Statement of Work (SOW)
for the Telephony Communications expansion
at CFB Petawawa, Petawawa, ON

19 November 2020

SSC IPO Technical Authority/Desk officers and Site POC:

Technical Authority/Desk officer:

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Contents

Acronyms

Acronyms used in this document

AES	Advanced Encryption Standard
APL	Approved Products List (DISA)
AS-SIP	Assured Services SIP
BTU	British Thermal Units
CAT6	Category 6 wired cabling
CLASS	Custom Local Area Signaling Services
CDP	Coordinated Dial Plan
CNSSI	Committee on National Security Systems Instructions
CSN	Canadian Switched Network (voice)
DISA	Defence Information Systems Agency
DNET	Defense Network (data)
DSN	Defense Switch Network (voice)
DTN	Defense Telephone Network
DWAN	Defence Wide Area Network
GDNS	Global Defence Network Services
IEC	International Electrotechnical Commission
IETF	Internet Engineering Task Force
IP	Internet Protocol
MLPP	Multilevel Precedence and Pre-emption
MTR	Main Telephone Room
MUDG	Military Unique Deployment Guide
NEMA	National Electrical Manufacturers Association
POE	Power Over Ethernet
PRI	Primary Rate Interface
PSTN	Public Service Telephone Network
QSIG	Protocol for Integrated Services Digital Network (ISDN) communications based on the Q.931 standard
SM	Single Mode (optical fibre)
SOW	Statement of Work
SRTP	Secure Real-time Transport Protocol
SSC	Shared Services Canada
SIP	Session Initiation Protocol
ST	Straight Tip (optical fiber connectors)
T1	Transmission System 1
TDM	Time Division Multiplex
TLS	Transport Layer Security
TSG	See CNSSI
UCR	Unified Capabilities Requirements
UTP	Unshielded Twisted Pair
VoIP	Voice over Internet Protocol
WCS	Workplace Communication Service

1. Introduction

CFB Petawawa - CSOR complex (Mattawa Plains, Petawawa, ON K0J 1J0) is a new Department of National Defence facility including several buildings required to meet operational requirements of the base and unit. Expansion to existing Avaya Aura telephony solution is needed to meet the new occupant's requirements of not more than 800 (eight hundred) users with total of 800 (eight hundred) telephony devices including offices, faxes, classrooms, conference/meeting rooms, workshops and staff rooms.

Existing Avaya Aura is front-ended by the base CS1000 PBX providing all PSTN and CSN dialing via Q.Sig PRIs.

Existing Avaya Aura system is identified as
SO 52697
LAC 19FZ049965F74AE46269E800,
LAC 19FZ954665F71793C38C6CFB
CMM 19KZ646786DB3554D6E7B73A

Existing Avaya CS1000E MG is a R5.0 DSN system.

CSOR complex buildings (nine main buildings with a couple outbuildings) are currently unoccupied due to the construction. Occupants are schedule with full occupancy scheduled for **May 2021.**
Main building designations: CS111, CS114, CS117, CS121, CS128, CS133, CS134, CS135, and CS136

Base MTR, where Avaya Aura core is currently installed, is available for installation and configuration. Current BOM is found in Appendix A

This site and buildings are an EMSEC and SECURE zones. All measures must be taken to deny unauthorized persons or entities information of value that might be derived from intercept and analysis of compromising emanations from communication and information systems.

This document details the work to be performed by the contractor, for Shared Services Canada (SSC) and Department of Defence (DND). This Statement of Work (SOW) is based on the best information provide to the SSC Desk officer from various sources and are described herein.

2. Project Objectives

Overall objective is to expand the current Avaya Aura telephony system at CFB Petawawa at building H101 to support CSOR complex of buildings, and implement telephony services at the CSOR complex.

To install, configure and provision Avaya Aura core system EXPANSION at H101 to support the increased number of users providing additional resiliency and redundancy as well improved operation such as moving voicemail service from CMM to IX Messaging.
To install, configure and provision media gateways & telephones supporting users of CSOR complex of buildings.
To train the trainers on telephone feature usage including voicemail and auto attendants.
To provide maintenance and support for one year.
All work to be coordinated with the SSC Desk officer and the Site POC.

3. Project Scope

The contractor will design, provide, configure and implement a contractor provided telephony/unified communications system solution expanding the current Avaya Aura system to support CSOR complex and users. On site contractor presence will be required at CFB Petawawa, Petawawa ON.
The solution and project's work as related to the contractor is defined as per the following:

3.1 Provide, install, and configure the following:

1. Expand the current Avaya Aura hybrid VoIP - TDM Telephony system in that new components must support the following:
    a. Must be 19" rack mountable,
        i. Contractor provide racking requirements including posts required, space, and clearances,
    b. Hardware and software must be listed on DISA APL and be JITC certified,
        i. Contractor must supply APL Memo and IO Certification memo;
        ii. Must state level of compliance to DISA UCR 2013 CHANGE 2;
        iii. Must support MLPP,
        iv. Contractor must supply MUDG;
        v. Contractor must supply all required documentation;
            1. At a minimum Installation manual(s),
            2. At a minimum Administration manual(s),
            3. At a minimum Maintenance manual(s),
            4. As built diagrams and documentation;
            5. If available, relevant Application Notes;
    c. Must have dual hot swappable power supplies supporting NEMA 5-15 grounded (Type B) receptacles; exception UPS which will have one 5-15P input plug,
    d. Contractor must provide power requirements as watts and BTUs for MTR housed components,
    e. Local storage of all system software and configuration files,
        i. Single backup media of all system software and configuration files provided to site POC,
2. Telephony solution must continue to support the following functions locally:
    a. Support DTN dial plan as per User Profile provided at award,
    b. In-building and On-base Extension dialing (4 digits),

   c. Uniform dial plan (UDP),
   d. Coordinated dial plan (CDP),
   e. E.164 dialing (15 digit dialing),
   f. Voicemail,
      i. Provide to the specifications defined in Appendix E: Voicemail Requirements;
   g. Auto attendant with dial by name capability,
      i. Provide to the specifications defined in Appendix F: Auto Attendant Requirements
   h. Local emergency call notification,
      i. Must provide notification to include minimum of emergency caller name, extension and time;
   i. Must support set paging,
   j. Must have a system/solution configuration management software defined in Appendix G,
   k. Must have the ability to manage through remote control operations by national configuration managers;
3. Telephony solution must continue to provide the following capabilities:
   a. Must be compatible with DND deployed Avaya CS1000 Release 5.0 (DSN),
      i. CS1000 Release 5.0 (DSN) software as defined in Appendix O,
      ii. Must support communication with Avaya CS1000 Release 5.0 (DSN) systems via PRI (T1 and E1) using QSIG;
      iii. QSig as specified in Appendix P,
   b. Must be compatible with WCS based on Avaya Aura release 7.1 (JITC),
      i. Must support communication with Avaya Aura systems via IP using Session Initiation Protocol (SIP);
      ii. As specified in Appendix Q;
   c. All VoIP media and signalling traffic must continue to be encrypted,
      i. Must support SRTP as per IETF;
      ii. Must support TLS as per IETF;
      iii. Must support minimum encryption of AES256 for VoIP media,
      iv. Must support minimum encryption of AES256 for VoIP signalling,
   d. Must support group set paging;
   e. Must continue to interoperate with public address systems;
      i. Public address systems may connect to PBX via FXS port, or SIP telephone/trunk;
      ii. Common DND public address systems are:
         1. Bogen systems;
         2. TAO systems;
   f. Must continue to provide support for SCIP telephone devices;
      i. Must support Sectera VIPER telephones in SIP mode;
         1. Contractor must describe compliance and functionality;
      ii. Must support Sectera VIPER telephones in PSTN mode;
         1. Contractor must describe compliance and functionality;
4. Must support telephony features/functions and user class of services as per Appendix D: DND User Class Of Service / Profiles;

