



## DEFENSE INFORMATION SYSTEMS AGENCY

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IN REPLY  
REFER TO: Joint Interoperability Test Command (JTE)

**6 May 11**

### MEMORANDUM FOR DISTRIBUTION

**SUBJECT:** Special Interoperability Test Certification of Interactive Intelligence<sup>®</sup>, Inc. Customer Interaction Center<sup>™</sup> (CIC) Release 3.0 Software Update (SU) 9

**References:** (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004  
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008  
(c) through (f), see Enclosure 1

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.

2. The Interactive Intelligence<sup>®</sup>, Inc. CIC Release 3.0 SU 9 is hereinafter referred to as the system under test (SUT). The SUT meets all of its critical interoperability requirements and is certified for joint use within the Defense Information System Network (DISN) as a Private Branch Exchange (PBX) 2. The PBX 2 switches have no Military Unique Features (MUFs) and can only serve Department of Defense (DoD), non-DoD, non-governmental, and foreign government users having no missions or communications requirement to ever originate or receive Command and Control (C2) communications. Since PBX 2s do not support MUF Requirements detailed in Reference (c), connectivity to the DSN is not authorized until a waiver is granted by the Chairman of the Joint Chiefs of Staff (CJCS) for each site in accordance with Reference (d). No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date the DISA Field Security Operations (FSO) provided a positive Certification and Accreditation (CA) Recommendation.

3. This finding is based on interoperability testing, review of the vendor's Letters of Compliance (LoC), and FSO CA Recommendation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 22 through 30 July 2010. Review of vendor's LoC was completed on 30 March 2011. The FSO provided a positive CA Recommendation on 5 April 2011 based on the security testing completed by DISA-led IA test teams and published in a separate report, Reference (e). Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The interoperability test summary of the SUT is indicated in Table 1. The PBX 2 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT’s ability to meet:

- a. DSN services for Network and Applications specified in Reference (c).
- b. PBX 2 interface and signaling requirements for trunks/lines specified in Reference (d) verified through JITC testing and/or vendor submission of LoC.
- c. PBX 2 CRs/FRs specified in Reference (d) verified through JITC testing and/or vendor submission of LoC.
- d. The overall system interoperability performance derived from test procedures listed in Reference (f).

**Table 1. SUT Interoperability Test Summary**

<b>DSN Trunk Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
T1 CAS (DTMF)	No <sup>1</sup>	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
E1 CAS (DTMF)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
T1 ISDN PRI NI 1/2 (ANSI T1.607)	No <sup>1</sup>	Certified	Met all critical CRs and FRs with the following minor exception: Secure voice calls fail at random intervals. <sup>2</sup>
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
<b>DSN Line Interfaces</b>			
<b>Interface &amp; Signaling</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs.
ISDN BRI NI 1/2	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT did not meet the IPv6 capability requirements. <sup>3</sup> The SUT incorrectly tagged OAM traffic. <sup>4</sup> The SUT does not support proper VLAN tagging with IP phones. The SUT is not certified for shared access. <sup>5</sup>
<b>DSN Features and Capabilities</b>			
<b>Feature/Capability</b>	<b>Critical</b>	<b>Status</b>	<b>Remarks</b>
Common Features	Yes	Certified	Met all critical CRs and FRs.
Attendant	No	Certified	Met all critical CRs and FRs.
Public Safety	Yes	Certified	Met all critical CRs and FRs for the basic 911.
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	No	Certified	Met all critical CRs and FRs with the T1 ISDN PRI interface.
Synchronization	Yes	Certified	Met all critical CRs and FRs.
VoIP System	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT did not meet the IPv6 capability requirements. <sup>3</sup>
Security	Yes	Certified	See note 6.

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

Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF)	No <sup>1</sup>	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	E1 CAS (DTMF)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No <sup>1</sup>	Certified	Met all critical CRs and FRs.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	Ground Start Line (GR-506-CORE)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	Loop Start Line (GR-506-CORE)	No	Certified	Met all critical CRs and FRs with an LOC.
<b>NOTES:</b>				
1 The UCR 2008 does not specify a required interface for a PBX 2. A PBX 2 switch must support at least T1 CAS or T1 PRI.				
2 After a non-secure call is established, the users can go secure. Secure calls fail at random intervals from two to eight minutes with no known explanation. After a secure call fails, the users can reestablish the non-secure call without hanging up by pressing the non-secure button. Since secure call capability is a conditional requirement for a PBX 2, there is minor operational impact.				
3 The SUT does not support IPv6. The vendor was granted a waiver by ASD-NII on 27 February 2011 with the following stipulation: The SUT must demonstrate IPv6 capability by 31 August 2011.				
4 The phones incorrectly used a different value for signaling. OAM traffic was either tagged at zero or 34 decimal. This was previously adjudicated by DISA as having a minor operational impact.				
5 Phones which offered shared access did not put voice traffic in a separate VLAN as required. The SUT is not certified for shared access, which is not required.				
6 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (e).				
<b>LEGEND:</b>				
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	ISDN	Integrated Services Digital Network	
ANSI	American National Standards Institute	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	
ASD/NII	Assistant Secretary of Defense for Networks and Information Integration	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements	
BRI	Basic Rate Interface	Mbps	Megabits per second	
CAS	Channel Associated Signaling	NI 1/2	National ISDN Standard 1 or 2	
CRs	Capability Requirements	OAM	Operations Administration and Management	
DISA	Defense Information Systems Agency	PBX 2	Private Branch Exchange 2	
DSN	Defense Switched Network	PRI	Primary Rate Interface	
DSS1	Digital Subscriber Signaling 1	PSTN	Public Switched Telephone Network	
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN	
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test	
FRs	Feature Requirements	T1	Digital Transmission Link Level 1 (1.544 Mbps)	
GR	Generic Requirement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1	
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	UCR	Unified Capabilities Requirements	
IEEE	Institute for Electrical and Electronics Engineers	VoIP	Voice over Internet Protocol	
IPv6	Internet Protocol version 6			

**Table 2. PBX 2 Requirements**

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No		<ul style="list-style-type: none"> <li>• Direct Inward Dialing (C)</li> <li>• National ISDN 1/2 Primary Access (C)</li> <li>• ITU-T ISDN Primary Access (Europe only) (C)</li> <li>• Normal Wink Start Operations (C)</li> <li>• Glare Operation (C)</li> <li>• Abnormal Wink Start (C)</li> <li>• Glare Resolution (C)</li> <li>• Call for Service Timing (C)</li> <li>• Guard Timing (C)</li> <li>• Satellite Interface (C)</li> <li>• Disconnect Control (C)</li> <li>• Reselect and Retrial (C)</li> <li>• Off-Hook Supervision Transition (C)</li> <li>• Dial-Pulse Signals (C)</li> <li>• DTMF Signaling (C)</li> <li>• DSN ISDN User-to-Network Signaling (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.3.2</li> <li>• UCR Section 5.2.1.3.4.1</li> <li>• UCR Section 5.2.1.3.4.2</li> <li>• UCR Section 5.2.4.3.3.1.1</li> <li>• UCR Section 5.2.4.3.3.1.2</li> <li>• UCR Section 5.2.4.3.3.2</li> <li>• UCR Section 5.2.4.3.3.2.2</li> <li>• UCR Section 5.2.4.3.5</li> <li>• UCR Section 5.2.4.3.6</li> <li>• UCR Section 5.2.4.3.7</li> <li>• UCR Section 5.2.4.3.8</li> <li>• UCR Section 5.2.4.3.9</li> <li>• UCR Section 5.2.4.3.10</li> <li>• UCR Section 5.2.4.4.1</li> <li>• UCR Section 5.2.4.4.2</li> <li>• UCR Section 5.2.4.7.1</li> </ul>
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Trunking	<ul style="list-style-type: none"> <li>• Application (C)</li> <li>• Physical Layer (C)</li> <li>• Data Link Layer (C)</li> <li>• Data Link Connection (C)</li> <li>• Peer-to-Peer Procedures of Data-Link Layer (C)</li> <li>• Layer 3 DSN User-to-Network Signaling (C)</li> <li>• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)</li> <li>• Sequence of Messages for DSN Circuit-Switched Calls (C)</li> <li>• Message Functional Definition and Content (C)</li> <li>• General Message Format and Information Elements Coding (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.1</li> <li>• UCR Section 5.2.4.7.1.2</li> <li>• UCR Section 5.2.4.7.1.3</li> <li>• UCR Section 5.2.4.7.1.3.1</li> <li>• UCR Section 5.2.4.7.1.3.2</li> <li>• UCR Section 5.2.4.7.1.4</li> <li>• UCR Section 5.2.4.7.1.4.2</li> <li>• UCR Section 5.2.4.7.1.4.3</li> <li>• UCR Section 5.2.4.7.1.4.4</li> <li>• UCR Section 5.2.4.7.1.4.5</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.607)	No		<ul style="list-style-type: none"> <li>• Supplementary Services (C)</li> <li>• Transmission (R)</li> <li>• PCM-24 Digital Trunk Interface (R)</li> <li>• Interface Characteristics (R)</li> <li>• Supervisory Channel Associated Signaling (C)</li> <li>• Clear Channel Capability (C)</li> <li>• Alarm and Restoral Requirements (C)</li> <li>• PCM-30 Digital Trunk Interface (Europe only) (C)</li> <li>• Interoperation of PCM-24 and PCM-30 (C)</li> <li>• Analog Trunk Interface (C)</li> <li>• Integrated Digital Loop Carrier (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.4.6</li> <li>• UCR Section 5.2.5</li> <li>• UCR Section 5.2.6.1</li> <li>• UCR Section 5.2.6.1.1</li> <li>• UCR Section 5.2.6.1.2</li> <li>• UCR Section 5.2.6.1.3</li> <li>• UCR Section 5.2.6.1.4</li> <li>• UCR Section 5.2.6.2</li> <li>• UCR Section 5.2.6.3</li> <li>• UCR Section 5.2.6.4</li> <li>• UCR Section 5.2.6.5</li> </ul>
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR 5.2.12.8.2.1</li> <li>• CJCSI 6215.01C</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
		Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (C: PRI only)</li> <li>• 64 kbps switched data (C: PRI only)</li> <li>• NX56 synchronous BER (C: PRI only)</li> <li>• NX64 synchronous BER (C: PRI only)</li> <li>• Secure data (STE/STU-III) (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• CJCSI 6215.01C</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>• FTR 1080B-2002</li> </ul>

