Statement of Work (SOW)
For the installation of Telephony Communications solution
At CFB Petawawa, Building P144, Petawawa, ON

04 January 2019

SSC IPO Technical Authority/Desk officers and Site POC:

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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
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
UCR	Unified Capabilities Requirements
UTP	Unshielded Twisted Pair
VoIP	Voice over Internet Protocol
WCS	Workplace Communication Service

1. Introduction

CFB Petawawa building P144 (250 Somme Street, Petawawa, ON K0J 1J0) is a new Department of National Defence building with a single internal zone: Operational in order for the facility to meet the health services requirements of the base. A new telephony solution is needed to meet the new occupant's requirements of not more than 240 (two hundred forty) users with total of not more than 400 (four hundred) telephony devices including offices, faxes, laboratories, staff rooms, treatment rooms, classrooms, conference/meeting rooms, and staff rooms.

Building P144 is currently unoccupied due to the construction. Occupants are schedule with full occupancy scheduled for 31 May 2019.

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

To implement a contractor provided telephony solution at CFB Petawawa for Bldg P144, phones, installation, configuration, provisioning, and one year of maintenance. All work to be coordinated with the SSC Desk officer and the Site POC.

3. Project Scope

The contractor will design, configure and implement a contractor provided a telephony/unified communications system solution connecting to BASE TELECOM SERVICES (T1-PRI: PSTN and CSN). All work to be coordinated with the SSC Desk officer and the Site POC.
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. Provide VoIP Telephony system that will 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 list 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. If available, relevant Application Notes;
    c. Must have dual hot swappable power supplies supporting NEMA 5-15 grounded (Type B) receptacles,
    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,

2. Telephony solution must support the following functions locally:
    a. Support DTN dial plan as per Appendix C,
    b. On base Extension dialing (4 digits),
    c. Uniform dial plan (UDP),
    d. Coordinated dial plan (CDP),
    e. Voicemail,
        i. Provide to the specifications defined in Appendix F1: Voicemail Requirements;
    f. Auto attendant with dial by name capability,
        i. Provide to the specifications defined in Appendix F2: Auto Attendant Requirements
    g. Local emergency call notification,
        i. Must provide notification to include minimum of emergency caller name, extension and time;
    h. Must have a system/solution configuration management software defined in Appendix F3,
    i. Must have the ability to manage through remote control operations by national configuration managers;
    j. Should support set paging,
3. Telephony solution must provide the following capacities:
    a. Minimum one (1) management port for local system administration,
    b. Minimum three (3) T1 PRI ports as tie trunks to base PBX,
    c. Minimum twenty four (24) FXS ports connecting to analog phones, faxes and public address system,
    d. Minimum 400 (four hundred) user telephones and devices,
    e. Minimum 240 (two hundred forty) voicemail boxes,
4. Telephony solution must provide the following capabilities:
    a. Must be compatible with DND deployed Avaya CS1000 Release 5.0 (DSN),
        i. Must support communication with Avaya CS1000 Release 5.0 (DSN) systems via PRI (T1 and E1) using QSIG;
        ii. QSig as specified in Appendix H,
    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 J;
    c. All VoIP media and signalling traffic must 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 AES128 (AES256 preferred) for VoIP signalling,

d. Must 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;

5. Telephony system must provide the following devices:
   a. Must provide analog telephones in compliance to specifications defined in Appendix I1: Analogue Telephone Requirements;
      i. Provide minimum quantity of twenty (twenty) Analogue Telephones;
      ii. Install no more than 10 (ten) Analogue Telephones;
   b. Must provide VoIP telephones in compliance to specifications defined in Appendix I2: VoIP Basic Telephone Requirements;
      i. Provide minimum quantity of 24 (twenty four) VoIP Basic Telephones;
      ii. Install no more than 20 (twenty) VoIP Basic Telephones;
   c. Must provide VoIP telephones in compliance to specifications defined in Appendix I3: VoIP Standard Telephone Requirements;
      i. Provide minimum quantity of 360 (three hundred sixty) VoIP Standard Telephones;
      ii. Install no more than 310 (three hundred ten) VoIP Telephones;
   d. Must provide VoIP conference telephones in compliance to specifications defined in Appendix I5: VoIP Conference Telephone Requirements
      i. Provide minimum quantity of 4 (four) VoIP Conference Telephones;
      ii. Install no more than 3 (three) VoIP Conference Telephones;
   e. Must provide wired headsets compatible with VoIP Standard Telephones supplied
      i. Provide minimum 40 (forty) one ear headsets for either left or right ear (universal) use,
      ii. Provide minimum 20 (twenty) two ear headsets;
   f. Must install telephone wallmounts as per table below defining quantities,
      i. Installation into drywall only;

