



## DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND

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IN REPLY  
REFER TO:

Battlespace Communications Portfolio (JTE)

30 October 2007

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

**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 S8400 Digital Switching System with Software Release CM 4.0 (R014x.00.2.731.7: Super Patch 14419) is hereinafter referred to as the System Under Test (SUT). The SUT met all of its critical interoperability requirements and is certified as interoperable for joint use within the Defense Switched Network (DSN). The SUT was tested and met the critical interoperability requirements for the following DSN switch types: Private Branch Exchange (PBX) 1, PBX 2, and Deployable Voice Exchange (DVX). The SUT is certified to support DSN Assured Services over Internet Protocol with any Assured Services Voice Application Local Area Network (ASVALAN) on the DSN Approved Products List (APL). The SUT is also certified for joint use with any Voice Application Local Area Network (VALAN) on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), Command and Control (C2) users and Special C2 users are not authorized to be served by the SUT connected to a VALAN.

The S8400 series media servers work in conjunction with the G650 complementary media gateways which support multi-protocol environments for concurrent support of Time Division Multiplex (TDM) and Internet Protocol (IP)-based telephony. The SUT is capable of supporting a maximum of five G650s from one port network. JITC, however, conducted testing on the SUT using only one G650. Based on this testing and through analysis, this certification applies to SUTs that are configured with up to five G650s. The SUT offers an internal Automated Call Distributor (ACD). The ACD was tested and is covered under this certification. The SUT does not offer an internal voicemail capability; however, the SUT is certified for external voicemail through the use of the 2-wire digital proprietary interface. The SUT is certified for conferencing through use of any external conferencing bridge that is on the APL. The identified test

discrepancies shown in the Certification Testing Summary (enclosure 2), which remained open after Super Patch 14419 was applied and regression tested, have an overall minor operational impact. 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 and review of the vendor's Letter of Compliance (LoC). Interoperability testing was conducted by JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 4 June through 13 July 2007. Regression testing was conducted from 7 through 10 August 2007. Review of the LoC was completed on 13 August 2007. Enclosure 2 documents the test results and describes the tested network and system configurations. No other configurations, features, or functions, except those cited within this report, are certified by the JITC or authorized by the Program Management Office for use within the DSN.

4. The interoperability test summary of the SUT is contained in table 1. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in table 2. If a switch meets the PBX 1 requirements, it meets the lesser requirements of a PBX 2. The comparison between PBX 1 and DVX requirements and interoperability summary is listed in table 3. This interoperability test status is based on the SUT's ability to meet:

- a. DSN services for Network and Applications specified in reference (c).
- b. PBX 1 and DVX interface and signaling requirements for trunks/lines specified in reference (d) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 1 and DVX 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 1-4, 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 (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, DP, MFR1)	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> The SUT T1 CAS preemption signal generation is out of tolerance. <sup>2</sup> The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. <sup>3</sup> During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an “In-Service/Active” state to a “Far-End-Busy” state. <sup>4</sup>
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. <sup>3</sup> During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an “In-Service/Active” state to a “Far-End-Busy” state. <sup>4</sup>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> Failure to maintain busy out condition after restart messages are received from the distant switch. <sup>5</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 this is not a required interface for a PBX 1 or DVX, there is no operational impact. <sup>6</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 with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
VoIP (IEEE 802.3)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
<b>Voicemail</b>			
<b>Interface</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
<b>Automated Call Distributor</b>			
Internal	No	Certified	Met all CRs and FRs.

**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 CRs and FRs with the following minor exception: Selective Call Rejection is not supported by the SUT. <sup>10</sup>	
Attendant	No	Certified	Met all CRs and FRs with the following minor exception: The SUT attendant console does not support the automatic recall feature. <sup>11</sup>	
Public Safety	Yes	Certified	Met all CRs and FRs with the following minor exception: Tandem call trace of a distant office DN is not supported by SUT. <sup>12</sup>	
Preset Conferencing	No	Certified	This feature is met through the use of the Compunetx Context <sup>®</sup> 240.	
Nailed-up Connections	No	Not Tested	This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact. <sup>13</sup>	
DSN Hotline Services	Yes	Certified	The SUT met all CRs and FRs. Hotline Services is required only for analog interfaces. The SUT supports Hotline Services only with analog stations.	
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs with the following minor exceptions: When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members. <sup>14</sup> When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. <sup>15</sup>	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 16.	See note 16.	
<b>VoIP</b>				
VoIP System	No	Certified	Met all CRs and FRs. The SUT is certified for VoIP with any VALAN or ASVALAN on the DSN APL. See note 17.	
<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)	No	Certified	Met all CRs and FRs.
	T1 CAS (MFR1)	No	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 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 this is not a required interface for a PBX 1 or DVX, there is no operational impact. <sup>6</sup>
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs. See note 18.

JITC Memo, JTE, Special Interoperability Test Certification of the Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)

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

LEGEND:	
802.3u	- Standard for carrier sense multiple access with collision detection at 100 Mbps
ANSI	- American National Standards Institute
APL	- Approved Products List
ASVALAN	- Assured Services Voice Application Local Area Network
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
DS1	- Digital Signal Level 1
DSS1	- Digital Subscriber Signaling 1
DTMF	- Dual Tone Multi-Frequency
DVX	- Deployable Voice Exchange
E1	- European Basic Multiplex Rate (2.048 Mbps)
EKTS	- Electronic Key Telephone System
FRs	- Feature Requirements
GR	- Generic Requirement
GR-506-CORE	- LSSGR: Signaling for Analog Interfaces
GSCR	- Generic Switching Center Requirements
IEEE	- Institute of Electrical and Electronics Engineers, Inc.
IPv4	- Internet Protocol version 4
IPv6	- Internet Protocol version 6
ISDN	- Integrated Services Digital Network
IT	- Information Technology
JITC	- Joint Interoperability Test Command
LSSGR	- Local Access and Transport Area (LATA) Switching System Generic Requirements
Mbps	- Megabits per second
MFR1	- Multi-Frequency Recommendation 1
MLHG	- Multi-Line Hunt Group
MLPP	- Multi-Level Precedence and Preemption
ms	- milliseconds
NI 1/2	- National ISDN Standard 1 or 2
PBX 1	- Private Branch Exchange 1
PM	- Program Manager
PRI	- Primary Rate Interface
PSTN	- Public Switched Telephone Network
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
VALAN	- Voice Application Local Area Network
VoIP	- Voice over Internet Protocol

**NOTES :**

- The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. The requirement states that the yellow alarm should be removed 15 seconds +/- 5 seconds upon DS1 restoration. The SUT removes the yellow alarm 30 seconds after the DS1 is restored. The operational impact is minor.
- The SUT T1 CAS preemption signal generation is out of tolerance. The preemption signal generated by the SUT was measured 2 ms outside the GSCR required preemption signal of 345 ms +/- 5 ms. The operational impact is minor.
- The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. The SUT recognizes wink start signals from 100 ms to 395 ms as valid. The GSCR requirement specifies the wink start recognition range to be between 100 ms and 350 ms. The operational impact is minor.
- During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. During this period of time, a ROUTINE call attempted over this span receives T-120 and a precedence above ROUTINE call receives Blocked Precedence Announcement. After the state is changed, the correct treatment, an Isolated Code Announcement, is provided to all calls attempted over this span. The operational impact is minor.
- When the SUT initiates a busy-out condition for a T1 PRI, and if the distant switch sends RESTART messages while the SUT has a busy-out condition, the SUT responds with RESTART ACKNOWLEDGEMENT messages; however, the SUT does not retransmit the SERVICE (Out-Of-Service) message for all of the busied channels. The result is that the distant switch idles the channels that the SERVICE (Out-Of-Service) messages were not retransmitted on. This condition can be eliminated by busying both ends. The operational impact is minor.
- The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Therefore, this interface is not certified by the JITC or authorized by the Program Management Office for use within the DSN. Since this is not a required interface for a PBX 1 or DVX, there is no operational impact.
- The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. Since the cadence is different than a ROUTINE ring cadence, the operational impact is minor.
- 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 the attendant, night service, or alternate DN. The operational impact is minor.
- Three-way conference members do not maintain their assigned precedence levels. Since the SUT classmarks the conference members at the highest precedence level, the operational impact is minor.
- Selective Call Rejection is not supported by the SUT. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.
- The SUT attendant console does not support the automatic recall feature. The SUT does permit the attendant console to extend (camp-on) a caller to a busy station. Since this is not a required feature for a PBX 1 or DVX and the SUT provides this for the subscriber as a feature access code, the operational impact is minor.
- Tandem call trace of a distant office DN is not supported by SUT. Since this is not a required feature for a PBX 1, there is no operational impact. Although it is a requirement for a DVX, the operational impact is minor.
- This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.
- 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.
- When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. In accordance with the GSCR specification, the EKTS intercom feature should not affect the busy/idle status of any of the DNs of the calling EKTS user. An EKTS station can have additional call appearances added to compensate for this discrepancy. The operational impact is minor.
- Security is tested by DISA-led Information Assurance test teams and published in a separate report.
- An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 31 December 2008:
  - Conformant with IPv6 standards profile contained in the DISR.
  - Maintaining interoperability in heterogeneous environments and with IPv4.
  - Commitment to upgrade as the IPv6 standard evolves.
  - Availability of contractor/vendor IPv6 technical support.
- Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

