TELSTRA IP TELEPHONY

CUSTOMER DETAILED INTEGRATION GUIDE

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1. About this Document

1.1. Introduction

This document is designed to assist the customer with its TIPT integration, training and deployment activities. Additional troubleshooting and tips are provided in the Appendix of this document.

> **Note:** This document does not constitute an IP Tel design but serves as a generic reference guide for the adoption of the TIPT Hosted suite of products and applications only within the customer's environment.

1.2. Minimum customer environment

The customer environment must at a minimum support:

CUSTOMER ENVIRONMENT	MINIMUM SUPPORT
LAN Environment	• IP Tel Enabled LAN Switches (VLAN tagging) with structured CAT5 (or better) cabling • VLANs to support VoIP (Voice VLAN) • Dynamic Host Configuration Protocol Server – Voice VLAN (with DHCP Scope Options) • Quality of Service (QoS) to deliver end-to-end classification & prioritisation (LLQ) • IEEE 802.3af Power Over Ethernet (recommended) • Cisco Discovery Protocol (CDP) to provide automatic VLAN configuration (recommended) Link Layer Discovery Protocol (LLDP) is supported with handset SIP firmware 3.3.0 and above. • NTP server access (MWAN will provide if deployed)
Desktop Integration and TIPT Service Management	• Domain Name Services (DNS) Server – Conditional DNS Forwarding • Telstra Telephony Toolbar, CommPilot Web and miRECEPTION console applications

1.3. Overview of the Telstra IP Telephony network

The diagram below provides an overview of the Telstra IP Telephony construct.

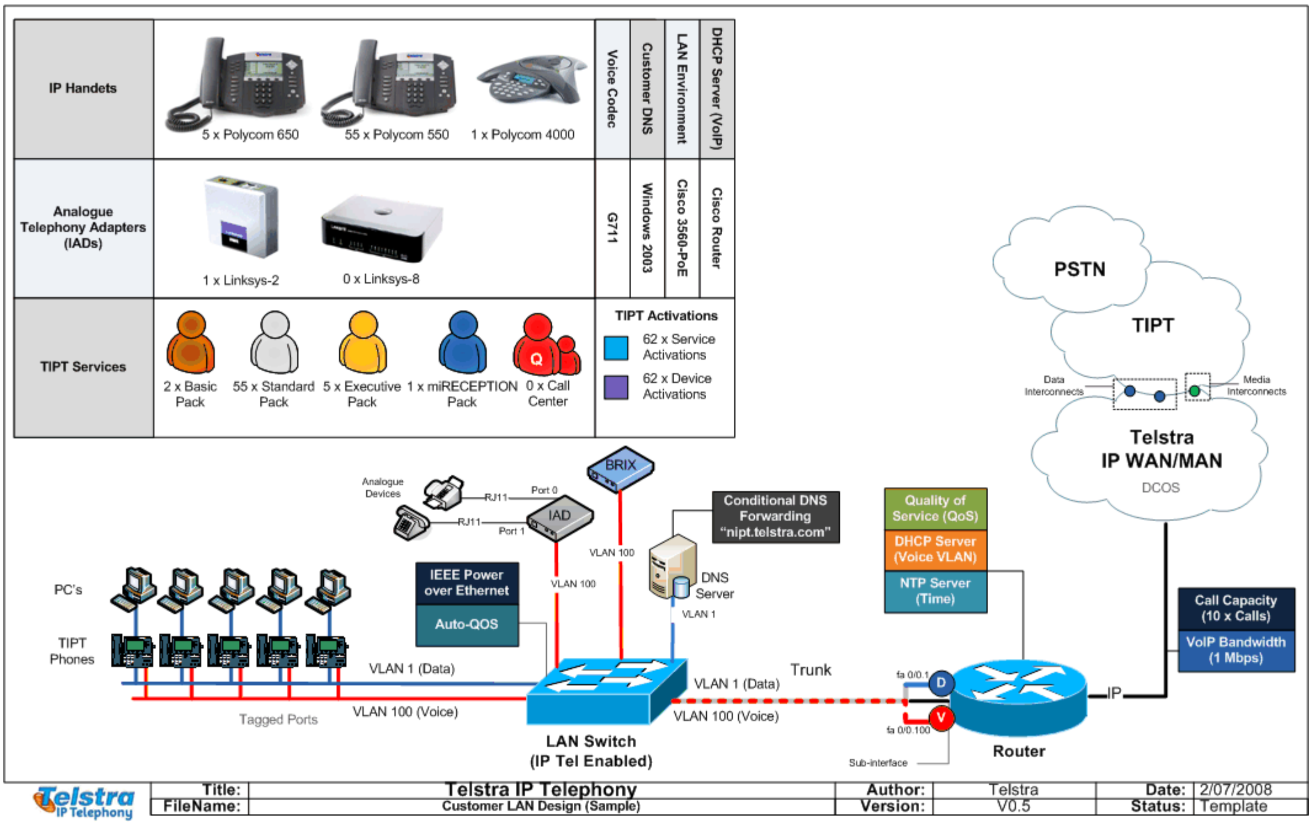


Figure 1 – Telstra IP Telephony (TIPT) Overview

2. Customer Integration

2.1. Contents

The following sections provide customer integration information to support the TIPT solution. Specifically the customer must configure and validate the following:

ITEM	DETAILS
IP Routing	
LAN environment	• Cabling • VLANS (VoIP VLAN) • Power over Ethernet (PoE) • NTP Server • DHCP Server (VoIP VLAN) • DNS Server • Firewalls • Quality of Service (QoS) • Web Proxy (Caching) Server
CPE Device Configuration Delivery	• Polycom IP Phones • Integrated Access Devices (IADs) • Polycom Group Series video conferencing (VC) endpoints
BRIX Verifier	
SIP Ping (SipSak Testing Tool)	The SipSak utility provides an invaluable pre-commissioning mechanism to test SIP connectivity to the TIPT platform by validating access to each Media POP.
Floor Plan (Professional Installation)	

2.2. Disclaimer

This guide is not intended to detail customer specific IP Telephony requirements. Refer to Telstra specific configuration (that is: Customer Design).

2.3. IP Routing

2.3.1. Introduction

Telstra requires that the customer has interconnects to the TIPT platform from the customer's Telstra IP-VPN as per the figure below. Connectivity is via the private IP network and not via the Internet.

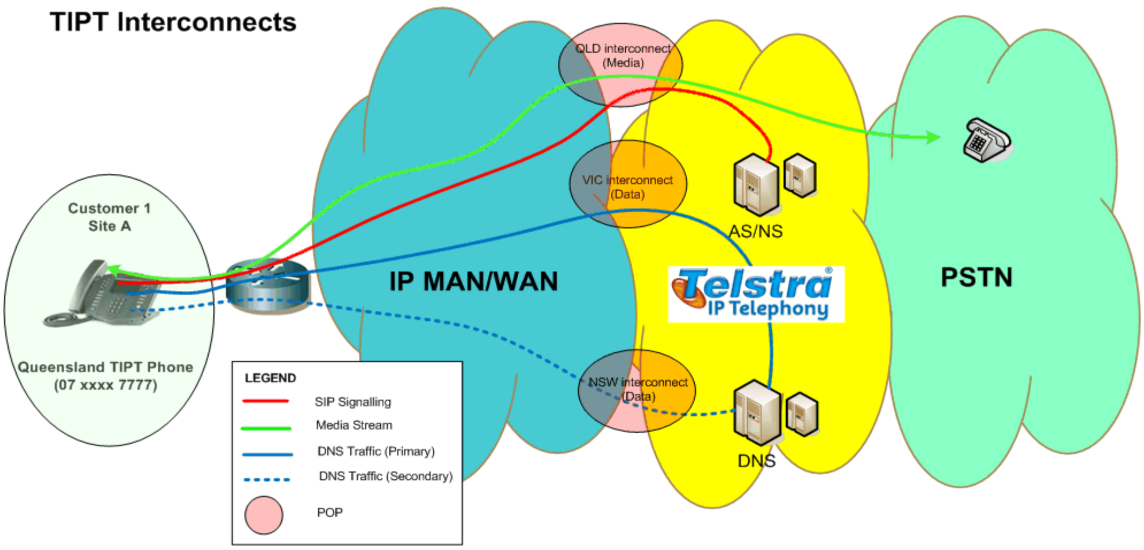


Figure 2 – TIPT Interconnects

2.3.2. Interconnects

Two data interconnects located in Melbourne and Sydney for access to the TIPT application server clusters are required as well as at least one Media interconnect per State dependant on customer requirements. Voice signalling and RTP streams (off-net) are routed via the TIPT Session Border Controllers (SBC) which are deployed in a High Availability, Stateful, Active / Standby configuration and provide Network Address Translation (NAT) and Voice firewall services.

The customer must be able to route to these networks to reach the TIPT platform. IP routing can be configured via static or dynamic routing protocols at the customer edge device (BGP is preferred).

A **traceroute** to these hosts should be performed prior to deployment. The traceroute will only show the first few hops as per sample below as ICMP is blocked inside the Telstra core. IP routing to these interconnects (POPs) must not occur via the Internet or via third party data-centres.

!DNS SERVERS (VIC & NSW DATA POPS)
c:\>tracert -d 203.52.0.221 Tracing to 203.52.0.221 over a maximum of 30 hops:
1 <1 ms 1 ms <1 ms 192.168.102.1 2 16 ms 17 ms 17 ms 10.200.0.2 3 18 ms 18 ms 18 ms 10.86.64.17 4 18 ms 18 ms 18 ms 10.86.64.18 5 * * * Request timed out.

```
c:\>tracert -d 203.52.1.222

Tracing to 203.52.1.222 over a maximum of 30 hops:

  1     1 ms     1 ms    <1 ms  192.168.102.1
  2    15 ms    16 ms    15 ms  10.200.0.2
  3    31 ms    31 ms    32 ms  10.86.65.17
  4    31 ms    31 ms    32 ms  10.86.65.18
  5     *        *        *     Request timed out.
```

!WEB SERVERS(VIC & NSW DATA POPS)

```
c:\>tracert -d 144.140.208.81

Tracing to 144.140.208.81over a maximum of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    16 ms    17 ms    17 ms  10.200.0.2
  3    18 ms    18 ms    18 ms  10.86.64.17
  4    18 ms    18 ms    18 ms  10.86.64.18
  5     *        *        *     Request timed out.
```

```
c:\>tracert -d 144.140.181.80

Tracing to 144.140.181.80 over a maximum of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    16 ms    17 ms    17 ms  10.200.0.2
  3    18 ms    18 ms    18 ms  10.86.64.17
  4    18 ms    18 ms    18 ms  10.86.64.18
  5     *        *        *     Request timed out.
```

```
c:\>tracert -d 144.140.222.33

Tracing to 144.140.222.33 over a maximum of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    16 ms    17 ms    17 ms  10.200.0.2
  3    18 ms    18 ms    18 ms  10.86.64.17
  4    18 ms    18 ms    18 ms  10.86.64.18
  5     *        *        *     Request timed out.
```

!MEDIA INTERCONNECTS (STATE MEDIA POPS)

```
c:\>tracert sbc-vic.nipt.telstra.com

Tracing to sbc-vic.nipt.telstra.com [203.52.0.167] over a maximum
of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    17 ms    17 ms    16 ms  10.200.0.2
  3    47 ms    45 ms    45 ms  203.52.0.162
  4     *        *        *     Request timed out.
```

```
c:\>tracert sbc-nsw.nipt.telstra.com

Tracing to sbc-nsw.nipt.telstra.com [203.52.1.167] over a maximum
of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    17 ms    17 ms    16 ms  10.200.0.2
  3    47 ms    45 ms    45 ms  203.52.1.162
  4     *        *        *     Request timed out.
```

```
c:\>tracert sbc-qld.nipt.telstra.com

Tracing to sbc-qld.nipt.telstra.com [203.52.1.167] over a maximum
of 30 hops:

  1    <1 ms     1 ms    <1 ms  192.168.102.1
  2    17 ms    17 ms    16 ms  10.200.0.2
  3    47 ms    45 ms    45 ms  203.52.1.162
  4     *        *        *     Request timed out.
```

```
c:\>tracert sbc-wa.nipt.telstra.com

Tracing to sbc-wa.nipt.telstra.com [203.52.1.167] over a maximum
of 30 hops:

  1     1 ms     1 ms    <1 ms  192.168.102.1
  2    81 ms    16 ms    17 ms  10.200.0.2
  3    60 ms    60 ms    59 ms  203.52.2.162
  4     *        *        *     Request timed out.
```

2.3.3. Interconnect Configuration

The table below shows the configuration for the Data, Media and AS interconnects.