Table: Devices Quantities Required

	Supplied (minimum)	Installed (maximum)	Wallmounts / Installed
Analog telephones	20	10	12 / 10
VoIP Basic Telephones	24	20	12 / 10
VoIP Standard Telephones	360	310	20 / 16
VoIP Conference Telephones	4	3	0
Additional Analog FXS ports	6	4	na
T1-PRI ports	3	2	na
Wired One ear headsets	40	na	na

Wired Two ear headsets	20	na	na

6. If telephony solution requires telephony fibre optic equipment, vendor must supply as per the following specifications:
    i. Define how many fibre pairs are required,
    ii. Must be compatible with existing DND deployed Luxcom OM-200 systems,
    iii. Must support communication with Avaya CS1000 Release 5.0 (DSN) systems via PRI (T1 and E1) using QSIG;
    iv. SSC and DND will provide SM fibre optic pairs with ST connectors between building P144 and building H101,
        1. Vendor must define fibre optical equipment rack space and power requirements to be placed in H101 MTR,
        2. Vendor must define fibre optical equipment rack space and power requirements to be placed in in P144 MTR,
        3. Vendor must supply optical equipment cabling (to ST),
        4. Vendor must supply PRI cabling for vendor supplied equipment,
        5. Fiber optic connectivity options as specified in appendix A,
    v. Must be 19” rack mountable,
    vi. Must have power supplies supporting NEMA 5-15 grounded (Type B) receptacles,
    vii. Must include spare equipment at a minimum a power supply (qty 2), fibre optic card (qty 2), and PRI card (qty 2),
    viii. Refer to appendix A for connectivity options,
7. Must support telephony features/functions and user class of services as per Appendix E: DND User Class Of Service;
    a. Contractor must state compliance and functionality;
8. Must support network class over services as per Appendix D: DND Network Class Of Service;
    a. Contractor must state compliance and functionality;
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. Install and configure one (1) UPS (Eaton model 5PX) and batteries at each location ;
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 software & hardware support, and
    a. Provide one (1) year ONSITE maintenance contract (operational hours are 12 / 5 with normal business hours of 0700 to 1700hrs) with response time of four (4) hours).

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;
    a. Telephony IP networks including media and signalling will be VLANs segregated and isolated from all other networks (absolutely no external VoIP network connectivity),
    b. Telephony IP network management networks will be separate VLANs that are managed as per DND network security (identical to LAN equipment management),
2. Telephony call routing conforming with the DTN national dialing plan and policies as provided by DND/SSC,
3. End users, devices programming, and voicemail programming into the system(s) according to 'User Profile Worksheet' which will be provided upon contract award,
4. Two (2) Bilingual Auto Attendant / IVR trees to two (2) levels depth with voice recordings from SSC/DND personnel,
5. Ensure emergency calls are routed according to base policy,
6. Enable on-site notification of emergency calls as directed by SSC/DND.

3.3 Interface with:

The system will be interfacing with the following SSC/DND provided eqpt:

1. Up to Two (2) T1 PRI trunks to the CFB Petawawa base PBX (See Appendix A),
2. One (1) analog telephone line or analog trunk to the paging/announcement (supplied by DND),
3. Ten (10) analog telephone station lines to the facsimile machines (supplied by DND),

3.4 Pre-staging tasks

Contractor can pre-stage the system to minimized on site time; possible pre-staging tasks are

- Stage equipment in lab environment when applicable,
- Configure/ Provision/ Program VoIP Telephony system hardware and software,
- Configure/ Provision/ Program the T1 PRI connectivity to base PBX,
- Configure/ Provision/ Program the voicemail(s), and auto attendant(s),
- Configure/ Provision/ Program the telephones and users,
- Ship all equipment to CFB Petawawa building P144, 250 Somme Street, Petawawa, ON K0J 1J0 site ensure label contains building P144 and building POC.