   a. Telephone templates must reflect Appendix D: DND User Class Of Service / Profiles and Appendix L: Sample Phone Button layout;
   b. Contractor must state compliance and functionality;
5. Must support network class of services as per Appendix C: DND Network Class Of Service;
   a. Contractor must state compliance and functionality;
6. Telephony solution must increment the current system capacities by ADDITIONAL support of:
   a. Minimum two (2) NEW T1 PRI ports as tie trunks to base PBX (MM710B)
   b. Minimum eleven (11) 24-port FXS cards (MM716) connecting to analog phones, faxes and public address system,
   c. Minimum eight hundred (800) user licensing (Core Suite),
   d. Minimum four fifty hundred (450) voicemail boxes,
7. Telephony system must provide the following ADDITIONAL devices:
   a. Must provide CNSSI Class A VoIP telephones in compliance to specifications defined in Appendix J2: CNSSI Class A VoIP Standard Telephone Requirements;
      i. Provide minimum quantity of forty (40) CNSSI Class A VoIP Telephones;
      ii. Install no more than thirty five (35) CNSSI Class A VoIP Telephones;
   b. Must provide VoIP telephones in compliance to specifications defined in Appendix J2: CNSSI Class B VoIP Standard Telephone Requirements;
      i. Provide minimum quantity of six hundred sixty (660) CNSSI Class B VoIP Standard Telephone Requirements;
      ii. Install no more than six hundred ten (610) CNSSI Class B VoIP Standard Telephone Requirements;
   c. Must provide analog conference telephones in compliance to specifications defined in Appendix N: Analog Conference Telephone Requirements
      i. Provide minimum quantity of fifteen (15) Analog Conference Telephones with ten (10) extended microphones;
      ii. Install no more than twelve (12) VoIP Conference Telephones;
   d. Must install telephone wallmounts as per table below defining quantities,
      i. Installation into drywall only;

Table: New Comms Closet Equipment Suggested

Building	G450	Eaton UPS (5PX 1000 – UPS + 5PX 48V)	Media Modules	Special Assemblies
CS111	1	1	Q1 - MM716	Fax Analog phones
CS114	1	1	Q1 - MM716 Q2- MM710B	Fax Analog phones PA Gate phones OSN/Crisis Alert Radio Call recording (Eventide)

CS117	1	1	Q1 - MM716	Fax Analog phones
CS121	1	1	Q1 - MM716	Fax Analog phones
CS128	2	2	Q1- S8300E Q2 - MM716	LSP Fax Analog phones
CS133	1	1	Q1 - MM716	Fax Analog phones
CS134	1	1	Q1 - MM716	Fax Analog phones
CS135	1	1	Q1 - MM716	Fax Analog phones
CS136	1	1	Q1 - MM716	Fax Analog phones
OverFlowGuardHouse	0	0		
H101			Q2- MM710B	Q.sig tie trunks to CS1000
H101			Q1- AVP	Resilient SM, CM, etc
H101			Q1 - AVP	IX Messaging (vmail & AA)

Table: Additional Telephone Quantities Required

	Supplied (minimum)	**Installed (maximum)**	**Wallmounts / Installed**
Analog telephones	0	6	0/0
CNSSI Class A VoIP Telephones	40	30	4 / 2
CNSSI Class B VoIP Standard Telephone	660	610	20 / 16
Analog Conference Telephones	15	12	0

8. Must provide cabling for telephone and equipment termination:
    a. A minimum quantity of two (2) STP RJ45/48 PRI cables of fifteen (15) metres length;
    b. A minimum quantity of twelve (12) cables of four (4) metres length terminating MM716 cards' amphenol ports to RJ11 rack;
    c. A minimum quantity of seven hundred (700) UTP Cat6 RJ45 of two (2) metre length;

   d. A minimum quantity of fifteen (15) UTP Cat3 or better RJ11 of four (4) metre length;

9. All Contractor or Manufacturer supplied security certificates must be valid for minimum three (3) years from date of equipment receipt;
10. Installation and configuration of provided telephony solution as per this Statement of Work;
11. Provide, install and configure twelve (12) UPS (Eaton 5PX 1000 – UPS plus a single 5PX 48V) and 2 post installation kit; one for each CSOR building with exception CS128 will have two installed,
12. Provide On Site training for USERS on phones, features and voice mail for key personnel (train the trainers) (no more than five (5) persons with duration not less than two (2) hours) with supporting material left and re-useable by the trainers,
13. Provide On Site training for SYSTEM ADMINISTRATORS covering adds, moves and changes of people and phones, system troubleshooting, and system auditing for key personnel (no more than five (5) personnel with duration not less than four (4) hours) with supporting material left and re-useable by the DND & SSC, and
14. Provide one (1) year ONSITE maintenance contract as defined in Appendix T.

3.2 Provisioning the solution to:

The VoIP Telephony solution will be provisioned to deliver:

1. Data network (IP) connectivity conforming with the data networking/IP addressing plan as provided by SSC/DND,
2. Telephony IP networks including media, signalling and management networks will be separate VLANs that are segregated and isolated from all other networks (absolutely no external IP network (INTERNET) connectivity),
3. Telephony call routing conforming with the DTN national dialing plan and policies as provided by DND/SSC,
4. End users, devices programming, and voicemail programming into the system(s) according to 'User Profile Worksheet',
5. Up to seven (7) Bilingual Auto Attendant / IVR trees to two (2) levels depth with voice recordings from SSC/DND personnel,
6. Ensure emergency calls are routed according to base policy,
7. Enable on-site notification of emergency calls as directed by SSC/DND.

3.3 Pre-staging tasks

Contractor can pre-stage the system to minimize on site time including

1. Review with the SSC rep the "User Profile" document, the equipment room layout and the site internal cabling,
2. As required, participate in all project meetings/conference calls with the SSC AOR and/or site POC,
3. Verify with SITE POC before shipping equipment;

Base MTR equipment will be sent to

CFB Petawawa - H101

179 Menin Rd
Petawawa, Ontario
K8H2X3
ensure any shipping label contains 'Building H101' and building POC

Remaining equipment will be sent to
CFB Petawawa – CSOR telephony
Mattawa Plains
Petawawa, Ontario
K8H2X3
ensure any shipping label contains 'Mattawa Plains' and building POC.

