



DEFENSE INFORMATION SYSTEMS AGENCY

JOINT INTEROPERABILITY TEST COMMAND
P.O. BOX 12798
FORT HUACHUCA, ARIZONA 85670-2798

IN REPLY
REFER TO:

Battlespace Communications Portfolio (JTE)

25 October 2006

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

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

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification. Additional references are provided in enclosure 1.
2. The Avaya S8300 Media Server with a G700 Media Gateway Software Release CM 3.0 (R013x.00.340.3: Super Patch 11815) is hereinafter referred to as the System Under Test (SUT). The SUT meets all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The manufacturer states that the SUT can handle up to ten G700 Media Gateways on a single S8300 processor; however, this capability was not tested and is not certified. The test was conducted with two G700 Media Gateways, which is the maximum number of gateways the S8300 is certified with for supporting assured voice services. The SUT meets the VoIP critical interoperability requirements with a certified Assured Services Voice Application Local Area Network listed on the Approved Product List found on the Telecom Switched Services Interoperability (TSSI) website at <http://jitic.fhu.disa.mil/tssi>. The identified test discrepancies shown in the Certification Testing Summary (enclosure 2), 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 finding is based on interoperability testing conducted by JITC and review of vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 19 December 2005 through 3 February 2006. Regression testing was conducted from 19 June through 21 July 2006. Review of vendor's LoC was completed on 25 September 2006. Enclosure 2 documents the test results and describes the tested network and system configurations. System interoperability should be verified before deployment in an operational environment that varies significantly from the test environment.

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

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

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

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	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 T1.619a)	Yes	Certified	Met all CRs and FRs.
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The E1 PRI interface is a not critical interface for a PBX1. It is supported by the SUT however it was not tested and is therefore not certified. There is no operational impact.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs with the following minor exception: The SUT offers only S/T ISDN BRI interface. ¹ The operational impact is minor.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP	No	Certified	Met all CRs and FRs with certified Assured Services Voice Application Local Area Network.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	No	Certified	All common features are conditional. The SUT met all CRs and FRs for the following common features: call waiting, three-way calling, call forwarding, and call pick-up with minor exceptions. ^{2,3} The SUT does not support the other common features. There is no operational impact because common features are not a critical requirement for a PBX 1.
Attendant	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
Public Safety	No	Certified	All public safety features are conditional. The SUT met all CRs and FRs for Basic Emergency Service 911. The SUT does not support the other public safety features. There is no operational impact because public safety is not a critical requirement for a PBX 1.
Preset Conferencing	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
Nailed-up Connections	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
PAT	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
DSN Hotline Services	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
Network Management	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs.	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 4.	See note 4.	
VoIP System	No	Certified	Met all CRs and FRs. ⁵	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	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 PRI (Q.931)	No	Not Tested	The E1 PRI interface is a not critical interface for a PBX1. It is supported by the SUT however it was not tested and is therefore not certified. There is no operational impact.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ⁶	Met all CRs and FRs.
LEGEND:				
<p>ANSI - American National Standards Institute BRI - Basic Rate Interface CAS - Channel Associated Signaling CRs - Capability Requirements DISA - Defense Information Systems Agency DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DSS1 - Digital Subscriber Signaling 1 DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 IT - Information Technology ISDN - Integrated Services Digital Network</p>				
<p>LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption NI 1/2 - National ISDN Standard 1 or 2 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network SS7 - Signaling System 7 S/T - 4-Wire ISDN BRI interface SUT - System Under Test TI - 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 TI TPC - Twisted Pair Copper U - 2-wire ISDN BRI interface VoIP - Voice over Internet Protocol</p>				
NOTES:				
<p>1 The SUT offers only an S/T ISDN BRI interface. A U-interface is not supported. ISDN BRI interface is a conditional interface for a PBX 1; therefore the operational impact is minor. 2 The SUT does not allow the classmarking of different precedence levels on each leg of a three-way conference. This is due to the fact that the SUT connects each party in a single timeslot. To mitigate the operational impact, the SUT classmarks each party at the highest precedence level of the conference. 3 When more than two members of a call pickup group are ringing at different precedence levels and a call pickup is attempted, the highest precedence level is not always picked up first. Since all unanswered precedence calls above ROUTINE placed to a call pickup group divert to an attendant console, night service, or alternate directory number within 15-45 seconds, the operational impact is minor. 4 Security is tested by DISA-led Information Assurance test teams and published in a separate report. 5 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008: a. Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DIRS). b. Maintaining interoperability in heterogeneous environments and with IPv4. c. Commitment to upgrade as the IPv6 standard evolves. d. Availability of contractor/vendor IPv6 technical support. 6 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.</p>				

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Table 2. PBX 1 Requirements

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

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

Table 2. PBX 1 Requirements (continued)

