



DEFENSE INFORMATION SYSTEMS AGENCY
JOINT INTEROPERABILITY TEST COMMAND
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FORT HUACHUCA, ARIZONA 85670-2798

IN REPLY
REFER TO: Networks and Transport Division (JTE)

31 January 2006

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Sphere Communications, Spherically Internet Protocol (IP) Digital Switching System with Software Release 4.2.0.12, Compile Date 17 August 2005

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

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification. Additional references are provided in enclosure 1.
2. The Sphere Communications, Spherically IP Digital Switching System with Software Release 4.2.0.12 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 Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.
3. This finding is based on interoperability testing conducted by JITC and review of the vendor's Letters of Compliance (LoC). Testing was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 1 August through 17 November 2005. Review of vendor's LoC was completed on 21 November 2005. Enclosure 2 documents the test results and describes the tested network and system configurations. System interoperability should be verified before deployment in an operational environment that varies significantly from the test environment.
4. The SUT was tested with multiple Assured Services Voice Application Local Area Networks (ASVALANs). Refer to the Telecom Switched Services Interoperability (TSSI) website <http://jitic.fhu.disa.mil/tssi> for the certified ASVALAN hardware and software components. The interoperability test summary of the SUT is indicated in table 1. The PBX 1 required and conditional 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 FRs/CRs specified in reference (d) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in reference (e).

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
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 (ANSI T1.619a)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	No	Certified	Met all CRs and FRs.
Attendant	No	Not Supported	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	Met all CRs and FRs.
Preset Conferencing	No	Not Certified	The SUT meets the requirements for Preset Conferencing with the following exception: The SUT does not support the minimum requirement of 10 simultaneous conferences with 20 conferees each. ¹ Since this is not a critical requirement, the operation impact is minor.
Nailed-up Connections	No	Not Supported	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
PAT	No	Not Supported	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 Supported	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 Supported	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	Not Supported	This feature is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 2.	
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6 capability. ³

Table 1. SUT Interoperability Test Summary (continued)

Network Gateways																																																																																
	Interface & Signaling	Critical	Status	Remarks																																																																												
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.																																																																												
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.																																																																												
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.																																																																												
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.																																																																												
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ^d	Met all CRs and FRs.																																																																												
<p>LEGEND:</p> <table border="0"> <tr> <td>ANSI</td> <td>- American National Standards Institute</td> <td>ITU-T</td> <td>- International Telecommunication Union – Telecommunication Standardization Sector</td> </tr> <tr> <td>BRI</td> <td>- Basic Rate Interface</td> <td>Mbps</td> <td>- Megabits per second</td> </tr> <tr> <td>CAS</td> <td>- Channel Associated Signaling</td> <td>MFR1</td> <td>- Multifrequency Recommendation 1</td> </tr> <tr> <td>CRs</td> <td>- Capability Requirements</td> <td>MLPP</td> <td>- Multi-Level Precedence and Preemption</td> </tr> <tr> <td>DISA</td> <td>- Defense Information Systems Agency</td> <td>NI 1/2</td> <td>- National ISDN 1 or 2</td> </tr> <tr> <td>DITSCAP</td> <td>- Department of Defense Information Technology Security Certification and Accreditation Process</td> <td>PAT</td> <td>- Precedence Access Threshold</td> </tr> <tr> <td>DoD</td> <td>- Department of Defense</td> <td>PBX 1</td> <td>- Private Branch Exchange 1</td> </tr> <tr> <td>DP</td> <td>- Dial Pulse</td> <td>PM</td> <td>- Program Manager</td> </tr> <tr> <td>DRSN</td> <td>- Defense Red Switch Network</td> <td>PRI</td> <td>- Primary Rate Interface</td> </tr> <tr> <td>DSN</td> <td>- Defense Switched Network</td> <td>PSTN</td> <td>- Public Switched Telephone Network</td> </tr> <tr> <td>DSS1</td> <td>- Digital Subscriber Signaling 1</td> <td>Q.931</td> <td>- Signaling Standard for ISDN</td> </tr> <tr> <td>DTMF</td> <td>- Dual Tone Multi-Frequency</td> <td>Q.955.3</td> <td>- ISDN Signaling standard for E1 MLPP</td> </tr> <tr> <td>E1</td> <td>- European Basic Multiplex Rate (2.048 Mbps)</td> <td>SS7</td> <td>- Signaling System 7</td> </tr> <tr> <td>EKTS</td> <td>- Electronic Key Telephone System</td> <td>SUT</td> <td>- System Under Test</td> </tr> <tr> <td>FRs</td> <td>- Feature Requirements</td> <td>T1</td> <td>- Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>GR</td> <td>- Generic Requirement</td> <td>T1.607</td> <td>- ISDN - Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1</td> </tr> <tr> <td>GSCR</td> <td>- Generic Switching Center Requirements</td> <td>T1.619a</td> <td>- SS7 and ISDN MLPP Signaling Standard For T1</td> </tr> <tr> <td>IPv6</td> <td>- Internet Protocol version 6</td> <td>TPC</td> <td>- Twisted Pair Copper</td> </tr> <tr> <td>ISDN</td> <td>- Integrated Services Digital Network</td> <td>VoIP</td> <td>- Voice over Internet Protocol</td> </tr> </table> <p>NOTES:</p> <ol style="list-style-type: none"> The SUT provides a voice conference bridge capability of up to a total of 60 simultaneous conferees for each Meeting Hub. The GSCR requires a minimum of 10 simultaneous conferences with 20 conferees each. The vendor stated during testing that the SUT can stack Meeting Hubs to attain this requirement, however, multiple Meeting Hubs were not tested; therefore preset conferencing is not covered under this certification. A waiver from the Joint Staff would be required in order for the SUT to be fielded using the tested preset conferencing configuration with a single Meeting Hub. DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades. 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. 					ANSI	- American National Standards Institute	ITU-T	- International Telecommunication Union – Telecommunication Standardization Sector	BRI	- Basic Rate Interface	Mbps	- Megabits per second	CAS	- Channel Associated Signaling	MFR1	- Multifrequency Recommendation 1	CRs	- Capability Requirements	MLPP	- Multi-Level Precedence and Preemption	DISA	- Defense Information Systems Agency	NI 1/2	- National ISDN 1 or 2	DITSCAP	- Department of Defense Information Technology Security Certification and Accreditation Process	PAT	- Precedence Access Threshold	DoD	- Department of Defense	PBX 1	- Private Branch Exchange 1	DP	- Dial Pulse	PM	- Program Manager	DRSN	- Defense Red Switch Network	PRI	- Primary Rate Interface	DSN	- Defense Switched Network	PSTN	- Public Switched Telephone Network	DSS1	- Digital Subscriber Signaling 1	Q.931	- Signaling Standard for ISDN	DTMF	- Dual Tone Multi-Frequency	Q.955.3	- ISDN Signaling standard for E1 MLPP	E1	- European Basic Multiplex Rate (2.048 Mbps)	SS7	- Signaling System 7	EKTS	- Electronic Key Telephone System	SUT	- System Under Test	FRs	- Feature Requirements	T1	- Digital Transmission Link Level 1 (1.544 Mbps)	GR	- Generic Requirement	T1.607	- ISDN - Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1	GSCR	- Generic Switching Center Requirements	T1.619a	- SS7 and ISDN MLPP Signaling Standard For T1	IPv6	- Internet Protocol version 6	TPC	- Twisted Pair Copper	ISDN	- Integrated Services Digital Network	VoIP	- Voice over Internet Protocol
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Table 2. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (DTMF, MFR1, 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 (DTMF, MFR1, 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	Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56-kbps switched data (R: ISDN PRI only) • 64-kbps switched data (R: ISDN PRI only) • NX56 synchronous BER (R: ISDN PRI only) • NX64 synchronous BER (R: ISDN 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 Sect. 3.10
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (C: ISDN PRI only) 	<ul style="list-style-type: none"> • DISR
DSN Line Interfaces				
2-Wire Analog (GR-506-CORE)	Yes	Access	<ul style="list-style-type: none"> • DN Identification (R) • Line signaling (R) • 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. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		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
ISDN BRI NI 1/2	No	Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56-kbps switched data (R: ISDN BRI only) • 64-kbps switched data (R: ISDN BRI only) • NX56 synchronous BER (R: ISDN BRI only) • NX64 synchronous BER (R: ISDN 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 Sect. 3.10
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: ISDN BRI only) 	<ul style="list-style-type: none"> • DISR