3.4 Onsite tasks

1. Meet with the POC and validate the telephony solution requirements,
2. Review with the SSC rep the "User Profile" document, the equipment room layout and the site internal cabling,
3. As required, participate in all project meetings/conference calls with the SSC AOR and/or site POC,
4. Work with the SSC/DND and other contractor teams (as designated by SSC/DND), Inventory the equipment prior to starting the installation,
5. Verify the rack power provide for the system install, ensure grounding as per manufacturer specifications,
6. Installation of related telephony hardware and cabling ,
7. Connect the system to the building wiring,
    a. To establish VoIP Telephony system connection with base PBX T1-PRI, (see Appendix A),
8. The installation, and implementation of Telephony system to include (but not limited to) the following,
    a. Digit dialing for building 346 telephony users and devices:
    b. Four (4) digits in-building and on-base dialing,
    c. PSTN access via access code 89 which is passed entirely to base PBX,
    d. CSN access via access code 86 which is passed entirely to base PBX,
9. Program users & telephones with associated voice mailboxes as per User Profile document provided at contract award,
10. Program and configure analog FXS lines for faxes,
11. Program and configure conference sets and additional microphones,
12. Program and configure one (1) analog FXS line for public announcement system (SSC provided),
13. Program and configure up to 2 (two) VoIP emergency calls On Site Notification telephone as directed by site POC,
14. Install and configure all telephones and/or lines,
    a. Place set in their respective location/offices,
15. Ensure 911 call routing is functioning correctly with local PSAP and building onsite notification;

16. Provide and Complete acceptance test plan of the solution with site POC and SSC/DND representatives:
    a. Sample test plan provided in Appendix R;
17. Participate in customer testing of all equipment to ensure end to end connectivity and expected feature operation,
18. Complete USER training session, and SYSTEM ADMINISTRATOR training session within two (2) weeks before or after of building Go Live date.

4. Other consideration

- Contractor will ensure that power and earth ground meet the manufacturer requirements for telephony hardware installation,
- Contractor will provide SSC/DND Desk officer with a copy of as-builds (to include (but limited to) site drawing/pictures, copy of the all programing, copy of all passwords),
- Implementation timeline as mutually agreed to with the SSC/DND Desk Officer and the Site POC.
- Upon request the contractor request will be provide a copy of pertinent site drawings,
- All Personal **MUST** hold a valid government of Canada **Enhanced Reliability** security clearance (may need to provide their clearance number each time they enter the base),
- All work will be done within normal business hours.

5. Site POC responsibilities

- The Site POC will provide the appropriate mounting equipment. For this rack-mount installation, site POC will provide 19” four post rack within MTR and each IDF,
- Provide a prime contact for all implementation issues,
- Provide a site contact list for the project and ensure availability of required resources for the duration of the project,
- Allow appropriate access to the Equipment / Office room(s) for performing the installation,
- Installation of necessary power distribution boxes, conduits, groundings, lightning protection, connectors, cables and associated hardware,
- Provide grounded UPS power within 1.5m of equipment to be installed,
- If applicable, provide laptop / PC, IP address, and/or LAN connection for the administration terminal,
- When applicable, provide floor plans and identify location of telephones, extension info, faxes, POS, etc.,
- The Building has been prewired; therefore SSC/DND will be responsible for the internal building wiring,
- Provide Power and Cooling,

- Site POC will supply earth ground in proximity of equipment installation which must be connected to the building structure's main ground,
- Providing extension numbering schema,
- Providing script for auto attendant greetings,
- Providing menu for auto attendant application,
- Provide voice talent or recordings (MP3 or WAV format) for auto attendant greetings,
- Providing the IP addressing methodology currently employed (if applicable),
- Provide logically segregated and isolated Ethernet VLANs for VoIP telephony, networks (Ethernet connectivity as per established by DND policy),
- Provide POE equipment sufficient to supply VoIP telephones power,
- POC and team must be available during implementation and training activities, and
- Provide all Ethernet and fibre patch cables not included in the telephony solution equipment configurations.

6. Acceptance Criteria and Sign-off

- Contractor to provide Site POC and the SSC/DND Desk officer with an Acceptance & Sign-off document to review and agree upon in advance of installation.

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Appendix A1: CFB Petawawa – Avaya Aura System current BOM

SO 52697
LAC 19FZ049965F74AE46269E800,
LAC 19FZ954665F71793C38C6CFB
CMM 19KZ646786DB3554D6E7B73A

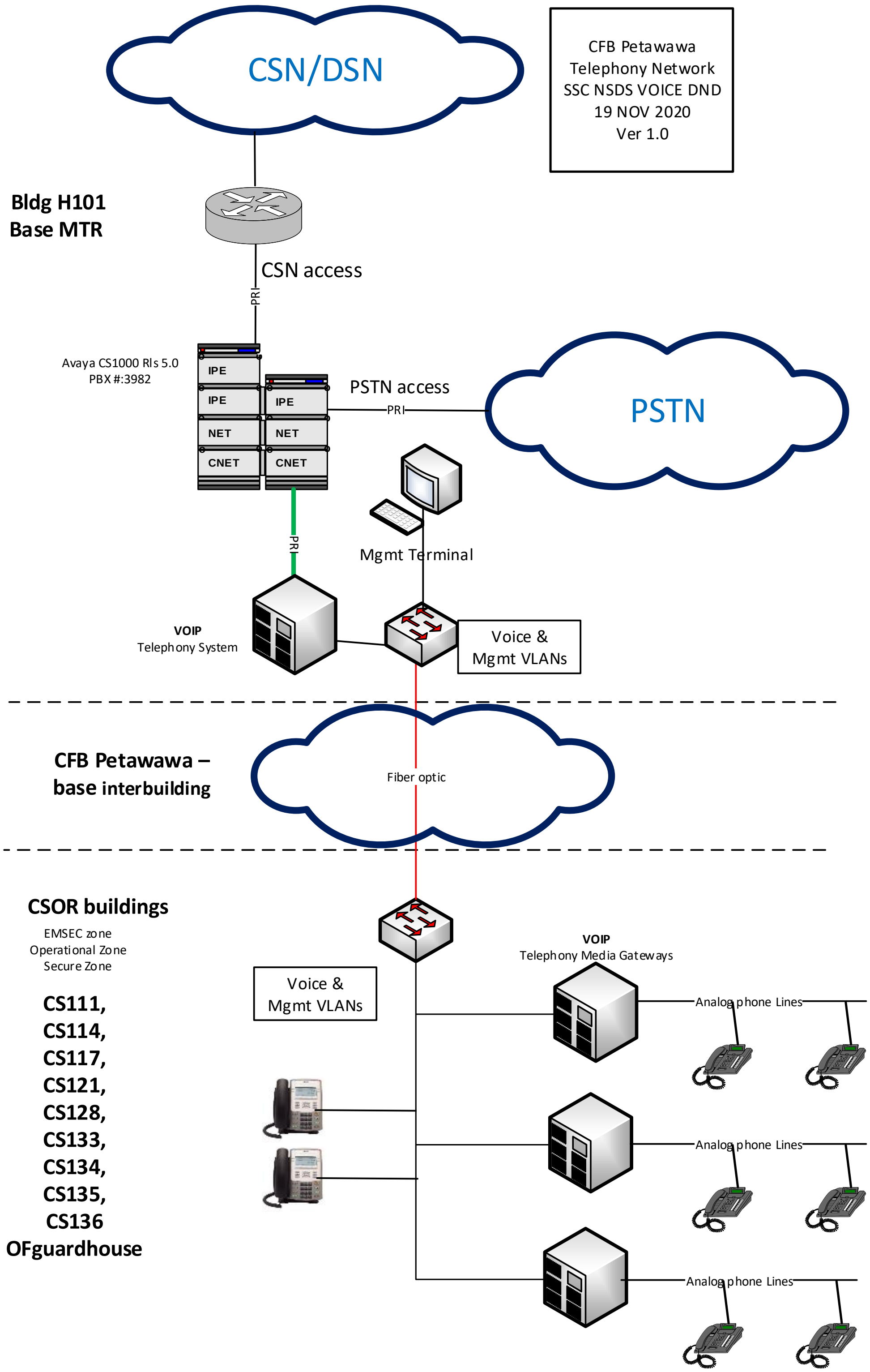
Comment	ACA1003020	1.00
259401	AVAYA RED : MEDIA ENCRYPTION R6+/MBT	1.00
380002	AVAYA RED : SM R7 VE VAPP LIC	1.00
380042	AVAYA RED : AURA R7 LARGE ENT SMPLX SOL TRK	1.00
380228	AVAYA RED : AURA SMGR R7 VE VAPP LICENSE	1.00
380349	AVAYA RED : AURA R7 CM VE VAPP SYS LIC	1.00
380361	AVAYA RED : AURA R7 UTILITY SVCS VE VAPP SYS LIC	1.00
381275	AVAYA RED : AVP R7+ SNGL CPU CMN LIC	1.00
391427	AVAYA RED : AURA UTILITY SVCS R7.1 AVP SYS LIC	1.00
396794	AVAYA RED : AURA R8 CORE NEW LIC	400.00
397108	AVAYA RED : AURA R8 ANALOG NEW/ADD LIC	26.00
397113	AVAYA RED : AURA R8 PRESENCE SERVICES R8 /E	400.00
397114	AVAYA RED : AURA R8 AES UNIFIED DESKTOP R8 /E	400.00
397115	AVAYA RED : AURA R8 ASBCE R8 STD /E	116.00
397116	AVAYA RED : AURA R8 ASBCE R8 ADV /E	58.00
397117	AVAYA RED : AURA R8 ASBCE R8 STD HA /E	116.00
397118	AVAYA RED : AURA R8 ASBCE R8 ADV HA /E	58.00
397120	AVAYA RED : AURA R8 AMM ENH USR /E	400.00
397124	AVAYA RED : AURA R8 EQUINOX IPAD /E	400.00