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

DSN Line Interfaces				
Interface	Critical	Requirements Required or Conditional		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>Individual Line (R)</li> <li>PBX Line (C)</li> <li>National ISDN 1/2 Basic Access (C)</li> </ul>	<ul style="list-style-type: none"> <li>UCR Section 5.2.1.1.1</li> <li>UCR Section 5.2.1.3.1</li> <li>UCR Section 5.2.1.3.3</li> </ul>
ISDN BRI NI 1/2	No		<ul style="list-style-type: none"> <li>Analog Line (C)</li> <li>Loop Start Line (R: 2-Wire Analog only)</li> <li>Reverse Battery (C)</li> <li>S/T Reference Point (ISDN BRI) (C)</li> </ul>	<ul style="list-style-type: none"> <li>UCR Section 5.2.1.3.5</li> <li>UCR Section 5.2.4.2.1</li> <li>UCR Section 5.2.4.3.1</li> <li>UCR Section 5.2.4.7.1.2.1</li> </ul>
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> <li>MOS (R)</li> <li>Secure Calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01C</li> <li>CJCSI 6215.01C</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>DISR</li> </ul>
		Data	<ul style="list-style-type: none"> <li>Modem (VBD) (R)</li> <li>Secure data (STE/STU-III) (C)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01C</li> <li>CJCSI 6215.01C</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>FTR 1080B-2002</li> </ul>
DSN Features & Capabilities				
Feature/Capability	Critical	Requirements Required or Conditional		References
Common Features	Yes	<ul style="list-style-type: none"> <li>Individual Lines (R)</li> <li>Call waiting (C)</li> <li>Three-way calling (C)</li> <li>Add-on transfer, conference calling, and call hold (C)</li> <li>Call Transfer Individual – All calls (C)</li> <li>Call Transfer - Internal Only (C)</li> <li>Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (C)</li> <li>Call Transfer – Outside (C)</li> <li>Call Transfer – Add-On Restricted Station (C)</li> <li>Call Transfer – Attendant (C)</li> <li>Call Hold (C)</li> <li>Conference Calling – Six Way Station Controlled (C)</li> <li>Call forwarding Variable (C)</li> <li>Call Forward Busy Line (C)</li> <li>Call Forwarding – Don't Answer – All Calls (C)</li> <li>Selective Call Forwarding (C)</li> <li>Call pick-up (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.4.7.1.2.1</li> <li>UCR Section 5.2.1.1.5.1</li> <li>UCR Section 5.2.1.1.6</li> <li>UCR Section 5.2.1.1.7</li> <li>UCR Section 5.2.1.1.7.1</li> <li>UCR Section 5.2.1.1.7.2</li> <li>UCR Section 5.2.1.1.7.3</li> <li>UCR Section 5.2.1.1.7.3</li> <li>UCR Section 5.2.1.1.7.4</li> <li>UCR Section 5.2.1.1.7.5</li> <li>UCR Section 5.2.1.1.7.6</li> <li>UCR Section 5.2.1.1.7.7</li> <li>UCR Section 5.2.1.1.7.8</li> <li>UCR Section 5.2.1.1.8.1</li> <li>UCR Section 5.2.1.1.8.2</li> <li>UCR Section 5.2.1.1.8.3</li> <li>UCR Section 5.2.1.1.8.4</li> <li>UCR Section 5.2.1.1.9.1</li> </ul>
Attendant	No	<ul style="list-style-type: none"> <li>Attendant Features (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.1.2</li> </ul>
Public Safety	Yes	<ul style="list-style-type: none"> <li>Emergency Service (911) Caller (R)</li> <li>Emergency Service (911) Public Safety Answering Service (C)</li> <li>Enhanced Emergency Service (E911) (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.1.4.1.1</li> <li>UCR Section 5.2.1.4.1.2</li> <li>UCR Section 5.2.1.4.1.3</li> </ul>
Call Processing	Yes	<ul style="list-style-type: none"> <li>Origination Treatment (R)</li> <li>Originating Busy (R)</li> <li>Termination Treatment (R)</li> <li>Busy or Idle Status (C)</li> <li>Release Treatment (R)</li> <li>Interruption Treatment (R)</li> <li>Connections (R)</li> <li>Class of Service (C)</li> <li>E&amp;M Lead Signaling States (C)</li> <li>4-Wire Analog User Access Lines (C)</li> <li>2-Wire User Access Lines (C)</li> <li>Interswitch and Intraswitch Dialing (C)</li> <li>Calling Name Delivery (C)</li> <li>Calling Number Delivery (C)</li> <li>Screening (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.3.1.1</li> <li>UCR Section 5.2.3.1.1.1</li> <li>UCR Section 5.2.3.1.2</li> <li>UCR Section 5.2.3.1.2.1</li> <li>UCR Section 5.2.3.1.3</li> <li>UCR Section 5.2.3.1.4</li> <li>UCR Section 5.2.3.1.5</li> <li>UCR Section 5.2.3.1.6</li> <li>UCR Section 5.2.3.3.1</li> <li>UCR Section 5.2.3.3.2</li> <li>UCR Section 5.2.3.3.3</li> <li>UCR Section 5.2.3.5.1.2</li> <li>UCR Section 5.2.3.5.1.8.1</li> <li>UCR Section 5.2.3.5.1.8.2</li> <li>UCR Section 5.2.3.5.8</li> </ul>

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

<b>DSN Features &amp; Capabilities (continued)</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
ISDN Services	No	<ul style="list-style-type: none"> <li>• BRI Access, Call Control and Signaling (C)</li> <li>• Uniform Interface Configuration for BRIs (C)</li> <li>• BRI Features (C)</li> <li>• PRI Access, Call Control and Signaling (C)</li> <li>• PRI Features (C)</li> <li>• Packet Data Features and Capabilities (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.9.2 Table 5.2.9-1</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-2</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-3</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-4</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-5</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-6</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (C)</li> <li>• Internal Stratum 4 (R)</li> <li>• Synchronization Performance Monitoring Criteria (C)</li> <li>• DS1 Traffic Interfaces (C)</li> <li>• DS0 Traffic Interconnects (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.10.1.1.2</li> <li>• UCR Section 5.2.10.1.2.2</li> <li>• UCR Section 5.2.10.2</li> <li>• UCR Section 5.2.10.3</li> <li>• UCR Section 5.2.10.4</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 3</li> </ul>
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 (R)</li> <li>• ITU-T G.711 PCM CODEC (R)</li> <li>• MLPP (C)</li> <li>• Security (R)</li> <li>• Network management (C)</li> <li>• System timing (R)</li> <li>• Latency ≤ 60 milliseconds (R)</li> <li>• IPv6 capable (R)</li> <li>• Service Class Tagging (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR section 5.2.12.8.2.1</li> <li>• UCR section 5.2.12.8.2.2</li> <li>• UCR section 5.2.12.8.2.3</li> <li>• UCR section 5.2.12.8.2.4</li> <li>• UCR section 5.2.12.8.2.5</li> <li>• UCR section 5.2.12.8.2.6</li> <li>• UCR section 5.2.12.8.2.7</li> <li>• UCR section 5.2.12.8.2.8</li> <li>• UCR section 5.2.12.8.2.9</li> </ul>
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
PSTN	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> <li>• Loop Start Line (C)</li> <li>• Ground Start Line (C)</li> <li>• Immediate Start (C)</li> <li>• Delay Dial (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.4.2.1</li> <li>• UCR Section 5.2.4.2.2</li> <li>• UCR Section 5.2.4.3.2</li> <li>• UCR Section 5.2.4.3.4</li> </ul>

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

<b>LEGEND:</b>					
ANSI	American National Standards Institute	FTR	Federal Telecommunications Recommendation	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	FTR 1080B	Video Teleconferencing Services	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	GR	Generic Requirement	PRI	Primary Rate Interface
C	Conditional	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling	H.320	Standard for Narrowband VTC	Q.931	Signaling Standard for ISDN Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	ISDN	Integrated Services Digital Network	R	ISDN BRI 4-wire interface
DIACAP	DoD Information Assurance Certification and Accreditation Process	IT	Information Technology	S/T	Secure Terminal Equipment
DISR	DoD IT Standards Registry	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	STIGs	Security Technical Implementation Guides
DoD	Department of Defense			STU-III	Secure Telephone Unit -3rd generation
DoDI	DoD Instruction	kbps	kilobits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DP	Dial Pulse	Mbps	Megabits per second	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DS0	Digital Signal Level 0	MFR1	Multi-Frequency Recommendation 1	T.4	Standardization of Group 3 facsimile terminals for document transmission
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MLPP	Multi-Level Precedence and Preemption	UCR	Unified Capabilities Requirements
DSN	Defense Switched Network	MOS	Mean Opinion Score	VBD	Variable bit data
DSS1	Digital Subscriber Signaling 1	NI 1/2	National ISDN Standard 1 or 2	VoIP	Voice over Internet Protocol
DTMF	Dual Tone Multi-Frequency	NX56	Data format restricted to multiples of 56 kbps	VTC	Video Teleconferencing
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps		
E911	Enhanced 911 Service	PBX	Private Branch Exchange		
E&M	Ear and Mouth	PBX 2	Private Branch Exchange 2		

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at <https://stp.fhu.disa.mil>. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at <http://jit.fhu.disa.mil> (NIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: [ucco@disa.mil](mailto:ucco@disa.mil).

6. The JITC point of contact is Ms. Anita Mananquil, DSN 879-5164, commercial (520) 538-5164, FAX DSN 879-4347, or e-mail to [anita.mananquil@disa.mil](mailto:anita.mananquil@disa.mil). The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0918002.

FOR THE COMMANDER:

2 Enclosures a/s

  
for BRADLEY A. CLARK  
Chief  
Battlespace Communications Portfolio

Distribution (electronic mail):

Joint Staff J-6

Joint Interoperability Test Command, Liaison, TE3/JT1

Office of Chief of Naval Operations, CNO N6F2

Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)

Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ

U.S. Marine Corps MARCORSSYSCOM, SIAT, MJI Division I

DOT&E, Net-Centric Systems and Naval Warfare

U.S. Coast Guard, CG-64

Defense Intelligence Agency

National Security Agency, DT

Defense Information Systems Agency, TEMC

Office of Assistant Secretary of Defense (NII)/DOD CIO

U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities  
Division, J68

Defense Information Systems Agency, GS23

## **ADDITIONAL REFERENCES**

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007
- (d) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008," 22 January 2009
- (e) Joint Interoperability Test Command, Memo, "Information Assurance (IA) Assessment of Interactive Intelligence Corporation (ININ) Customer Interaction Center (CIC) Release (Rel.) 3.0 (TN0918002)," 6 April 2011
- (f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

## CERTIFICATION TESTING SUMMARY

**1. SYSTEM TITLE.** Interactive Intelligence<sup>®</sup>, Inc. Customer Interaction Center<sup>™</sup> (CIC) Release 3.0 Software Update (SU) 9; hereinafter referred to as the System Under Test (SUT).

**2. PROPONENT.** Defense Contract Management Agency (DCMA) Integrated Technology Directorate.

**3. PROGRAM MANAGER.** Mr. Dave Gaulden, DCMAIT-FC, Post Office Box 3990, Columbus, Ohio 43218-3990, Email: dave.gaulden@dcma.mil.

**4. TESTER.** Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

**5. SYSTEM UNDER TEST DESCRIPTION.** The SUT is an all-in-one Enterprise Voice over Internet Protocol (VoIP) Public Branch Exchange (PBX) 2 and Contact Center solution. The SUT includes an auto-attendant, an Interactive Voice Response (IVR) service, and an Automatic Call Distributor (ACD) for call centers with skills-based routing and voice recording. The SUT consists of the following components: AudioCodes Mediant 3000 (M3K), CIC Server, CIC Switchover Server, Database Server, AudioCodes Element Management System (EMS) Server, Media Server, Polycom SoundPoint IP 3/4/5/6 Series Phones, site-provided ININ-Vista Management Workstation, site-provided ININ-Experience (XP) Management Workstation, site-provided AudioCodes EMS Client-XP Management Workstation, AudioCodes MediaPack (MP)-124, and analog telephones.