3.5 Onsite tasks

- Meet with the POC and validate the telephony solution requirements,

- Review with the SSC rep the "User Profile" document, the equipment room layout and the site internal cabling,
- As required, participate in all project meetings/conference calls with the SSC Desk officer and/or site POC,
- Work with the SSC/DND and other contractor teams (as designated by SSC/DND), Inventory the equipment prior to starting the installation,
- Verify the rack power provide for the system install, ensure grounding as per manufacturer specifications,
- Installation of related telephony hardware and cabling (Note: depend on the configuration and/or building cabling contractor may choose to use either 25PR Pigtails or RJ45 plugs),
- Connect the system to the building wiring,
  - To establish VoIP Telephony system connection with base PBX T1-PRI, (see Appendix A),
- The installation, and implementation of Telephony system to include (but not limited to) the following,
  - Digit dialing for building P144 telephony users and devices:
    - Four (4) digits in-building and on base dialing,
    - PSTN access via access code 89 which is passed entirely to base PBX,
    - CSN access via access code 86 which is passed entirely to base PBX,
  - Program and configure voicemail service,
  - Program/setup 2 (two) Auto Attendant(s) / IVR tree(s) (maximum 2 levels) as directed,
- Program a maximum of three hundred fifty (350) locals/sets with associated two hundred ten (210) voice mailboxes including:
- Program and configure up to 2 (two) analog FXS lines for faxes,
- Program and configure up to two (2) VoIP conference sets and additional microphones,
- Program and configure one (1) analog FXS line for public announcement system (SSC provided),
- Program and configure up to 2 (two) VoIP emergency calls On Site Notification telephones as directed by site POC,
- Install and configure all telephones and/or lines,
  - Place set in their respective location/offices,
- Ensure 911 call routing is functioning correctly with local / base PSAP with building onsite notification;
- Provide and Complete acceptance test plan of the solution with site POC and SSC/DND representatives:
  - Sample test plan provided in Appendix K
- Participate in customer testing of all equipment to ensure end to end connectivity and expected feature operation,
- Complete USER training session, and SYSTEM ADMINISTRATOR training session within two (2) weeks before or after of building P144 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),