**Table 2. PBX 1 Requirements**

<b>DSN Trunk Interfaces</b>				
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
T1 CAS (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>• Alarm and Restoral Requirements (R)</li> <li>• Alarm and Restoral Requirements (C)</li> <li>• WWNDP (R)</li> <li>• Outpulsing digit formats (C: CAS only)</li> <li>• Routing (C)</li> <li>• Trunk Groups (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. 7.1.4</li> <li>• GSCR Sect. 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. 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)		<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>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Facsimile	<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>
		Data	<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>
		VTC		
<b>DSN Line Interfaces</b>				
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line signaling (R)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Analog Ground Start (R)</li> <li>• Alerting Signals and Tones (R)</li> <li>• WWNDP (R)</li> <li>• Origination Treatment (R)</li> <li>• Termination Treatment (R)</li> <li>• Release Treatment (R)</li> <li>• Interruption Treatment (R)</li> <li>• Connections (R)</li> <li>• Class of Service (C)</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 5.2.2</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.1.1</li> <li>• GSCR Sect. 4.1.2</li> <li>• GSCR Sect. 4.1.3</li> <li>• GSCR Sect. 4.1.4</li> <li>• GSCR Sect. 4.1.5</li> <li>• GSCR Sect. 4.1.6</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)	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>
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 (IEEE 802.3)	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. PBX 1 Requirements (continued)**

<b>SUT Voice Mail Interfaces</b>			
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
2 Wire Digital Proprietary	No	<ul style="list-style-type: none"> <li>• FCC Part15/Part 68 (R): Analog only</li> <li>• DTMF outpulsing (C)</li> <li>• DISR compliance as applicable (R)</li> <li>• ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)</li> <li>• TIA/EIA-470-B (R): Analog only</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR A7.5</li> <li>• GSCR A7.5, 5.4.1, 5.4.2</li> <li>• GSCR A7.5</li> <li>• GSCR A7.5.5</li> <li>• GSCR A7.5.1</li> </ul>
<b>Automated Call Distributor Interfaces</b>			
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Internal	No	<ul style="list-style-type: none"> <li>• DTMF outpulsing (C)</li> <li>• DISR compliance as applicable (R)</li> <li>• ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. A7.5, 5.4.1, 5.4.2</li> <li>• GSCR Sect. A7.5</li> <li>• GSCR Sect. A7.5</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	No	<ul style="list-style-type: none"> <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. 2.1.3</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>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing 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. 2.4.5</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>

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

<b>DSN Features &amp; Capabilities</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
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>
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 (replacement for 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 ≤ 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 Section 1, paragraph 1.7</li> </ul>
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
PSTN <sup>1</sup>	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> </ul>
DRSN <sup>2</sup>	Yes	Access	<ul style="list-style-type: none"> <li>• Alerting Signals and Tones (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• Analog busy/idle (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR 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>

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

<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
CoS	- Class of Service
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
PBX	- Private Branch Exchange
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
QoS	- Quality of Service
R	- Required
Sect.	- Section
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

**NOTES:**  
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.

**Table 3. SUT PBX 1/DVX Comparison and Interoperability Test Summary**

<b>GSCR Paragraph</b>	<b>Requirement</b>	<b>PBX 1 Critical</b>	<b>DVX Critical</b>	<b>Status</b>	<b>Remarks</b>
2.1.4	Code Restriction and Diversion	No	Yes	Certified	Met all critical CRs and FRs.
2.4.2	Trace of Terminating Calls	No	Yes	Certified	Met all critical CRs and FRs.
2.4.3	Outgoing Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.4	Tandem Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.5	Trace of a Call in Progress	No	Yes	Certified	Met all critical CRs and FRs.
2.5.4.2	Manual Test of Trunks	No	Yes	Certified	Met all critical CRs and FRs.
2.5.5	Trunk Group Remove from Service (Make Busy)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.6	Trunk Group Return to Service (Make Idle)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.7	Carrier Group Alarm	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.1	Preset Conferencing	No	Yes	Certified	Met all critical CRs and FRs.
2.12.1	Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.2	Hotline Service Protection	No	Yes	Certified	Met all critical CRs and FRs.
2.12.3	Non-Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.4	Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
3.2.3	MLPP Trunk Selection	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.1	Calls from non-MLPP Networks	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.2	Precedence Calls to non-MLPP	No	Yes	Certified	Met all critical CRs and FRs.
3.4.1	Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.
3.7	ISDN MLPP Primary Rate Interface (ANSI T1.619a)	Yes	No	Certified	Met all critical CRs and FRs.
3.14	Data Collection	No	Yes	Certified	Met all critical CRs and FRs.
4.1.6	Class of Service	No	Yes	Certified	Met all critical CRs and FRs.
4.2	Primary and Alternate Routing	No	Yes	Certified	Met all critical CRs and FRs.
4.3.1	E&M Lead Signaling States	No	Yes	Certified	Met all critical CRs and FRs.

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**Table 3. SUT PBX 1/DVX Comparison and Interoperability Test Summary (continued)**

<b>GSCR Paragraph</b>	<b>Requirement</b>	<b>PBX 1 Critical</b>	<b>DVX Critical</b>	<b>Status</b>	<b>Remarks</b>																																																
4.3.2	Four Wire E&M Analog User Access Lines	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.4.2	Terminating Call Processing	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.5.2	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.5.1.8	Emergency Service 911 Conflict Resolution	Yes	No	Certified	Met all critical CRs and FRs.																																																
Table 4-9	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.																																																
Table 4-10	DSN Switch DTMF Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.5.5	Base Services – Abbreviated Numbers	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.5.7	Digit Registration Capacity	No	Yes	Certified	Met all critical CRs and FRs.																																																
4.5.8	Screening	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.3.1.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.3.1.2	Glare Operation	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.3.2.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.3.2.2	Glare Resolution	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.7	Satellite Interface	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.8	Disconnect Control	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.9	Reselect or Retrial	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.3.10	Off-Hook Supervision	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.4.1	Dial Pulse Signals	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.4.2	MFR1 Signaling	No	Yes	Certified	Met all critical CRs and FRs.																																																
5.4.3	MFR1 2/6 Signaling	No	Yes	Certified	Met all critical CRs and FRs.																																																
7.1.2	Supervisory Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.																																																
7.2	PCM-30 Digital Trunk Interface	No	Yes	Certified	Met all critical CRs and FRs.																																																
7.3	Interoperation of PCM-24 and PCM-30 Systems	No	Yes	Certified	Met all critical CRs and FRs.																																																
A2.5.2.5	DISA Network Traffic Management Operating System (NTMOS)	No	Yes	Certified	Met all critical CRs and FRs.																																																
A2.5.2.6	Data Quality	No	Yes	Certified	Met all critical CRs and FRs.																																																
9.2.2.1.1	Traffic Measurements	No	Yes	Certified	Met all critical CRs and FRs.																																																
9.8	Remote Access to Switch	No	Yes	Certified	Met all critical CRs and FRs.																																																
A2.5.2.3	DVX Switch ISDN Outpulsing Digit Formats	No	Yes	Certified	Met all critical CRs and FRs.																																																
12.2	PBX Availability	Yes	No	Certified	Met all critical CRs and FRs.																																																
Section 13	Security	Yes	No	Certified	Met all critical CRs and FRs.																																																
<b>LEGEND:</b> <table border="0"> <tr> <td>A</td> <td>- Appendix</td> <td>GSCR</td> <td>- Generic Switching Center Requirements</td> </tr> <tr> <td>ANSI</td> <td>- American National Standards Institute</td> <td>ISDN</td> <td>- Integrated Services Digital Network</td> </tr> <tr> <td>BRI</td> <td>- Basic Rate Interface</td> <td>MFR1</td> <td>- Multi-Frequency Recommendation 1</td> </tr> <tr> <td>CDR</td> <td>- Call Detail Recording</td> <td>MLPP</td> <td>- Multi-Level Precedence and Preemption</td> </tr> <tr> <td>CRs</td> <td>- Capability Requirements</td> <td>NI 1/2</td> <td>- National ISDN Standard 1 or 2</td> </tr> <tr> <td>DISA</td> <td>- Defense Information Systems Agency</td> <td>PBX</td> <td>- Private Branch Exchange</td> </tr> <tr> <td>DSN</td> <td>- Defense Switched Network</td> <td>PCM-24</td> <td>- Pulse Code Modulation - 24 Channels</td> </tr> <tr> <td>DSS1</td> <td>- Digital Subscriber Signaling 1</td> <td>PCM-30</td> <td>- Pulse Code Modulation - 30 Channels</td> </tr> <tr> <td>DTMF</td> <td>- Dual Tone Multi-Frequency</td> <td>SS7</td> <td>- Signaling System 7</td> </tr> <tr> <td>DVX</td> <td>- Deployable Voice Exchange</td> <td>SUT</td> <td>- System Under Test</td> </tr> <tr> <td>E&amp;M</td> <td>- Ear and Mouth</td> <td>T1</td> <td>- Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>FRs</td> <td>- Feature Requirements</td> <td>T1.619a</td> <td>- SS7 and ISDN MLPP Signaling Standard for T1</td> </tr> </table>						A	- Appendix	GSCR	- Generic Switching Center Requirements	ANSI	- American National Standards Institute	ISDN	- Integrated Services Digital Network	BRI	- Basic Rate Interface	MFR1	- Multi-Frequency Recommendation 1	CDR	- Call Detail Recording	MLPP	- Multi-Level Precedence and Preemption	CRs	- Capability Requirements	NI 1/2	- National ISDN Standard 1 or 2	DISA	- Defense Information Systems Agency	PBX	- Private Branch Exchange	DSN	- Defense Switched Network	PCM-24	- Pulse Code Modulation - 24 Channels	DSS1	- Digital Subscriber Signaling 1	PCM-30	- Pulse Code Modulation - 30 Channels	DTMF	- Dual Tone Multi-Frequency	SS7	- Signaling System 7	DVX	- Deployable Voice Exchange	SUT	- System Under Test	E&M	- Ear and Mouth	T1	- Digital Transmission Link Level 1 (1.544 Mbps)	FRs	- Feature Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1
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<b>NOTE:</b> The requirements for PBX 1s and DVXs are identical except for those listed in above.																																																					

