



DEFENSE INFORMATION SYSTEMS AGENCY

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

23 July 2008

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Mitel 3300 Internet Protocol Communications Platform (ICP) with Software Release 8.0.8.12_2

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
(c) through (e), see enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Mitel 3300 ICP with Software Release 8.0.8.12_2 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 SUT is certified to support DSN Assured Services over Internet Protocol with any Assured Services Voice Application Local Area Network (ASVALAN) on the DSN APL. The SUT is also certified for joint use with any Voice Application Local Area Network (VALAN) on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), Command and Control (C2) users and Special C2 users are not authorized to be served by the SUT connected to a VALAN. The identified test discrepancies shown in the Certification Testing Summary (enclosure 2) have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC, or authorized by the Program Management Office (PMO) for use within the DSN. This certification expires upon system changes that 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 1 October through 9 November 2007, and 10 March through 4 April 2008. The vendor submitted their LoC on 30 May 2008, and review was completed by JITC on 3 June 2008. Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The interoperability test summary of the SUT is indicated in table 1. The PBX 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 1-4, verified through vendor submission of an LoC.
- e. 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 Certified	T1 CAS is offered by the SUT; however it does not support the critical MLPP requirements. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
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	E1 PRI is supported by the SUT; however it was not tested. The SUT E1 PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface 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	No	Not Tested	ISDN BRI is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP (100BaseT)	No	Certified	Met all CRs and FRs. The SUT is certified with any ASVALAN or VALAN on the DSN APL. ¹
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: Selective call rejection, Denied originating service, Code restriction and diversion, Add-on transfer, conference calling, and call hold. Common features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.
Attendant	No	Certified	Met all CRs and FRs for Attendant Services.

Table 1. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities				
Features and Capabilities	Critical	Status	Remarks	
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: Trace of terminating calls, Outgoing call trace. Public safety features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.	
Preset Conferencing	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Precedence Access Threshold	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Network Management	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
ISDN Services (EKTS)	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
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)	Not Tested	E1 CAS is not supported by the SUT. Since this is not a required interface for a PBX 1, there is no operational impact.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 PRI (ITU-T Q.931)	No	Not Tested	E1 PRI is supported by the SUT; however it was not tested and is therefore not certified. Since this is not a required interface for a PBX 1, there is no operational impact.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN ⁶	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.

JITC Memo, JTE, Special Interoperability Test Certification of Mitel 3300 Internet Protocol Communications Platform (ICP) with Software Release 8.0.8.12_2

Table 1. SUT Interoperability Test Summary (continued)

LEGEND:	
100BaseT	- 100 Mbps (Baseband Operation, Twisted Pair) Ethernet
ANSI	- American National Standards Institute
APL	- Approved Products List
ASVALAN	- Assured Services Voice Application Local Area Network
BRI	- Basic Rate Interface
C2	- Command and Control
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
ISDN	- Integrated Services Digital Network
IT	- Information Technology
ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector
JITC	- Joint Interoperability Test Command
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
PBX 1	- Private Branch Exchange 1
PM	- Program Manager
PMO	- Program Management Office
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
VALAN	- Voice Application Local Area Network
VoIP	- Voice over Internet Protocol
NOTES:	
1 The SUT is certified to support DSN Assured Services over Internet Protocol with any ASVALAN on the DSN APL. The SUT is also certified for joint use with any VALAN on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), C2 users and Special C2 users are not authorized to be served by the SUT connected to a VALAN.	
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:	
a. Conformance 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.	

Table 2. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> Framing (R) Line Code (R) Signaling (R: PRI only) Alarm and Restoral Requirements (R) Alarm and Restoral Requirements (C) WWNDP (R) Outpulsing digit formats (C: CAS only) Routing (C) Trunk Groups (C) Call Processing (R) 	<ul style="list-style-type: none"> GSCR Section 7.1.1 GSCR Section 7.1 and 7.2 GSCR Section 5 GSCR Section 7.1.4 GSCR Section 7.2.2 GSCR Section 4.5.1 GSCR Section 4.5.2 GSCR Section 4.2 GSCR Section 2.5.5 & 2.5.6 GSCR Section 4.1, 4.1.2, 4.1.3, 4.1.4, 4.1.5, 4.3.3, 4.5.1 GSCR Section 7.3 GSCR Section 2.3.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> PCM-24/PCM-30 Interoperation (C) Direct Inward Dialing (C) 	<ul style="list-style-type: none"> GSCR Section 7.3 GSCR Section 2.3.2
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Voice	<ul style="list-style-type: none"> MOS (R) MLPP (R) 	<ul style="list-style-type: none"> CJCSI 6215.01C GSCR Section 3.1, 3.2, 3.2.1, 3.2.2, 3.3, 3.4.2, 3.5.1 CJCSI 6215.01C
		Facsimile	<ul style="list-style-type: none"> Secure calls (R) Analog: ITU-T T.4 (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 6215.01C GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 CJCSI 6215.01C
		VTC	<ul style="list-style-type: none"> ITU-T H.320 (R: PRI only) 	<ul style="list-style-type: none"> DISR
DSN Line Interfaces				
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) 	<ul style="list-style-type: none"> GSCR Section 2.1.1 GSCR Section 5.2.1 & 5.2.2 GSCR Section 5.2.1 GSCR Section 5.5 GSCR Section 4.5.1.2, 4.5.1.3, 4.5.1.4 & 4.5.1.5 GSCR Section 4.1.1 – 4.1.5 GSCR Section 4.3.3 GSCR Section 4.3.4.1
ISDN BRI NI 1/2 (ANSI T1.619a)	No		<ul style="list-style-type: none"> 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"> CJCSI 6215.01C GSCR Section 3.1.3 GSCR Section 3.4.3/3.9 GSCR Section 5.2.2 CJCSI 6215.01C
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> MOS (R) Announcements (R) MLPP (R) Ground Start Line (R) Secure Calls (R) 	<ul style="list-style-type: none"> CJCSI 6215.01C GSCR Section 3.1.3 GSCR Section 3.4.3/3.9 GSCR Section 5.2.2 CJCSI 6215.01C
		Facsimile	<ul style="list-style-type: none"> Analog: ITU-T T.4 (R) 	<ul style="list-style-type: none"> DISR
VoIP 100BaseT	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 6215.01C GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 CJCSI 6215.01C
		VTC	<ul style="list-style-type: none"> ITU-T H.320 (R: BRI only) 	<ul style="list-style-type: none"> DISR