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

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	Critical	Requirements Required or Conditional	References
Network Management	No	<ul style="list-style-type: none"> • Interfaces (C) • Measurements and data generation (C) • Fault management (C) • Configuration management (C) • Accounting management (C) • Performance management (C) • NM controls (C) • Remote access (C) 	<ul style="list-style-type: none"> • GSCR Sect. 9.1 • GSCR Sect. 9.2 • GSCR Sect. 9.3 • GSCR Sect. 9.4 • GSCR Sect. 9.5 • GSCR Sect. 9.6 • GSCR Sect. 9.7 • GSCR Sect. 9.8
ISDN Services	No	<ul style="list-style-type: none"> • Electronic Key Telephone System (C) 	<ul style="list-style-type: none"> • GSCR Sect. 10, table 10-3
Synchronization	Yes	<ul style="list-style-type: none"> • Line timing mode (R) • Internal Stratum 4 (R) 	<ul style="list-style-type: none"> • GSCR Sect. 11.1.1.2 • GSCR Sect. 11.1.2.2
Reliability	Yes	<ul style="list-style-type: none"> • GR-512-CORE (R) 	<ul style="list-style-type: none"> • GSCR Sect. 12
Security	Yes	<ul style="list-style-type: none"> • GR-815, STIGs, and DIACAP (replacement for DITSCAP) (R) 	<ul style="list-style-type: none"> • GSCR Sect. 13
VoIP			
Feature/ Capability	Critical	Requirements Required or Conditional	References
VoIP System	No	<p>VoIP function is conditional. If VoIP is provided, all of the following requirements must be met:</p> <ul style="list-style-type: none"> • MOS 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Coder/Decoder (CODEC) • Traffic Engineering • Security • NM • Line timing • Internal Clock • Latency \leq 60 milliseconds • Packet Loss • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3 • GSCR App. 3, paragraph 1.7

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

Table 2. PBX 1 Requirements (continued)

Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN ¹	No	Trunking	<ul style="list-style-type: none"> Positive Identification Control (C) On-Netting (C) Off-Netting (C) 	<ul style="list-style-type: none"> CJCSI 6215.01B CJCSI 6215.01B CJCSI 6215.01B
DRSN ²	Yes	Access	<ul style="list-style-type: none"> Alerting Signals and Tones (R) Call Processing (R) Call Treatments (R) Analog busy/idle (R) 	<ul style="list-style-type: none"> GSCR Sect. 5.5 GSCR Sect. 4.4 GSCR Sect. 4.1 GSCR Sect. 4.3.4.1
		Voice	<ul style="list-style-type: none"> MOS (C) MLPP (C) Secure calls (C) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Sect. 3 CJCSI 6215.01B
LEGEND: 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 CJCSI - Chairman of the Joint Chiefs of Staff Instruction D/A - Digital to Analog Conversion DIACAP - DoD Information Assurance Certification and Accreditation Process DISR - DoD IT Standards Registry DITSCAP - DoD Information Technology Security Certification and Accreditation Process 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) EIA - Electronic Industries Alliance G.711 - Standard for PCM of Voice Frequencies GR-512 - LSSGR: Reliability, Section 12 GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union - Telecommunication Standardization Sector kbps - kilobits per second KXX - K=any number 2-8, X=any number 1-9 LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score NI 1/2 - National ISDN Standard 1 or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required Sect. - Section SS7 - Signaling System 7 STE - Secure Terminal Equipment STIGs - Security Technical Implementation Guides STU-III - Secure Telephone Unit-3 rd generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-465-A - Group 3 Facsimile Apparatus for Document Transmission VBD - Variable bit data VoIP - Voice over Internet Protocol VTC - Video Teleconferencing WWNDP - Worldwide Numbering and Dialing Plan				
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.				

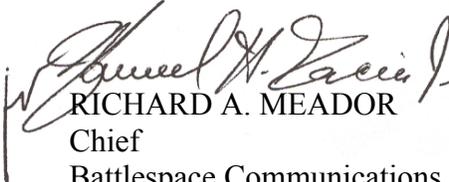
5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive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

JITC Memo, JTE, Special Interoperability Test Certification of Avaya S8300 Media Server with a G700 Media Gateway Software Release Communication Manager (CM) 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP)

6. The JITC point of contact is Mr. Joseph Schulte, DSN 879-5164, commercial (520) 538-5164, FAX DSN 879-4347, or e-mail to joseph.schulte@disa.mil. The tracking number for the SUT is 43051.

FOR THE COMMANDER:

2 Enclosures a/s



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Distribution:

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Joint Interoperability Test Command, Liaison, ATTN: TED/JT1, 2W24-8C, P.O. Box 4502, Falls Church, VA 22204-4502

Defense Information Systems Agency, Net-Centricity Requirements and Assessment Branch, ATTN: GE333, Room 244, P.O. Box 4502, Falls Church, VA 22204-4502

Office of Chief of Naval Operations (N71CC2), CNO N6/N7, 2000 Navy Pentagon, Washington, DC 20350

Headquarters U.S. Air Force, AF/XICF, 1800 Pentagon, Washington, DC 20330-1800

Department of the Army, Office of the Secretary of the Army, CIO/G6, ATTN: SAIS-IOQ, 107 Army Pentagon, Washington, DC 20310-0107

U.S. Marine Corps (C4ISR), MARCORSSYSCOM, 2200 Lester St., Quantico, VA 22134-5010

DOT&E, Net-Centric Systems and Naval Warfare, 1700 Defense Pentagon, Washington, DC 20301-1700

U.S. Coast Guard, CG-64, 2100 2nd St. SW, Washington, DC 20593

Defense Intelligence Agency, 2000 MacDill Blvd., Bldg 6000, Bolling AFB, Washington, DC 20340-3342

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

Director, Defense Information Systems Agency, ATTN: GS235, Room 5W24-8A, P.O. Box 4502, Falls Church, VA 22204-4502

Office of Assistant Secretary of Defense (NII)/DoD CIO, Crystal Mall 3, 7th Floor, Suite 7000, 1851 S. Bell St., Arlington, VA 22202

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

U.S. Joint Forces Command, J68, Net-Centric Integration, Communications, and Capabilities Division, 1562 Mitscher Ave., Norfolk, VA 23551-2488

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

ADDITIONAL REFERENCES

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (d) Defense Information Systems Agency (DISA), "Defense Switched Network (DSN) Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005
- (e) Executive Office of the President, "Transition Planning for Internet Protocol version 6 (IPv6)," 2 August 2005
- (f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 1, Revision 1," 1 June 2005

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Avaya S8300 Media Server with a G700 Media Gateway Software Release CM 3.0 (R013x.00.340.3: Super Patch 11815) including Voice over Internet Protocol (VoIP) is hereinafter referred to as the System Under Test (SUT).

