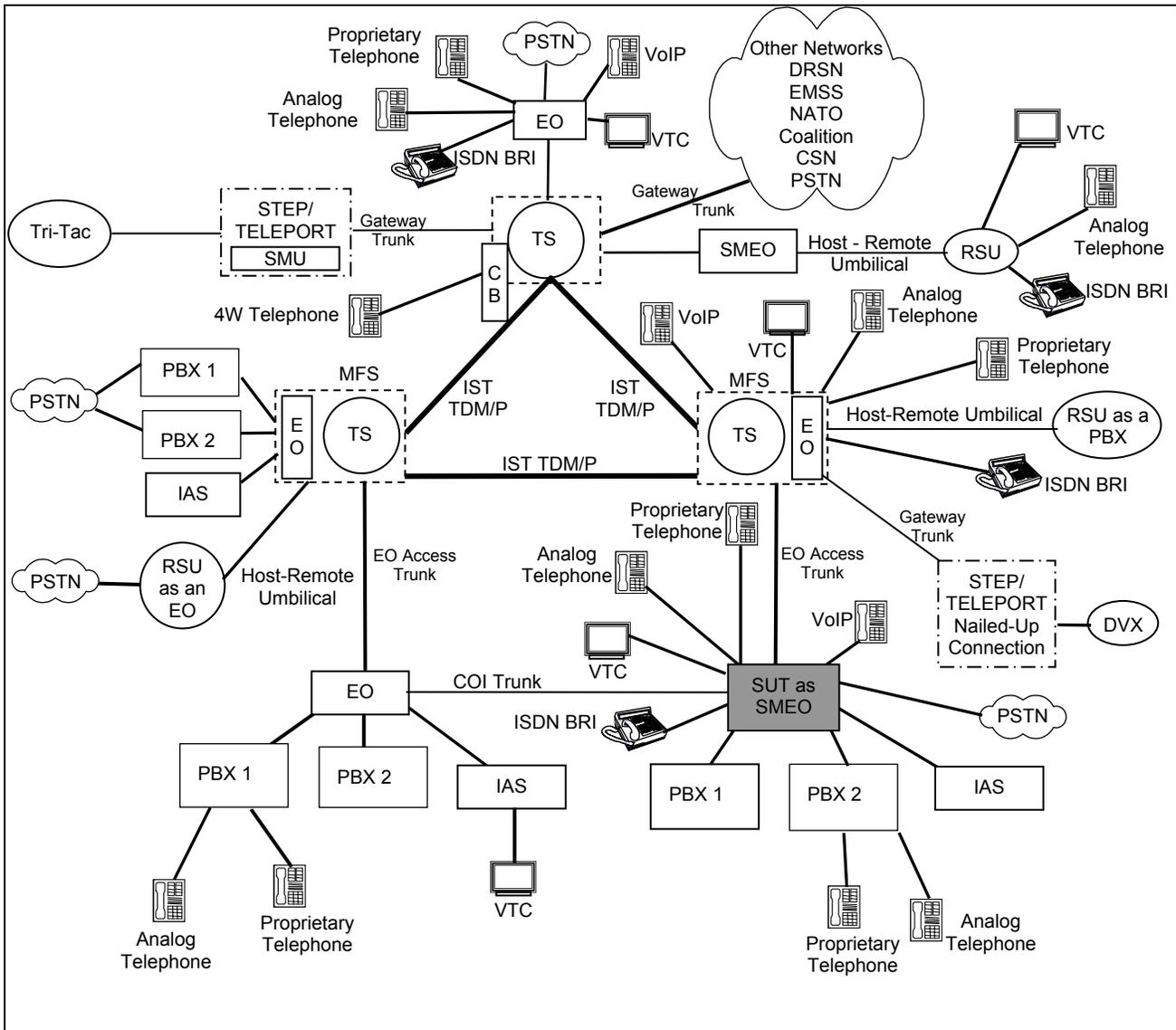
**2. PROPONENTS.** Defense Information Systems Agency (DISA).

**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 19-inch rack mount, Intel Processor-based server, running the Linux operating system. The SUT uses two processors which can be separated up to 10 kilometers (6.3 miles). The SUT can process up to 300,000 Busy Hour Call Completions and support up to 36,000 stations, of which 12,000 can be Internet Protocol (IP) stations, and 8,000 trunks. The SUT configuration tested included two servers (one server in hot standby mode) running the REDHAT 8.0 Linux Operating System and three peripheral cabinets: Multi-Carrier Cabinet (MCC), Single-Carrier Cabinet (SCC), and the G650 that were all interconnected using multi-mode fiber optic. The three cabinets served as media gateways and each supported the same peripheral cards. The SUT offers a VoIP capability that was successfully tested and is covered by this certification. Avaya's S8700 digital switching system is currently in use within the Defense Information System Network providing Small End Office (SMEO) Switch and Private Branch Exchange (PBX) functionality.