397125	AVAYA RED : AURA R8 EQUINOX WIN /E	400.00
397126	AVAYA RED : AURA R8 EQUINOX MOBILE /E	400.00
397127	AVAYA RED : AURA R8 EC500 SM /E	400.00
397128	AVAYA RED : AURA R8 ONE-X CES /E	400.00
397129	AVAYA RED : AURA R8 VIDEO /E	400.00
397130	AVAYA RED : AURA R8 EQUINOX MAC /E	400.00
397131	AVAYA RED : AURA R8 ONE-X COMM /E	400.00
397132	AVAYA RED : AURA R8 COMM FOR MS LYNC /E	400.00
397133	AVAYA RED : AURA R8 EQUINOX FOR WEB /E	400.00
397134	AVAYA RED : AURA R8 INTG MGMT ADMIN R6 /E	1.00
397135	AVAYA RED : AURA R8 BREEZE R3 USER /E	400.00
397184	AVAYA RED : AURA R8 MSG SEAT BASIC R7 /E	400.00
397287	AVAYA RED : AURA SMGR R8 LIC	1.00
397941	AVAYA RED : SM R8 SYSTEM LIC	1.00
397945	AVAYA RED : SM BRANCH (BSM) R8 SYSTEM LIC	1.00
405362641	AVAYA RED : PWR CORD USA	2.00
700510424	AVAYA RED : AV APP VRTL PLTFRM R7 MEDIA KIT	1.00
700514194	AVAYA RED : ACP 120 DELL SERVER PROFILE 4 WITH AVP BUNDLE	1.00
405362641	AVAYA RED : PWR CORD USA	2.00

700466634	AVAYA RED : MM710B E1/T1 MEDIA MODULE - NON GSA	3.00
700506956	AVAYA RED : G450 MP160 MEDIA GATEWAY NON GSA	1.00
700507394	AVAYA RED : G450 R2 POWER SUPPLY	1.00
700395445	AVAYA RED : 120A CSU CABLE 50FT RHS	3.00
293630J	AVAYA RED : SA PREF C/D AAVP R7 SNGL CPU CMN 1YPP	1.00
340525J	AVAYA RED : SA PREF C/D SM R7 VE VAPP LIC 1YPP	1.00
344221J	AVAYA RED : SA PREF ADP AURA R8 ANALOG 1YPP	26.00
344277J	AVAYA RED : SA PREF ADP AURA R8 CORE 1YPP	400.00
344365	AVAYA RED : SA AURA SW SUPT R7 BUY TO CURRENT	1.00
344368	AVAYA RED : SA CALL CENTER SW SUPT R7 BUY TO CURRENT	1.00
344369	AVAYA RED : SA SESSION MGR SUPT R7 BUY TO CURRENT	1.00
344587J	AVAYA RED : SA PREF ADP SM R8 SYSTEM 1YPP	1.00
344625J	AVAYA RED : SA PREF ADP SYS MGR R8 LIC 1YPP	1.00
Comment		1.00
Comment	ACA1005692	1.00
Comment		1.00
380349	AVAYA RED : AURA R7 CM VE VAPP SYS LIC	1.00
380361	AVAYA RED : AURA R7 UTILITY SVCS VE VAPP SYS LIC	1.00
381276	AVAYA RED : AVP R7+ SNGL CPU EMBD LIC	1.00
381547	AVAYA RED : SM BRANCH (BSM) R7 VE VAPP LIC	1.00
391427	AVAYA RED : AURA UTILITY SVCS R7.1 AVP SYS LIC	1.00
700510424	AVAYA RED : AV APP VRTL PLTFRM R7 MEDIA KIT	1.00
193806	AVAYA RED : UTILITY TRIGGER REM GATEWAY NEW SITE	1.00
272731	AVAYA RED : APS NTWK READINESS ASSESSMENT-VENDOR PRV	1.00
405362641	AVAYA RED : PWR CORD USA	2.00
700406267	AVAYA RED : S8300/S8400 CD/DVD ROM DRIVE RHS	1.00
700466626	AVAYA RED : MM711 ANLG MEDIA MODULE - NON GSA	2.00
700466642	AVAYA RED : MM716 ANLG MEDIA MOD 24FXS - NON GSA	1.00