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),
- 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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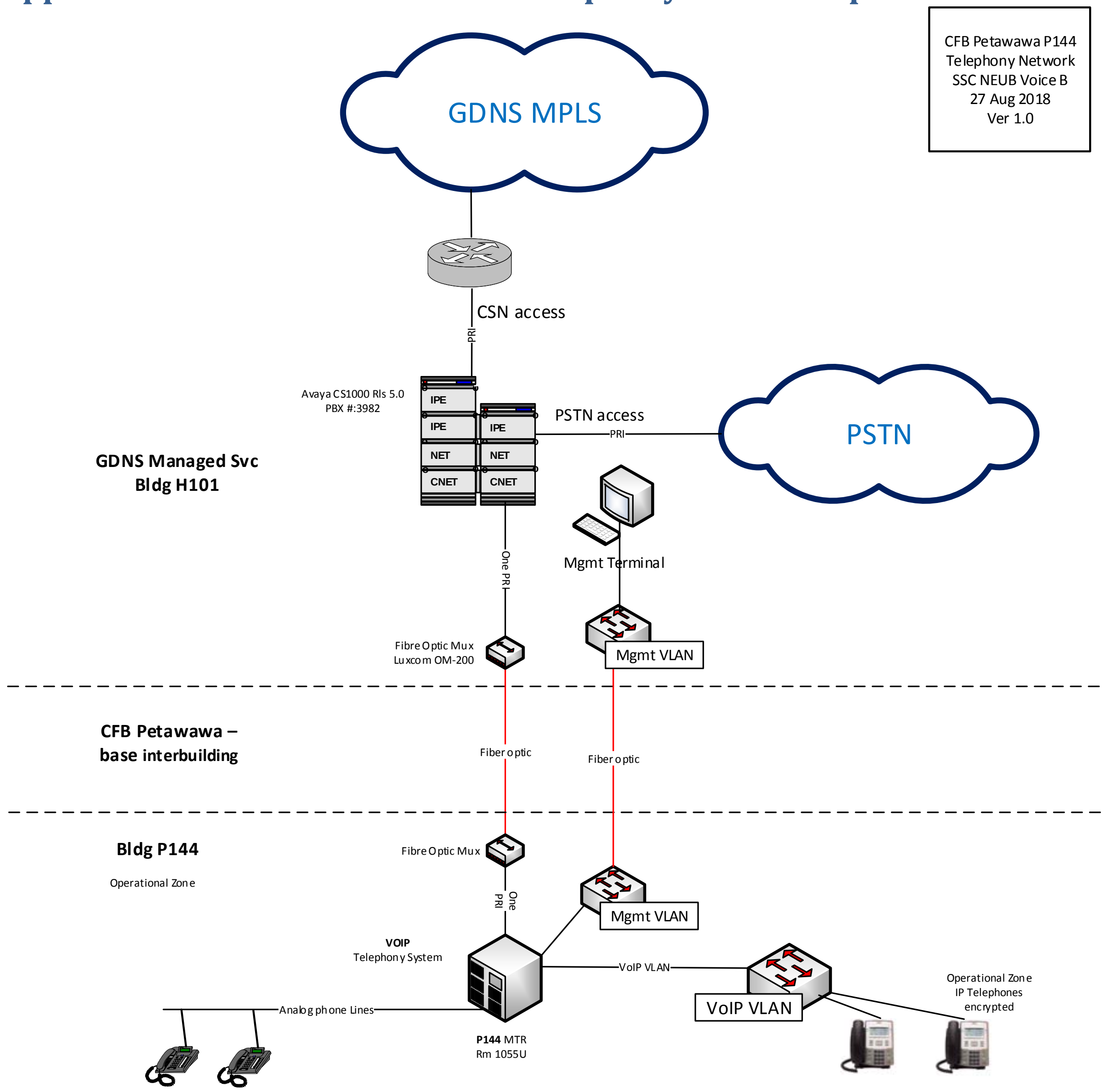
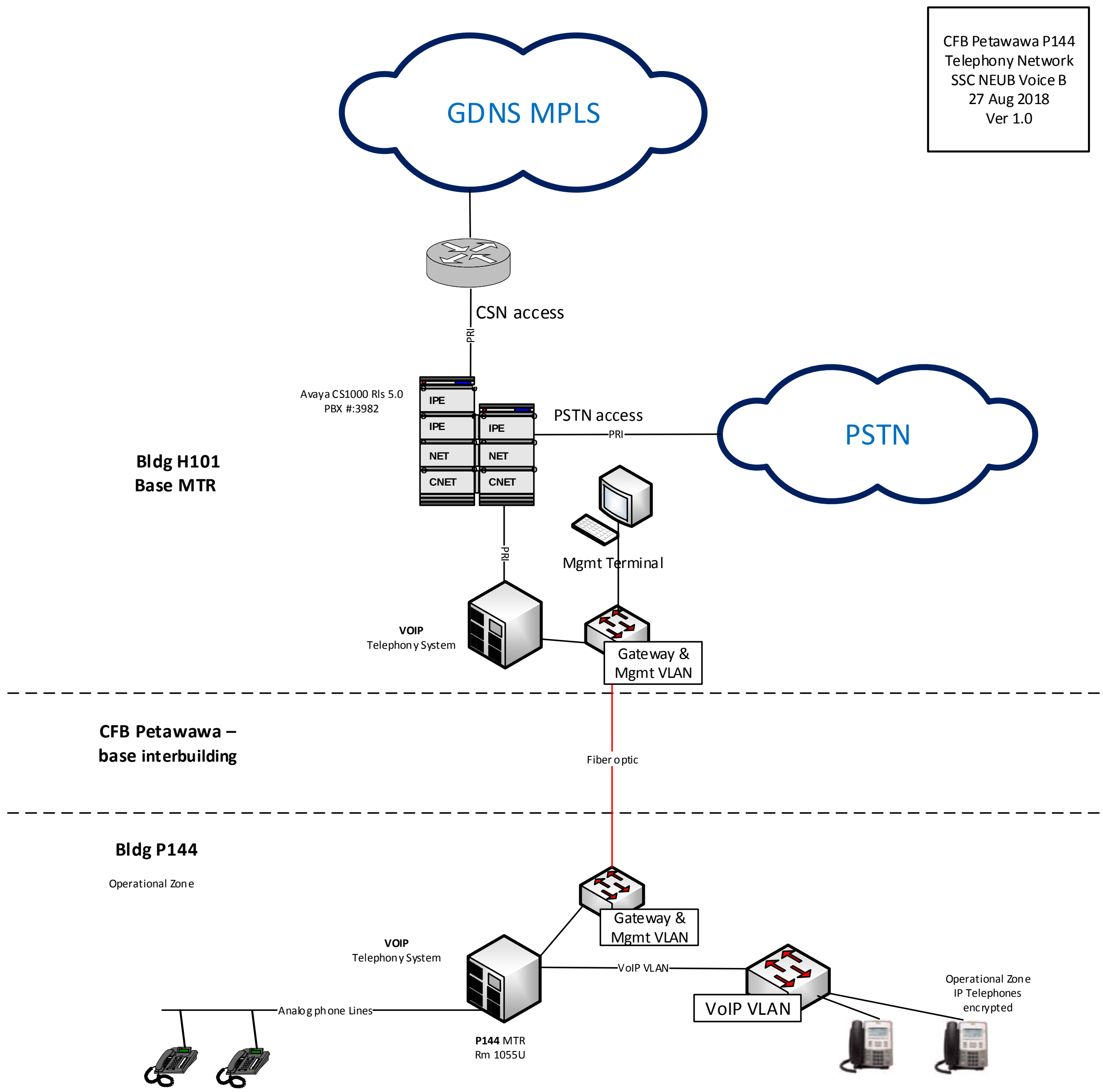
Appendix A: CFB Petawawa P144 Telephony Solution Options