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, conference calling, and call hold (C) • Call forwarding (C) • Call pick-up (C) 	<ul style="list-style-type: none"> • GSCR Section 2.1.2 • GSCR Section 2.1.3 • GSCR Section 2.1.4 • GSCR Section 2.1.5 • GSCR Section 2.1.6 • GSCR Section 2.1.7 • GSCR Section 2.1.8 • GSCR Section 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 Section 2.2.1 • GSCR Section 2.2.2 • GSCR Section 2.2.3 • GSCR Section 2.2.4 • GSCR Section 2.2.5 • GSCR Section 2.2.6 • GSCR Section 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) 	<ul style="list-style-type: none"> • GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3
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 Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6.1 • GSCR Section 2.6.2 • GSCR Section 2.6.3 • GSCR Section 2.6.4 • GSCR Section 2.6.5 • GSCR Section 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 Section 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 Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 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) 	<ul style="list-style-type: none"> • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4
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

JITC Memo, JTE, Special Interoperability Test Certification of Mitel 3300 Internet Protocol Communications Platform (ICP) with Software Release 8.0.8.12_2

Table 2. PBX 1 Requirements (continued)

DSN Features & Capabilities																																																																
Feature/ Capability	Critical	Requirements Required or Conditional		References																																																												
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 Section 11.1.1.2 GSCR Section 11.1.2.2 																																																												
Reliability	Yes	<ul style="list-style-type: none"> GR-512-CORE (R) 		<ul style="list-style-type: none"> GSCR Section 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 Section 13 																																																												
DSN Features & Capabilities (continued)																																																																
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"> Voice Quality with MOS of 4.0 or better Class of Service (CoS) and Quality of Service (QoS) ITU-T G.711 PCM Codec Traffic Engineering Security Network management Line timing Internal Clock Latency ≤ 60 milliseconds Packet Loss IPv6 capable 		<ul style="list-style-type: none"> GSCR Appendix 3, para. A3.2.1 GSCR Appendix 3, para. A3.3.2 & A3.3.3 GSCR Appendix 3, para. A3.2.2 GSCR Appendix 3, para. A3.3.4.4 GSCR Appendix 3, para. A3.2.4 GSCR Appendix 3, para. A3.2.5 GSCR Appendix 3, para. A3.2.6 GSCR Appendix 3, para. A3.2.6 GSCR Appendix 3, para. A3.2.7 GSCR Appendix 3, para. A3.3.1.3 GSCR Section 1, para. 1.7 and Appendix 3, para. A3.2.8 																																																												
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 6215.01C CJCSI 6215.01C CJCSI 6215.01C 																																																												
DRSN ²	Yes	Access	<ul style="list-style-type: none"> Alerting Signals and Tones (R) Call Processing (C) Call Treatments (R) Analog busy/idle (R) 	<ul style="list-style-type: none"> GSCR Section 5.5 GSCR Section 4.4 GSCR Section 4.1 GSCR Section 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 6215.01C GSCR Section 3 CJCSI 6215.01C 																																																												
<p>LEGEND:</p> <table border="0"> <tr> <td>100BaseT - 100 Mbps (Baseband Operation, Twisted Pair) Ethernet</td> <td>G.711 - Standard for PCM of Voice Frequencies</td> <td>PAT - Precedence Access Threshold</td> </tr> <tr> <td>2W - 2-Wire</td> <td>GR - Generic Requirement</td> <td>PBX 1 - Private Branch Exchange 1</td> </tr> <tr> <td>A/D - Analog to Digital Conversion</td> <td>GR-512-CORE - LSSGR; Reliability, Section 12</td> <td>PCM - Pulse Code Modulation</td> </tr> <tr> <td>ANSI - American National Standards Institute</td> <td>GR-815 - Generic Requirements For Network Element/Network System (NE/NS) Security</td> <td>PCM-24 - Pulse Code Modulation - 24 Channels</td> </tr> <tr> <td>BER - Bit Error Ratio</td> <td>GSCR - Generic Switching Center Requirements</td> <td>PCM-30 - Pulse Code Modulation - 30 Channels</td> </tr> <tr> <td>BRI - Basic Rate Interface</td> <td>H.320 - Standard for Narrowband VTC</td> <td>PRI - Primary Rate Interface</td> </tr> <tr> <td>C - Conditional</td> <td>IPv6 - Internet Protocol version 6</td> <td>PSTN - Public Switched Telephone Network</td> </tr> <tr> <td>CAS - Channel Associated Signaling</td> <td>ISDN - Integrated Services Digital Network</td> <td>Q.955.3 - ISDN Signaling Standard for E1 MLPP</td> </tr> <tr> <td>CCS - Common Channel Signaling</td> <td>IT - Information Technology</td> <td>R - Required</td> </tr> <tr> <td>CJCSI - Chairman of the Joint Chiefs of Staff Instruction</td> <td>ITU-T - International Telecommunication Union-Telecommunication Standardization Sector</td> <td>SS7 - Signaling System 7</td> </tr> <tr> <td>D/A - Digital to Analog Conversion</td> <td>kbps - kilobits per second</td> <td>STE - Secure Terminal Equipment</td> </tr> <tr> <td>DIACAP - DoD Information Assurance Certification and Accreditation Process</td> <td>KXX - K= any number 2-8; X= any number 1-9</td> <td>STIGs - Security Technical Implementation Guides</td> </tr> <tr> <td>DISR - DoD IT Standards Registry</td> <td>LSSGR - Local Access and Transport Area (LATA) Switching Systems Generic Requirements</td> <td>STU-III - Secure Telephone Unit -3rd generation</td> </tr> <tr> <td>DITSCAP - DoD IT Security Certification and Accreditation Process</td> <td>Mbps - Megabits per second</td> <td>T1 - Digital Transmission Link Level 1 (1.544 Mbps)</td> </tr> <tr> <td>DoD - Department of Defense</td> <td>MFR1 - Multi-Frequency Recommendation 1</td> <td>T1.619a - SS7 and ISDN MLPP Signaling Standard for T1</td> </tr> <tr> <td>DP - Dial Pulse</td> <td>MLPP - Multi-Level Precedence and Preemption</td> <td>T.4 - Standardization of Group 3 facsimile terminals for document transmission</td> </tr> <tr> <td>DRSN - Defense Red Switch Network</td> <td>MOS - Mean Opinion Score</td> <td>VBD - Variable bit data</td> </tr> <tr> <td>DSN - Defense Switched Network</td> <td>NI 1/2 - National ISDN Standard 1 or 2</td> <td>VoIP - Voice over Internet Protocol</td> </tr> <tr> <td>DTMF - Dual Tone Multi-Frequency</td> <td>NX56 - Data format restricted to multiples of 56 kbps</td> <td>VTC - Video Teleconferencing</td> </tr> <tr> <td>E1 - European Basic Multiplex Rate (2.048 Mbps)</td> <td>NX64 - Data format restricted to multiples of 64 kbps</td> <td>WWNDP - Worldwide Numbering and Dialing Plan</td> </tr> </table>					100BaseT - 100 Mbps (Baseband Operation, Twisted Pair) Ethernet	G.711 - Standard for PCM of Voice Frequencies	PAT - Precedence Access Threshold	2W - 2-Wire	GR - Generic Requirement	PBX 1 - Private Branch Exchange 1	A/D - Analog to Digital Conversion	GR-512-CORE - LSSGR; 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<p>NOTES:</p> <p>1 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.</p> <p>2 Facsimile, data, and VTC services are not provided via the DSN to DRSN interface.</p>																																																																

