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Dual Voice IE GUARANTEE

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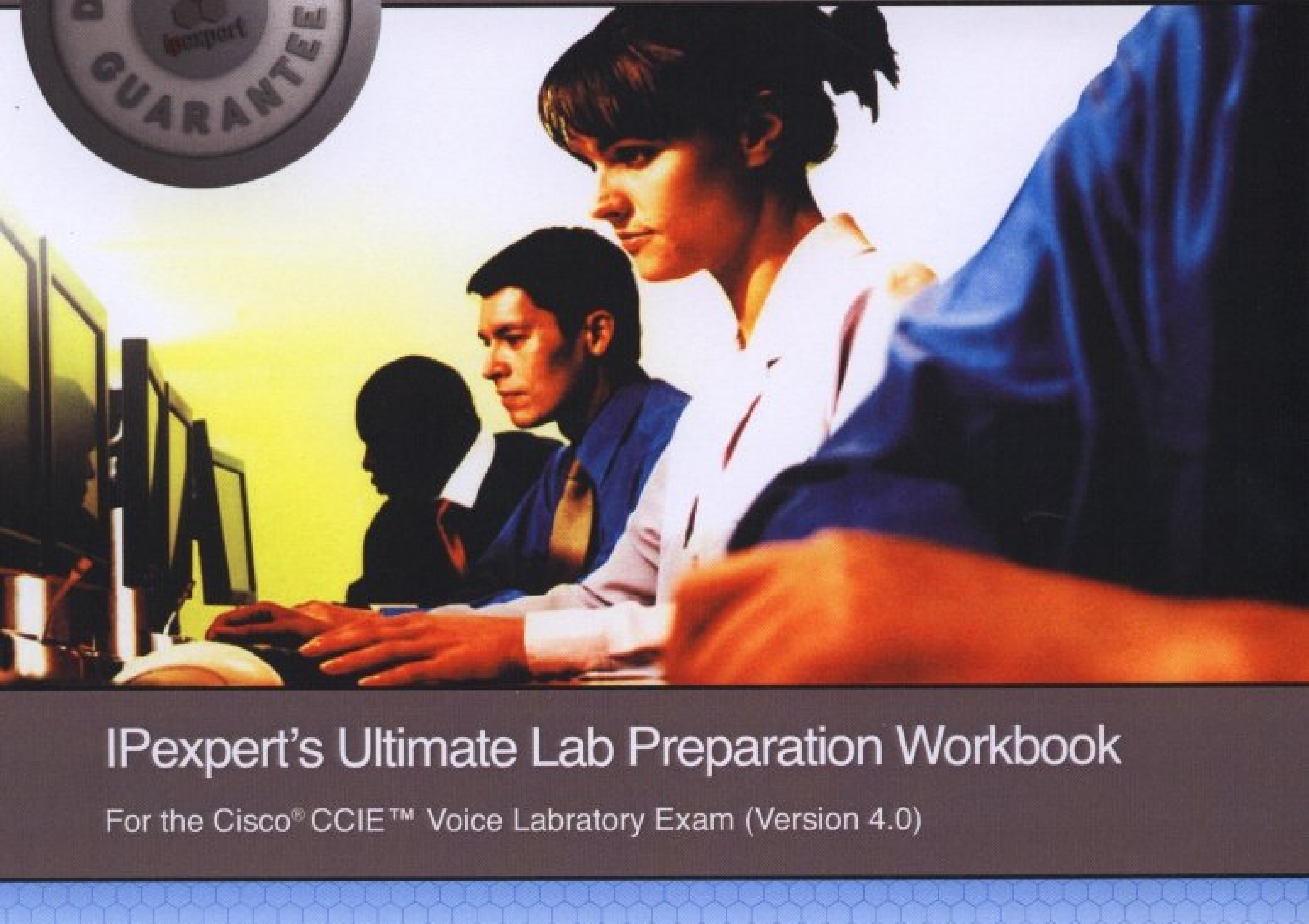

IPexpert's Ultimate Lab Preparation Workbook

For the Cisco® CCIE™ Voice Labratory Exam (Version 4.0)

ipexpert™

powered by PROCTOR LABS

IPexpert's Ultimate Preparation Workbook for the Cisco® CCIE™ Voice Laboratory Exam (Version 4.0)

Before We Begin

Congratulations! **You now possess the ULTIMATE CCIE™ Voice Lab preparation resource available today!** This resource was produced by senior engineers, technical instructors, and authors boasting decades of internetworking experience. Although there is no way to guarantee a 100% success rate on the CCIE™ Voice Lab exam, we feel **VERY** confident that your chances of passing the Lab will improve dramatically after completing this industry-recognized Workbook!

At the beginning of each section, you will be referred to a diagram of the network topology, as illustrated in Diagram A (located on page 4). All sections utilize the same physical topology, which can be rented at www.ProctorLabs.com.

In addition, for your convenience, ALL technical configurations, diagrams, and documentation are now immediately available via download in your IPexpert Member's Area. Please visit the following web site for instructions: http://www.ipexpert.com/configs

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About The Authors

IPexpert employs only the best and brightest CCIE developers and instructors in the industry. Our celebrated team of diverse experts holds multiple CCIE certifications gained from substantial and highly relevant real-world experience. These key attributes give IPexpert the leading edge for delivering the most effective training possible.

Wayne A. Lawson II
CCIE #5244 (R&S), CCNA, CCDA, Nortel NCSE, MCP, MCSE (NT 4.0), MCSE +I, CNA, CNE (4.0), CNX Ethernet, Cisco Wireless LAN Design Specialist, Cisco IP Telephony Design Specialist
Founder & President – IPexpert, Inc.
With 15 years of networking, sales and marketing experience, Mr. Lawson possesses the technical competency, leadership and visionary talent possessed only by the most successful entrepreneurs around the globe. Wayne has served as a highly effective contributing member of five major organizations, including the United States Marine Corps (USMC), International Network Services (INS), Cisco Systems, Vertical Networks and IPexpert, Inc. He has been published on the topics of "Building Cisco Remote Access Networks" (ISBN: 1-928993-13-X) and "Configuring Cisco AVVID" (ISBN: 1-928994-14-8), and has written for various technical and entrepreneurial magazines. Mr. Lawson founded IPexpert in 2001 and continues to revolutionize the way engineers prepare for the coveted CCIE Lab certification. Wayne's unique visionary approach to cutting-edge technologies and enterprise network solutions, coupled with a fanatical dedication to customer satisfaction, propel the engine of success at IPexpert. With a talent for revolutionizing products, services and solutions, and a drive to achieve perfection, his leadership and business ethics have molded IPexpert into the clear leader in CCIE Lab training. In addition to acting as the President and Senior Director of IPexpert, Inc., Wayne is also preparing for his CCIE Voice Lab exam.

Scott Morris
Quad CCIE #4713 (R&S, ISP-Dial, Security and Service Provider), CCDP, CCSP, Cisco Cable Communications Specialist, Cisco IP Telephony Support Specialist, Cisco IP Telephony Design Specialist, CCNA (WAN Switching), MCSE (NT 4.0), Juniper Networks JNCIP and JCNIS, RiverStone Networks RCNP, NSA/CNSS INFOSEC Professional, TIA Convergence Technology Professional (CTP), and CISSP #37445.
Senior Technical Instructor and Developer – IPexpert, Inc.
Boasting more than 18 years of technical training and consulting experience and a wealth of technical certifications, Scott Morris has proven himself among the elite in the technical training industry. Scott is one of the few people in the world currently holding four separate CCIE certifications, and he is actively preparing for his fifth – the CCIE Voice. Scott has an outstanding track record of success in editing, writing and reviewing training books for Cisco Press, Wylie, Sybex, Que Publishing and McGraw-Hill, and teaching CCIE lab preparation materials. He has served as a contributing author for works including Cisco Press' Managing Cisco Network Security book (ISBN: 1578701031) - Chapters on the PIX Firewall; and Cisco Press' CCIE Practical Studies, Vol. 2 (ISBN: 1587050722) - Chapter on Multicast. Scott has also written various articles for Packet Magazine and TCP Mag.

Vik Malhi
CCIE #13890 Voice, CCVP, Cisco IP Telephony Support Specialist, Cisco IP Telephony Operations Specialist, Cisco IP Telephony Design Specialist and Cisco Wireless LAN Design Specialist.
Sr. Voice Technical Instructor and Developer – IPexpert, Inc.
With nearly 10 years of IP Telephony training and consulting experience and a wealth of technical certifications, Vik Malhi has proven that he? one of the top Cisco voice instructors and consultants in the world! Vik was the first engineer to install CM 3.0 in Europe, Has over 6 years of AVVID consulting and implementation experience and has taught CCIE Voice Lab classes for the past several months. Vik has joined IPexpert's accredited team of experts and will be in charge of updating, supporting and teaching IPexpert's CCIE Voice-related products, services and classes.

Mark Snow
CCIE #14073 Voice, CCVP, CCNP, CCDP, CSE, CQS-CIPCCES, CQS-CIPTDS, CQS-CIPTOS, CQS-CIPTSS, MCSE.
Sr. Voice Technical Instructor and Developer – IPexpert, Inc.
From the age of 5 in his father's (patented inventor) laboratory, Mark passion for technology has never stopped growing. With over 10 years working professionally in the IT industry and over 5 years spent consulting internationally with a focus on large-scale, Cisco IP Telephony, Mark brings a wealth of knowledge to the training arena. Mark holds a CCIE in Voice, as well as many other Cisco and Microsoft certifications. Mark plans to begin working on his next CCIE in Security. Mark is responsible for IPexpert's CCIE Voice training, self-paced product development and support.

Feedback

Do you have a suggestion or other feedback regarding this book or other IPexpert products? At IPexpert, we look to you – our valued clients – for the real world, frontline evaluation that we believe is necessary to improve continually. Please send an email with your thoughts to feedback@ipexpert.com or call 1.866.225.8064 (international callers dial +1.810.326.1444).

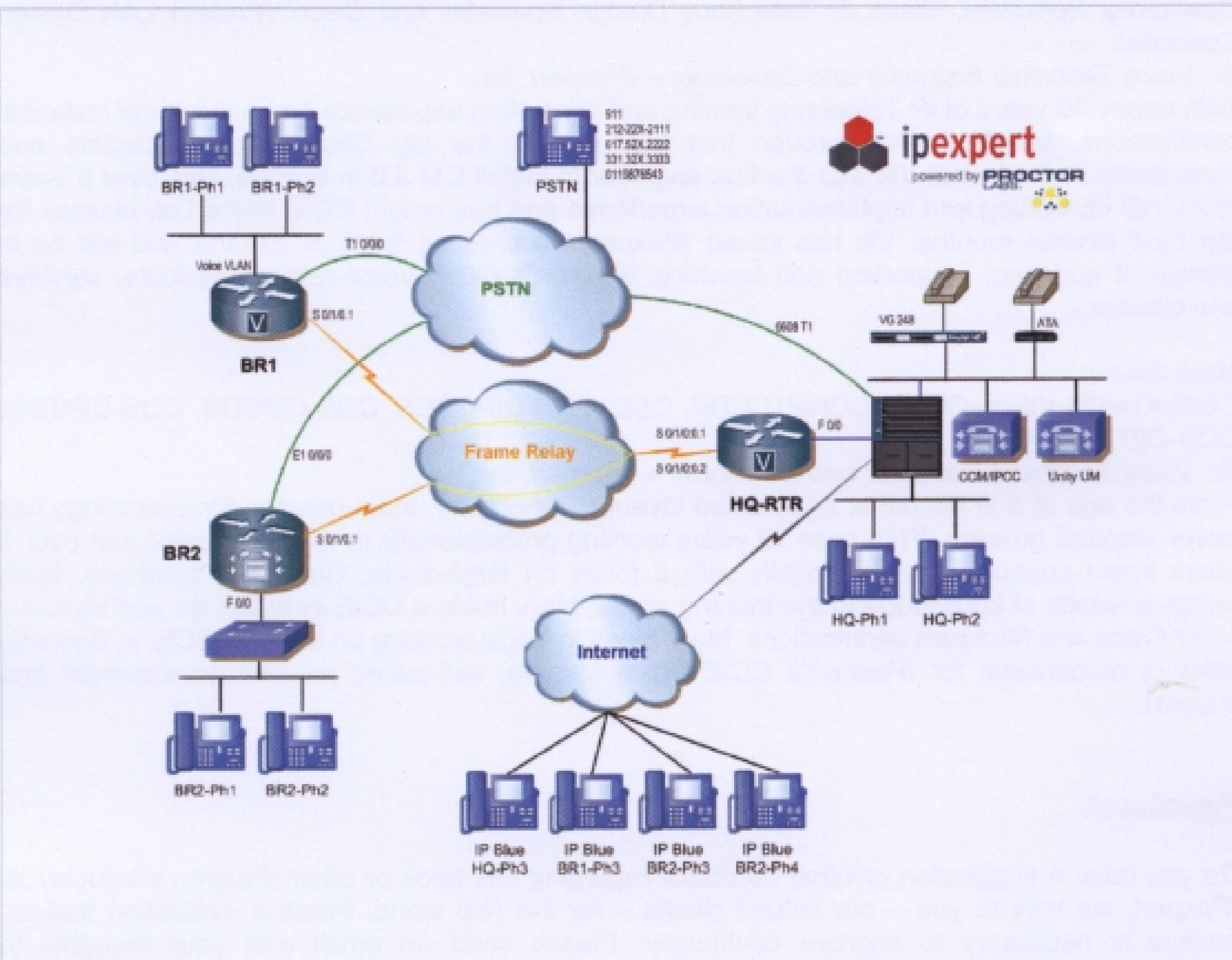
In addition, when you pass the CCIE™ Lab exam, we want to hear about it! Email your CCIE™ number to success@ipexpert.com and let us know how IPexpert helped you succeed. We would like to send you a gift of thanks and congratulations.

Additional CCIE™ Preparation Material

IPexpert, Inc. is committed to developing the most effective Cisco CCIE™ R&S, Security, Service Provider, and Voice Lab certification preparation tools available. Our team of certified networking professionals develops the most up-to-date and comprehensive materials for networking certification, including self-paced workbooks, online Cisco hardware rental, classroom training, online (distance learning) instructor-led training, audio products, and video training materials. Unlike other certification-training providers, we employ the most experienced and accomplished team of experts to create, maintain, and constantly update our products. At IPexpert, we are focused on making your CCIE™ Lab preparation more effective.

IPexpert features a variety of CCIE™ training materials to suit your needs and learning preferences. Please review the catalog that has been incorporated into this book for additional products that are available to you!

Diagram A (Standard Topology)



A message from the Authors:

The scenarios covered in this workbook were developed strictly by Voice CCIEs to help you prepare for the Cisco CCIE Voice laboratory. It is strongly recommended that you also use other reading materials in addition to this workbook.

Training is not the CCIE Voice workbook objective. The intent of these labs is to test your knowledge and ability of implementing Cisco Enterprise Voice Solutions.

Time management is very important, if you get stuck on a lab scenario be sure to write it down. Formulate a checklist for skipped sections and then return to those sections once you have gone through the entire lab. Be sure to revisit the questions that you do not understand.

For more information on the CCIE Voice lab, please visit
http://www.cisco.com/go/ccie
and click on the link for Voice on the top-right of the page.

Helpful Hints

- Keep It Simple, try to avoid any extra work (example: adding descriptions)
- Always try to use "Cisco Best Practices" http://www.cisco.com/go/srnd
- Save your router configurations (write memory)
- Restart the CallManager service periodically.

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Table of Contents

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Table 2 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PODS 11-19: X=Last Digit of POD Number		
PODS 20-22: X=Both Digits of POD Number		
*****Server is not present on Network – but still configure CCM as if it were***		

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM/Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM
12	2/11	POD12 CCM
13	2/17	POD13 CCM
14	2/23	POD14 CCM
15	2/29	POD15 CCM
16	2/35	POD16 CCM
17	2/41	POD17 CCM
18	2/47	POD18 CCM
19	3/5	POD19 CCM
20	3/11	POD20 CCM
21	3/17	POD21 CCM
22	3/23	POD22 CCM

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003
IP Blue	3004

Table 8 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

NOTE:

There is not CallManager Publisher server present on the Proctor Labs network for reasons of speed per pod – HOWEVER, you should configure every aspect of your network AS IF the publisher were alive, used as a secondary call processing engine (in CCM Groups) and utilizing the IP Address given in **Table 2**. Don't forget this when configuring MGCP and H323 GWs among other things.

Section 1 Configuration Tasks

1. Ensure that the link between the HQ/BR2 routers and appropriate switches are configured as dot1Q trunks with no option of becoming an access port. Give the voice sub-interface the appropriate IP address from **Table 2**. Check connectivity between all sites and CallManager/Unity.

2. Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 1**. Use **Table 3** for 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3. Configure all phone ports such that they bypass the spanning-tree listening and learning states.

4. Configure Microsoft DHCP Server on the CallManager server. HQ and Branch 1 phones should get IP Address and other relevant information from the CallManager Server. Use **Table 2** for subnet information; only xx.xx.xx.120 – xx.xx.xx.129 from the voice subnet should be assigned from the server.

5. Configure IOS DHCP on the Branch 2 router; only Branch 2 phones should get IP address and relevant information from this DHCP server. Use **Table 2** for subnet information; only xx.xx.xx.120 – xx.xx.xx.129 should be assigned from the server.

6. Set the hardware clock on the HQ router (use EST as the time zone which is 5 hours behind GMT). Configure the HQ-RTR to become an authoritative time source which distributes the time via NTP. Configure the BR1 and BR2 routers to synchronize their clock with HQ-RTR.

7. Configure CallManager for NTP; use HQ-RTR as the NTP authoritative source.

8. Register HQ and BR1 phones to CallManager based on **Tables 5** and **6**. Ensure that all DNs have 6 media channels, while leaving 3 free for outgoing calls.

NOTE:

You may **not** use auto-registration due to the fact that the CallManager is already in a 'mixed-mode' regarding security. Also, due to this fact, **do not disable** the Services "Cisco CTL Provider" or "Cisco Certificate Authority Proxy Function" **at any time** or your section for Security *may* not work properly.

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9. Start the telephony-service on BR2 router using the voice sub-interface as the source address. Register BR2 phones based on **Table 7**. Configure all lines such that two calls can be active per single DN.

Technical Verification and Support

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Section 2: Call Manager Fundamentals

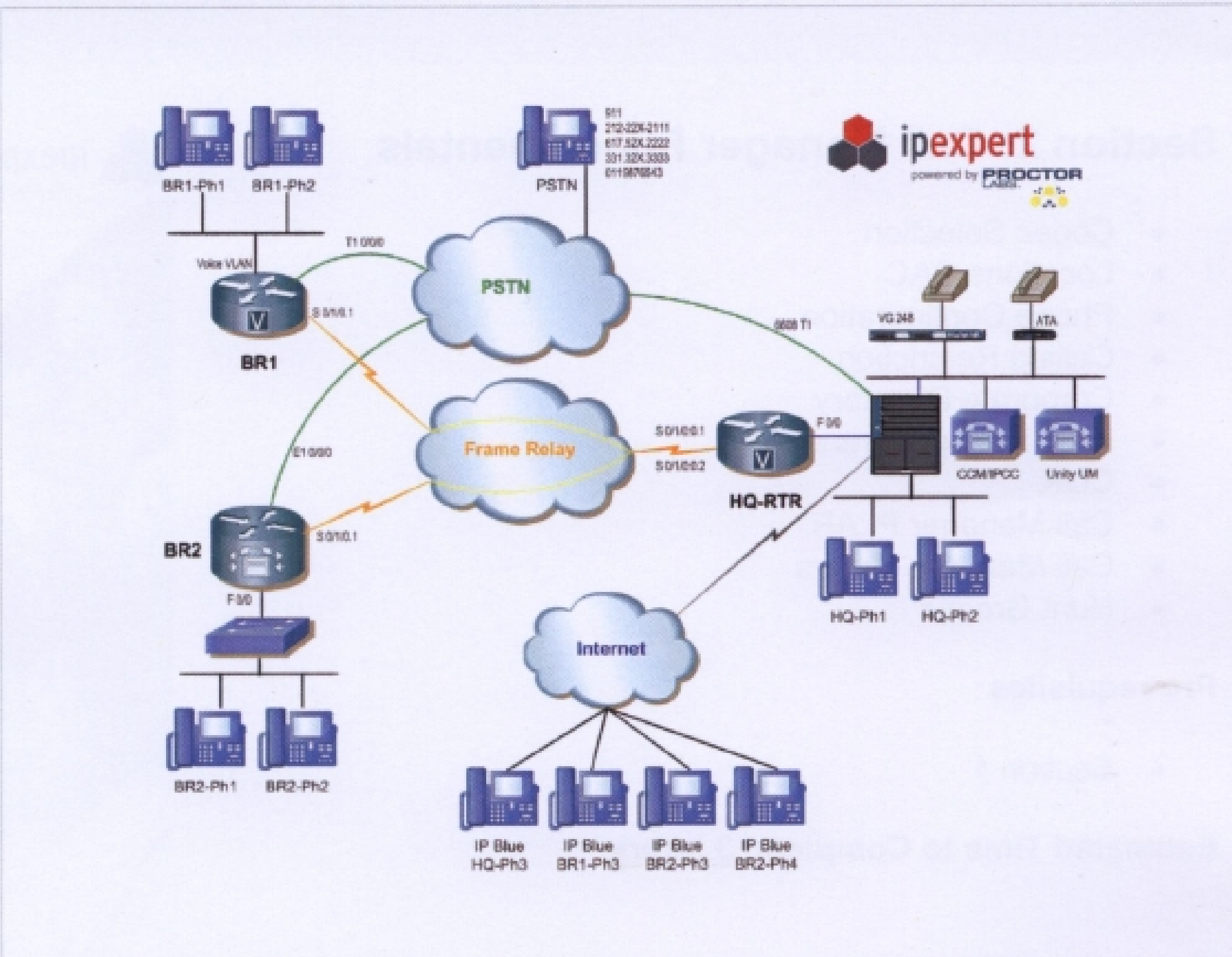
- Codec Selection
- Locations CAC
- Phone Configuration
- Calling Restriction
- Corporate Directory
- Multi-line appearance
- CDR
- Call Manager PLAR
- Call Manager Timers
- Hunt Groups

Pre-requisites

- Section 1

Estimated Time to Complete: <u>3 hours</u>

Section 2 Topology



Section 2 Tables

Table 1 – Calling Restriction

	HQ/BR1 Phone1	All other phones at each site
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

Table 2 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
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CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
***PODS 11-19: X=Last Digit of POD Number**		
***PODS 20-22: X=Both Digits of POD Number**		
*****Server is not present on network – but configure everything as if it were**		

Table 3 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 4 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Section 2 Configuration Tasks

1. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

2. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

3. Change the display on the HQ and BR1 phones so that the clock is using 12-hour format.

4. Ensure that when a user presses the 'Services' or 'Directories' button the error message is not displayed.

5. Configure Calling Restriction based on **Table 1**.

6. Assume a Publisher exists (**Table 2**) and every phone is registered in a CCM Group first to a Subscriber, and then to the Publisher. Configure the keepalive interval between any IP Phone and the Publisher server CallManager to be set to 40 seconds.

7. Set the phones that reside in the POD (not all 79XX phones are physically accessible to you) to auto-answer. Phones must wait 5 seconds before auto answering.

8. Configure CallManager so that the Inter-digit timeout is set to 7 seconds.

9. Enable the Corporate Directory. Add two users with UserID 'hqphn3' and 'br1phn3' and associate relevant devices. Text on the phone should say 'PODXX Directory' instead of 'Corporate Directory'. [When creating users password should be "cisco" and PIN should be "12345"].

10. Configure CallManager such that users logging into CCMUSER web page should **not** be able to subscribe or configure IP Phone Services.

11. Configure CallManager so that CDRs are created for all connected calls. Ensure only calls that are answered create a CDR. A maximum of 500 000 CDRs should be held in the database.

12. Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring simultaneously when this DN is called. You may not use a shared line to accomplish this task.

Technical Verification and Support

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Section 3: CME Fundamentals

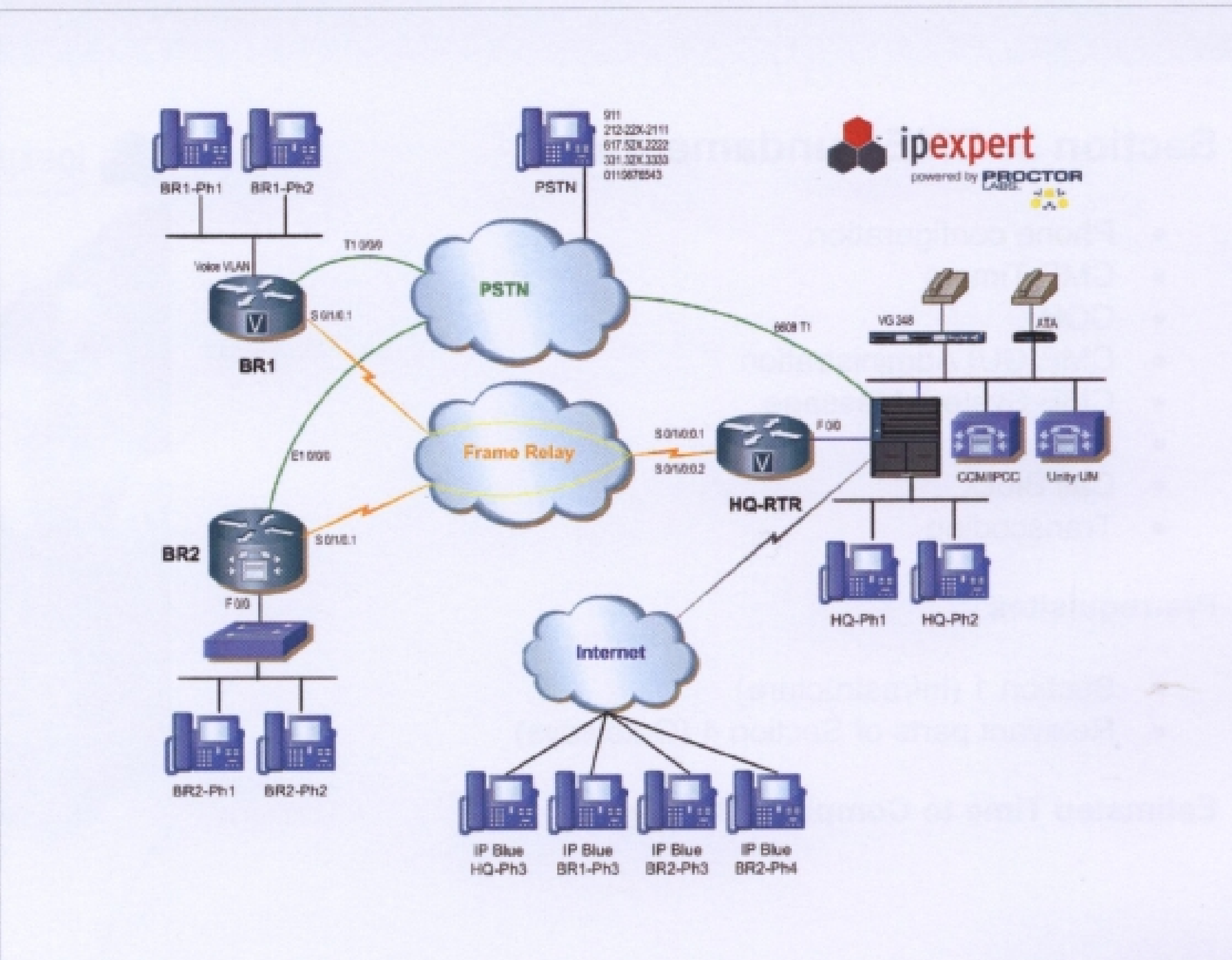
- Phone configuration
- CME Timers
- COR
- CME GUI Administration
- CME System Message
- Paging
- Call Block
- Transcoding

Pre-requisites

- Section 1 (Infrastructure)
- Relevant parts of Section 4 (Gateways)

Estimated Time to Complete: 3 hours

Section 3 Topology



Section 3 Tables

Table 1 – Class of Restriction

	BR2 Phone1	All other phones at site
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

Table 2 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Section 3 Configuration Tasks

1. Create a shared line on the BR2 phones 1 and 3 with DN 3010. The first call coming into the shared line must ring both phones. When a second call comes into the shared line (while the first call to the shared line is still connected) the unused phone that has the shared line must ring **while at the same time** displaying 'call waiting' on the first phone. Ensure that the shared lines have the second media channel enabled.