CFB Petawawa P144
Telephony Network
SSC NEUB Voice B
27 Aug 2018
Ver 1.0
GDNS MPLS
CSN access
PRI
Avaya CS1000 Rls 5.0
PBX #:3982
IPE
IPE
IPE
NET
NET
CNET
CNET
PSTN access
PRI
PSTN
Bldg H101
Base MTR
PRI
Mgmt Terminal
VOIP
Telephony System
Gateway &
Mgmt VLAN
CFB Petawawa –
base interbuilding
Fiber optic
Bldg P144
Operational Zone
Gateway &
Mgmt VLAN
VOIP
Telephony System
VoIP VLAN
VoIP VLAN
Operational Zone
IP Telephones
encrypted
Analog phone Lines
P144 MTR
Rm 1055U

Appendix B: Building P144 Fibre Optic Requirements

Following sections define fibre optic requirements for the P144 telephony solution.

1. CFB Petawawa Inter-Building Fibre Optics
   1.1. CFB Petawawa PBX 3982 can provide a T1 PRI (PSTN & CSN) service via CFB Petawawa inter-building fibre optic cabling as per the above specifications.

   1.2. DND will ensure inter-building cable distances are within cable and signaling specifications.

   1.3. CFB Petawawa utilizes 1310 nm single mode optics. ST optical connectors are used to terminate the base fibre optic cabling.

   1.4. CFB Petawawa PBX 3982 fibre optic service terminates in building P144 MTR room 1055U.

   1.5. CFB Petawawa inter-building fibre optic multiplexers currently used are Luxcom OM200.

   1.6. Interoperability with Luxcom OM200 OCA Sonet OC3 framing and signalling is required.
      1.6.1. https://www.luxcom.com/product/om200/

Appendix C: DND Dial Plan

Appendix C1: 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 C2: 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 D: DND Network Class Of Service

Appendix D1: Basic NCOS Structure

NCOS	FRL	Description
0	0	Internal, CDP, 911
1	1	Local, Toll Free, 711 (TRS)
2	2	Tie Trunks
3	3	CSN, DSN Zone 1 (312, 315, 317, 319), EAS
4	4	Canada and USA Toll, 411, 555 (L/D Info), 700-555-4141
5	5	All Operator Calls (0, 0 Plus, 01, 011), IVSN, 600
6	6	DSN Zone 4, Australia (715), Iridium (707/717)
7	7	Unrestricted, 611, NTAS (310)

Appendix D2: Advanced NCOS Structure

NCOS	FRL	Description
41	4	Canada and USA Toll with Expensive Route Warning Tone
51	5	Toll with ERWT
61	6	Toll with ERWT
71	7	MLPP Priority
72	7	MLPP Immediate
73	7	MLPP Flash
74	7	MLPP Flash Over-ride
99	7	Attendant Position