INTERCONNECTS	CONFIGURATION	DESCRIPTION
Data interconnects (Mandatory)	203.52.0.0/23	Kent St and Lonsdale St – Media and Data
	203.41.188.96/28	Lonsdale St – TIPT Web Servers
	203.42.70.224/28	Kent St Data – TIPT Web Servers
	144.140.208.16/29	Exhibition St – Digital Business
	144.140.208.32/28	Exhibition St – TIPT DMS
	144.140.208.80/28	Exhibition St – TIPT Web Servers
	144.140.162.40/29	Pitt St – Digital Business
	144.140.162.48/28	Pitt St – TIPT DMS
	144.140.162.80/28	Pitt St – TIPT Web Servers
	144.140.181.80/28	St Leonards – TIPT Web Servers
	144.140.222.0/24	Clayton – TIPT & Digital Business Web Traffic
Media interconnects	203.52.0.0/24	Victoria – SBC

(As required)	203.52.1.0/24	NSW – SBC
	203.52.3.160/28	Queensland – SBC
	203.44.43.160/28	South Australia – SBC
	203.52.2.160/28	Western Australia – SBC

Table 1 – TIPT Interconnects (Data & Media POPs)

NSW, SA and VIC exchanges will be used for ACT, NT and TAS SBC's

2.4. LAN Environment

2.4.1. LAN switches

IP Telephony enabled LAN switches supporting VLANs, IEEE8021.Q (VLAN Tagging), IEEE 802.3af (Power over Ethernet), Quality of Service (QoS) must be used. VLAN segmentation is required to demarcate and allow prioritisation of voice traffic. VLAN ID 100 is recommended as the Voice VLAN to ensure configuration consistency across all sites. PC's will typically piggy-back from the IP Phone but reside in the Data VLAN (that is: VLAN ID 1).

The use of Cisco switches with CDP provides automatic VLAN assignment of Polycom IP Phones where Voice VLAN's have been configured. In non-Cisco LAN switch environments the VLAN ID (that is: VLAN 100) will need to be manually configured in each IP handset unless the IP handset has SIP firmware 3.3.0 or above where there is support for LLDP.

Polycom Group Series video endpoints can also use the Voice VLAN.

2.4.2. WAN router

Additionally the WAN router must be designed and configured with a suitable QoS policy to categorise and prioritise VoIP traffic to ensure Grade of Service. A LAN design must be undertaken to ensure that the converged solution meets TIPT requirements. Site documentation (e.g. network diagrams) should be updated to support the IP Telephony solution.

Key: Data VLAN = Green, Voice VLAN = Orange

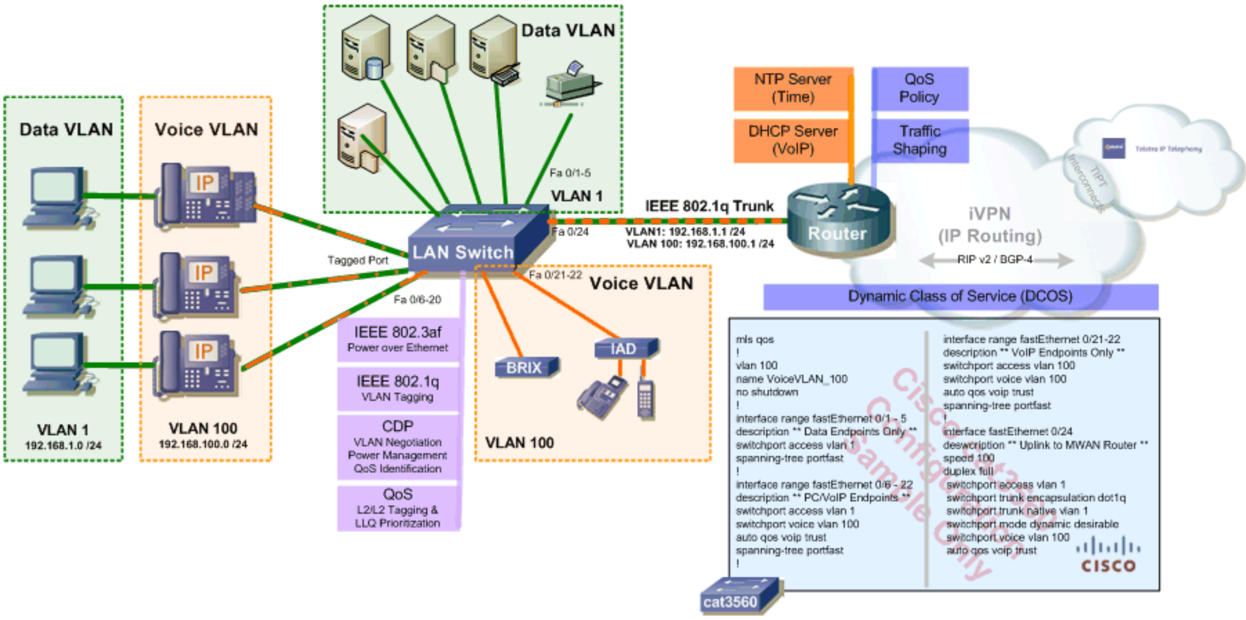


Figure 3 – Customer LAN Switches - Voice VLANs (Design & Build)

2.4.3. Switch port

The customers LAN switch(s) must be configured with an additional Voice VLAN to support TIPT handsets and video conferencing (VC) endpoints. The LAN switch is typically connected to the Telstra MWAN router using a trunk port which transports both Data (i.e. VLAN 1) and Voice (i.e. VLAN 100) VLANs. At larger sites Customers can choose to implement Layer 3 switches where appropriate infrastructure requirements dictate.

Switch ports connected to IP phones must be configured as an IEEE802.1Q tagged port which is used to trunk the Voice and Data VLANs. In this configuration the Data VLAN is untagged and the Voice is tagged. Existing customer devices such as servers (including Customer Data DHCP server) and printers must reside in the Data VLAN only using an untagged port.

The DHCP server supporting the Voice VLAN must assign the correct IP configuration to the CPE.

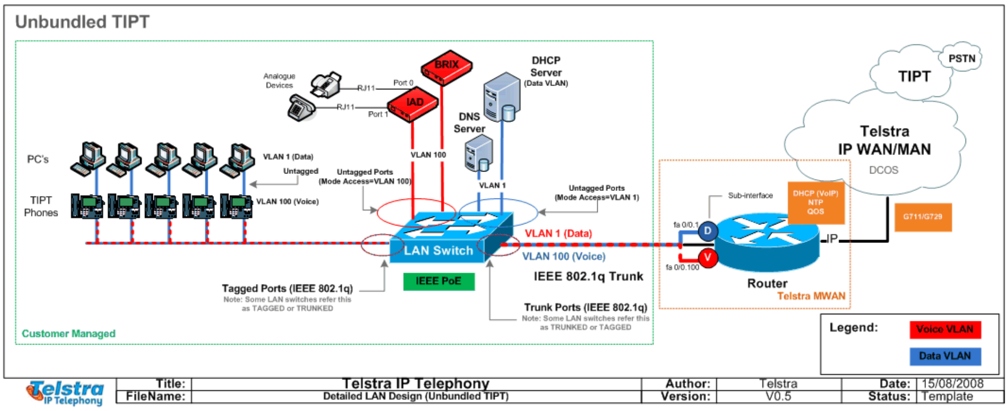


Figure 4 – LAN Configuration (Sample)

Specialist equipment including Integrated Analogue Devices (IADs) and BRIX verifiers must be connected directly into the Voice VLAN as per sample figure 5 below.

At least 1 x LAN switch-port is required to be configured into the Voice VLAN for management (diagnostic\troubleshooting) purposes.

> **Note:** These devices will not function correctly if connected via the Data VLAN.

2.4.4. Legacy switches

The customer should pay close attention to legacy switches (non-Cisco) or switches located in remote buildings which may not present VLAN or PoE to its intended location.

2.4.5. Testing requirements

The LAN build (including DHCP) must be completed and tested (Voice & Data) at each location prior to IP handset/IAD or Video Conferencing (VC) endpoint deployment. This includes testing both IP Phone and PC connectivity at all locations and across all access switches prior to deployment.

2.4.6. Disclaimer

Customer specific IP Tel LAN design is outside the scope of this document.

2.5. Cabling

2.5.1. Customer cabling

The customers LAN must have structured cabling of CAT5 or better in order to support IP Telephony. Enterprise grade IP Telephony places higher requirements on network cabling and LAN infrastructure.

All cabling including network patching is the responsibility of the customer and must be completed before the on-site arrival of professional phone installers.

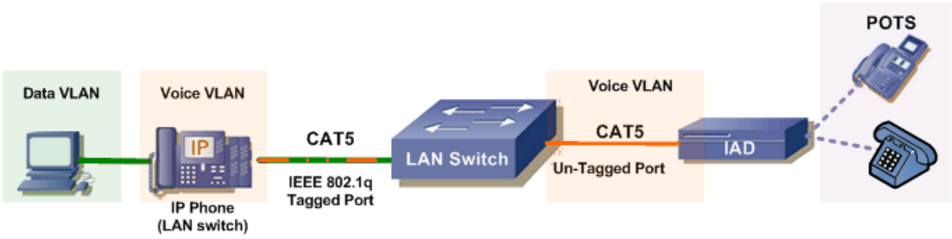


Figure 5 – Cabling & Network Patching Requirements

2.6. VLANs

2.6.1. Voice VLAN

VLANs are required to segment and provide grade of service to IP Phones. The use of a separate Voice VLAN for VoIP is essential for all deployments of IP Telephony. VLAN configuration and validation must be performed prior to IP handset deployment.

The Voice VLAN ports are nominally configured as IEEE802.1Q tagged and the Data VLAN ports configured as untagged. This allows the IP Phone to reside in the Voice VLAN whilst the PC resides in the Data network (e.g. VLAN 1). Printers and Servers must remain in the Data VLAN.