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



**LEGEND:**

4W	- 4-Wire	PBX	- Private Branch Exchange
BRI	- Basic Rate Interface	PBX 1	- Private Branch Exchange 1
CB	- Channel Bank	PBX 2	- Private Branch Exchange 2
COI	- Community of Interest	PSTN	- Public Switched Telephone Network
CSN	- Canadian Switch Network	RSU	- Remote Switching Unit
DRSN	- Defense Red Switch Network	SMEO	- Small End Office
DSN	- Defense Switched Network	SMU	- Switched Multiplex Unit
DVX	- Deployable Voice Exchange	STEP	- Standardized Tactical Entry Point
EMSS	- Enhanced Mobile Satellite System	SUT	- System Under Test
EO	- End Office	TDM/P	- Time Division Multiplex/Packetized
IAS	- Integrated Access Switch	Tri-Tac	- Tri-Service Tactical Communications Program
ISDN	- Integrated Services Digital Network	TS	- Tandem Switch
IST	- Interswitch Trunk	VoIP	- Voice over Internet Protocol
MFS	- Multifunction Switch	VTC	- Video Teleconferencing
NATO	- North Atlantic Treaty Organization		

**Figure 2-1. DSN Architecture**

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

- a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, “Policy for Department of Defense Voice Services.”
- b. GSCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC).
- c. GSCR SMEO Capability 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 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.

**Table 2-1. SMEO Requirements**

DSN Line Interfaces				
Interface	Critical	Requirements Required or Conditional		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line signaling (R)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Alerting Signals and Tones (R)</li> <li>• WWNDP (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• 2W user access (R: 2-Wire Analog only)</li> <li>• Analog busy/idle (R: 2-Wire Analog only)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.1</li> <li>• GSCR Sect. 5.2</li> <li>• GSCR Sect. 5.2.1</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.3</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes		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>
2W Digital Proprietary	No	Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
VoIP	No	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R)</li> <li>• 64 kbps switched data (R: BRI only)</li> <li>• NX56 synchronous BER (R: BRI only)</li> <li>• NX64 synchronous BER (R: BRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
		VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: BRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>

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

<b>DSN Trunk Interfaces</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
T1 SS7 (ANSI T1.619a)	No	Trunking	<ul style="list-style-type: none"> <li>• Framing (R)</li> <li>• Line Code (R)</li> <li>• Signaling (R)</li> <li>• Alarms (R)</li> <li>• WWNDP (R)</li> <li>• Outputting digit formats (R: CAS only)</li> <li>• Routing (R)</li> <li>• Trunk Groups (R)</li> <li>• Call Processing (R)</li> <li>• CAS to CCS trunk interworking (C)</li> <li>• PCM-24/PCM-30 Interoperation (R)</li> <li>• Direct Inward Dialing (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 7</li> <li>• GSCR Sect. 5</li> <li>• GSCR Sect. 2.5.7, 7.1.4 &amp; 7.2.2</li> <li>• GSCR Sect. 4.5.1</li> <li>• GSCR Sect. 4.5.2</li> <li>• GSCR Sect. 4.2</li> <li>• GSCR Sect. 2.5.5 &amp; 2.5.6</li> <li>• GSCR Sect. 4</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 7.3</li> <li>• GSCR Sect. 2.3.2</li> </ul>	
E1 SS7 (ITU-T Q.735.3)	No (Europe only)				
T1 CAS (MFR1)	No				
T1 CAS (DTMF, DP)	Yes		Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
E1 CAS (MFR1, DTMF, DP)	Yes (Europe only)		Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• GSCR Sect. 3.10</li> <li>• CJCSI 6215.01B</li> </ul>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> <li>• ITU-T H.320 (R: PRI only)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>	
<b>DSN Features &amp; Capabilities</b>					
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
Common Features	Yes	<ul style="list-style-type: none"> <li>• Selective call rejection (C)</li> <li>• Denied originating service (C)</li> <li>• Code restriction and diversion (R)</li> <li>• Call waiting (C)</li> <li>• Three-way calling (C)</li> <li>• Add-on transfer and conference calling and call hold (C)</li> <li>• Call forwarding (C)</li> <li>• Call pick-up (C)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.1.2</li> <li>• GSCR Sect. 2.1.3</li> <li>• GSCR Sect. 2.1.4</li> <li>• GSCR Sect. 2.1.5</li> <li>• GSCR Sect. 2.1.6</li> <li>• GSCR Sect. 2.1.7</li> <li>• GSCR Sect. 2.1.8</li> <li>• GSCR Sect. 2.1.9</li> </ul>	
Attendant	No	<ul style="list-style-type: none"> <li>• Initiate all precedence levels (C)</li> <li>• Visual display (C)</li> <li>• Override class of service (C)</li> <li>• Override busy line (C)</li> <li>• Call deflection (C)</li> <li>• Auto recall (C)</li> <li>• Waiting queue (C)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.2.1</li> <li>• GSCR Sect. 2.2.2</li> <li>• GSCR Sect. 2.2.3</li> <li>• GSCR Sect. 2.2.4</li> <li>• GSCR Sect. 2.2.5</li> <li>• GSCR Sect. 2.2.6</li> <li>• GSCR Sect. 2.2.7</li> </ul>	
Public Safety	Yes	<ul style="list-style-type: none"> <li>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (R)</li> <li>• Outgoing call trace (R)</li> <li>• Tandem call trace (R)</li> <li>• Trace of a call in progress (R)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. 2.4.1</li> <li>• GSCR Sect. 2.4.2</li> <li>• GSCR Sect. 2.4.3</li> <li>• GSCR Sect. 2.4.4</li> <li>• GSCR Sect. 2.4.5</li> </ul>	