JITC Memo, JTE, Special Interoperability Test Certification of Mitel 3300 Internet Protocol Communications Platform (ICP) with Software Release 8.0.8.12_2

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

6. The JITC point of contact is Mr. Steven Lesneski, DSN 879-5400, commercial (520) 538-5400, FAX DSN 879-4347, or e-mail to steven.lesneski@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0726201.

FOR THE COMMANDER:



RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

2 Enclosures a/s

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

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Washington, DC 20301

U.S. Joint Forces Command, J68, Net-Centric Integration, Communications, and Capabilities
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Defense Information Systems Agency (DISA), ATTN: GS23 (Mr. McLaughlin), Room 5W23,
5275 Leesburg Pike (RTE 7), Falls Church, VA 22041

ADDITIONAL REFERENCES

- (c) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007
- (d) Defense Information Systems Agency, "Department of Defense Voice Networks Generic Switching Center Requirements (GSCR), Errata Change 2," 14 December 2006, Revised 27 March 2007
- (e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Mitel 3300 Internet Protocol Communications Platform (ICP) with Software Release 8.0.8.12_2, hereinafter referred to as the System Under Test (SUT).

2. PROPONENT. United States Army National Guard.

3. PROGRAM MANAGER. Mr. Brian Gunter, JFHQ GA, 5019 GA Highway, Room 171, Ellenwood, Georgia, 30249, E-mail: brian.gunter@us.army.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 is built upon Mitel's Data Integrated Voice Applications architecture delivering call management, applications, and desktop solutions. The SUT delivers a scalable call control that utilizes Internet Protocol (IP) while supporting Time Division Multiplexing (TDM)-based telephony for legacy devices and Public Switched Telephone Network (PSTN) and Defense Switched Network (DSN) connectivity. Mitel's architecture uses the IP network to connect IP telephony devices and provides a supplementary TDM subsystem to switch calls between traditional telephone devices. The SUT provides call setup, tear down, and signaling between Ethernet IP-connected telephones. For traditional telephony, such as PSTN and DSN trunks, call handling is also handled by the SUT through a conventional TDM circuit-switched subsystem.

The SUT components include: the Mitel 3300 Compact eXchange integrated (CXi) Controller, 3300 Medium eXchange expandable (MXe), two Mitel Universal Network Service Units (NSU), Mitel Universal Analog Service Unit (ASU) II, and the Mitel Peripheral Node. The SUT was tested and is certified with the following optional peripherals: NSU and Peripheral Node. The SUT is certified with or without any combination of these optional peripherals. The Mitel 3300 ICP solution is managed with a workstation running Microsoft Windows XP Service Pack 2 and Internet Explorer browser Version 6. In addition, the workstation is loaded with Java Runtime library Version 6, Update 1 and HyperTerminal. The system provides secure access through a web interface to an embedded menu-based tool for configuration and maintenance.