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)

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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>.

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](mailto:joseph.schulte@disa.mil). The tracking number for the SUT is 0700902.

FOR THE COMMANDER:

2 Enclosures a/s



MANUEL H. GARCIA, JR.

Chief

Battlespace Communications Portfolio

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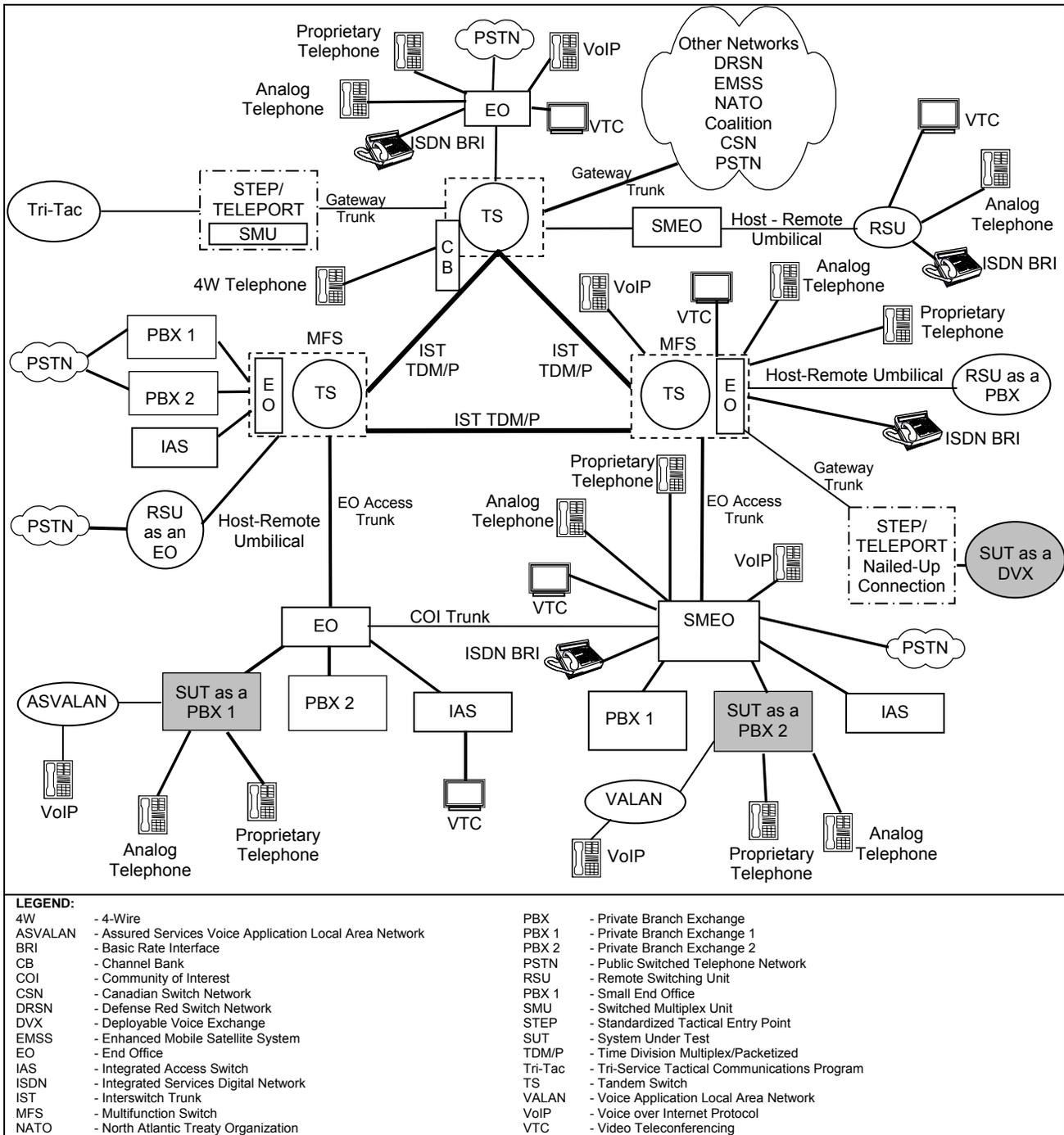
Defense Information Systems Agency (DISA), ATTN: GS23 (Mr. McLaughlin), Room 5W23, 5275 Leesburg Pike (RTE 7), Falls Church, VA 22041

## **ADDITIONAL REFERENCES**

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (d) Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007
- (e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

## CERTIFICATION TESTING SUMMARY

- 1. SYSTEM TITLE.** Avaya S8400 Digital Switching System with Software Release Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419) is hereinafter referred to as the System Under Test (SUT).
- 2. PROPONENTS.** Program Executive Office Command, Control, Communications, Computers and Intelligence (PEO C4I) Shore and Expeditionary Program Office (PMW 790).
- 3. PROGRAM MANAGER.** Ms. Shirley Dolengo, Shore Telephony, 4301 Pacific Hwy, OT4 Rm 2043, San Diego, California, 92110, e-mail: shirley.dolengo@navy.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 a 3.2 Gigahertz (GHz) processor. The SUT can process up to 45,000 Busy Hour Call Completions and support 400 trunks and up to 900 stations, of which 225 can be Internet Protocol (IP) stations. The SUT offers the following trunk interfaces: Digital Transmission Link Level 1 (T1) Channel Associated Signaling (CAS), European Basic Multiplex Rate (E1) CAS, T1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI), and E1 ISDN PRI. The SUT E1 ISDN PRI interface was not tested and is not covered under this certification. The SUT offers the following line interfaces: 2-wire analog, ISDN Basic Rate Interface (BRI), 2-wire proprietary digital, and Institute of Electrical and Electronics Engineers, Inc. (IEEE) 802.3u. The SUT configuration tested included one G650 which houses the S8400 server card. The G650 is a media gateway which also contains the various IP and Time Division Multiplex (TDM) cards used in the test. The SUT can support up to five G650s. The SUT offers a Voice over Internet Protocol (VoIP) capability that was successfully tested and is covered by this certification. Avaya's S8400 digital switching system is currently in use within the Defense Information System Network providing 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 the Private Branch Exchange 1 (PBX 1). The Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. This 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. If a switch meets the PBX 1 requirements, it meets the lesser requirements of a PBX 2. The comparison between PBX 1 and Deployable Voice Exchange (DVX) requirements and interoperability summary is listed in table 2-2. These requirements are derived from:

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

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

c. GSCR PBX 1 and DVX 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 1-4, verified through vendor submission of LoC signed by the Vice President of the company.