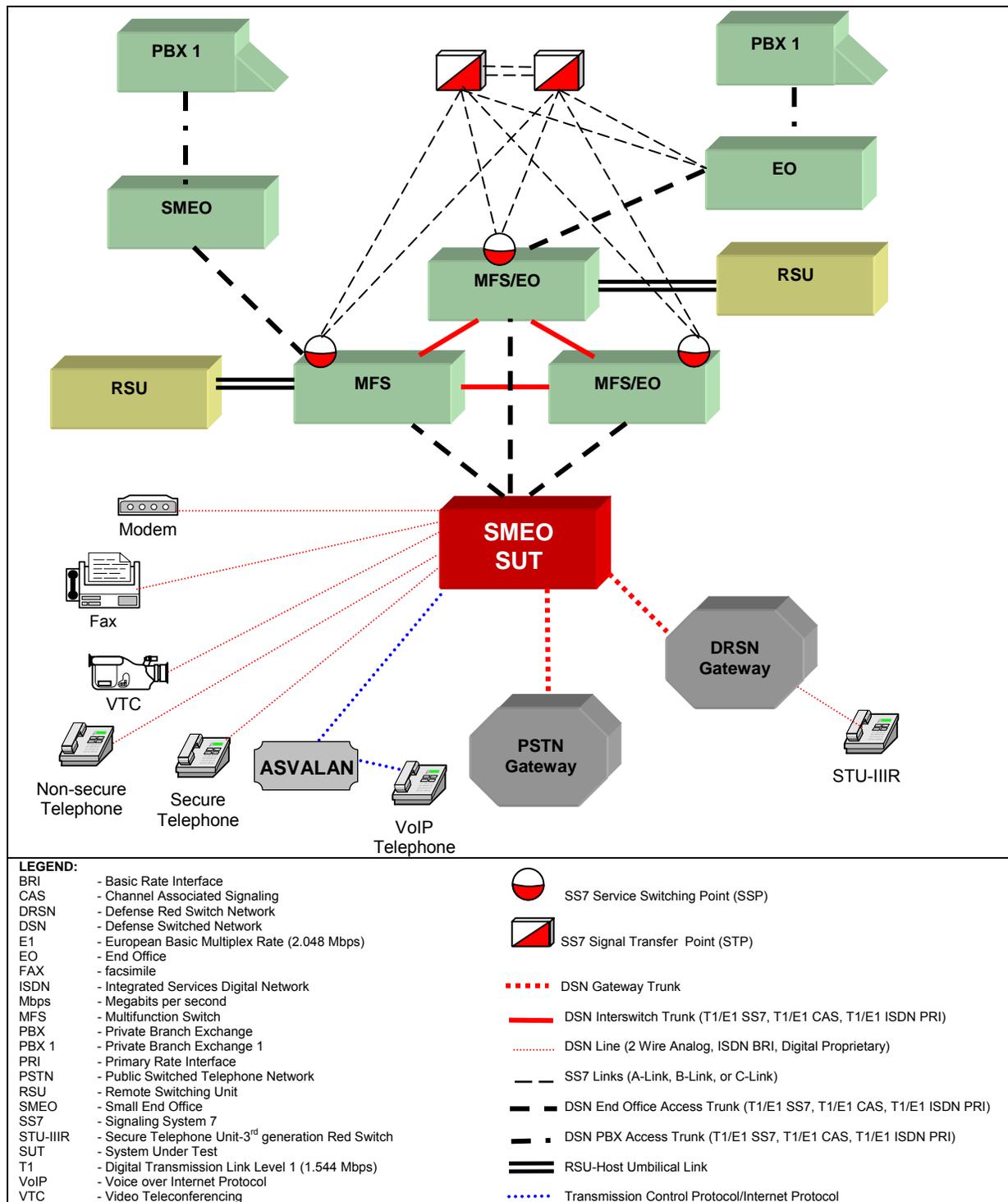
**Table 2-1. SMEO Requirements (continued)**