6. OPERATIONAL ARCHITECTURE. The 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.

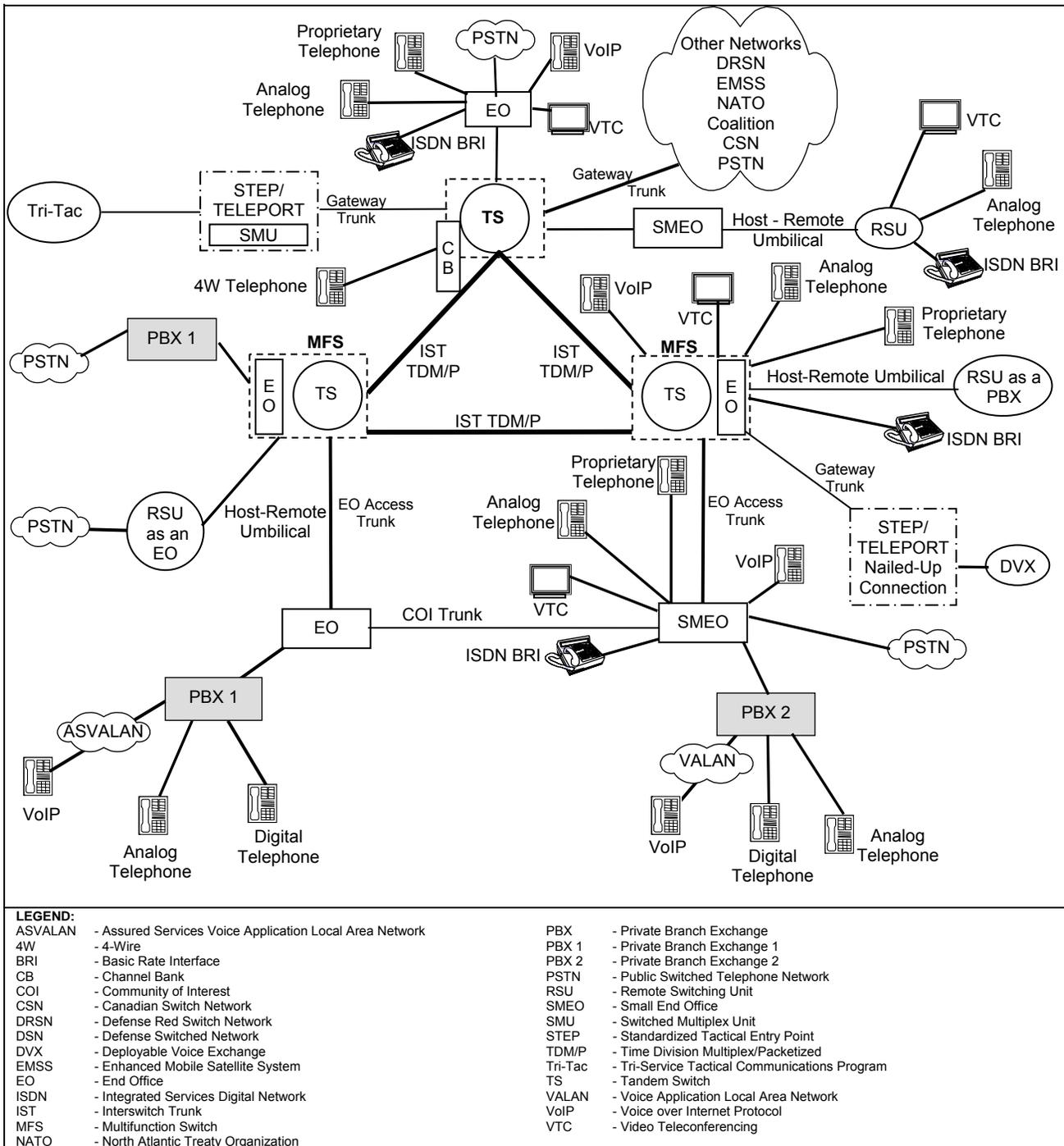


Figure 2-1. DSN Architecture

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

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

b. 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 1-4, verified through vendor submission of LoC.

Table 2-1. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	Critical	Requirements Required or Conditional		References
T1 CAS (MFR1, DTMF, DP)	No	Trunking	<ul style="list-style-type: none"> Framing (R) Line Code (R) Signaling (R: PRI only) Alarm and Restoral Requirements (R) Alarm and Restoral Requirements (C) WWNDP (R) Outpulsing digit formats (C: CAS only) Routing (C) Trunk Groups (C) Call Processing (R) 	<ul style="list-style-type: none"> GSCR Section 7.1.1 GSCR Section 7.1.4 GSCR Section 5 GSCR Section 7.1.4 GSCR Section 7.2.2 GSCR Section 4.5.1 GSCR Section 4.5.2 GSCR Section 4.2 GSCR Section 2.5.5 & 2.5.6 GSCR Section 4.1, 4.1.2, 4.1.3, 4.1.4, 4.1.5, 4.3.3, 4.5.1 GSCR Section 7.3 GSCR Section 2.3.2
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		<ul style="list-style-type: none"> PCM-24/PCM-30 Interoperation (C) Direct Inward Dialing (C) 	
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 6215.01C GSCR Section 3.1, 3.2, 3.2.1, 3.2.2, 3.3, 3.4.2, 3.5.1 CJCSI 6215.01C
		Facsimile	<ul style="list-style-type: none"> Analog: ITU-T T.4 (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 6215.01C GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 GSCR Section 3.10 CJCSI 6215.01C
		VTC	<ul style="list-style-type: none"> ITU-T H.320 (R: PRI only) 	<ul style="list-style-type: none"> DISR