2. Configure Class of Restriction (COR) based on **Table 1**.

3. Set up the CME GUI and allow the administrator to add/remove DNs through the web interface. Also allow the administrator to manage time through the web interface.

4. Configure the inter-digit timer to 7 seconds.

5. Configure the BR2 phones so that the system display shows the message "BR2 site" instead of "Cisco CME".

6. Create 3 paging groups with DN = 1007, 1008, and 1009. When 1007 is dialed BR2 phone 1 will be paged. When 1008 is dialed BR2 phone 3 will be paged. When 1009 is dialed all the BR2 phones will be paged.

7. International calls should be blocked from all phones Mon-Fri outside office hours. Office hours are 9am-5pm.

8. A call from another site supporting only G729 may ring into a BR2 site CME phone, and that phone may be busy and have set to forward busy calls into CUE VM. If this is the case, ensure that G729 call into CUE will not fail.

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Section 4: Gateways

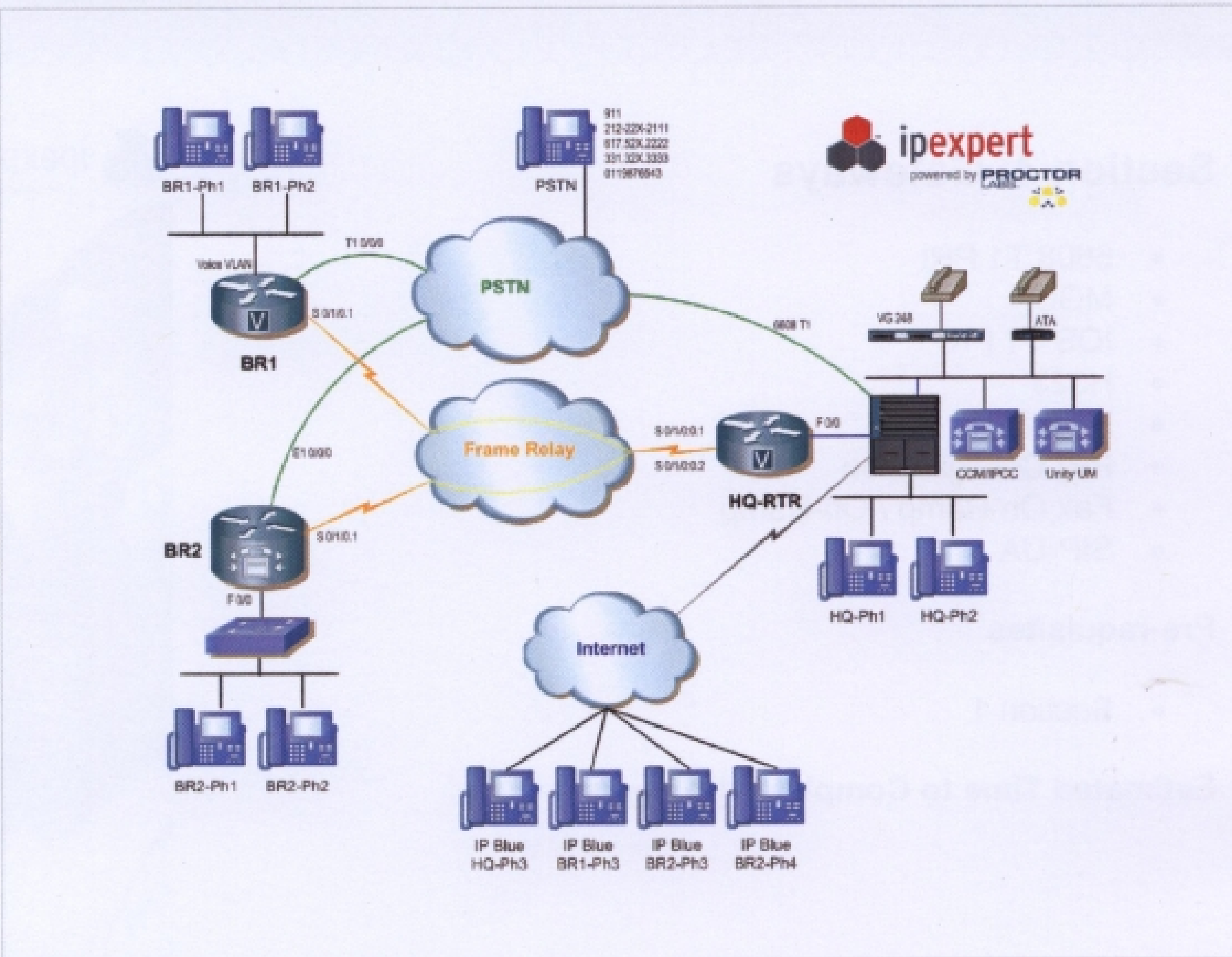
- 6608 T1 PRI
- MGCP
- IOS T1 PRI
- H323
- E1 R2
- PRI Configuration
- Fax On-Ramp / Off-Ramp
- SIP-UA

Pre-requisites

- Section 1

Estimated Time to Complete: 1.5 hours

Section 4 Topology



Section 4 Tables

Table 1 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 2 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
X=Last Digit of POD number including pods 20-23	

Table 3 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
**** X=Last Digit of POD number including pods 20-23***	

Table 4 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 5 – WS-6608 T1 Port Assignment

	6608 PRI
POD11	4/1
POD12	4/4
POD13	4/7
POD14	5/2
POD15	5/5
POD16	5/8
POD17	6/3
POD18	6/6
POD19	7/1
POD20	7/4
POD21	7/7
POD22	8/2

Table 7 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 8 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Table 9 – CME DNs

Branch 2	DN
Phone1	3001
Phone2	3002
Phone3	3003
Phone4	3004

Table 10 – CCM DNs

Site	DN
HQ Phone1	1001
HQ Phone2	1002
HQ Phone3	1003
HQ Phone4	1004
BR1 Phone1	2001
BR1 Phone2	2002
BR1 Phone3	2003

Table 11 – SIP Registrar

FQDN	IP Address
sip1.ipexpert.com	192.20.200.51
sip2.ipexpert.com	192.20.200.52

Section 4 Configuration Tasks

NOTE:

- Please note that clocking will be provided by the network for all locations.
- A test PSTN phone is provided for testing purposes.

1. Configure the HQ 6608 T1 PRI gateway based on **Table 1**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed.

2. Configure BR1 as an MGCP gateway, based on information in **Table 1**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed.

3. Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

4. Configure BR2 as an E1 R2/H323 gateway based on the information in **Table 1**. Outbound ANI should be displayed (10 digits).

5. Configure the E1 R2 such that calls from the PSTN are set up 3 seconds quicker (where possible) than the default. Our carrier instructs that they will send us 10 digits.

6. Block incoming calls to the BR2 router that have no Caller ID.

7. Assume the PSTN has SIP Registrar servers according to **Table 11.** Configure the BR2 gateway to register all of its FXS and soon to be ephone-dn numbers to these 2 servers as primary and fallback respectively. Increase the default retry registration time by half of one second. You must use FQDNs to accomplish this task.

8. Unity needs to be able to accept in-bound faxes at the mailbox of IncomingFax@voip.lab (we will configure Unity for this task later). Unity will also be your DNS server. Configure the BR1 GW to intercept these faxes from the PSTN PRI as an On-Ramp gateway and email all incoming faxes to this mailbox. Refer to **Table 7** of this section for FaxDN ranges. All the files you will need are in BR1 Flash memory.

9. Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 9** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed to CCM via SIP using G711ulaw (see **Table 10** for DNs at in CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

10. Unity also needs to be able to send out-bound faxes to the PSTN (we will configure Unity for this task later). Configure the BR1 GW to accept these faxes via email from Unity and then to send them out through the PSTN PRI as an Off-Ramp gateway. Refer to **Table 8** of this section for PSTN FaxDN ranges. Again, all the files you will need are in BR1 Flash memory.

Technical Verification and Support

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Section 5: Gatekeeper

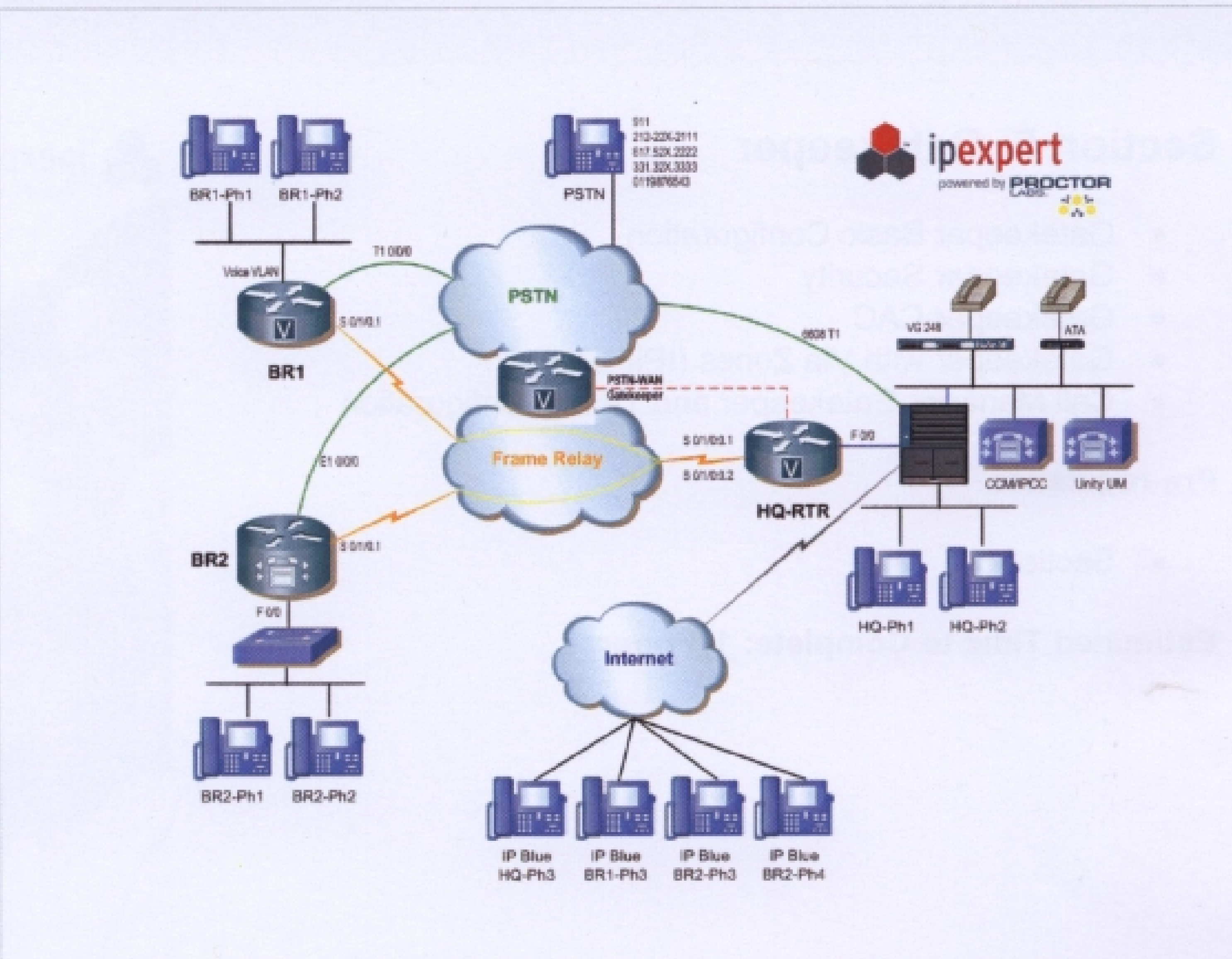
- Gatekeeper Basic Configuration
- Gatekeeper Security
- Gatekeeper CAC
- Gatekeeper with Via Zones (IPIPGW)
- Call Manager Gatekeeper and Trunk Configuration

Pre-requisites

- Section 1

Estimated Time to Complete: 1.5 hours

Section 5 Topology



Section 5 Tables

Table 1 – Loopback IP Addresses

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number	
PODS 20-22: X = BOTH Digits of POD Number	

Table 2 – Device IP Address Assignment

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
***PODS 11-19: X=Last Digit of POD Number**		
***PODS 20-22: X=Both Digits of POD Number**		
****Server is not present on Network – but still configure CCM as if it were**		

Section 5 Configuration Tasks

1. Configure gatekeeper on HQ-RTR with the following information:

   Local zone= HQ-RTR
   domain name = ipexpert.com
   [use loopback interface for local zone]

   Remote zone= PSTN-WAN
   domain name= ipexpert.com
   ip address= 10.X.200.2 [X=Last digit of POD number]

   Register CallManager to the gatekeeper using the default tech prefix 1#

2. Configure CAC to allow one G711 call and one G729 call.

3. Calls should attempt to use G711. If the call fails due to there not being enough bandwidth over the WAN provisioned, then the G729 codec should be selected.

4. Enable security on the local gatekeeper such that only devices that need to register CAN register (e.g. CallManager); other rogue devices should be prevented from registering.

5. Route calls beginning with '011' to the remote gatekeeper.

6. Add on to your Gatekeeper configuration and create a new zone to enable the GK to route calls through your IPIPGW located on the same HQ-RTR. Allow your ATA to register with a new DN of 2080 to the GK in a new local zone. Ensure that calls from CCM routed to the ATA will succeed and terminate both sides of their RTP stream via the IPIPGW. Also ensure the call comes from CCM in a G711 format and flows to the ATA using a G729 codec.

   New Local zone for CCM = CCM-GK
   domain name = ipexpert.com

   New Local zone for IPIPGW = VGK
   domain name = ipexpert.com

   New Local zone for ATA = ATA-GK
   domain name = ipexpert.com

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Section 6: Dial Plan

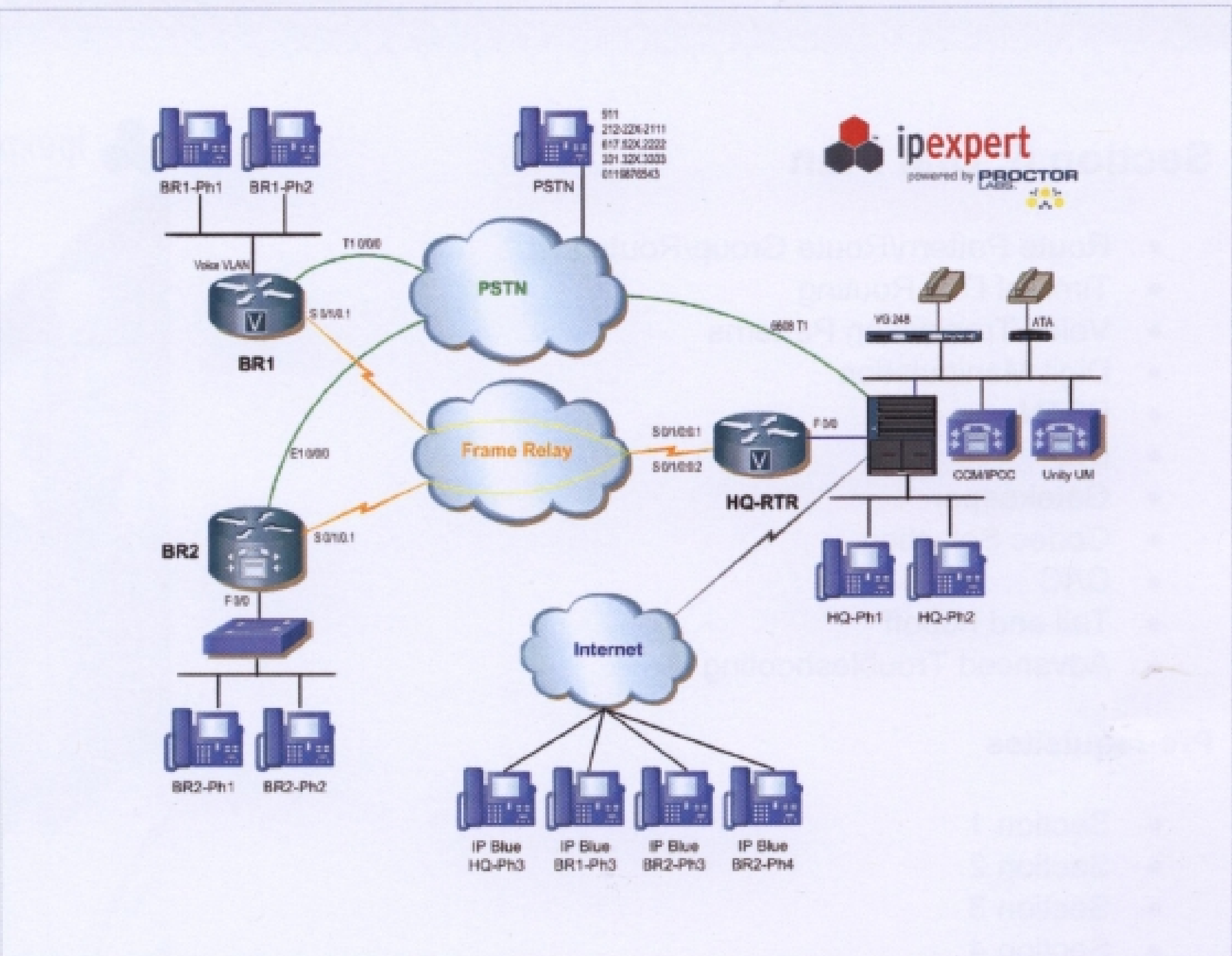
- Route Pattern/Route Group/Route List
- Time of Day Routing
- Voice Translation Patterns
- Digit Manipulation
- PSTN
- DID
- Gatekeeper
- Codec Selection
- CAC
- Tail end hopoff
- Advanced Troubleshooting

Pre-requisites

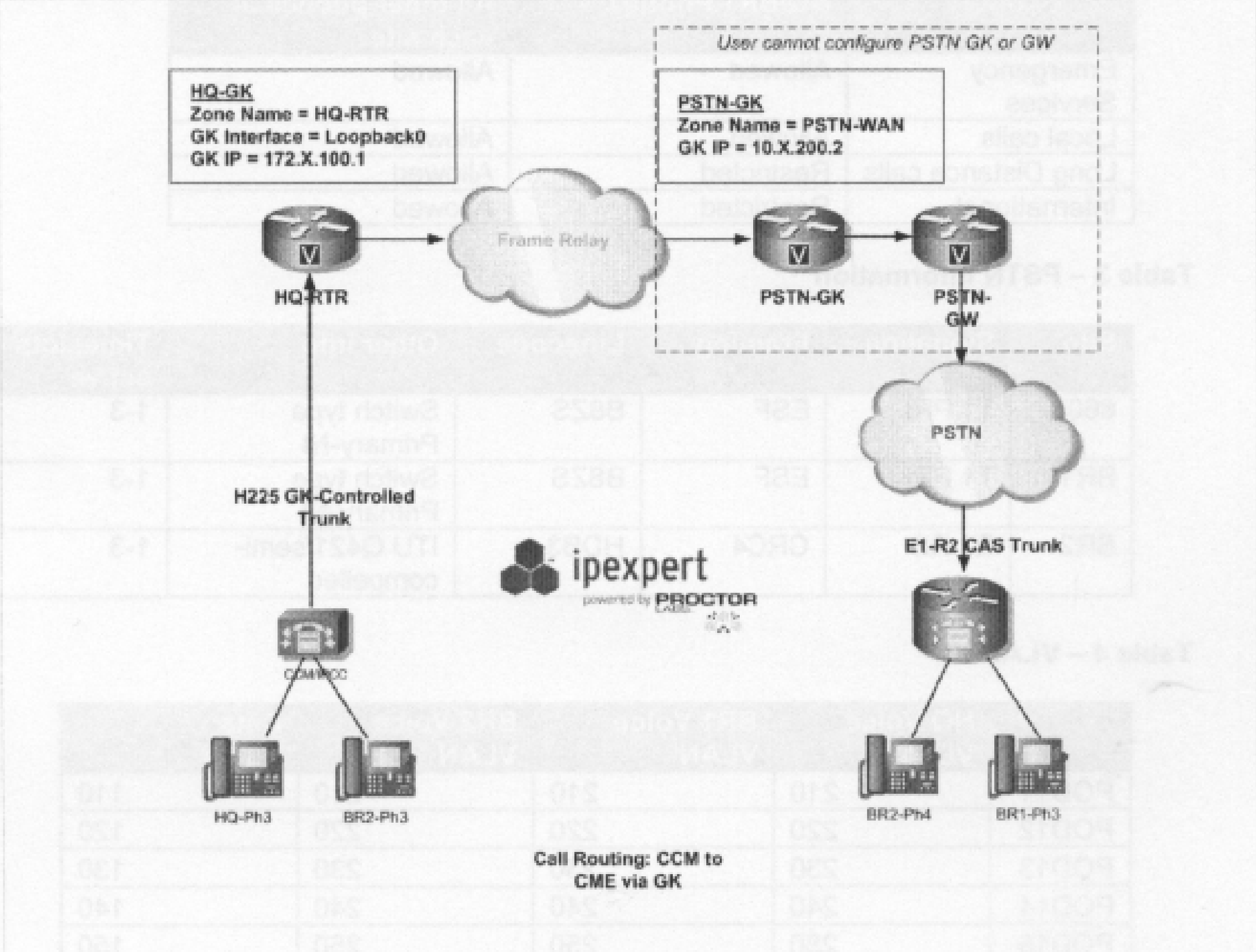
- Section 1
- Section 2
- Section 3
- Section 4
- Section 5

Estimated Time to Complete: 4 hours

Section 6 Standard Topology



Gatekeeper Topology



Section 6 Tables

Table 1 – PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

Table 2 – Calling Restriction

	HQ/BR1 Phone1	All other phones at each site
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

Table 3 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 4 – VLAN ID

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 5 – WS-6608 Port Assignment

6608 PRI	
POD11	4/1
POD12	4/4
POD13	4/7
POD14	5/2
POD15	5/5
POD16	5/8
POD17	6/3
POD18	6/6
POD19	7/1
POD20	7/4
POD21	7/7
POD22	8/2

Table 6 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 7 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 8 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 9 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1...
Branch 1*	617-52X-2...
Branch 2*	011-331-32X-3...
** X=last digit of POD Number*	

Table 10 – Public Phone DN Assignment

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
***X=Last Digit of POD number including pods 20-23**	

Table 11 – Loopback IP Address Assignment

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number *PODS 20-22: X = BOTH Digits of POD Number**	

Table 14 – Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Section 6 Configuration Tasks

For differing types of PSTN Calls refer to Route/Destination patterns defined in **Table 1**.

For exact topology of Gatekeeper call routing refer to the topology diagram.

NOTE:

The call is not VoIP every leg of the call between endpoints.

1. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

2. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

3. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed first out of the local Gatekeeper and then through the HQ-IPIPGW as a backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. (**Test both** call routing methods one at a time before moving on to the next question).

NOTE:

The gatekeeper is expecting the full E164 number; i.e. international prefix '011' plus 10 digits.

---

4. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway. You must utilize 4 different types of methods to manipulate the digits sent to the PSTN (e.g. one method is by using the 'forward-digits' command inside the POTS dial-peer).
5. Calls originating from BR2 to CallManager (both HQ and BR1) should use VoIP through the HQ IPIPGW and the PSTN as backup. 4-digit dialing must be preserved. Also, you must use the minimum amount of dial-peers possible.
6. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used).
7. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 14** for a time schedule. Also, all members of this HuntGroup should get the message in their VM box and see their MWI for any message left for 1005.

NOTE:

For all sites, the Telco is sending 10 digits. At BR2, you are not allowed to use translation rules or the 'num-exp' command.

Technical Verification and Support

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Section 7: Media

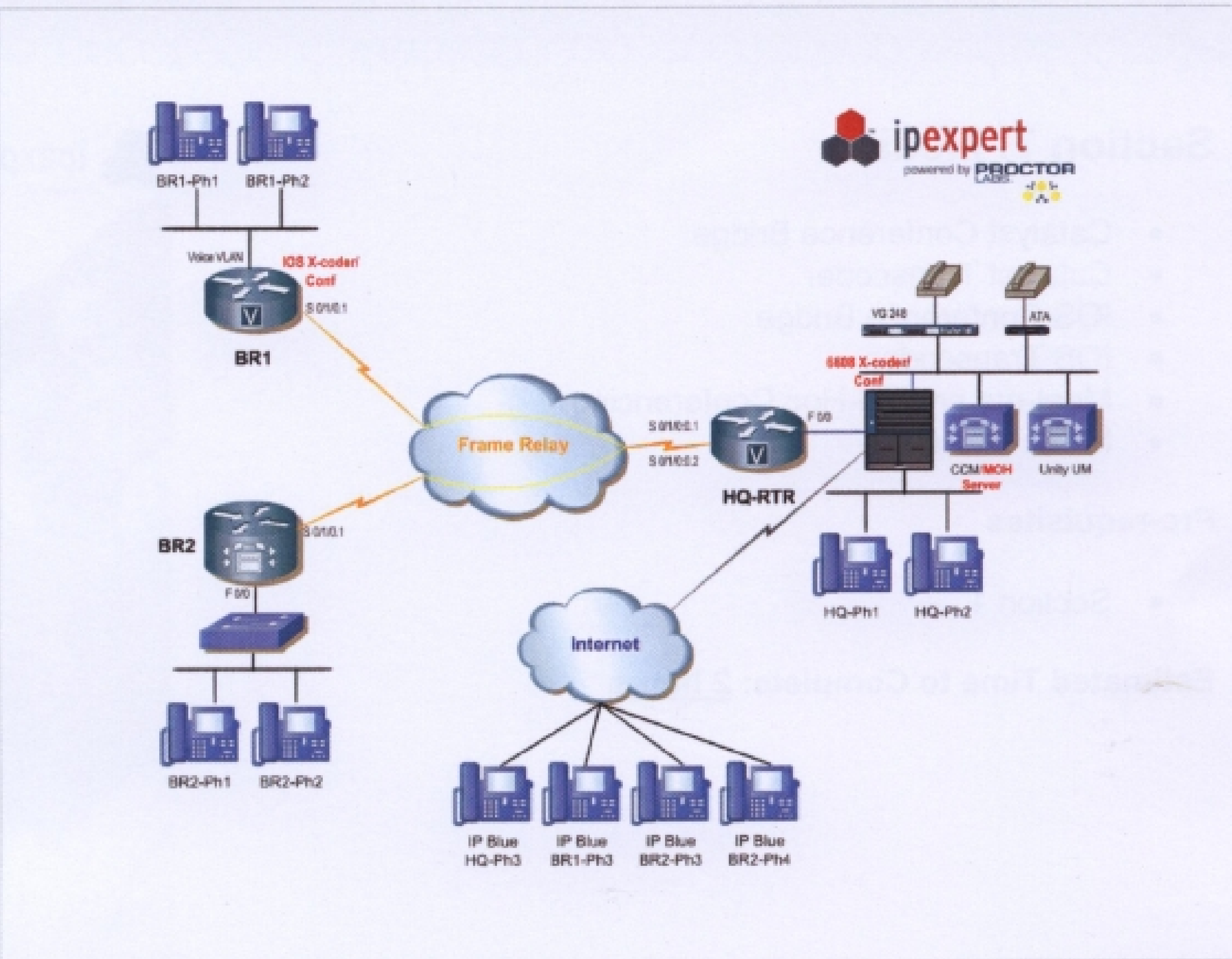
- Catalyst Conference Bridge
- Catalyst Transcoder
- IOS Conference Bridge
- IOS Transcoder
- Meet-me and Ad-Hoc Conferencing
- Music on Hold

Pre-requisites

- Section 1

Estimated Time to Complete: 2 hours

Section 7 Topology



Section 7 Tables

Table 1 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 2 – Device IP Address Assignment

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
***PODS 11-19: X=Last Digit of POD Number**		
***PODS 20-22: X=Both Digits of POD Number**		
****Server is not present on Network – but still configure CCM as if it were**		

Section 7 Configuration Tasks

1 Configure your assigned 6608 port as a conference bridge.

2 Configure your assigned 6608 port as a transcoder.

3 Configure a conference bridge resource on BR1 router for CCM to use. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

4 Configure a transcoder resource on BR1 router for CCM to use. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

5 HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should use the IOS resources and the 6608 resources as backup.

6 Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source.