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Preset Conferencing	No	<ul style="list-style-type: none"> <li>• Support 10 bridges; 1 originator and 20 conferees per bridge (C)</li> <li>• Assign up to 20 address numbers per bridge (C)</li> <li>• Use KXX codes for bridge access (C)</li> <li>• Conference notification recorded announcement (C)</li> <li>• Auto retrial and alternate address (C)</li> <li>• Bridge release (C)</li> <li>• Lost connection (C)</li> <li>• Secondary conferencing (C)</li> <li>• Address translation (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6</li> <li>• GSCR Sect. 2.6.1</li> <li>• GSCR Sect. 2.6.2</li> <li>• GSCR Sect. 2.6.3</li> <li>• GSCR Sect. 2.6.4</li> <li>• GSCR Sect. 2.6.5</li> <li>• GSCR Sect. 2.7</li> </ul>
Nailed-up Connections	No	<ul style="list-style-type: none"> <li>• Between any two like terminations (C)</li> <li>• PCM-24 and PCM-30, both CAS and CCS (C)</li> <li>• Supervision passed end-to-end for A/D or D/A (C)</li> <li>• Monitored and auto reconfigure (C)</li> <li>• Support at least 10% of circuits as nailed-up (C)</li> <li>• Non-preemptable (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.8</li> </ul>
PAT	No	<ul style="list-style-type: none"> <li>• Classmark for/not for PAT screening (C)</li> <li>• 7 PAT mechanisms (C)</li> <li>• Outgoing call screening (C)</li> <li>• Functional structure (C)</li> <li>• Simultaneous calls limitation (C)</li> <li>• Overflow process (C)</li> <li>• Decrementing call-in-progress count (C)</li> <li>• Call treatment (C)</li> <li>• Queuing (C)</li> <li>• Attendant calls (C)</li> <li>• Operations measurement registers (C)</li> <li>• Maintenance and Administration of thresholds (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1</li> <li>• GSCR Sect. 2.11.1.1</li> <li>• GSCR Sect. 2.11.1.2</li> <li>• GSCR Sect. 2.11.1.3</li> <li>• GSCR Sect. 2.11.1.4</li> <li>• GSCR Sect. 2.11.1.5</li> <li>• GSCR Sect. 2.11.1.6</li> <li>• GSCR Sect. 2.11.1.7</li> <li>• GSCR Sect. 2.11.1.8</li> <li>• GSCR Sect. 2.11.1.9</li> <li>• GSCR Sect. 2.11.1.10</li> </ul>
DSN Hotline Services	Yes	<ul style="list-style-type: none"> <li>• Hotline restrictions (R)</li> <li>• Auto initiate (R)</li> <li>• Analog and digital (R)</li> <li>• Subscription basis (R)</li> <li>• Protected hotline calling (R)</li> <li>• WWNDP interoperable (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12</li> <li>• GSCR Sect. 2.12.1-4</li> <li>• GSCR Sect. 2.12.5</li> </ul>
Network Management	Yes	<ul style="list-style-type: none"> <li>• Interfaces (R)</li> <li>• Measurements and data generation (R)</li> <li>• Fault management (R)</li> <li>• Configuration management (R)</li> <li>• Accounting management (R)</li> <li>• Performance management (R)</li> <li>• NM controls (R)</li> <li>• Remote access (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 9.1</li> <li>• GSCR Sect. 9.2</li> <li>• GSCR Sect. 9.3</li> <li>• GSCR Sect. 9.4</li> <li>• GSCR Sect. 9.5</li> <li>• GSCR Sect. 9.6</li> <li>• GSCR Sect. 9.7</li> <li>• GSCR Sect. 9.8</li> </ul>
ISDN Services	No	<ul style="list-style-type: none"> <li>• EKTS (C)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 10, table 10-3</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (R)</li> <li>• Internal Stratum 4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 11.1.1.2</li> <li>• GSCR Sect. 11.1.2.2</li> </ul>
Reliability	Yes	<ul style="list-style-type: none"> <li>• GR-512-CORE (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 12</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DIACAP (formerly known as DITSCAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 13</li> </ul>

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

VoIP				
Feature/ Capability	Critical	Requirements Required or Conditional		References
VoIP System	No	VoIP function is conditional. If VoIP is provided, all of the following requirements must be met: <ul style="list-style-type: none"> <li>• Voice Quality with MOS of 4.0 or better</li> <li>• CoS and 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 ms</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR App. 3</li> <li>• GSCR App. 3, paragraph 1.7</li> </ul>
Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN <sup>1</sup>	Yes	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (R)</li> <li>• On-Netting (R)</li> <li>• Off-Netting (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> </ul>
DRSN <sup>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 Sect. 5.5</li> <li>• GSCR Sect. 4.4</li> <li>• GSCR Sect. 4.1</li> <li>• GSCR Sect. 4.3.4.1</li> </ul>
		Voice	<ul style="list-style-type: none"> <li>• MOS (C)</li> <li>• MLPP (C)</li> <li>• Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
<b>LEGEND:</b> 2W - 2-Wire A/D - Analog to Digital Conversion ANSI - American National Standards Institute App. - Appendix BER - Bit Error Ratio BRI - Basic Rate Interface C - Conditional CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCS - Chairman of the Joint Chiefs of Staff CJCSI - CJCS Instruction D/A - Digital to Analog Conversion DIACAP - DoD Information Assurance Certification and Accreditation Process DISR - DoD IT Standards Registry DITSCAP - DoD IT Security Certification and Accreditation Process DN - Directory Number DoD - Department of Defense DP - Dial Pulse DSN - Defense Switched Network DRSN - Defense Red Switch Network DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EKTS - Electronic Key Telephone System EIA - Electronic Industries Alliance G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement (Telcordia) 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 LAN - Local Area Network LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score ms - milliseconds NI 1/2 - National ISDN Standard 1or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.735.3 - SS7 Signaling Standard for E1 MLPP Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required Sect. - Section SMEO - Small End Office SS7 - Signaling System 7 STE - Secure Terminal Equipment STIGs - Security Technical Implementation Guides STU-III - Secure Telephone Unit – 3 <sup>rd</sup> Generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-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. This test was conducted using the test configuration shown in figure 2-2.



**Figure 2-2. SUT Test Configuration**

**9. SYSTEM CONFIGURATIONS.** Table 2-2 provides the system configurations, hardware and software components tested with the SUT.