CISCO LAN SWITCH CONFIGURATION
interface FastEthernet0/xx
description “** TIPT Phone Port **”
switchport access vlan 1
switchport voice vlan 100
spanning-tree portfast
auto qos voip trust
interface FastEthernet0/xx
description “** TIPT IAD / Brix Port **”
switchport access vlan 100
spaning-tree portfast
auto qos voip trust
interface FastEthernet0/xx
description “** Data DHCP Server **”

```
switchport access vlan 1

spanning-tree portfast

auto qos voip trust


Note: xx is the port number on your device
```

Table 2 – Cisco Switch Configuration (Sample VLAN Configuration)

(E.g. Cisco IOS: ***show vlan***).

PORT FUNCTION	SWITCH CONFIG
Phone / PC port	Port is set “Untagged” in Data VLAN (e.g. 1) & “Tagged” in Voice VLAN (e.g. 100)
IAD / Brix port(s)	Port set to “Untagged” in Voice VLAN (e.g. 100) only.
Switch to Router Uplink	Port is set “Tagged or Untagged” in Data VLAN (e.g. 1) & “Tagged” in Voice VLAN (e.g. 100)
Data DHCP Server port(s)	Port set to “Untagged” in Data VLAN (e.g. 1) only.

Table 3 – Juniper / HP / Linksys / D-Link / Netgear TIPT Switch Configs

2.7. VLAN allocation

2.7.1. Cisco environment using CDP

On Cisco LAN switches Cisco Discovery Protocol (CDP) is used to transparently assign the Voice VLAN ID to the Polycom IP handset. Cisco LAN switches also use CDP to manage PoE advertisement and QoS identification.

(E.g. Cisco IOS: **show cdp neigh <detail>**).

2.7.2. Non Cisco environments without CDP (LLDP)

In non-Cisco LAN switch environments the VLAN ID must be set via DHCP option 128 (VLAN discovery) or manually configured into each IP Phone by the phone installer, as per procedure below for Polycom phones:

Process to “Hard Code” the VLAN ID into a phone that is just booting:

1. Press the Setup key just as the phone is booting (enter password of 456)
2. Down arrow to Ethernet Menu: (Select)
3. Down arrow to VLAN ID: (Edit)
4. Enter appropriate Voice VLAN ID: (Ok) (Back) (Back)
5. Select Save & Reboot Phone.

Process to “Hard Code” the VLAN ID into a phone that is already booted:

1. Select Menu
2. Press 3. Settings
3. Press 2. Advanced (password: 456 Enter)
4. Press 1. Admin Settings
5. Press 1. Network Configuration
6. Follow steps 2 – Step 5 above

Where Polycom handsets have been upgraded to SIP firmware 3.3.0 and above they support the use of the vendor neutral standard LLDP to allocate Voice VLAN ID, PoE and QoS.

Polycom Group Series video endpoints should have the VLAN ID configured during installation.

2.8. Power over Ethernet (PoE)

2.8.1. Using PoE

The use of UPS and surge protected IEEE 802.3af PoE switches is strongly recommended.

Using PoE allows flexibility in the deployment of IP Telephony within the customer site. The LAN design must also factor in PoE budgeting (especially in high density environments) to ensure the LAN is capable of supporting the necessary PoE requirements of the site.

The following table documents the PoE requirements of the Polycom IP phones.

PHONE MODEL	POWER BUDGET (POE)
Polycom VVX1500	14 Watts
Polycom IP670	14 Watts (15.4 Watts with Expansion Module).
Polycom IP650	12 Watts (15.4 Watts with Expansion Module). Manually configurable to 6 Watts.
Polycom IP560	8 Watts

Table 4 – Power over Ethernet (PoE)

Polycom Group Series video endpoints do not use PoE.

2.8.2. AC Power Packs

It is recommended that all IP Phones be plugged in as soon as possible to baseline power loading within the customer's environment. The use of AC power (plug) packs is considered clumsy and should be avoided. If AC Power Packs are used an accessible and available 240V power outlet is required at each IP phone location. . Power packs for Polycom handsets will be made available as appropriate on request.

(E.g. Cisco IOS: ***show power inline***).

2.9. Network Time Protocol (NTP)

2.9.1. Using NTP

The customer should utilise a NTP server (e.g. local Cisco Router) in order for the IP handsets to display the correct date and time. Under the new Device Management Solution (see section 2.4), NTP source is more critically needed for HTTPS certificate validation, and if the NTP source is not available, new devices may fail to boot, and existing devices fail to load a configuration or firmware update. In large environments it is recommended that an NTP hierarchy be used to ensure accurate time synchronisation whilst minimising WAN time traffic.

The location of the Time Server is specified using DHCP Option 42. The DHCP server must be configured correctly to provide this parameter.

(E.g. Cisco IOS: ***show ntp status***).

2.10. DHCP Server

2.10.1. Configuring DHCP server

The DHCP server used for the TIPT Voice VLANs (local or centralised) at each site must be configured to provide DHCP Options. The DHCP server may be configured by Telstra on the MWAN router (preferred deployment scenario) if required.

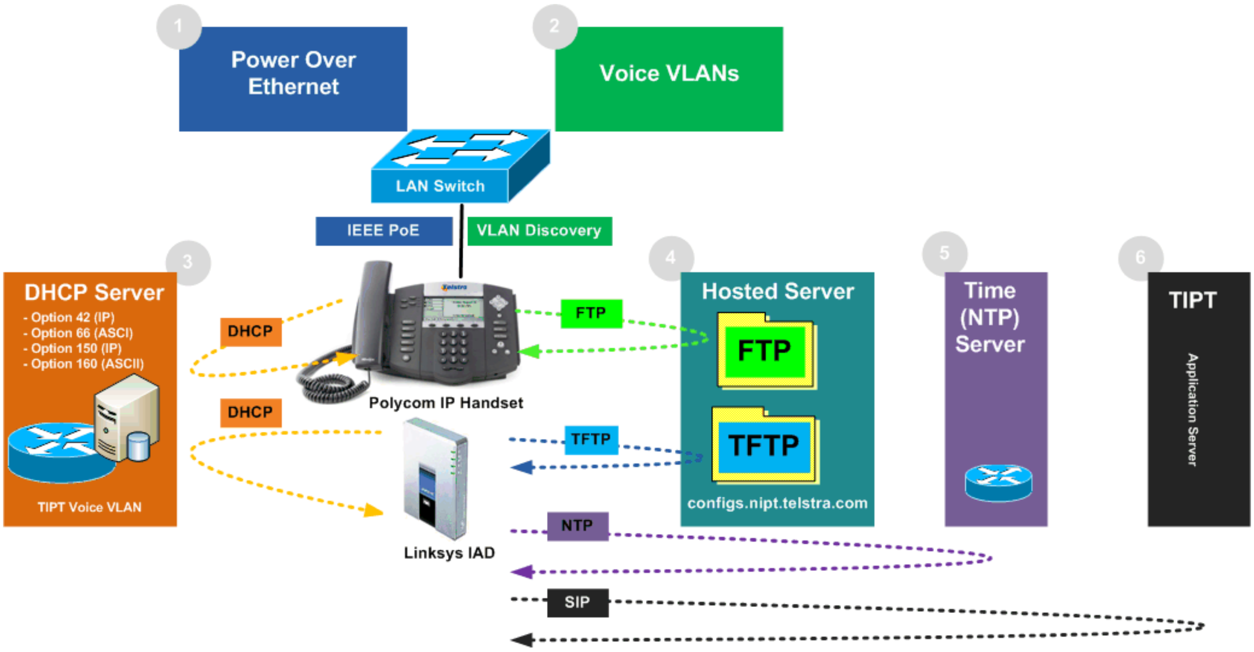


Figure 6 – DHCP Server Configuration (Voice VLAN)

2.10.2. DHCP options

DHCP Options are used to provide location parameters to the IP handsets as per the table below.

DHCP OPTION	FIELD FORMAT	VALUE
Option 42	IP Address	IP Address of NTP (Time) Server
Option 66	ASCII	FQDN of Hosted T/FTP Servers as a String
Option 150	IP Address	IP Address of Hosted TFTP Servers (for SIP connect IAD device)

Table 5 – DHCP server options for IP phone support (Voice VLAN) for CPE device configuration delivery via Hosted Config Server (old system)

DHCP OPTION	FIELD FORMAT	VALUE
Option 42	IP Address	IP Address of NTP (Time) Server
Option 66	ASCII	FQDN of T/FTP bootstrap Servers as a String
Option 150	IP Address	IP Address of Hosted TFTP Server(s) (for SIP connect IAD device)
Option 160	ASCII	FQDN of HTTP/S Device Management Solution Bootstrap Server

Table 6 – DHCP server options for IP phone support (Voice VLAN) for CPE device configuration delivery via Device Management Solution (new system)

Polycom Group Series video endpoints require NTP to be configured as above, but do not require FTP or TFTP.

2.10.3. Testing before the phone boot process

The customer must test that IP Phones at each site correctly obtain IP configuration information (Voice VLAN) and locate the various destinations defined above via DHCP Options. PC's which are piggy-backed behind the IP phone must obtain an IP address from the Data VLAN. This must be tested prior to deployment.

Messages pertaining to "Unable to obtain DHCP" or "Unable to locate Boot Server" must not be presented during the phone boot process. If you get one of these messages it indicates a DHCP configuration issue.

2.10.4. Cisco IOS DHCP server

The following Cisco IOS DHCP Server (Voice VLAN) configuration template may be used. Please note requirement for Telstra TIPT DNS to be used in the customer Voice VLAN scope.

<table>
<tr><th colspan="2">CISCO IOS DHCP SERVER CONFIGURATION</th></tr>
<tr><td colspan="2">service dhcp server
ip dhcp excluded-address 192.168.100.1 192.168.100.10
ip dhcp pool TIPT_VoiceVLAN
  network 192.168.100.0 255.255.255.0
  default-router 192.168.100.1
  dns-server 203.52.0.221 203.52.1.222
lease 7
  option 42 ip 192.168.100.1</td></tr>
<tr><td>Hosted Configuration Server</td><td>Device Management Solution</td></tr>
<tr><td>option 66 ascii "configs.nipt.telstra.com"
option 150 ip 203.52.0.206</td><td>option 66 ascii "dms.digitalbusiness.telstra.com"
option 150 ip 203.52.0.206 203.52.1.206
option 160 ascii</td></tr>
</table>

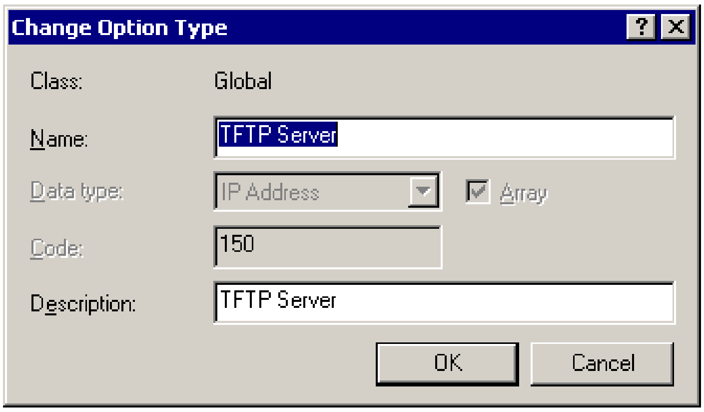
203.52.1.206	"http://polydms.digitalbusiness.telstra.com/dms/bootstrap"