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Interface	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, 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"> • E911 (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 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) • Operation 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

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Interface	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"> • EKTS (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 ¹	Yes	<ul style="list-style-type: none"> • DITSCAP (R) 	<ul style="list-style-type: none"> • GSCR Sect. 13
VoIP			
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 • ITU-T G.711 PCM Codec • Security in accordance with DITSCAP • NM • Line timing • Internal Clock • Latency ≤ 60 ms • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3
LANs	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"> • LAN parameters • CoS/QoS • VLANs • IEEE Standards Conformance • .99999 availability • Modular devices • 2 second link restoral • LAN NM • Traffic Engineering 	<ul style="list-style-type: none"> • GSCR App. 3
Network Gateways			
Gateway	Critical	Requirements Required or Conditional	References
PSTN	No	<p>Trunking</p> <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	<p>Access</p> <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
		<p>Voice</p> <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

JITC Memo, JTE, Special Interoperability Test Certification of Sphere Communications, Spherically Internet Protocol (IP) Digital Switching System with Software Release 4.2.0.12, Compile Date 17 August 2005

Table 2. PBX 1 Requirements (continued)

LEGEND:	
2W	- 2-Wire
A/D	- Analog to Digital
ANSI	- American National Standards Institute
App	- Appendix
BER	- Bit Error Ratio
BRI	- Basic Rate Interface
CAS	- Channel Associated Signaling
CCS	- Common Channel Signaling
CJCSI	- Chairman of the Joint Chiefs of Staff Instruction
CoS	- Class of Service
C	- Conditional
D/A	- Digital to Analog
DISA	- Defense Information Systems Agency
DISR	- Department of Defense Information Technology Standards Registry
DITSCAP	- Department of Defense Information Technology Security Certification and Accreditation Process
DN	- Directory Number
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)
E911	- 911 Emergency Service
EIA	- Electronic Industries Alliance
EKTS	- Electronic Key Telephone System
G.711	- Standard for PCM of Voice Frequencies
GR	- Generic Requirement
GSCR	- Generic Switching Center Requirements
H.320	- Standard for Narrowband VTC
IEEE	- Institute of Electrical and Electronics Engineers, Inc.
IPv6	- Internet Protocol version 6
ISDN	- Integrated Services Digital Network
ITU-T	- International Telecommunication Union-Telecommunication Standardization Sector
kbps	- kilobits per second
KXX	- K= any number 2-8; X= any number 1-9
LAN	- Local Area Network
Mbps	- Megabits per second
MFR1	- Multi-Frequency Recommendation 1
MLPP	- Multi-Level Precedence and Preemption
MOS	- Mean Opinion Score
ms	- milliseconds
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
QoS	- Quality of Service
R	- Required
Sect.	- section
SS7	- Signaling System 7
STE	- Secure Terminal Equipment
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
VLAN	- Virtual Local Area Network
VoIP	- Voice over Internet Protocol
VTC	- Video Teleconferencing
WWNDP	- Worldwide Numbering and Dialing Plan

NOTES:
1 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.
2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.

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

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

FOR THE COMMANDER:

2 Enclosures a/s



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Spherically Internet Protocol (IP) Digital Switching System with Software Release 4.2.0.12,
Compile Date 17 August 2005

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ADDITIONAL REFERENCES

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01B, "Policy for Department of Defense Voice Services," 23 September 2001
- (d) Defense Information Systems Agency (DISA), "Defense Switched Network (DSN) Generic Switching Center Requirements (GSCR), Incorporated Change 1," 1 March 2005
- (e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP)," 23 April 2004

CERTIFICATION TESTING SUMMARY

1. **SYSTEM TITLE.** Special Interoperability Test Certification of Sphere Communications Spherical Internet Protocol (IP) Digital Switching System with Software Release 4.2.0.12, Compile Date 17 August 2005, 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 suited for small stand-alone or remote applications. The SUT is an IP-based Private Branch Exchange (PBX) 1, which supports American National Standards Institute (ANSI) T1.619a Digital Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI). The SUT is designed to be installed on standard Commercial Off the Shelf (COTS) servers running Windows 2000 or 2003 server operating systems. The SUT is engineered with two Media Gateway Controllers (MGC). One MGC acts as the primary and the other the secondary. The primary and secondary MGCs both participate in call control and load balancing of devices. The SUT consists of two VG3 Media Gateway Central Office (CO)Hubs, which each support a single T1 PRI interface. The VG3 Media Gateway Phone Hub provides 24 analog telephone ports. The Spherical Manager Application software provides call control and telephony features. The VG3 Media Gateways convert analog stations and T1 ISDN trunks to Transmission Control Protocol/Internet Protocol (TCP/IP). The Spherical Administrator provides a Graphical User Interface-based configuration management tool. The Spherical Manager is hosted on COTS servers and provides call-processing functionality within the Voice over Internet Protocol (VoIP) network. The SUT also comprises of the end instruments (phones) listed in Table 2-2.
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 PBX 1s to the other DSN switch types.