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

System Name		Software Release	
Nortel MSL-100 (MFS, EO)		SE08	
Siemens EWSD (MFS, EO)		19d with Patch Set 46	
Lucent 5ESS (MFS, EO)		5E16.2 SU 05.0005, 5E16.2 SU 06-0002	
Raytheon Secure Digital Switch (DRSN)		15.02.03	
Raytheon Digital Small Switch (DRSN)		8.05.00	
Software Release	Component	Sub-Component (see note)	Hardware/Firmware/Software
<b>SUT CM 3.0 (R013x.00.0.340.3: Super Patch 11815)</b>	MCC  SCC  G650	Linux Server	REDHAT 8.0 Operating System
		Avaya P133G2 Router	Software Version 2.5.3
		Power Unit 631DA1/DB1 (MCC only)	N/A
		Maintenance/Test TN771DP	HW03 FW019
		Maintenance TN775C	Vintage 000001
		Maintenance/Test TN771DP	Vintage 000006
		Tone/Clock TN2182C	Vintage 000004
		BRI Line Card TN2198	Vintage 000003
		BRI Line Card TN556D	Vintage 000001
		Power Unit 1217A (SCC only)	N/A
		IP Server Interface TN2312AB	HW 06 FW023
		IP Server Interface TN2312AP	HW02 FW031
		IP Media Processor Board TN2302AP	HW20 FW107, HW59 FW006, HW15 FW105
		Expansion Interface Board TN570D	Vintage 000005
		Announcement Card TN2501AP	HW01 FW007
		Control LAN Card TN799DP	HW01 FW016
		Analog Card TN793B	Vintage 000002, 000007, 000013
		Analog Card TN793CP V2	HW00 FW002
		Analog Card TN793B	Vintage 000007
		Power Unit 655A (G650 only)	N/A
		DS1 Interface TN2464HP	HW00 FW17
		DS1 Interface TN2464BP	HW05 FW008
DS1 Interface TN464GP	HW06 FW17, HW02 FW017		
Call Classifier Card TN744E	Vintage 000004		
Digital Line Card TN2224CP	HW08 FW015		
Digital Line Card 754B	Vintage 000002		
Digital Line Card TN2224B	Vintage 000011 000018		
TELEPHONE INSTRUMENTS			
Interface Type		Model/Release	
2-Wire Analog		Panasonic KX-TS15-W	
2-Wire Digital Proprietary		2420 Standard	
ISDN BRI		8510T	
ISDN BRI		Tone Commander phones with Release 01.06.12: 6210U, 6210T, 6220U, 6220T, and 6220T TSG	
VoIP		4620IP, 4620SW, 4621SW, 4625SW	
<b>LEGEND:</b>			
5ESS	- Class 5 Electronic Switching System	MCC	- Multi-Carrier Cabinet
BRI	- Basic Rate Interface	MFS	- Multifunction Switch
CM	- Communications Manager	MSL	- Meridian Switching Load
DRSN	- Defense Red Switch Network	N/A	- Not Applicable
DS1	- Digital Signal Level 1	SCC	- Single-Carrier Cabinet
EO	- End Office	SE	- Succession Enterprise
EWSD	- Elektronisches Wählsystem Digital	SU	- Software Update
FW	- Firmware	SUT	- System Under Test
HW	- Hardware	T	- Part designator for S/T interface
IP	- Internet Protocol	TSG	- Telephone Secure Group
ISDN	- Integrated Services Digital Network	U	- 2-wire BRI Interface
LAN	- Local Area Network	VoIP	- Voice over Internet Protocol
NOTE: The sub-components are interchangeable in each component unless specified to be only in one component.			

**10. TEST 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 Telecom Switched Services Interoperability (TSSI) web page: [http://jitc.fhu.disa.mil/tssi/cert/cert\\_avaya.html](http://jitc.fhu.disa.mil/tssi/cert/cert_avaya.html).

**(a)** The SUT does not send an alerting message over American National Standards Institute (ANSI) T1.619a Integrated Services Digital Network (ISDN) Digital Transmission Link Level 1 (T1) Primary Rate Interface (PRI) until the preempted SUT subscriber goes on hook. The failure of the SUT to send an alerting message can result in the remote switch timing out and disconnecting the call due to no response (no alerting message) from the SUT. The timeout is approximately 8-10 seconds and can be adjusted with each switch. The SUT does send an alerting message when the preempted SUT subscriber hangs up promptly. Since the user hangs up on average within three seconds of being preempted and the timeout can be adjusted, the operational impact is minor.

**(b)** The SUT fails to send a follow on service message (out of service) when connected to the Nortel MSL-100 via an ANSI T1.619a ISDN T1 PRI. In the case where the SUT places 18 or more channels on this particular interface in a maintenance busy state, the MSL-100 will send a Restart message after 2 minutes to check the status of the trunk(s) and the SUT responds with a Restart Acknowledgement message and no follow on service message (out of service) message. When the Nortel MSL-100 fails to receive the service message (out of service) from the SUT, the MSL-100 marks the channel(s) back in service. This anomaly only occurs with the MSL-100 and can be mitigated by placing channels in a maintenance busy condition at the MSL-100 vice the SUT. The operational impact is minor.