Table 2-1. PBX 1 Requirements (continued)

DSN Line Interfaces				
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) 	<ul style="list-style-type: none"> • GSCR Section 2.1.1 • GSCR Section 5.2.1 & 5.2.2 • GSCR Section 5.2.1 • GSCR Section 5.5 • GSCR Section 4.5.1.2, 4.5.1.3, 4.5.1.4 & 4.5.1.5 • GSCR Section 4.1.1 – 4.1.5 • GSCR Section 4.3.3 • GSCR Section 4.3.4.1
ISDN BRI NI 1/2 (ANSI T1.619a)	No		<ul style="list-style-type: none"> • Call Treatments (R) • 2W user access (R: 2-Wire Analog only) • Analog busy/idle (R: 2-Wire Analog only) 	
2-Wire Proprietary Digital	No	Voice	<ul style="list-style-type: none"> • MOS (R) • Announcements (R) • MLPP (R) • Ground Start Line (R) • Secure Calls (R) 	<ul style="list-style-type: none"> • CJCSI 6215.01C • GSCR Section 3.1.3 • GSCR Section 3.4.3/3.9 • GSCR Section 5.2.2 • CJCSI 6215.01C
		Facsimile	<ul style="list-style-type: none"> • Analog: ITU-T T.4 (R) 	<ul style="list-style-type: none"> • DISR
VoIP 100BaseT	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 6215.01C • GSCR Section 3.10 • GSCR Section 3.10 • GSCR Section 3.10 • GSCR Section 3.10 • CJCSI 6215.01C
		VTC	<ul style="list-style-type: none"> • ITU-T H.320 (R: BRI 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, conference calling, and call hold (C) • Call forwarding (C) • Call pick-up (C) 		<ul style="list-style-type: none"> • GSCR Section 2.1.2 • GSCR Section 2.1.3 • GSCR Section 2.1.4 • GSCR Section 2.1.5 • GSCR Section 2.1.6 • GSCR Section 2.1.7 • GSCR Section 2.1.8 • GSCR Section 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 Section 2.2.1 • GSCR Section 2.2.2 • GSCR Section 2.2.3 • GSCR Section 2.2.4 • GSCR Section 2.2.5 • GSCR Section 2.2.6 • GSCR Section 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) 		<ul style="list-style-type: none"> • GSCR Section 2.4.1 • GSCR Section 2.4.2 • GSCR Section 2.4.3
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 Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6 • GSCR Section 2.6.1 • GSCR Section 2.6.2 • GSCR Section 2.6.3 • GSCR Section 2.6.4 • GSCR Section 2.6.5 • GSCR Section 2.7

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	Critical	Requirements Required or Conditional	References
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 Section 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 Section 2.11.1 • GSCR Section 2.11.1 • GSCR Section 2.11.1.1 • GSCR Section 2.11.1.2 • GSCR Section 2.11.1.3 • GSCR Section 2.11.1.4 • GSCR Section 2.11.1.5 • GSCR Section 2.11.1.6 • GSCR Section 2.11.1.7 • GSCR Section 2.11.1.8 • GSCR Section 2.11.1.9 • GSCR Section 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) 	<ul style="list-style-type: none"> • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12 • GSCR Section 2.12.1-4
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 Section 11.1.1.2 • GSCR Section 11.1.2.2
Reliability	Yes	<ul style="list-style-type: none"> • GR-512-CORE (R) 	<ul style="list-style-type: none"> • GSCR Section 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 Section 13
DSN Features & Capabilities (continued)			
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"> • Voice Quality with MOS of 4.0 or better • Class of Service (CoS) and Quality of Service (QoS) • ITU-T G.711 PCM Codec • Traffic Engineering • Security • Network management • Line timing • Internal Clock • Latency ≤ 60 milliseconds • Packet Loss • IPv6 capable 	<ul style="list-style-type: none"> • GSCR Appendix 3, para. A3.2.1 • GSCR Appendix 3, para. A3.3.2 & A3.3.3 • GSCR Appendix 3, para. A3.2.2 • GSCR Appendix 3, para. A3.3.4.4 • GSCR Appendix 3, para. A3.2.4 • GSCR Appendix 3, para. A3.2.5 • GSCR Appendix 3, para. A3.2.6 • GSCR Appendix 3, para. A3.2.6 • GSCR Appendix 3, para. A3.2.7 • GSCR Appendix 3, para. A3.3.1.3 • GSCR Section 1, para. 1.7 and Appendix 3, para. A3.2.8

Table 2-1. 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 6215.01C CJCSI 6215.01C CJCSI 6215.01C
DRSN ²	Yes	Access	<ul style="list-style-type: none"> Alerting Signals and Tones (R) Call Processing (C) Call Treatments (R) Analog busy/idle (R) 	<ul style="list-style-type: none"> GSCR Section 5.5 GSCR Section 4.4 GSCR Section 4.1 GSCR Section 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 6215.01C GSCR Section 3 CJCSI 6215.01C
LEGEND: 100BaseT - 100 Mbps (Baseband Operation, Twisted Pair) Ethernet 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 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 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) G.711 - Standard for PCM of Voice Frequencies GR - Generic Requirement GR-512-CORE - 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 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 SS7 - Signaling System 7 STE - Secure Terminal Equipment STIGs - Security Technical Implementation Guides 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 T.4 - Standardization of Group 3 facsimile terminals 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.				

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 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.