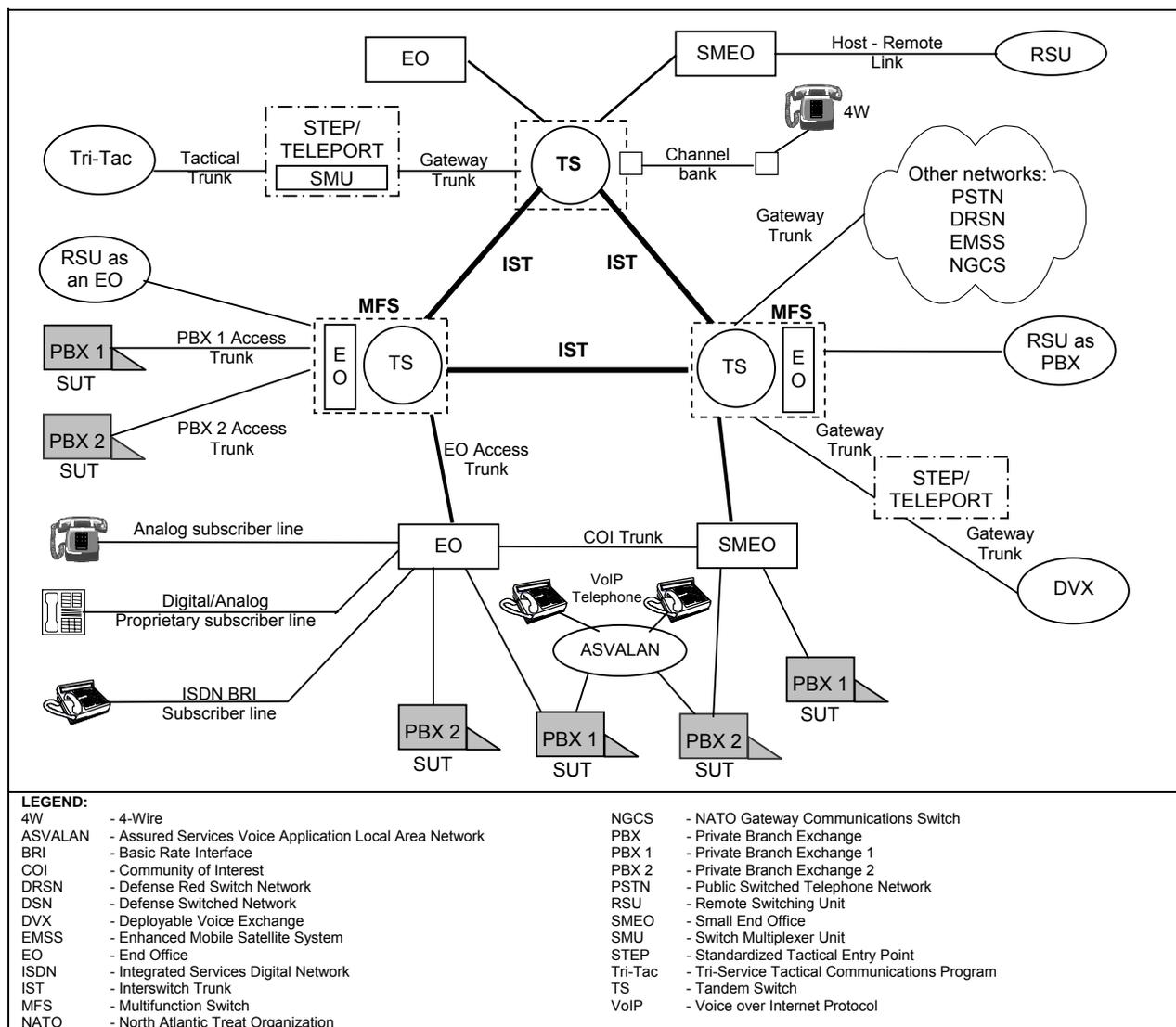


Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in table 2-1. If a switch satisfies PBX 1 criteria, it will satisfy the lesser standards of a PBX 2. These requirements are derived from:

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

Table 2-1. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (DTMF, MFR1, 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 (DTMF, MFR1, 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
		Data	<ul style="list-style-type: none"> • Modem (VBD) (R) • 56-kbps switched data (R: ISDN PRI only) • 64-kbps switched data (R: ISDN PRI only) • NX56 synchronous BER (R: ISDN PRI only) • NX64 synchronous BER (R: ISDN 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 Sect. 3.10
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	VTC	<ul style="list-style-type: none"> • ITU-T H.320 (C: ISDN PRI only) 	<ul style="list-style-type: none"> • DISR
DSN Line Interfaces				
2-Wire Analog (GR-506-CORE)	Yes	Access	<ul style="list-style-type: none"> • DN Identification (R) • Line signaling (R) • 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. 5.5 • GSCR Sect. 4.5 • GSCR Sect. 4.1 • GSCR Sect. 4.3.3 • GSCR Sect. 4.3.4.1
		Voice	<ul style="list-style-type: none"> • MOS (R) • MLPP (R) • Announcements (R) • Secure Calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01B • GSCR Sect. 3.4.3, 3.9 • GSCR Sect. 3.1.3 • CJCSI 6215.01B
ISDN BRI NI 1/2	No	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: ISDN BRI only) • 64-kbps switched data (R: ISDN BRI only) • NX56 synchronous BER (R: ISDN BRI only) • NX64 synchronous BER (R: ISDN 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 Sect. 3.10
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: ISDN BRI only) 	<ul style="list-style-type: none"> • DISR

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities			
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, 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"> • 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 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) • Operation 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

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Interface	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"> • EKTS (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 ¹	Yes	<ul style="list-style-type: none"> • DITSCAP (R) 	<ul style="list-style-type: none"> • GSCR Sect. 13
VoIP			
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 • ITU-T G.711 PCM Codec • Security in accordance with DITSCAP • NM • Line timing • Internal Clock • Latency ≤ 60 ms • IPv6 capable 	<ul style="list-style-type: none"> • GSCR App. 3
LANs	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"> • LAN parameters • CoS/QoS • VLANs • IEEE Standards Conformance • .99999 availability • Modular devices • 2 second link restoral • LAN NM • Traffic Engineering 	<ul style="list-style-type: none"> • GSCR App. 3

Table 2-1. PBX 1 Requirements (continued)

Network Gateways				
Gateway	Critical	Requirements Required or Conditional		References
PSTN	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) Announcements (R) Secure calls (C) 	<ul style="list-style-type: none"> CJCSI 6215.01B GSCR Sect. 3 GSCR Sect. 3.1.3 CJCSI 6215.01B
LEGEND: 2W - 2-Wire A/D - Analog to Digital ANSI - American National Standards Institute App - Appendix BER - Bit Error Ratio BRI - Basic Rate Interface CAS - Channel Associated Signaling CCS - Common Channel Signaling CJCSI - Chairman of the Joint Chiefs of Staff Instruction CoS - Class of Service C - Conditional D/A - Digital to Analog DISA - Defense Information Systems Agency DISR - Department of Defense Information Technology Standards Registry DITSCAP - Department of Defense Information Technology Security Certification and Accreditation Process DN - Directory Number 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) E911 - 911 Emergency Service EIA - Electronic Industries Alliance EKTS - Electronic Key Telephone System G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement GSCR - Generic Switching Center Requirements H.320 - Standard for Narrowband VTC IEEE - Institute of Electrical and Electronics Engineers, Inc. IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU-T - International Telecommunication Union-Telecommunication Standardization Sector kbps - kilobits per second KXX - K= any number 2-8; X= any number 1-9 LAN - Local Area Network Mbps - Megabits per second MFR1 - Multi-Frequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption MOS - Mean Opinion Score ms - milliseconds 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 QoS - Quality of Service R - Required Sect. - section SS7 - Signaling System 7 STE - Secure Terminal Equipment 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 VLAN - Virtual Local Area Network VoIP - Voice over Internet Protocol VTC - Video Teleconferencing WWNDP - Worldwide Numbering and Dialing Plan				
NOTES: 1 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 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. The system’s required functions and features were tested using the notional test configuration depicted in figure 2-2. The SUT was tested as the end-point in relation to the other switches. Figure 2-3 depicts the switch configuration.