- **AudioCodes M3K.** The AudioCodes M3K is a media gateway that provides connection to traditional Time-Division Multiplexing (TDM) networks by way of standard Public Switched Telephone Network (PSTN) telecommunications interfaces.

- **CIC Server.** The CIC Server runs on the Hewlett-Packard (HP) DL 320, 360, and 380 series of servers. The CIC Server is a VoIP-based, all-in-one Internet Protocol (IP) PBX, IVR, and ACD. The CIC Server connects with IP telephones, digital gateways to connect third party switches with legacy analog telephones, and analog VoIP gateways to connect legacy telephone networks with IP-based telephony networks.

- **CIC Switchover Server.** The CIC Switchover Server runs on the HP DL 320, 360, and 380 series of servers. The CIC Switchover is an identical copy of the CIC server, which can be used in case of a failure or maintenance of the primary server. The Switchover Server is running in stand-by mode and can be made active (automatically as well as manually) if required. In case of a switchover, established calls will not be affected and will continue.

- **Database Server.** The Database Server is a Microsoft (MS) Structured Query Language (SQL) 2008 Server, which runs on the HP DL 320, 360, and 380 Series of servers. The CIC 3.0 uses the Microsoft SQL database to store reporting information

about the calls. This information is used to provide standard and custom historical reporting.

- **AudioCodes EMS Server.** The AudioCodes EMS Server is installed on a Sun Fire V215. The EMS is a server-based application that provides the management and provisioning of the AudioCodes M3K and AudioCodes MP-124 media gateway network elements.

- **Media Server.** The Media Server runs on the HP DL 320, 360, and 380 series of servers and is capable of load balancing. The Media Server processes the audio used by the CIC Server. It can play IVR prompts, announcements, or on-hold music to callers. The Media Server can also handle Dual-Tone Multi-Frequency (DTMF) input while CIC maintains control of the call. The server also allows calls to be recorded (based on privileges) by the server and later moved to the CIC Server for management.

- **Polycom SoundPoint IP Telephones.** Most Polycom SoundPoint IP telephones support High-Definition (HD) Voice. The HD voice does not conform to the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) G.711 standard, which is the required codec.

**Polycom SoundPoint IP 3 Series – 321, 321G, 331, 331G, 335, 335G.** The SoundPoint IP 3 series is Polycom's entry level phones. The 321, 331, and corresponding G models support dual lines and Power over Ethernet (PoE). The 321 has one Ethernet port and the 331 has two Ethernet ports. This series supports 10/100 Megabits per second (Mbps). The 335 and 335G have the same feature set as the 331 with the addition of High-Definition (HD) Voice.

**Polycom SoundPoint IP 4 Series – 450 and 450G.** The SoundPoint IP 4 series is Polycom's mid-range model. It supports three lines, PoE, HD voice, and two Ethernet ports. It comes standard with a Liquid-Crystal Display (LCD). The 450 series supports 10/100 Mbps.

**Polycom SoundPoint IP 5 Series – 550, 550G, 560, 560G.** The SoundPoint IP 5 Series is a performance model supporting four lines, PoE, HD voice, and two Ethernet ports. They support a graphical LCD display in grayscale. The 550 series supports 10/100 Mbps and the 560 series supports 10/100/1000 Mbps.

**Polycom SoundPoint IP 6 Series – 650, 650G, 670, 670G.** The SoundPoint IP 6 Series is the high-performance model and supports six lines natively and up to 12 lines with an expansion module installed. This series supports PoE, HD voice, two Ethernet ports, and a Universal Serial Bus (USB) port to allow for local call recording. This series supports a graphical LCD display in color. The 650 series supports 10/100 Mbps and the 670 series supports 10/100/1000 Mbps.

**Polycom Expansion Module.** The expansion module can be added to the SoundPoint IP 5 and 6 Series phones to turn the phone into an attendant console. A

single phone has the capability to attach three modules, each allowing an additional 14 lines to be managed. The modules are hot swappable, plug and play, and require no additional power.

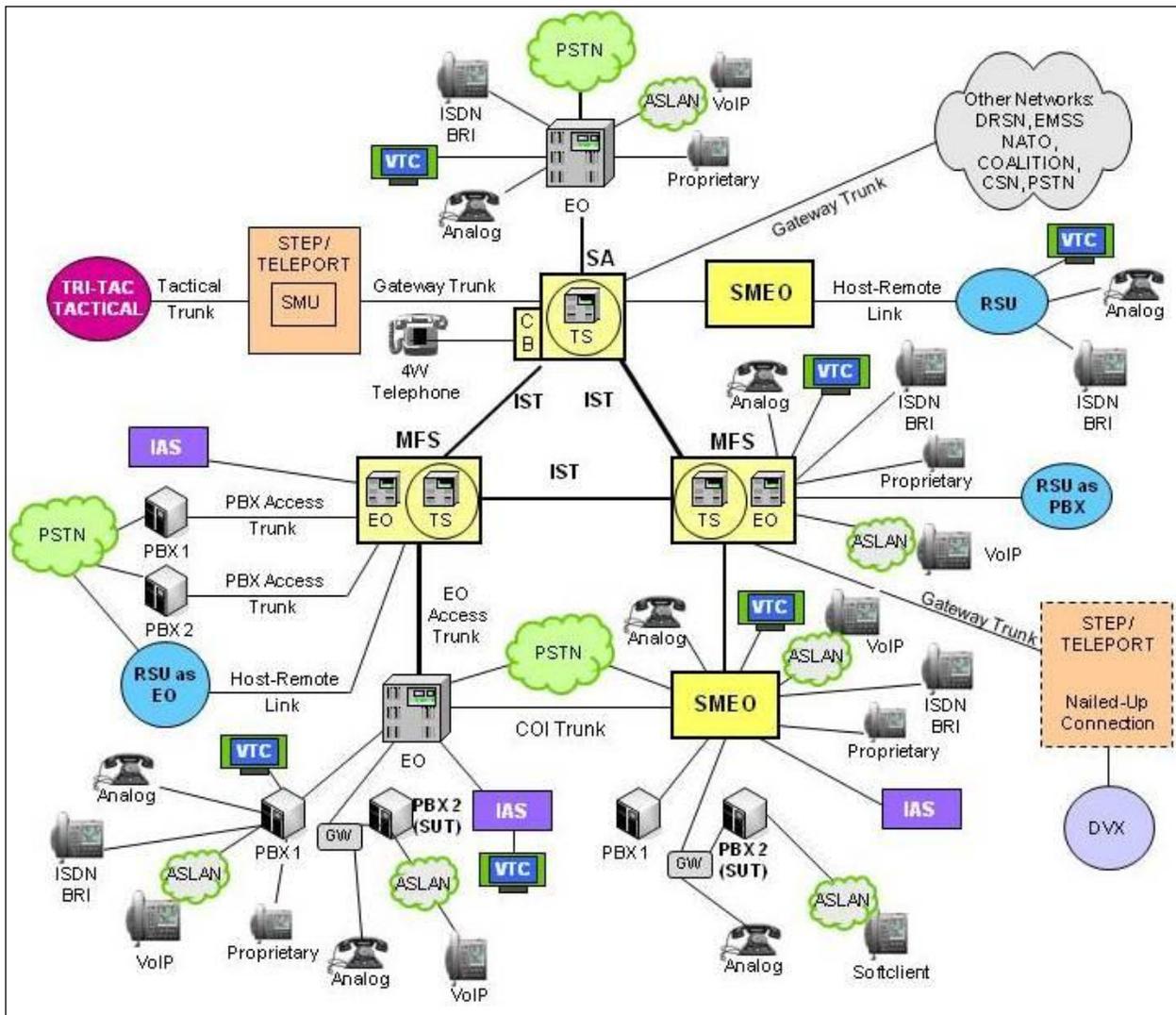
- **ININ-Vista or ININ-XP Management Workstation.** The ININ-Vista or ININ-XP Management Workstation is a site-provided workstation, which runs the Interaction Center User Application 3.0. The CIC 3.0 Client, which is included as part of the Interaction Center User Application, allows the user a set of features that can be defined based on roles. Some of the features include the ability to place calls based on call classifications such as local, long-distance, international, and 911. It also includes the right to control call pickup, forwarding, transferring, recording, multiple-line appearances, and access to the phonebook information, which includes AD users and extensions. The Interaction Center Business Manager Application 3.0 includes Interaction Supervisor, and may also run on the site-provided machine.

- **AudioCodes EMS Client-XP Management Workstation.** The AudioCodes EMS Client-XP is installed on the site-provided Microsoft XP Desktop. The EMS Client application coordinates with the EMS Server to manage the AudioCodes M3K and AudioCodes MP-124 elements providing a Graphical User Interface (GUI) for the user to manage and provision these VoIP network components.

- **AudioCodes MP-124.** The AudioCodes MP-124 is an analog VoIP gateway providing superior voice technology for connecting legacy telephones with IP-based telephony networks.

- **Analog Telephones.** The analog telephones are translated by the MP-124 into the VoIP network.

**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 PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 2s to the other DSN switch types.



**LEGEND:**

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

**Figure 2-1. DSN Architecture**

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

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

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

c. UCR PBX 2 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC.