2. PROPONENT. Defense Information Systems Agency (DISA).

3. PROGRAM MANAGER. Mr. Howard Osman, GS23, Room 5W23, 5275 Leesburg Pike, Falls Church, VA 22041, E-mail: Howard.Osman@disa.mil.

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

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Private Branch Exchange (PBX) 1. The SUT consists of one S8300 Media Server module and two G700 Media Gateways connected together via a serial backplane. The SUT delivers industry standard call control, quality of service, and management functions. The S8300 Media Server uses an Intel Pentium Class Server with a 30 gigabit hard-drive and 512 Megabytes of Random Access Memory and runs a LINUX operating system. The S8300 Media Server module is inserted into and takes up one media module bay in a G700 Media Gateway. The G700 Media Gateway is a 19-inch 2U chassis containing four media module bays and one Avaya P330 Expansion Module bay. The G700 Media Gateway's media module bays support card types: Pulse Code Modulation (PCM)-24 [Digital Transmission Link Level 1 (T1) Channel Associated Signaling (CAS), T1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI)], PCM-30 [European Basic Multiplex Rate (E1) CAS], analog, digital proprietary, and 4-wire ISDN lines. The SUT offers an E1 ISDN PRI trunk interface; however, the interface was not tested and is not covered under this certification. The G700 Media Gateway contains an integrated VoIP engine that can support up to 64 International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) G.711 Time Division Multiplex (TDM)/Internet Protocol (IP) simultaneous calls via an IP connection to an Assured Services Voice Application Local Area Network (ASVALAN). ITU-T G.729 and ITU-T G.723 are also supported by the SUT; however, they were not tested and are not covered under this certification. The G700 has two 10/100 Ethernet ports and a serial port for Command Line Interface access.

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 Generic Switching Center Requirements (GSCR) operational DSN Architecture is depicted in figure 2-1. The architecture depicts the relationship of Military Department PBXs to the other DSN switch types.