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

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> <li>• Framing (R)</li> <li>• Line Code (R)</li> <li>• Signaling (R)</li> <li>• Alarm and Restoral Requirements (R)</li> <li>• Alarm and Restoral Requirements (C)</li> <li>• WWNDP (R)</li> <li>• Outpulsing digit formats (C: CAS only)</li> <li>• Routing (C)</li> <li>• Trunk Groups (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. 7.1.4</li> <li>• GSCR Sect. 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. 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)			
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• MLPP (R)</li> <li>• Secure calls (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR Sect. 3</li> <li>• CJCSI 6215.01B</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>• Analog: TIA/EIA-465-A (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (R: PRI only)</li> <li>• 64 kbps switched data (R: PRI only)</li> <li>• NX56 synchronous BER (R: PRI only)</li> <li>• NX64 synchronous BER (R: PRI only)</li> <li>• Secure data (STE/STU-III) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• GSCR 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)**

<b>DSN Line Interfaces</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>• DN Identification (R)</li> <li>• Line signaling (R)</li> <li>• Loop Start Line (R: 2-Wire Analog only)</li> <li>• Analog Ground Start (R)</li> <li>• Alerting Signals and Tones (R)</li> <li>• WWNDP (R)</li> <li>• Origination Treatment (R)</li> <li>• Termination Treatment (R)</li> <li>• Release Treatment (R)</li> <li>• Interruption Treatment (R)</li> <li>• Connections (R)</li> <li>• Class of Service (C)</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. 5.2.2</li> <li>• GSCR Sect. 5.5</li> <li>• GSCR Sect. 4.5</li> <li>• GSCR Sect. 4.1.1</li> <li>• GSCR Sect. 4.1.2</li> <li>• GSCR Sect. 4.1.3</li> <li>• GSCR Sect. 4.1.4</li> <li>• GSCR Sect. 4.1.5</li> <li>• GSCR Sect. 4.1.6</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)	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. 3.1, 3.2, 3.2.1, 3.2.2</li> <li>• CJCSI 6215.01B</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 (IEEE 802.3)	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>• NX64 synchronous BER (R: BRI only)</li> <li>• NX64 asynchronous 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>SUT Voice Mail Interfaces</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
2 Wire Digital Proprietary	No	<ul style="list-style-type: none"> <li>• FCC Part15/Part 68 (R): Analog only</li> <li>• DTMF outpulsing (C)</li> <li>• DISR compliance as applicable (R)</li> <li>• ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)</li> <li>• TIA/EIA-470-B (R): Analog only</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR A7.5</li> <li>• GSCR A7.5, 5.4.1, 5.4.2</li> <li>• GSCR A7.5</li> <li>• GSCR A7.5.5</li> <li>• GSCR A7.5.1</li> </ul>	
<b>Automated Call Distributor Interfaces</b>					
<b>Interface</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>	
Internal	No	<ul style="list-style-type: none"> <li>• DTMF outpulsing (C)</li> <li>• DISR compliance as applicable (R)</li> <li>• ROUTINE precedence only in accordance with GSCR, Section 3.3 (R)</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR Sect. A7.5, 5.4.1, 5.4.2</li> <li>• GSCR Sect. A7.5</li> <li>• GSCR Sect. A7.5</li> </ul>	

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

<b>DSN Features &amp; Capabilities</b>			
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>	<b>References</b>
Common Features	No	<ul style="list-style-type: none"> <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. 2.1.3</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>• Basic Emergency Service (911) (C)</li> <li>• Trace of terminating calls (C)</li> <li>• Outgoing 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. 2.4.5</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>
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>
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 (replacement for DITSCAP) (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR Sect. 13</li> </ul>

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

VoIP					
Feature/ Capability	Critical	Requirements Required or Conditional		References	
VoIP System	No	VoIP function is conditional. If VoIP is provided, <b>all</b> 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>• Class of Service (CoS) and Quality of Service (QoS)</li> <li>• ITU-T G.711 PCM Codec</li> <li>• Traffic Engineering</li> <li>• Security</li> <li>• NM</li> <li>• Line timing</li> <li>• Internal Clock</li> <li>• Latency ≤ 60 ms</li> <li>• Packet Loss</li> <li>• IPv6 capable</li> </ul>		<ul style="list-style-type: none"> <li>• GSCR App. 3</li> <li>• GSCR Section 1, paragraph 1.7</li> </ul>	
Network Gateways					
Gateway	Critical	Requirements Required or Conditional		References	
PSTN <sup>1</sup>	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> <li>• CJCSI 6215.01B</li> </ul>	
DRSN <sup>2</sup>	Yes	Access	<ul style="list-style-type: none"> <li>• Alerting Signals and Tones (R)</li> <li>• Call Processing (R)</li> <li>• Call Treatments (R)</li> <li>• Analog busy/idle (R)</li> </ul>	<ul style="list-style-type: none"> <li>• GSCR 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	EKTS	- Electronic Key Telephone System	NX56	- Data format restricted to multiples of 56 kbps
A/D	- Analog to Digital Conversion	EIA	- Electronic Industries Alliance	NX64	- Data format restricted to multiples of 64 kbps
ANSI	- American National Standards Institute	G.711	- Standard for PCM of Voice Frequencies	PBX	- Private Branch Exchange
App.	- Appendix	GR	- Generic Requirement (Telcordia)	PCM	- Pulse Code Modulation
BER	- Bit Error Ratio	GR-512	- LSSGR: Reliability, Section 12	PCM-24	- Pulse Code Modulation - 24 Channels
BRI	- Basic Rate Interface	GR-815	- Generic Requirements For Network Element/Network System (NE/NS) Security	PCM-30	- Pulse Code Modulation - 30 Channels
C	- Conditional	GSCR	- Generic Switching Center Requirements	PRI	- Primary Rate Interface
CAS	- Channel Associated Signaling	H.320	- Standard for Narrowband VTC	PSTN	- Public Switched Telephone Network
CCS	- Common Channel Signaling	IPv6	- Internet Protocol version 6	Q.735.3	- SS7 Signaling Standard for E1 MLPP
CJCS	- Chairman of the Joint Chiefs of Staff	ISDN	- Integrated Services Digital Network	Q.955.3	- ISDN Signaling Standard for E1 MLPP
CJCSI	- CJCS Instruction	IT	- Information Technology	QoS	- Quality of Service
CoS	- Class of Service	ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector	R	- Required
D/A	- Digital to Analog Conversion	LAN	- Local Area Network	Sect.	- Section
DIACAP	- DoD Information Assurance Certification and Accreditation Process	LSSGR	- Local Access and Transport Area (LATA) Switching Systems Generic Requirements	SS7	- Signaling System 7
DISR	- DoD IT Standards Registry	kbps	- kilobits per second	STE	- Secure Terminal Equipment
DITSCAP	- DoD IT Security Certification and Accreditation Process	KXX	- K= any number 2-8; X= any number 1-9	STIGs	- Security Technical Implementation Guides
DN	- Directory Number	Mbps	- Megabits per second	STU-III	- Secure Telephone Unit – 3 <sup>rd</sup> Generation
DoD	- Department of Defense	MFR1	- Multi-Frequency Recommendation 1	T1	- Digital Transmission Link Level 1 (1.544 Mbps)
DP	- Dial Pulse	MLPP	- Multi-Level Precedence and Preemption	T1.619a	- SS7 and ISDN Signaling Standard for T1
DSN	- Defense Switched Network	MOS	- Mean Opinion Score	TIA	- Telecommunications Industry Association
DRSN	- Defense Red Switch Network	ms	- milliseconds	TIA/EIA-465-A	- Group 3 Facsimile Apparatus for Document Transmission
DTMF	- Dual Tone Multi-Frequency	NI 1/2	- National ISDN Standard 1or 2	VBD	- Variable bit data
E1	- European Basic Multiplex Rate (2.048 Mbps)	NM	- Network Management	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.					