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

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No		<ul style="list-style-type: none"> <li>• Direct Inward Dialing (C)</li> <li>• National ISDN 1/2 Primary Access (C)</li> <li>• ITU-T ISDN Primary Access (Europe only) (C)</li> <li>• Normal Wink Start Operations (C)</li> <li>• Glare Operation (C)</li> <li>• Abnormal Wink Start (C)</li> <li>• Glare Resolution (C)</li> <li>• Call for Service Timing (C)</li> <li>• Guard Timing (C)</li> <li>• Satellite Interface (C)</li> <li>• Disconnect Control (C)</li> <li>• Reselect and Retrial (C)</li> <li>• Off-Hook Supervision Transition (C)</li> <li>• Dial-Pulse Signals (C)</li> <li>• DTMF Signaling (C)</li> <li>• DSN ISDN User-to-Network Signaling (C)</li> <li>• Application (C)</li> <li>• Physical Layer (C)</li> <li>• Data Link Layer (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.1.3.2</li> <li>• UCR Section 5.2.1.3.4.1</li> <li>• UCR Section 5.2.1.3.4.2</li> <li>• UCR Section 5.2.4.3.3.1.1</li> <li>• UCR Section 5.2.4.3.3.1.2</li> <li>• UCR Section 5.2.4.3.3.2</li> <li>• UCR Section 5.2.4.3.3.2.2</li> <li>• UCR Section 5.2.4.3.5</li> <li>• UCR Section 5.2.4.3.6</li> <li>• UCR Section 5.2.4.3.7</li> <li>• UCR Section 5.2.4.3.8</li> <li>• UCR Section 5.2.4.3.9</li> <li>• UCR Section 5.2.4.3.10</li> <li>• UCR Section 5.2.4.4.1</li> <li>• UCR Section 5.2.4.4.2</li> <li>• UCR Section 5.2.4.7.1</li> </ul>
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Trunking	<ul style="list-style-type: none"> <li>• Data Link Connection (C)</li> <li>• Peer-to-Peer Procedures of Data-Link Layer (C)</li> <li>• Layer 3 DSN User-to-Network Signaling (C)</li> <li>• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)</li> <li>• Sequence of Messages for DSN Circuit-Switched Calls (C)</li> <li>• Message Functional Definition and Content (C)</li> <li>• General Message Format and Information Elements Coding (C)</li> <li>• Supplementary Services (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.1</li> <li>• UCR Section 5.2.4.7.1.2</li> <li>• UCR Section 5.2.4.7.1.3</li> <li>• UCR Section 5.2.4.7.1.3.1</li> <li>• UCR Section 5.2.4.7.1.3.2</li> <li>• UCR Section 5.2.4.7.1.4</li> <li>• UCR Section 5.2.4.7.1.4.2</li> <li>• UCR Section 5.2.4.7.1.4.3</li> <li>• UCR Section 5.2.4.7.1.4.4</li> <li>• UCR Section 5.2.4.7.1.4.5</li> </ul>
T1 ISDN PRI NI 1/2 (ANSI T1.607)	No		<ul style="list-style-type: none"> <li>• Transmission (R)</li> <li>• PCM-24 Digital Trunk Interface (R)</li> <li>• Interface Characteristics (R)</li> <li>• Supervisory Channel Associated Signaling (C)</li> <li>• Clear Channel Capability (C)</li> <li>• Alarm and Restoral Requirements (C)</li> <li>• PCM-30 Digital Trunk Interface (Europe only) (C)</li> <li>• Interoperation of PCM-24 and PCM-30 (C)</li> <li>• Analog Trunk Interface (C)</li> <li>• Integrated Digital Loop Carrier (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR Section 5.2.4.7.1.4.6</li> <li>• UCR Section 5.2.5</li> <li>• UCR Section 5.2.6.1</li> <li>• UCR Section 5.2.6.1.1</li> <li>• UCR Section 5.2.6.1.2</li> <li>• UCR Section 5.2.6.1.3</li> <li>• UCR Section 5.2.6.1.4</li> <li>• UCR Section 5.2.6.2</li> <li>• UCR Section 5.2.6.3</li> <li>• UCR Section 5.2.6.4</li> <li>• UCR Section 5.2.6.5</li> </ul>
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Voice	<ul style="list-style-type: none"> <li>• MOS (R)</li> <li>• Secure calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>• UCR 5.2.12.8.2.1</li> <li>• CJCSI 6215.01C</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>• Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>• DISR</li> </ul>
		Data	<ul style="list-style-type: none"> <li>• Modem (VBD) (R)</li> <li>• 56 kbps switched data (C: PRI only)</li> <li>• 64 kbps switched data (C: PRI only)</li> <li>• NX56 synchronous BER (C: PRI only)</li> <li>• NX64 synchronous BER (C: PRI only)</li> <li>• Secure data (STE/STU-III) (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• UCR Section 5.2.2.9.6</li> <li>• CJCSI 6215.01C</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>• FTR 1080B-2002</li> </ul>

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

DSN Line Interfaces				
Interface	Critical	Requirements Required or Conditional		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> <li>Individual Line (R)</li> <li>PBX Line (C)</li> <li>National ISDN 1/2 Basic Access (C)</li> <li>Analog Line (C)</li> </ul>	<ul style="list-style-type: none"> <li>UCR Section 5.2.1.1.1</li> <li>UCR Section 5.2.1.3.1</li> <li>UCR Section 5.2.1.3.3</li> <li>UCR Section 5.2.1.3.5</li> </ul>
ISDN BRI NI 1/2	No		<ul style="list-style-type: none"> <li>Loop Start Line (R: 2-Wire Analog only)</li> <li>Reverse Battery (C)</li> <li>S/T Reference Point (ISDN BRI) (C)</li> </ul>	<ul style="list-style-type: none"> <li>UCR Section 5.2.4.2.1</li> <li>UCR Section 5.2.4.3.1</li> <li>UCR Section 5.2.4.7.1.2.1</li> </ul>
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> <li>MOS (R)</li> <li>Secure Calls (C)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01C</li> <li>CJCSI 6215.01C</li> </ul>
		Facsimile	<ul style="list-style-type: none"> <li>Analog: ITU-T T.4 (R)</li> </ul>	<ul style="list-style-type: none"> <li>DISR</li> </ul>
		Data	<ul style="list-style-type: none"> <li>Modem (VBD) (R)</li> <li>Secure data (STE/STU-III) (C)</li> </ul>	<ul style="list-style-type: none"> <li>CJCSI 6215.01C</li> <li>CJCSI 6215.01C</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>FTR 1080B-2002</li> </ul>
DSN Features & Capabilities				
Feature/Capability	Critical	Requirements Required or Conditional		References
Common Features	Yes	<ul style="list-style-type: none"> <li>Individual Lines (R)</li> <li>Call waiting (C)</li> <li>Three-way calling (C)</li> <li>Add-on transfer, conference calling, and call hold (C)</li> <li>Call Transfer Individual – All calls (C)</li> <li>Call Transfer - Internal Only (C)</li> <li>Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (C)</li> <li>Call Transfer – Outside (C)</li> <li>Call Transfer – Add-On Restricted Station (C)</li> <li>Call Transfer – Attendant (C)</li> <li>Call Hold (C)</li> <li>Conference Calling – Six Way Station Controlled (C)</li> <li>Call forwarding Variable (C)</li> <li>Call Forward Busy Line (C)</li> <li>Call Forwarding – Don't Answer – All Calls (C)</li> <li>Selective Call Forwarding (C)</li> <li>Call pick-up (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.4.7.1.2.1</li> <li>UCR Section 5.2.1.1.5.1</li> <li>UCR Section 5.2.1.1.6</li> <li>UCR Section 5.2.1.1.7</li> <li>UCR Section 5.2.1.1.7.1</li> <li>UCR Section 5.2.1.1.7.2</li> <li>UCR Section 5.2.1.1.7.3</li> <li>UCR Section 5.2.1.1.7.3</li> <li>UCR Section 5.2.1.1.7.4</li> <li>UCR Section 5.2.1.1.7.5</li> <li>UCR Section 5.2.1.1.7.6</li> <li>UCR Section 5.2.1.1.7.7</li> <li>UCR Section 5.2.1.1.7.8</li> <li>UCR Section 5.2.1.1.8.1</li> <li>UCR Section 5.2.1.1.8.2</li> <li>UCR Section 5.2.1.1.8.3</li> <li>UCR Section 5.2.1.1.8.4</li> <li>UCR Section 5.2.1.1.9.1</li> </ul>
Attendant	No	<ul style="list-style-type: none"> <li>Attendant Features (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.1.2</li> </ul>
Public Safety	Yes	<ul style="list-style-type: none"> <li>Emergency Service (911) Caller (R)</li> <li>Emergency Service (911) Public Safety Answering Service (C)</li> <li>Enhanced Emergency Service (E911) (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.1.4.1.1</li> <li>UCR Section 5.2.1.4.1.2</li> <li>UCR Section 5.2.1.4.1.3</li> </ul>
Call Processing	Yes	<ul style="list-style-type: none"> <li>Origination Treatment (R)</li> <li>Originating Busy (R)</li> <li>Termination Treatment (R)</li> <li>Busy or Idle Status (C)</li> <li>Release Treatment (R)</li> <li>Interruption Treatment (R)</li> <li>Connections (R)</li> <li>Class of Service (C)</li> <li>E&amp;M Lead Signaling States (C)</li> <li>4-Wire Analog User Access Lines (C)</li> <li>2-Wire User Access Lines (C)</li> <li>Interswitch and Intraswitch Dialing (C)</li> <li>Calling Name Delivery (C)</li> <li>Calling Number Delivery (C)</li> <li>Screening (C)</li> </ul>		<ul style="list-style-type: none"> <li>UCR Section 5.2.3.1.1</li> <li>UCR Section 5.2.3.1.1.1</li> <li>UCR Section 5.2.3.1.2</li> <li>UCR Section 5.2.3.1.2.1</li> <li>UCR Section 5.2.3.1.3</li> <li>UCR Section 5.2.3.1.4</li> <li>UCR Section 5.2.3.1.5</li> <li>UCR Section 5.2.3.1.6</li> <li>UCR Section 5.2.3.3.1</li> <li>UCR Section 5.2.3.3.2</li> <li>UCR Section 5.2.3.3.3</li> <li>UCR Section 5.2.3.5.1.2</li> <li>UCR Section 5.2.3.5.1.8.1</li> <li>UCR Section 5.2.3.5.1.8.2</li> <li>UCR Section 5.2.3.5.8</li> </ul>

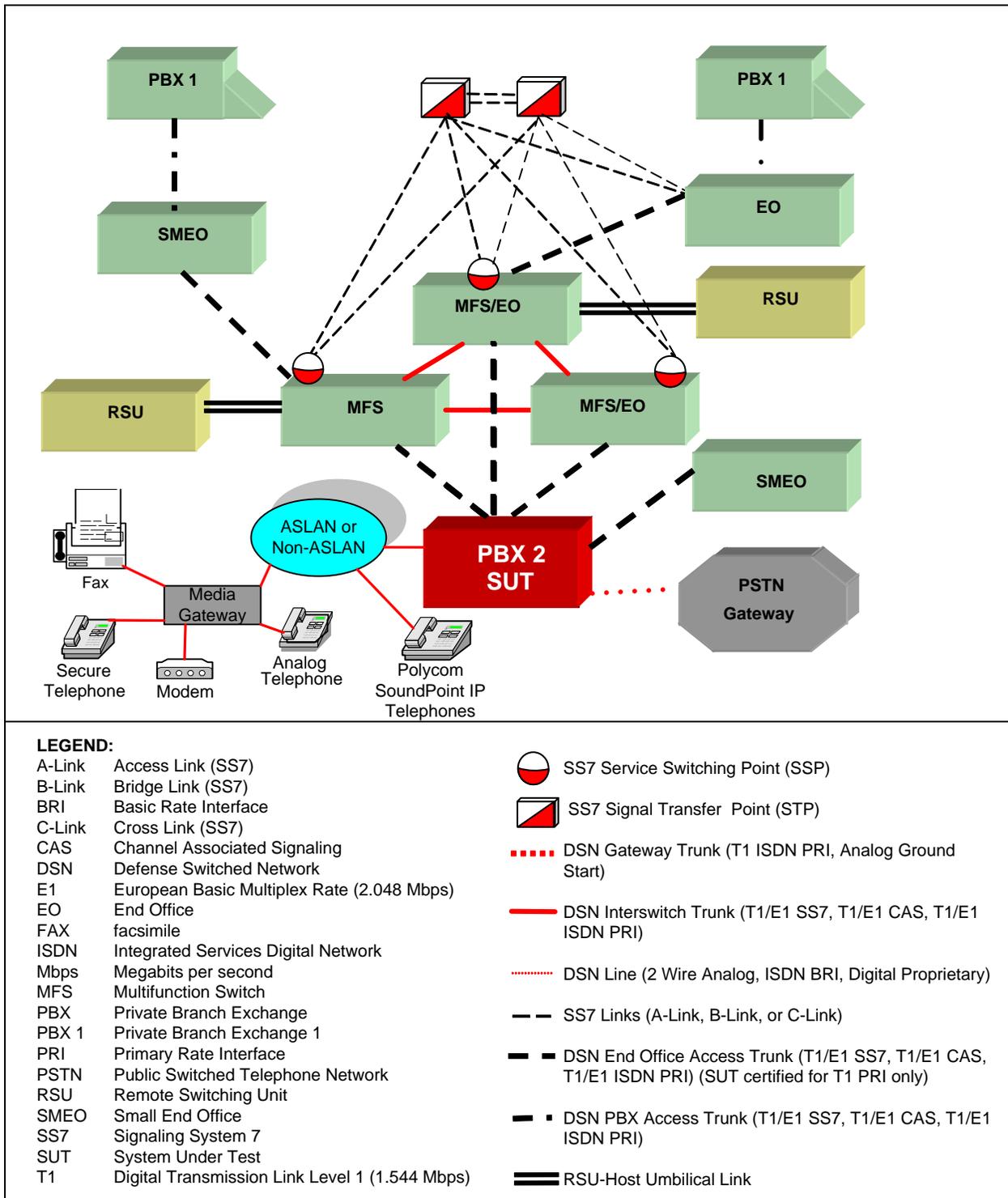
**Table 2-1. PBX 2 Requirements (continued)**