**(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 the Isolation Code Announcement (ICA) for ROUTINE precedence calls. ROUTINE precedence calls receive a T60 slow busy treatment rather than the required ICA. The ICA is received by calls above ROUTINE precedence. The operational impact is minor.

**(b)** The SUT does not allow classmarking of each separate connection of a three-way call at different precedence levels. When a three-way call is established, the GSCR requires that "each connection shall maintain its assigned precedence level." The SUT, however, connects a three-way call in a single time slot and classmarks all

parties at the highest precedence level of the two connections. Since all members of the conference are classmarked at the highest precedence of each connection there is no operational impact.

(c) The SUT call pickup feature doesn't retrieve the call with the highest precedence first. The SUT retrieves unanswered call pickup group calls above ROUTINE in a random sequence. The GSCR requires that "If a call pickup group has more than one party in an unanswered condition and the unanswered parties are at different precedence levels, a call pickup attempt in that group shall retrieve the highest precedence call first." All unanswered precedence calls above ROUTINE in the pickup group do divert after 15-45 seconds if unanswered and are positively connected to either the attendant, night service, or alternate DN. The operational impact 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) The SUT attendant console does not support the automatic recall feature. However, a subscriber can initiate this feature through the use of a feature access code. The operational impact is minor.

(b) The SUT does not fully support the Busy Override Tone when the Attendant Console is overriding a busy line condition. The attendant provides an initial tone before entering a call but does not periodically continue to provide the tone during the duration of the override. The operational impact is minor.

(c) When an Electronic Key Telephone Systems (EKTS) member is assigned to a Multi-Line Hunt Group (MLHG), a call to that EKTS member fails to ring the other EKTS members. When a call is sent to a MLHG pilot number that causes an EKTS member to ring, all members of the EKTS group should have an incoming call appearance. EKTS is certified with the SUT for standalone only without the MLHG configuration. Since EKTS is a conditional requirement, the operational impact is minor.

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

**(5) Voice over Internet Protocol (VoIP) System.** The SUT is certified with any certified Assured Services Voice Application Local Area Network (ASVALAN), which is posted on the (TSSI) web page (<http://jitic.fhu.disa.mil/tssi>) approved product list. The following paragraphs detail the results of the SUT VoIP solution.

(a) **Voice Quality.** In accordance with the GSCR, appendix 3, section A3.2.1, VoIP calls shall have a Mean Opinion Score (MOS) of 4.0 or better as measured in accordance with Department of Defense Information Technology Standards Registry

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 4.31 with all calls having an MOS of at least 4.0. The average inter-switch MOS was measured at 4.31 with all calls having an MOS of at least 4.08. This was based on a total of 85 intra-switch and inter-switch calls.

**(b) 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 assigning an 802.1p/Q tag. Switches within the topology were configured with multiple Virtual Local Area Networks (VLANs) to separate data from voice traffic. The 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice VLAN traffic was assigned to a high priority queue, ensuring voice traffic took precedence over data traffic. Priority bits for L2 voice signaling was set for 6 and voice media was set for 5. The L3 DSCP value for voice signaling was set for 48 and voice media for 46, in the tested configuration. Packet captures indicated all tags were set properly.

**(c) Coder/Decoder (CODEC).** In accordance with the GSCR, appendix 3, section A3.2.2, the International Telecommunication Union - Telecommunication Standardization Sector G.711 Pulse-Code Modulation CODEC with a 20-millisecond (ms) packet fill was required and was met by the SUT VoIP solution.

#### **(d) Traffic Engineering.**

**1. Phones.** The Avaya IP phones that met the critical interoperability requirements for certification were the 4620IP, 4620IPSW, 4621SW, and 4625SW. Although the Avaya IP phones are capable of shared access [i.e., same switch port is shared by Personal Computer and IP phone], this capability was not tested and is not covered under this certification. Only dedicated access was tested (separate ports for voice and data) and certified. All phones and Local Area Network (LAN) ports were set to 100/FULL.

**2. Scalability.** The SUT can support 200 MedPro cards, which limits the maximum IP subscribers to 12,800. However, the manufacturer recommendation for release CM 3.0 is not to exceed 12,000 users.

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

**(f) Network Management (NM).** The GSCR, appendix 3, defines the overall NM requirements for VoIP systems. The NM requirements for the SUT LAN were satisfied with the vendor LoC.

**(g) 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 5-minute latency over 85 calls was measured to be 55.12 ms.

**(h) 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 over a 24-hour period.

**(i) IPv6.** Criteria for IPv6 capability is defined in GSCR, paragraph 1.7 table 1-3. This requirement was satisfied by a vendor LoC signed by the Vice President of their company. In accordance with reference (e), the vendor stated, in writing, compliance to the following criteria by 30 June 2008:

**1.** An IPv6 capable system or product 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).

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

**3.** Maintaining interoperability in heterogeneous environments and with IPv4.

**4.** Commitment to upgrade as the IPv6 standard evolves.

**5.** Availability of contractor/vendor IPv6 technical support.

**b. Summary.** The SUT is certified for joint use in the DSN as a SMEO with VoIP in accordance with the requirements set forth in the GSCR. The identified test discrepancies shown that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The interoperability test summary is shown in table 2-3.