700506956	AVAYA RED : G450 MP160 MEDIA GATEWAY NON GSA	1.00
700507394	AVAYA RED : G450 R2 POWER SUPPLY	1.00
700508955	AVAYA RED : S8300E SERVER - NON GSA	1.00
700383326	AVAYA RED : 96XX RPLCMNT LINE CORD	384.00
700501539	AVAYA RED : AVAYA B100 SER EXP MIC 1PR	4.00
700504740	AVAYA RED : AVAYA B179 SIP CONF PHONE POE ONLY	4.00
700507946	IP TELEPHONE 9608G Gray GIGABIT ETHERNET (TAA)	384.00
293648J	AVAYA RED : SA PREF C/D AAVP R7 SNGL CPU EMBD 1YPP	1.00
340531J	AVAYA RED : SA PREF C/D SM BR (BSM) R7 VE VAPP 1YPP	1.00
344345J	AVAYA RED : SA PREF ADP REMOTE SITE TRKG AURA R8	1.00
700383375	AVAYA RED : 9620/08/11 AND 94/9500 WALL MOUNT	44.00
700514051	AVAYA RED : AV L119 HEADSET LEATHER RJ9 MONO	40.00
700514054	AVAYA RED : AV L149 HEADSET LEATHER QD STEREO	20.00
700514324	AVAYA RED : AV QD RJ9 HDST CORD 1.2M STRA	20.00
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
5PX1000RTUPS	5PX RACK/TOWER UPS. 1000 VA/	2.00
5PXEBM48RT	EATON INDUSTRIES : 5PX 48V EBM R/T 2U 1000-2200VA MODELS	4.00
103007018-5591	EATON 2 POST RM RAIL KIT FOR 130, 9130	2.00
A1265-0000-10-05	AASTRA 9116LP Analog Phone	20.00

Appendix A2: CFB Petawawa – Telephony Diagram



Appendix B: DND Dial Plan

Appendix B1: CSN/DSN and PSTN Dialing

Dialing	Actions	Digits
Emergency Services	Off hook and dial	911, 86-911, and 89-911
Extension within base PBX	Off hook and dial	XXXX 4 to 7 digits
CSN within same NPA	Off hook and dial	86-NXX – XXXX ACOD + 7 digits
CSN within same NPA	Off hook and dial	NXX – XXXX 7 digits
CSN/DSN outside caller's NPA	Off hook and dial	86-NPA – NXX – XXXX 10 digits
CSN/DSN outside caller's NPA	Off hook and dial	NPA – NXX – XXXX 10 digits
Local Public	Off hook and dial	89 - NPA – NXX - XXXX
Local Public	Off hook and press Public Dialing button	NPA – NXX - XXXX
Long distance Public NA	Off hook and dial	89 – 1 -NPA – NXX - XXXX
Long distance Public NA	Off hook and press Public Dialing button	1 -NPA – NXX - XXXX
Long distance Public INTL	Off hook and dial	89 – 011 – CC- NPA –NXX - XXXX
Long distance Public INTL	Off hook and press Public Dialing button	011 –CC - NPA – NXX - XXXX

Appendix B2: MLPP Dialing

Dialing	Actions	Digits
MLPP to CSN with same NPA	Off hook	86-*8X - NXX – XXXX
MLPP to CSN with same NPA	Off hook	*8X - NXX – XXXX
MLPP to CSN with same NPA	Off hook and press MLPP button	86-NXX – XXXX

MLPP to CSN with same NPA	Off hook and press MLPP button	NXX – XXXX
MLPP to CSN/DSN outside caller’s NPA	Off hook	86-*8X – NPA- NXX – XXXX
MLPP to CSN/DSN outside caller’s NPA	Off hook	*8X – NPA- NXX – XXXX
MLPP to CSN/DSN outside caller’s NPA	Off hook and press MLPP button	86-NPA - NXX – XXXX
MLPP to CSN/DSN outside caller’s NPA	Off hook and press MLPP button	NPA - NXX – XXXX

Appendix C: DND Network Class of Service

Not all classes applicable to this specific location. User Profile defines used NCOS.

Access	Permitted Calling Profile Level												
	0	1	2	3	4	5	6	7	8	9	10	11	12
INTERNAL, 911	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Local, Toll Free, 711 (TDY)		✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Tie Trunks (as applicable)			✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
CSN, DSN Zone 1 (312, 315, 317, 319 and other DSN area codes as specified by Canada), EAS				✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Canada and USA Toll, 411, 555, 700-555-4141 +611(re-route)					✓	✓	✓	✓	✓	✓	✓	✓	✓
Operator Calls (0, 0 Plus, 01, 011), NCN (formerly IVSN), 600						✓	✓	✓	✓	✓	✓	✓	✓
DSN Zone 4, Australia (715), Iridium (707/717) and other DSN area codes as specified by Canada							✓	✓	✓	✓	✓	✓	✓
Unrestricted NTAS (310)								✓	✓	✓	✓	✓	✓
MLPP Routine On Net CSN/DSN				✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
MLPP Priority On Net CSN/DSN									✓	✓	✓	✓	✓
MLPP Intermediate On Net CSN/DSN										✓	✓	✓	✓
MLPP Flash On Net CSN/DSN											✓	✓	✓
MLPP Flash Override On Net CSN/DSN												✓	✓

Appendix D: DND User Class of Service / Profiles

Not all classes applicable to this specific location. User Profile defines used NCOS.

DND/CF National User Profiles		User Profile													
Functions	Access Code (as applicable)	Public Phone	Conf Room	Basic User	Help Desk	Adv User Routine	Adv User Priority	Adv User Immediate	Adv User Flash	Adv User Flash Override	Exec User Routine	Exec User Priority	Exec User Immediate	Exec User Flash	Exec User Flash Override
Call Hold			x	x	x	x	x	x	x	x	x	x	x	x	x
911 Emergency		x	x	x	x	x	x	x	x	x	x	x	x	x	x
Music on Hold			x	x	x	x	x	x	x	x	x	x	x	x	x
Call Waiting				x	x	x	x	x	x	x	x	x	x	x	x
Voicemail				x	x	x	x	x	x	x	x	x	x	x	x
Speed Dialing			x	x	x	x	x	x	x	x	x	x	x	x	x
Last Number Redial			x	x	x	x	x	x	x	x	x	x	x	x	x
Call Forward All				x	x	x	x	x	x	x	x	x	x	x	x
Call Forward No Answer				x	x	x	x	x	x	x	x	x	x	x	x
Call Forward Busy				x	x	x	x	x	x	x	x	x	x	x	x
Call Forward Unregistered				x	x	x	x	x	x	x	x	x	x	x	x
Enterprise Directory Access			x	x	x	x	x	x	x	x	x	x	x	x	x
Dial-In Directory Access (PSTN)				x	x	x	x	x	x	x	x	x	x	x	x
Call Logs				x	x	x	x	x	x	x	x	x	x	x	x
Mute		x	x	x	x	x	x	x	x	x	x	x	x	x	x
Call Reject			x	x	x	x	x	x	x	x	x	x	x	x	x
Caller ID			x	x	x	x	x	x	x	x	x	x	x	x	x
Click to Dial (GAL)				x	x	x	x	x	x	x	x	x	x	x	x
Do Not Disturb			x	x	x	x	x	x	x	x	x	x	x	x	x
Vertical service code			x	x	x	x	x	x	x	x	x	x	x	x	x
Unattended Call Transfer				x	x	x	x	x	x	x	x	x	x	x	x
Attended Call Transfer				x	x	x	x	x	x	x	x	x	x	x	x
Ad-hoc Audio Conferencing			x	x	x	x	x	x	x	x	x	x	x	x	x
Blocking Caller ID				x		x	x	x	x	x	x	x	x	x	x
Ring Again			x	x	x	x	x	x	x	x	x	x	x	x	x
Hands-free capability			x	x	x	x	x	x	x	x	x	x	x	x	x
Message Waiting Indicator				x	x	x	x	x	x	x	x	x	x	x	x
Presence				x	x	x	x	x	x	x	x	x	x	x	x
Call Pick-up Group					x	x	x	x	x	x					
Meet-Me Audio Conferencing											x	x	x	x	x
Multiple Directory Number Appearances											x	x	x	x	x
Shared Line Features											x	x	x	x	x
Preset Audio Conference											x	x	x	x	x
ACD					x										
Extension Mobility					x	x	x	x	x	x	x	x	x	x	x
Local (PSTN)	89	x	x	x	x	x	x	x	x	x	x	x	x	x	x
Long Distance (PWGSC)	89		x	x	x	x	x	x	x	x	x	x	x	x	x
International (PWGSC)	89				x	x	x	x	x	x	x	x	x	x	x
CSN/DSN	86		x	x	x	x	x	x	x	x	x	x	x	x	x
Local Site (Building/Base)		x	x	x	x	x	x	x	x	x	x	x	x	x	x
IVSN	81				x	x	x	x	x	x	x	x	x	x	x
MITNET	83				x	x	x	x	x	x	x	x	x	x	x
DOG (CSN)	86				x	x	x	x	x	x	x	x	x	x	x
MLPP Routine		x	x	x	x	x					x				
MLPP Priority	82						x					x			
MLPP Immediate	85							x					x		
MLPP Flash	88								x					x	
MLPP Flash Override	80									x					x
Default Calling Profile Level (NCOS)		1	5	5	12	7	8	9	10	11	7	8	9	10	11