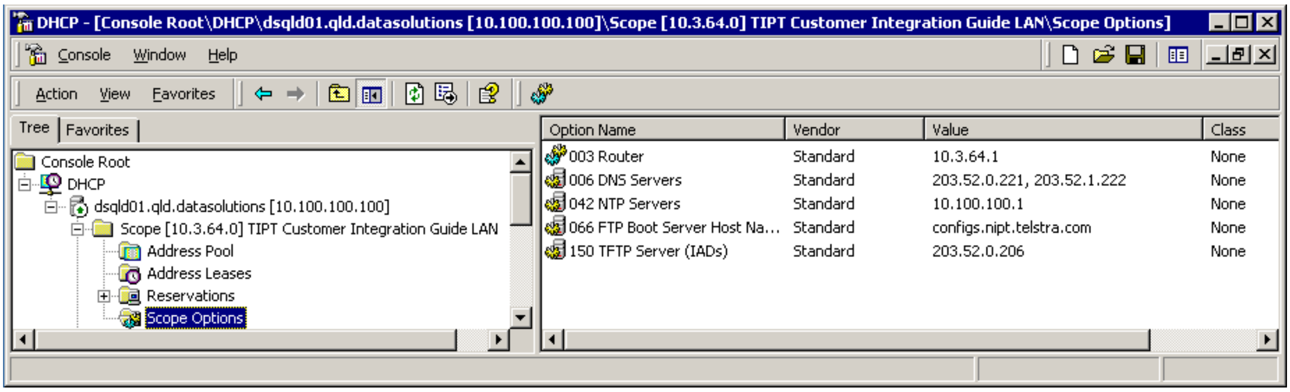
Table 7 – Cisco IOS DHCP Server Template (Voice VLANs)

2.10.5. Microsoft Windows DHCP server

The following Microsoft Windows 2000/2003/2008 DHCP Server (Voice VLAN) configuration template may be used.

Microsoft Windows DHCP Server Configuration

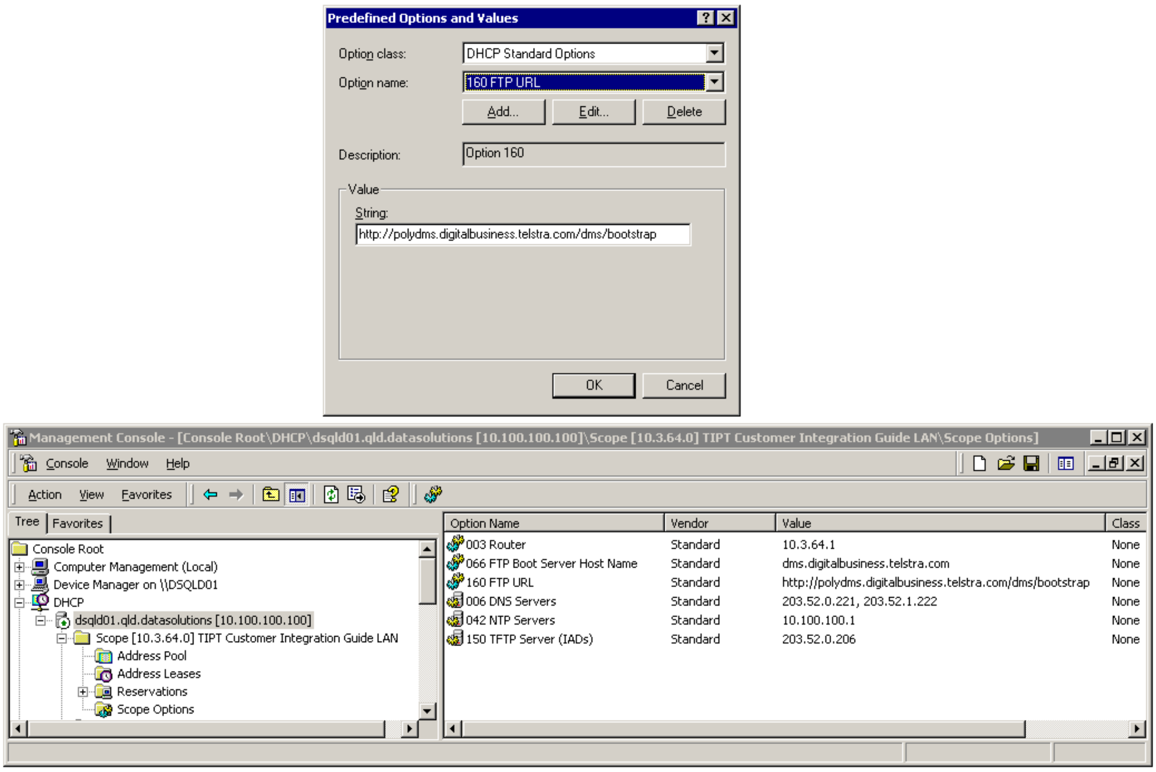




Microsoft Windows DHCP Server Template (Voice VLANs) - Hosted Configuration Server

Notes: DHCP Option 150 (IP Address - Array) must be added via “Set Predefined Options”.

Ensure the DHCP Option Data Type is set correctly.



Microsoft Windows DHCP Server Template (Voice VLANs) – Device Management Solution

> **Notes:** DHCP Option 150 (IP Address - Array) must be added via “Set Predefined Options”.
>
> Ensure the DHCP Option Data Type is set correctly.

2.11. DNS Server

2.11.1. Configuring the DNS server

If the customer has a premise based DNS server(s) it must be configured for DNS conditional forwarding for the “nipt.telstra.com” and “tipt.telstra.com” domains as per the table below.

This is required to utilise the TIPT applications including; Toolbar, miRECEPTION and CommPilot web. DNS conditional forwarding is supported by Windows 2003 but not Windows 2000 Server.

PARAMETER	CONFIGURATION
Forwarder – DNS Domain Name	**nipt.telstra.com** **tipt.telstra.com**
Primary Domain Forwarder	203.52.0.221
Secondary Domain Forwarder	203.52.1.222

Table 11 – Conditional DNS forwarding

2.11.2. Integrating the on-site DNS

If the customer cannot integrate its on-site DNS (e.g. Windows DNS 2000) then the integration is performed via existing Internet DNS referrals. If the customer selects this sub-optimal architecture then TIPT applications may be impacted by an Internet outage. IP Phones within the Voice VLAN are not affected.

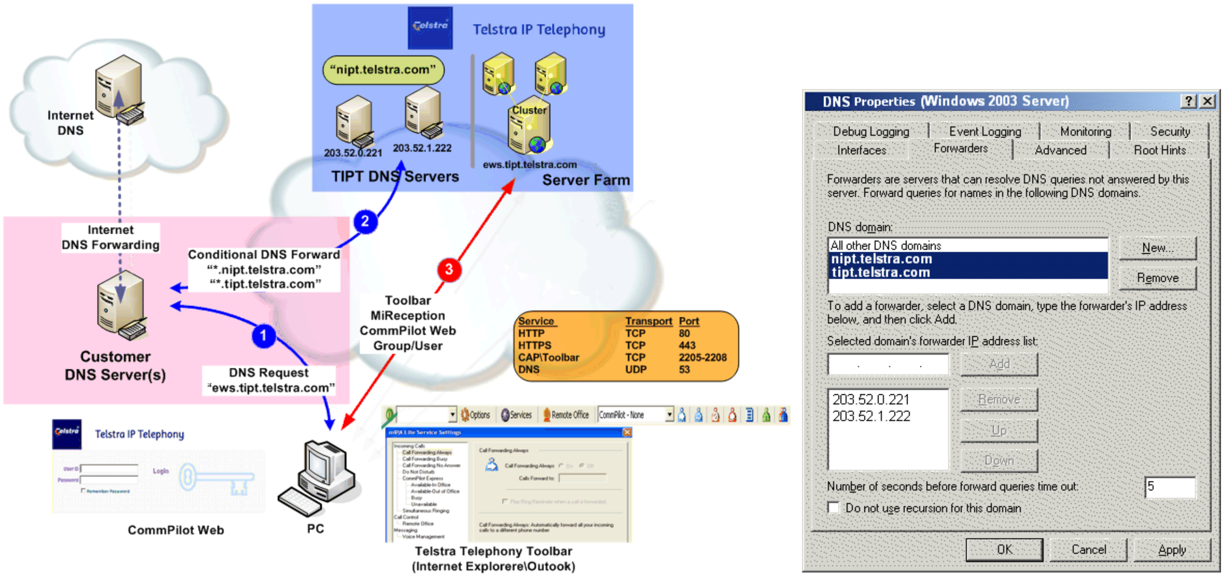


Figure 7 – DNS Server Configuration (Conditional DNS Forwarding)

2.11.3. DNS name value validation

The "**nslookup**" tool **must** be used from **both** the customers DNS server(s) and an on-site PC client to verify name resolution to the TIPT hosts as per below. The nslookup should also be performed on the Voice VLAN to ensure name resolution from the IP phone.

NAME RESOLUTION VALIDATION	WEB HOSTS
> **nslookup ews.tipt.telstra.com** Addresses : 144.140.208.81, 144.140.181.80 > **nslookup sbc-vic.nipt.telstra.com** Address: 203.52.0.167	Additionally customers must also be able to access the following hosts: https://ews.tipt.telstra.com **Note:** Access to these sites will only be available once interconnects are established.

Table 12 – DNS Name Resolution Validation ("nslookup")

2.12. Web Proxy (Caching) Server

2.12.1. Configuring the caching server

The customers Web Proxy server should be configured for http(s) cache bypass for all "*.nipt.telstra.com" and "*.tipt.telstra.com" web requests. Alternatively the customer can configure Internet Explorer as below.

If cache bypass is not configured the customer administrator may find that stale web pages are presented whilst trying to manage TIPT via CommPilot Web. This is especially critical where the customer is using an internal web proxy server to gain access to the Internet.

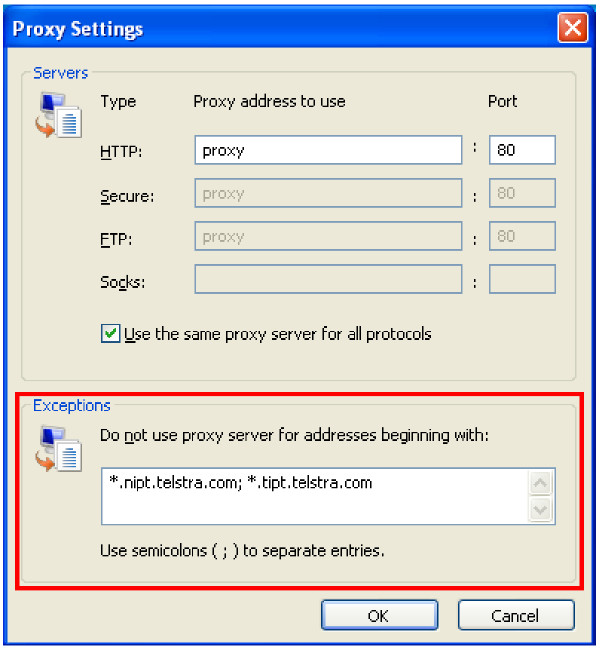


Figure 8 – Proxy Bypass Configuration (Internet Explorer)

2.13. Firewalls

2.13.1. TIPT and firewalls

The use of customer firewalls (or NAT devices) to limit Voice Traffic - SIP (signalling) and dynamic RTP (Media) is not recommended and should be avoided. VoIP firewall traversal has the potential to impact performance (voice quality due to additional network delay) and service quality.