7 Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You cannot use the transcoder to achieve this task.

8 Ensure that the HQ and BR1 phones receive multicast music on hold.

9 Configure music on hold on the BR2 CME for PSTN callers to hear.

10 Create a meet-me conference with DN=1900. Only HQ Phone 3 should be able to initiate the conference; other devices should be able to join/initiate this conference using DN=1901. Set the maximum number of participants of a single Meet-me conference to 6 conferencees.

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Section 8: High Availability

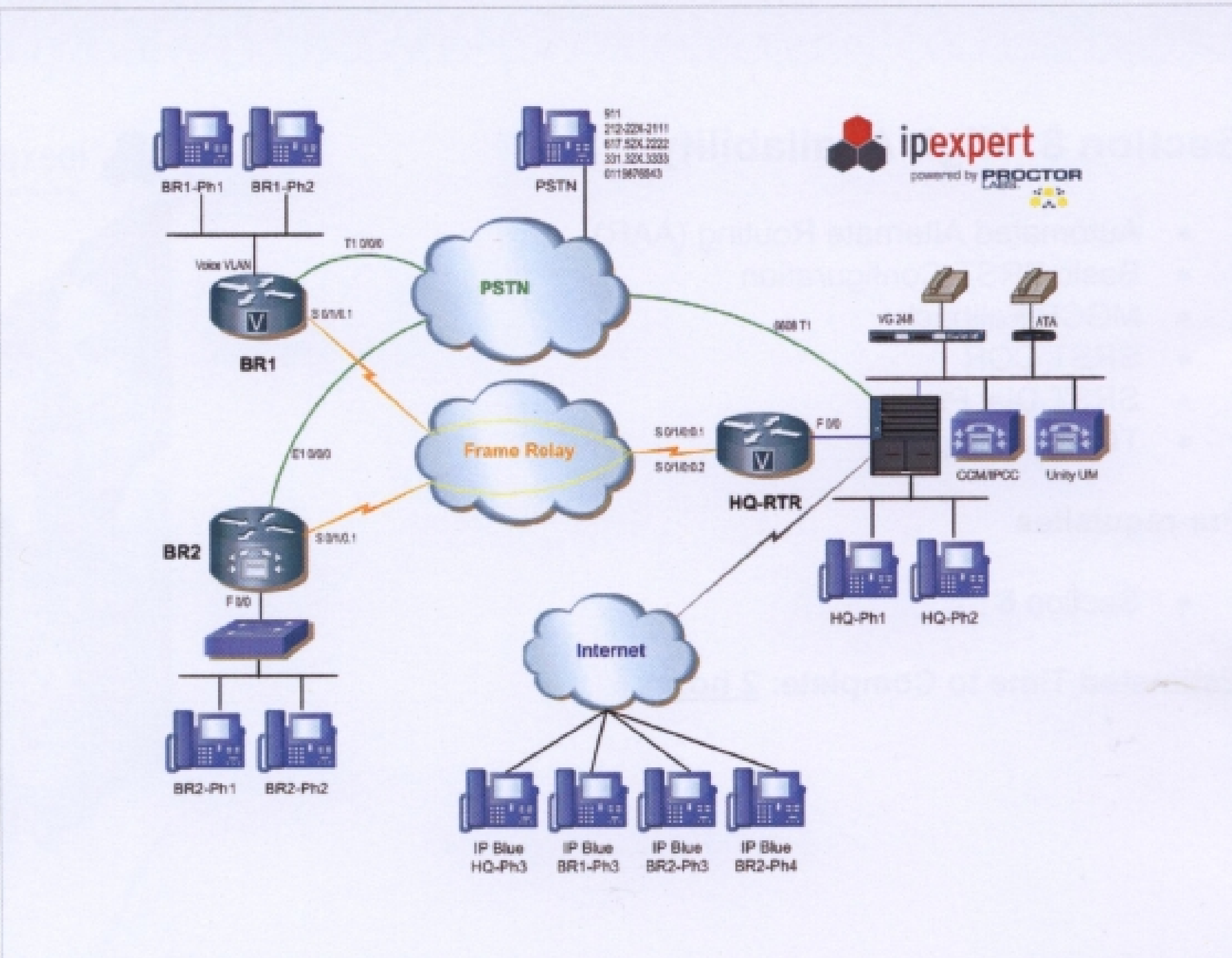
- Automated Alternate Routing (AAR)
- Basic SRST Configuration
- MGCP Fallback
- SRST COR
- SRST Dial Plan
- TCL Scripting

Pre-requisites

- Section 6

Estimated Time to Complete: 2 hours

Section 8 Topology



Section 8 Tables

Table 1 – Voice and Data VLAN IDs

	BR2 Phone1	BR2 Phone2 and 3
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

Table 2 – PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

Table 3 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
***X=Last Digit of POD number including pods 20-23**	

Section 8 Configuration Tasks

1. Configure AAR such that calls between HQ and BR1 will be rerouted over the PSTN when there is not enough bandwidth over the WAN. You must preserve 10 digit Calling Number display. The text "*Network Congestion, Rerouting*!!" must be displayed on the phone when AAR is being used.

2. Configure SRST at BR1 so that in the event of a WAN failure users can still make/receive calls from the PSTN. Use the voice sub-interface as the source address BUT assume that this is not the default gateway. All phones registered must have the second channel enabled on their lines.

3. Configure the SCCP heartbeat timer to 20 seconds and ensure the clock displayed on the phone is using the 12 hour display. Also ensure that the inter-digit timeout is 7 seconds.

4. Configure SRST such that one 3-party conference is allowed.

5. Configure Music on Hold for all phones in SRST fallback.

6. Configure SRST such that the phones will only re-register back to CallManager after normal service has been resumed for 5 minutes. (Normal service is defined as the WAN is operational and CallManager is running).

7. Calls to HQ and BR2 must be preserved using 4 digit dialing. You cannot use the 'prefix' command or translation rules to achieve this task.

8. Ensure that the same Class of restriction is preserved when phones are in SRST fallback.

9. Configure Class of Restriction such that no PSTN caller can dial BR1 phone 2.

10. Use the TCL script already in IOS Flash to provide an IVR Auto-attendant in the case of a WAN down – SRST situation. The pilot DN for CallManager in a non-SRST event is 2000. The pilot DN for the SRST event should also be 2000. If no extension can be reached all calls should ring 2001.

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Section 9: Unity

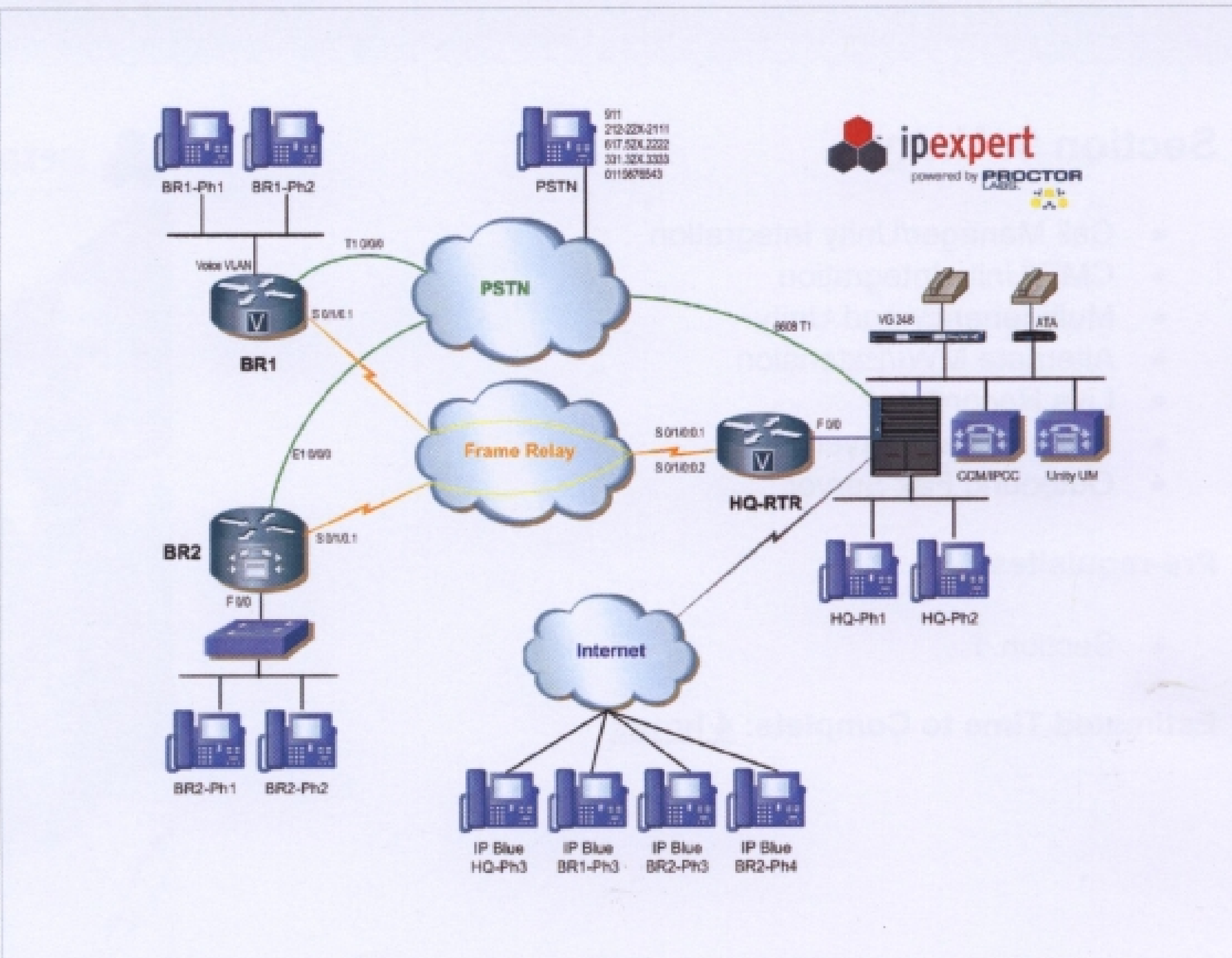
- Call Manager/Unity Integration
- CME/Unity Integration
- Multi-tenancy and Unity
- Alternate MWI/Extension
- Live Record
- Inbound Fax Server
- Outbound Fax Server

Pre-requisites

- Section 1

Estimated Time to Complete: 4 hours

Section 9 Topology



Section 9 Tables

Table 1 – HQ DN Assignment

HQ	VoiceDN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 2 – Branch 1 DN Assignment

Branch 1	VoiceDN
Phone1	2001
Phone2	2002
Phone3 (IPBlue)	2003

Table 3 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003
IP Blue	3004

Table 4 – Device IP Address Assignment

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
***PODS 11-19: X=Last Digit of POD Number** ***PODS 20-22: X=Both Digits of POD Number** ****Server is not present on Network – but still configure CCM as if it were**		

Table 5 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 6 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Section 9 Configuration Tasks

1 Integrate CallManager with Unity with the following information:

- Voice Mail Pilot = 1600
- Voice Mail Ports = 1600-1603
- MWI On/Off = 1999/1998

2. Integrate CME into Unity with the following information:

- Voice Mail Pilot = 3600
- MWI On/Off = 3999/3998

3. Ensure that with both integrations the 4^{th} port is dedicated to MWI and that users **cannot** dial the MWI ext.

4. Phone 1 at HQ should be configured with a Unity subscriber account with DN=1001. You must create this subscriber account using the Bulk Import Tool and a CSV file.

5. Phone 3 at HQ should use the corresponding subscriber account of Phone 1. For example ext '1003' will use the voicemail account for '1001'. Ensure the Phone 1 subscriber greeting is heard when Phone 3 is forwarded to voicemail and that MWI lights up both phones.

6. Create a Unity subscriber account for BR1 phone 1 with DN = 22001. Ensure that the correct subscriber greeting is heard and that MWI is working correctly. You may NOT use Alternate Extension or Alternate MWI to achieve this task.

7. Configure an auto-attendant on Unity with DN = 1570. Record a prompt that says "press 1 for sign-in, 2 for extension of '1001' and 3 for voicemail of 1001' ". Allow caller input during this greeting. The user should press '1' for sign-in, '2' for transfer to the extension '1001', '3' for transfer to voicemail for '1001'. For any other entry, forward back to the auto-attendant using the error greeting.

8. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

9. Continuing from question 4.8 and using **Table 5**; Configure Unity to be an Inbound only Fax Server. Create the necessary VM box for BR1 Phone 2 and ensure that Faxes coming in from the BR1 Gateway arrive in the appropriate user's mailbox.

10. Continuing from question 4.10; Configure Unity to be an Outbound (and Inbound) Fax Server. Ensure that Faxes going out to the PSTN number in **Table 6** from the BR1 Gateway arrive and are viewable.

Technical Verification and Support

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Section 10: IPCC Express

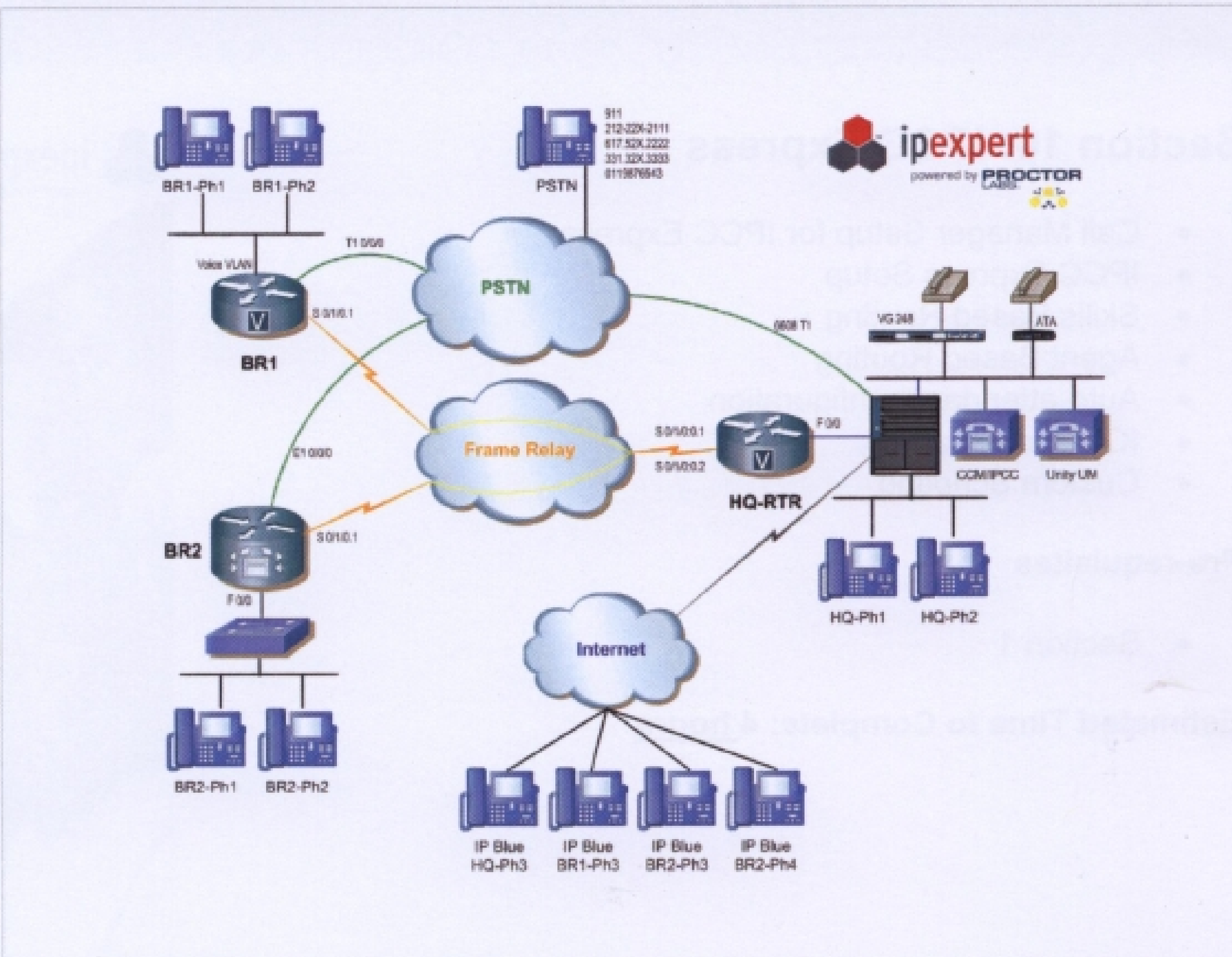
- Call Manager Setup for IPCC Express
- IPCC Express Setup
- Skills Based Routing
- Agent Based Routing
- Auto-attendant configuration
- ICD configuration
- Custom Scripting

Pre-requisites

- Section 1

Estimated Time to Complete: 4 hours

Section 10 Topology



Section 10 Tables

Table 1 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 2 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Section 10 Configuration Tasks

1. Configure Auto-attendant and ICD with the following information:

- AA Route Point = 1710
- Script = aa.aef
- CTI Ports = 1711, 1712

- ICD Route Point = 1700
- Script = icd.aef
- CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine)
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ)

2. Modify the "icd.aef" script with the following information (you may wish to rename it to avoid loosing the original script)

    - You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'

    - If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds

    - Regardless of whether the call is routed to a specific agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue')

3. Configure IPCC Phone Agent for both phones and also ensure that they only have to press the services button once (i.e. that they don't have to provide User/Ext/Pin information on the phone when logging in).

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Section 11: Call Manager Features

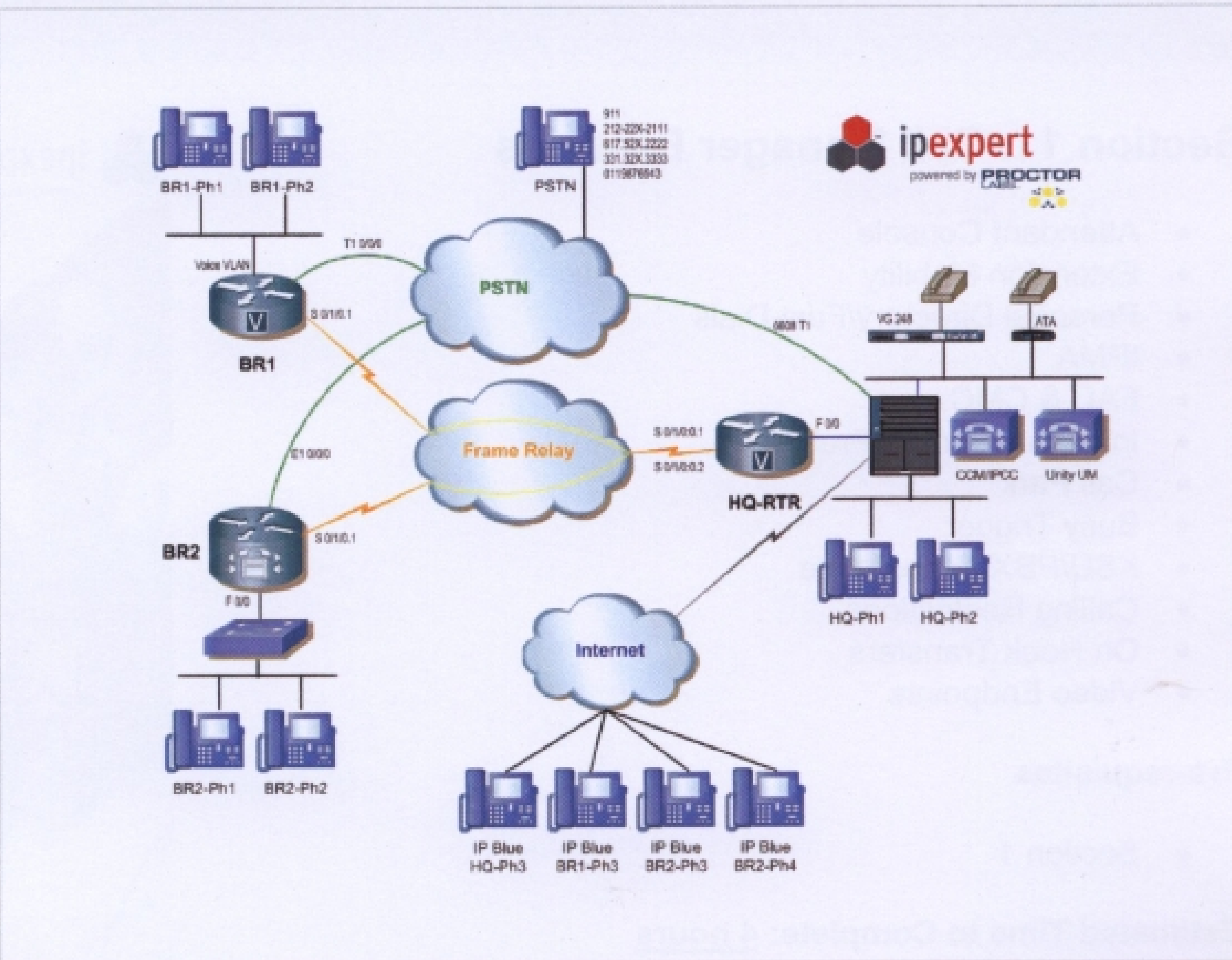
- Attendant Console
- Extension Mobility
- Personal Directory/Fast Dials
- IPMA
- FAC & CMC
- Inbound Number Prefix
- Call Park
- Busy Trigger
- KSU/PBX appearance
- Calling Restrictions
- On Hook Transfers
- Video Endpoints

Pre-requisites

- Section 1

Estimated Time to Complete: <u>4 hours</u>

Section 11 Topology



Section 11 Tables

Table 1 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 2 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 3 – CMC Codes

Branch 1	DN
PSTN Phone number for BR1	701
All other LD numbers	555

Table 4 – FAC Codes

Branch 1	DN
LD Code	9558
International Code	9889

Section 11 Configuration Tasks

1. Configure Attendant Console with Pilot number = '1550'. Add HQ phone 3 and BR1 phone 3 into the huntgroup.

2. Change the 'ac' user password to 'cisco'.

3. Enable Circular Hunting within the AC huntgroup. Also ensure that if there are up to 20 callers simultaneously calling 1550, and both IP Phones are presently taking calls, that callers will not be dropped but any number greater that 20 callers will be dropped and not sent to VM. Also, a call should not be dropped no matter how long it remains waiting.

4. Configure Extension Mobility such that a Device Profile with DN = '1551' is assigned to the following user.

    - UserID='em'
    - Password='adgjm'
    - PIN='12345'

    This user must be allowed to log into any device in the BR1 site. The user should be allowed to log into a device while already logged into another device; this event should cause the user to be logged out of the first device automatically. Finally, configure CallManager such that the user is automatically logged out of a device after 6 hours.

5. Create a user with the following information:

    - UserID='br1phn3'
    - Password='adgjm'
    - PIN='12345'
    - Associate phone 3 at the BR1 site

    Configure Personal Directory and Fast Dials for this user. Create some entries in the Personal Directory and Fast Dials of your choice.

6. Configure IPMA with the following information: BR1 phone 3 will be used as the manager phone and HQ phone 3 will be used as the assistant phone.

Username: br1phn3	**Username: Assistant**
Primary Line: 2003	Primary Line: 1003
SD to Intercom	Proxy Line: 1560
	Incoming Intercom

7. If a call rings into HQ Phone 3, that user must have the option of sending that call to VM without exhausting the CallFwdNoAn timer.

8. Configure calls such that a BR1 user is able to enter one of 2 client codes (see **Table 3**) when dialing a LD number.

9. Ensure that missed calls coming in to an IP Phone do not need any user intervention in order for them to be redialed as an outgoing call (you may not use a translation pattern to accomplish this task).

10. Assign the Call Park range for both Pub and Sub to be the same DNs and allow for 10 slots beginning with DN 1701.

11. Make sure that if during a call, the user presses the Transfer softkey, dials the extension, and immediately hangs up, that the transfer succeeds (without having to press the transfer key a second time).

12. Enable support for a VTA camera on HQ Phone 3.

13. Configure CCM to 'appear' as an older KSU would have appeared at a remote site - so that if a BR1 Phone3 was to pick up their handset and select what they believed to be a "line" to dial out of, that they would not need to first dial a 9 in order to access a trunk. Make this "line" access separate from their main extension DN. Also ensure that the Caller does **not** see the 9 before their dialed number when they place the call.

14. Restrict internal callers from BR1 only, so that they may not see CNAM information only regarding who they are calling or who is calling them; however, ensure that BR1 Phone 3 **can** see all CNAM information.

15. Configure calls such that a HQ user must enter a forced auth code with a level of at least 20 or better in order to be allowed to dial an LD number and one of 30 or better in order to be allowed to dial an International number. See **Table 4** for Auth Codes.

16. Assume that an H323 Video endpoint is at our HQ site but may be taken over to the BR1 site at any time without notice. This Video endpoint has a DN of 1815 and should be allowed to call Local and LD. Ensure that when this Video endpoint moves, that no admin intervention is required on CCM.

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Section 12: Quality of Service

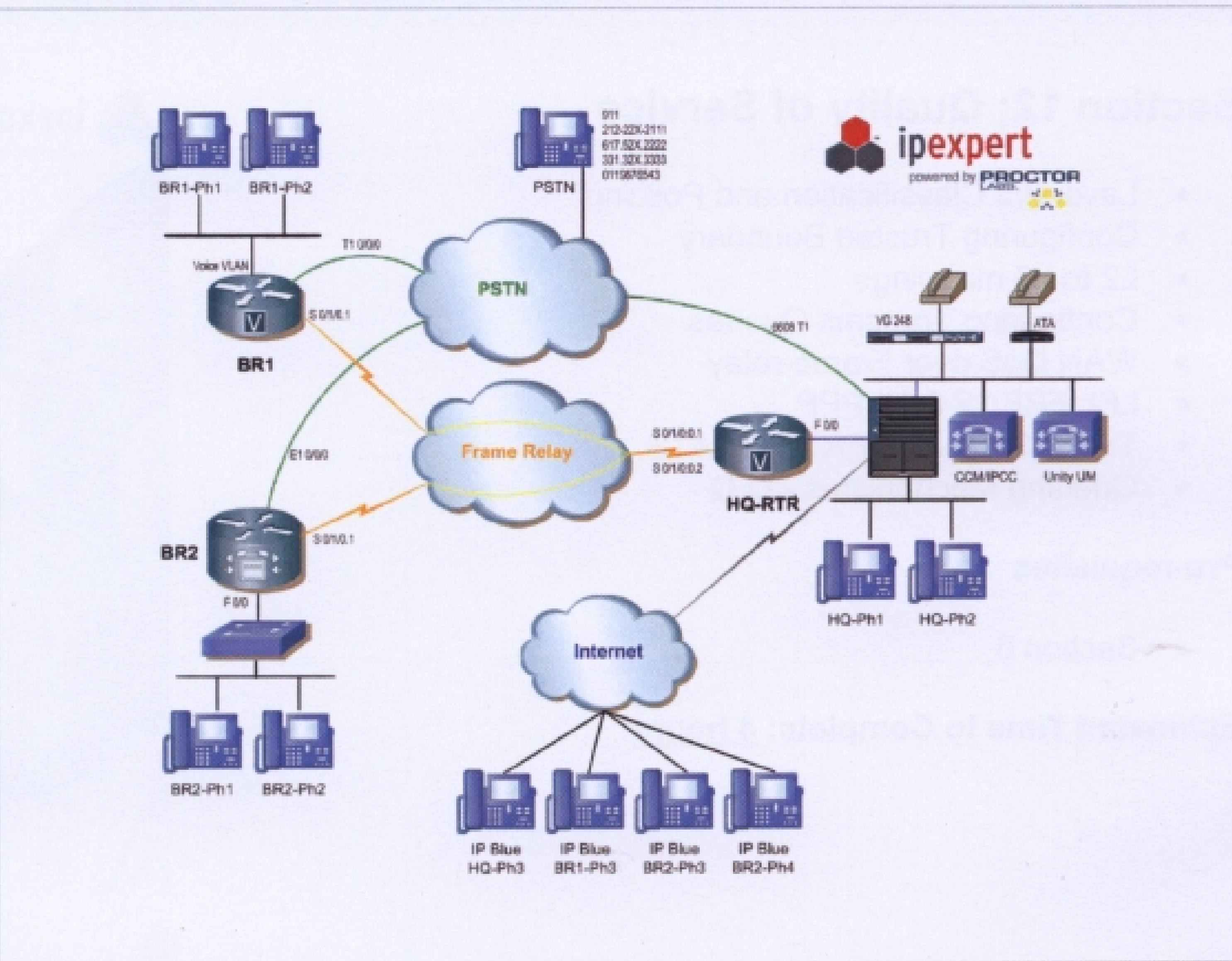
- Layer 2/3 Classification and Policing
- Configuring Trusted Boundary
- L2 to L3 mappings
- Configuring Transmit Queues
- WAN QoS over Frame-relay
- LFI: FRF.12 & MLPPP
- Traffic Shaping – FRTS & VATS
- Queuing Mechanisms - LLQ

Pre-requisites

- Section 6

Estimated Time to Complete: <u>4 hours</u>

Section 12 Topology



Section 12 Tables

Table 1 – Voice and Data VLAN IDs

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** ***Server is not present on Network – but still configure CCM as if it were*		

Table 2 – WAN IP Address Assignment

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
***PODS 11-19: X = Single Last Digit of POD Number**	
***PODS 20-22: X = BOTH Digits of POD Number**	

Table 3 – 6500 Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM
12	2/11	POD12 CCM
13	2/17	POD13 CCM
14	2/23	POD14 CCM
15	2/29	POD15 CCM
16	2/35	POD16 CCM
17	2/41	POD17 CCM
18	2/47	POD18 CCM
19	3/5	POD19 CCM
20	3/11	POD20 CCM
21	3/17	POD21 CCM
22	3/23	POD22 CCM

Table 4 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Section 12 Configuration Tasks

1. Assume the single cable solution is used; configure the switches to trust the Layer 2 QoS classification of the IP phones but not the attached PC. (**NOTE**: The ports where the phones are attached are not shown in the **Tables** – use CDP to figure this out)

2. Configure the Catalyst 6500 to mark all VOIP control traffic from the CallManager to the appropriate L3 setting.

3. Configure the Catalyst 6500 to move VOIP control traffic to the 2nd queue and 1st threshold.

4. Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

5. The speed (CIR) of the FR PVC between the HQ and Branch 2 is 768 kbps. Configure FRTS for such.

6. Add the LFI mechanism of FRF.12 between the HQ and Branch 2 to the previous question and apply this to the PVC. Configure such that the serialization delay is 10 ms.