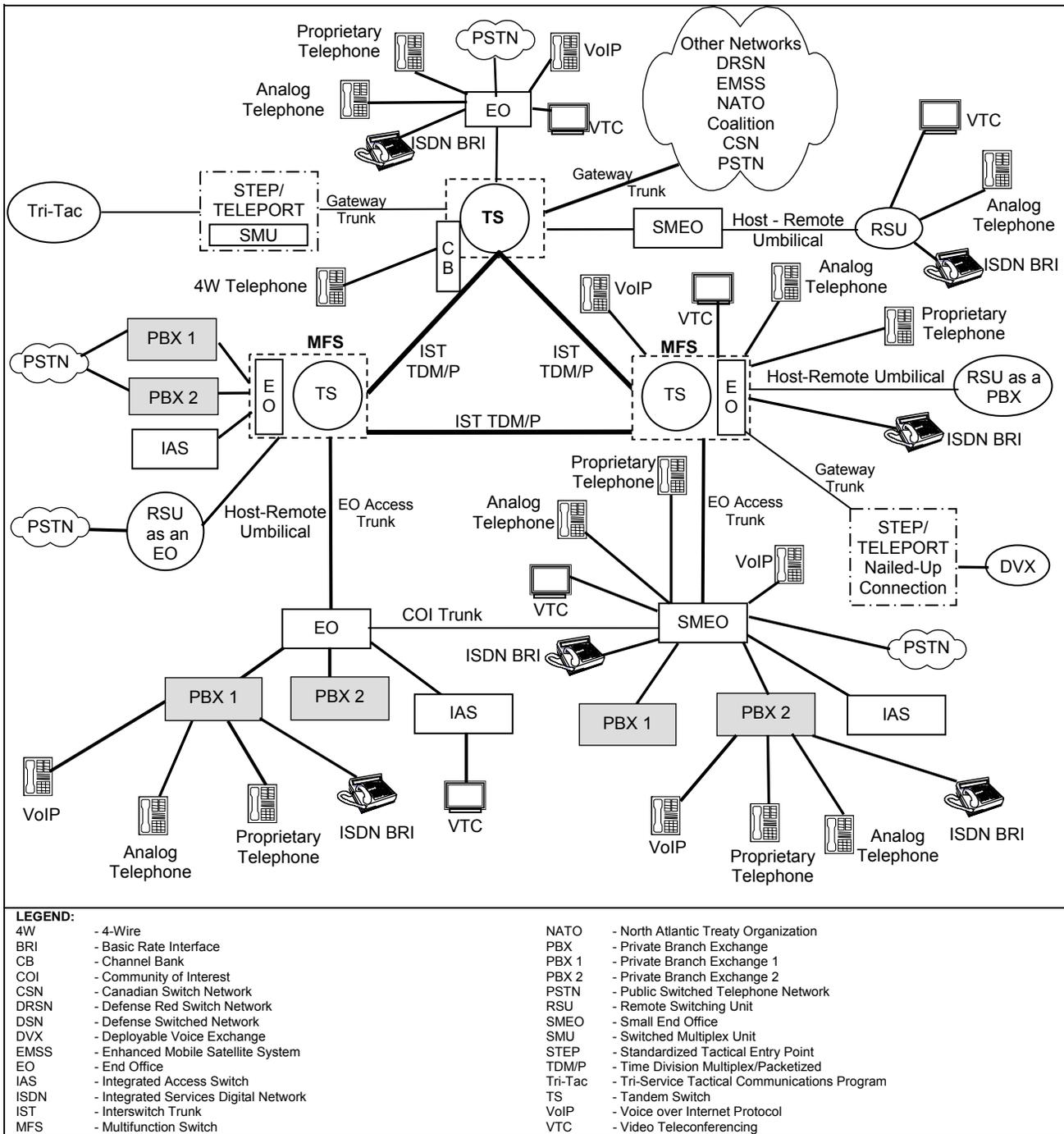


Figure 2-1. DSN Architecture

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

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

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

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

d. Internet Protocol version 6 (IPv6) requirements specified in reference (d), paragraph 1.7, table 1-3, by 30 June 2008 in accordance with reference (e) verified through vendor submission of LoC.

Table 2-1. PBX 1 Requirements

DSN Line Interfaces				
Interface	Critical	Requirements Required or Conditional		References
2-Wire Analog	Yes	Access	<ul style="list-style-type: none"> • Directory Number Identification (R) • Line signaling (R) • Loop Start Line (R: 2-Wire Analog only) • Alerting Signals and Tones (R) • WWNDP (R) • Call Treatments (R) • 2W user access (R: 2-Wire Analog only) • Analog busy/idle (R: 2-Wire Analog only) 	<ul style="list-style-type: none"> • GSCR Sect. 2.1.1 • GSCR Sect. 5.2 • GSCR Sect. 5.2.1 • GSCR Sect. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
2-Wire Digital (Proprietary)	No			
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Voice	<ul style="list-style-type: none"> • MOS (R) • Announcements (R) • MLPP (R) • Secure Calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Sect. 3.1.3 • GSCR Sect. 3.4.3/3.9 • CJCSI 6215.01B
		Facsimile	<ul style="list-style-type: none"> • Analog: TIA/EIA-465-A (R) 	<ul style="list-style-type: none"> • DISR
VoIP	No	Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56 kbps switched data (R: BRI only) • 64 kbps switched data (R: BRI only) • NX56 synchronous BER (R: BRI only) • NX64 synchronous BER (R: BRI only) • Secure data (STE/STU-III) (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • GSCR Sect. 3.10 • CJCSI 6215.01B
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: BRI only) 	<ul style="list-style-type: none"> • DISR

Table 2-1. PBX 1 Requirements (continued)

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

Table 2-1. PBX 1 Requirements (continued)

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

Table 2-1. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	Critical	Requirements Required or Conditional		References
VoIP System	No	VoIP function is conditional. If VoIP is provided, all of the following requirements must be met: <ul style="list-style-type: none"> • MOS 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM CODEC • Traffic Engineering • Security • NM • Line timing • Internal Clock • Latency ≤ 60 milliseconds • Packet Loss • IPv6 capable 		<ul style="list-style-type: none"> • GSCR App. 3 • GSCR App. 3, paragraph 1.7
Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN ¹	No	Trunking	<ul style="list-style-type: none"> • Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • CJCSI 6215.01B • CJCSI 6215.01B
DRSN ²	Yes	Access	<ul style="list-style-type: none"> • Alerting Signals and Tones (R) • Call Processing (R) • Call Treatments (R) • Analog busy/idle (R) 	<ul style="list-style-type: none"> • GSCR Sect. 5.5 • GSCR Sect. 4.4 • GSCR Sect. 4.1 • GSCR Sect. 4.3.4.1 • CJCSI 6215.01B • GSCR Sect. 3 • CJCSI 6215.01B
		Voice	<ul style="list-style-type: none"> • MOS (C) • MLPP (C) • Secure calls (C) 	<ul style="list-style-type: none"> • GSCR Sect. 3 • CJCSI 6215.01B
LEGEND: 2W - 2-Wire A/D - Analog to Digital Conversion ANSI - American National Standards Institute BER - Bit Error Ratio BRI - Basic Rate Interface C - Conditional CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCS - Chairman of the Joint Chiefs of Staff CJCSI - CJCS Instruction D/A - Digital to Analog Conversion DISA - Defense Information Systems Agency DISR - DoD IT Standards Registry DITSCAP - DoD IT Security Certification and Accreditation Process DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EIA - Electronic Industries Alliance EMSS - Enhanced Mobile Satellite System G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement GR-512 - LSSGR: Reliability, Section 12 GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IEEE - Institute of Electrical and Electronics Engineers, Inc. IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network IT - Information Technology ITU-T - International Telecommunication Union-Telecommunication Standardization Sector kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 LAN - Local Area Network Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score NATO - North Atlantic Treaty Organization NGCS - NATO Gateway Communication Switch NI 1/2 - National ISDN 1 or 2 NM - Network Management NX56 - Data format restricted to multiples of 56 kbps NX64 - Data format restricted to multiples of 64 kbps PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PCM - Pulse Code Modulation PCM-24 - Pulse Code Modulation - 24 Channels PCM-30 - Pulse Code Modulation - 30 Channels PRI - Primary Rate Interface PSTN - Public Switched Telephone Network Q.955.3 - ISDN Signaling Standard for E1 MLPP R - Required SMEO - Small End Office SS7 - Signaling System 7 STE - Secure Terminal Equipment STU-III - Secure Telephone Unit -3rd generation T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.619a - SS7 and ISDN MLPP Signaling Standard for T1 TIA - Telecommunications Industry Association TIA/EIA-465-A - Group 3 Facsimile Apparatus for Document Transmission VBD - Variable bit data VLAN - Virtual LAN 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.				