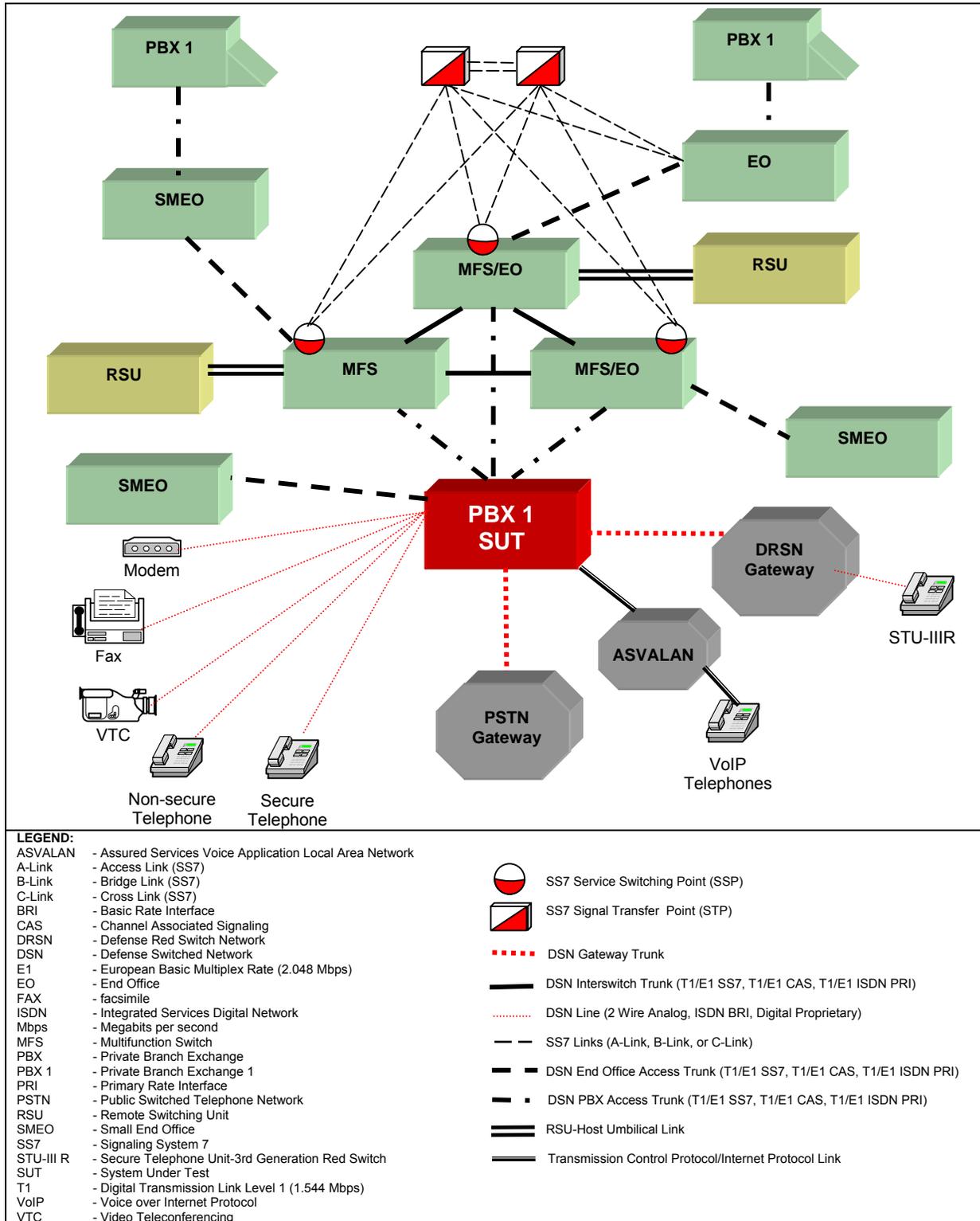


Figure 2-2. Notional Test Configuration

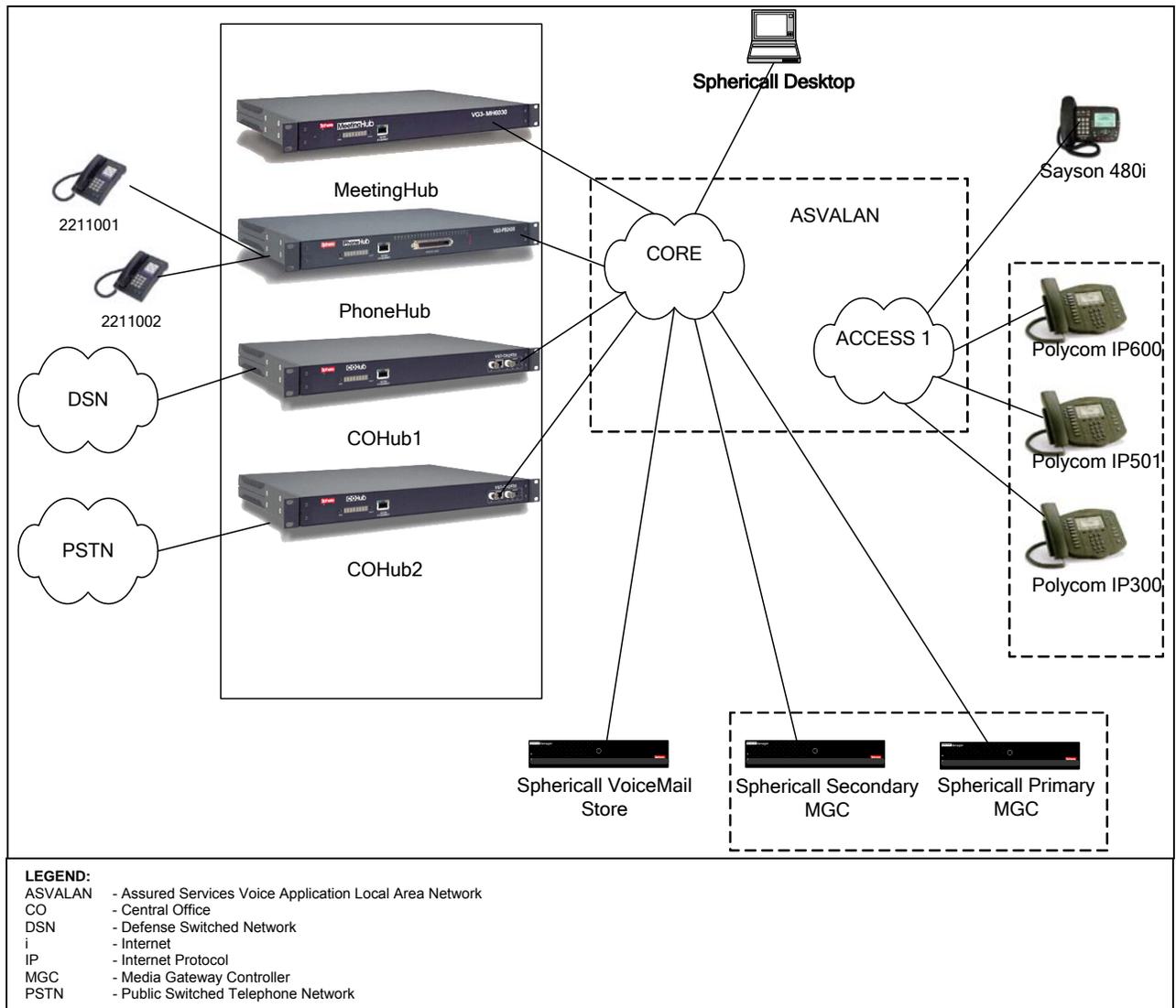


Figure 2-3. Switch Configuration

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

Table 2-2. Tested System Configurations

System Name		Software Release	
Nortel Networks MSL-100 (MFS, EO)		SE06	
Siemens EWSD (MFS, EO)		19d with Patch Set 44	
Avaya S8700 (SMEO, PBX 1, PBX 2)		Communication Manager (CM) 3.0	
Nortel Networks Option 11C (PBX1, PBX2)		Succession 3.0	
Redcom IGX (SMEO, PBX1, PBX2)		6.1A R1P3	
SUT	VG3 Media Gateway Meeting Hub MH360-10/100 (Not Certified)	4.2.0.12	
	VG3 Media Gateway Phone Hub PB2430-10/100	4.2.0.12	
	VG3 Media Gateway COHub CH2430-10/100	4.2.0.12	
	Spherical Primary MGC Server	W2K3 Server with SP1	4.2.0.12
	Spherical Secondary MGC Server, DC		4.2.0.12
	Voice Message Store Exchange Server		4.2.0.419
	MGC ^{see note}	4.2.0.506	
	DB Server ^{see note}	4.2.0.506	
	Call Logger ^{see note}	4.2.0.506	
	Desktop Manager ^{see note}	4.2.0.506	
	MediaServer ^{see note}	4.2.0.506	
	SUT Telephones		
Telephone Type		Firmware	
Polycom IP300, IP500, IP600		V 2.1.9.0058	
Panasonic KX-TS15-W		NA	
Sayson Aastra 480i		V.1.0.2.36	
LEGEND: CH - CO Hub CO - Central Office DB - Database DC - Domain Controller EO - End Office EWSD - Elektronisches Wählsystem Digital IGX - Integrated Services Digital Network (ISDN) Gateway Exchange IP - Internet Protocol MFS - Multifunction Switch MGC - Media Gateway Controller MH - Meeting Hub MSL - Meridian Switching Load NA - Not Applicable PBX 1 - Private Branch Exchange 1 PBX 2 - Private Branch Exchange 2 PB - Phone Hub SE - Succession Enterprise SMEO - Small End Office SP1 - Service Pack 1 SUT - System Under Test VG - Voice Gateway W2K3 - Windows 2003			
NOTE: These items are components that reside on the Spherical Media Gateways.			

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 NI 1/2 (ANSI T1. 619a. Detailed trunk configurations and associated lessons learned could be found on the JITC Telecom Switched Services Interoperability (TSSI) program web page: <http://jitc.fhu.disa.mil/tssi>.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for a 2-Wire Analog (GR-506-CORE) DSN line interface.

(3) Features and Functions. The SUT met all critical interoperability certification requirements for Features and Functions.

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

Network Gateways. All Sphere Gateways have a configurable jitter buffer. The minimum buffer size is set and the maximum buffer size will be twice the minimum. The minimum buffer can be set to as low as one packet, and the maximum buffer size is twice the minimum.

(5) VoIP. The SUT VoIP solution is a centric switch and is certified with any certified Assured Services Voice Application Local Area Network (ASVALAN). The results for the overall VoIP system, as defined by the GSCR, appendix 3, are presented below. All certified ASVALANs could be found on the TSSI web page: <http://jitc.fhu.disa.mil/tssi>.

(a) VoIP System. The GSCR, appendix 3, section A3.2, outlines the end-to-end VoIP requirements (i.e., encompasses both the circuit switch and ASVALAN). The following paragraphs detail the SUT VoIP solution test results.

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 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 met the requirement with all calls having a MOS of at least 4.0 and an average MOS of 4.38. The inter-switch calls had a MOS of at least 4.0 and the average MOS was 4.33. This average was based on a total of 100 intra-switch and inter-switch calls.

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 (L2) and Differentiated Services Code Point (DSCP) at the Network Layer (L3) were two CoS mechanisms that the certified network products employed. The SUT provides CoS by assigning of an 802.1p/Q tag. Switches within the topology were configured with multiple Virtual Local Area Networks (VLANs) to separate data from voice traffic. The 802.1Q tags were used to uniquely identify and separate traffic as it passed through network connections. Voice VLAN traffic was assigned to a high priority queue, ensuring voice traffic took precedence over data traffic. Priority bits for L2 voice signaling were set for 6 and voice media were set for 5. The L3 DSCP value for voice signaling was set for 48 and voice media for 46, in the tested configuration. A data load of 1.2 times the total link aggregate was put on the certified ASVALAN to insure that all CoS and QoS settings were working properly. Captures indicated all tags were set properly.