The following table provides a list of the protocols and ports used by TIPT.

SERVICE	PROTOCOL	PORT	DESCRIPTION
SIP	UDP	5060	Signalling protocol used by IP Handsets, video endpoints and IADs.
RTP	UDP	Dynamic	Real-time Transport Protocol (Media) used to deliver audio between VoIP end-points.
RTCP	UDP	Dynamic	Real-time Transport Control Protocol used to provide QoS status to end-points.
DNS	UDP	53	Used for Name Resolution via TIPT DNS servers.
CAP	TCP	2205-2208	Toolbar and miRECEPTION desktop application use the Client Access Protocol for call control.
HTTP(s)	TCP	80, 443	CommPilot Web use HTTP and SSL for TIPT administration. BRIX and the TIPT Online Resource Centre also use HTTP(s) Phone configuration files for the Device Management Solution
FTP	TCP	20-21	File Transfer Protocol. Polycom CPE use FTP to download firmware and device configuration files (MAC Address) using the

			Hosted Configuration Server solution.
TFTP	UDP	69	Trivial File Transfer Protocol. Linksys IADs and Cisco IP Phones use TFTP to download firmware and device configuration files (MAC Address) using the Hosted Configuration Server solution.
NTP	UDP	123	Network Time Protocol. Handsets will obtain their NTP time source via a specified synchronised time source
Adobe Flash	TCP	843	Used to manage incoming media for Adobe flash
SOCKS-5	TCP	1081	File Transfers within UC-One
XMPP	TCP	5222	Instant Messaging and Presence within UC-One
Bosh over HTTP/HTTPS	TCP	5280/5281	Instant Messaging and Presence within UC-One
XMPP	TCP	5269	XSI Federation for 3rd party access
HTTPS	TCP	8443	Desktop Share within UC One

Table 83 – TIPT Ports and Protocols

2.14. Quality of Service (QoS)

2.14.1. Introduction

Quality of Service (QoS) is the ability to provide differential levels of treatment to specific classes of traffic. This traffic, being voice, video or data, must be identified and sorted into different classes to which differential treatment is applied.

Implementation of a network wide QoS design on site switches and routers is proposed to ensure that voice receives prioritisation within the Local Area Network and that network congestion is obviated through the implementation of techniques such as “Modular QoS”.

2.14.2. Class of Service

Implementation of Telstra’s IP MAN/IP WAN Dynamic CoS will ensure that voice receives prioritisation within the Wide Area Network and that network congestion is obviated through the implementation of techniques such as “Egress Queuing”.

- A Quality of Service Policy will address the key issues of:
- Classification and Marking
- Congestion Management
- Congestion Avoidance
- Traffic Policing and Shaping, and
- Link Efficiency.

In order to provide an enterprise grade of service for TIPT is required that the customer’s network support end-to-end QoS. The WAN edge router must be configured to categorise and prioritise TIPT traffic accordingly.

DIFFERENTIATED SERVICE (DSCP)	CLASS OF SERVICE (COS)	VALUE	QUEUING METHOD
EF	5	RTP Audio	Bandwidth allocation Strict Priority Queuing (LLQ)
AF (31) (Video devices – eg VVX1500)	3	RTP Video H.264	Bandwidth %
CS3	3	SIP (Signalling), RTCP	Bandwidth %
BE (0)	0	Other (that is: CAP, DNS, HTTP(s), FTP , TFTP)	

Table 14 – QoS Configuration (WAN Router)

2.15. CPE Device Configuration Delivery

2.15.1. Hosted configuration server

The Hosted configuration server provides a network delivered T/FTP server for the delivery of IP handset configuration data and firmware.

Polycom IP Phones load their configuration from the network Hosted server using the FTP protocol. Polycom devices locate the Boot Server via DHCP options from the DHCP Server (Voice VLAN).

Polycom device configurations are generated automatically provided that the correct Device Type and MAC address are assigned in the TIPT platform. MAC Addresses must be unique and entered as a 12 digit HEX number only (that is: 0011abcd1234).

2.15.2. Device Management Solution

In April 2011, a new Device Management Solution will be used instead of the Hosted Configuration Server for all new customers/enterprises, and all new sites/groups for existing customers will use this system.

The DHCP options and protocols used are different than the previous Hosted Configuration Server solution (as detailed in section 2.2.6). This solution is "internet ready", with the use of HTTPS for security. The Device Management Solution will also deliver enhanced firmware management capability and new user features.

The other key change brought about by DMS, is that provisioning/activation of devices is based on user name and password, rather than MAC address. This provides more flexibility for provisioning, upgrading and replacing phones. You will be advised of your Username and Password from your Customer Group Administrator. The username will be your phone number.

2.15.3. Phone install procedure with QSetup

If you are required to initially configure a brand new Polycom IP phone you will see a QSetup soft key on the screen on your phone. Pressing the QSetup soft key takes the user directly to the menu where the supplied username and password for that device needs to be entered. Once the username and password have been entered the user will be prompted to save the new configuration.

If a phone has already been previously configured (with user name and password) via Qsetup, the QSetup soft key will not be displayed on the screen when the phone is powered up – eg after the phone has been unplugged or otherwise experienced a power supply interuption.

At the introduction of DMS, all existing Polycom phones are supported (ie 30X / 430 / 50X / 550 / 560 / 60X / 650 / 670 / VVX1500 / 4000 / 6000 / 7000), although older phones models IP301, IP430, IP501, IP600, IP601 and IP4000 do not support the QSetup soft key. The IP300 and IP500 Polycoms are NOT supported by the DMS solution and will be swapped out as part of the DMS migration process. Detailed step by step instructions for each phone model can be found at :

http://www.telstraenterprise.com/support/tiptresources/Pages/PhoneUsers.aspx

2.15.4. Install of Video Conferencing endpoints

For Polycom Group Series video endpoints, the install procedure does NOT use QSetup but a Telstra scheduler/co-ordinator will be in contact to schedule an appointment for a Telstra technician to visit the site in order to install and configure the video endpoint.

It’s very important prior to the technician appointment that the customer network (eg router, switch, VLAN, etc.) is ready or has been configured in preparation prior to the video endpoint installation. Please refer to section 2 of this document on customer integration and preparation of the network prior to the video endpoint installation.

2.15.5. Adds, Moves and Changes

Using the CommPilot web portal a suitably trained customer administrator can perform their own Add Moves and Changes (AMCs). Alternatively this function can be performed by the Telstra Managed Voice Service (MVS) Help Desk if that service has been purchased.

2.16. IP Handsets –Polycom IP Phones

2.16.1. Available models

There are seven models of TIPT IP phones and two IP conference phone that can be deployed. The Polycom IP550 phone is the recommended standard office telephony phone type.

Polycom VVX1500 (6 Line IP Video Phone – VVX is GigE)

Polycom IP670 (6 Line IP Phone – expands to 12 lines with Expansion Module - IP670 is GigE)

Polycom IP650 (6 Line IP Phone – expands to 12 lines with Expansion Module)

Polycom IP550/560 (4 Line IP Phone – IP560 is GigE)

Polycom IP450 (3 Line IP Phone)

Polycom IP331 (2 Line IP Phone)

Polycom IP235T (1 Line IP Phone)

Polycom IP7000 Conference Phone

Polycom IP6000 Conference Phone

Figure 9 – Polycom IP and Conference Phones

2.16.2. Polycom configurations and diagnostics

Polycom IP Handsets obtain their configurations from the either the Hosted Configuration or Device Management Solution Servers. Configured Polycom IP phones will display a solid black phone icon next to the phone number indicating that the phone is successfully configured and registered against the TIPT platform.

The Polycom IP phone has a diagnostic menu that can be used to verify the phones' audio quality and keypad functions

Figure 10 – Polycom IP Phone Display (Registered Phone)

2.16.3. VLANID

The VLANID is set as per section 2.7 VLAN Allocation

2.16.4. Video Conferencing (VC) endpoints

There are two models of TIPT VC CPE endpoints that can be deployed. The Polycom Group Series 300 and 500.

Polycom RPG 300 (VC endpoint)

Polycom RPG 500 (VC endpoint)

Figure 11.1 – Polycom Group Series (VC CPE endpoints)

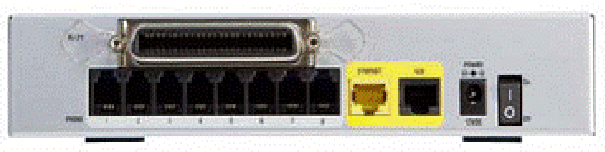
2.17. Integrated Access Devices – Linksys IADs

2.17.1. Configuring the Linksys IAD

The Linksys SPA2102 and SPA8000 IADs obtain their configuration from the Hosted Configuration or Device Management Solution servers. Configured IADs will present dial tone via an analogue handset.

The LAN Switch port to which the IAD is connected must be programmed in the Voice VLAN (e.g. VLAN 100). Also, the SPA2102 must be connected to the LAN switch using the "Internet" port (Blue colour port) and not the port marked Ethernet.

> **NOTE:** If the IAD is connected with the wrong port, this will stop IP phones from working as the IAD will act as a DHCP server providing IP subnet network addresses from the 192.168.0.0/24 range.



Linksys SPA8000

Linksys SPA2102

Figure 12 – Linksys IADs

2.18. Soft Client – Mobile and PC – UC-One

2.18.1. Connectivity Requirements for Desktop Client

The Desktop device that will be running UC-One will need to have the following URL's resolvable and connectivity to

XSP:

- https://cmanager.tipt.telstra.com/uc-one/pc
- https://xsi-actions.tipt.telstra.com

SBC

- The SBC that the customers CPE points to.

2.18.2. Connectivity Requirements for Mobile Client

The mobile device that will be running UC-One will need to have the following URL's resolvable and connectivity to as above except they will also need this additional address resolvable:

- https://cmanager.tipt.telstra.com/uc-one/mobile

SBC

- The SBC that the customers CPE points to.

This will need to be checked via the customers APN, Office Wi-Fi or VPN if the customer wishes to deploy such a solution.

2.18.3. Port Requirements

SERVICE	PORT
Signalling	5060
RTP for Voice	16384 - 32767
RTP for Video	8600 - 8698
HTTPS/HTTP	443/80
SOCKS-5	1081
XMPP	5222
Bosh over HTTP/HTTPS	5280/5281
XMPP	5269
HTTPS	8443

2.18.4. Quality of service for UC-One clients (QoS)

It must be noted that there is No Guarantee of Voice Quality on calls via UC-One, however Implementation of QOS at a site may assist to mitigate network performance issues that may be experienced with voice/video quality.

2.19. XSI Call Control

Any Client using XSI Call Control will need access to the XSI Interfaces:

- https://xsi-actions.tipt.telstra.com/com.broadsoft.xsi-actions/test
- https://xsi-events.tipt.telstra.com/com.broadsoft.xsi-events/test

2.19.1. Port Requirements

SERVICE	PORT
HTTPS	443

3. BRIX Verifier and Reporting

3.1. Introduction

The Brix BV-100M verifier is a Telstra device that gathers and provides service assurance information to Telstra and the customer. A BRIX BV-100M is deployed at TIPT sites with greater than 30 handsets and must be connected via the Voice VLAN. The customer is required to provide a dedicated switch port which is configured as an access port in the Voice VLAN.