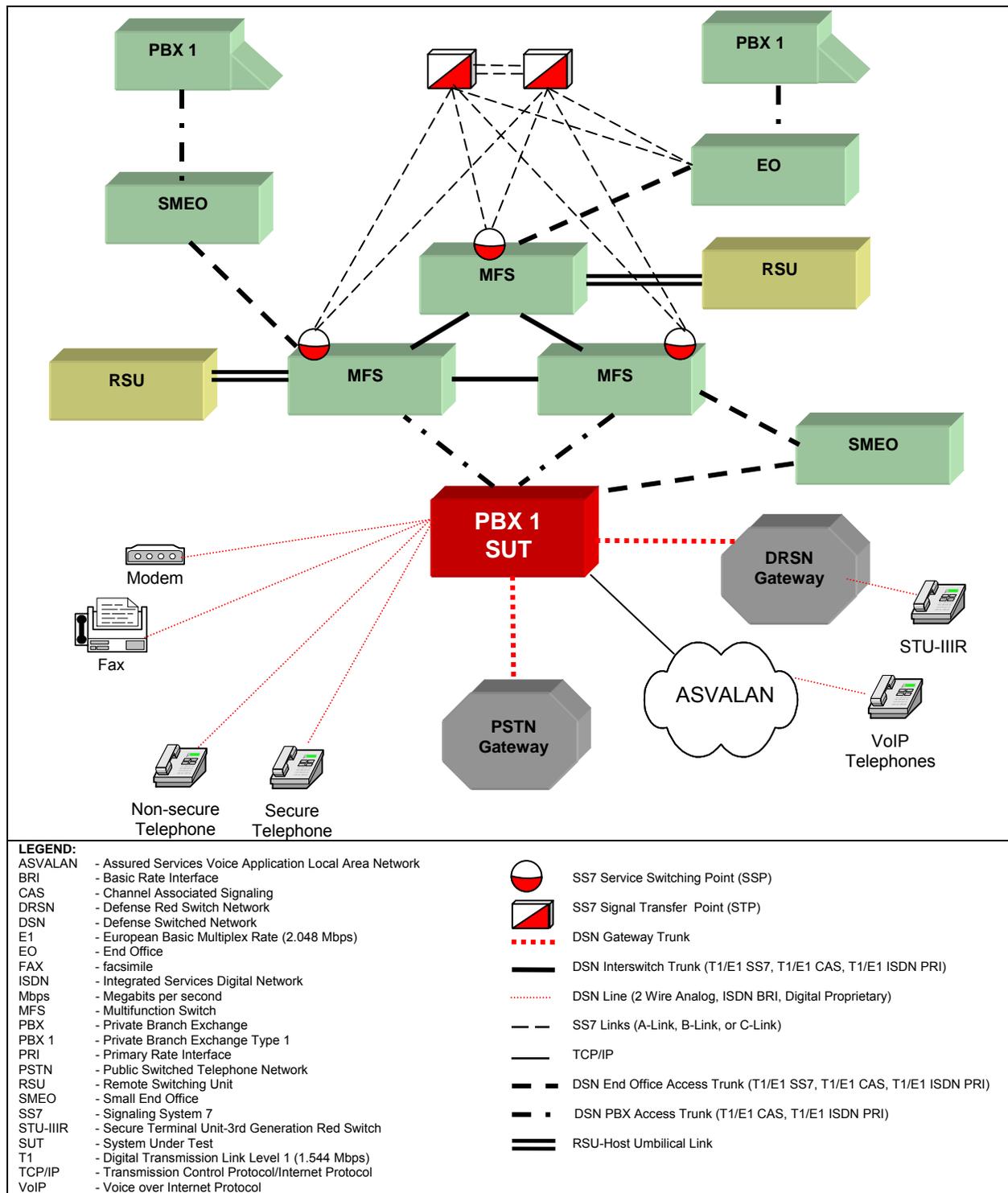
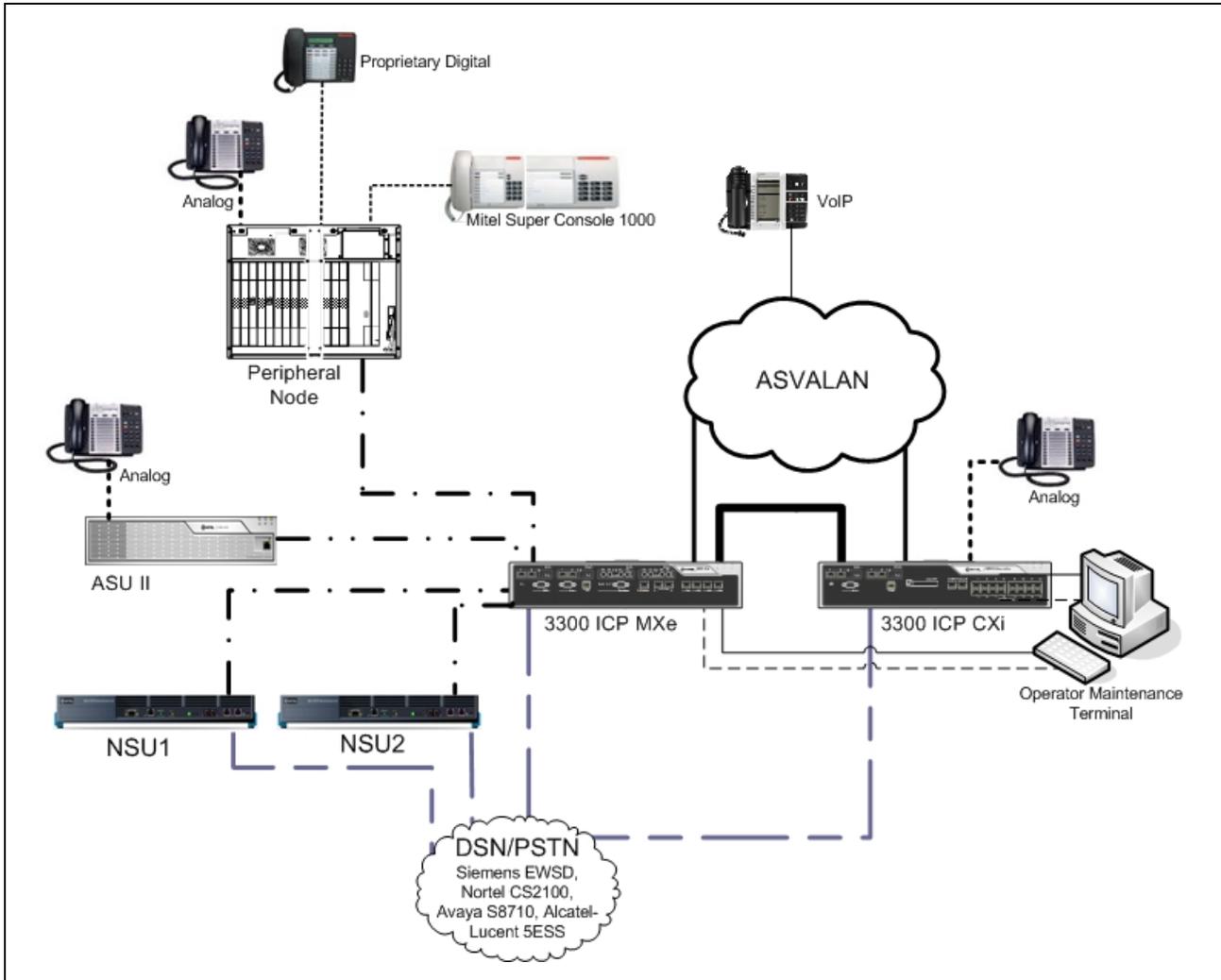