Appendix E: Voicemail Requirements

Following section defines the requirements of the telephony solution as applicable to voicemail services.

1. Each designated phone, user or position will have a voicemail box,
    a. Each individual voicemail box will store a minimum 20 (twenty) minutes of voice,
    b. Each individual voicemail box will support minimum 2 (two) recorded greetings (busy and away) of one minute minimum for each greeting,
2. Must provide message waiting indication to the connected PBX and telephones,
    a. May provide message waiting indication to networked PBXs,
3. Must provide a multilevel administration interface as defined by DND/SSC at a minimum as follows;
    a. Voicemail system configuration (network & PBX connectivity, admin setup);
    b. Voicemail administration for creation and deletion of voicemail boxes;
    c. Voicemail password resets (self-administration option welcome);

Appendix F: Auto Attendant Requirements

Following section defines the requirements of the telephony solution as applicable to an auto attendant.

1. Support a minimum of two independent auto attendant trees,
2. Provide a minimum features of
    a. Call List
    b. Dial By Name
    c. Disconnect
    d. Replay Greetings / Menu
    e. Transfer to User / Group
    f. Transfer to Operator
3. Must provide a multilevel administration interface as defined by DND/SSC at a minimum as follows;
    a. AA system configuration (PBX connectivity, admin setup);
    b. AA administration for creation and deletion of menu trees;
    c. AA administration for recorded announcements and tree activation;

Appendix G: PBX Management Platform

a) Must be listed on DISA APL and JITC certified;
   - Contractor must supply APL Memo and IO Certification memo;
   - Contractor must supply any MUDG;
   - Contractor must supply all required documentation;
     - At a minimum Installation manual(s),
     - At a minimum Administration manual(s),
     - At a minimum Maintenance manual(s),
     - If available, Application Notes;

b) Must provide ability to facilitate graphically following functionality:
   - Configuration management,
   - Fault management,
   - Performance management,
   - Change management,
   - Inventory management,
   - Security and Accounting management,
   - Contractor must describe compliance and functionality;

c) Must provide a multilevel administration interface as defined by DND/SSC at a minimum as follows;
   - System configuration (network & PBX connectivity, admin & user setup);
   - Administration for creation and deletion of PBX components, services and management users;
   - Administration of management abilities on a per PBX user class or individual PBX user basis;

d) Must provide a central management capability;
   - Must provide a visual view of the entire PBX system and components (Graphic User Interface -- GUI);
   - Must have capability to remotely manage in real-time deployed PBX systems,
   - Contractor must describe compliance and functionality;

e) Must provide a local management capability;
   - Must provide a visual view of the local PBX system(s) and components (Graphic User Interface - GUI);
   - Must operate when network isolated;
   - Contractor must describe compliance and functionality;

f) Should support English and French languages;
   - Contractor must describe compliance and functionality;

Appendix H: Analogue Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to analogue telephones.

1. Must be black or grey,
2. Must be wall mountable,
3. Must have minimum 2 line display with 16 characters each line,
4. Must have iconic labels or be bilingual (French & English) labelling,
5. Support dual-tone multi-frequency signaling (DTMF),
6. Must have ringer volume control,
7. Must have handset volume control,
8. Must have mute capabilities and Mute button,
9. Must have visual ring indication,
10. May have optional feature support such as redial, call history, message waiting indication, call timer, and date & time, and
11. Must have compliance with CLASS features.

Appendix I: Basic VoIP Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to VoIP telephones.

1. Must be black or grey,
2. Must be wall mountable,
3. Must have minimum 2 line display with 16 characters each line,
4. Must have iconic labels or be bilingual (French & English) labelling,
5. Support dual-tone multi-frequency signaling (DTMF),
6. Must support more than one line,
7. Must support Ethernet
8. Must support Power over Ethernet (POE),
9. Must support secure real time protocol (SRTP) with AES-256,
10. Must have ringer volume control,
11. Must have handset volume control,
12. Must have mute capabilities and Mute button,
13. Must have visual ring indication,
14. Must have visual message waiting indication,
15. Must have configurable (on/off) duplex speakerphone capabilities,
16. Must have capability to support standard wired headset,
17. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
18. Must be capable of supporting SIP without hardware modification.

Appendix J1: Standard VoIP Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to VoIP telephones.

1. Must be black or grey,
2. Must be wall mountable,
3. Must have a pixel graphical greyscale display,
4. Must have iconic labels or be bilingual (French & English) labelling,
5. Support dual-tone multi-frequency signaling (DTMF),
6. Must support minimum three (3) lines,
7. Must support Ethernet
8. Must support Power over Ethernet (POE),
9. Must support secure real time protocol (SRTP) with AES-256,
10. Must have ringer volume control,
11. Must have handset volume control,
12. Must have mute capabilities and Mute button,
13. Must have visual ring indication,
14. Must have visual message waiting indication,
15. Must have configurable (on/off) duplex speakerphone capabilities,
16. Must have capability to support standard wired headset,
17. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
18. Must be capable of supporting SIP without hardware modification.

Appendix J2: Standard CNSSI5006 Class A (TSG6) VoIP Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to VoIP telephones.