The verifier measures essential network and service performance information that is complied into end-to-end performance reports that are used by Telstra to support maintenance and fault diagnostic functions.

Note: The BRIX verifier is not part of the Connect IP Tel offering

3.1.1. Operational status

Patch the LAN/Test port to the configured switch port.

A solid GREEN status LED indicates the device is operational and communicating with the Telstra BRIX platform.

3.1.2. Pre-commissioning requirement

Once operational the BRIX verifier must reside in the Voice VLAN.

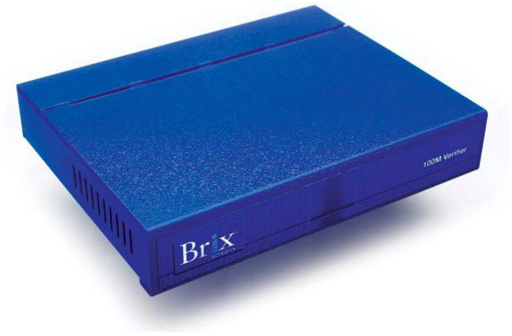


Figure 13 – Brix 100M (Service Assurance Verifier)

4. SIP Ping (SipSak Testing Tool)

4.1. Introduction

The SipSak utility provides an invaluable pre-commissioning mechanism to test SIP connectivity to the TIPT platform by validating access to each Media POP.

4.2. Downloading and running SipSak

SipSak can be downloaded from: http://www.iptel.org/download

> **Note:** Should be run from a command prompt using the specific batch files provided (for example: QLD-SBC.bat).

4.3. Using SIP Ping

The following output illustrates a successful SIP ping using the provided “<STATE>-SBC.bat” batch file.

> **Note:** This is not mandatory and is required only for the first site within each state

SIPSAK (SIP PING – TESTING TOOL)
C:\ > **QLD-SBC.bat** (Note: Use the correct batch file based on your site’s location) **using A record: sbc-qld.nipt.telstra.com** message received: SIP/2.0 200 OK Via: SIP/2.0/UDP 192.168.200.32:1421;branch=z9hG4bK.2d5d737b;alias;rport=1422 From: <sip:sipsak@192.168.200.32:1421>;tag=4c590013 To: sip:ping@as.nipt.telstra.com Call-ID: 1280901139@192.168.200.32 CSeq: 1 OPTIONS Accept: application/sdp,application/broadsoft,text/plain

```
Allow: ACK,BYE,CANCEL,INFO,INVITE,MESSAGE,OPTIONS,PRACK,REFER,REGISTER,SUBSCRIBE,
NOTIFY,UPDATE
Supported: 100rel
Content-Length: 0
```

**** reply received after 88.000 ms ****
SIP/2.0 200 OK
final received

4.4. Troubleshooting

Failure of the SIP PING (SipSak) tests may indicate:

- No IP Route to TIPT Interconnect (perform a traceroute to the platform)
- No Name Resolution (check DNS using a nslookup)

Firewall or Access-list blocking SIP access to platform

5. Floor Plan (Professional Installation)

5.1. Preparing for installation

The customer must provide a detailed floor plan or alternatively the customer can locate IP phone boxes (pre-labelled) on the user's desk prior to CPE deployment (to ensure correct handset placement).

The floor plan must accurately map user TIPT extensions (TIPT phone number) to location as per the diagram below. During IP phone installation PC's will be disconnected from the network then daisy-chained into the "PC" port of the Polycom handset.

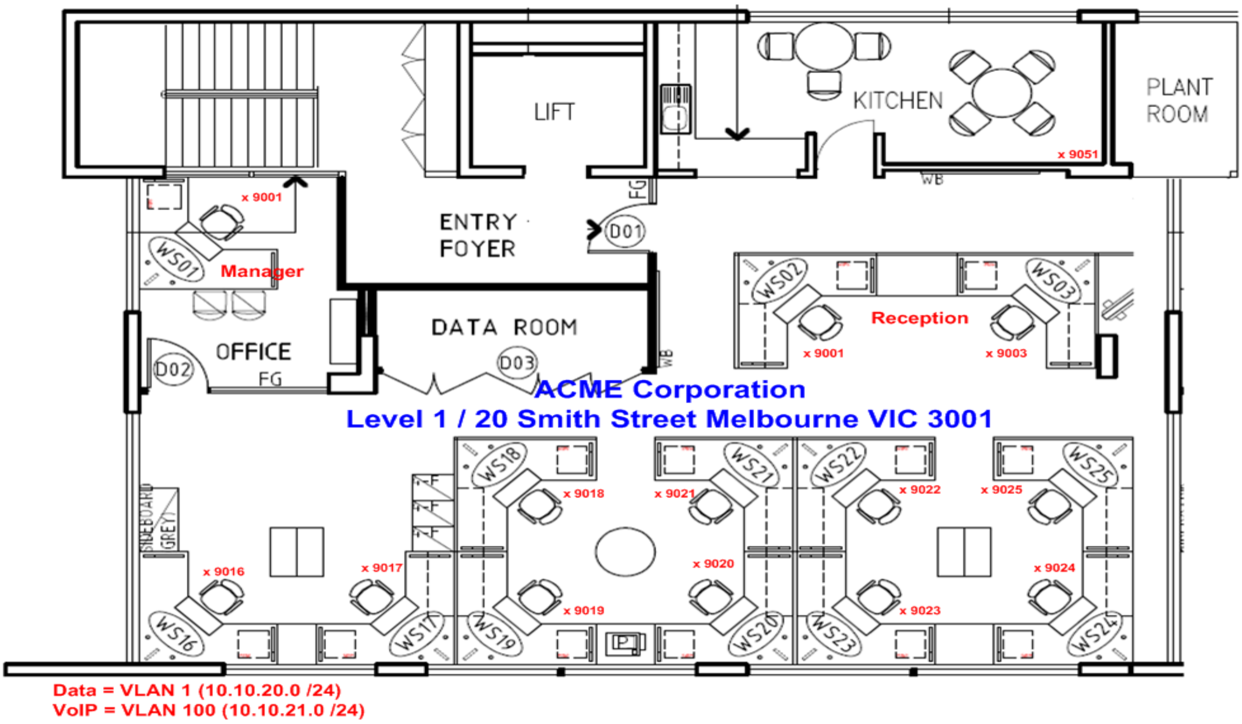


Figure 14 – Detailed Floor Plan (IP Handset Installation)

6. Desktop Integration

6.1. Introduction

The following sections provide customer desktop integration information to support the TIPT solution. This guide is not intended to detail customer specific requirements. Refer to Telstra specific configuration, that is: Customer Design. TIPT software should be validated within the customer's environment prior to deployment.

6.2. Telstra Telephony Toolbar (TTT)

6.2.1. Downloading the TTT

The Telstra Telephony Toolbar software can be downloaded from the TIPT Resource Centre:

http://insight.telstra.com.au/t5/Downloads/TIPT-Application-Downloads/ta-p/129

> **Note:** Access to these sites will only be available once interconnects are established.

6.2.2. Installation, Integration and deployment

The Toolbar supports Internet Explorer and Microsoft Outlook only.

> **Note**: Outlook Contacts integration is only available in the Executive feature pack.

Toolbar requires desktop administrative access to install. All user applications must be closed before installation of the toolbar application. The toolbar must be installed and configured prior to training.

Desktop deployment (including interoperability testing) and configuration of the Toolbar is the responsibility of the customer. Toolbar configuration requires both the case-sensitive username, for example: FNN@domainname.com.au and password to be specified.

Refer to Toolbar installation and configuration instructions at:

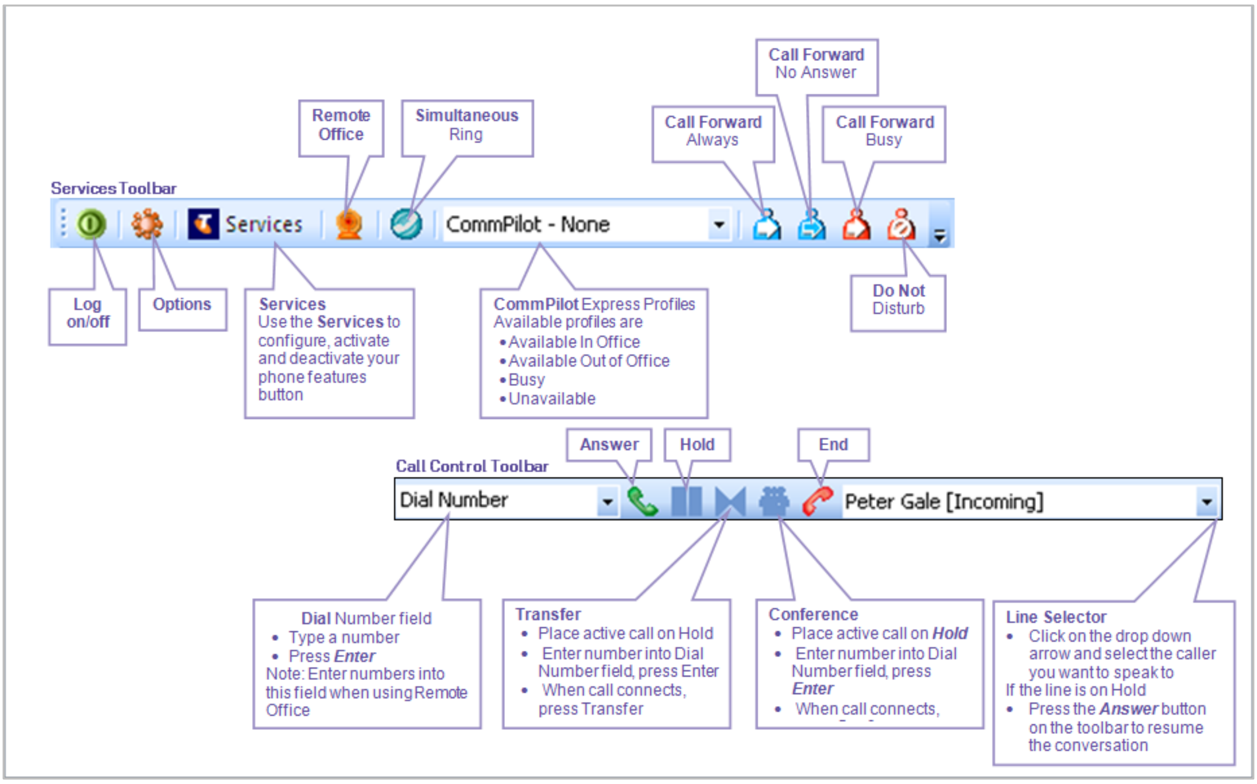
http://www.telstra.com.au/business-enterprise/download/document/business-telephony-toolbar-install-uninstall-reference-guide.pdf

6.2.3. TTT features

The following Quick Reference guide describes the features of the Telstra Telephony Toolbar. Full details are available from:

http://www.telstra.com.au/business-enterprise/download/document/business-telephony-toolbar-user-guide.pdf

> **Note:** Access to these sites will only be available once interconnects are established.



6.2.4. TTT Tags

Existing Web directories can be easily enhanced to support Toolbar click to call and click to blind transfer using the following HTML Toolbar tags as per the sample below.