Figure 2-2. Test Configuration



LEGEND:

- 5ESS - Class 5 Electronic Switching System
- 100BaseT - 100 Mbps (Baseband Operation, Twisted Pair) Ethernet
- ASVALAN - Assured Services Voice Application Local Area Network
- ASU - Analog Service Unit
- CAS - Channel Associated Signaling
- CS - Communication Server
- CXi - Compact eXchange integrated
- DSN - Defense Switched Network
- EIA - Electronic Industries Alliance
- EIA-232 - Standard for defining the mechanical and electrical characteristics for connecting Data Terminal Equipment (DTE) and Data Circuit-terminating Equipment (DCE) data communications devices
- EWSD - Elektronisches Wählsystem Digital
- ICP - Internet Protocol Communications Platform
- Mbps - Megabits per second
- MXe - Medium eXchange expandable
- NSU - Network Service Unit
- PRI - Primary Rate Interface
- PSTN - Public Switched Telephone Network
- SUT - System Under Test
- T1 - Digital Transmission Link Level 1 (1.544 Mbps)
- VoIP - Voice over Internet Protocol

- 2-Wire Analog
- Proprietary Digital Signaling
- Proprietary Signaling on Copper
- Proprietary Signaling on Fiber
- EIA-232 Serial Management Port
- Management 10/100BaseT
- ASVALAN 10/100BaseT
- T1 PRI to DSN/T1 CAS to PSTN
- T1 Failover Link CXi to MXe

Figure 2-3. SUT VoIP Test Configuration

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

Table 2-2. Tested System Configurations

System Name		Software Release	
Nortel CS2100		Succession Enterprise (SE) 09.1	
Siemens EWSD		19d with Patch Set 46	
Alcatel-Lucent 5ESS		5E16.2, Broadcast Warning Message (BWM) 07-0003	
Avaya S8710		Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)	
Mitel 3300 ICP with Software Release 8.0.8.12_2 (SUT)	Hardware	Part Number	Software/Firmware
	ASU II	50005105 Rev C	ARC International - MQX V 2.4. Embedded
	Peripheral Cabinet	50004203 Rev H	Mitel - Proprietary Task Scheduler Software V 38.0.5.11 Embedded
	3300 ICP MXe	50005080 Rev B	Wind River - VxWorks V 2.6, Linux 2.4.18 Embedded Mitel Call Control Engine - Proprietary OS Embedded Go-Ahead Web Server 2.1.8. Embedded
	3300 ICP CXi	50005097 Rev B	Wind River - VxWorks V 2.6, Linux 2.4.18 Embedded Mitel Call Control Engine - Proprietary OS Embedded Go-Ahead Web Server 2.1.8. Embedded
	NSU	50004990 Rev A	Wind River - VxWorks V 2.6. Embedded
Peripheral Equipment		Software	
Operator Maintenance Terminal		Windows XP Service Pack 2, Internet Explorer browser Version 6, Java Runtime Library Version 6, Update 1, HyperTerminal	
Telephones			
Telephone Type	Model	Firmware	
Mitel DNI	Superset 4150	Main software release 6.4, Boot software 4.4	
	Superset 4025	Main software release 4.3, Boot software 4.3	
	Superconsole 1000	N/A	
Mitel IP	5310 Conference Unit	N/A	
	5303 Conference Saucer	N/A	
	5330 IP Set	ARC International - MQX V 2.50	
	5340 IP Set	RTCS V 2.93	
2-Wire Analog	Panasonic KX-TS15-W	Not Applicable	
LEGEND:			
5ESS - Class 5 Electronic Switching System		MXe - Medium eXchange expandable	
ARC - Advanced Research Corporation		N/A - Not Applicable	
ASU II - Analog Service Unit II		NSU - Network Service Unit	
CS - Communication Server		OS - Operating System	
CXi - Compact eXchange integrated		Rev - Revision	
DNI - Digital Network Interface		RTCS - Realtime TCP/IP Communications Suite	
EWSD - Elektronisches Wählsystem Digital		SUT - System Under Test	
ICP - IP Communications Platform		TCP/IP - Transmission Control Protocol/Internet Protocol	
IP - Internet Protocol		V - Version	
MQX - Message Queue eXecutive			