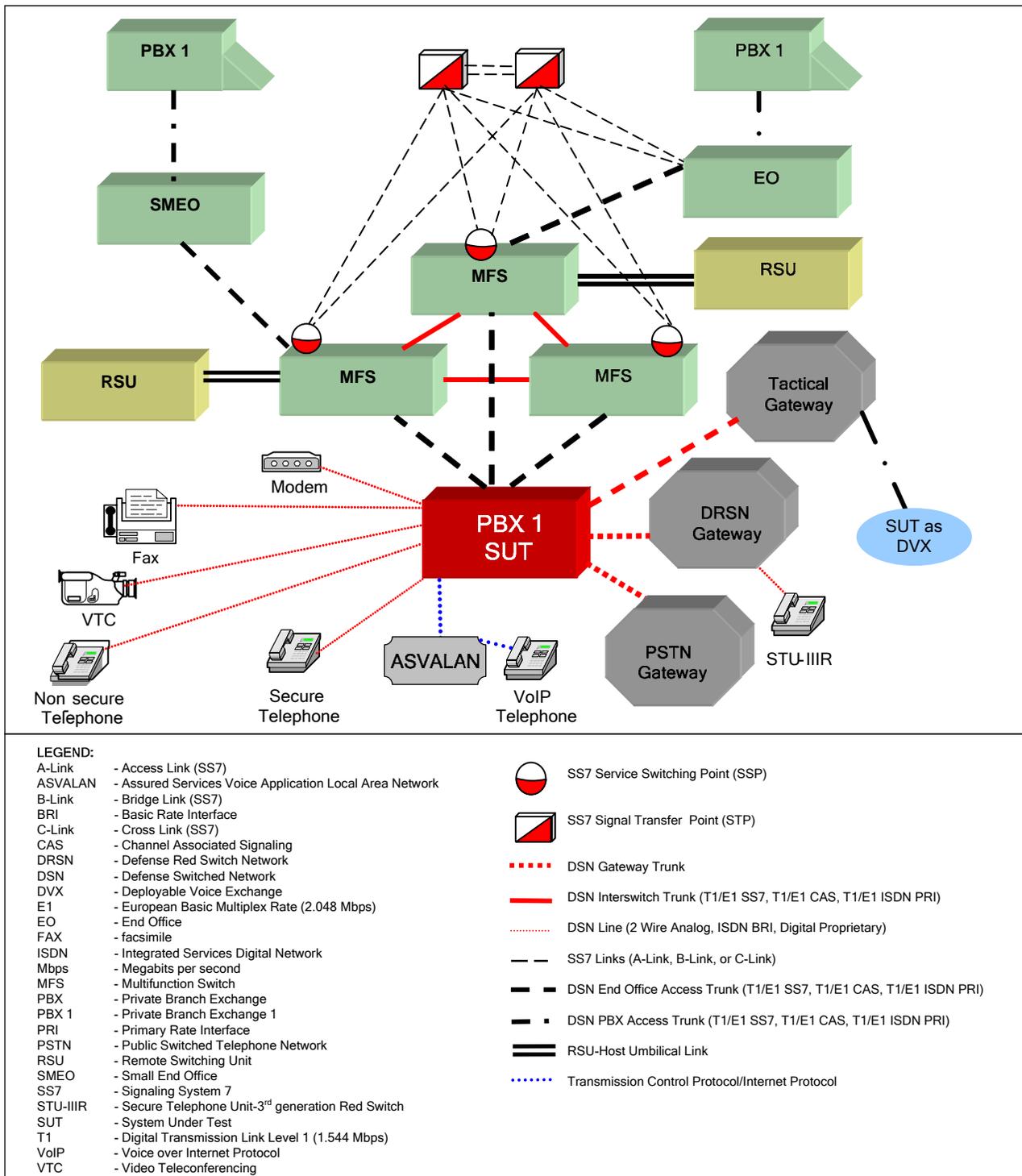
**Table 2-2. SUT PBX 1/DVX Comparison and Interoperability Test Summary**

<b>GSCR Paragraph</b>	<b>Requirement</b>	<b>PBX 1 Critical</b>	<b>DVX Critical</b>	<b>Status</b>	<b>Remarks</b>
2.1.4	Code Restriction and Diversion	No	Yes	Certified	Met all critical CRs and FRs.
2.4.2	Trace of Terminating Calls	No	Yes	Certified	Met all critical CRs and FRs.
2.4.3	Outgoing Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.4	Tandem Call Trace	No	Yes	Certified	Met all critical CRs and FRs.
2.4.5	Trace of a Call in Progress	No	Yes	Certified	Met all critical CRs and FRs.
2.5.4.2	Manual Test of Trunks	No	Yes	Certified	Met all critical CRs and FRs.
2.5.5	Trunk Group Remove from Service (Make Busy)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.6	Trunk Group Return to Service (Make Idle)	No	Yes	Certified	Met all critical CRs and FRs.
2.5.7	Carrier Group Alarm	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.1	Preset Conferencing	No	Yes	Certified	Met all critical CRs and FRs.
2.12.1	Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.2	Hotline Service Protection	No	Yes	Certified	Met all critical CRs and FRs.
2.12.3	Non-Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
2.12.4	Pair Protected Hotline Calling	No	Yes	Certified	Met all critical CRs and FRs.
3.2.3	MLPP Trunk Selection	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.1	Calls from non-MLPP Networks	No	Yes	Certified	Met all critical CRs and FRs.
3.2.4.2	Precedence Calls to non-MLPP	No	Yes	Certified	Met all critical CRs and FRs.
3.4.1	Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.
3.7	ISDN MLPP Primary Rate Interface (ANSI T1.619a)	Yes	No	Certified	Met all critical CRs and FRs.
3.14	Data Collection	No	Yes	Certified	Met all critical CRs and FRs.
4.1.6	Class of Service	No	Yes	Certified	Met all critical CRs and FRs.
4.2	Primary and Alternate Routing	No	Yes	Certified	Met all critical CRs and FRs.
4.3.1	E&M Lead Signaling States	No	Yes	Certified	Met all critical CRs and FRs.
4.3.2	Four Wire E&M Analog User Access Lines	No	Yes	Certified	Met all critical CRs and FRs.
4.4.2	Terminating Call Processing	No	Yes	Certified	Met all critical CRs and FRs.
4.5.2	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
4.5.1.8	Emergency Service 911 Conflict Resolution	Yes	No	Certified	Met all critical CRs and FRs.
Table 4-9	DSN Switch MFR1 Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
Table 4-10	DSN Switch DTMF Outpulsing Digit Format	No	Yes	Certified	Met all critical CRs and FRs.
4.5.5	Base Services – Abbreviated Numbers	No	Yes	Certified	Met all critical CRs and FRs.
4.5.7	Digit Registration Capacity	No	Yes	Certified	Met all critical CRs and FRs.
4.5.8	Screening	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.1.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.1.2	Glare Operation	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.2.1	Wink Start	No	Yes	Certified	Met all critical CRs and FRs.
5.3.3.2.2	Glare Resolution	No	Yes	Certified	Met all critical CRs and FRs.
5.3.7	Satellite Interface	No	Yes	Certified	Met all critical CRs and FRs.
5.3.8	Disconnect Control	No	Yes	Certified	Met all critical CRs and FRs.
5.3.9	Reselect or Retrial	No	Yes	Certified	Met all critical CRs and FRs.
5.3.10	Off-Hook Supervision	No	Yes	Certified	Met all critical CRs and FRs.
5.4.1	Dial Pulse Signals	No	Yes	Certified	Met all critical CRs and FRs.
5.4.2	MFR1 Signaling	No	Yes	Certified	Met all critical CRs and FRs.
5.4.3	MFR1 2/6 Signaling	No	Yes	Certified	Met all critical CRs and FRs.
7.1.2	Supervisory Channel Associated Signaling	No	Yes	Certified	Met all critical CRs and FRs.
7.2	PCM-30 Digital Trunk Interface	No	Yes	Certified	Met all critical CRs and FRs.

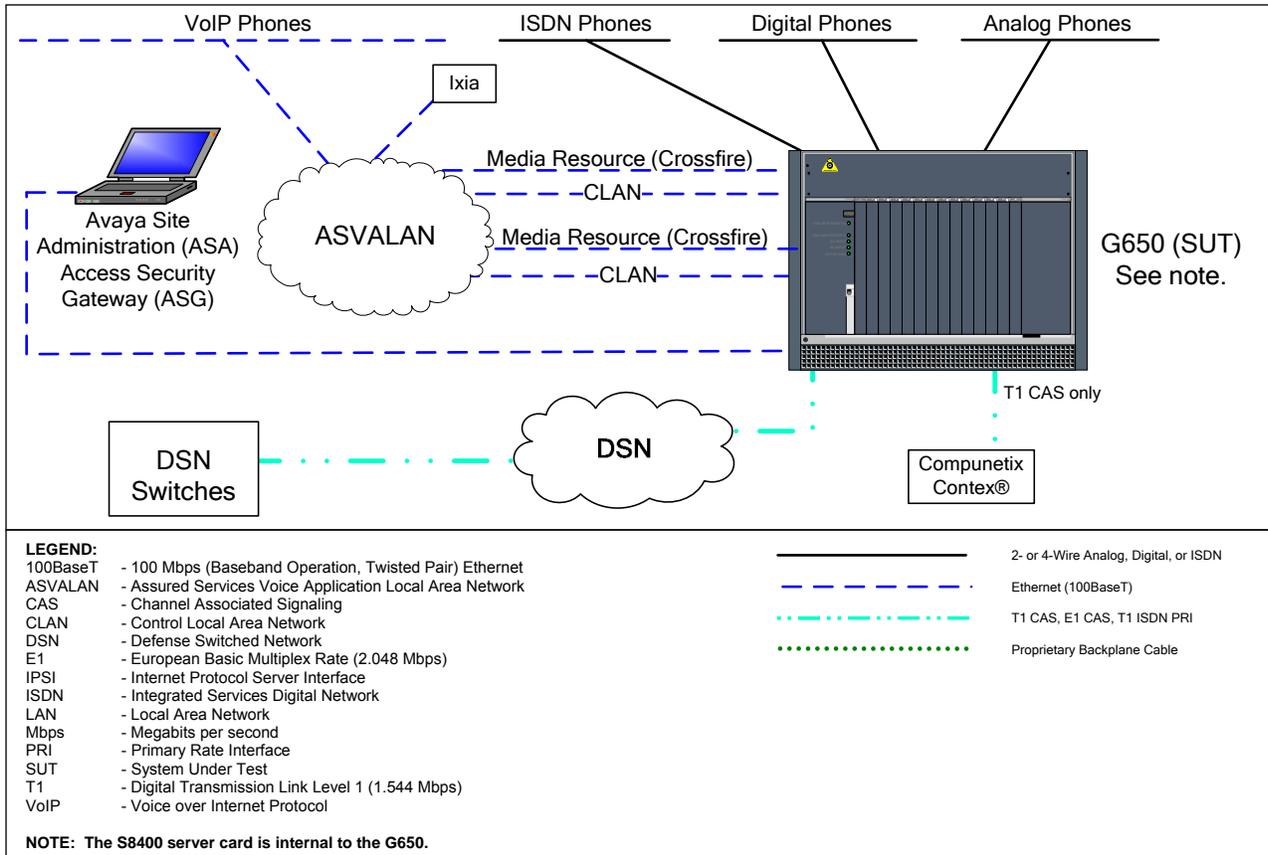
**Table 2-2. SUT PBX 1/DVX Comparison and Interoperability Test Summary (continued)**