- To dial a number use the following as a Hyperlink “**TT:<PhoneNumber>?Dial**”
- To perform a blind transfer whilst on a call use the following as a Hyperlink “**TT:<PhoneNumber>? BlndTransfer**”

SERVICE	NUMBER	DIAL	BLIND TRANSFER
Telstra Time Service	1194		
Telstra Weather Service	1196		
TAXI	131 008		

Table 17 – Toolbar Smart Tags (Sample)

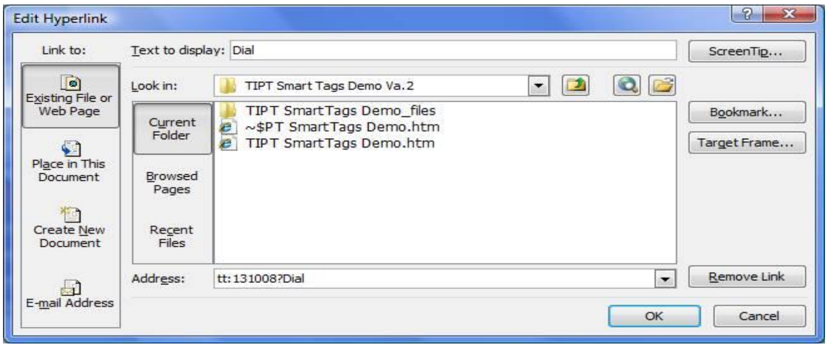


Figure 15 – Toolbar Smart Tags

6.3. miRECEPTION

6.3.1. Requirements

The miRECEPTION application, which requires the optional miRECEPTION feature pack, is used in conjunction with a Polycom IP Phone to provide a reception console solution. Additionally calls can be queued through the use of the optional Call Centre pack.

The miRECEPTION solution requires JAVA which is required to be downloaded.

The miRECEPTION application requires a valid login that must be validated prior to training.

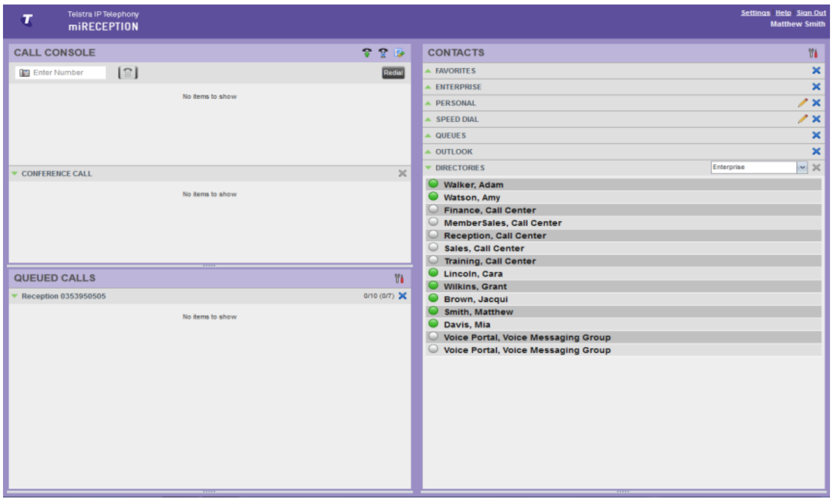


Figure 16 - miRECEPTION Console

6.3.2. Installation and configuration instructions

Refer to the links below for installation & configuration instructions;

http://insight.telstra.com.au/t5/Downloads/TIPT-Application-Downloads/ta-p/129

> **Note:** Access to these sites will only be available once interconnects are established.

7. Miscellaneous Integration

7.1. Introduction

The following sections provide the customer with additional information in relation is miscellaneous integration components.

7.2. Music on Hold (MoH)

7.2.1. Music on Hold Customisation

The customer administrator can load a static WAV file as their Music on Hold (MoH). The WAV file must be encoded as G711 A-law or U-law G711 (A-law is preferred). The wav file must be 8KHz, 8 bit, mono.

The WAV file should not be more than 10 minutes duration (5 MB in size).

Music on Hold can be customised at the group (default) or department level and is loaded via the Custom Music File option. Please note Video on Hold is not supported.

8.

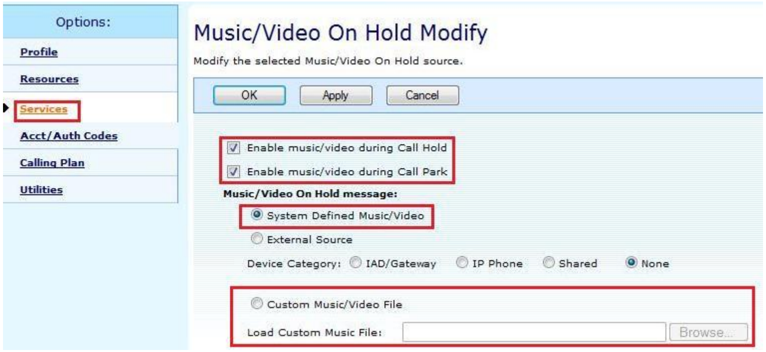


Figure 17 – Music on Hold Customisation

8.1. Headsets

Not all existing headsets work with IP phones. Telstra does not support or formally recommend TIPT headsets. However a list of Polycom approved headsets may be found at;

http://support.polycom.com/global/documents/support/technical/products/voice/Headset_Compatibility_List_SoundPoint_IP_TB37477.pdf?elq=8c9d484767d74a92a541c34f9770e898

Sennheiser link:

http://www.sennheiserusa.com/cordless-wireless-headsets-office

Jabra link:

http://www.jabra.com/au-cp/unified-communications/pages/discoverproducts.aspx

8.2. Customer Logo Placement on Polycom Handsets: Logo File Specifications

You may elect to have a Logo displayed on some Polycom handsets. This table is a guide to providing the correct file type and format.   Although logos smaller than those listed in the table are supported, larger logos will be truncated and may interfere with areas of the user interface.

Model	File Type	Width (Pixels)	Height (Pixels)	Colour Depth
SoundPoint IP 450	BMP	170	73	4-bit grayscale or monochrome
SoundPoint IP 550/560/650	BMP	213	111	4-bit grayscale or monochrome
SoundPoint IP 670	BMP	213	111	12-bit colour

SoundPoint IP 7000	BMP	255	75	32-bit grayscale or monochrome
VVX1500	JPG	800	400	Full colour

Table 17 – Customer Logo File Specifications

7.5 LDAP Integration

The TIPT platform can integrate into the directory using LDAP. TO make this change happen, you will need to provide us with the details that will allow our platform to communicate with your integration server, and will approximate the following list.

Directory address – LDAP server name e.g. ldapvic1.core.dir.telstra.com

Port – LDAP Port number e.g. 389

Encrypt Connection – SSL connection required (Yes/ No)

Use Authentication – (Yes/ No)

Authentication Name – username with access privilege to the LDAP server e.g. CN=n010428,OU=Non User Accounts,OU=Administration,DC=core,DC=dir,DC=telstra,DC=com

Password – password for the above user id.

Search base – the directory name against which to search the keyword/ contacts for e.g.

OU=People,OU=eProfile,DC=core,DC=dir,DC=telstra,DC=com

ecursive – whether a recursive search must be performed (– yes/no)

Search filter – the keyword will be matched against the specified filter e.g.

cn=*__SEARCH_TEXT__*)

Retrieved attribute – the result of a successful query will display these fields

e.g. cn,sn,dn,telephoneNumber,mobile,homePhone,mail

9.

10. Reception Solutions

10.1. Simple Reception Solution

The simplest solution for a small reception environment utilises a Polycom 650 IP phone with an optional Backlit Expansion Module (BEM). A Hunt Group is used to present an inbound pilot number to incoming callers. Using this cost effective solution the receptionist can manage up to a maximum of 6 (using Multi-line Appearance) concurrent calls via the handset. The receptionist transfers calls using the IP phone as shown below.

This solution is intended for basic reception environments with low call volumes and without the need for extension presence (i.e. extension monitoring).

The BEM provides additional on phone speed-dials (14 per BEM) to allow the operator to manage calls more efficiently and to provide diversions using simple Feature Access Codes.

In higher call volume environments the Hunt Group can be replaced with an optional Call Centre service which is used to queue calls in the network. The Call Centre provides an entrance and periodic comfort message with Music on Hold. The Hunt Group or Call Centre can be diverted using Call Forward Always (CFA) in the occurrence of a Disaster Recovery situation.

After hours diversion is performed automatically using the Call Forward Selective Feature in conjunction with a defined time schedule or manually using a Feature Access Code (FAC) to divert all calls to either Voice Messaging or another number.

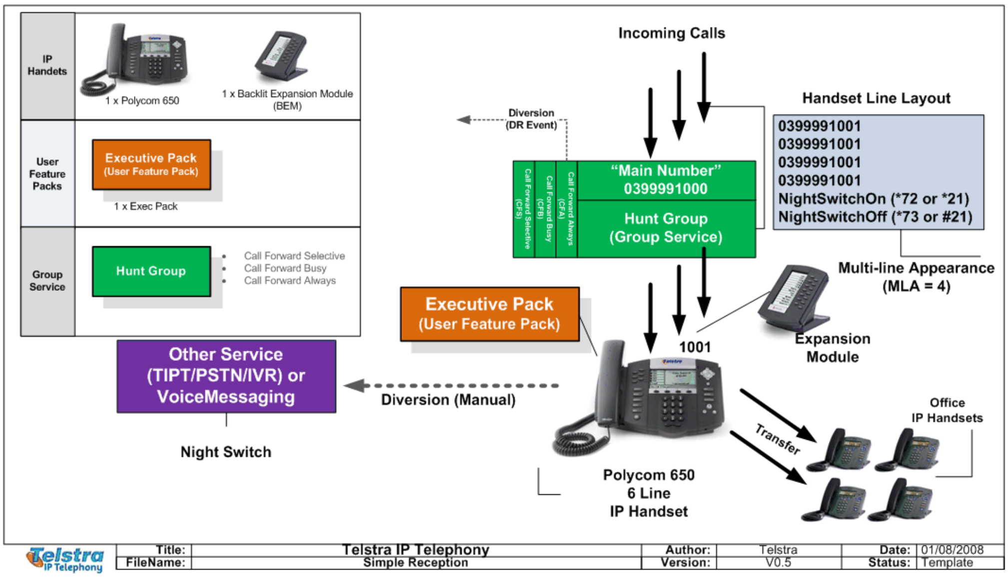


Figure 187 – Simple Reception Solution

10.2. Premium Reception Solution

The best solution for a typical reception environment utilises a miRECEPTION application in conjunction with a Polycom 650 IP phone with a Backlit Expansion Module (BEM) and Executive User Pack. A Call Centre (or Hunt Group) is used to present a pilot number and the receptionist can handle up to 6 concurrent calls. The receptionist transfers calls using either the miRECEPTION application or IP phone as shown below.

The solution also allows for an Auto-Attendant (IVR) to front the Call Centre to provide predefined menu options via a customisable Business Hours or After Hours simple IVR menu.

This solution is intended for typical receptionist environments with high call volumes and where extension monitoring (using miRECEPTION) is required. Additional miRECEPTION applications (for additional reception users) can be added to support multi-application environments.

After hours diversion can be performed automatically via the Auto Attendant or manually using the miRECEPTION application or via Feature Access Codes to divert all calls to either Voice-Messaging or another number.

The BEM provides additional on phone speed-dials (14 per BEM) to allow the operator to manage calls more efficiently and to provide diversions using simple Feature Access Codes.