3. Codec. In accordance with the GSCR, appendix 3, section A3.2.2, the International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) G.711 Pulse-Code Modulation (PCM) codec with a 20 millisecond (ms) packet fill was required and was met by the SUT VoIP solution.

4. Traffic Engineering

a. IP Phones. The IP phones that met the critical interoperability requirements for certification were the Polycom IP300, IP500, and IP600 phones, along with the Aastra 480i phone. The Polycom phones are capable of only "auto negotiation" and cannot be set of 100 Mbps Full duplex. If an access device is configured for 100 Mbps full duplex, the Polycom phones will negotiate to 100 Mbps half duplex. This will cause excessive frame loss resulting in poor quality calls. Ports on access devices connecting to a Polycom phone need to be configured to "auto negotiation" in order to meet the certified configuration. Although the phones are capable of shared access (i.e., same switch port is shared by PC and IP phone), dedicated access was tested (separate ports for phones and PCs). The IP phones' switch ports were not tested and are not covered by this certification. Jitter buffers on the Polycom IP phones were set to 0/1, which is the minimum and maximum number of packets that can be buffered to compensate for jitter and still meet the latency requirements.

b. Scalability. Table 2-3 shows the minimum Media Gateway Server requirements and the maximum number of IP subscribers, which can be supported by each server. The SUT is certified with any certified ASVALAN, which can be found on the TSSI web page: <http://jitc.fhu.disa.mil/tssi>. The ASVALAN can be scaled to meet the maximum subscribers as long as it is composed of the equipment and software listed in this certification, and meets the traffic engineering constraints listed in table 2-3.

Table 2-3. Minimum Server Requirements

Minimum Hardware Server Configuration:	Maximum number of users per server
Intel Pentium 4 Processor, 2.8 GHz, 1GB RAM, 240 GB hard drive,	1500 Primary Ports
Intel Pentium 4 or Xeon Processor, 2.4 GHz, 1 GB RAM, 240 GB Hard Drive	1000 Primary Ports
Intel Pentium 4 Processor, 2.0 GHz, 1GB RAM, 160 GB hard drive,	500 Primary Ports
Intel Pentium III 1.0 GHz or Celeron 2.4 GHz Processor, 512 GB RAM, 80 GB hard drive,	250 Primary Ports
Intel Pentium III 850 MHz, 512 MB RAM, 20 GB Hard Drive	100 Primary Ports
LEGEND: GB - Gigabyte GHz - Gigahertz MB - Megabyte MHz - Megahertz RAM - Random Access Memory	

Information Assurance (IA) requires the security logs not be overwritten; therefore when the security logs are full the system will reboot and will not allow any action to take place, which would require a new event in the log. This prevents the secondary database from accessing the primary database and updating configuration changes. To overcome this anomaly, the event log must be configured to automatically archive and clear when it becomes full.

5. Multi-Level Precedence and Preemption (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 IA Test Plan. Results of the security testing are reported in a separate test report generated by the DISA IA test personnel.

(c) Network Management (NM). In accordance with the GSCR, section 9, PBX 1 switches are not required to support NM. The SUT does not offer this feature, and it was therefore not tested. The NM requirements for the ASVALAN are covered under a separate certification letter.

(d) Internal Clock. The switching system Internal Clock requirements in accordance with the GSCR, appendix 3, are not required for a PBX 1 and were not tested.