<b>DSN Features &amp; Capabilities (continued)</b>				
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
ISDN Services	No	<ul style="list-style-type: none"> <li>• BRI Access, Call Control and Signaling (C)</li> <li>• Uniform Interface Configuration for BRIs (C)</li> <li>• BRI Features (C)</li> <li>• PRI Access, Call Control and Signaling (C)</li> <li>• PRI Features (C)</li> <li>• Packet Data Features and Capabilities (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.9.2 Table 5.2.9.-1</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-2</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-3</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-4</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-5</li> <li>• UCR Section 5.2.9.2 Table 5.2.9-6</li> </ul>
Synchronization	Yes	<ul style="list-style-type: none"> <li>• Line timing mode (C)</li> <li>• Internal Stratum 4 (R)</li> <li>• Synchronization Performance Monitoring Criteria (C)</li> <li>• DS1 Traffic Interfaces (C)</li> <li>• DS0 Traffic Interconnects (C)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 5.2.10.1.1.2</li> <li>• UCR Section 5.2.10.1.2.2</li> <li>• UCR Section 5.2.10.2</li> <li>• UCR Section 5.2.10.3</li> <li>• UCR Section 5.2.10.4</li> </ul>
Security	Yes	<ul style="list-style-type: none"> <li>• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR Section 3</li> </ul>
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 (R)</li> <li>• ITU-T G.711 PCM CODEC (R)</li> <li>• MLPP (C)</li> <li>• Security (R)</li> <li>• Network management (C)</li> <li>• System timing (R)</li> <li>• Latency ≤ 60 milliseconds (R)</li> <li>• IPv6 capable (R)</li> <li>• Service Class Tagging (R)</li> </ul>		<ul style="list-style-type: none"> <li>• UCR section 5.2.12.8.2.1</li> <li>• UCR section 5.2.12.8.2.2</li> <li>• UCR section 5.2.12.8.2.3</li> <li>• UCR section 5.2.12.8.2.4</li> <li>• UCR section 5.2.12.8.2.5</li> <li>• UCR section 5.2.12.8.2.6</li> <li>• UCR section 5.2.12.8.2.7</li> <li>• UCR section 5.2.12.8.2.8</li> <li>• UCR section 5.2.12.8.2.9</li> </ul>
<b>Network Gateways</b>				
<b>Gateway</b>	<b>Critical</b>	<b>Requirements Required or Conditional</b>		<b>References</b>
PSTN	No	Trunking	<ul style="list-style-type: none"> <li>• Positive Identification Control (C)</li> <li>• On-Netting (C)</li> <li>• Off-Netting (C)</li> <li>• Loop Start Line (C)</li> <li>• Ground Start Line (C)</li> <li>• Immediate Start (C)</li> <li>• Delay Dial (C)</li> </ul>	<ul style="list-style-type: none"> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• CJCSI 6215.01C</li> <li>• UCR Section 5.2.4.2.1</li> <li>• UCR Section 5.2.4.2.2</li> <li>• UCR Section 5.2.4.3.2</li> <li>• UCR Section 5.2.4.3.4</li> </ul>

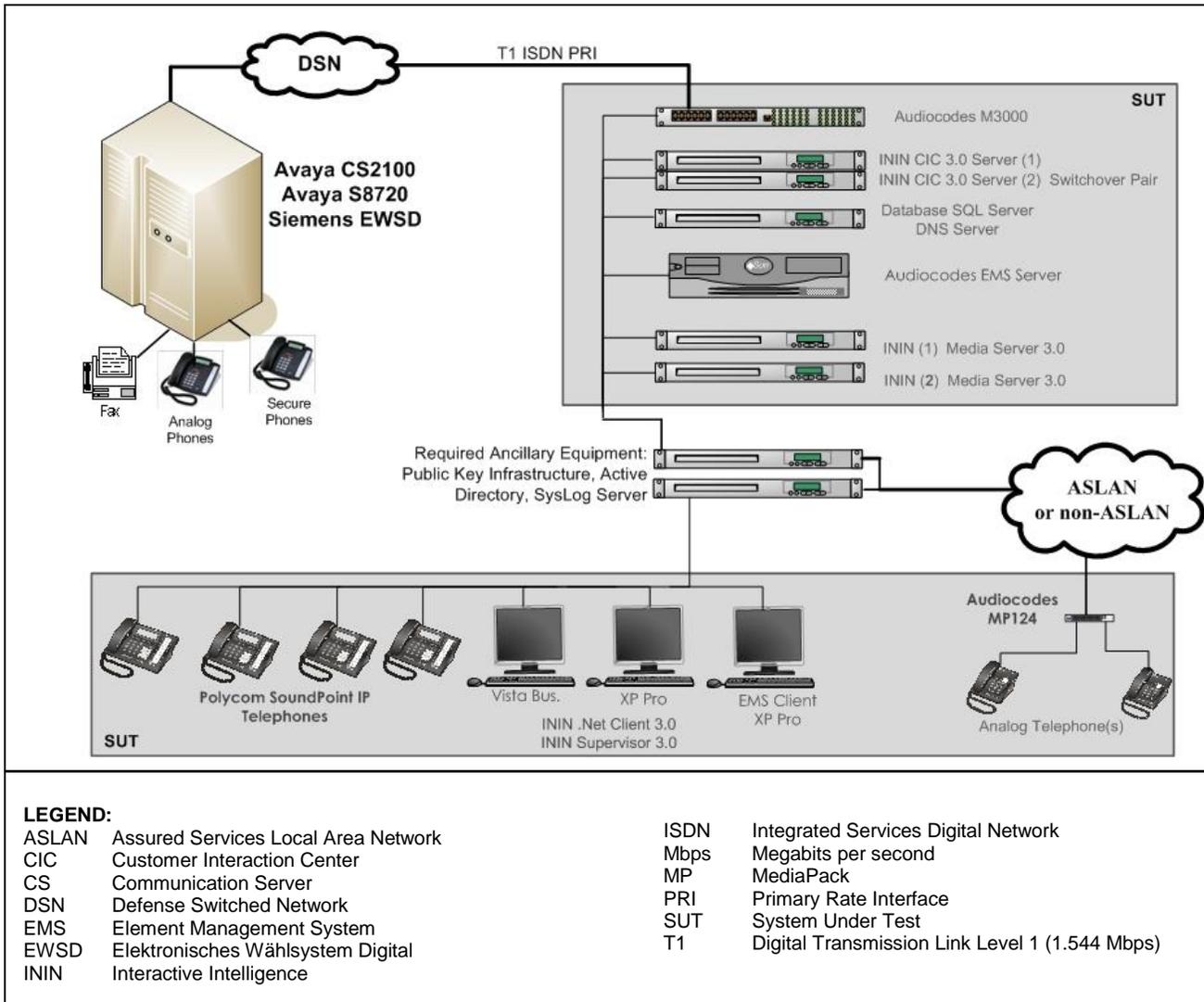
**Table 2-1. PBX 2 Requirements (continued)**

<b>LEGEND:</b>					
ANSI	American National Standards Institute	FTR 1080B	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	GR	Generic Requirement	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PRI	Primary Rate Interface
C	Conditional			PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling			Q.931	Signaling Standard for ISDN Required
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	H.320	Standard for Narrowband VTC	R	Required
DIACAP	DoD Information Assurance Certification and Accreditation Process	ISDN	Integrated Services Digital Network	S/T	ISDN BRI 4-wire interface
DISR	DoD IT Standards Registry	IT	Information Technology	STE	Secure Terminal Equipment
DoD	Department of Defense	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	STIGs	Security Technical Implementation Guides
DoDI	DoD Instruction			STU-III	Secure Telephone Unit -3rd generation
DP	Dial Pulse	kbps	kilobits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS0	Digital Signal Level 0	Mbps	Megabits per second	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	MFR1	Multi-Frequency Recommendation 1		
DSN	Defense Switched Network	MLPP	Multi-Level Precedence and Preemption		
DSS1	Digital Subscriber Signaling 1	MOS	Mean Opinion Score	T.4	Standardization of Group 3 facsimile terminals for document transmission
DTMF	Dual Tone Multi-Frequency	NI 1/2	National ISDN Standard 1 or 2	UCR	Unified Capabilities Requirements
E1	European Basic Multiplex Rate (2.048 Mbps)	NX56	Data format restricted to multiples of 56 kbps	VBD	Variable bit data
E911	Enhanced 911 Service	NX64	Data format restricted to multiples of 64 kbps	VoIP	Voice over Internet Protocol
E&M	Ear and Mouth			VTC	Video Teleconferencing
FTR	Federal Telecommunications Recommendation	PBX	Private Branch Exchange		
		PBX 2	Private Branch Exchange 2		

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



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



**Figure 2-3. SUT Test Configuration**

**9. SYSTEM CONFIGURATIONS.** Table 2-2 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 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