In lower call volume environments the Call Centre can be downgraded to a Hunt Group. The advantage of the Call Centre is that is used to queue calls in the network without consuming resources within the customer network. The Call Centre provides an entrance and periodic comfort message with customisable Music on Hold. The Hunt Group or Call Centre can be diverted using Call Forward Always (CFA) in the occurrence of a Disaster Recovery situation.

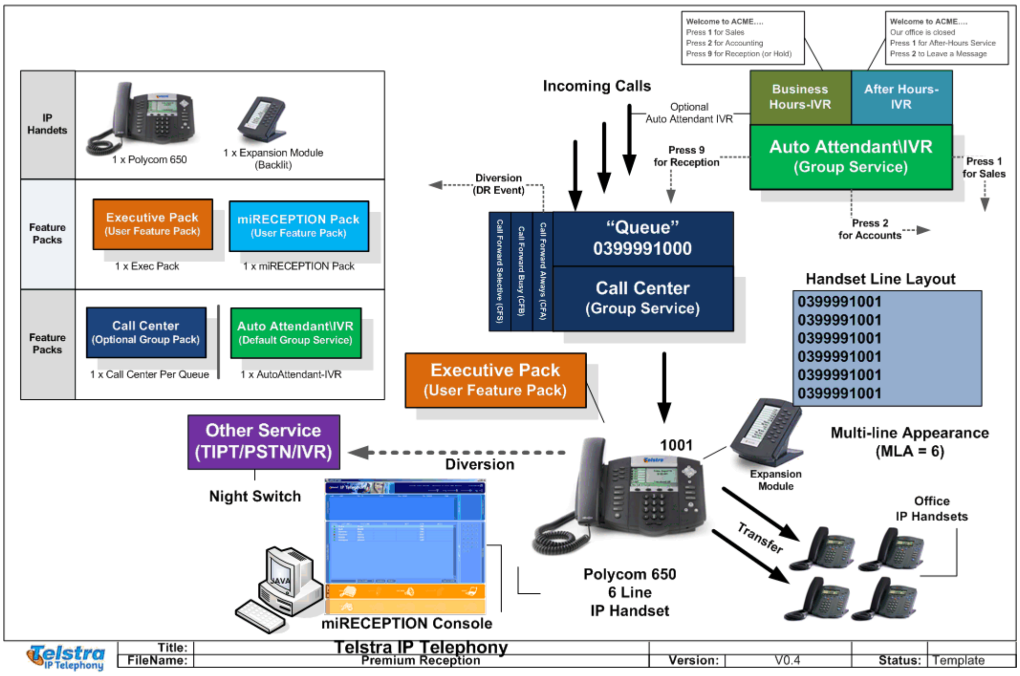


Figure 198 – Premium Reception Solution

11. Appendices

11.1. Appendix A: Glossary

The following terms, acronyms and abbreviations are referred to in this document.

TERM	DEFINITION
BEM	Backlit Expansion Module
CAP	Client Access Protocol
CDP	Cisco Discovery Protocol
CIP	Connect IP
CoS	Class of Service
DHCP	Dynamic Host Configuration Protocol
DMS	Device Management Solution
DNS	Domain Name Services
FTP	File Transfer Protocol
HTTP	Hyper Text Transfer Protocol
HTTPS	Hyper Text Transfer Protocol Secure
IAD	Integrated Access Device

IP	Internet Protocol
IP MAN	Internet Protocol Metropolitan Area Network
IP WAN	Internet Protocol Wide Area Network
LAN	Local Area Network
LLDP	Link Layer Discovery Protocol
MWAN	Managed Wide Area Network
NIPT	Network Internet Protocol Telephony
NTP	Network Time Protocol
PABX	Private Automated Branch Exchange
PBX	Private Branch Exchange
PoE	Power over Ethernet
POP	Point Of Presence
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTCP	Real Time Transport Control Protocol
RTP	Real Time Transport Protocol
SIP	Session Initiation Protocol
TFTP	Trivial File Transfer Protocol
TIPT	Telstra IP Telephony
VLAN	Virtual Local Area Network
VoIP	Voice over Internet Protocol
VPN	Virtual Private Network
WAN	Wide Area Network

11.2. Appendix B: TIPT CPE Troubleshooting & Quick Tips

11.2.1. CPE configuration & troubleshooting

This section provides additional detailed steps to ensure that the CPE is configured correctly within the customer's environment. Additionally this section also serves as an advanced CPE troubleshooting section.

<table>
<tr><th>PARAMETER</th><th>CONFIGURATION</th><th>STEPS</th></tr>
<tr><td>Service Activation</td><td>Confirm that you can make both inbound and outbound calls.

For a new TIPT site the “Intercept Group Service” must be turned off by Telstra to activate services at each site.</td><td>If the “This line has been placed out of service…” RVA is played out then contact the Telstra PM to get the Intercept service removed.</td></tr>
<tr><td>VLAN</td><td>Confirm VLAN settings on IP handset are correct.

(that is: Voice VLAN = 100)

Ensure you can get an IP address within the Voice VLAN. Connecting a PC to the Voice VLAN and performing an ipconfig /all from the command line will indicate if DHCP is functional.</td><td>Polycom:

At IP Phone start hold down the “About” soft key and verify VLAN ID is correct.

See section2.2.3 for details.</td></tr>
<tr><td>Hosted Configuration Server and Device Management Solution</td><td>Confirm CPE is obtaining configuration.</td><td>Polycom:

After the phone has booted verify the configuration files are loaded;

“MENU > Status > Platform > Configuration> “

If the phone fails to contact the boot server, the phone will display a “fail to contact boot server” error message. This would suggest connectivity issue. Under DMS, lack of NTP may also be a cause of boot failure.</td></tr>
<tr><td>Power over Ethernet</td><td>IEEE 802.31af PoE must be used.
<table>
<tr><th>IEEE CLASS</th><th>MAX. POWER OUTPUT</th></tr>
<tr><td>0</td><td>15.4 W</td></tr>
<tr><td>1</td><td>4.0 W</td></tr>
<tr><td>2</td><td>7.0 W</td></tr>
<tr><td>3</td><td>15.4 W</td></tr>
</table>
Polycom 301 & 501 requires the use of PoE cable (which is keyed).</td><td>Install all CPE to ensure LAN switch has sufficient power budget to support all CPE. A symptom of low PoE budget is that newly deployed IP phones will not power up.

Use diagnostic commands on switch to verify PoE consumption that is: ;

CISCO LAN Switch:

show power inline</td></tr>
</table>

PARAMETER	CONFIGURATION	STEPS
DNS Settings	Confirm DNS settings on IP handset are correct. IP handsets must be using the TIPT and not the customers DNS Non-compliance may result in post-dial delays.	Polycom: "MENU > Status > Network … > TCP/IP Parameters…> " DNS Server: 203.52.0.221 DNS Alt. Server: 203.52.1.222
Messages Button (Voice Portal)	Confirm the IP Phones messages button is configured for the TIPT Voice Portal at each site.	Press the "Messages" button on IP phone and a "Welcome to your CommPilot Voice Portal…" RVA will be played out. A PIN is required to access the service.
VoIP Codec	Confirm VoIP Codec as either G711 (Uncompressed) or G729 (Compressed) as per customer requirements.	Polycom: During an active call (e.g. Voice Portal) press "MENU > Status > Network … > Packet Statistics…> Call>" Tx Codec: G.729AB Rx Codec: G.729AB
Voice Quality	The Polycom phone provides basic QoS performance feedback on the handset. Ideally the Network Jitter should be less than 50 ms (maximum) and packet loss should be minimal. Whilst the end-to-end delay should be less than 150-200 ms.	Polycom: During an active call (e.g. Voice Portal) press "MENU > Status > Network … > Packet Statistics…> Call>" Lost Packets: 0 (Small Number) Jitter (ms): 7 Max. Jitter (ms): 25
MAC Address for IAD's	The IAD's obtain their configuration using the MAC address of the device.	Note: If swapping an IAD out, the MAC address will need to be updated in CommPilot by the Customer Administrator.
MAC Address for Phones	For Phone configs delivered over Hosted Config Server MAC address identifies the config for the phone to load.	Polycom: The MAC Address (Serial Number) can be found by pressing "MENU 2-2-2" to obtain the 12 digit HEX MAC address (e.g. 004f201ABCD)

PARAMETER	CONFIGURATION	STEPS
Time (NTP)	Confirm that the phone is getting correct time from the local simple NTP (Network Time Protocol) server and that the NTP server has the correct time and is synchronised (that is: upstream time server). The NTP server is provided to the Polycom Handset using DHCP Option 42, confirm that NTP source can actually be reached (there are third party tools available on the web to test if required).	Polycom: “MENU > Status > Network Settings… > TCP/IP Parameters…> “ SNTP Address: <192.168.100.1> GMT Offset: <36000(10.0)> If the phone or NTP server is not synchronised the phone will display a date of “JAN 01”
IADs	Ensure the Integrated Access Device (IAD) is plugged into the Voice VLAN (the switch port will need to be configured as untagged). Ensure the IAD is connected via the Internet port (and not the network port).	The Linksys IADs do not support the piggy-backing of a PC on the network port. To enter Interactive Voice Response Menu Press ****. Do not press any other keys until you hear, “Linksys configuration menu. To check the devices IP Address press **110#**
Factory Reset	Before returning a faulty IP handset perform a *Device Reset* to ensure customer details are removed from the handset.	Polycom: Press and hold “**468*”** and enter 456 as the password to perform a device reset. Press and hold “**1357”** and enter 456 as the password to perform a device reset (Polycom 330). Press and hold “**68***” and enter 456 as the password to perform a device reset (Polycom 4000. Please ensure that CDP/LLDP is enabled and that the correct VLANID is used. This is required in order for the handset to locate the network.

Table 18 – CPE Testing & Troubleshooting

11.2.2. CPE quick tips

This section provides a number of quick tips to assist in the setup of common parameters.

PARAMETER	CONFIGURATION	TIPS
Music on Hold	MoH can be customised by loading of a G711 A-law encoded wav file by the Customer Administrator. MoH can be loaded at the Group and Department level. Recommended MoH audio duration should not exceed 10 minutes.	Audacity can be used to encode MoH and change volume levels. Audacity is free cross-platform sound editor can be downloaded from: http://audacity.sourceforge.net
Commonly Used User Features	The commonly used User Features are: Call Forward Always (CFA) Call Forward No Answer (CFNA) Call Forward Busy (CFB) Call Waiting (On/Off) Voice Messaging (On/Off)	These features are configured via Toolbar or CommPilot Web
Power User Features	The Power User Features are: SimRing Remote Office CommPilot Express Profiles Available - In Office Available - Out of Office Busy Unavailable	These features are configured via Toolbar or CommPilot Web

Table 19 – TIPT Quick Tips

12. Document Control Sheet

Contact for Enquiries and Proposed Changes

If you have any questions suggestion for improving this document, contact by e-mail:

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Revision History of Document

Version No	**Version Date**	**Nature of Change**	**Author of change**
Baseline Document	19/06/2014	There have been numerous versions of this document which have not been captured in this revision history	N/A
14 0	20/06/2014	Addition of Document Control Sheet. Addition of file specifications for customer logos.	Mark Hammond
14.1	3/09/2014	Addition of Polycom Group Series for TIPT UC Video	TJ / WT