1. Must be compliant with CNSSI5006 Class A; see Avaya model 700514742
2. Must be black or grey,
3. Must be wall mountable,
4. Must have a pixel graphical greyscale display,
5. Must have iconic labels or be bilingual (French & English) labelling,
6. Support dual-tone multi-frequency signaling (DTMF),
7. Must support minimum three (3) lines,
8. Must support Ethernet
9. Must support Power over Ethernet (POE),
10. Must support secure real time protocol (SRTP) with AES-256,
11. Must have ringer volume control,
12. Must have handset volume control,

13. Must have mute capabilities and Mute button,
14. Must have visual ring indication,
15. Must have visual message waiting indication,
16. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
17. Must be capable of supporting SIP without hardware modification.

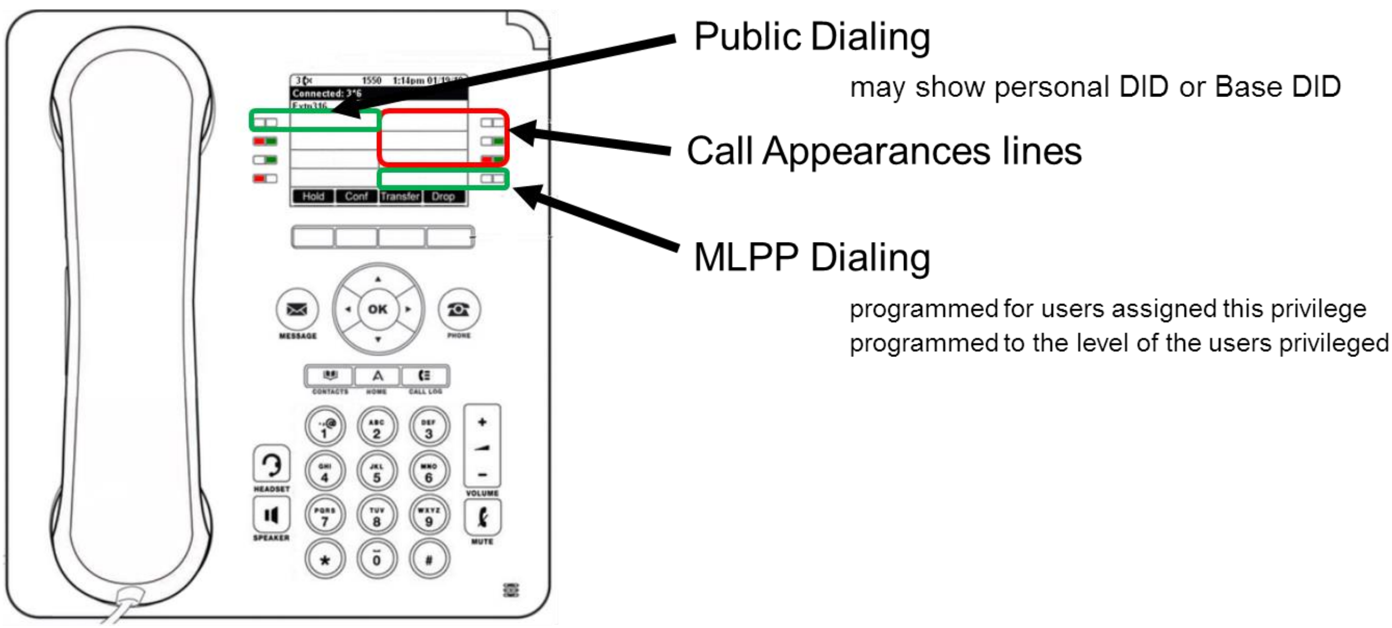
Appendix J2: Standard CNSSI5006 Class B (TSG6) VoIP Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to VoIP telephones.

1. Must be compliant with CNSSI5006 Class B; see Avaya model 700514745
2. Must be black or grey,
3. Must be wall mountable,
4. Must have a pixel graphical greyscale display,
5. Must have iconic labels or be bilingual (French & English) labelling,
6. Support dual-tone multi-frequency signaling (DTMF),
7. Must support minimum three (3) lines,
8. Must support Ethernet
9. Must support Power over Ethernet (POE),
10. Must support secure real time protocol (SRTP) with AES-256,
11. Must have ringer volume control,
12. Must have handset volume control,
13. Must have mute capabilities and Mute button,
14. Must have visual ring indication,
15. Must have visual message waiting indication,
16. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
17. Must be capable of supporting SIP without hardware modification.

Appendix L: Sample Phone Button Layout

Following is a sample template for Telephone buttons layout (from existing DND telephones). Refer to features for each User COS in Appendix D



Appendix M: VoIP Conference Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to VoIP conference telephones.

1. Must be black or grey,
2. Must have minimum 2 line display with 16 characters each line,
3. Must have iconic labels or be bilingual (French & English) labelling,
4. Support dual-tone multi-frequency signaling (DTMF),
5. Must support more than one line,
6. Must support Ethernet
7. Must support Power over Ethernet (POE),
8. Must support secure real time protocol (SRTP) with AES-256,
9. Must support voice codecs listed below;
    a. G.711A
    b. G.723.1
    c. G.726 (16,24,32,40 kb/s)
    d. G.729 A & B
10. Must have ringer volume control,
11. Must have volume control,
12. Must have mute capabilities, mute button, and mute visual indicator,
13. Must have visual ring indication,
14. Must have visual message waiting indication,
15. Must support optional extended microphones minimum of two (2),
16. May have optional feature support such as redial, call history, call timer, conference, transfer and date & time, and
17. Must be capable of supporting SIP without hardware modification.

Appendix N: Analog Conference Telephone Requirements

Following section defines the requirements of the telephony solution as applicable to analog station conference telephones.

1. Must be black or grey,
2. Must have minimum 2 line display with 16 characters each line,
3. Must have iconic labels or be bilingual (French & English) labelling,
4. Support dual-tone multi-frequency signaling (DTMF),
5. Must support minimum one line analog station line (FXS),
6. Must have ringer volume control,
7. Must have volume control,
8. Must have mute capabilities, mute button, and mute visual indicator,
9. Must have visual ring indication,
10. Must have visual message waiting indication,
11. Must support optional extended microphones minimum of two (2), and
12. May have optional feature support such as redial, call history, call timer, conference, transfer and date & time.

Appendix O: CFB Petawawa PBX Specifications

CFB Petawawa PBX is an Avaya/Nortel CS1000 PBX. The following configuration and specifications provided to ensure interoperability with existing telephony services. Contractor system must be interoperable at the highest level.

Avaya Communication Server 1000MG

System Parameters:
- Software Version : 3621
- System Type : Option 81C
- Call Processor : CP PIV
- Release : 5
- Issue : 00 W +

Appendix P: Avaya Communication Server 1000 Q.931 (QSig) Features

CS 1000 R5.0 DSN QSIG Features	Title	Identifier
	Basic Call	QSIG-BC
	Calling Line Identification Presentation	SS-CLIP
	Calling Line Identification Restriction	SS-CLIR
	Connected Line Identification Presentation	SS-COLP
	Connected Line Identification Restriction	SS-COLR
	Identification Supplementary Service	SS-ISSD
	Name Identification Supplementary Service	SS-NISD
	Calling Name Identification Presentation	SS-CNIP
	Connected Name Identification Presentation	SS-CONP
	Calling/Connected Name Restriction	SS-CNIR
	Generic Functional Protocol (transport)	QSIG-GF
	Call Diversion	SS-CFSD
	Call Forwarding Unconditional	SS-CFU
	Call Forwarding on Busy	SS-CFB
	Call Forwarding on No Reply	SS-CFNR
	Call Deflection	SS-CD
	Path Replacement	ANF-PR
	Call Completion to Busy Subscribers	SS-CCBS
	Call Completion on No Reply	SS-CCNR
	Message Waiting Indication	SS-MWI
	Transit Counter	n/a

Appendix Q: Workplace Communication Service (WCS) Specifications

Following section defines the specifications of Canada’s WCS solution. The telephony solution of building 346 will be required to connect to WCS.