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

System Name		Software Release
Avaya CS2100		Succession Enterprise (SE) 09.1
Siemens EWSD		19d with Patch Set 46
Avaya S8720		Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)
Other Required Equipment		
Required Ancillary Equipment		Public Key Infrastructure, Active Directory, SysLog Server
Interactive Intelligence CIC 3.0 (SUT)	SUT Component	Component Hardware/Software
	M3K Media Gateway	M3K8410/DC – Simple Chassis: Mediant 3000 Rel. 5.60AV.007, pSOS 2.5
	CIC Server	HP DL 380 G5 Dual Core: CIC 3.09.11490, Internet Explorer 8, Microsoft .Net Framework 3.5 SP 1, MSXML 6.20.2003.0 SP2, Dialogic Host Media Processing (HMP) R2.0 SU 248, Interactive Intelligence HTTP Stack and OpenSLL Library 0.9.8, Windows Server 2003 SP2
	CIC Switchover Server	HP DL 380 G5 Dual Core: CIC 3.09.11490, Internet Explorer 8, Microsoft .Net Framework 3.5 SP 1, MSXML 6.20.2003.0 SP2, Dialogic Host Media Processing (HMP) R2.0 SU 248, Interactive Intelligence HTTP Stack and OpenSLL Library 0.9.8, Windows Server 2003 SP2
	Database Server	HP Server DL 320 G5: Microsoft SQL Sever 2008 Standard Edition (32 bit) 10.0.1600, Internet Explorer 8, Microsoft Data Access Component 2006.086.395, MSXML 6.20.2003.0 SP2, Microsoft .Net Framework 3.5 SP 1, Windows Server 2003 SP2
	AudioCodes EMS Server	SUN V.215 Server: Solaris 10 11/06 s10s_u3wos_10 SPARC, Java for Solaris 1.6.0_20-b02, Oracle 11.1.0.7.0, Java for Oracle 1.5.0_24-b02, OpenSSL 0.9.8n 24 Mar 2010, SSH Sun_SSH_1.1.3, Apache 2.2.14, Solaris Kerberos, EMS V5.8.83 + security patch bundle
	Media Server	HP Server DL 360 G5: Media Sever 3.0 SU9, Internet Explorer 8, Interactive Intelligence HTTP stack and OpenSLL Library 0.9.8, MSXML 6.20.2003.0 SP2, Microsoft .Net Framework 3.5 SP 1, Windows Server 2003 SP2
	ININ-Vista Management Workstation (site-provided)	Interaction Center User Application 3.0 SU9, Interaction Center Business Manager Application 3.0 SU9, Active Client CAC x86 (8.2.0.27), Tumbleweed Desktop Validator V4.10.0.344, Microsoft Visual C++ 2005 Redistributable, Microsoft Visual C++ 2005 ATL Update (KB973923 -x86 8.0.50727.4053), MSXML 4.0 SP2, Microsoft .Net Framework 3.5 SP1, Microsoft Windows Vista SP2
	ININ-XP Management Workstation (site-provided)	Interaction Center User Application 3.0 SU9, Interaction Center Business Manager Application 3.0 SU9, Active Client CAC x86 (8.2.0.27), Tumbleweed Desktop Validator V4.10.0.344, Microsoft Visual C++ 2005 Redistributable, Microsoft Visual C++ 2005 ATL Update (KB973923 -x86 8.0.50727.4053), MSXML 4.0 SP2, Microsoft .Net Framework 3.5 SP1, Microsoft Windows XP Professional SP3
	AudioCodes EMS Client-XP Management Workstation (site-provided)	Interaction Center User Application 3.0 SU9, Interaction Center Business Manager Application 3.0 SU9, Active Client CAC x86 (8.2.0.27), Tumbleweed Desktop Validator V4.10.0.344, Microsoft Visual C++ 2005 Redistributable, Microsoft Visual C++ 2005 ATL Update (KB973923 -x86 8.0.50727.4053), MSXML 4.0 SP2, Microsoft .Net Framework 3.5 SP1, Microsoft Windows XP Professional SP3
MP-124	Mediant MP-124/FXS Gateway: MP-124/FXS 5.60AM.025.011, pSOS 2.5	

**Table 7. Tested System Configuration (continued)**

SUT Telephone Telephones		
Telephone type	Model	Software/Firmware
Polycom SoundPoint IP 3/4/5/6 Series	IP331	Firm. ver. 3.3.0.1098 / boot code 4.2.0.0310
	IP450	
	IP550	
	IP670	
	IP670 Extension Module	
Analog	Generic	NA
<b>LEGEND:</b> CAC Common Access Card MP MediaPack CIC Contact Interaction Center MS Microsoft MSXML Microsoft XML Core Services CS Communication Server NA Not Applicable EMS Element Management System OpenSSL Open Secure Sockets Layer EWSD Elektronisches Wählsystem Digital SP Service Pack Firm. Firmware SPARC Sun Microsystems Architecture FXS Foreign Exchange Station SSH Secure Shell HP Hewlett-Packard SU Service Update HTTP Hypertext Transfer Protocol V Version ININ Interactive Intelligence Corp Ver. Version IP Internet Protocol XML Extensible Markup Language KB Knowledge Base XP Experience		

**10. TESTING LIMITATIONS.** None.

**11. TEST RESULTS**

**a. Discussion.** All requirements were met through testing and review of the vendor's LoC.

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs in accordance with UCR 2008 Section 5.2.1.3 for T1 ISDN PRI National ISDN (NI) 1/2 (American National Standards Institute [ANSI] T1.607) interfaces with the following minor exception: After a non-secure call is established, the users can go secure. Secure calls fail at random intervals with no known explanation. After a secure call fails, the users can reestablish the non-secure call without hanging up by pressing the non-secure button. Since secure call capability is a conditional requirement for a PBX 2, there is minor operational impact.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE) in accordance with UCR 2008 Section 5.2.1.3 and VoIP DSN line interfaces in accordance with UCR 2008 Section 5.2.12.8.2.

(3) DSN Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Common Features.

(b) Attendant. The SUT met all critical CRs and FRs.

(c) Public Safety. The SUT met all critical CRs and FRs for the only required Public Safety feature which is basic 911.

(d) Call Processing. The SUT met all critical CRs and FRs.

(e) ISDN Services. The SUT met all critical CRs and FRs with the T1 ISDN PRI interface.

(f) Synchronization. The SUT met all critical CRs and FRs. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

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

(4) VoIP. The SUT is certified with any Assured Services Local Area Network (ASLAN) or non-ASLAN on the UC APL.

(a) VoIP System. The UCR, paragraph 5.2.12.8.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, paragraph 5.2.12.8.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and from handset to gateway trunk in the DSN. The SUT meets MOS requirements with an average of 4.37.

2. Codec. In accordance with the UCR, paragraph 5.2.12.8.2.2, the ITU-T G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill is required and was met by the SUT VoIP solution.

3. MLPP. In accordance with the UCR, paragraph 5.2.12.8.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, Section 5.2.2. This feature is not supported by the SUT and is conditional for a PBX 2. There is no risk associated with the SUT not supporting this feature.

4. Security. Security requirements in accordance with the UCR, paragraph 5.2.12.8.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, Reference (e).

5. Network Management (NM). In accordance with the UCR, paragraph 5.2.12.8.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement

was verified via a LoC because of the numerous third party systems and applications capable of performing this function. The SUT is certified with any ASLAN or ASLAN components on the UC APL. The ASLAN management system is covered under the ASLAN report.

6. Synchronization. In accordance with the UCR, paragraph 5.2.12.8.2.6, the VoIP system shall meet all synchronization requirements identified in UCR, paragraph 5.2.10. The SUT derived synchronization with line timing mode via traditional T1 TDM-based interfaces. Line side timing was provided by the SUT.

3. Latency. The UCR, paragraph 5.2.12.8.2.7, states that one-way system latency for the VoIP system must be 60ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress trunk. The SUT meets this requirement with a measured maximum delay of 51 ms. The minimum delay seen was 46 ms.

4. Internet Protocol version 6 (IPv6). In accordance with UCR, section 5.3.5, all systems submitted for testing must be IPv6 capable. Dual Stack solutions are preferred and tunneling solutions are unacceptable. IPv6 Capable-products, in accordance with UCR, section 4.3.1.3, can create or receive, process, and send or forward (as appropriate) IPv6 packets in mixed Internet Protocol version 4 (IPv4)/IPv6 environments. IPv6 capable products shall be able to interoperate with other IPv6 capable products on networks supporting only IPv4, only IPv6, or both IPv4 and IPv6, and shall also:

a. Conform to the requirements of the Department of Defense (DoD) IPv6 Standard Profiles for IPv6 Capable Products document contained in the DoD Information Technology Standards Registry (DISR).

b. Possess a migration path and/or written commitment to upgrade from the developer (company Vice President or equivalent) as the IPv6 standard evolves.

c. Ensure product developer IPv6 technical support is available.

d. Conform to National Security Agency (NSA) and/or Unified Cross Domain Management Office requirements for Information Assurance products.

The SUT does not support IPv6. The vendor was granted a waiver by ASD-NII on 27 February 2011 with the following stipulation: The SUT must demonstrate IPv6 capability by 31 August 2011.

5. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP session control components (i.e. Media Gateway and Session Control Agent) shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 6-bit traffic class in the IPv6 header and DSCPs field in the IPv4 header. Packets were DSCP tagged with 46 decimal for media and 48 decimal for signaling from the media gateway. The SUT session control components used 6-bit service class tagging in the IP header, which meets the requirement for the media gateways. The phones incorrectly used a different value for signaling. Operations Administration and Management (OAM) traffic was either tagged at zero or 34 decimal. This was previously adjudicated by DISA as having a minor operational impact.

b. All session control components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. In accordance with the UCR, the DSCP field of the IP traffic associated with the distinct service classes of the session control components can be assigned a unique value by the SUT which meets this requirement.

c. For VoIP, video, and data end products, any end system that supports convergence (i.e., more than one media) the end-system must pre-assign the virtual LAN (VLAN) using Institute of Electrical and Electronics Engineers (IEEE) 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media (i.e., voice or video or data), the LAN can assign the VLAN based on port-based VLAN assignment. The SUT VoIP session control components provide IEEE 802.1Q 2-byte TGI VID, which meets the requirement. In addition, although the SUT is not required to support the IEEE 802.1Q Priority header information, the SUT session control components have arbitrary values coded in them. DISA adjudicated this as minor on 2 September 2009. Phones which offered shared access did not put voice traffic in a separate VLAN as required. The SUT is not certified for shared access, which is not required.

6. In accordance with the UCR, section 5.2.12.8.2.9, the VoIP system end user devices shall meet the following requirements:

a. All end instrument components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. The DSCPs may be assigned by either having the end instrument itself assign the traffic class and DSCP tag to the distinct service class or having the call control portion of the VoIP system tell the end instrument what distinct service class to assign. The SUT end instruments have the capability to be assigned any DSCP value of 0-63.

b. All end instruments shall be capable of implementing Service Class tagging using the 6-bit traffic class in the IPv6 header and DSCPs field in the IPv4 header. The SUT end instruments did not support IPv6 dual stack. The SUT does not support IPv6. The SUT does not support IPv6. The vendor was granted a waiver by ASD-NII on 27 February 2011 with the following stipulation: The SUT must demonstrate IPv6 capability by 31 August 2011.

c. For VoIP, video, and data end products, any end system that supports convergence (i.e., more than one media) the end-system must pre-assign the VLAN using IEEE 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media (i.e., voice or video or data), the LAN can assign the VLAN based on port-based VLAN assignment. No VLAN tagging was seen in any captures from any IP phone. The SUT is not certified for shared access, which is not required.