(e) Synchronization. The SUT met the GSCR, section 11 requirements for synchronization. Synchronization between gateways was achieved by designating one gateway as the network synchronization master and all other gateways as slaves. A timing signal was passed via the IP network from the master to all of the slaves, and each slave synchronizes its internal clock to the received timing signal. The master gateway must be a CoHub connected to a trunk that can provide a reliable and accurate timing source. The CoHub's internal clock is locked to the line timing from the trunk, and generates a synchronization packet every 1000ms. All synchronization packets are sent via Real Time Protocol (RTP) to a designated multicast address. Each of the media gateways that are designated as slaves listen for the RTP traffic on the designated multicast address. As the network synchronization packets are received, the internal clocks of each slave gateway are adjusted so that they are synchronous to the arrival of the network synchronization packets. To allow for jitter in the network, the slave gateway synchronizes to the average inter-packet interval of the network synchronization RTP packets.

(f) Latency. The GSCR, appendix 3, section A3.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress trunk. The SUT average latency over 100 calls was 51.4 ms.

(g) Packet Loss. The GSCR, appendix 3, section A3.3.1.3, states packet loss shall not exceed 0.05 percent averaged over any five-minute period. The SUT had 0.00 percent packet loss.

(h) Internet Protocol version 6 (IPv6). Although IPv6 capability is a required feature, currently the SUT's inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the Department of Defense (DoD) current overall IPv6 capabilities. To meet DoD's goal of transitioning to IPv6, the operational risk of the SUT's IPv6 capability will be reassessed during future hardware and software upgrades.

b. System Interoperability Results. The SUT 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 2-4 and the detailed interoperability test status is shown table 2-5.

Table 2-4. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	Critical	Status	Remarks
T1 CAS (DTMF, MFR1, DP)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all CRs and FRs.
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
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 (ANSI T1.619a)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
DSN Features and Capabilities			
Features and Capabilities	Critical	Status	Remarks
Common Features	No	Certified	Met all CRs and FRs.
Attendant	No	Not Supported	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	Met all CRs and FRs.
Preset Conferencing	No	Not Certified	The SUT meets the requirements for Preset Conferencing with the following exception: The SUT does not support the minimum requirement of 10 simultaneous conferences with 20 conferees each. ¹ Since this is not a critical requirement, the operational impact is minor.
Nailed-up Connections	No	Not Supported	This interface is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
PAT	No	Not Supported	This interface is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
DSN Hotline Services	No	Not Supported	This interface is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
Network Management	No	Not Supported	This feature not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
ISDN Services (EKTS)	No	Not Supported	This feature is not supported. The risk of not testing is minor because it is not a critical requirement for a PBX 1.
Synchronization	Yes	Certified	Met all CRs and FRs.
Reliability	Yes	Certified	Met all CRs and FRs.
Security	Yes	See note 2.	
VoIP System	No	Certified	Met all CRs and FRs with the exception of IPv6 capability. ³

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

Network Gateways				
	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Not Supported	This interface is not supported. There is no operational impact because it is not a critical requirement for a PBX 1.
DRSN	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified ⁴	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 DITSCAP - Department of Defense Information Technology Security Certification and Accreditation Process 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 GSCR - Generic Switching Center Requirements IPv6 - Internet Protocol version 6 ISDN - Integrated Services Digital Network ITU-T - International Telecommunication Union – Telecommunication Standardization Sector Mbps - Megabits per second MFR1 - Multifrequency Recommendation 1 MLPP - Multi-Level Precedence and Preemption NI 1/2 - National ISDN 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 Q.931 - Signaling Standard for ISDN Q.955.3 - ISDN Signaling standard for E1 MLPP SS7 - Signaling System 7 SUT - System Under Test T1 - Digital Transmission Link Level 1 (1.544 Mbps) T1.607 - ISDN - Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1 T1.619a - SS7 and ISDN MLPP Signaling Standard For T1 TPC - Twisted Pair Copper VoIP - Voice over Internet Protocol				
NOTES: 1 The SUT provides a voice conference bridge capability of up to a total of 60 simultaneous conferees for each Meeting Hub. The GSCR requires a minimum of 10 simultaneous conferences with 20 conferees each. The vendor stated during testing that the SUT can stack Meeting Hubs to attain this requirement, however, multiple Meeting Hubs were not tested; therefore preset conferencing is not covered under this certification. A waiver from the Joint Staff would be required in order for the SUT to be fielded using the tested preset conferencing configuration with a single Meeting Hub. 2 DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report. 3 Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades. 4 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>.

Table 2-5. SUT Interoperability Requirements/Status

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

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
E1 CAS	No (Europe only)	Not Supported ¹	Trunking	Framing (R)	GSCR Sect. 7	Not Supported	
				Line Code (R)	GSCR Sect. 7	Not Supported	
				Signaling (R)	GSCR Sect. 5	Not Supported	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Not Supported	
				WWNDP (R)	GSCR Sect. 4.5.1	Not Supported	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Not Supported	
				Routing (C)	GSCR Sect. 4.2	Not Supported	
				Trunk Groups(C)	GSCR Sect. 2.5.5 & 2.5.6	Not Supported	
				Call Processing (C)	GSCR Sect. 4	Not Supported	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Not Supported	
			PCM-24/PCM-30 Interoperation(C)	GSCR Sect. 7.3	Not Supported		
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Not Supported		
			Voice	MOS (R)	CJCSI 6215.01B	Not Supported	
				MLPP (R)	GSCR Sect. 3	Not Supported	
				Secure calls (R)	CJCSI 6215.01B	Not Supported	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Not Supported	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Supported	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported	
				NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported	
NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported					
VTC	Secure data (STE/STU-III) (R)	CJCSI Sect. 3.10	Not Supported				
	ITU-T H.320 (C: ISDN PRI only)	DISR	Not Supported				

Table 2-5. SUT Interoperability Requirements/Status (continued)

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

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Supported ¹	Trunking	Framing (R)	GSCR Sect. 7	Not Supported	
				Line Code (R)	GSCR Sect. 7	Not Supported	
				Signaling (R)	GSCR Sect. 5	Not Supported	
				Alarms (R)	GSCR Sect. 2.5.7, 7.1.4 & 7.2.2	Not Supported	
				Timing (R)	GSCR Sect. 11.1.1.2	Not Supported	
				WWNDP (R)	GSCR Sect. 4.5.1	Not Supported	
				Outpulsing digit formats (C)	GSCR Sect. 4.5.2	Not Supported	
				Routing (C)	GSCR Sect. 4.2	Not Supported	
				Trunk Groups(C)	GSCR Sect. 2.5.5 & 2.5.6	Not Supported	
				Call Processing (C)	GSCR Sect. 4	Not Supported	
				CAS to CCS trunk interworking (C)	GSCR Sect. 3.10	Not Supported	
				PCM-24/PCM-30 Interoperation(C)	GSCR Sect. 7.3	Not Supported	
			Direct Inward Dialing (C)	GSCR Sect. 2.3.2	Not Supported		
			Voice	MOS (R)	CJCSI 6215.01B	Not Supported	
				MLPP (R)	GSCR Sect. 3	Not Supported	
				Secure calls (R)	CJCSI 6215.01B	Not Supported	
			Facsimile	Analog: TIA/EIA-465-A (R)	DISR	Not Supported	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Supported	
				56-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported	
				64-kbps switched data (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported	
NX56 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported					
NX64 synchronous BER (R: ISDN PRI only)	GSCR Sect. 3.10	Not Supported					
	Secure data (STE/STU-III) (R)	CJCSI Sect. 3.10	Not Supported				
VTC	ITU-T H.320 (R: ISDN PRI only)	DISR	Not Supported				