7. Configure Low Latency Queuing (LLQ) for the HQ and Branch 2. Allocate 33% of the total bandwidth to Media and 2% of the total to control traffic. Ensure that all of the rest of the traffic is WFQ in software.

8. Configure LLQ between the HQ and Branch 1 sites. Allocate 256Kbps for media and 8Kbps for signaling. Assume the speed of the PVC between HQ and Branch 1 is 1544 Kbps.

9. Use the Catalyst 6500 policer to police all control traffic originating from CallManager to 32 Kbps; the exceed action should be to remark control traffic to DSCP 10.

10. Re-configure FRTS between HQ and BR2 such that the traffic shaper only engages when Voice traffic is present on the link. For this task you may assume that the FR port speed is 768kbps and that the CIR provided by the carrier is 384. A proper LFI mechanism should be engaged at all times and should be relevant to the CIR not the Port speed.

11. Now re-configure LFI for the link between HQ and Branch 2. This time you may **not** use FRF.12. Configure such that the serialization delay is 10 ms.

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Section 13: Fax

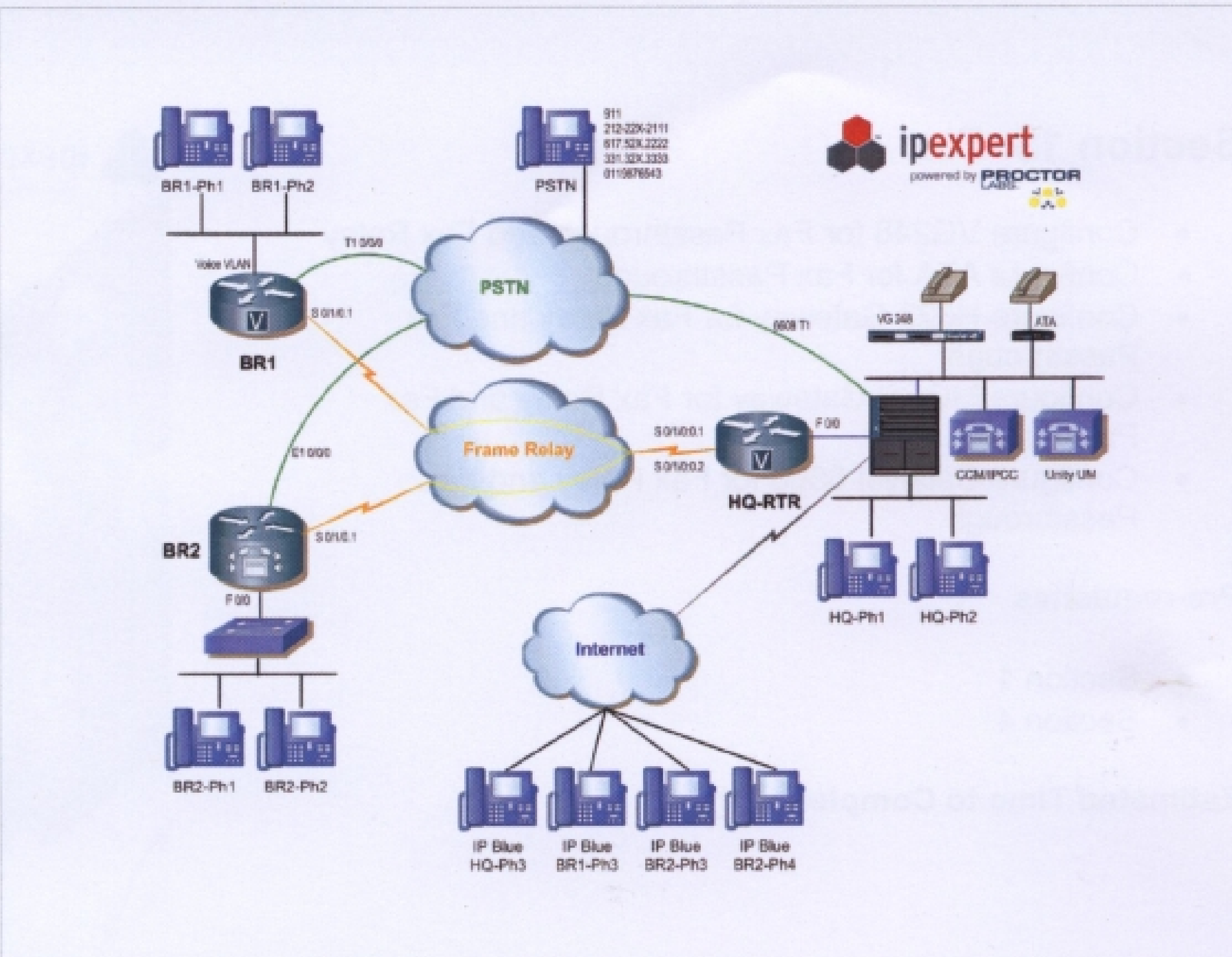
- Configure VG248 for Fax Passthrough and Fax Relay
- Configure ATA for Fax Passthrough
- Configure H323 Gateway for Fax Relay and Fax Passthrough
- Configure MGCP Gateway for Fax Relay and Fax Passthrough
- Configure Catalyst 6608 for Fax Relay and Fax Passthrough

Pre-requisites

- Section 1
- Section 4

Estimated Time to Complete: 1 hour

Section 13 Topology



Section 13 Tables

Table 1 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Section 13 Configuration Tasks

1. Configure VG248 port 1 and 6 for Fax Passthrough.
2. Configure the ATA at HQ for Fax Passthrough.
3. Configure the BR2 H323 Gateway for Fax Passthrough.
4. Configure the BR1 MGCP Gateway for Fax Passthrough.
5. Configure HQ Catalyst 6608 Gateway for Fax Passthrough.

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Section 14: CME Features

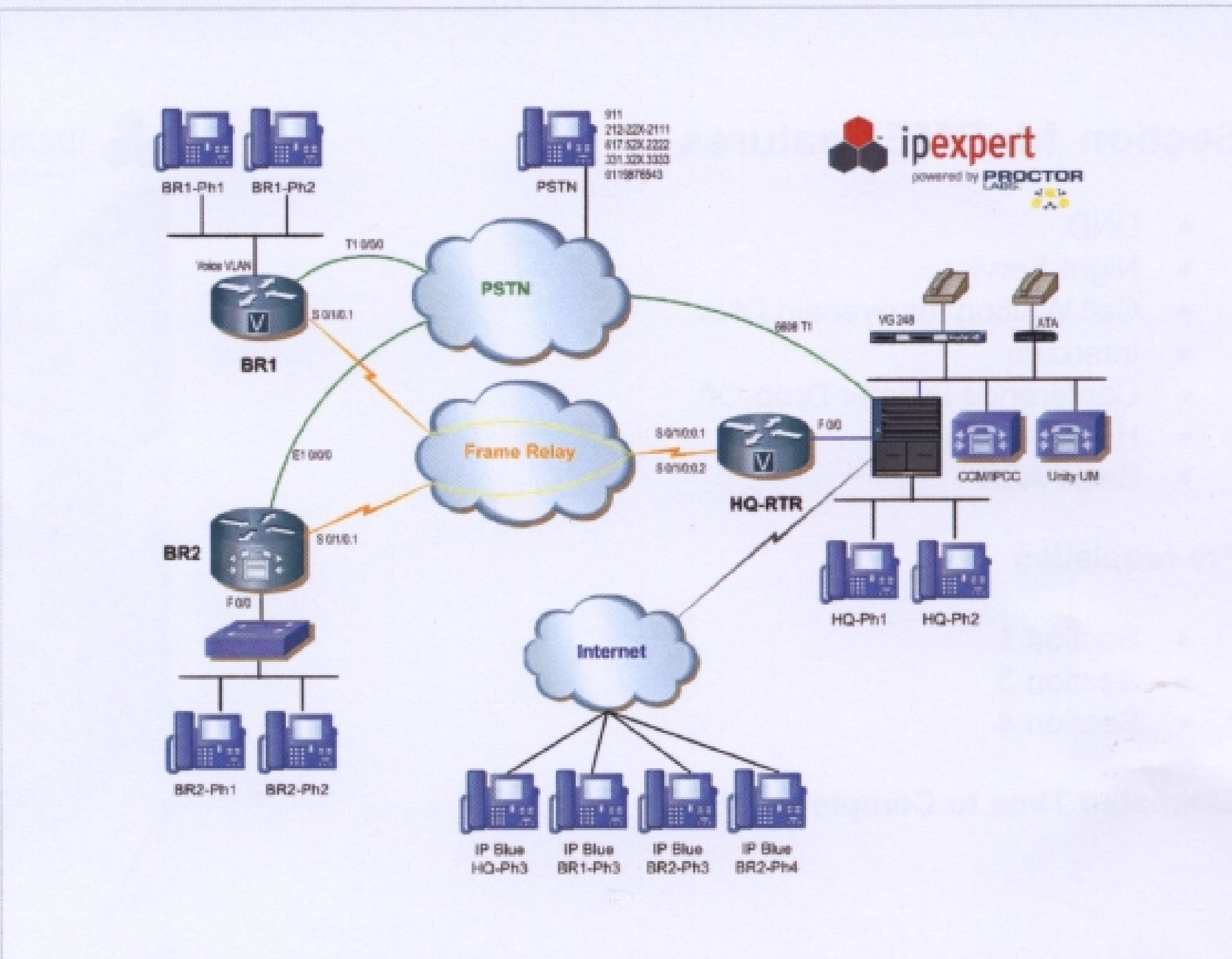
- DND
- Night Service
- Call Waiting for overlaid DNs
- Intercom
- Conference Initiator Drop-Off
- Hunt Group Login
- Basic ACD

Pre-requisites

- Section 1
- Section 3
- Section 4

Estimated Time to Complete: 2 hours

Section 14 Topology



Section 14 Tables

Table 1 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Section 14 Configuration Tasks

1. Change Phone 3 at BR2 so that the user cannot permanently invoke the Do-Not-Disturb function.

2. Change Phone 2 at BR2 so that the user may not utilize the Callback feature.

3. Configure phones 1 and 2 at BR2 so that they can intercom each other and have an immediate 2-way conversation – security should be in place so that no one else can dial their respective intercom numbers. Use whatever DNs you wish for this. Also, if another intercom call happened to be present on phone 1 when phone 2 places the intercom call, the first call should be automatically put on hold.

4. Allow that if a user dials *67 and then a pattern (PSTN or Internal), that the caller's ANI will not show up on the other side.

5. Continuing from Task 3.7; Allow phone 3 at BR2 to be able to enter a code in order to make International calls after hours, and make phone 2 to never be restricted for after hours international calls.

6. Ensure that any conference call at BR2 in which the conference initiator hangs up – that the conference terminates upon that person's leaving. However Phone 3 should be allowed to hang up or press the 'end-call' softkey and leave a conference but allow it to continue running.

7. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 and 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member.

8. Send all IP Phone requests at BR2 for IP XML Services to the CallManager main Services URL.

9. Create an incoming AutoAttendant at the DN of 3000. Also Create a Basic ACD using the support team hunt group you just created (The necessary TCL scripts are already loaded in BR2 router's flash memory). Have the AA script automatically hand-off the callers into the support ACD hunt group when a user presses 2. Allow no more than 20 callers in the Q at any one point. Play a prompt for the user every 30 seconds to let them know that all agents are busy. Allow the users to dial-by-extension by pressing 4. Ensure that the Q is collecting statistics and view them as part of your troubleshooting.

10. Ensure that all calls that are placed internally (from IP Phone to IP Phone) receive not only CLID but also CNAM display with their respective names (you may name them whatever you wish).

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Section 15: CUE Fundamentals

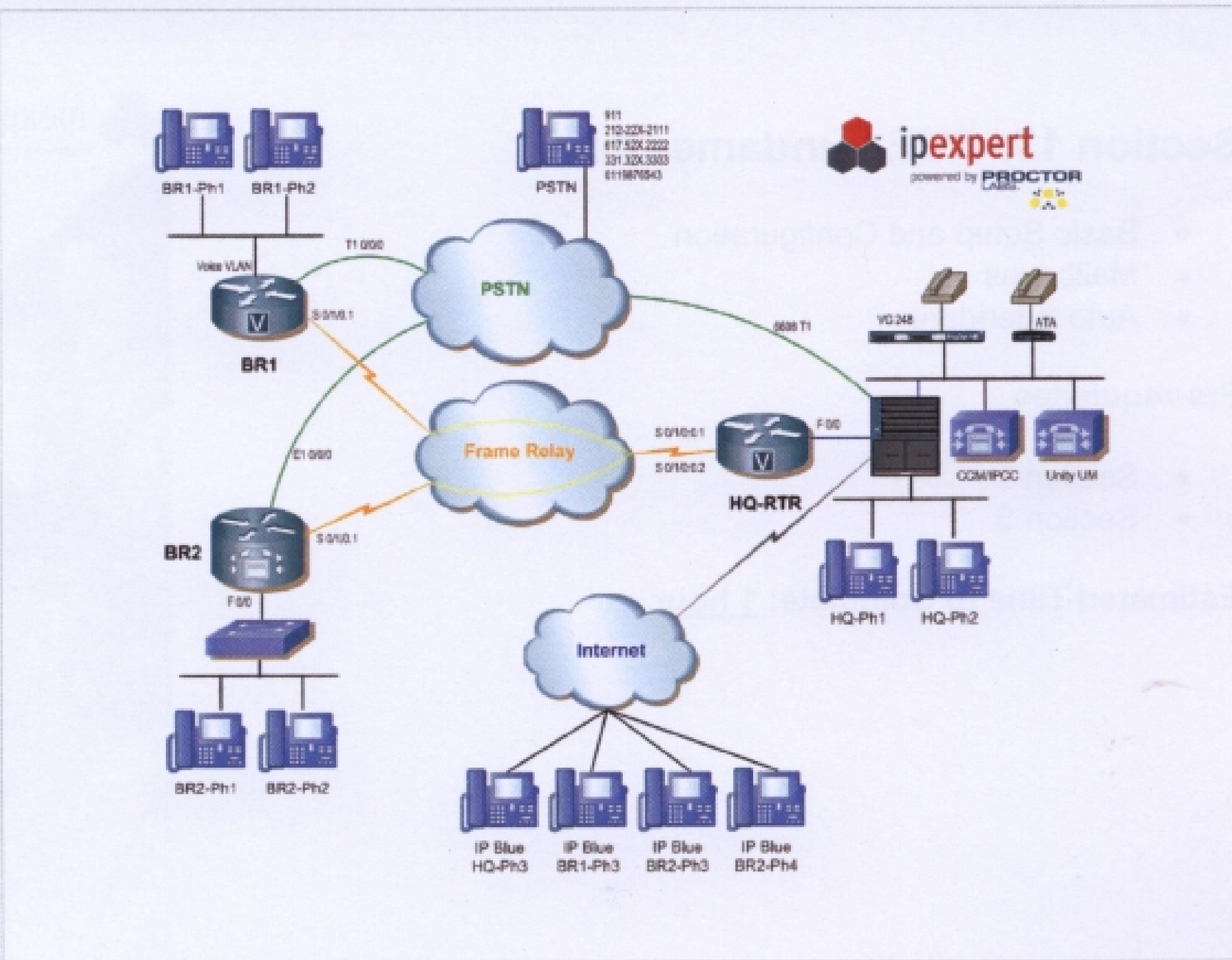
- Basic Setup and Configuration
- Mailboxes
- Auto Attendant

Pre-requisites

- Section 1
- Section 3

Estimated Time to Complete: 1 hour

Section 15 Topology



Section 15 Tables

Table 1 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 2 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
***PODS 11-19: X = Single Last Digit of POD Number** *PODS 20-22: X = BOTH Digits of POD Number*	

Section 15 Configuration Tasks

1. Configure the BR2 router to support the CUE module using information from **Table 2**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate user's extensions and mailboxes.

   Un-integrate CME from Unity (performed in Task 9.2) and integrate into Unity Express using the same information as follows:

   - Voice Mail Pilot = 3600
   - MWI On/Off = 3999/3998
   - AA DN = 3100
   - TUI = 3200

   Finally, setup mailboxes for all 3 phones at BR2.

2. Ensure that if a call is forwarded from any one of the 3 phones for reasons of No-Answer or Busy that the call goes to the appropriate mailbox.

3. Ensure that when users listen to their voicemail messages, that they hear the ANI announced of the phone who left them the message.

4. Create a new Auto-Attendant application to replace the default one setup for you during initialization. (You may not use TCL or Unity for this operation) The incoming DN will be 3100 and the AA should give the option to dial-by-name by pressing 1, dial-by-extension at any time during the prompt, and also to be able to press 2 and be connected to the Support Hunt Group (the Queuing feature provided by the TCL script is not necessary to fulfill this requirement).

5. Assume that CUE does not mark any of its traffic with correct DSCP bits. Ensure that in the router, this traffic is marked correctly as soon as it comes from CUE and ensure that it follows the same standards of marking set from CCM regarding voice and call control.

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Section 16: CUE Features

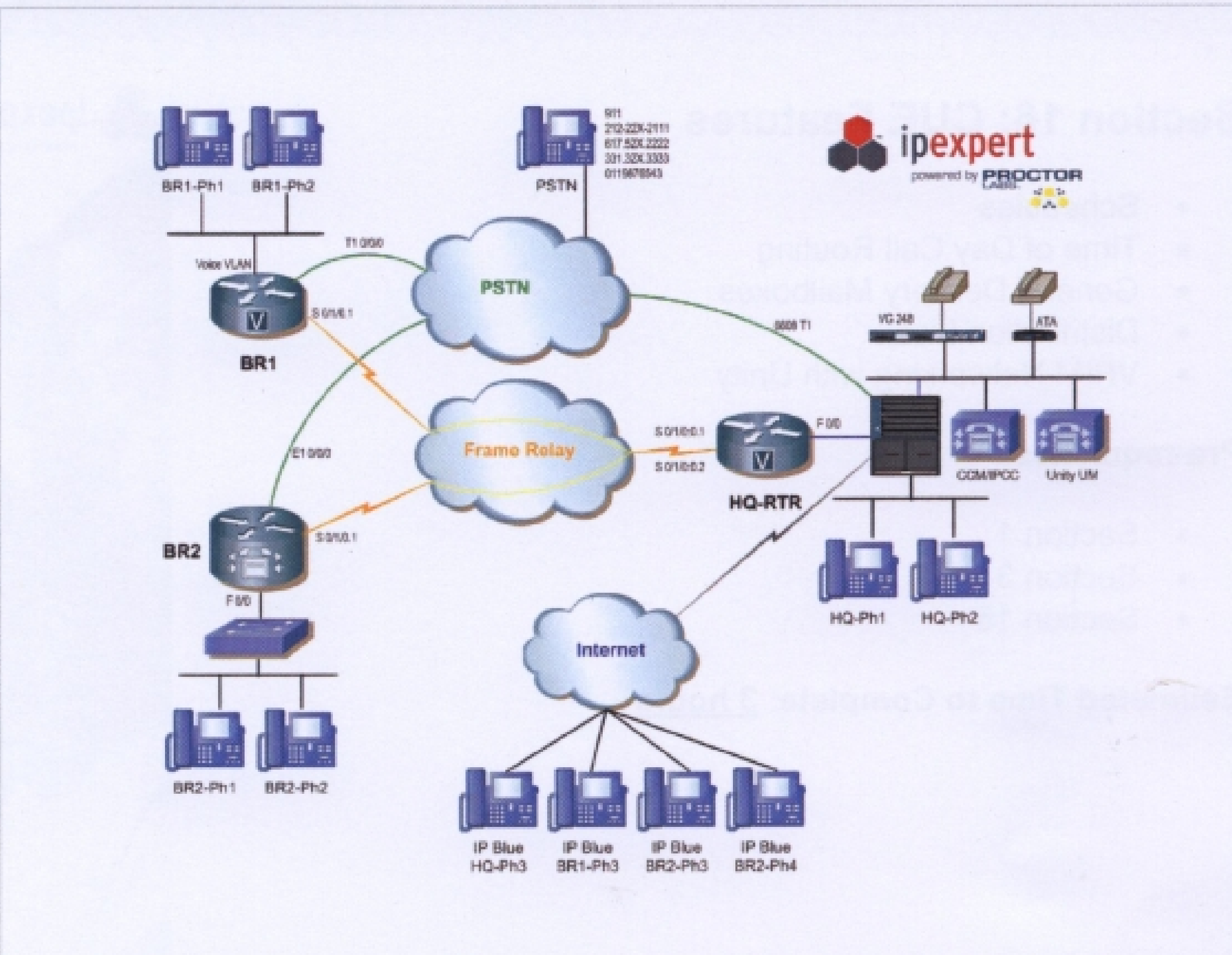
- Schedules
- Time of Day Call Routing
- General Delivery Mailboxes
- Distribution Lists
- VPIM Networking with Unity

Pre-requisites

- Section 1
- Section 3
- Section 15

Estimated Time to Complete: 3 hours

Section 16 Topology



Section 16 Tables

Table 1 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 2 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
****PODS 11-19: X = Single Last Digit of POD Number*** ***PODS 20-22: X = BOTH Digits of POD Number***	

Section 16 Configuration Tasks

1. Create a schedule in CUE that defines normal business hours as Mon-Fri 8am-5pm and Saturday 10am-2pm.

2. Create a General Delivery Mailbox for the Support Queue but give it the extension of 3215. Ensure that any phone in the office can access this mailbox by pressing "9" after they sign into their VM box. Finally, modify the Support Queue (not the hunt group) so that if agents are un-available they will go to this GDM. Also ensure that all BR2 phones see if there is a message waiting in the GDM.

3. Modify the AA script you created to cause calls that come into the CUE AA during normal business hours go to the standard menu, but that if the call comes in after hours or during holidays, to send it directly to the general delivery mailbox for the Support Q. Create a holiday for Dec 25.

4. Create a Distribution List that allows users to be able to forward important messages to extension 3250 and all phones will receive the message in their own mailbox. The GDM must also receive the message in its box. You may **not** directly select phone extensions when creating this List. You also may **not** use the default 'everyone' list.

5. Network CUE with Unity. Allow messages to be seamlessly forwarded back and forth.

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- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 17: Security

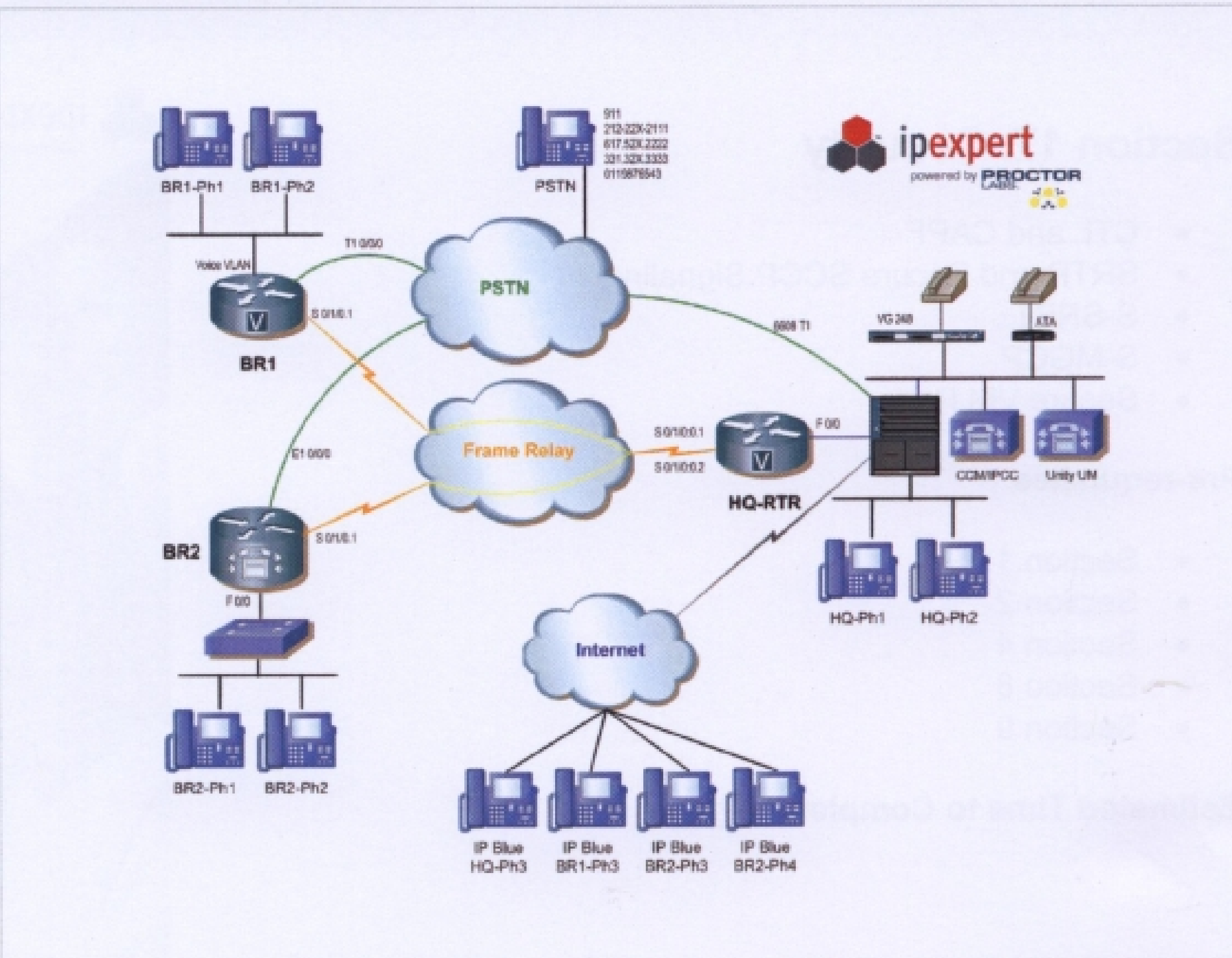
- CTL and CAPF
- SRTP and Secure SCCP Signaling
- S-SRST
- S-MGCP
- Secure VM Ports

Pre-requisites

- Section 1
- Section 2
- Section 4
- Section 8
- Section 9

Estimated Time to Complete: <u>5 hours</u>

Section 17 Topology



Section 17 Tables

Table 1 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Section 17 Configuration Tasks

1. CallManager has already been converted to Mixed Mode and has 2 USB Security Tokens installed in the CTL. Ensure that the proper Services in CallManager are activated and running in order to allow the CTL and CAPF functions to operate correctly (**DO NOT change their status – this may break security without physical access to the USB ports and Tokens – which you do not have**).

2. Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2, will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that no user interaction be required on the physical IP Phone.

3. Ensure that if BR1 gateway goes into fallback mode, that it has already created its own certificate, forwarded that certificate to CallManager and that CallManager in turn has installed that certificate into the IP Phones at BR1, so that signaling and media continue to be encrypted during any fallback occurrence.

4. Ensure that RTP traffic between any capable IP Phone and the BR1 MGCP gateway is encrypted with AES128.

5. Ensure that all MGCP signaling between CallManager and BR1 gateway is encrypted with 3DES 168bit encryption using a SHA hashing algorithm.

6. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
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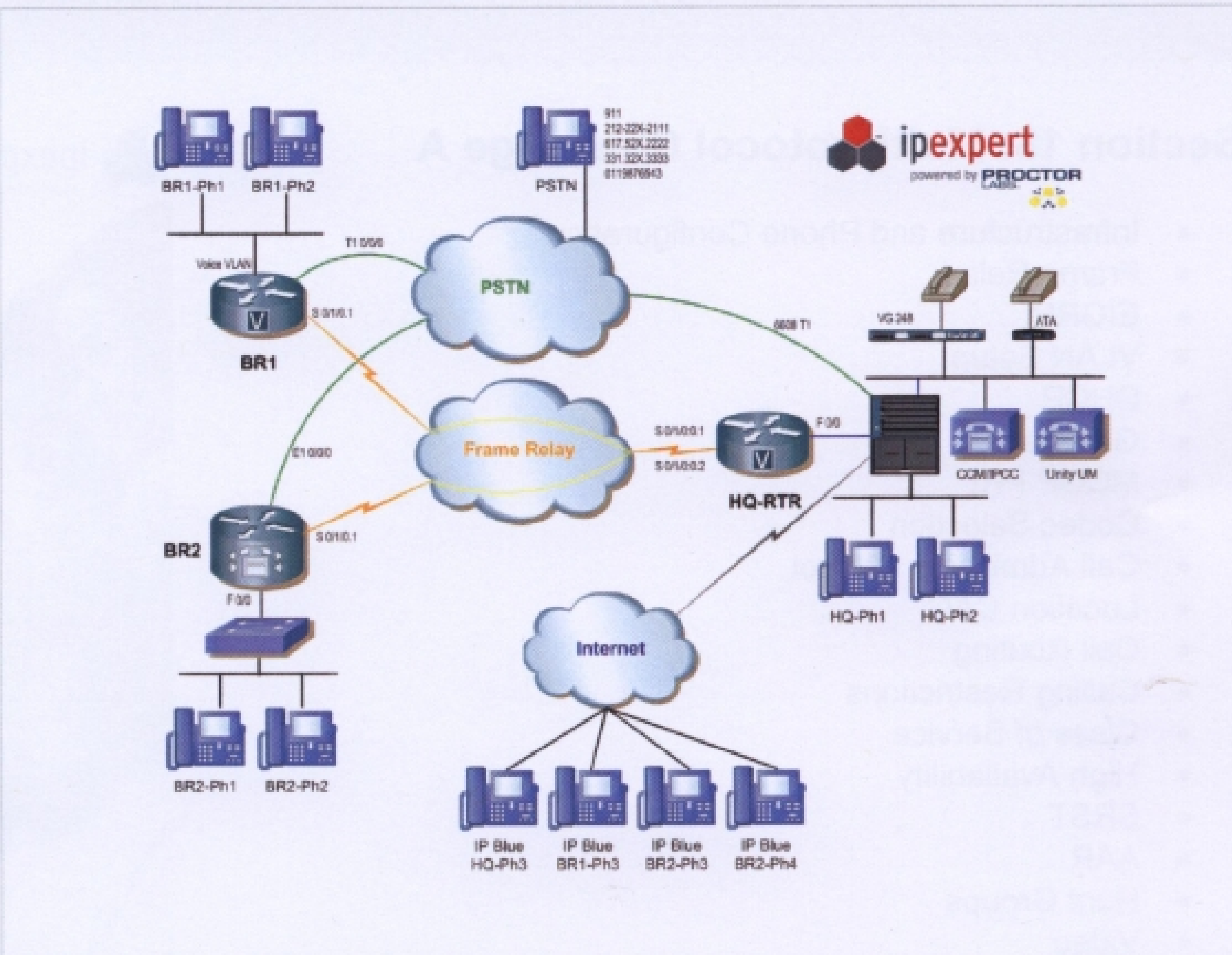
- Online Forum: http://www.CertificationTalk.com

Section 18: Multiprotocol Challenge A

- Infrastructure and Phone Configuration
- Frame Relay
- EIGRP
- VLAN Setup
- DHCP
- Gateways
- MCGP PRI
- Codec Selection
- Call Admission Control
- Location CAC
- Call Routing
- Calling Restrictions
- Class of Service
- High Availability
- SRST
- AAR
- Hunt Groups
- Video
- Time of Day Routing

Estimated Time to Complete: 6 hours

Section 18 Topology



Section 18 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2

PODS 11-19: X=Last Digit of POD Number
PODS 20-22: X=Both Digits of POD Number
*****Server is not present on network – but configure everything as if it were***

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM/Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23**	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
*** X=Last Digit of POD number including pods 20-23**	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Section 18 Configuration Tasks

Infrastructure and Phone Configuration

1. Check all of the Frame Relay connections and OSPF connectivity between sites.

2. Configure voice and data VLANs based on **Table 2**.

3. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for the IP phones, VG-248 and 6608. For Branch 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

4. Manually register all phones to CallManager and assign directory number to devices based on **Tables 5, 6 and 7**.

5. Assume a Publisher exists (**Table 1**) and every phone is registered in a CCM Group first to a Subscriber, and then to the Publisher. Configure the keepalive interval between any IP Phone and the Publisher server CallManager to be set to 40 seconds.

Gateways

NOTE:

- Please note that clocking will be provided by the network for all locations.
- Please note that the PSTN at each site is sending the last 10 digits for incoming calls.
- Top-Down signaling should be used in all cases.

---

6. Configure PRI for one of the 6608 ports as defined in **Table 8**. Refer to **Table 13** for 6608 port assignment.

7. Configure MGCP PRI for the Branch 1 site. Refer to **Table 8** for PSTN configuration.

8. Configure E1 R2 for Branch 2. Use **Table 8** for PSTN details. Register to CallManager as an H323 gateway. H323 packets should be sourced from the voice sub-interface.

Codec Selection and Call Admission Control

9. Configure CallManager with bandwidth values such that Audio calls within each site are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between any of the sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Region and respective Device Pool.

10. Use CallManager CAC to limit the number of calls over the WAN between any site to one G729 audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

Hunt Groups and ToD Routing

11. Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring simultaneously when this DN is called. You may not use a shared line to accomplish this task.

12. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 11** for a time schedule.

Call Routing and Calling Restrictions

13. Configure the route patterns as follows:

911	Emergency
9.911	Emergency
9.[2-9]XXXXXX	Local
9.1XXXXXXXXXX	Long Distance
9.011!	International
9.011!#	International

14. Configure CallManager to provide Class of Service such that phone 1 can call everywhere, phone 2 can call 911 and local, other phones can call 911, local and long distance.

15. All calls from HQ and Branch 1 should go through local gateway only.

16. Local calls from Branch 2 will be routed through local gateway first and use 6608 as backup. International calls will be routed through 6608 first and use local gateway as backup. 911 and Long Distance calls will be routed through local gateway only.

17. When someone places a call from Branch 2 to the area code in HQ, calls should be hopped off using the 6608 and use local gateway as backup.

High Availability

18. Configure Survivable Remote Site Telephony for Branch 1. Allow maximum of 5 phones and 10 DNs.

19. Configure SRST to provide calling restriction as specified earlier.

20. Configure SRST such that PSTN caller can reach IP phones directly.

21. Any inbound unknown numbers should be redirected to 2003.

22. Configure the BR1 gateway such that users at Branch 1 can dial 4 digits to reach the HQ and Branch 2.

23. Configure AAR between the HQ and Branch 2.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

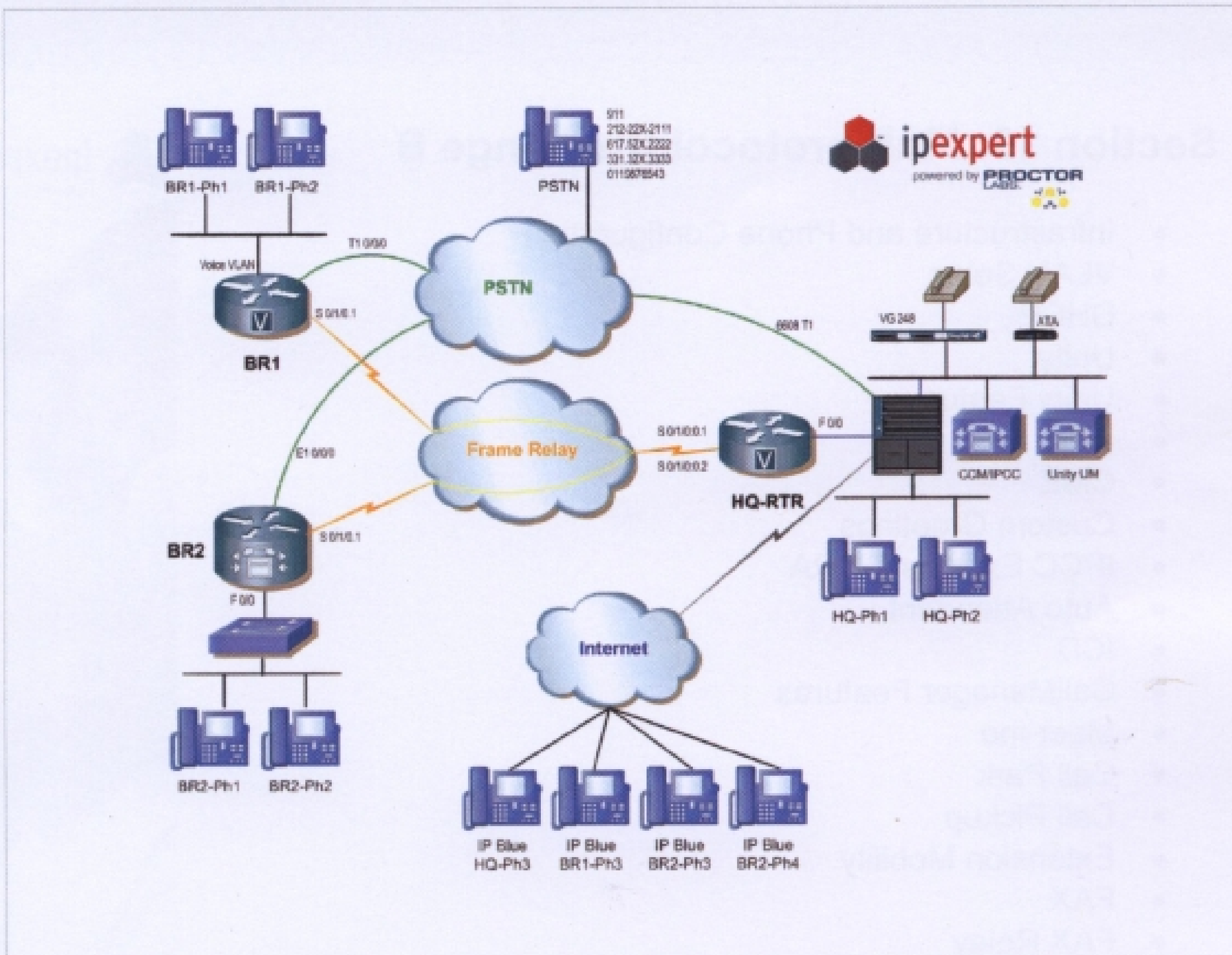
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 19: Multiprotocol Challenge B

- Infrastructure and Phone Configuration
- VLAN Setup
- DHCP
- Unity
- Unity Features
- MWI
- CME
- Custom Greetings
- IPCC Express – CRA
- Auto Attendant
- ICD
- CallManager Features
- Meet-me
- Call Park
- Call Pickup
- Extension Mobility
- FAX
- FAX Relay

Estimated Time to Complete: 6 hours

Section 19 Topology



Section 19 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number**	
***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
** X=Last Digit of POD number including pods 20-23*	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
** X=Last Digit of POD number including pods 20-23*	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Section 19 Configuration Tasks

Infrastructure

1. Configure each switch port connecting to IP phones and ATA to support both data and voice traffic according to VLAN IDs specified in **Table 2**.

2. In the HQ site, configure MS DHCP server to enable DHCP service for IP phones to get IP addresses and download configurations and phone loads.

3. In each branch site, configure each router/gateway to provide DHCP service. Configure the lease time to be 8 hours. Exclude the IP address of X.X.X.1 and X.X.X.254.

4. Configure all the phones with 4 digit extensions as shown in **Table 5** through **7** (**NOTE:** Branch 2 router should be configured as CME). Configure all the phones such that each phone displays its own full E.164 number on its LCD display.

5. Configure all the phones in Branch 1 and HQ to forward calling name and number. When calling PSTN, each phone should forward full DID DN to PSTN phone and calling names. The PSTN gateway is sending10 digits to all gateways.

6. Configure the CallManager and Branch 2 CME to synchronize its clock to the NTP authoritative source located on the PSTN-WAN router (**Table 1**).

Cisco Unity

7. Setup Unity voice mail boxes for Phone 1 in HQ, and Branch 1. Ensure you can call and reach mailboxes as well as leave messages. Ensure you can light the MWI light.

   - Pilot Number: 1600
   - Unity Port Number: 1601 – 1604
   - Set the default password to 54321
   - MWI Light – 1998 off
   - MWI Light – 1999 on

8. Integrate the Branch 2 CME with Unity with the following information and provide MWI and voice mail boxes for Phone 1 and 2 in the Branch 2 CME.

    - Pilot Number: 3600
    - Unity Port Number: 3600 – 3603
    - Set the default password to 54321
    - MWI Light – 3998 off
    - MWI Light – 3999 on

9. Configure each mail box such that when a subscriber logs in for the first time, the subscriber is not asked to change the password or to do self-enrollment.

10. For Unity integration, configure such that only the last Voice Mail port is used to send out a MWI notification.

11. Configure Unity such that MWI is synchronized daily at 12:00AM.

12. Set the minimum extension length for locations to be 4 and set the maximum recording time to be 5 minutes.

13. Configure CM and Unity such that when a caller dials 1570, the caller should hear "Welcome to IPexpert" in the greeting, and then be transferred to Phone 1 in the HQ. When transferred to Phone 1, the caller should be asked to announce the name of caller and have the Phone 1 subscriber to reject or accept the call.

14. Configure Unity such that call transfers from Unity are allowed only for 4 digit internal numbers. Assume that this is not done by default.

15. Configure CM and Unity such that if a caller dials 8 plus 4 digit extension, the call goes directly to the mailbox of the subscriber instead of ringing the phone.

16. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

IPCC Express – Customer Response Applications

17. Configure Auto-attendant and ICD with the following information:

    - AA Route Point = 1710
    - Script = aa.aef
    - CTI Ports = 1711, 1712

    - ICD Route Point = 1700
    - Script = icd.aef
    - CTI Ports = 1701, 1702

    Create the following users that will be used in this task:

    - "**jtapi**" to be used for the JTAPI Subsystem.
    - "**rmjtapi**" to be used for the ICD Subsystem.
    - "**telecaster**" to be used for the ICD Subsystem and enterprise data.
    - "**crsadmin**" which must be the designated administrator for IPCC Express.

- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine)
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ)

18. Modify the "icd.aef" script with the following information (you may wish to rename it to avoid loosing the original script)

    - You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'

    - If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds

    - Regardless of whether the call is routed to a specififc agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue')

19. Configure IPCC Phone Agent for both phones.

CallManager Features

20. Configure CM such that Phone 2 in the HQ cannot do 'call forward all' to international numbers.

21. Configure a meet-me conference with the meet-me number of 1900. Use appropriate 6608 port specified in **Table 13** as a conference bridge.

22. Configure CM such that an incoming call can be parked and picked up from another phone. Use the call park number of 12345 – this number should be the same from both CMs.

23. There is a user in the organization the travels frequently to Branch 1. Configure Extension mobility for this user. The username will be ad, password is abcde. Ensure that his/her number will be the following at each of the sites: 1005 and 2005 for HQ and Branch 1 respectively. Configure EM such that the user be automatically logged out after 4 hours and the user be able to log out any time. Ensure that at each site that when the user dials 911 that the call goes out to the local gateway.

24. Configure Attendant Console with the pilot number of 1550. Callers should be able to dial 0 as well as 1550 and reach 1003, and 2003. Enable Circular Hunting within the AC huntgroup. Also ensure that if there are up to 20 callers simultaneously calling 1550, and both IP Phones are presently taking calls, that callers will not be dropped but any number greater that 20 callers will be dropped and not sent to VM. Also, a call should not be dropped no matter how long it remains waiting.

Fax

25. Configure FAX relay between HQ and Branch 1 with the fax rate of 7200 bps.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

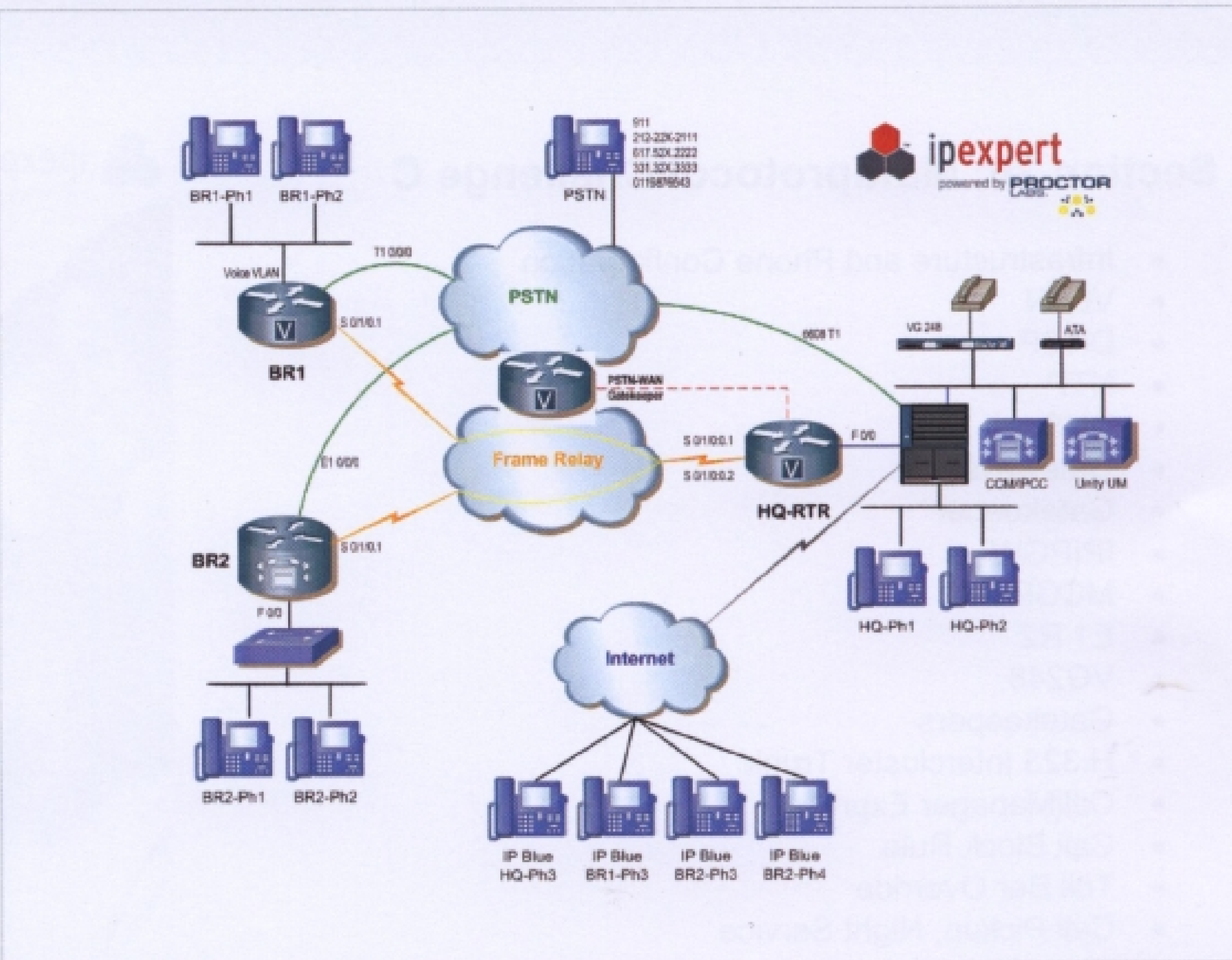
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 20: Multiprotocol Challenge C

- Infrastructure and Phone Configuration
- VLAN
- DHCP
- NTP
- DNS
- Gateways
- Gatekeeper
- IPIPGW
- MCGP PRI
- E1 R2
- VG248
- Gatekeepers
- H.323 Intercluster Trunk
- CallManager Express
- Call Block Rule
- Toll Bar Override
- Call Pickup, Night Service
- Cisco Unity
- Distribution Lists
- CUE
- MWI

Estimated Time to Complete: <u>6 hours</u>

Section 20 Topology



Section 20 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
****PODS 11-19: X=Last Digit of POD Number*** ***PODS 20-22: X=Both Digits of POD Number* ***Server is not present on network – but configure everything as if it were*		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
**PODS 11-19: X = Single Last Digit of POD Number *PODS 20-22: X = BOTH Digits of POD Number*	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
** X=Last Digit of POD number including pods 20-23*	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
** X=Last Digit of POD number including pods 20-23*	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Section 20 Configuration Tasks

Infrastructure and Phone Configuration

1. Configure the Voice and Data VLANs for all locations detailed in **Table 2**.

2. Provide DHCP services for IP phones and gateways for all locations using IOS DHCP service only. Configure the Unity Server as the DNS service; be sure to use this for all name resolution.

Location	IP Range
HQ*	10.X.200.10-20/24
Branch 1*	10.X.201.10-20/24
Branch 2*	10.X.202.10-20/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

3. Configure time synchronization for CallManager using the NTP protocol. Configure the backbone router address (**Table 1**) as your NTP source address.

4. Configure HQ and Branch 1 Phones to register with CallManager and configure the appropriate 4 digit extensions. **Tables 5** and **6** provide the phone information.

Gateways and Gatekeepers

NOTE:

Each location has connectivity to the PSTN. Clocking (Layer 1) should be setup to receive from the network. Use channels 1-3 for each site. PSTN calls should display Calling Name and Calling Number. PSTN will send 10 digits to all gateways. Make the appropriate gateway configuration to receive the calls. Use 9 to access the PSTN, but be sure to strip the 9 before sending to the PSTN.

---

5. Configure the Headquarters 6608 port for MGCP T1 PRI, line coding B8ZS and framing is ESF. Use **Table 13** for port allocation.

6. Configure Branch 2's E1 controller for CRC4 framing and HDB3 encoding. Configure the signaling for R2 ITU Q421 DTMF. Use CAS Custom to understand how to specify the country code.

7. Configure the Branch 1 T1 controller for MGCP PRI. Switch-type NI2 and line coding is B8ZS and framing is ESF.

8. Configure the gatekeeper with the following parameters:

   Local zone= HQ-RTR
   domain name = ipexpert.com
   [use loopback interface for local zone]

   Remote zone= PSTN-WAN
   domain name= ipexpert.com
   ip address= 10.X.200.2 [X=Last digit of POD number]

   Register CallManager to the HQ-RTR gatekeeper.

   Calls from HQ and BR1 destined for BR2 should be sent out of the gatekeeper. Full E164 number should be sent to gatekeeper including an International prefix (see **Table 12**).

9. Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 7** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed to CCM via SIP using G711ulaw (see **Tables 5** and **6** for DNs at CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

CallManager Express

10. Configure the router in Branch 2 for CallManager Express. Register the devices and configure the DNs described in **Table 7**.

11. Configure a Call Block rule for all CME phones. Block all 900 and 976 calls. Create a user "IPExpert" and allow that user to override the block using Toll Bar Override. Deactivate the user's login after 25 minutes when the phone becomes idle.

12. Create a Call Pickup group for both IP phones using 55 for the group number.

13. Configure both CME IP Phones with night service. Define the night service period of Monday through Friday 5:01pm to 7:59am and all weekend. Allow phone 1 to enter a code of *123 to manually disable it.

Unity

14. Configure Voice Mail Integration for CallManager. The information is provided below.

    - Pilot Number: 1600
    - Unity Port Number: 1601 – 1604
    - Set the default password to 54321
    - MWI Light – 1998 off
    - MWI Light – 1999 on

Unity Express

15. Configure the BR2 router to support the CUE module using information from **Table 16**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate user's extensions and mailboxes.

    Integrate CME into Unity Express using the same information as follows:

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    Finally, setup mailboxes for all 3 phones at BR2.

16. Configure all IP Phones with Unity or Unity Express Voicemail (depending on site). Use names that are descriptive to the site (e.g. HQ Phone1).

17. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

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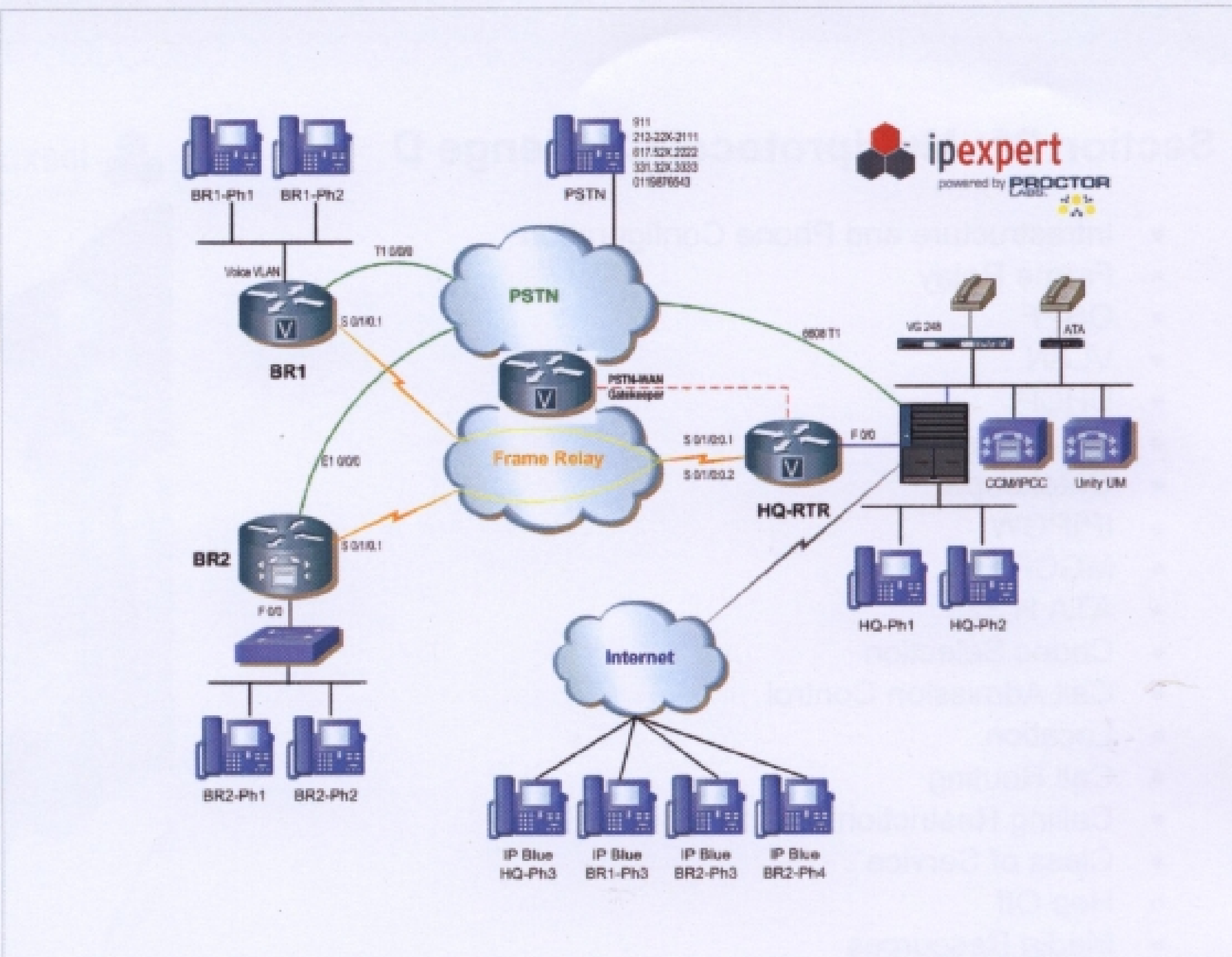
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 21: Multiprotocol Challenge D

- Infrastructure and Phone Configuration
- Frame Relay
- OSPF
- VLAN
- DHCP
- Gateways
- Gatekeeper
- IPIPGW
- MGCP PRI
- ATA H.323
- Codec Selection
- Call Admission Control
- Location
- Call Routing
- Calling Restriction
- Class of Service
- Hop Off
- Media Resources
- MOH
- Conference
- CallManager Features
- Video
- Attendant Console
- IPMA
- Callback
- FAX
- FAX Relay

Estimated Time to Complete: <u>6 hours</u>

Section 21 Topology



Section 21 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number**	
***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23**	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
*** X=Last Digit of POD number including pods 20-23**	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Section 21 Configuration Tasks

Infrastructure and Phone Configuration

1. Check all of the Frame Relay connections and connectivity between sites.

2. Check all basic OSPF has been set up correctly – If not assign all interfaces to area 0 and verify the routing.

3. Configure both voice and data VLANs based on **Table 2**.

4. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for the IP phones, VG-248 and 6608. For Branch 2, use IOS DHCP to allocate IP address for the IP phones.