Appendix E: DND User Class Of Service

DND/CF National Dial Plan (Dec 2011)		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	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
MITNET	83				x	???	x	x	x	x	???	x	x	x	x
DOG (CSN)	86				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

Appendix F: Telephony System Applications Requirements

Appendix F1: 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 F2: 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 F3: PABX 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 G: CFB Petawawa PBX Specifications

CFB Petawawa PBX 3982 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 +

Software packages installed

TYPE	pkg
OPTF	1
CUST	2
CDR	4
CTY	5
RAN	7
TAD	8
DNDI	9
EES	10
INTR	11
ANI	12
ANIR	13
BRTE	14
DNDG	16
MSB	17
SS25	18
DDSP	19
ODAS	20
DI	21
CHG	23
CAB	24
BAUT	25
CASM	26
CASR	27
BQUE	28
NTRF	29
NCOS	32
CPRK	33
SSC	34
IMS	35

DTD	138
FFC	139
DCON	140
MPO	141
ICP	143
ABCD	144
ISDN	145
PRA	146
ISL	147
NTWK	148
IEC	149
DNXP	150
CDRE	151
FXS	152
IAP3P	153
PRI2	154
ACNT	155
THF	157
FGD	158
NAS	159
FNP	160
ISDN_INTL_SUP	161
SAR	162
MINT	163
LAPW	164
GPRI	167
COOP	169
ARIE	170
CPGS	172
ECCS	173
AAA	174
NMS	175
EOVF	178
HVS	179
DKS	180
SACP	181
TFM	182
VNS	183
OVLP	184
EDRG	185
POVR	186
RPA	187
L1MF	188
SVCT	189
SECL	191
ORC-RVQ	192

RCK	193
FAXS	195
OHOL	196
FFCSF	198
IPRA	202
XPE	203
XCT0	204
XCT1	205
MLWU	206
NACD	207
HSE	208
MLM	209
MAID	210
MLIO	211
VAWU	212
EAR	214
ECT	215
BRI	216
IVR	218
MWI	219
MSDL	222
FC68	223
M911	224
CWNT	225
MSDL SDI	227
MSDL STA	228
SSAU	229
DNWK	231
PEMD	232
BRIT	233
FCDR	234
BRIL	235
ACRL	236
MCMO	240
MULTI_USER	242
ALRM_FILTER	243
SYS_MSG_LKUP	245
VMBA	246
CALL_ID	247
M911 ENH	249
DPNA	250
SCDR	251
ARFW	253
PHTN	254
INBD	255
ADMINSET	256

NMCE	364
FIBN	365
BNE	367
CPP_CNI	368
STS_MSG	380
CDIR	381
VIRTUAL_OFFICE	382
ATAN	384
NI2NAME	385
M3900_PROD_ENH	386
VIR_OFF_ENH	387
ACDE	388
PONW	389
UUI	393
OAS	394
ICON	397
PCA	398
H323_VTRK	399
LOCX	400
PVQM	401
GRPRIM	404
SIP	406
CAC	407
MS_CONV	408
HIGH_AVAIL	410

Appendix H: 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 I: Telephone Requirements

Appendix I1: 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 I2: VoIP Basic 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. Optional display,
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 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
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. Must have configurable (on/off) duplex speakerphone capabilities,
17. Must have capability to support standard wired headset,

18. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
19. Must be capable of supporting SIP without hardware modification.

Appendix I3: VoIP Standard 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 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
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. Must have configurable (on/off) duplex speakerphone capabilities,
17. Must have capability to support standard wired headset,
18. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
19. Must be capable of supporting SIP without hardware modification.

Appendix I4: CNSSI (TSG6) VoIP Telephone Requirements (if applicable)

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

1. Must be CNSSI 5000 and CNNSI 5006 (TSG6) certified,
2. Must be a Class A device,
3. Optional Push to Talk handset must be available,
4. Must be black or grey,
5. Must be wall mountable,
6. Must have minimum 2 line display with 16 characters each line,
7. Must have iconic labels or be bilingual (French & English) labelling,

8. Support dual-tone multi-frequency signaling (DTMF),
9. Must support more than one line,
10. Must support Ethernet
11. Must support Power over Ethernet (POE),
12. Must support secure real time protocol (SRTP) with AES-256,
13. 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
14. Must have ringer volume control,
15. Must have handset volume control,
16. Must have mute capabilities and Mute button,
17. Must have visual ring indication,
18. Must have visual message waiting indication,
19. Must have configurable (on/off) duplex speakerphone capabilities,
20. Must have capability to support standard wired headset,
21. May have optional set feature support such as redial, call history, call timer, conference, transfer and date & time, and
22. Must be capable of supporting SIP without hardware modification.