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks
2-Wire Analog (GR-506-CORE)	Yes	Certified	Access	DN Identification (R)	GSCR Sect 2.1.1	Met	
				Line signaling (R)	GSCR Sect 5.2	Met	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Met	
				WWNDP (R)	GSCR Sect. 4.5	Met	
				Call Treatments (R)	GSCR Sect. 4.1	Met	
				2W user access (R)	GSCR Sect 4.3.3	Met	
			Voice	Analog busy/idle (R)	GSCR Sect 4.3.4.1	Met	
				MOS (R)	CJCSI 6215.01B	Met	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Met	
				Announcements (R)	GSCR3.1.3	Met	
			Facsimile	Secure calls (R)	CJCSI 6215.01B	Met	
				Analog: TIA/EIA-465-A (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Met	
Secure data (STE/STU-III) (R)	CJCSI Sect. 3.10	Met					
ISDN BRI NI 1/2	No	Not Supported ¹	Access	DN Identification (R)	GSCR Sect 2.1.1	Not Supported	
				Line signaling (R)	GSCR Sect 5.2	Not Supported	
				Alerting Signals and Tones (R)	GSCR Sect 5.5	Not Supported	
				WWNDP (R)	GSCR Sect. 4.5	Not Supported	
				Call Treatments (R)	GSCR Sect. 4.1	Not Supported	
			Voice	MOS (R)	CJCSI 6215.01B	Not Supported	
				MLPP (R)	GSCR Sect. 3.4.3, 3.9	Not Supported	
				Announcements (R)	GSCR3.1.3	Not Supported	
				Secure calls (R)	CJCSI 6215.01B	Not Supported	
			Data	Modem (VBD) (R)	CJCSI 6215.01B	Not Supported	
				56-kbps switched data (R))	GSCR Sect. 3.10	Not Supported	
				64-kbps switched data (R)	GSCR Sect. 3.10	Not Supported	
				NX56 synchronous BER (R)	GSCR Sect. 3.10	Not Supported	
				NX64 synchronous BER (R)	GSCR Sect. 3.10	Not Supported	
				Secure data (STE/STU-III) (R)	CJCSI Sect. 3.10	Not Supported	
			VTC	ITU-T H.320 (R: ISDN BRI only)	DISR	Not Supported	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features & Capabilities						
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional	Reference	Test Results	Remarks
Common Features	No	Certified	Selective call rejection (C)	GSCR Sect. 2.1.2	Met	
			Denied originating service (C)	GSCR Sect. 2.1.3	Met	
			Code restriction and diversion (C)	GSCR Sect. 2.1.4	Met	
			Three-way calling (C)	GSCR Sect. 2.1.5	Met	
			Add-on transfer and conference calling, and call hold (C)	GSCR Sect. 2.1.6	Met	
			Call forwarding (C)	GSCR Sect. 2.1.7	Met	
			Call pick-up (C)	GSCR Sect. 2.1.8	Met	
Call waiting (C)	GSCR Sect. 2.1.9	Met				
Attendant	No	Not Supported	Initiate all precedence levels (C)	GSCR Sect. 2.2.1	Not Supported	Minor ¹
			Visual display (C)	GSCR Sect. 2.2.2	Not Supported	Minor ¹
			Override class of service (C)	GSCR Sect. 2.2.3	Not Supported	Minor ¹
			Override busy line (C)	GSCR Sect. 2.2.4	Not Supported	Minor ¹
			Call deflection (C)	GSCR Sect. 2.2.5	Not Supported	Minor ¹
			Auto recall (C)	GSCR Sect. 2.2.6	Not Supported	Minor ¹
Waiting queue (C)	GSCR Sect. 2.2.7	Not Supported	Minor ¹			
Public Safety	No	Certified	E911 (C)	GSCR Sect. 2.4.1	Met	
			Trace of terminating calls (C)	GSCR Sect. 2.4.2	Met	
			Outgoing call trace (C)	GSCR Sect. 2.4.3	Met	
			Tandem call trace (C)	GSCR Sect. 2.4.4	Not Supported	Minor ¹
			Trace of a call in progress (C)	GSCR Sect. 2.4.5	Not Supported	Minor ¹
Preset Conferencing	No	Not Certified	Support 10 bridges; 1 originator and 20 conferees per bridge (C)	GSCR Sect. 2.1.6	Not Met	Minor ²
			Assign up to 20 address numbers per bridge (C)	GSCR Sect. 2.6	Met	
			Use KXX codes for bridge access (C)	GSCR Sect. 2.6	Met	
			Conference notification recorded announcement (C)	GSCR Sect. 2.6.1	Met	
			Auto retrieval and alternate address (C)	GSCR Sect. 2.6.2	Met	
			Bridge release (C)	GSCR Sect. 2.6.3	Met	
			Lost connection (C)	GSCR Sect. 2.6.4	Met	
			Secondary conferencing (C)	GSCR Sect. 2.6.5	Met	
Address translation (C)	GSCR Sect. 2.7	Met				
Nailed-Up Connections	No	Not Supported	Between any two like terminations (C)	GSCR Sect. 2.8	Not Supported	Minor ¹
			PCM-24 and PCM-30, both CAS and CCS (C)	GSCR Sect. 2.8	Not Supported	Minor ¹
			Supervision passed end-to-end for A/D or D/A (C)	GSCR Sect. 2.8	Not Supported	Minor ¹
			Monitored and auto reconfigure (C)	GSCR Sect. 2.8	Not Supported	Minor ¹
			Support at least 10% of circuits as nailed-up (C)	GSCR Sect. 2.8	Not Supported	Minor ¹
Non-preemptable (C)	GSCR Sect. 2.8	Not Supported	Minor ¹			