5. Configure CallManager to allow ALL devices to register. Assign directory number to devices based on **Tables 5, 6** and **7**.

6. Configure ATA-186 with H323 firmware image and allocate appropriate DN. Configure such that line 1 can use G711 and G729.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

---

7. Configure PRI for the allocated 6608 port as follows. Refer to **Table 13** for 6608 port assignment.

    - Linecode – B8ZS
    - Framing – ESF
    - PRI Protocol - PRI NI2
    - Protocol Side - User
    - Channel Selection - Top Down

8. Configure MGCP PRI for Branch 1

    - Linecode – B8ZS
    - Framing – ESF
    - PRI Protocol - PRI NI2
    - Protocol Side - User
    - Channel Selection - Top Down
    - Use the first 3 channels only

9. Configure E1 R2 for Branch 2. Use **Table 8** for PSTN details. Register to CallManager as an H323 gateway. H323 packets should be sourced from the voice sub-interface.

10. Configure the HQ router as a gatekeeper to perform call routing from CallManager to ATA-186. Use the zone name – HQ-RTR. Register ATA to the gatekeeper (See **Table 5**).

Codec Selection and Call Admission Control

11. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

12. Configure CallManager to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

Call Routing and Call Restrictions

13. Configure the dial plan as follows:

911	Emergency
9.911	Emergency
9.[2-9]XXXXXX	Local
9.1XXXXXXXXXX	Long Distance
9.011!	International
9.011!#	International

14. Configure CallManager to provide Class of Service such that phone 1 can call everywhere, phone 2 can call 911 and local, other phones can call 911, local and long distance.

15. 4 digit dialing should be preserved to reach the ATA from all other sites. Calls will be sent to the gatekeeper for call routing.

16. For Branch 1, local calls should be sent to the local gateway and then use 6608 as backup. All other calls should be sent to local gateway only. For HQ, all calls should be sent to local gateway only.

17. Local calls from Branch 2 will be routed through local gateway first and use 6608 as backup. International calls will be routed through 6608 first and use local gateway as backup. 911 and Long Distance calls will be routed through local gateway only.

18. When someone places the calls from Branch 2 to the area code in HQ, calls should be hopped off using the 6608 and use local gateway as backup.

Media Resources

19. Based on **Table 13**, configure one of the 6608 ports as a transcoder.

20. Based on **Table 13**, configure one of the 6608 ports as a conference bridge.

21. Configure media resources using DSPs on Branch 1 gateway. Configure two transcoder and one conference bridge sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

22. Configure the MoH server to support both G.711 and G.729 region.

23. Configure the MoH server and the network to support multicast music on hold. HQ should receive unicast MoH, Branch 1 should get multicast MoH and PSTN caller should get tone on hold.

24. Branch 1 IP phones should receive music stream from the IOS gateway – not from CCM.

25. Configure media resource redundancy for the HQ so it will use 6608 as Conference Bridge and BR1 Conference Bridge as backup.

CallManager Features

26. Configure the Attendant Console as follows:

    - Pilot Point Number: 1550
    - Members: 1001, 2001 and 2002

    - Users cannot call 1550 directly. They need to dial 0 instead to reach these members. Calls should be distributed in a circular fashion.

27. Configure the IPMA for CallManager as follows:

Manager Primary Extension	2003
Manager Intercom	2333
Speeddial to Assistant Intercom	

Assistant Primary Extension	1003
Assistant Proxy Line	1333
Assistant Intercom	1334
Speeddial to Manager Intercom	

    The IPMA Assistant Console can be installed on the Publisher.

    Make sure that the IPMA can intercept the calls to the manager's primary extension.

    Calls from HQ, Branch 2 and PSTN should be handled by the assistant. Calls from Branch 1 should be able to reach manager directly.

28. Make sure CallBack service is available for all phones.

29. If a call rings into HQ Phone 3, that user must have the option of sending that call to VM without exhausting the CallFwdNoAn timer.

30. Ensure that missed calls coming in to an IP Phone do not need any user intervention in order for them to be redialed as an outgoing call (you may not use a translation pattern to accomplish this task).

Fax

31. Configure Cisco FAX relay for 6608, VG248 and Branch 2 gateway.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

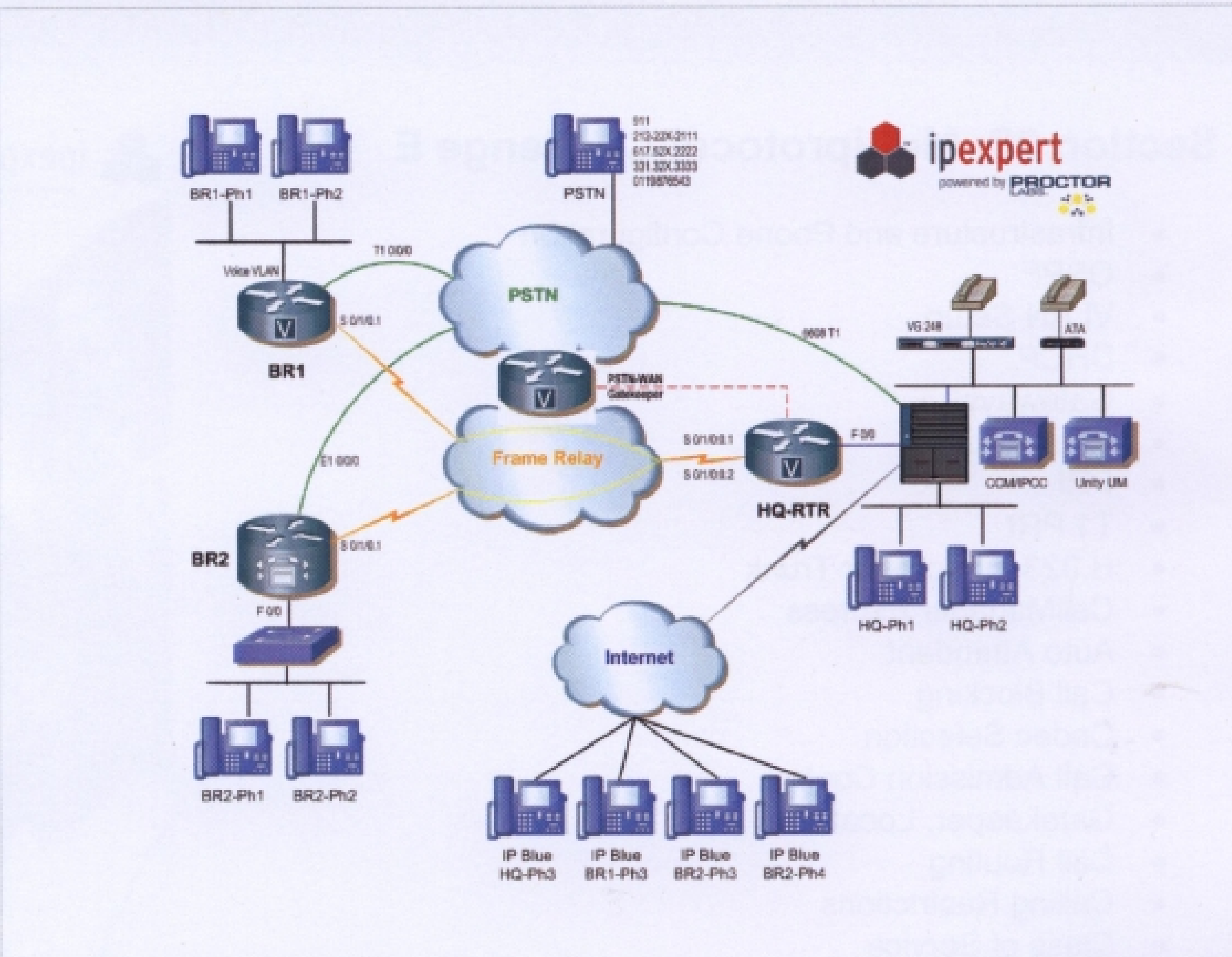
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 22: Multiprotocol Challenge E

- Infrastructure and Phone Configuration
- OSPF
- VLAN Setup
- DHCP
- Gateways
- Gatekeepers
- PRI
- T1 PRI
- H.323 Intercluster Trunk
- CallManager Express
- Auto Attendant
- Call Blocking
- Codec Selection
- Call Admission Control
- Gatekeeper, Location
- Call Routing
- Calling Restrictions
- Class of Service
- Media Resources
- Conf Bridge
- Video
- MOH
- CME MOH
- BACD
- QoS
- LLQ
- FRF.12
- IPCC Express – CRS
- Custom Application
- FAX
- FAX Passthrough

Estimated Time to Complete: <u>8 hours</u>

Section 22 Topology



Section 22 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number		
PODS 20-22: X=Both Digits of POD Number		
***Server is not present on network – but configure everything as if it were*		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
****PODS 11-19: X = Single Last Digit of POD Number* **PODS 20-22: X = BOTH Digits of POD Number***	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
**** X=Last Digit of POD number including pods 20-23***	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1...
Branch 1*	617-52X-2...
Branch 2*	011-331-32X-3...
**** X=Last Digit of POD number including pods 20-23***	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Section 22 Configuration Tasks

Infrastructure

1. Check and configure OSPF as a routing protocol in the network so that all the router interfaces including loopback interfaces are reachable from anywhere in the network.

2. Configure each switch port connecting to IP phones and ATA to support both data and voice traffic according to VLAN IDs specified in **Table 2**.

3. In the HQ site, configure MS DHCP server to enable DHCP service for IP phones to get IP addresses and download configurations and phone loads.

4. In each branch site, configure each router/gateway to provide DHCP service. Configure the lease time to be 8 hours. Exclude the IP address of X.X.X.1 and X.X.X.254.

5. Configure all the phones in HQ and Branch 1 with 4 digit extensions as shown in **Tables 5** and **6**. Configure all the phones such that each phone displays its own full E.164 number on its LCD display.

6. Configure the Branch 2 router as a CME and register all the phones based on **Table 7**. When calling PSTN, each phone should forward full DID DN and calling name to the PSTN phone.

7. Configure CallManager to be synchronized to the NTP authoritative source which is the backbone PSTN gateway (**Table 1**). Use the correct time zone (EST) for the server.

8. Configure a user to associate with the HQ phone 1 and use the web interface to configure a speed dial on Phone 1 to dial 212-22X-1111.

9. Configure Phone 1 and Phone 2 in the HQ site to share an extension of 1101 so that an incoming call to this extension will ring both phones.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

10. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

11. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed.

12. Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

13. Configure BR2 as a Gateway based on the information in **Table 8**.

14. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    [use loopback interface for local zone]

    Remote zone= PSTN-WAN
    domain name= ipexpert.com
    ip address= 10.X.200.2 [X=Last digit of POD number]

    Register CallManager to the HQ-RTR gatekeeper.

CallManager Express

15. Register phones to CME based on **Table 7**. Configure a second line of 3005 on Phone 1. If the first line whose DN is 3001 is busy, the call should roll to the second line.

16. Configure the CME phone LCD to display "Welcome to IPexpert".

17. Configure CME such that outbound calls to 1 900 XXX-XXXX from Phone 1 and 2 are blocked.

18. Configure CME such that outbound calls from Phone 2 do not forward caller ID. All other phones from CME should forward caller IDs.

19. Configure the inter-digit timeout to be 3 seconds on CME.

20. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 and 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member.

21. Create an incoming AutoAttendant at the DN of 3000. Also Create a Basic ACD using the support team hunt group you just created (The necessary TCL scripts are already loaded in BR2 router's flash memory). Have the AA script automatically hand-off the callers into the support ACD hunt group when a user presses 2. Allow no more than 20 callers in the Q at any one point. Play a prompt for the user every 30 seconds to let them know that all agents are busy. Allow the users to dial-by-extension by pressing 4. Ensure that the Q is collecting statistics and view them as part of your troubleshooting.

Codec Selection and Call Admission Control

22. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 10 calls at the minimum bandwidth supported by a Cisco VTA camera in H263 mode and between each site with values such that Video calls be allowed to use 2 calls at the minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

23. In Branch 2, use G729 codec when placing calls to any of the other sites.

24. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to 3 audio calls and 2 video calls using the minimum bandwidth allowable by a VTA camera in H263 mode.

25. Allow only four concurrent G.711 calls through the HQ site gatekeeper from anywhere.

Call Routing

26. Configure Calling Restriction using the **Table** below.

Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

27. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

28. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

29. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out of the local gatekeeper with the appropriate local gateway acting as backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. **NOTE:** The gatekeeper is expecting the full E164 number; i.e. international prefix '011' plus 10 digits.

30. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway. You must utilize 4 different types of methods to manipulate the digits sent to the PSTN (e.g. one method is by using the 'forward-digits' command inside the POTS dial-peer).

31. Calls originating from BR2 to CallManager (both HQ and BR1) should use VoIP and PSTN as backup. If you decide to register the CME to the gatekeeper use your local HQ-RTR gatekeeper as opposed to the PSTN-WAN gatekeeper. 4-digit dialing must be preserved.

32. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

High Availability

33. Configure in CallManager such that if there is not enough bandwidth to make calls to and from Branch 1, use PSTN to complete the calls.

34. Setup SRST on the Branch 1 gateway such that when the CallManager or WAN is down, subscribers can still make calls to PSTN. Calls from PSTN should still work. Configure the SRST gateway such that subscribers can still dial 4 digit extensions to reach other sites.

Media Resources

35. Configure the appropriate 6608 port as a conference bridge. Configure a conference bridge on the Branch 1 router: Allow 4 Sessions. Use **Table 13** for port allocation.

36. Configure MoH server on the CallManager such that multicast MoH is provided to phones in Branch 1.

37. Configure MoH on the Branch 2 CME such that PSTN phones calling into BR2 phones should get music when calls from or to CME are placed on hold.

Quality of Service

38. In the HQ site Cat 6500, mark the DSCP PHB label as CS3 for SCCP traffic between IP phones and CM.

39. In the Branch 2 Cat 3550, mark the DSCP PHB label as EF for RTP traffic from the port connected to IP phones.

40. Map CoS 3 to CS3 and CoS 5 to EF on 3550 in the Branch 2 site. Configure the appropriate priority queuing for the CoS 5 traffic on the phone ports of the Branch 2 switch.

41. The link speed between HQ and Branch 2 is 256 Kbps. Employ Traffic Shaping as well as a well-known technique to minimize serialization delay. There is no need to do this on the link between HQ and Branch 1 since its link speed is 1544kbps.

42. From HQ Site to Branch 2, reserve 50% of the bandwidth for voice RTP traffic on a priority queue and reserve 5% minimum guaranteed bandwidth for voice control traffic.

43. From Branch 2 to HQ Site, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 12kbps for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queueing.

44. From the HQ site to the Branch 1 site, reserve 50% of bandwidth for voice RTP traffic on a high priority queue and 5 % of bandwidth for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queueing.

IPCC Express – Customer Response Applications

45. Create a custom application with the HQ CM such that when a caller from PSTN places a call to 1202, the application should ask the caller to enter a 4 digit PIN. If the entered PIN matches "1234", route the call to Phone 2 in the HQ site.

Fax

46. Configure FAX pass through between the HQ VG248 FAX and Branch 1 FAX phones.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
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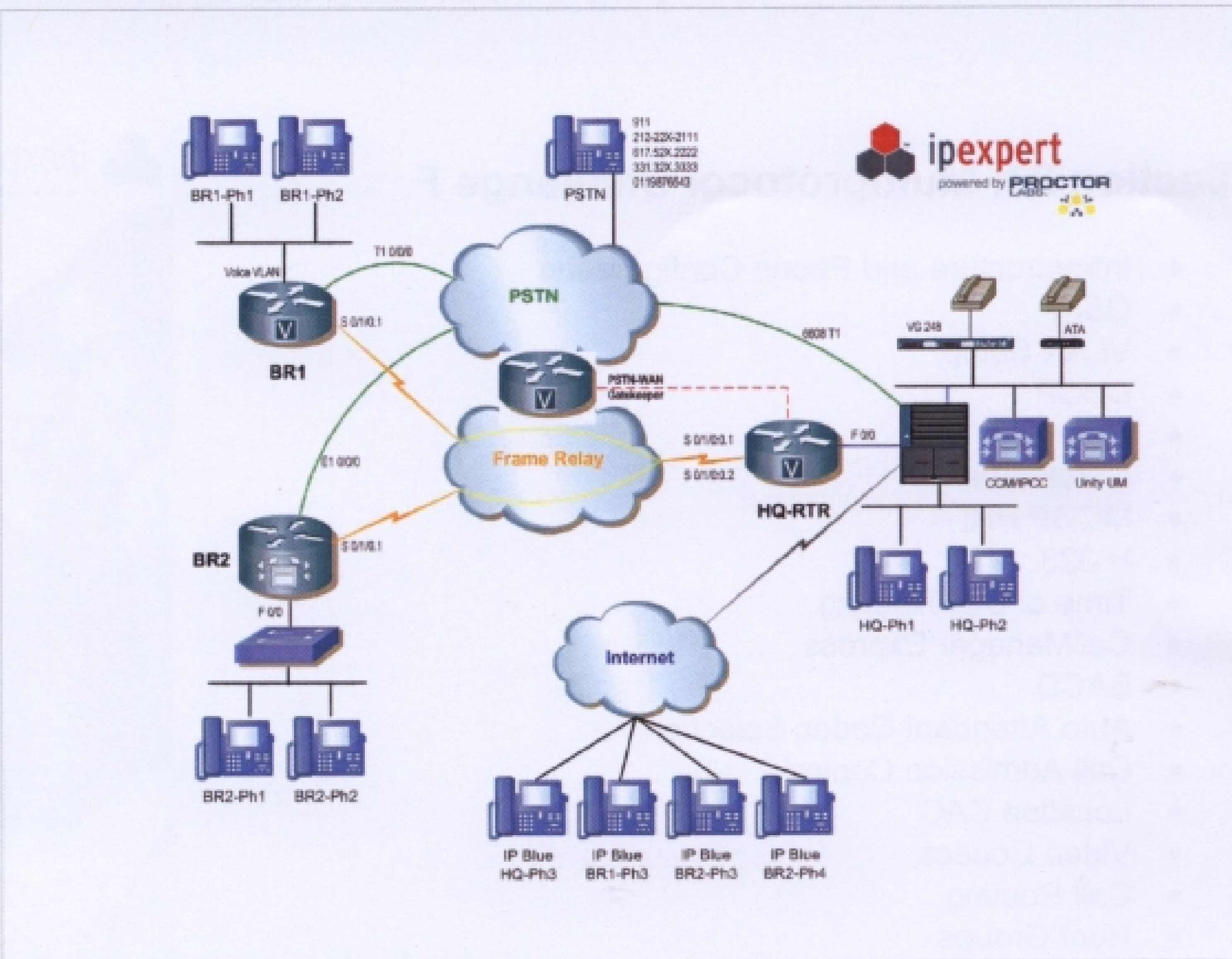
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 23: Multiprotocol Challenge F

- Infrastructure and Phone Configuration
- OSPF
- VLAN Setup
- DHCP
- Gateways
- Gatekeeper
- MCGP PRI
- H.323
- Time of Day Routing
- CallManager Express
- BACD
- Auto Attendant Codec Selection
- Call Admission Control
- Location CAC
- Video Codecs
- Call Routing
- Hunt Groups
- Call Coverage
- Calling Restriction
- Route Filters
- Media Resources
- Conferencing
- Transcoding
- MOH
- QoS
- CoS to DSCP, LLQ
- Cisco Unity
- Live Reply
- Optional Conversation
- CUE
- VPIM Networking
- CallManager Features
- Personal Address Book
- Fast Dial IP Phone Services

Estimated Time to Complete: <u>8 hours</u>

Lab 23 Topology



Section 23 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
** X=Last Digit of POD number including pods 20-23*	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
** X=Last Digit of POD number including pods 20-23*	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Section 23 Configuration Tasks

Infrastructure and Phone Configuration

1. Check and configure basic OSPF – If not configured assign all interfaces to area 0 using the IP Network addresses found in **Table 1**. Configure Loopback interfaces on each router from **Table 9**.

2. Configure the Voice and Data VLANs for all locations detailed in **Table 2**.

3. Provide DHCP services for IP phones and gateways at the Headquarters location using the CallManager Microsoft DHCP service. For Branch 1 and 2, use the local routers for IOS DHCP service. Configure the lease time to be 168 hours at each site.

Location	IP Range
HQ*	10.X.200.5-15/24
Branch 1*	10.X.201.5-15/24
Branch 2*	10.X.202.5-15/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

4. Register HQ and BR1 Phones to CallManager and configure with the appropriate 4 digit extensions. **Tables 5** and **6** provide the phone information.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

5. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

6. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed.

7. Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

8. Configure BR2 as an H323 gateway based on the information in **Table 8**.

9. Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 7** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed to CCM via SIP using G711ulaw (see **Table 5** and **6** for DNs at CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

10. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    [use loopback interface for local zone]

    Remote zone= PSTN-WAN
    domain name= ipexpert.com
    ip address= 10.X.200.2 [X=Last digit of POD number]

    Register CallManager to the HQ-RTR gatekeeper.

CallManager Express

11. Configure the router in Branch 2 for CallManager Express. Register the devices and configure the DNs described in **Table 7**. Enable the ability to add directory numbers through the web interface.

12. Create an incoming AutoAttendant at the DN of 3000. Also Create a Basic ACD using the support team hunt group you just created (The necessary TCL scripts are already loaded in BR2 router's flash memory). Have the AA script automatically hand-off the callers into the support ACD hunt group when a user presses 2. Allow no more than 20 callers in the Q at any one point. Play a prompt for the user every 30 seconds to let them know that all agents are busy. Allow the users to dial-by-extension by pressing 4. Ensure that the Q is collecting statistics and view them as part of your troubleshooting.

Codec Selection and Call Admission Control

13. Configure the gatekeeper to allow only Four G.729 calls to the remote zone.

14. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

Call Routing and Calling Restrictions

15. Configure the following Call Restrictions

Restricted	Internal 911 Toll By-pass
Local and Long Distance	Internal 911 Toll By-pass Local Long Distance
International	Internal 911 Toll By-pass Local Long Distance International

16. Configure the Phones with the following Class of Restrictions (COR).

Headquarters

Name	**Restrictions**
Phone 1	International
Phone 2 (ATA)	Long Distance
IP Phone 3 (IP Blue)	Long Distance
VG248	Restricted

Branch 1

Name	Restrictions
Phone 1	International
Phone 2	Restricted
Phone 3	Long Distance

Branch 2

Name	Restrictions
Phone 1	International
Phone 2	Long Distance
Phone 3	Restricted

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

17. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

18. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

19. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed first out of the local Gatekeeper and then through the HQ-IPIPGW as a backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. (**Test both** call routing methods, one at a time, before moving on to the next question). **NOTE:** The gatekeeper is expecting the full E164 number, i.e. international prefix '011' plus 10 digits.

20. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway. You must 4 different types of methods to manipulate the digits sent to the PSTN (e.g. one method is by using the 'forward-digits' command inside the POTS dial-peer).

21. Calls originating from BR2 to CallManager (both HQ and BR1) should use VoIP and PSTN as backup. If you decide to register the CME to the gatekeeper use your local HQ-RTR gatekeeper as opposed to the PSTN-WAN gatekeeper. 4-digit dialing must be preserved.

22. Enable DID for PSTN users dialing into the CallManager (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

23. All users with a class of service that includes access to PSTN numbers should be given access to Toll Free Services. Block access to all 1-900 and 1-976 numbers.

24. Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring when this DN is called. You may not use a shared line to accomplish this task. The call should alternate back and forth between these 2 phones equally.

Time of Day and extended Time of Day Call Coverage Routing

25. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 11** for a time schedule. Also, all members of this HuntGroup should get the message in their VM box and see their MWI for any message left for 1005 (once Unity is configured).

26. Create a new DN of 1010. When this DN is called, all calls should be directed to the DN of 1005. The call should ring this DN for approximately 2 rings. If for any reason this DN does not answer, the call should then be forwarded to VM if the call originated from an internal number, however if the call originated from an external number, the call should then be forwarded to the PSTN phone at '221-1111'.

Media Resources

27. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

28. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

29. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

30. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

31. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should load share the IOS resources and 6608 resources.

32. Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source. You may not configure music sources on line, device or device pool to accomplish this task.

33. Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You must use the transcoder to achieve this task.

34. Ensure that the HQ and BR1 phones receive multicast music on hold.

35. Configure music on hold on the BR2 CME.

36. Create a meet-me conference with DN=1900.

Quality of Service

37. Configure the CoS-to-DSCP Mappings for the Catalyst 6500 and the Catalyst 3550 switches. The recommended settings are DSCP of CS3 for VoIP control plane and DSCP of EF for VoIP bearer plane.

38. Configure video traffic so it is mapped from the DSCP value AF41 to the 802.1Q CoS value 4. Apply this policy only to the data/video VLAN FE sub-interface in Branch 2.

39. Configure FRTS between HQ and BR2 such that the traffic shaper only engages when Voice traffic is present on the link. For this task you may assume that the FR port speed is 768kbps and that the CIR provided by the carrier is 384. A proper LFI mechanism should be engaged at all times and should be relevant to the CIR not the Port speed.

40. Configure LLQ between Headquarters and Branch 2 location. From Headquarters to Branch 2 configure 32kbps control traffic and 256kbps bearer traffic. From Branch 2 to Headquarters configure 6% control traffic and 40% bearer traffic. All other traffic should be given a fair-queue configuration in software.

Unity

41. Configure Voice Mail Integration for CallManager. The information is provided below.

    - Pilot Number: 1600
    - Unity Port Number: 1601 – 1604
    - Set the default password to 54321
    - MWI Light – 1998 off
    - MWI Light – 1999 on

42. Subscribers have recently migrated from Octel Voicemail system to Unity and expect Unity to have the same Message Retrieval Menus. Configure Unity such that when users retrieve messages the prompts will be transparent.

43. Configure a subscriber mailbox for phone 2 at the HQ and BR1 locations. Record the subscriber's greeting to say "I am not in the office right now. If you need to reach me press 3 to be transferred to my cell phone". Have callers transferred from Voicemail to the External Numbers listed below.