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs for Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2) American National Standards Institute (ANSI) T1.619a. The SUT offers a T1 Channel Associated Signaling (CAS) interface; however, it does not support the critical Multi-Level Precedence and Preemption (MLPP) requirements and is therefore not certified by JITC, or authorized by the DSN Program Management Office (PMO) for use within the DSN. The SUT offers a European Basic Multiplex Rate (E1) PRI interface; however, it was not tested and is not covered under this certification. The SUT E1 PRI interface is not authorized for use within the DSN by the PMO.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Analog (GR-506-CORE), 2-Wire Digital (Proprietary) and Voice over Internet Protocol (VoIP) DSN line interfaces.

(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 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 Assured Services voice Application Local Area Network (ASVALAN) listed on the DSN Approved Products List (APL). The SUT is also certified for joint use with any Voice Application Local Area Network (VALAN) on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), Command and Control (C2)

users and Special C2 users are not authorized to be served by the SUT connected to a VALAN.

(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 4.31. The average inter-switch MOS was 4.29. This average was based on a total of 150 intra-switch and inter-switch calls with the lowest intra-switch MOS being 4.16 and the lowest inter-switch MOS of 4.02.

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 Mitel IP phones met the critical interoperability requirements for certification: 5330 IP and 5340 IP. Shared access (i.e., same switch port is shared by PC and IP phone), was not tested and is not certified with this configuration. It was noted during testing that the vendor's original configuration was first tested using DHCP (Dynamic Host Configuration Protocol). With proper configuration of the ASVALAN, this set up did work. The Mitel 3300 was later tested with static IP addresses assigned to all components in the SUT with out any interoperability issues. Statically assigning the IP addresses is the preferred method for

deployment. The 5330 and 5340 IP phones met all critical requirements as determined in the GSCR, appendix 3 when connected to a certified ASVALAN.

b. Scalability. The manufacturer states that the SUT can support any combination of up to 250 MXe / CXi controllers in a single system. This capability was not tested. The test was conducted using one MXe and one CXi controller. Per the manufacturer, each MXe can support a maximum of 1400 IP phone extensions and the CXi can support a maximum of 100 IP phone extensions.

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 100 independent, five-minute calls was measured to be 55.6 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 percent averaged over any five-minute period. The SUT average packet loss was measured at 0.00 percent for 100 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:

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 the PSTN and the Defense Red Switch Network (DRSN) Gateways. The interfaces certified for PSTN are T1 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	Not Certified	T1 CAS is offered by the SUT; however it does not support the critical MLPP requirements. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
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	E1 PRI is supported by the SUT; however it was not tested. The SUT E1 PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface 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	No	Not Tested	ISDN BRI is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Certified	Met all CRs and FRs.
VoIP (100BaseT)	No	Certified	Met all CRs and FRs. The SUT is certified with any ASVALAN or VALAN on the DSN APL. ¹

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

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: Selective call rejection, Denied originating service, Code restriction and diversion, Add-on transfer, conference calling, and call hold. Common features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.	
Attendant	No	Certified	Met all CRs and FRs for Attendant Services.	
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: Trace of terminating calls, Outgoing call trace. Public safety features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.	
Preset Conferencing	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Precedence Access Threshold	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
Network Management	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
ISDN Services (EKTS)	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.	
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)	Not Tested	E1 CAS is not supported by the SUT. Since this is not a required interface for a PBX 1, there is no operational impact.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all CRs and FRs.
	E1 PRI (ITU-T Q.931)	No	Not Tested	E1 PRI is supported by the SUT; however it was not tested and is therefore not certified. Since this is not a required interface for a PBX 1, there is no operational impact.
	Ground Start Line	Yes	Certified	Met all CRs and FRs.
DRSN ⁶	TPC 2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all CRs and FRs.

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

LEGEND:	
100BaseT	- 100 Mbps (Baseband Operation, Twisted Pair) Ethernet
ANSI	- American National Standards Institute
APL	- Approved Products List
ASVALAN	- Assured Services Voice Application Local Area Network
BRI	- Basic Rate Interface
C2	- Command and Control
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
ISDN	- Integrated Services Digital Network
IT	- Information Technology
ITU-T	- International Telecommunication Union - Telecommunication Standardization Sector
JITC	- Joint Interoperability Test Command
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
PBX 1	- Private Branch Exchange 1
PM	- Program Manager
PMO	- Program Management Office
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
VALAN	- Voice Application Local Area Network
VoIP	- Voice over Internet Protocol
NOTES:	
1 The SUT is certified to support DSN Assured Services over Internet Protocol with any ASVALAN on the DSN APL. The SUT is also certified for joint use with any VALAN on the DSN APL. However, since VALANs do not support the Assured Services Requirements detailed in reference (c), C2 users and Special C2 users are not authorized to be served by the SUT connected to a VALAN.	
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:	
a. Conformance 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>.