(b) Scalability. The SUT can support up to 15,000 ports users.

**(5) Network Gateways.** The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateways with the T1 ISDN PRI and Loop Start Line interfaces.

**b. System Interoperability Results.** The SUT is certified for joint use in the Defense Information System Network (DISN) as a PBX 2 in accordance with the requirements set forth in the UCR. The SUT is certified with any certified ASLAN or ASLAN components on the UC APL. The interoperability test summary is shown in Table 2-3. The SUT Interoperability Requirements/Status is shown in Table 2.

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

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF)	No <sup>1</sup>	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
E1 CAS (DTMF)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
T1 ISDN PRI NI 1/2 (ANSI T1.607)	No <sup>1</sup>	Certified	Met all critical CRs and FRs with the following minor exception: Secure voice calls fail at random intervals. <sup>2</sup>
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs.
ISDN BRI NI 1/2	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
VoIP (Ethernet IEEE 802.3u)	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT did not meet the IPv6 capability requirements. <sup>3</sup> The SUT incorrectly tagged OAM traffic. <sup>4</sup> The SUT does not support proper VLAN tagging with IP phones. The SUT is not certified for shared access. <sup>5</sup>

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

DSN Features and Capabilities				
Feature/Capability	Critical	Status	Remarks	
Common Features	Yes	Certified	Met all critical CRs and FRs.	
Attendant	No	Certified	Met all critical CRs and FRs.	
Public Safety	Yes	Certified	Met all critical CRs and FRs for the basic 911.	
Call Processing	Yes	Certified	Met all critical CRs and FRs.	
ISDN Services	No	Certified	Met all critical CRs and FRs with the T1 ISDN PRI interface.	
Synchronization	Yes	Certified	Met all critical CRs and FRs.	
VoIP System	No	Certified	Met all critical CRs and FRs with the following minor exception: The SUT did not meet the IPv6 capability requirements. <sup>3</sup>	
Security	Yes	Certified	See note 6.	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF)	No <sup>1</sup>	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	E1 CAS (DTMF)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No <sup>1</sup>	Certified	Met all critical CRs and FRs.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	Ground Start Line (GR-506-CORE)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	Loop Start Line (GR-506-CORE)	No	Certified	Met all critical CRs and FRs with an LOC.
<b>NOTES:</b>				
1 The UCR 2008 does not specify a required interface for a PBX 2. A PBX 2 switch must support at least T1 CAS or T1 PRI.				
2 After a non-secure call is established, the users can go secure. Secure calls fail at random intervals from two to eight minutes with no known explanation. After a secure call fails, the users can reestablish the non-secure call without hanging up by pressing the non-secure button. Since secure call capability is a conditional requirement for a PBX 2, there is minor operational impact.				
3 The SUT does not support IPv6. The vendor was granted a waiver by ASD-NII on 27 February 2011 with the following stipulation: The SUT must demonstrate IPv6 capability by 31 August 2011.				
4 The phones incorrectly used a different value for signaling. OAM traffic was either tagged at zero or 34 decimal. This was previously adjudicated by DISA as having a minor operational impact.				
5 Phones which offered shared access did not put voice traffic in a separate VLAN as required. The SUT is not certified for shared access, which is not required.				
6 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (e).				

<b>LEGEND:</b>			
802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps	ISDN	Integrated Services Digital Network
ANSI	American National Standards Institute	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
ASD/NII	Assistant Secretary of Defense for Networks and Information Integration	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
BRI	Basic Rate Interface	Mbps	Megabits per second
CAS	Channel Associated Signaling	NI 1/2	National ISDN Standard 1 or 2
CRs	Capability Requirements	OAM	Operations Administration and Management
DISA	Defense Information Systems Agency	PBX 2	Private Branch Exchange 2
DSN	Defense Switched Network	PRI	Primary Rate Interface
DSS1	Digital Subscriber Signaling 1	PSTN	Public Switched Telephone Network
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	SUT	System Under Test
FRs	Feature Requirements	T1	Digital Transmission Link Level 1 (1.544 Mbps)
GR	Generic Requirement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	UCR	Unified Capabilities Requirements
IEEE	Institute for Electrical and Electronics Engineers	VoIP	Voice over Internet Protocol
IPv6	Internet Protocol version 6		

**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). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at <http://jitc.fhu.disa.mil/tssi>. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: [ucco@disa.mil](mailto:ucco@disa.mil).

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
T1 CAS (DTMF)	No	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (C)	UCR Section 5.2.4.3.3.2	Not Tested	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Glare Resolution (C)	UCR Section 5.2.4.3.5	Not Tested	
				Call for Service Timing (C)	UCR Section 5.2.4.3.6	Not Tested	
				Guard Timing (C)	UCR Section 5.2.4.3.7	Not Tested	
				Satellite Interface (C)	UCR Section 5.2.4.3.8	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.9	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.10	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.4.1	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.2	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.7.1	Not Tested	
				Transmission (R)	UCR Section 5.2.5	Not Tested	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Not Tested	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Not Tested	
				Supervisory Channel Associated Signaling (C)	UCR Section 5.2.6.1.2	Not Tested	
			Clear Channel Capability (C)	UCR Section 5.2.6.1.3	Not Tested		
			Alarm and Restoral Requirements (C)	UCR Section 5.2.6.1.4	Not Tested		
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (C)	CJCSI 6215.01C	Not Tested	
Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested				
Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested				
	56 kbps switched data (C)	UCR Section 5.2.2.9.6	Not Tested				
	NX56 synchronous BER (C)	UCR Section 5.2.2.9.6	Not Tested				
	Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Not Tested				

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
E1 CAS (DTMF)	No (Europe only)	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Not Tested	
				Line Signaling (R)	UCR Section 5.2	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.2.4.3.3.1.1	Not Tested	
				Glare Operation (C)	UCR Section 5.2.4.3.3.1.2	Not Tested	
				Abnormal Wink Start (C)	UCR Section 5.2.4.3.3.2	Not Tested	
				Glare Resolution (C)	UCR Section 5.2.4.3.3.2.2	Not Tested	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Not Tested	
				Guard Timing (C)	UCR Section 5.2.4.3.6	Not Tested	
				Satellite Interface (C)	UCR Section 5.2.4.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.2.4.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.2.4.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.2.4.4.2	Not Tested	
			Transmission (R)	UCR Section 5.2.5	Not Tested		
			PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested		
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested		
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (C)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				56 kbps switched data (C)	UCR Section 3.10	Not Tested	
				64 kbps switched data (C)	UCR Section 3.10	Not Tested	
NX56 synchronous BER (C)	UCR Section 3.10	Not Tested					
NX64 synchronous BER (C)	UCR Section 3.10	Not Tested					
	Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Not Tested				

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
T1 ISDN PRI NI 1/2 (ANSI T1.607)	Yes	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Met	
				National ISDN 1/2 Primary Access (C)	UCR Section 5.2.1.3.4.1	Met	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Met	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Met	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Met	
				DSN ISDN User-to-Network Signaling (C)	UCR Section 5.2.4.7.1	Met	
				Application (C)	UCR Section 5.2.4.7.1.1	Met	
				Physical Layer (C)	UCR Section 5.2.4.7.1.2	Met	
				Data Link Layer (C)	UCR Section 5.2.4.7.1.3	Met	
				Data Link Connection (C)	UCR Section 5.2.4.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (C)	UCR Section 5.2.4.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (C)	UCR Section 5.2.4.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)	UCR Section 5.2.4.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (C)	UCR Section 5.2.4.7.1.4.3	Met	
				Message Functional Definition and Content (C)	UCR Section 5.2.4.7.1.4.4	Met	
				General Message Format and Information Elements Coding (C)	UCR Section 5.2.4.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.2.4.7.1.4.6	Met	
				Transmission (R)	UCR Section 5.2.5	Met	
				PCM-24 Digital Trunk Interface (R)	UCR Section 5.2.6.1	Met	
				Interface Characteristics (R)	UCR Section 5.2.6.1.1	Met	
			Clear Channel Capability (C)	UCR Section 5.2.6.1.3	Met		
			Alarm and Restoral Requirements (C)	UCR Section 5.2.6.1.4	Met		
			Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	See note 2.	
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested	See note 2.	
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (C)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
	Modem (VBD) (R)	CJCSI 6215.01C	Met				
	56 kbps switched data (C: PRI only)	UCR Section 5.2.2.9.6	Met				
	64 kbps switched data (C: PRI only)	UCR Section 5.2.2.9.6	Met				
	NX56 synchronous BER (C: PRI only)	UCR Section 5.2.2.9.6	Met				
	NX64 synchronous BER (C: PRI only)	UCR Section 5.2.2.9.6	Met				
	Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Met				
VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 2.			

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

DSN Trunk Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 5.2.1.3.2	Not Tested	
				ITU-T ISDN Primary Access (C)	UCR Section 5.2.1.3.4.2	Not Tested	
				Call for Service Timing (C)	UCR Section 5.2.4.3.5	Not Tested	
				Disconnect Control (C)	UCR Section 5.2.4.3.8	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.2.4.3.10	Not Tested	
				DSN ISDN User-to-Network Signaling (C)	UCR Section 5.2.4.7.1	Not Tested	
				Application (C)	UCR Section 5.2.4.7.1.1	Not Tested	
				Physical Layer (C)	UCR Section 5.2.4.7.1.2	Not Tested	
				Data Link Layer (C)	UCR Section 5.2.4.7.1.3	Not Tested	
				Data Link Connection (C)	UCR Section 5.2.4.7.1.3.1	Not Tested	
				Peer-to-Peer Procedures of Data-Link Layer (C)	UCR Section 5.2.4.7.1.3.2	Not Tested	
				Layer 3 DSN User-to-Network Signaling (C)	UCR Section 5.2.4.7.1.4	Not Tested	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)	UCR Section 5.2.4.7.1.4.2	Not Tested	
				Sequence of Messages for DSN Circuit-Switched Calls (C)	UCR Section 5.2.4.7.1.4.3	Not Tested	
				Message Functional Definition and Content (C)	UCR Section 5.2.4.7.1.4.4	Not Tested	
				General Message Format and Information Elements Coding (C)	UCR Section 5.2.4.7.1.4.5	Not Tested	
				Supplementary Services (C)	UCR Section 5.2.4.7.1.4.6	Not Tested	
				Transmission (R)	UCR Section 5.2.5	Not Tested	
				PCM-30 Digital Trunk Interface (C)	UCR Section 5.2.6.2	Not Tested	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 5.2.6.3	Not Tested	
			Integrated Digital Loop Carrier (C)	UCR Section 5.2.6.5	Not Tested		
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (C)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				56 kbps switched data (C: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
				64 kbps switched data (C: PRI only)	UCR Section 5.2.2.9.6	Not Tested	
NX56 synchronous BER (C: PRI only)	UCR Section 5.2.2.9.6	Not Tested					
NX64 synchronous BER (C: PRI only)	UCR Section 5.2.2.9.6	Not Tested					
	Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Not Tested				
VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested				