    - HQ phone 2 transfers to 212-22X-1111
    - BR1 phone 2 transfers to 617-52X-2222

44. Enable the Live Reply feature in Unity. Allow the subscriber to call back another subscriber immediately after listening to the voice message, through the Telephone User Interface (TUI). Use the first line for phone 1 at Headquarters and Branch 1 locations.

Unity Express

45. Configure the BR2 router to support the CUE module using information from **Table 16**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate user's extensions and mailboxes.

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    Finally, setup mailboxes for all Phone 2 at BR2.

46. Ensure that if a call is forwarded from any one of the 3 phones for reasons of No-Answer or Busy that the call goes to the appropriate mailbox.

47. Ensure that when users listen to their voicemail messages, that they hear the ANI announced of the phone who left them the message.

48. Network CUE with Unity. Allow messages to be seamlessly forwarded back and forth.

CallManager Features

49. Configure the Personal Address Book and Fast Dial IP Phone Services on CallManager. Have all phones at this Branch 1 subscribe to this service. Use the Cisco CallManager Lightweight Directory Access Protocol (LDAP). Enter three IP Phones user information from the Headquarters site into the address book. Use names that are descriptive to the site (Example: HQ Phone1 / 212-22X-1001).

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

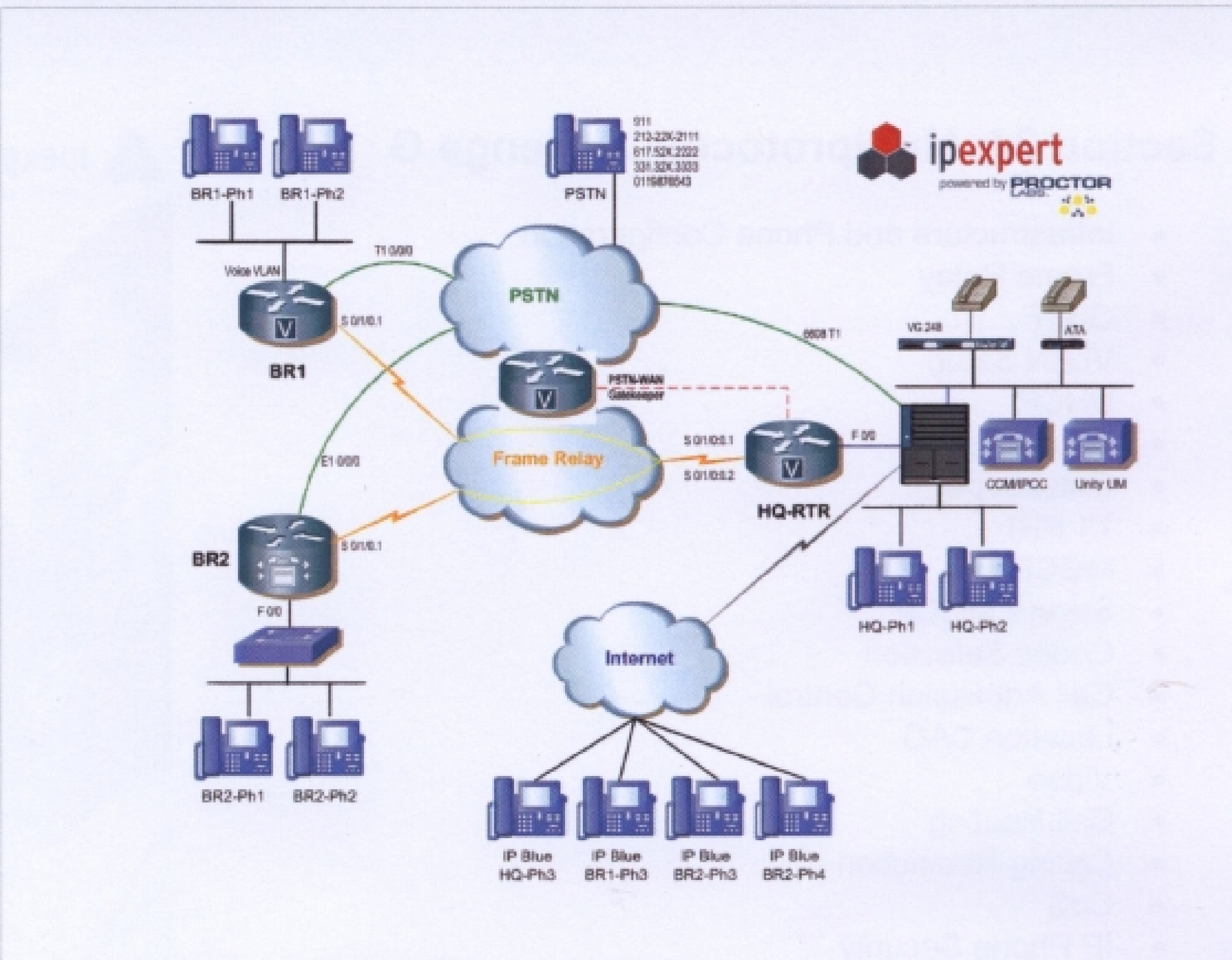
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 24: Multiprotocol Challenge G

- Infrastructure and Phone Configuration
- Frame Relay
- OSPF
- VLAN Setup
- DHCP
- Gateways
- Gatekeeper
- T1 PRI
- MGCP PRI
- Secure MGCP
- Codec Selection
- Call Admission Control
- Location CAC
- Video
- Call Routing
- Calling Restriction
- CoS
- IP Phone Security
- Multi-Tenant Overlapping
- High Availability
- SRST
- Secure SRST
- Hunt Group
- AAR
- Media Resources
- Conf Bridge
- Transcoding
- MOH
- Cisco Unity
- Secure Unity
- MWI
- Auto Attendant
- CallManager Features
- Extension Mobility

Estimated Time to Complete: <u>10 hours</u>

Lab 24 Topology



Section 24 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2

PODS 11-19: X=Last Digit of POD Number
PODS 20-22: X=Both Digits of POD Number
*****Server is not present on network – but configure everything as if it were***

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number *PODS 20-22: X = BOTH Digits of POD Number	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
X=Last Digit of POD number including pods 20-23	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1...
Branch 1*	617-52X-2...
Branch 2*	011-331-32X-3...
X=Last Digit of POD number including pods 20-23	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Lab 24 Configuration Tasks

Infrastructure and Phone Configuration

1. Check the configuration of basic Frame Relay connections.

2. Check the configuration of basic OSPF – all interfaces should be assigned to area 0. Verify the connectivity between sites.

3. Configure both voice and data VLANs based on **Table 2**.

4. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for the IP phones, VG-248 and 6608. For Branch 2, use IOS DHCP to allocate IP address for the IP phones.

5. Configure CallManager to register devices manually. Assign directory number to devices based on **Tables 5, 6** and **7**.

6. Ensure that Security based services are started – that tokens exist, and that all media and signaling traffic between HQ Phone 3 and BR1 Phone 3 are secured with 128bit AES encryption.

Gateway and Gatekeeper

7. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

8. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed.

9. Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

10. Ensure that RTP traffic between the any capable IP Phone and the BR1 MGCP gateway is encrypted with AES128.

11. Ensure that all MGCP signaling between CallManager and BR1 gateway is encrypted with 3DES 168bit encryption using a SHA hashing algorithm.

12. Configure BR2 as an H323 gateway based on the information in **Table 8**. Register the gateway to CallManager using the voice subinterface.

13. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    [use loopback interface for local zone]

    Remote zone= PSTN-WAN
    domain name= ipexpert.com
    ip address= 10.X.200.2 [X=Last digit of POD number]

    Register CallManager to the HQ-RTR gatekeeper.

Codec Selection and Call Admission Control

14. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

15. Use Locations CAC to limit the number of calls over the WAN between the HQ and BR1/BR2 to one audio call. Limit video to one call per site using the minimum bandwidth allowable by a VTA camera in H263 mode.

16. Ensure that when a user presses the 'Services' or 'Directories' button the error message is not displayed.

Call Routing and Calling Restriction

17. Configure the dial plan as follows:

911	Emergency
9.911	Emergency
9.[2-9]XXXXXX	Local
9.1XXXXXXXXXX	Long Distance
9.011!	International
9.011!#	International

18. Configure CallManager to provide Class of Service such that phone 1 can call everywhere, phone 2 can call 911 and local, other phones (if applicable) can call 911, local and long distance. Configure device and line CSS to achieve this configuration. You cannot restrict any calls from the device CSS level.

19. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

20. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

21. Calls originating from both the HQ and BR1 sites destined for any international number including the BR2 PSTN numbers should be routed out of the local gatekeeper with the appropriate local gateway acting as backup. The gatekeeper is expecting the full E164 number, i.e. international prefix '011' plus 10 digits.

22. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway.

23. Enable DID for PSTN users dialing into the CallManager (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

24. There are multiple tenants and overlapping extension scenario in the HQ and Branch 1 sites. Create a line with DN=1010 and assign to both HQ phone 3 and BR1 phone 3. Ensure that when HQ users dial 1010 HQ phone 3 will ring and when BR1 users ring 1010 BR1 phone 3 will ring. Also ensure that calls coming in from the PSTN destined for the overlapping DN will ring the local phone, for example calls coming in from the 6608 for 1010 will ring HQ phone 3.

High Availability

25. Configure Survivable Remote Site Telephony for Branch 1. Allow maximum of 5 phones and 10 DNs. Ensure that if BR1 gateway does go into fallback mode, that it has already created its own certificate, forwarded that certificate to CallManager and that CallManager in turn has installed that certificate into the IP Phones at BR1, so that signaling and media continue to be encrypted during SRST.

26. When "Message" button is pressed, users should be able to hear the general greeting from Unity.

27. Configure SRST to provide calling restriction as specified in question #7.

28. Configure Branch 1 gateway such that users at Branch 1 can dial 4 digits to reach the other sites.

29. Configure hunt group in SRST. Incoming calls will be routed to 2002 and then 2003.

30. Configure AAR between the HQ and Branch 2. Calls should only use the local gateway at each site (and not the gatekeeper).

Media Resources

31. Based on **Table 13**, configure one of the 6608 ports as a transcoder.

32. Based on **Table 13**, configure one of the 6608 ports as a conference bridge.

33. Configure media resources using DSPs on BR1 gateway. Configure two transcoder and one conference bridge sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

34. Configure the MoH server to support both G.711 and G.729 codecs.

35. Configure the MoH server and the network to support multicast music on hold. HQ should receive unicast MoH, Branch 1 and 2 should receive multicast MoH and PSTN should receive tone on hold.

Cisco Unity

36. Configure Voice Mail Integration for CallManager. The information is provided below.

    - Pilot Number: 1600
    - Unity Port Number: 1601 – 1604
    - Set the default password to 54321
    - MWI Light – 1998 off
    - MWI Light – 1999 on

37. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

38. Create 4 subscribers shown below in Unity using the Bulk Import Tool.

First Name	Last Name	Extension
John	Bee	1001
Bill	Dee	1003
Cindy	Tee	2003
Frank	See	3003

39. HQ needs Auto Attendant feature. Configure Unity to provide such functionality. Use 1570 as the extension for the call handler. User should hear a greeting saying "Press 1 for directory handler, 2 for HQ phone 1". When caller press 1, send the call to directory handler. When caller press 2, send the call to phone 1 at HQ. Any other entry should take the caller back to the original greeting.

40. Configure Unity so that MWI will be resynchronized at 3:00am every day.

CallManager Features

41. Configure Extension Mobility as follows:

- Username: ptw
- Password: gjmpt

- HQ Device Profile: 1666
- Branch 2 Device Profile: 3666

42. Configure Phone 3 at HQ and Phone 2 at Branch 2 to support Extension Mobility. User's calling restrictions should be 911 only. Call routing shall be inherited from the device the user is logged in.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

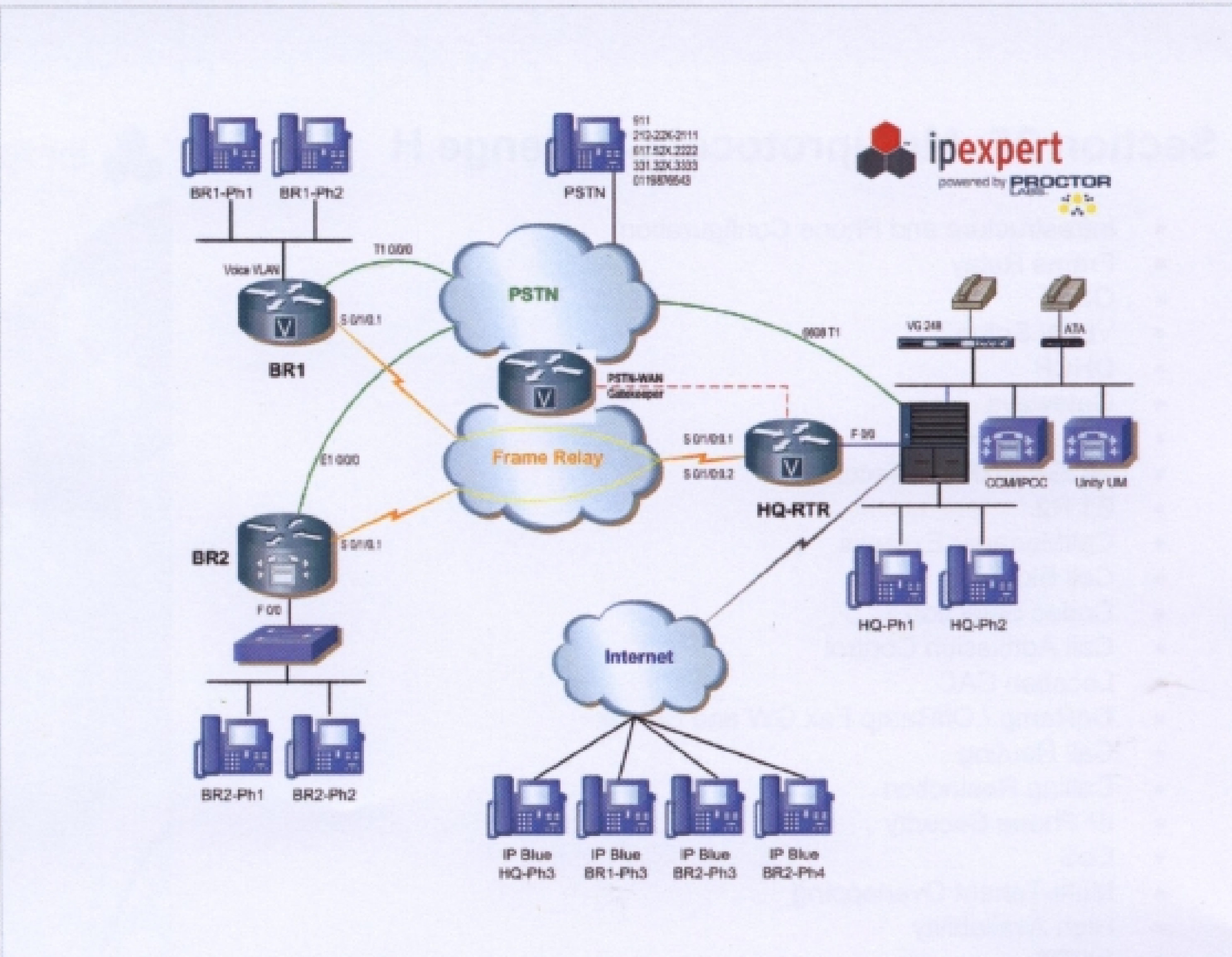
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 25: Multiprotocol Challenge H

- Infrastructure and Phone Configuration
- Frame Relay
- OSPF
- VLAN Setup
- DHCP
- Gateways
- Gatekeeper
- MCGP PRI and Security
- E1 R2
- CallManager Express
- Call Blocking
- Codec Selection
- Call Admission Control
- Location CAC
- OnRamp / OffRamp Fax GW and Server
- Call Routing
- Calling Restriction
- IP Phone Security
- CoS
- Multi-Tenant Overlapping
- High Availability
- SRST
- Secure SRST
- AAR
- Media Resources
- Conf Bridge
- Transcoding
- Hunt Groups and Call Coverage
- MOH
- QoS
- CoS-to-DSCP Mapping
- FRF.12
- LLQ
- Cisco Unity and Security
- Audio Text Application
- IPCC Express – CRS
- ICD – Skills Based Routing
- CallManager Features
- CUE
- TAPS, Extension Mobility
- FAX
- FAX Passthrough

Estimated Time to Complete: 9 hours

Lab 25 Topology



Section 25 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number* **PODS 20-22: X=Both Digits of POD Number* ***Server is not present on network – but configure everything as if it were		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number* *PODS 20-22: X = BOTH Digits of POD Number	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
X=Last Digit of POD number including pods 20-23	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
X=Last Digit of POD number including pods 20-23	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Section 25 Configuration Tasks

Infrastructure and Phone Configuration

1 Ensure that the link between the HQ/BR2 routers and appropriate switches are configured as dot1Q trunks. Give the voice sub-interface the appropriate ip address from **Table 1**. Check connectivity between all sites and to CallManager/Unity.

2 Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 2**. Use **Table 3** for HQ site 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3 Configure all phone ports such that they bypass the spanning-tree listening and learning states.

4 Set the clock on the BR2 router to the correct time.

5 For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for all Devices. For Branch 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

6 Configure CallManager to register devices. Assign directory number to devices based on **Tables 5** and **6**. Also Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2 will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that **no** user interaction is required on the IP Phone.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

---

7 Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

8 Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed. Ensure that RTP packets as well as signaling packets on this GW are encrypted so that a sniffer would not be able to decode the traffic.

9 Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

10 Configure BR2 as an H323 gateway based on the information in **Table 8**.

11 Configure the E1 R2 such that calls to the PSTN are set up 3 seconds quicker (where possible) than the default. You may assume the maximum length digit string is 11 digits.

12 Configure gatekeeper on HQ-RTR with the following information:

Local zone= HQ-RTR
domain name = ipexpert.com
[use loopback interface for local zone]

Remote zone= PSTN-WAN
domain name= ipexpert.com
ip address= 10.X.200.2 [X=Last digit of POD number]

Register CallManager to the HQ-RTR gatekeeper.

13 Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 7** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed to CCM via SIP using G711ulaw (see **Table 5** and **6** for DNs at CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

CallManager Express

14 Configure Branch 2 gateway for CallManager Express. Allow maximum of 10 phones and 20 DNs.

15 Configure Branch 2 gateway as H.323 gateway for CallManager. Use the loopback address as the source address. Assuming there is another CallManager (**Table 1**) in the cluster, configure the redundant dial-peers and reduce the H.225 timeout value to 3 seconds.

16 Allow that if a user dials *67 and then a pattern (PSTN or Internal), that the caller's ANI will not show up on the other side.

17 Ensure that any conference call at BR2 in which the conference initiator hangs up – that the conference terminates upon that person's leaving. However, Phone 3 should be allowed to hang up or press the 'end-call' softkey and leave a conference but allow it to continue running

18 Configure phones 1 and 2 at BR2 so that they can intercom each other and have an immediate 2-way conversation – security should be in place so that no one else can dial their respective intercom numbers. Use whatever DNs you wish for this. Also, if another intercom call happened to be present on phone 1 when phone 2 places the intercom call – the first call should be automatically put on hold.

19 Ensure that all calls that are placed internally (from IP Phone to IP Phone) receive not only CLID but also CNAM display with their respective names (you may name them whatever you wish).

Codec Selection and Call Admission Control

20 Calls within the HQ and BR1 sites must use the G711 codec. Calls between the two sites must only be allowed to use G729 codec.

21 Use Locations CAC to limit the number of calls over the WAN between the HQ and BR1 to one call. Configure CallManager so that Locations tracing is enabled.

22 Configure CallManager Express such that the codec selection is flexible between CME and CM.

Call Routing and Calling Restrictions

23 Configure the dial plan as follows:

911	Emergency
9.911	Emergency
9.[2-9]XXXXXX	Local
9.1XXXXXXXXXX	Long Distance
9.011!	International
9.011!#	International

24 Configure CallManager to provide Class of Service such that phone 1 can call everywhere, phone 2 can call 911 and local, and all other phones (if applicable) can call 911, local and long distance. Configure device and line CSS to achieve this configuration. You cannot restrict any calls from the device CSS level.

25 From HQ or Branch 1, dial 4 digits to reach Branch 2. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed first out of the local Gatekeeper (which is expecting a e164 call with a 011 prefix) and then through the HQ-IPIPGW as a backup. (**Test both** call routing methods, one at a time, before moving on to the next question). All other calls from the HQ will be routed to local gateway only.

26 Calls originating from BR2 to CallManager (both to HQ and to BR1) should use VoIP through the HQ IPIPGW as a first choice, directly to the CCM as a second choice, and to the PSTN as third choice backup. 4-digit dialing must be preserved all the way around. Also, you must use the minimum amount of dial-peers possible. All other calls will be routed via PSTN.

27 Local calls from Branch 1 will be routed through local gateway first and use 6608 as backup. International calls will be routed through 6608 first and use local gateway as backup. 911 and Long Distance calls will be routed through local gateway only.

28 When someone places the calls from Branch 1 to the area code in HQ, calls should be hopped off using the 6608 and use local gateway as backup.

High Availability

29 Configure Survivable Remote Site Telephony for Branch 1. Allow maximum of 5 phones and 10 DNs. Ensure that by the time IP Phones register to this SRST device, that the necessary certificate exchange has occurred in order to continue to keep phones capable of a secure conversation as such.

30 When "Message" button is pressed, users should be able to hear the general greeting from Unity.

31 Configure SRST to provide same calling restrictions as specified in question #24.

32 Configure Branch 1 gateway such that users at Branch 1 can dial 4 digits to reach the HQ.

33 Configure AAR between the HQ and Branch 1.

Media Resources

34 Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

35 Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

36 Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

37 Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

38 HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should use the IOS resources and the 6608 resources as backup.

39 Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source.

40 Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You cannot use the transcoder to achieve this task.

41 Ensure that the HQ and BR1 phones receive multicast music on hold.

42 Configure music on hold on the BR2 CME.

Quality of Service

43 Assume the single cable solution is used, configure the switches to trust the IP phones and not the attached PC.

44 Configure the Catalyst 6500 to move VOIP control traffic to the 2^{nd} queue and 1^{st} threshold.

45 Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

46 Configure the Catalyst 6500 to mark all VOIP control traffic from the CallManager as CS3.

47 Assume one of Gigabit Ethernet connections is used for the uplink to the distribution switch, configure the Catalyst 6500 such that the tagged packets will have access to PQ in the receive queue.

48 The speed of the PVC between the HQ and Branch 2 is 768 kbps. Configure FRTS and apply to the PVC.

49 Configure FRF.12 between the HQ and Branch 2. Configure such that the serialization delay is 10 ms.

50 Configure Low Latency Queuing (LLQ) for the HQ and Branch 2. Allocate 33% of the total bandwidth to Priority Queue and 2% as one of the CBWFQ for the control traffic.

Cisco Unity and Unity Express

51 Configure Voice Mail Integration for CallManager. The information is provided below.

- Pilot Number: 1600
- Unity Port Number: 1601 – 1604
- Set the default password to 54321
- MWI Light – 1998 off
- MWI Light – 1999 on

52 Integrate the Branch 2 CME with Unity Express (IP address in **Table 16**) with the following information and provide MWI and voice mail boxes for Phone 1 and 2 in the Branch 2 CME. Also ensure that callers receive the CallerID for whomever left the message.

- Voice Mail Pilot = 3600
- MWI On/Off = 3999/3998
- AA DN = 3100
- TUI = 3200

53 Create 3 subscribers shown below in Unity using the Bulk Exchange Tool.

First Name	Last Name	Extension
John	Bee	1001
Bill	Dee	1003
Cindy	Tee	2003

54 HQ needs Auto Attendant feature. Configure Unity to provide such functionality. Use 1570 as the extension for the call handler. User should hear a greeting saying "Press 1 for directory handler, 2 for HQ phone 1". When caller press 1, send the call to directory handler. When caller press 2, send the call to phone 1 at HQ. Any other entry should take the caller back to the original greeting.

55 Configure Unity to be an Inbound and an Outbound Fax Server. Create the necessary VM box for BR1 Phone 2 and ensure that Faxes coming in from the BR1 Gateway arrive in the appropriate user's mailbox. Ensure that Faxes going out to the PSTN number in **Table 18** from the BR1 Gateway arrive and are viewable. Also setup the BR1 Gateway as an OnRamp and an OffRamp Faxing GW. Unity needs to be able to accept in-bound faxes at the mailbox of IncomingFax@voip.lab. Unity also can be your DNS server for this task. Refer to **Table 17** of this section for FaxDN ranges. All the files you will need are in BR1 Flash memory.

56 Configure Unity so that only "John Bee" can manage the greeting using TUI and they have access to the call handler (1570) only. John will need to dial 1571 in order to manage the Greeting.

IPCC Express – Customer Response Applications

57 Configure Integrated Contact Distribution (ICD) using the following information:

- CTI Route Point: 1700
- CTI Ports: 1711-1714
- Contact Service Queue Name: ipexpert
- Agent ID: tw
- Agent password: jmptw
- Script: icd.aef

Configure phone 3 at HQ to be used as an agent phone. Use skill based routing instead of resource group.

CallManager Features

58 Configure Extension Mobility as follows:

- Username: ptw
- Password: gjmpt

- HQ Device Profile: 1666
- Branch 2 Device Profile: 3666

59 Configure Phone 3 at HQ and Phone 2 at Branch 2 to support Extension Mobility. User's calling restriction should be preserved. Long distance call should be routed through 6608. All other calls should be routed through local gateway only.

60 Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring simultaneously when this DN is called. You may not use a shared line to accomplish this task. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 11** for a time schedule. Also, all members of this HuntGroup should get the message in their VM box and see their MWI for any message left for 1005. Finally create a new DN of 1010. When this DN is called, all calls should be directed to the DN of 1005. The call should ring this DN for approximately 2 rings. If for any reason this DN does not answer, the call should then be forwarded to VM if the call originated from an internal number, however if the call originated from an external number, the call should then be forwarded to the PSTN phone at '221-1111'.