**Table 2-3. SUT Interoperability Test Summary**

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.
T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs.
E1 CAS (DTMF, MFR1, DP)	Yes (Europe only)	Certified	Met all critical CRs and FRs.
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 send an alerting message after preemption until party goes on hook. <sup>1</sup> The SUT, when connected to the Nortel MSL-100, fails to send a follow on service message (out of service). <sup>2</sup>

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

<b>DSN Trunk Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since it is not a critical requirement for a SMEO, the operational impact is minor.
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
ISDN BRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
2-Wire Proprietary Digital	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT provides the Isolation Code Announcement only for precedence calls above ROUTINE. <sup>3</sup> The SUT connects a three-way call in a single time slot and classmarks all parties at the highest precedence level. <sup>4</sup> Call Pickup does not retrieve the call with the highest precedence. <sup>5</sup>
VoIP	No	Certified	The SUT met all critical CRs and FRs. <sup>6</sup>
<b>DSN Features and Capabilities</b>			
<b>Features and Capabilities</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
Common Features	No	Certified	Met all critical CRs and FRs.
Attendant	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT attendant console does not support the automatic recall feature. However, a subscriber through the use of a feature access code can initiate this feature. <sup>7</sup> The SUT does not support periodic Busy Override Tone when the Attendant Console is overriding a busy line condition. <sup>8</sup>
Public Safety	Yes	Certified	Met all CRs and FRs.
Preset Conferencing	No	Not Tested	See note 9.
Nailed-up Connections	No	Not Tested	See note 9.
PAT	No	Not Tested	See note 9.
DSN Hotline Services	Yes	Certified	Met all critical CRs and FRs.
Network Management	Yes	Certified	Met all critical CRs and FRs.
ISDN Services (EKTS)	Yes	Certified	Met all critical CRs and FRs with the following minor exception: When an EKTS member is assigned to a MLHG a call to that EKTS member fails to ring the other EKTS members. <sup>10</sup>
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Not Tested	See note 11.
<b>VoIP</b>			
<b>Features and Capabilities</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
VoIP System	Yes	Certified	Met all critical CRs and FRs. <sup>9</sup> The SUT is certified with any ASVALAN listed on the TSSI website at <a href="http://jtc.fhu.disa.mil/tssi">http://jtc.fhu.disa.mil/tssi</a> .