1. Based on Avaya Aura release 7.1,
2. Includes key elements of Avaya Communication Manager, Session Manager, and G-series Media Gateways
3. Will support Session Initiation Protocol (SIP), and
4. All PSTN and CSN calls will be routed via WCS.

Appendix R: Sample Test Plan

Test Item #	Dial Test Plan
1.00	Telephony
1.01	Receive and make in-building extension calls
1.02	Receive and make base extension calls
1.03	Receive and make local PSTN calls
1.04	Receive and make long distance North American PSTN call
1.05	Make a long distance International PSTN call
1.06	Receive and make CSN calls
1.07	Receive and make DSN calls
1.08	Confirm Calling Line ID and Called Party Name Display are acceptable for all call scenarios
1.09	Receive and make fax calls
1.10	Test 9-1-1 calling; verify location/address
1.11	Verify voicemail
1.12	Verify auto attendant
1.13	Test public address system
1.14	Test paging; test set paging
1.15	Test SCIP (encrypted) devices
1.16	Test radio devices
1.17	Test administration tools/software
1.18	Test maintenance tools/software

Appendix S: UPS Equipment

To be supplied by Vendor

Requirement for UPS		
Eaton PN	**Description**	**Qty.**
5PX1000RT	Eaton 5PX 1000 120V 2U Rack Tower, 8ft 5-15P Input Cord, (8)5-15R Outlets	2
5PXEBM48RT	Eaton 5PX 48V EBM R/T 2U, 1000-2200va Models	4
103007018-5591	2-post rail kit	2

Appendix T: Maintenance Services

Maintenance Services

1) When requested by Canada in a Service Order for a hardware Product, the Contractor must provide On-Site Maintenance for the Product that requires the Contractor to arrive on site within the SLT-MTO with all components and tools required to restore the Product to Fully Functional Operation 24 hours / day and seven (7) days per week.
2) When requested by Canada in a Service Order for a software Product, the Contractor must provide Software Maintenance for the Product that requires the Contractor to maintain the Product in good working order and at an up-to-date revision level according to the Software Publisher's specifications that includes:
   a) software bug fixes;
   b) preventative maintenance software updates;
   c) version upgrades/updates; and
   d) feature set upgrades/updates/releases.
3) The Contractor must create an Incident Ticket for each Maintenance Event and assign a severity level and priority as specified by Canada when Canada notifies the Contractor of the Incident Ticket.
4) The Contractor must revise the severity level and priority of an Incident Ticket when requested to do so by Canada within 15 minutes of the request.
5) The Contractor must escalate Incident Tickets based on the Incident Ticket type, severity, impact, importance to Canada, and the time that an Event has remained open.
6) The Contractor must have an operational and management escalation matrix that defines the resources, with alternates (of equal authority) for each level of escalation for Incidents and related Incident Tickets, and clear contact instructions.
7) The Contractor must provide Canada with notification of Incidents according to the operational and management escalation matrices.
8) The Contractor must escalate Incidents as requested by Canada.
9) The Contractor must create at least 1 Incident Ticket for each Incident identified by the Contractor or reported by Canada.
10) The Contractor's Incident Tickets must include, at least the following dedicated information fields:
   a) ticket number;
   b) Event description;
   c) related Event tickets;
   d) date and time stamp when ticket initiated;
   e) date and time stamp when ticket closed;
   f) ticket type;
   g) ticket severity level;
   h) ticket priority;
   i) ticket status (i.e. open, closed, in progress, suspended, cancelled etc.);
   j) Canada's ticket number;
   k) affected SDPs;
   l) Contractor contact (name, telephone number and email address);
   m) Client identifier;
   n) Canada contact information (name, telephone number and email address);

o) activity log including all actions taken by third parties;
p) resolution description and cause;
q) outage time (for closed tickets only);
r) make, model and serial number of affected equipment;
s) service technician identifier; and
t) resolution description and cause.

11) The Contractor must include additional Incident Ticket information fields for Incidents as requested by Canada.
12) The Contractor must document all escalations and interactions with third parties for Incidents in the Incident Ticket information log.
13) The Contractor must open an Incident Ticket within 5 minutes for both Contractor-determined and Canada-reported Incidents.
14) The Contractor must update the Incident Ticket within 15 minutes of a change in status of the Incident as evidenced by the Incident Ticket timestamp.
15) The Contractor must update the Incident Ticket information log for an Incident within 5 minutes of a request by Canada.
16) The Contractor must track and report the outage time of each Incident in the associated Incident Tickets.
17) The outage time for an Incident must start at the time (start time) that the Incident is detected by the Contractor, or reported to the Contractor by Canada.
18) The outage time for an Incident must stop at the time that Canada has approved the closure of the associated Incident Tickets
19) The Contractor must request access to a SDP when such access is required for an Incident.
20) The Contractor must suspend outage time for an Incident at Canada's request or where the Contractor has requested:
    a) access to a SDP necessary to resolving an Incident and Canada is unable to provide access; or
    b) closure of an Incident Ticket pending Canada's approval, and Canada is not available to consider the request.
21) The Contractor must restart the outage time for an Incident where the outage time has been suspended, when requested by Canada or when:
    a) SDP access was required by the Contractor and Canada grants access to the SDP; or
    b) Canada is available to review the request to close an Incident and has determined that the Incident must remain open.
22) The Contractor must obtain Canada's approval before closing an Incident Ticket

Service Level Target - Maximum Time On-Site (SLT-MTO)

23) The SLT-MTO represents the maximum allowable time for the Contractor to arrive in site to perform On-Site Maintenance.
24) The SLT-MTO is 2.0 hours.
25) The SLT-MTO is calculated as the total elapsed time from when Canada or the Contractor identifies as an Incident that requires On-Site Maintenance, whichever is the earliest, and until the Contractor arrives at the site as the documented in the Incident ticket.

Service Level Target – Service Desk Response (SLT-SDR)

26) The SLT-SDR must meet or exceed 80.0% of all calls received by the Contractor Service Desk in a calendar month.
27) The SLT-SDR must be calculated as follows: (number of calls answered within two hundred and forty (240) seconds + number of calls abandoned within two hundred and forty (240) seconds)/ (total number of calls answered + total number of abandoned calls) * 100.
28) The calculation of the time to answer a call starts from the time the telephone call is connected to the Contractor Service Desk telephone system and ends when the Contractor Service Desk agent answers the call.
29) An abandoned call is a telephone call that is connected to the Contractor Service Desk telephone system and the Calling Party terminates the call before a Contractor Service Desk agent answers the call.