Appendix I5: VoIP Conference Telephone Requirements

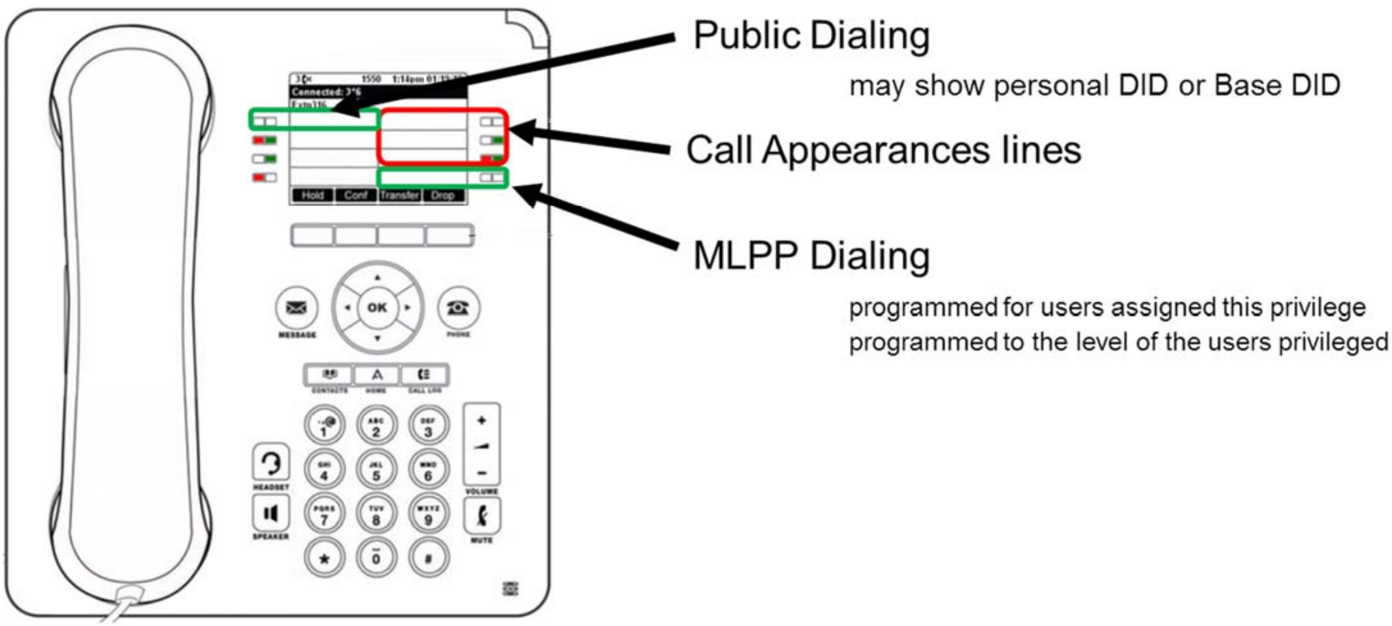
Following section defines the requirements of the telephony solution as applicable to VoIP conference telephones.

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 I6: Sample Phone Button Layout

Following is a sample template for Telephone buttons layout (from existing DND systems):



Appendix J: Workplace Communication Service (WCS) Specifications

Following section defines the specifications of Canada's WCS solution. The telephony solution of building P144 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 K: 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 L: UPS Equipment

To be supplied by SSC/DND

Requirement for UPS		
Eaton PN	**Description**	**Qty.**
5PX1000RT	Eaton 5PX 1000 120V 2U Rack Tower, 8ft 5-15P Input Cord, (8)5-15R Outlets	1
5PXEBM48RT	Eaton 5PX 48V EBM R/T 2U, 1000-2200va Models	3
103007018-5591	2-post rail kit	1