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

Network Gateways																																																																																												
Gateway	Interface & Signaling	Critical	Status	Remarks																																																																																								
PSTN	T1 CAS (DTMF, DP)	Yes	Certified	Met all critical CRs and FRs.																																																																																								
	T1 CAS (MFR1)	No	Certified	Met all critical CRs and FRs.																																																																																								
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all critical CRs and FRs.																																																																																								
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs.																																																																																								
	E1 ISDN PRI	No	Not Tested	The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Since it is not a critical requirement for a SMEO, the operational impact is minor.																																																																																								
	Ground Start Line	Yes	Certified	Met all critical CRs and FRs.																																																																																								
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified <sup>12</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>IT</td> <td>- Information Technology</td> </tr> <tr> <td>ASVALAN</td> <td>- Assured Services Voice Application LAN</td> <td>ISDN</td> <td>- Integrated Services Digital Network</td> </tr> <tr> <td>BRI</td> <td>- Basic Rate Interface</td> <td>LAN</td> <td>- Local Area Network</td> </tr> <tr> <td>CAS</td> <td>- Channel Associated Signaling</td> <td>LSSGR</td> <td>- Local Access and Transport Area (LATA) Switching System Generic Requirements</td> </tr> <tr> <td>CRs</td> <td>- Capability Requirements</td> <td>Mbps</td> <td>- Megabits per second</td> </tr> <tr> <td>DISA</td> <td>- Defense Information Systems Agency</td> <td>MFR1</td> <td>- Multifrequency Recommendation 1</td> </tr> <tr> <td>DISR</td> <td>- DoD IT Standards Registry</td> <td>MLHG</td> <td>- Multi-Line Hunt Group</td> </tr> <tr> <td>DN</td> <td>- Directory Number</td> <td>MLPP</td> <td>- Multi-Level Precedence and Preemption</td> </tr> <tr> <td>DoD</td> <td>- Department of Defense</td> <td>MSL</td> <td>- Meridian Switching Load</td> </tr> <tr> <td>DP</td> <td>- Dial Pulse</td> <td>NI 1/2</td> <td>- National ISDN Standard 1 or 2</td> </tr> <tr> <td>DRSN</td> <td>- Defense Red Switch Network</td> <td>PAT</td> <td>- Precedence Access Threshold</td> </tr> <tr> <td>DSN</td> <td>- Defense Switched Network</td> <td>PM</td> <td>- Program Manager</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>SMEO</td> <td>- Small End Office</td> </tr> <tr> <td>EKTS</td> <td>- Electronic Key Telephone System</td> <td>SS7</td> <td>- Signaling System 7</td> </tr> <tr> <td>FRs</td> <td>- Feature Requirements</td> <td>SUT</td> <td>- System Under Test</td> </tr> <tr> <td>GR</td> <td>- Generic Requirement</td> <td>T1</td> <td>- Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>GR-506</td> <td>- LSSGR: Signaling for Analog Interfaces</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>IPv4</td> <td>- Internet Protocol version 4</td> <td>TPC</td> <td>- Twisted Pair Copper</td> </tr> <tr> <td>IPv6</td> <td>- Internet Protocol version 6</td> <td>VoIP</td> <td>- Voice over Internet Protocol</td> </tr> </table> <p><b>NOTES:</b></p> <ol style="list-style-type: none"> <li>The SUT does not send an alerting message over ANSI T1.619a ISDN T1 PRI until the preempted SUT subscriber goes on hook. The failure of the SUT to send an alerting message can result in the DN switch timing out and disconnecting the call due to no response (no alerting message) from the SUT. The timeout is approximately 8-10 seconds and can be adjusted with a longer timeout. The SUT does send an alerting message when the preempted SUT subscriber hangs up promptly. Since the user hangs up on average within three seconds of being preempted and the timeout can be adjusted, the operational impact is minor.</li> <li>The SUT fails to send a follow on service message (out of service) when connected to the Nortel MSL-100 via an ANSI T1.619a ISDN T1 PRI. In the case where the SUT places 18 or more channels on this particular interface in a maintenance busy state, the MSL-100 will send a Restart message after 2 minutes to check the status of the trunk(s) and the SUT responds with a Restart Acknowledgement message and no follow on service message (out of service) message. When the Nortel MSL-100 fails to receive the service message (out of service) from the SUT, the MSL-100 marks the channel(s) back in service. This anomaly only occurs with the MSL-100 and can be mitigated by placing channels in a maintenance busy condition at the MSL-100 vice the SUT. The operational impact is minor.</li> <li>Met all DSN Announcement requirements except for Isolation Code Announcement. The SUT provides this announcement only for precedence calls above ROUTINE. ROUTINE precedence calls receive a T60 slow busy treatment. The operational impact is minor.</li> <li>The SUT does not allow classmarking of each separate connection of a three-way call at different precedence levels. When a three-way call is established, the GSCR requires that "each connection shall maintain its assigned precedence level." The SUT, however, connects a three-way call in a single time slot and classmarks all parties at the highest precedence level of the two connections. Since all members of the conference are classmarked at the highest precedence of each connection there is no operational impact.</li> <li>The SUT call pickup feature doesn't retrieve the call with the highest precedence first when more than one party is ringing within the pickup group. The calls are answered in a random sequence. Since unanswered calls above ROUTINE are diverted to the attendant, night service, or an alternate DN, the operational impact is minor.</li> <li>An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008:             <ol style="list-style-type: none"> <li>Conformant with IPv6 standards profile contained in the DISR.</li> <li>Maintaining interoperability in heterogeneous environments and with IPv4.</li> <li>Commitment to upgrade as the IPv6 standard evolves.</li> <li>Availability of contractor/vendor IPv6 technical support.</li> </ol> </li> <li>The SUT attendant console does not support the automatic recall feature. However, a subscriber can initiate this feature through the use of a feature access code. The operational impact is minor.</li> <li>The SUT does not fully support Busy Override Tone when the Attendant Console is overriding a busy line condition. The attendant provides an initial tone before entering a call but does not periodically continue to provide the tone during the duration of the override. The operational impact is minor.</li> <li>This feature is not supported. There is no operational impact because it is not a critical requirement.</li> <li>When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members. When a call is sent to a MLHG pilot number that causes an EKTS member to ring, all members of the EKTS group should have an incoming call appearance. The EKTS feature is certified as standalone and not when assigned as a member of a MLHG. MLHG interaction with EKTS is a conditional requirement; therefore, the operational impact is minor.</li> <li>Security is tested by DISA-led Information Assurance test teams and published in a separate report.</li> <li>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.</li> </ol>					ANSI	- American National Standards Institute	IT	- Information Technology	ASVALAN	- Assured Services Voice Application LAN	ISDN	- Integrated Services Digital Network	BRI	- Basic Rate Interface	LAN	- Local Area Network	CAS	- Channel Associated Signaling	LSSGR	- Local Access and Transport Area (LATA) Switching System Generic Requirements	CRs	- Capability Requirements	Mbps	- Megabits per second	DISA	- Defense Information Systems Agency	MFR1	- Multifrequency Recommendation 1	DISR	- DoD IT Standards Registry	MLHG	- Multi-Line Hunt Group	DN	- Directory Number	MLPP	- Multi-Level Precedence and Preemption	DoD	- Department of Defense	MSL	- Meridian Switching Load	DP	- Dial Pulse	NI 1/2	- National ISDN Standard 1 or 2	DRSN	- Defense Red Switch Network	PAT	- Precedence Access Threshold	DSN	- Defense Switched Network	PM	- Program Manager	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)	SMEO	- Small End Office	EKTS	- Electronic Key Telephone System	SS7	- Signaling System 7	FRs	- Feature Requirements	SUT	- System Under Test	GR	- Generic Requirement	T1	- Digital Transmission Link Level 1 (1.544 Mbps)	GR-506	- LSSGR: Signaling for Analog Interfaces	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	IPv4	- Internet Protocol version 4	TPC	- Twisted Pair Copper	IPv6	- Internet Protocol version 6	VoIP	- Voice over Internet Protocol
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**12. TEST AND ANALYSIS REPORT.** No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at <https://stp.fhu.disa.mil>. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at <http://jit.fhu.disa.mil> (NIPRNet), or <http://199.208.204.125> (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>.