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

DSN Line Interfaces							
Interface	Critical	Interface Status	UCR Requirement		Reference	Test Results	Remarks
2-Wire Analog	Yes	Certified	Access	Individual Line (R)	UCR Section 5.2.1.1.1	Met	
				PBX Line (C)	UCR Section 5.2.1.3.1	Met	
				Analog Line (C)	UCR Section 5.2.1.3.5	Met	
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.4.2.1	Met	
				Reverse Battery (C)	UCR Section 5.2.4.3.1	Met	See note 2.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (C)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Met					
ISDN BRI NI 1/2 (ANSI T1.607)	No	Not Tested (See note 1.)	Access	Individual Line (R)	UCR Section 5.2.1.1.1	Not Tested	
				National ISDN 1/2 Basic Access (C)	UCR Section 5.2.1.3.3	Not Tested	
				S/T Reference Point (R)	UCR Section 5.2.4.7.1.2.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (C)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
Secure data (STE/STU-III) (C)	CJCSI 6215.01C	Not Tested					
VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Not Tested				
2-Wire Proprietary Digital	No	Not Tested (See note 1.)	Access	Individual Line (R)	UCR Section 5.2.1.1.1	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	

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

DSN Features and Capabilities						
Feature/ Capability	Critical	Feature Status	UCR Requirement	Reference	Test Results	Remarks
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 5.2.4.7.1.2.1	Met	
			Call waiting (C)	UCR Section 5.2.1.1.5.1	Met	
			Three-way calling (C)	UCR Section 5.2.1.1.6	Met	
			Add-on transfer, conference calling, and call hold (C)	UCR Section 5.2.1.1.7	Met	
			Call Transfer Individual – All calls (C)	UCR Section 5.2.1.1.7.1	Met	
			Call Transfer - Internal Only (C)	UCR Section 5.2.1.1.7.2	Met	
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (C)	UCR Section 5.2.1.1.7.3	Not Tested	See note 2.
			Call Transfer – Outside (C)	UCR Section 5.2.1.1.7.4	Not Tested	See note 2.
			Call Transfer – Add-On Restricted Station (C)	UCR Section 5.2.1.1.7.5	Not Tested	See note 2.
			Call Transfer – Attendant (C)	UCR Section 5.2.1.1.7.6	Met	
			Call Hold (C)	UCR Section 5.2.1.1.7.7	Met	
			Conference Calling – Six Way Station Controlled (C)	UCR Section 5.2.1.1.7.8	Met	
			Call Forwarding Variable (C)	UCR Section 5.2.1.1.8.1	Met	
			Call Forward Busy Line (C)	UCR Section 5.2.1.1.8.2	Not Met	See note 3.
			Call Forwarding – Don't Answer – All Calls (C)	UCR Section 5.2.1.1.8.3	Met	
Selective Call Forwarding (C)	UCR Section 5.2.1.1.8.4	Met				
Call pick-up (C)	UCR Section 5.2.1.1.9.1	Met				
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 5.2.1.2	Met	
Public Safety	Yes	Certified	Emergency Service (911) Caller (R)	UCR Section 2.4.1.1	Met	
			Emergency Service (911) Public Safety Answering Service (C)	UCR Section 2.4.1.2	Not Tested	See note 2.
			Enhanced Emergency Service (E911) (C)	UCR Section 2.4.1.3	Not Tested	See note 2.
Call Processing	Yes	Certified	Origination Treatment (R)	UCR Section 5.2.3.1.1	Met	
			Originating Busy (R)	UCR Section 5.2.3.1.1.1	Met	
			Termination Treatment (R)	UCR Section 5.2.3.1.2	Met	
			Busy or Idle Status (C)	UCR Section 5.2.3.1.2.1	Met	
			Release Treatment (C)	UCR Section 5.2.3.1.3	Met	
			Interruption Treatment (C)	UCR Section 5.2.3.1.4	Met	
			Connections (R)	UCR Section 5.2.3.1.5	Met	
			Class of Service (C)	UCR Section 5.2.3.1.6	Not Tested	See note 2.
			E&M Lead Signaling States (C)	UCR Section 5.2.3.3.1	Not Tested	See note 2.
			4-Wire Analog User Access Lines (C)	UCR Section 5.2.3.3.2	Met	
			2-Wire User Access Lines (ROUTINE Only) (C)	UCR Section 5.2.3.3.3	Met	
			Interswitch and Intraswitch Dialing (C)	UCR Section 5.2.3.5.1.2	Met	
			Calling Name Delivery (C)	UCR Section 5.2.3.5.1.8.1	Not Tested	See note 2.
Calling Number Delivery (C)	UCR Section 5.2.3.5.1.8.2	Not Tested	See note 2.			
Screening (C)	UCR Section 5.2.3.5.8	Met				

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

<b>DSN Features and Capabilities</b>							
<b>Feature/ Capability</b>	<b>Critical</b>	<b>Feature Status</b>	<b>UCR Requirement</b>	<b>Reference</b>	<b>Test Results</b>	<b>Remarks</b>	
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (C)	UCR Section 5.2.9.2 Table 5.2.9.-1	Not Tested	See note 2.	
			Uniform Interface Configuration for BRIs (C)	UCR Section 5.2.9.2 Table 5.2.9.-2	Not Tested	See note 2.	
			BRI Features (C)	UCR Section 5.2.9.2 Table 5.2.9.-3	Not Tested	See note 2.	
			PRI Access, Call Control and Signaling (R)	UCR Section 5.2.9.2 Table 5.2.9.-4	Met		
			PRI Features (R)	UCR Section 5.2.9.2 Table 5.2.9.-5	Met		
			Packet Data Features and Capabilities (C)	UCR Section 5.2.9.2 Table 5.2.9.-6	Met		
Synchroniz- ation	Yes	Certified	Line timing mode (C)	UCR Section 5.2.10.1.1.2	Met		
			Internal Stratum 4 (R)	UCR Section 5.2.10.1.2.2	Met		
			Synchronization Performance Monitoring Criteria (C)	UCR Section 5.2.10.2	Met		
			DS1 Traffic Interfaces (C)	UCR Section 5.2.10.3	Met		
			DS0 Traffic Interconnects (C)	UCR Section 5.2.10.4	Met		
Security	Yes	Certified	GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 3	Met	See note 4.	
VoIP System	No	Certified	Voice Quality with MOS of 4.0 or better (R)	UCR Section 5.2.12.8.2.1	Met		
			ITU-T G.711 PCM CODEC (R)	UCR Section 5.2.12.8.2.2	Met		
			MLPP (C)	UCR Section 5.2.12.8.2.3	Met		
			Security (R)	UCR Section 5.2.12.8.2.4	Met		
			Network management (C)	UCR Section 5.2.12.8.2.5	Met		
			System timing (R)	UCR Section 5.2.12.8.2.6	Met		
			Latency ≤ 60 milliseconds (R)	UCR Section 5.2.12.8.2.7	Met		
			IPv6 capable (R)	UCR Section 5.2.12.8.2.8	Not Tested	See note 5.	
Service Class Tagging (R)	UCR Section 5.2.12.8.2.9	Met	See notes 6 and 7.				
<b>Network Gateways</b>							
PSTN	No	Certified	Trunking	Positive Identification Control (C)	CJCSI 6215.01C	Met	
				On-Netting (C)	CJCSI 6215.01C	Met	
				Off-Netting (C)	CJCSI 6215.01C	Met	
				Loop Start Line (C)	UCR Section 5.2.4.2.1	Met	
				Ground Start Line (C)	UCR Section 5.2.4.2.2	Not Tested	See note 2.
				Immediate Start (C)	UCR Section 5.2.4.3.2	Not Tested	See note 2.
				Delay Dial (C)	UCR Section 5.2.4.3.4	Not Tested	See note 2.

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

**NOTES:**

- 1 This interface is not supported by the SUT and is not required for a PBX 2.
- 2 This capability/feature is not supported by the SUT and is not required for a PBX 2.
- 3 Implementing this feature would require a configuration change. The configuration change was not tested; therefore, this feature is not covered under this certification. This is not required for a PBX 2.
- 4 Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (e).
- 5 The SUT does not support IPv6. The vendor was granted a waiver by ASD-NII on 27 February 2011 with the following stipulation: The SUT must demonstrate IPv6 capability by 31 August 2011.
- 6 The phones incorrectly used a different value for signaling. OAM traffic was either tagged at zero or 34 decimal. This was previously adjudicated by DISA as having a minor operational impact.
- 7 Phones which offered shared access did not put voice traffic in a separate VLAN as required. The SUT is not certified for shared access, which is not required.

**LEGEND:**

ANSI	American National Standards Institute	FTR	Federal Telecommunications Recommendation	PCM	Pulse Code Modulation
ASD/NII	Assistant Secretary of Defense for Networks and Information Integration	FTR 1080B	Video Teleconferencing Services	PCM-24	Pulse Code Modulation - 24 Channels
BER	Bit Error Ratio	G.711	PCM of voice frequencies	PCM-30	Pulse Code Modulation - 30 Channels
BRI	Basic Rate Interface	GR	Generic Requirement	PRI	Primary Rate Interface
C	Conditional	GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security	PSTN	Public Switched Telephone Network
CAS	Channel Associated Signaling			Q.931	Signaling Standard for ISDN
CJCSI	Chairman of the Joint Chiefs of Staff Instruction	H.320	Standard for Narrowband VTC	R	Required
DIACAP	DoD Information Assurance Certification and Accreditation Process	IPv6	Internet Protocol version 6	S/T	ISDN BRI 4-wire interface
DISA	Defense Information Systems Agency	ISDN	Integrated Services Digital Network	STE	Secure Terminal Equipment
DISR	DoD IT Standards Registry	IT	Information Technology	STIGs	Security Technical Implementation Guides
DoD	Department of Defense	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	STU-III	Secure Telephone Unit -3rd generation
DoDI	Department of Defense Instruction			SUT	System Under Test
DS0	Digital Signal Level 0 (64 kbps)	kbps	kilobits per second	T1	Digital Transmission Link Level 1 (1.544 Mbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)	Mbps	Megabits per second	T1.607	ISDN – Layer 3 Signaling Specifications for Circuit Switched Bearer Service for DSS1
DSS1	Digital Subscriber Signal 1	MLPP	Multi-Level Precedence and Preemption	T.4	Standardization of Group 3 facsimile terminals for document transmission
DSN	Defense Switched Network	MOS	Mean Opinion Score	UCR	Unified Capabilities Requirements
DTMF	Dual Tone Multi-Frequency	NI 1/2	National ISDN Standard 1 or 2	VBD	Variable bit data
E&M	Ear and Mouth	NX56	Data format restricted to multiples of 56 kbps	VoIP	Voice over Internet Protocol
E1	European Basic Multiplex Rate (2.048 Mbps)	NX64	Data format restricted to multiples of 64 kbps	VTC	Video Teleconferencing
		PBX	Private Branch Exchange		
		PBX 2	Private Branch Exchange 2		