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

VoIP test configuration is depicted in figure 2-3. The SUT was tested as the end-point in relation to the other switches.

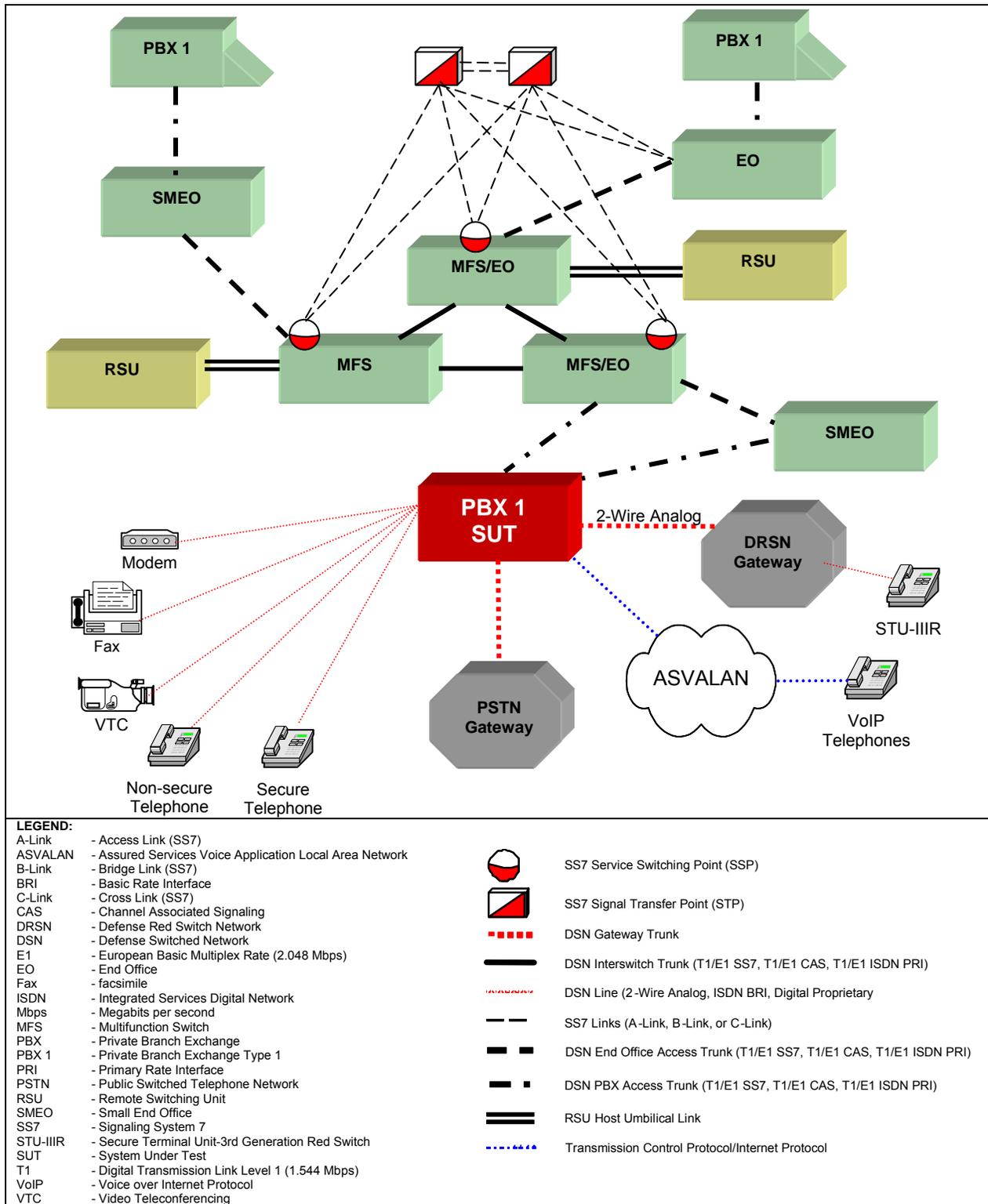


Figure 2-2. Notional Test Configuration

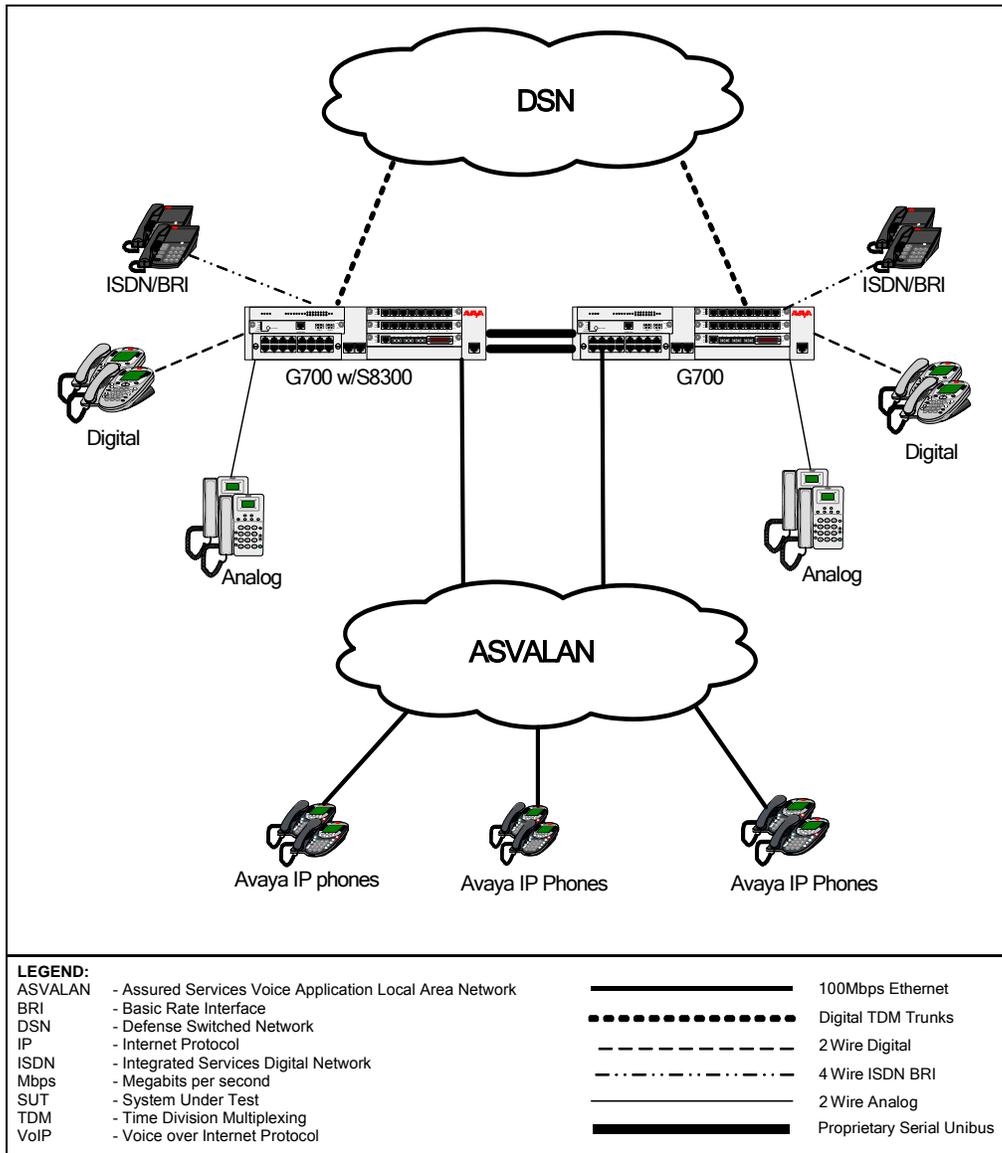


Figure 2-3. SUT VoIP Test Configuration

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware and software used in the test.

Table 2-2. Tested System Configurations

System Name		Software Release	
Nortel Networks MSL-100 (MFS, EO)		SE08	
Siemens EWSD (MFS, EO)		19d with Patch Set 46	
Lucent 5ESS (MFS, EO, SMEO, PBX 1, PBX 2)		5E16.2 SU 05-0005	
Avaya S8700/S8710 (SMEO, PBX 1, PBX 2)		CM 3.0 (R013x.00.0.340.3)	
Avaya S8300 Media Server with a G700 Media Gateway Software Release CM 3.0 (R013x.00.340.3 Super Patch 11815) (SUT)	Sub-component	Firmware	Function
	ICC	V3	Processing/Signaling (S8300 only)
	MM7711AP	VH6	TDM Interface 8-port Analog card
	MM710AP	VH4	TDM Interface Span card configurable as T1 CAS, T1 PRI, E1 CAS
	MM710AP	VH5	TDM Interface Span card configurable as T1 CAS, T1 PRI, E1 CAS
	MM720AP	VH5	TDM Interface 8-port Basic Rate Interface card
MM712AP	VH3	TDM Interface 8-port Digital card	
Telephones			
Interface Type	Model		Firmware
IP Phone	4620 IP		2.2
IP Phone	4620SW IP		2.2
IP Phone	4621SW IP		2.2
IP Phone	4625SW IP		2.5
ISDN BRI	Tone Commander 6210-B-02K		01.06.12
ISDN BRI	Tone Commander 6220-TSG-B-02D		01.06.12
Digital Set	6408D+		Not Applicable
2-Wire Analog	Panasonic KX-TS15-W		Not Applicable