<b>GSCR Paragraph</b>	<b>Requirement</b>	<b>PBX 1 Critical</b>	<b>DVX Critical</b>	<b>Status</b>	<b>Remarks</b>
7.3	Interoperation of PCM-24 and PCM-30 Systems	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.5	DISA Network Traffic Management Operating System (NTMOS)	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.6	Data Quality	No	Yes	Certified	Met all critical CRs and FRs.
9.2.2.1.1	Traffic Measurements	No	Yes	Certified	Met all critical CRs and FRs.
9.8	Remote Access to Switch	No	Yes	Certified	Met all critical CRs and FRs.
A2.5.2.3	DVX Switch ISDN Outpulsing Digit Formats	No	Yes	Certified	Met all critical CRs and FRs.
12.2	PBX Availability	Yes	No	Certified	Met all critical CRs and FRs.
Section 13	Security	Yes	No	Certified	Met all critical CRs and FRs.
<b>LEGEND:</b>					
A	- Appendix	GSCR	- Generic Switching Center Requirements		
ANSI	- American National Standards Institute	ISDN	- Integrated Services Digital Network		
BRI	- Basic Rate Interface	MFR1	- Multi-Frequency Recommendation 1		
CDR	- Call Detail Recording	MLPP	- Multi-Level Precedence and Preemption		
CRs	- Capability Requirements	NI 1/2	- National ISDN Standard 1 or 2		
DISA	- Defense Information Systems Agency	PBX	- Private Branch Exchange		
DSN	- Defense Switched Network	PCM-24	- Pulse Code Modulation - 24 Channels		
DSS1	- Digital Subscriber Signaling 1	PCM-30	- Pulse Code Modulation - 30 Channels		
DTMF	- Dual Tone Multi-Frequency	SS7	- Signaling System 7		
DVX	- Deployable Voice Exchange	SUT	- System Under Test		
E&M	- Ear and Mouth	T1	- Digital Transmission Link Level 1 (1.544 Mbps)		
FRs	- Feature Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard for T1		
<b>NOTE:</b> The requirements for PBX 1s and DVXs are identical except for those listed in above.					

**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 configurations shown in figures 2-2 and 2-3. Figure 2-2 depicts the general DSN test configuration. Figure 2-3 depicts the SUT test system configuration.



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



**Figure 2-3. SUT Tested System Configuration**

**9. SYSTEM CONFIGURATIONS.** Table 2-3 provides the system configurations, hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in table 2-3. The DSN switches listed in table 2-3 only depict the tested configuration. Table 2-3 is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the DSN Approved Products List (APL) that offer the same certified interfaces.

**Table 2-3. Tested System Configurations**

System Name	Software Release
Nortel CS2100	Succession Enterprise (SE) 08
Siemens EWSD	19d with Patch Set 46
Lucent 5ESS	5E16.2 Software Update 07-0003
Nortel CS1000M Cabinet	4.5W
REDCOM HDX	2.0A R3P0
Digital Subscriber Switch (DSS) (DRSN)	8.07.03
Secure Digital Switch (SDS) (DRSN)	15.04.00
ASVALAN (See note.)	Extreme ASVALAN with Native Operating System 11.6.1.9

**Table 2-3. Tested System Configurations (continued)**

Component		Software	Sub-Component	Hardware/Firmware/Software	
SUT	S8400	Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)	G650	TN8400AP	Hardware Vintage 1
				BRI Line Card TN556D	Vintage 000001
				BRI Line Card TN2198	Vintage 000003
				TN8412	Hardware Vintage 2
				IP Media Processor Board TN2602AP	HW59 FW030; HW02 FW030; HW2 FW204; HW4 FW031
				Expansion Interface Board TN570D	Vintage 000005
				Announcement Card TN2501AP	HW01 FW017
				Control LAN Card TN799DP	HW01 FW024
				Analog Card TN793CP	HW09 FW009
				Power Unit 655A	N/A
				DS1 Interface TN464GP	HW06 FW17; HW06 FW019
DS1 Interface TN464HP	HW04 FW019				
Digital Line Card TN2224CP	HW05 FW015				
Telephones, Voicemail, and Conference Bridge Components					
Interface Type		Model/Release			
2-Wire Analog		Panasonic KX-TS15-W			
2-Wire Digital Proprietary		Avaya 6402D, Avaya 2420, Avaya 6408D, Avaya 6416D+M, Avaya 6402			
ISDN BRI		Avaya 8510T			
		Tone Commander phones with Release 01.07.22: 6210U, 6210T, 6220U, 6220T, and 6220T TSG			
VoIP		Avaya 4620SW IP (Firmware: A20d01b2_8.bin, Bootstrap: B20d01a2_8.bin)			
		Avaya 4621SW IP (Firmware: A21d01b2_8.bin, Bootstrap: B21d01a2_8.bin)			
		Avaya 4625SW IP (Firmware: A25d01b2_8.bin, Bootstrap: B25d01a2_8.bin)			
		Avaya 9610 (Firmware: ha9610ua1_50r21st.bin, Bootstrap: ha9610ua1_50r21st.bin)			
		Avaya 9620 (Firmware: ha9620ua1_50r21st.bin, Bootstrap: ha9620ua1_50r21st.bin)			
		Avaya 9630 (Firmware: ha9630ua1_50r21st.bin, Bootstrap: ha9630ua1_50r21st.bin)			
Callware Technologies Callegra.UC™ Server		Avaya 9640 (Firmware: ha9640ua1_50r21st.bin, Bootstrap: ha9640ua1_50r21st.bin)			
Compunetix Inc. Context® 240		Software Release 6.14-JITC			
Compunetix Inc. Context® 240		Conference Engine Release 1.0.0			
<b>LEGEND:</b> 5ESS - Class 5 Electronic Switching System APL - Approved Products List ASVALAN - Assured Services Voice Application Local Area Network BRI - Basic Rate Interface CS - Communication Server DRSN - Defense Red Switch Network DS1 - Digital Signal Level 1 DSN - Defense Switched Network EWSD - Elektronisches Wählsystem Digital FW - Firmware GB - Gigabyte HDX - High Density Exchange HW - Hardware IP - Internet Protocol ISDN - Integrated Services Digital Network JITC - Joint Interoperability Test Command LAN - Local Area Network MB - Megabyte N/A - Not Applicable RAM - Random Access Memory SUT - System Under Test T - Part designator for S/T interface (ISDN BRI 4-wire interface) TSG - Telephone Secure Group U - Part designator for U interface (ISDN BRI 2-wire Interface) UPS - Uninterruptible Power Supply VoIP - Voice over Internet Protocol					
NOTE: The SUT is certified to support Assured Services when used in conjunction with any ASVALAN found on the DSN APL					

**10. TEST LIMITATIONS.** None.

**11. TEST RESULTS**

**a. Discussion**

**(1) DSN Trunk Interfaces.** The SUT offers the following DSN trunk interfaces: T1 CAS, E1 CAS, T1 ISDN PRI, and E1 ISDN PRI. The E1 ISDN PRI interface was not tested and not covered under this certification. Therefore, the E1 ISDN PRI interface is not authorized nor approved for use within the DSN. There is no operational impact because it is not a critical interface for a PBX 1 or DVX. The SUT met all critical

interoperability certification requirements for T1 CAS, E1 CAS, and T1 ISDN PRI DSN trunk interfaces with the exceptions noted in the following subparagraphs. The overall operational impact of the following discrepancies is minor.