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features & Capabilities (continued)						
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional	Reference	Test Results	Remarks
PAT	No	Not Supported	Classmark for/not for PAT screening (C)	GSCR Sect. 2.11.1	Not Supported	Minor ¹
			7 PAT mechanisms (C)	GSCR Sect. 2.11.1	Not Supported	Minor ¹
			Outgoing call screening (C)	GSCR Sect. 2.11.1.1	Not Supported	Minor ¹
			Functional structure (C)	GSCR Sect. 2.11.1.2	Not Supported	Minor ¹
			Overflow Process (C)	GSCR Sect. 2.11.1.3	Not Supported	Minor ¹
			Simultaneous calls limitation (C)	GSCR Sect. 2.11.1.4	Not Supported	Minor ¹
			Decrementing call-in-progress count (C)	GSCR Sect. 2.11.1.5	Not Supported	Minor ¹
			Call treatment (C)	GSCR Sect. 2.11.1.6	Not Supported	Minor ¹
			Queuing (C)	GSCR Sect. 2.11.1.7	Not Supported	Minor ¹
			Attendant calls (C)	GSCR Sect. 2.11.1.8	Not Supported	Minor ¹
DSN Hotline Services	No	Not Supported	Operations measurement registers (C)	GSCR Sect. 2.11.1.9	Not Supported	Minor ¹
			Maintenance and Administration of thresholds (C)	GSCR Sect. 2.11.1.10	Not Supported	Minor ¹
			Hotline restrictions (C)	GSCR Sect. 2.12	Not Supported	Minor ¹
			Auto initiate (C)	GSCR Sect. 2.12	Not Supported	Minor ¹
			Analog and digital (C)	GSCR Sect. 2.12	Not Supported	Minor ¹
			Subscription basis (C)	GSCR Sect. 2.12	Not Supported	Minor ¹
Network Management	No	Not Supported	Protected hotline calling (C)	GSCR Sect. 2.12.1-4	Not Supported	Minor ¹
			WWNDP interoperable (C)	GSCR Sect. 2.12.5	Not Supported	Minor ¹
			Interfaces (C)	GSCR Sect. 9.1	Not Supported	Minor ¹
			Measurements and data generation (C)	GSCR Sect. 9.2	Not Supported	Minor ¹
			Fault management (C)	GSCR Sect. 9.3	Not Supported	Minor ¹
			Configuration management (C)	GSCR Sect. 9.4	Not Supported	Minor ¹
			Accounting management (C)	GSCR Sect. 9.5	Not Supported	Minor ¹
ISDN Services	No	Not Supported	Performance management (C)	GSCR Sect. 9.6	Not Supported	Minor ¹
			NM controls (C)	GSCR Sect. 9.7	Not Supported	Minor ¹
Synchronization	Yes	Certified	Remote access (C)	GSCR Sect. 9.8	Not Supported	Minor ¹
			EKTS (C)	GSCR Sect. 10, table 10-3	Not Supported	Minor ¹
Reliability	Yes	Certified	Line timing mode (R)	GSCR Sect. 11.1.1.2	Met	
Security	Yes	See note 3.	Internal Stratum 4 (R)	GSCR Sect. 11.1.2.2	Met	
			GR-512-CORE (R)	GSCR Sect. 12	Met	
VoIP System	No	Certified	DITSCAP (R)	GSCR Sect. 13	See note 3.	
			MOS 4.0 or better	GSCR App. 3	Met	
			ITU-T G.711 PCM Codec (R)	GSCR App. 3	Met	
			Security in accordance with DITSCAP (R)	GSCR App. 3	Met	
			NM (R)	GSCR App. 3	Met	
			Line Timing	GSCR App. 3	Met	
			Internal Clock	GSCR App. 3	Met	
Latency @ 60 ms or less (R)	GSCR App. 3	Met				
			IPV6 capable (R)	GSCR App. 3	Not Supported	See note 4.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features & Capabilities (continued)								
Features/ Capabilities	Critical	Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks	
LANs	No	Certified	LAN parameters (R)		GSCR App. 3	Met		
			CoS/QoS (R)		GSCR App. 3	Met		
			VLANs (R)		GSCR App. 3	Met		
			IEEE Standards Conformance (R)		GSCR App. 3	Met		
			.99999 availability (R)		GSCR App. 3	Met		
			Modular devices (R)		GSCR App. 3	Met		
			2 second link restoral (R)		GSCR App. 3	Met		
			LAN NM (R)		GSCR App. 3	Met		
Traffic Engineering (R)		GSCR App. 3	Met					
Network Gateway								
Gateway	Critical	Interface Status	GSCR Requirement Required or Conditional		Reference	Test Results	Remarks	
PSTN T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Trunking	Positive Identification Control (C)		CJCSI 6215.01B	Met	
				On-Netting (C)		CJCSI 6215.01B	Met	
				Off-Netting (C)		CJCSI 6215.01B	Met	
DRSN ⁵	Yes	Certified	Access	Alerting Signals and Tones (R)		GSCR Sect. 5.5	Met	
				Call Processing (R)		GSCR Sect. 4.4	Met	
				Call Treatments (R)		GSCR Sect. 4.1	Met	
				Analog busy/idle (R)		GSCR Sect. 4.3.4.1	Met	
			Voice	MOS (C)		CJCSI 6215.01B	Met	
				MLPP (C)		GSCR Sect. 3	Met	
Secure Calls (C)		CJCSI 6215.01B	Met					

Table 2-5. SUT Interoperability Requirements/Status (continued)

LEGEND:		
2W	- 2-Wire	EKTS - Electronic Key Telephone System
A/D	- Analog to Digital Conversion	G.711 - Standard for PCM of voice frequencies
ANSI	- American National Standards Institute	GR - Generic Requirement
App.	- Appendix	GSCR - Generic Switching Center Requirements
BER	- Bit Error Ratio	H.320 - Standard for narrowband VTC
BRI	- Basic Rate Interface	IEEE - Institute of Electrical and Electronics Engineers, Inc.
C	- Conditional	IPv6 - Internet Protocol version 6
CAS	- Channel Associated Signaling	ISDN - Integrated Services Digital Network
CCS	- Common Channel Signaling	ITU - International Telecommunication Union
CJCSI	- Chairman of the Joint Chiefs of Staff Instruction	ITU-T - ITU -Telecommunication Standardization Sector
CoS	- Class of Service	kbps - kilobits per second
D/A	- Digital to Analog Conversion	KXX - K= any number 2-8; X= any number 1-9
DISA	- Defense Information Systems Agency	LAN - Local Area Network
DISR	- Department of Defense Information Technology Standards Registry	Mbps - Megabits per second
DITSCAP	- Department of Defense Information Technology Security and Accreditation Process	MLPP - Multi-Level Precedence and Preemption
DN	- Directory Number	MOS - Mean Opinion Score
DoD	- Department of Defense	ms - milliseconds
DRSN	- Defense Red Switch Network	NI 1/2 - National ISDN Standard 1 or 2
DSN	- Defense Switched Network	NM - Network Management
DSS1	- Digital Subscriber Signaling 1	NX56 - Data format restricted to multiples of 56 kbps
E1	- European Basic Multiplex Rate (2.048 Mbps)	NX64 - Data format restricted to multiples of 64 kbps
E911	- 911 Emergency Service	PAT - Precedence Access Threshold
EIA	- Electronic Industries Alliance	PCM - Pulse Code Modulation
		PCM-24 - Pulse Code Modulation - 24 Channels
		PCM-30 - Pulse Code Modulation - 30 Channels
		PM - Program Manager
		PRI - Primary Rate Interface
		PSTN - Public Switched Telephone Network
		Q.955.3 - ISDN Signaling Standard for E1 MLPP
		QoS - Quality of Service
		R - Required
		Sect. - Section
		SS7 - Signaling System 7
		S/T - ISDN BRI 4-Wire Interface
		STE - Secure Terminal Equipment
		STU-III - Secure Telephone Unit-3rd generation
		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
		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 Conferencing
		WWNDP - Worldwide Numbering and Dialing Plan
NOTES:		
1	This interface, feature, or capability is not supported by the SUT. There is no operational impact because it is not a critical requirement for a PBX 1.	
2	The SUT provides a voice conference bridge capability of up to a total of 60 simultaneous conferees for each Meeting Hub. The GSCR requires a minimum of 10 simultaneous conferences with 20 conferees each. The vendor stated during testing that the SUT can stack Meeting Hubs to attain this requirement, however, multiple Meeting Hubs were not tested; therefore preset conferencing is not covered under this certification. A waiver from the Joint Staff would be required in order for the SUT to be fielded using the tested preset conferencing configuration with a single Meeting Hub.	
3	DITSCAP information assurance testing is accomplished via DISA-led Information Assurance test teams and published in a separate report.	
4	Although IPv6 is a required capability, the SUT's current inability to provide this capability is deemed to have only a minor operational risk when operating within the DSN. This is based on the current overall IPv6 capabilities of the DoD. To meet the DoD's goal of transitioning to IPv6, the SUT's IPv6 capability will be reassessed during future hardware and software upgrades.	
5	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.	