LEGEND: 5ESS - Class 5 Electronic Switching System BRI - Basic Rate Interface CAS - Channel Associating Signal CM - Communications Manager E1 - European Basic Multiplex Rate (2.048 Mbps) EO - End Office EWSD - Elektronisches Wählsystem Digital ICC - Internal Call Controller IP - Internet Protocol ISDN - Integrated Services Digital Network Mbps - Megabits per second MFS - Multifunction Switch MM - Media Module MSL - Meridian Switching Load PBX 1 - Private Branch Exchange 1 PBX 2 - Private Branch Exchange 2 PRI - Primary Rate Interface SE - Succession Enterprise SMEO - Small End Office SU - Software Update SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) TDM - Time Division Multiplexing			
NOTE: The SUT E1 card is configurable for E1 ISDN PRI; however, it was not tested and is not covered under this certification.			

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs for T1 ISDN PRI National ISDN Standard 1 or 2 (NI 1/2) American National Standards Institute (ANSI) T1.619a. The SUT met all critical CRs and FRs for Defense Switched Network T1 and E1 CAS. The SUT offers an E1 PRI; however, it was not tested and is not covered under this certification. Detailed trunk configurations and associated lessons learned can be found on the Telecom Switched Services Interoperability (TSSI) web page: http://jitc.fhu.disa.mil/tssi/cert/cert_avaya.html.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Analog (GR-506-CORE), 2-Wire Digital (Proprietary), ISDN Basic Rate Interface (BRI) 4-Wire ISDN BRI interface (S/T) only, and VoIP DSN Line Interfaces. The SUT does not support an ISDN BRI U-Interface. The ISDN BRI line interface is a conditional requirement for a PBX 1; therefore the operational impact is minor.