(a) The SUT fails to remove a yellow alarm condition after a Digital Signal Level 1 (DS1) has been broken and restored within GSCR specification. The requirement states that the yellow alarm should be removed 15 seconds +/- 5 seconds upon DS1 restoration. The SUT removes the yellow alarm 30 seconds after the DS1 is restored. The operational impact is minor.

(b) The SUT T1 Channel Associated Signaling (CAS) preemption signal generation is out of tolerance. The preemption signal generated by the SUT was measured 2 milliseconds (ms) outside the GSCR required preemption signal of 345 ms +/- 5 ms. The operational impact is minor.

(c) The SUT recognizes T1 CAS and E1 CAS wink start signals from 100 ms to 395 ms as valid. The GSCR requirement specifies the wink start recognition range to be between 100 ms and 350 ms. The operational impact is minor.

(d) During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. During this period of time, a ROUTINE call attempted over this span receives T-120 and a precedence above ROUTINE call receives Blocked Precedence Announcement. After the state is changed, the correct treatment, an Isolated Code Announcement, is provided to all calls attempted over this span. The operational impact is minor.

(e) The SUT fails to send follow-up SERVICE (out-of-service) messages for all channels when connected to the Nortel Communication Server (CS)2100 via an American National Standards Institute (ANSI) T1.619a ISDN T1 PRI. In the case where the SUT places this particular interface in a maintenance busy state, the Nortel CS2100 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 subsequent SERVICE (out-of-service) messages for some of the channels. Because the Avaya acknowledged the Nortel CS2100's RESTART messages, the SUT is required to retransmit the SERVICE (out-of-service) messages for each channel to maintain the busy-out condition at the distant switch. When the Nortel CS2100 fails to receive the SERVICE (out-of-service) message from the SUT, the Nortel CS2100 marks the channel(s) back in service. This anomaly was observed with the Nortel CS2100 only; however, it could be duplicated on any switch that is configured to transmit RESTART messages during a T1 PRI remote busy-out condition. It can be mitigated by placing channels in a maintenance busy condition at the Nortel CS2100 only or both ends instead of the SUT. The operational impact is minor.

**(2) DSN Line Interfaces.** The SUT offers the following line interfaces: 2-wire analog, ISDN BRI, 2-wire proprietary digital, and IEEE 802.3u. The SUT met all critical

interoperability certification requirements for DSN line interfaces with the exceptions noted in the following subparagraphs. Refer to table 2-3 for specific instrument models tested under this certification test. The overall operational impact of these discrepancies is minor.

(a) The precedence above ROUTINE ring cadence is not in accordance with the GSCR specification. Since the cadence is different than a ROUTINE ring cadence, the operational impact is minor.

(b) 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 the attendant, night service, or alternate Directory Number (DN). The operational impact is minor.

(c) 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." However, the SUT 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.

**(3) Voicemail.** The SUT does not offer an internal voicemail capability; however, the SUT met all CRs and FRs for voicemail through the 2-wire digital proprietary interface.

**(4) Automated Call Distributor (ACD).** The SUT offers an internal ACD which met all critical interoperability certification requirements.

## **(5) Features and Capabilities**

(a) Common Features. The SUT met all CRs and FRs with the following minor exception: Selective Call Rejection is not supported by SUT. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.

(b) Attendant. The SUT met all CRs and FRs with the following minor exception: The SUT attendant console does not support the automatic recall feature. The SUT does permit the attendant console to extend (camp-on) a caller to a busy station. Since the SUT provides this for the subscriber as a feature access code and this is not a required feature for a PBX 1 or DVX, there is no operational impact.

(c) Public Safety. The SUT met all CRs and FRs with the following minor exception: Tandem call trace of a distant office DN is not supported by SUT. Since this is not a required feature for a PBX 1, there is no operational impact. Although it is a requirement for a DVX, the operational impact is minor.

(d) Preset Conferencing. The SUT met all CRs and FRs for preset conferencing using the Compunetix Context® 240.

(e) Nailed-up Connections. This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.

(f) Precedence Access Threshold. This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.

(g) DSN Hotline Services. The SUT met all CRs and FRs. Hotline Services is required only for analog interfaces. The SUT supports Hotline Services only with analog stations.

(h) Network Management. All critical interoperability certification CRs and FRs for this feature were met by the SUT and verified by vendor LoC.

(i) ISDN Services Electronic Key Telephone Systems (EKTS). The SUT met all CRs and FRs with the following minor exceptions:

1. When an EKTS member is assigned to a Multiline Hunt Group (MLHG), a call to that EKTS member rings only the primary EKTS member and not the other members sharing the same DN. 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. Since all unanswered precedence calls above ROUTINE place to the MLHG DN divert to an attendant, alternate DN, or night service, the operational impact is minor.

2. When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. In accordance with the GSCR specification, the EKTS intercom feature should not affect the busy/idle status of any of the DNs of the calling EKTS user. An EKTS station can have additional call appearances added to compensate for this discrepancy. The operational impact is minor.

(j) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

(k) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and verified by vendor LoC.

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

**(6) VoIP System.** The SUT is certified to support DSN Assured Services over IP with any Assured Services Voice Application Local Area Network (ASVALAN) on the DSN APL. The SUT is also certified for joint use with any Voice Application Local Area Network (VALAN) on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), Command and Control (C2) users and Special C2 users are not authorized to be served by the SUT connected to a VALAN. 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 (DISR) voice quality standards. MOS 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 4.21 with all calls having a MOS of 4.0 or better. The average inter-switch MOS was measured at 4.37 with all calls having a MOS of 4.02 or better. The MOS average was based on a total of 99 intra-switch and inter-switch calls. Automatic Gain Control (AGC) must be disabled on the IP phones in order to achieve a passing MOS.

**(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 of L2 voice signaling was set for a value of 6 and voice media was set for a value of 5. In the tested configurations, the L3 DSCP value for voice signaling was set for a value of 48 and voice media was a value of 46. Packet captures indicated all tagging was correct with the exception of initial initialization requests from the IP phones. This initial traffic is TCP, and not User Datagram Protocol (UDP). It was determined that packets required for the setup and initialization of the phones are signaling, and should have an L2 value of 6 and an L3 DSCP value of 48. The 9600 series of phones utilize an L3 DSCP value of 34 during the configuration process, but prior to the initial contact with the Control Local Area Network (CLAN) card. The 9600 phone did use the correct and expected L2 value for tagging. After the IP phone receives the initial configuration information from the CLAN card, it corrects its L3 signaling value to the expected value of 48. Additional testing was performed with the ASVALAN loaded at 125 percent of rated capacity using the Ixia IxExplorer while the phone was going through the initialization and configuration process. The phone, which utilizes fault-tolerant TCP traffic for initialization and configuration, was able to start up

normally 100 percent of the time. There was no operational impact observed, captured, or measured associated with the incorrect L3 tagging of the 9600 series phones during the initialization and configuration process.

**(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 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 4620SW, 4621SW, 4625SW, 9610, 9620, 9630, and 9640. 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 nor authorized for use within the DSN. 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 Megabits per second full duplex. In order for the IP phone to function properly on the ASVALAN, the TCP Signaling Socket setting must be set to "N" in the switch configuration for the IP phones.

**2. Scalability.** The SUT supports one port network with up to five G650s. The SUT would have about 28 available free slots to populate with CLAN and Media Resource (Crossfire) cards when expanded to five G650 chassis. This allows for the typical complement of TDM cards which would also be installed. Each pair of Media Resource cards can support up to 320 media streams, or 320 TDM to IP calls. The IP to IP calls are normally configured to bypass the Media Resource cards, so these media streams do not involve the Media Resource cards except during initial connection. Each pair of CLAN cards support up to 500 sockets, which translates to 250 active VoIP calls. In a configuration where 12 Media Resource cards and 16 CLAN cards are used, 1920 IP to TDM calls could be active. The manufacturer recommendation for release CM 4.0 is not to exceed 4,000 VoIP users total.

In order to meet reliability requirements, the CLAN and Media Resource cards must be configured in fail-over pairs. To meet reliability requirements, the SUT's IP Media Resource cards will not operate in a failover redundant fashion unless "critical reliable bearer" is set to a "y". This selection is found under the "change-ip-interface" menu. Based on values that were determined for scalability, these formulas will calculate the number of Media Resource and CLAN cards which will be required for fail-over, redundant operation. In addition, in order for IP phones to fail over to the CLAN properly, the ITU-T H.323 IP endpoint parameters in the "change system-parameters ip-options" menu must be set to the following values: Link Loss Delay Timer: 60 minutes, Primary Search time: 1800 seconds, and Periodic Registration Timer: 20 minutes. The ITU-T H.323 IP Endpoints parameters in the "change ip-network-region 1" must be set to the following values: H.323 Link Bounce Recovery to a "y", Idle Traffic Interval 20 seconds, Keep-Alive Interval to 5 seconds and the Keep-Alive count set to "5."