(3) Features and Capabilities. The SUT met all critical interoperability certification requirements for features and capabilities as follows. The following features and capabilities are met by the SUT: synchronization, reliability, security, ISDN services (electronic key telephone systems), and VoIP. The following non-critical features and capabilities are not offered by the SUT and are therefore not covered by this certification: attendant console, preset conference, nailed up connections, precedence access threshold, DSN hotline, and network management. All public safety features are conditional. The SUT met all CRs and FRs for Basic Emergency Service 911. The SUT does not support the other public safety features. All common features are conditional. The SUT met all CRs and FRs for the following common features: call waiting, three-way calling, call forwarding, and call pick-up with the following minor exceptions:

- The SUT does not allow the classmarking of different precedence levels on each leg of a three-way conference. This is due to the fact that the SUT connects each party in a single timeslot. To mitigate the operational impact, the SUT classmarks each party at the highest precedence level of the conference.
- When more than two members of a call pickup group are ringing at different precedence levels and a call pickup is attempted, the highest precedence level is not always picked up first. Since all unanswered precedence calls above ROUTINE placed to a call pickup group divert to an attendant console, night service, or alternate directory number within 15-45 seconds, the operational impact is minor.

(4) VoIP. The SUT is certified with any certified ASVALAN listed on the Approved Products List located on the TSSI web page: <http://jitic.fhu.disa.mil/tssi>.

(a) VoIP System. The GSCR, appendix 3, section A3.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the GSCR, appendix 3, section A3.2.1, VoIP calls shall have a Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with Department of Defense Information Technology Standards Registry (DISR) voice quality standards. This applies from handset to handset and from handset to gateway trunk in the DSN. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.15. The average inter-switch MOS was 4.21.

This average was based on a total of 90 intra-switch and inter-switch calls with the lowest intra-switch MOS being 4.09 and the lowest inter-switch MOS of 4.05.

2. Class of Service (CoS) and Quality of Service (QoS). The GSCR, appendix 3, section A3.3.2, outlines several methodologies to implement CoS and QoS. The 802.1p/Q at the Data Link Layer 2 (L2) and Differentiated Services Code Point (DSCP) at the Network Layer 3 (L3) were two CoS mechanisms that the certified network products employed. The SUT provides CoS by assignment of an 802.1p/Q tag. The 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice Virtual Local Area Network traffic was assigned to a high priority queue, ensuring voice traffic took precedence over data traffic. The priority bit for L2 voice signaling was tagged with a value of 6 and the voice media was tagged with a value of 5. The L3 DSCP bits for voice signaling, was tagged with 48 and voice media was tagging 46, in the tested configuration. By using the Ixia test equipment, a data load of 1.4 times the total link aggregate, was injected on the certified ASVALAN to insure that all CoS and QoS settings were working properly. Packet captures indicated all L2 and L3 tagging was properly tagged by the SUT.

3. Coder/Decoder (CODEC). In accordance with the GSCR, appendix 3, section A3.2.2, the ITU-T G.711 PCM CODEC with a 20 millisecond (ms) packet fill was required and was met by the SUT VoIP solution.

4. Traffic Engineering.

a. Phones. The following Avaya IP phones met the critical interoperability requirements for certification: 4620, 4620SW, 4621SW, and 4625SW. Shared access (i.e., same switch port is shared by PC and IP phone), was not tested per vendor request and is not certified with this configuration. It was noted during testing that auto negotiation failed to negotiate the correct physical layer parameter (i.e. half duplex, full duplex, etc.) All Avaya IP phones must have the access ports hard set to 100 Megabits per second full duplex. This will alleviate any auto negotiate problems that can occur during an active call. The 4620, 4620SW, 4621SW, and 4625SW IP phones passed all tests as required by the GSCR, appendix 3 when connected to a certified ASVALAN.

b. Scalability. This certification covers the Avaya S8300 with the G700 Media Gateway. The manufacturer states that the SUT can handle up to ten G700 Media Gateways on a single S8300 processor; however, this capability was not tested and is not certified. The test was conducted with two G700 Media Gateways, which is the maximum number of gateways the S8300 is certified with for supporting assured voice services. Each G700 Media Gateway can support a maximum of 450 extensions. The G700 contains an integrated VoIP engine that can support a maximum of 64 simultaneous ITU-T G.711 IP to TDM calls. Each G700 has a maximum of four slots that can hold either a T1/E1 card, digital line card (eight lines per card) or analog line card (eight lines per card). If the S8300 processor is included with the G700 Media gateway, then only three slots are available. To guarantee assured services with the

VoIP to TDM limitation of 64 simultaneous calls, each G700 Media Gateway must be configured with no more than two T1 cards and the remaining slots populated with line cards or one E1 card and the remaining slots populated with line cards.

5. MLPP. The GSCR, section 3, details the requirements for MLPP. All critical MLPP features and functions were met by the SUT.

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

(c) Network Management (NM). The GSCR, appendix 3, defines the overall NM requirements that VoIP systems must meet. The NM requirements for the SUT LAN were satisfied with vendor LoC. The switching system NM requirements in accordance with the GSCR, section 9, are not required for a PBX 1 and are not offered by the SUT.

(d) Synchronization. Synchronization is required for overall voice platforms to include VoIP systems. For the SUT solution, synchronization in accordance with the GSCR, section 11, was met. The SUT derived synchronization with line timing mode via traditional (PCM-24 or PCM-30) TDM-based interfaces.

(e) Latency. The GSCR, appendix 3, section A3.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress trunk. The SUT met the requirement. The average latency for 90 independent, five-minute calls was measured to be 56.55 ms, with none of the five-minute calls having a latency exceeding 60ms.

(f) Packet Loss. The GSCR, appendix 3, section A3.3.1.3, states packet loss shall not exceed 0.05% averaged over any five-minute period. The SUT average packet loss was measured at 0.00% for 90 five-minute calls placed.

(g) Internet Protocol version 6 (IPv6). An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of Internet Protocol version 4 (IPv4). IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008:

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

(5) Network Gateways. The SUT met all critical interoperability certification requirements for Public Switched Telephone Network (PSTN) and the Defense Red Switch Network (DRSN) Gateways. The interfaces certified for PSTN are T1 CAS, E1 CAS, T1 ISDN PRI NI 1/2, and 2-wire ground start line. The only interface certified for the DRSN is Twisted Pair Copper 2-wire analog.

b. System Interoperability Results. The SUT including VoIP is certified for joint use in the DSN as a PBX 1 and PBX 2 in accordance with the requirements set forth in the GSCR. The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The interoperability test summary is shown in table 2-3.

Table 2-3. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	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 T1.619a)	Yes	Certified	Met all CRs and FRs.
E1 PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The E1 PRI interface is a not critical interface for a PBX1. It is supported by the SUT however it was not tested and is therefore not certified. There is no operational impact.
DSN Line Interfaces			
Interface & Signaling	Critical	Status	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.
ISDN BRI NI 1/2	No	Certified	Met all CRs and FRs with the following minor exception: The SUT offers only S/T ISDN BRI interface. ¹ The operational impact is minor.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP	No	Certified	Met all CRs and FRs with certified Assured Services Voice Application Local Area Network.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	No	Certified	All common features are conditional. The SUT met all CRs and FRs for the following common features: call waiting, three-way calling, call forwarding, and call pick-up with minor exceptions. ^{2,3} The SUT does not support other common features. There is no operational impact because it is not a critical requirement for a PBX 1.
Attendant	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
Public Safety	No	Certified	All public safety features are conditional. The SUT met all CRs and FRs for Basic Emergency Service (911). The SUT does not support other public safety features. There is no operational impact because it is not a critical requirement for a PBX 1.
Preset Conferencing	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.

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

DSN Features and Capabilities (continued)				
Features and Capabilities	Critical	Status	Remarks	
Nailed-up Connections	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
PAT	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
DSN Hotline Services	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
Network Management	No	Not Tested	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.	
ISDN Services (EKTS)	No	Certified	Met all CRs and FRs.	
Synchronization	Yes	Certified	Met all CRs and FRs.	
Reliability	Yes	Certified	Met all CRs and FRs.	
Security	Yes	See note 4.	See note 4.	
VoIP System	No	Certified	Met all CRs and FRs. ⁵	
Network Gateways				
Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	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 PRI (Q.931)	No	Not Tested	The E1 PRI interface is a not critical interface for a PBX1. It is supported by the SUT however it was not tested and is therefore not certified. There is no operational impact.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ⁶	Met all CRs and FRs.
LEGEND: ANSI - American National Standards Institute BRI - Basic Rate Interface CAS - Channel Associated Signaling CRs - Capability Requirements DISA - Defense Information Systems Agency DoD - Department of Defense DP - Dial Pulse DRSN - Defense Red Switch Network DSN - Defense Switched Network DSS1 - Digital Subscriber Signaling 1 DTMF - Dual Tone Multi-Frequency E1 - European Basic Multiplex Rate (2.048 Mbps) EKTS - Electronic Key Telephone System FRs - Feature Requirements GR - Generic Requirement GR-506-CORE - LSSGR: Signaling for Analog Interfaces GSCR - Generic Switching Center Requirements IPv4 - Internet Protocol version 4 IPv6 - Internet Protocol version 6 IT - Information Technology ISDN - Integrated Services Digital Network LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption NI 1/2 - National ISDN Standard 1 or 2 PAT - Precedence Access Threshold PBX 1 - Private Branch Exchange 1 PM - Program Manager PRI - Primary Rate Interface PSTN - Public Switched Telephone Network SS7 - Signaling System 7 S/T - 4-Wire ISDN BRI interface 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 U - 2-wire ISDN BRI interface VoIP - Voice over Internet Protocol				
NOTES: 1 The SUT offers only an S/T ISDN BRI interface. A U-interface is not supported. ISDN BRI interface is a conditional interface for a PBX 1; therefore the operation impact is minor. 2 The SUT does not allow the classmarking of different precedence levels on each leg of a three-way conference. This is due to the fact that the SUT connects each party in a single timeslot. To mitigate the operational impact, the SUT classmarks each party at the highest precedence level of the conference. 3 When more than two members of a call pickup group are ringing at different precedence levels and a call pickup is attempted, the highest precedence level is not always picked up first. Since all unanswered precedence calls above ROUTINE placed to a call pickup group divert to an attendant console, night service, or alternate directory number within 15-45 seconds, the operational impact is minor. 4 Security is tested by DISA-led Information Assurance test teams and published in a separate report. 5 An IPv6 capable system or product, as defined in the GSCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria by 30 June 2008: a. Conformant with IPv6 standards profile contained in the DoD IT Standards Registry (DISR). b. Maintaining interoperability in heterogeneous environments and with IPv4. c. Commitment to upgrade as the IPv6 standard evolves. d. Availability of contractor/vendor IPv6 technical support. 6 Interoperability certification of the SUT does not constitute DRSN PM approval for connectivity to the DRSN. It is the user's responsibility to request connectivity approval directly from the PM.				

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