**Total number of pairs of Media Resource cards = total VoIP users /320.**  
**Total number of pairs of CLAN cards = total VoIP users /250.**

**(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 met this requirement with an average latency of 60 ms or less averaged over 827 five-minute calls with an overall average latency measured to be 50.9 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 met this requirement with packet loss being measured at 0.00 percent over a 24-hour period.

**(i) IPv6.** An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of Internet Protocol version 4 (IPv4). IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 31 December 2008:

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

**(7) Network Gateways.** The SUT met all critical interoperability certification requirements for Public Switched Telephone Network (PSTN) and Defense Red Switch Network (DRSN) Gateways with no exceptions. The following interfaces are certified for the PSTN gateway: T1 CAS, E1 CAS, and T1 ISDN PRI. The DRSN gateway certified interface is Twisted Pair Copper 2-wire analog.

**b. Summary.** The SUT is certified for joint use in the DSN as a PBX 1, PBX 2, and DVX 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 2-4.

**Table 2-4. 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, MFR1)	No	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> The SUT T1 CAS preemption signal generation is out of tolerance. <sup>2</sup> The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. <sup>3</sup> During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. <sup>4</sup>
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. <sup>3</sup> During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. <sup>4</sup>
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs with the following minor exceptions: The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. <sup>1</sup> Failure to maintain busy out condition after restart messages are received from the distant switch. <sup>5</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 this is not a required interface for a PBX 1 or DVX, there is no operational impact. <sup>6</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 with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
VoIP (IEEE 802.3)	No	Certified	Met all CRs and FRs with the following minor exceptions: The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. <sup>7</sup> The call pick-up feature does not pick-up the call with the highest precedence or longest ringing call first. <sup>8</sup> Three-way conference members do not maintain their assigned precedence levels. <sup>9</sup>
<b>Voicemail</b>			
<b>Interface</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.

**Table 2-4. SUT Interoperability Test Summary (continued)**

<b>Automated Call Distributor</b>				
Internal	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 CRs and FRs with the following minor exception: Selective Call Rejection is not supported by the SUT. <sup>10</sup>	
Attendant	No	Certified	Met all CRs and FRs with the following minor exception: The SUT attendant console does not support the automatic recall feature. <sup>11</sup>	
Public Safety	Yes	Certified	Met all CRs and FRs with the following minor exception: Tandem call trace of a distant office DN is not supported by SUT. <sup>12</sup>	
Preset Conferencing	No	Certified	This feature is met through the use of the Compunetx Context <sup>®</sup> 240.	
Nailed-up Connections	No	Not Tested	This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact. <sup>13</sup>	
DSN Hotline Services	Yes	Certified	The SUT met all CRs and FRs. Hotline Services is required only for analog interfaces. The SUT supports Hotline Services only with analog stations.	
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs with the following minor exceptions: When an EKTS member is assigned to a MLHG, a call to that EKTS member fails to ring the other EKTS members. <sup>14</sup> When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. <sup>15</sup>	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 16.	See note 16.	
<b>VoIP</b>				
VoIP System	No	Certified	Met all CRs and FRs. The SUT is certified for VoIP with any VALAN or ASVALAN on the DSN APL. See note 17.	
<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)	No	Certified	Met all CRs and FRs.
	T1 CAS (MFR1)	No	Certified	Met all CRs and FRs.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Certified	Met all CRs and FRs.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 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 this is not a required interface for a PBX 1 or DVX, there is no operational impact. <sup>6</sup>
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs. See note 18.

**Table 2-4. SUT Interoperability Test Summary (continued)**

LEGEND:	
802.3u	- Standard for carrier sense multiple access with collision detection at 100 Mbps
ANSI	- American National Standards Institute
APL	- Approved Products List
ASVALAN	- Assured Services Voice Application Local Area Network
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
DS1	- Digital Signal Level 1
DSS1	- Digital Subscriber Signaling 1
DTMF	- Dual Tone Multi-Frequency
DVX	- Deployable Voice Exchange
E1	- European Basic Multiplex Rate (2.048 Mbps)
EKTS	- Electronic Key Telephone System
FRs	- Feature Requirements
GR	- Generic Requirement
GR-506-CORE	- LSSGR: Signaling for Analog Interfaces
GSCR	- Generic Switching Center Requirements
IEEE	- Institute of Electrical and Electronics Engineers, Inc.
IPv4	- Internet Protocol version 4
IPv6	- Internet Protocol version 6
ISDN	- Integrated Services Digital Network
IT	- Information Technology
JITC	- Joint Interoperability Test Command
LSSGR	- Local Access and Transport Area (LATA) Switching System Generic Requirements
Mbps	- Megabits per second
MFR1	- Multi-Frequency Recommendation 1
MLHG	- Multi-Line Hunt Group
MLPP	- Multi-Level Precedence and Preemption
ms	- milliseconds
NI 1/2	- National ISDN Standard 1 or 2
PBX 1	- Private Branch Exchange 1
PM	- Program Manager
PRI	- Primary Rate Interface
PSTN	- Public Switched Telephone Network
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
VALAN	- Voice Application Local Area Network
VoIP	- Voice over Internet Protocol

**NOTES :**

- The SUT fails to remove a yellow alarm condition after a DS1 has been broken and restored within GSCR specification. The requirement states that the yellow alarm should be removed 15 seconds +/- 5 seconds upon DS1 restoration. The SUT removes the yellow alarm 30 seconds after the DS1 is restored. The operational impact is minor.
- The SUT T1 CAS preemption signal generation is out of tolerance. The preemption signal generated by the SUT was measured 2 ms outside the GSCR required preemption signal of 345 ms +/- 5 ms. The operational impact is minor.
- The SUT recognizes E1 and T1 CAS wink start signals greater than the maximum interval as valid. The SUT recognizes wink start signals from 100 ms to 395 ms as valid. The GSCR requirement specifies the wink start recognition range to be between 100 ms and 350 ms. The operational impact is minor.
- During a remote busy condition on a T1 CAS or E1 CAS, the SUT takes approximately 5 minutes to change the status of the timeslots from an "In-Service/Active" state to a "Far-End-Busy" state. During this period of time, a ROUTINE call attempted over this span receives T-120 and a precedence above ROUTINE call receives Blocked Precedence Announcement. After the state is changed, the correct treatment, an Isolated Code Announcement, is provided to all calls attempted over this span. The operational impact is minor.
- When the SUT initiates a busy-out condition for a T1 PRI, and if the distant switch sends RESTART messages while the SUT has a busy-out condition, the SUT responds with RESTART ACKNOWLEDGEMENT messages; however, the SUT does not retransmit the SERVICE (Out-Of-Service) message for all of the busied channels. The result is that the distant switch idles the channels that the SERVICE (Out-Of-Service) messages were not retransmitted on. This condition can be eliminated by busying both ends. The operational impact is minor.
- The SUT offers an E1 ISDN PRI interface; however, this interface was not tested and is not covered under this certification. Therefore, this interface is not certified by the JITC or authorized by the Program Management Office for use within the DSN. Since this is not a required interface for a PBX 1 or DVX, there is no operational impact.
- The precedence above ROUTINE ring cadence is not in accordance with GSCR specification. Since the cadence is different than a ROUTINE ring cadence, the operational impact is minor.
- 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 the attendant, night service, or alternate DN. The operational impact is minor.
- Three-way conference members do not maintain their assigned precedence levels. Since the SUT classmarks the conference members at the highest precedence level, the operational impact is minor.
- Selective Call Rejection is not supported by the SUT. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.
- The SUT attendant console does not support the automatic recall feature. The SUT does permit the attendant console to extend (camp-on) a caller to a busy station. Since this is not a required feature for a PBX 1 or DVX and the SUT provides this for the subscriber as a feature access code, the operational impact is minor.
- Tandem call trace of a distant office DN is not supported by SUT. Since this is not a required feature for a PBX 1, there is no operational impact. Although it is a requirement for a DVX, the operational impact is minor.
- This feature is not supported. Since this is not a required feature for a PBX 1 or DVX, there is no operational impact.
- 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.
- When an intercom call is placed on an EKTS station, the primary DN of the calling EKTS user is used and the station is made busy. In accordance with the GSCR specification, the EKTS intercom feature should not affect the busy/idle status of any of the DNs of the calling EKTS user. An EKTS station can have additional call appearances added to compensate for this discrepancy. The operational impact is minor.
- Security is tested by DISA-led Information Assurance test teams and published in a separate report.
- An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of their company. The vendor stated, in writing, compliance to the following criteria by 31 December 2008:
  - Conformant with IPv6 standards profile contained in the DISR.
  - Maintaining interoperability in heterogeneous environments and with IPv4.
  - Commitment to upgrade as the IPv6 standard evolves.
  - Availability of contractor/vendor IPv6 technical support.
- Interoperability Certification of the SUT does not constitute DRSN PM's approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.

**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>.