Fax

61 Configure FAX Passthrough for 6608, ATA-186, VG248 and Branch 2 gateway.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

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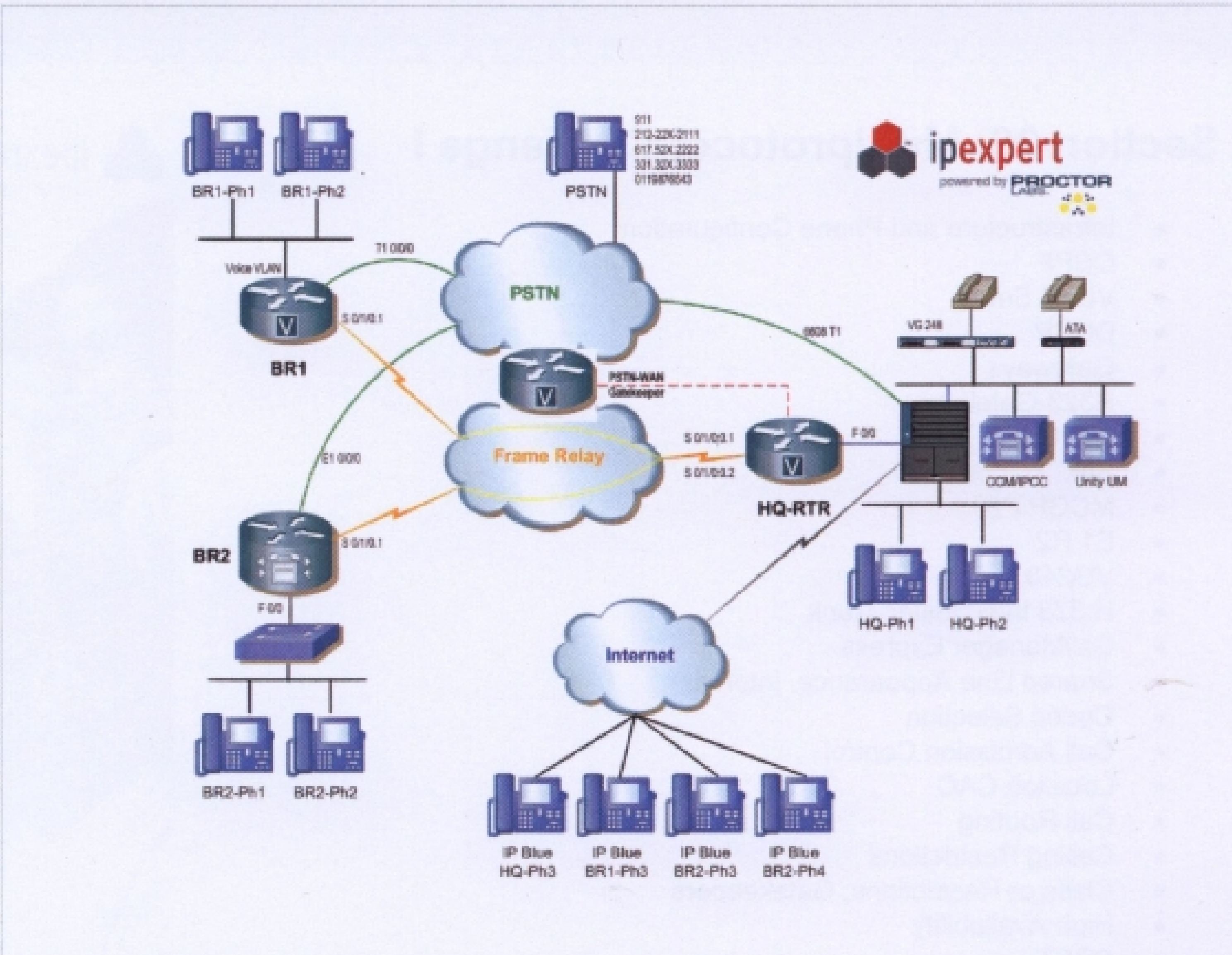
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 26: Multiprotocol Challenge I

- Infrastructure and Phone Configuration
- OSPF
- VLAN Setup
- DHCP
- Gateways
- H323 Gatekeeper
- IPIPGW
- SIP
- MCGP PRI
- E1 R2
- VG248
- H.323 Intercluster Trunk
- CallManager Express
- Shared Line Appearance, Intercom
- Codec Selection
- Call Admission Control
- Location CAC
- Call Routing
- Calling Restrictions
- Class or Restrictions, Gatekeepers
- High Availability
- SRST
- AAR
- Media Resources
- Transcoding, MOH
- QoS
- FRF.12
- Cisco Unity
- MWI
- CME
- CUE
- TCL
- B-ACD
- IPCC Express – CRA
- ICD
- CallManager Features
- IPMA
- FAX
- FAX Passthrough

Estimated Time to Complete: <u>9 hours</u>

Section 26 Topology



Section 26 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
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Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number *PODS 20-22: X=Both Digits of POD Number* ***Server is not present on network – but configure everything as if it were*		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
**** X=Last Digit of POD number including pods 20-23***	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
**** X=Last Digit of POD number including pods 20-23***	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number* *PODS 20-22: X = BOTH Digits of POD Number	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number* *PODS 20-22: X = BOTH Digits of POD Number	

Section 26 Configuration Tasks

Infrastructure

1. Configure OSPF as a routing protocol in the network so that all the router interfaces including loopback interfaces are reachable from anywhere in the network. Use area 0.

2. Configure each switch port connecting to IP phones and ATA to support both data and voice traffic according to VLAN IDs specified in **Table 2**.

3. Configure each switch port connecting to router/gateway in each site to only allow traffic from data and voice VLANs and not from any other VLANs.

4. In the HQ site, configure MS DHCP server to enable DHCP service for IP phones to get IP addresses and download phone loads.

5. In each branch site, configure each router/gateway to provide DHCP service. Configure the lease time to be 8 hours.

6. Configure HQ and BR1 phones with 4 digit extensions as shown in **Tables 5** and **6**.

7. Configure each phone forward calling name and number. When calling PSTN, each phone should forward full DID DN to PSTN phone and calling names.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

---

8. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

9. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed. Ensure that RTP packets as well as signaling packets on this GW are encrypted so that a sniffer would not be able to decode the traffic.

10. Ensure that the correct transport for DTMF digits is used for the BR1 gateway.

11. Configure BR2 as an E1 R2 gateway based on the information in **Table 8**.

12. Configure the E1 R2 such that calls to the PSTN are set up 3 seconds quicker (where possible) than the default. You may assume the maximum length digit string is 11 digits.

13. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= CCM-GK
    domain name = ipexpert.com
    [use loopback interface for local zone]
    Register CallManager to this zone.

    Local zone= BR2-GK
    domain name = ipexpert.com
    Register the BR2 CME to this zone.

    Local zone= VGK
    domain name = ipexpert.com
    Register the IPIPGW to this zone.

CallManager Express

14. Configure a shared line of 3010 on Phone 1 and Phone 2. When a call comes into this number, it should only ring Phone 1, but not Phone 2.

15. A call from another site supporting only G729 may ring into a BR2 site CME phone, and that phone may be busy and have set to forward busy calls into CUE VM. If this is the case, ensure that G729 call into CUE will not fail.

Codec Selection and Call Admission Control

16. Calls CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode.

17. Use a CallManager CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

18. In the HQ gatekeeper, allow only two concurrent G.711 calls to the gatekeeper.

Call Routing

19. Configure the following Dial Plan and Classes of Restrictions.

20. Configure the international call route pattern to account for dialing # at the end of their number to avoid a long inter-digit timeout. International calls should also complete without dialing # at the end.

Dial Plan

Partition Name	Description
911 and Internal	911 and 9911 emergency All internal phones
Local Long Distance	9 + 7 Digits for local calls. First digit use 2 – 9 9 + 1 + 10 digit number. 1st and 4th Digit can use only 2-9. 4 digits to call Branch 2 phones via gatekeeper and local gateways with prefix of 011-33-1-32X-3…
International	9 + 011 + variable length number

Class of Restrictions

Restricted	Internal 911
Local and Long Distance	Internal 911 Local Long Distance
International	Internal 911 Local Long Distance International

HQ

Name	Restrictions
Phone 1	International
Phone 2	Long Distance
Phone 3	Restricted
Phone 4	Restricted

Branch 1

Name	Restrictions
Phone 1	International
Phone 2	Restricted
Phone 3	International

Branch 2

Name	Restrictions
Phone 1	International
Phone 2	Long Distance
Phone 3	Long Distance

21. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

22. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

23. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out of the HQ-RTR gatekeeper while terminating their RTP streams at the IPIPGW on the HQ-RTR, with the appropriate local gateway acting as backup. Users must be able to dial the BR2 site using access-code of 7 and the users' 4 digit extensions. Seeing as we are using a Gatekeeper for our primary route, calls use H323 signaling end-to-end. Calls will leave using codec G711ulaw but arrive at BR2 CME as G729.

24. In the Branch 2 site, use an access-code of 7 and the users' 4 digit extensions to call other sites (HQ and BR1). Primary route must be via an IPIPGW on the BR1 router inbound as a SIP call, outbound to CCM as H323. If the IPIPGW were to fail for some reason, try a direct call to both CallManagers using H323. Finally if this method were to fail then you must send the calls out the local PSTN trunk. Accomplish this with the **minimum** number of dial-peers. Use the local gateway for all other PSTN calls.

25. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

High Availability

26. Configure in CallManager such that if there is not enough bandwidth to make calls to and from Branch 1, use PSTN to complete the calls.

27. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 and 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member.

Media Resources

28. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

29. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

30. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

31. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

32. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should load share between the IOS resources and the 6608 resources.

33. Configure G.711 unicast MoH for HQ. Configure G729 multicast for Branch 1. Configure MoH for the Branch 1 so that the gateway provides MoH instead of CM.

Quality of Service

34. In the HQ site Cat 6500, Configure on the HQ router to set the CoS of voice RTP to 5 and CoS of voice control traffic to 3 on traffic sent to Catalyst 6500 in voice VLAN. Configure the switch ports connected to IP phones not to trust COS marking from PCs connected to the phones.

35. In the Branch 2 Cat 3550, configure the port connected to IP phones not to trust CoS marking from PCs connected to the phones.

36. Assume that ATA does not mark voice RTP packets correctly to CoS of 5. Configure the switch port connected to ATA to correct this problem.

37. Configure FRTS and FRF.12 between HQ Site and Branch 2. Assume the PVC speed between HQ and BR2 is 256Kbps.

38. From HQ Site to Branch 2, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 8kbps for voice control traffic.

39. From Branch 2 to HQ Site, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 8kbps for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queueing.

40. From the HQ site to the BR1 site, reserve 50% of bandwidth for voice RTP traffic on a high priority queue and 5% of bandwidth for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queueing. The link speed between HQ and BR1 is 1544Kbps.

Cisco Unity

41. Setup Unity voice mail boxes for Phone 1 in HQ, and Branch 1. Ensure you can call and reach mailboxes as well as leave messages. Ensure you can light the MWI light.

    - Pilot Number: 1600
    - Unity Port Number: 1601 – 1604
    - Set the default password to 54321
    - MWI Light – 1998 off
    - MWI Light – 1999 on

42. Integrate the Branch 2 CME with Unity Express with the following information (**Table 16**).

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    - Provide voice mail boxes for Phone 1 and Phone 3 at Branch 2.

43. When dialing 1570, configure Unity and CM such that a caller hears a custom greeting. After hearing the greeting, if a caller dials 0, the call should be transferred to HQ Phone 1 (1001).

44. Ensure that when users listen to their voicemail messages in CUE, that they hear the ANI announced of the phone who left them the message.

45. Create an Auto-Attendant application using CUE. The incoming DN will be 3313213000 or 3000 and the AA should give the option to dial-by-name by pressing 1, dial-by-extension at any time during the prompt, and be transferred into a B-ACD TCL Queue at DN 3500 by pressing 2 and be connected to the Support Hunt Group with Queueing and Statistics.

IPCC Express

46. Configure Auto-attendant and ICD with the following information:

    - AA Route Point = 1710
    - Script = aa.aef
    - CTI Ports = 1711, 1712

    - ICD Route Point = 1700
    - Script = icd.aef
    - CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine)
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ)

47. Modify the "icd.aef" script with the following information (you may wish to rename it to avoid loosing the original script)

    - You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'

    - You must also determine what time the call is coming into the script – and if between the hours of 8 AM to 5 PM local server time – then you must determine if any agents are logged into the CSQ and if they are, continue processing the call in the script (do not worry about logged in agents for agent based routing here) If no agents are logged into the CSQ or if you determine that the time is outside of the specified hours, then the call must be routed to VM. You must route the call to the VM subscriber box of 1580 (create this mailbox), however you may **not** use any forwarding device in CCM (e.g. CTI Port, CTI RP, Phone, etc …) **or** any Call Routing Rules in Unity to get this call to the correct mailbox.

    - If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds

- Regardless of whether the call is routed to a specififc agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue')

CallManager Features

48. Configure IPMA with the following information: BR1 phone 3 will be used as the manager phone and HQ phone 3 will be used as the assistant phone.

Username: Manager	**Username: Assistant**
Primary Line: 2003	Primary Line: 1003
SD to Intercom	Proxy Line: 1560
	Incoming Intercom

Security

49. Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2 will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that **no** user interaction is required on the IP Phone.

50. Ensure that if BR1 gateway goes into fallback mode, that it has already created its own certificate, forwarded that certificate to CallManager and that CallManager in turn has installed that certificate into the IP Phones at BR1, so that signaling and media continue to be encrypted during any fallback occurrence.

51. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

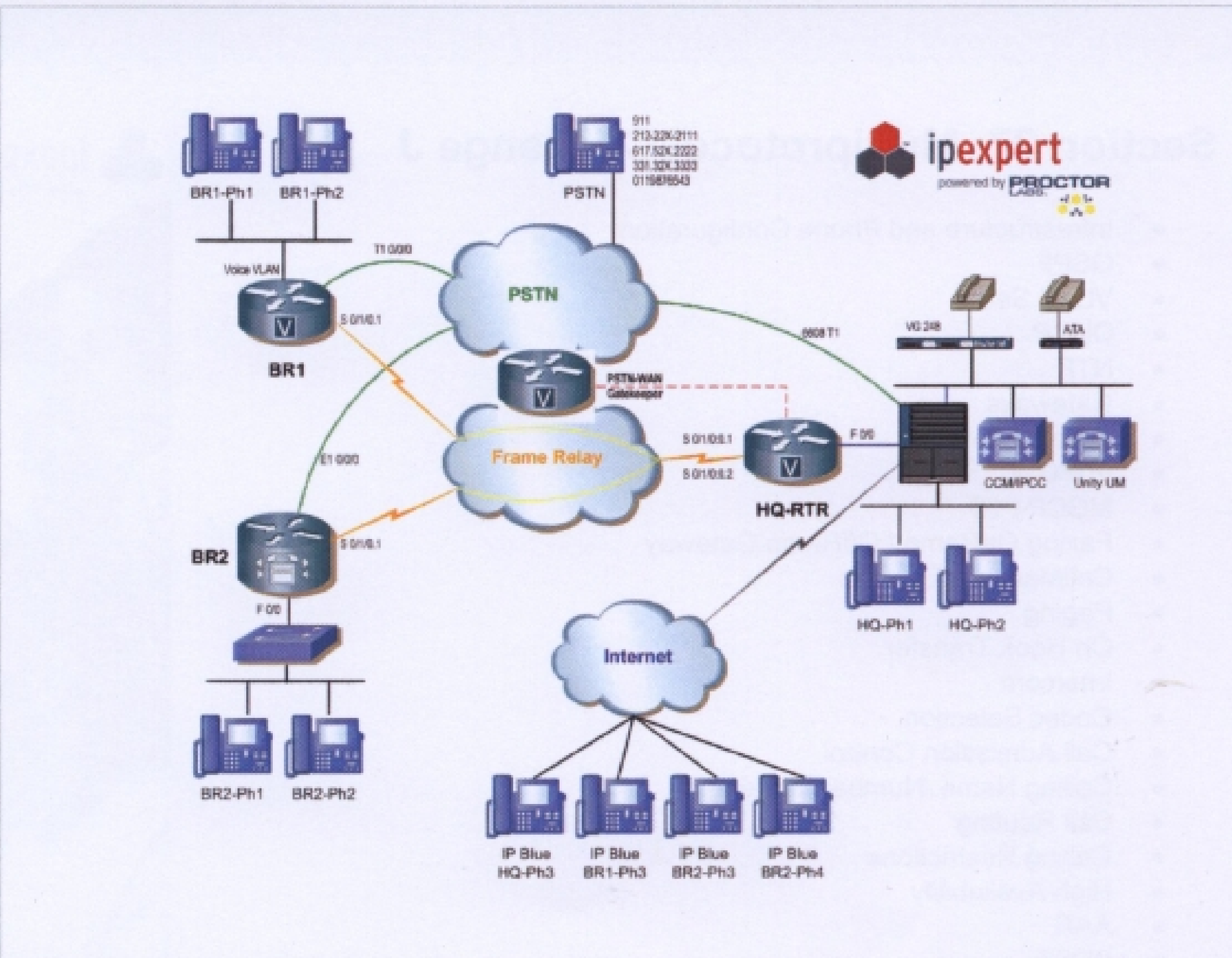
- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 27: Multiprotocol Challenge J

- Infrastructure and Phone Configuration
- OSPF
- VLAN Setup
- DHCP
- NTP
- Gateways
- Gatekeeper
- IPIPGW
- MGCP PRI
- Faxing OnRamp / OffRamp Gateway
- CallManager Express
- Paging
- On Hook Transfer
- Intercom
- Codec Selection
- Call Admission Control
- Calling Name /Number Masking
- Call Routing
- Calling Restrictions
- High Availability
- AAR
- SRST
- Media Resources
- Conf Bridge
- Transcoding
- MOH
- QoS
- CoS-to-DSCP
- LLQ
- Cisco Unity
- Faxing Inbound / Outbound Server
- Holiday Greeting
- Clustering
- IPCC Express – CRA
- Auto Attendant
- CallManager Features
- Idle URL
- Video
- FAX
- FAX Relay

Estimated Time to Complete: 10 hours

Section 27 Topology



Section 27 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23***	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
*** X=Last Digit of POD number including pods 20-23***	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Section 27 Configuration Tasks

Infrastructure

1. Configure OSPF as a routing protocol in the network so that all the router interfaces including loopback interfaces are reachable from anywhere in the network. Use area 0.

2. Configure each switch port connecting to IP phones and ATA to support both data and voice traffic according to VLAN IDs specified in **Table 2**.

3. Configure each switch port connecting to router/gateway in each site to only allow traffic from data and voice VLANs and not from any other VLANs.

4. In the HQ site, configure MS DHCP server to enable DHCP service for IP phones to get IP addresses and download phone loads.

5. In each branch site, configure each router/gateway to provide DHCP service. Configure the lease time to be 8 hours.

6. Configure HQ and BR1 phones with 4 digit extensions as shown in **Tables 5** and **6**. Security must be enforced on these phones so that one using a sniffer would not be able to decipher signaling or voice packets

7. Configure each phone forward calling name and number. When calling PSTN, each phone should forward full DID DN to PSTN phone and calling names.

Gateway and Gatekeeper

NOTE:

Please note that clocking will be provided by the network for all locations.

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8. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

9. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed. Security must be enforced on this gateway so that one using a sniffer would not be able to decipher signaling or voice packets.

10. Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 7** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed through the IPIPGW to CCM via SIP using G711ulaw (see **Table 5** and **6** for DNs at CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

11. Configure BR2 as an E1 R2 gateway based on the information in **Table 8**.

12. Configure the E1 R2 such that calls to the PSTN are set up 3 seconds quicker (where possible) than the default. You may assume the maximum length digit string is 11 digits.

13. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    [use loopback interface for local zone]

    Register CallManager to the HQ-RTR gatekeeper with a default technology prefix. Register CME to the gatekeeper with 4 digit extension numbers only.

CallManager Express

14. Configure the router in Branch 2 for CallManager Express. Register the devices and configure the DNs described in **Table 7**. Assign the appropriate restrictions.

15. Configure paging for the two IP phones located in Branch One, use line 2 as the intercom so that when ephone 1 presses line 2, a call is placed to line 1 of ephone 2. When ephone 2 presses line 2, a call is placed to line 1 of ephone 1. Assign 3101 to line 2 on IP Phone 1 and 3102 on line 2 of IP Phone 2.

16. Specify the maximum entries and minutes of the call history to 500.

Codec Selection and Call Admission Control

17. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

18. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

19. In the HQ gatekeeper, allow only two concurrent G.729 calls to the gatekeeper.

Call Routing and Calling Restriction

20. Configure Calling Restriction using the **Table** below.

Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

CallManager Express

14. Configure the router in Branch 2 for CallManager Express. Register the devices and configure the DNs described in **Table 7**. Assign the appropriate restrictions.

15. Configure paging for the two IP phones located in Branch One, use line 2 as the intercom so that when ephone 1 presses line 2, a call is placed to line 1 of ephone 2. When ephone 2 presses line 2, a call is placed to line 1 of ephone 1. Assign 3101 to line 2 on IP Phone 1 and 3102 on line 2 of IP Phone 2.

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18. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

19. In the HQ gatekeeper, allow only two concurrent G.729 calls to the gatekeeper.

Call Routing and Calling Restriction

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Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

21. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the BR1 gateway with the HQ gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

22. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway.

23. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed first out of the local Gatekeeper and then through the HQ-IPIPGW as a backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. (**Test both** call routing methods, one at a time, before moving on to the next question).

24. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway.

25. Calls originating from BR2 to CallManager (both HQ and BR1) should use VoIP through the HQ IPIPGW and the PSTN as backup. 4-digit dialing must be preserved. Also, you must use the minimum amount of dial-peers possible.

26. Unity needs to be able to accept in-bound faxes at the mailbox of IncomingFax@voip.lab. Unity will also be your DNS server. Configure the BR1 GW to intercept these faxes from the PSTN PRI as an On-Ramp gateway and email all incoming faxes to this mailbox. Refer to **Table 17** of this section for FaxDN ranges. All the files you will need are in BR1 Flash memory. Unity also needs to be able to send out-bound faxes to the PSTN (we will configure Unity for this task later). Configure the BR1 GW to accept these faxes via email from Unity and then to send them out through the PSTN PRI as an Off-Ramp gateway. Refer to **Table 18** of this section for PSTN FaxDN ranges. Again, all the files you will need are in BR1 Flash memory.

27. All users with a class of service that includes access to PSTN numbers should be given access to Toll Free Services. Block access to all 1-900 and 1-976 numbers.

High Availability

28. Configure the Branch 1 router to provide SRST service. Only allow three phones to register. In this exercise to block calls you must use a super set of COR lists applied for outgoing calls and subset of COR lists applied for incoming calls. Use descriptive names like Restricted, LD, INTL, Manager and Employee. Route Patterns must be transparent to the users.

29. Configure AAR for Headquarters and Branch 1.

Media Resources

30. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

31. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

32. Configure conference bridge on BR1 router. Use a maximum of 1 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

33. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

34. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should load share the IOS resources and 6608 resources.

35. Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source. You may not configure music sources on line, device or device pool to accomplish this task.

36. Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You must use the transcoder to achieve this task.

37. Ensure that the HQ and BR1 phones receive multicast music on hold.

38. Configure music on hold on the BR2 CME.

39. Create a meet-me conference with DN=1900.

Quality of Service

40. Configure the CoS-to-DSCP Mappings for the Catalyst 6500 and the Catalyst 3550 switches. The recommended settings are DSCP of CS3 for VoIP control plane and DSCP of EF for VoIP bearer plane.

41. Configure LLQ between Headquarters and BR1 site- configure 32kbps control traffic and 256kbps bearer traffic. From BR1 to Headquarters configure 6% control traffic and 40% bearer traffic. Link speed is 1544Kbps.

Cisco Unity

42. Setup Unity voice mail boxes for Phone 1 in HQ, and Branch 1. Ensure you can call and reach mailboxes as well as leave messages. Ensure you can light the MWI light.

    - Pilot Number: 1600
    - Unity Port Number: 1601 – 1604
    - Set the default password to 54321
    - MWI Light – 1998 off
    - MWI Light – 1999 on

43. Integrate the Branch 2 CME with Unity with the following information.

    - Pilot Number: 3600
    - Unity Port Number: 3600 – 3603
    - Set the default password to 54321
    - MWI Light – 3998 off
    - MWI Light – 3999 on

    - Provide voice mail boxes for Phone 1 and Phone 3 at Branch 2.

44. Configure Unity to observe holidays for the next five years.

• New Year's Day	January 1
• President's Day	February 16
• Memorial Day	May 31
• Independence Day	July 4-5
• Labor Day	September 6
• Thanksgiving	November 25-26
• Christmas	December 24-25
• New Year's Eve	December 31

45. Setup a Unity Call Handler for each site with DN 1570 / 2570 / 3570. During active hours (8-5pm) Monday through Friday, a standard transfer will release the call to phone 1 at each site. During closed business hours send the caller to a standard greeting that says a message like "Your call is important to us...the office is now closed, our normal hours are M-F between 8 a.m. and 5 pm. If you would like to leave a message" and allow users to leave a message. During a holiday schedule send the call to a standard greeting that says a message like "Your call is important to us...the office is now closed due to holiday observance, our normal hours are M-F between 8 a.m. and 5 pm if you would like to leave a message" and allow users to leave a message.

46. Configure Unity to be an Inbound/Outbound Fax Server. Ensure that Faxes going out to the PSTN number in **Table 18** from the BR1 Gateway arrive and are viewable. Ensure that Faxes coming into BR1 GW destined for the DN in **Table 17** arrive and are viewable (This fax will be viewable from the Proctor Labs web interface).

47. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

CallManager Features

48. Configure the CallManager IP Phones with idle URL using the Cisco Logo image. (Graphic provided on the Desktop: ciscologo.cip)

49. Restrict internal callers from BR1 only, so that they may not see CNAM information only regarding who they are calling or who is calling them, however ensure that BR1 Phone 3 **can** see all CNAM information.

50. Configure CCM to 'appear' as an older KSU would have appeared at a remote site - so that if a BR1 Phone3 was to pick up their handset and select what they believed to be a "line" to dial out of, that they would not need to first dial a 9 in order to access a trunk. Make this "Line" access separate from their main extension DN. Also ensure that the Caller does **not** see the 9 before their dialed number when they place the call.

51. Make sure that if during a call, the user presses the Transfer softkey, dials the extension, and hangs up, that the transfer succeeds without having to press the transfer key a second time.

52. Enable support for a VTA camera on HQ Phone 3.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 28: Multiprotocol Challenge K

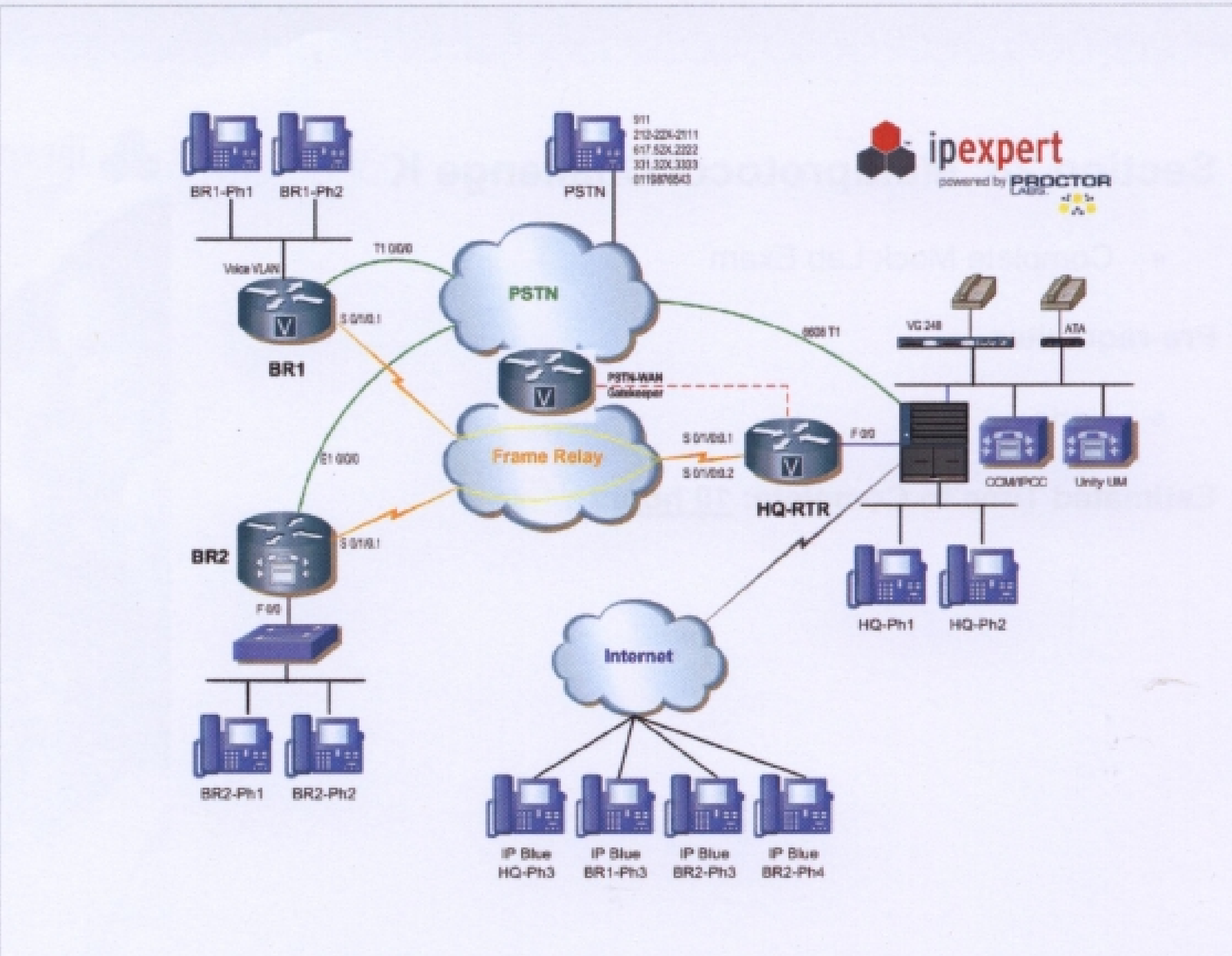
- Complete Mock Lab Exam

Pre-requisites

- None

Estimated Time to Complete: **<u>10 hours</u>**

Section 28 Topology



Section 28 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
**** X=Last Digit of POD number including pods 20-23***	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
**** X=Last Digit of POD number including pods 20-23***	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
****PODS 11-19: X = Single Last Digit of POD Number* **PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
****PODS 11-19: X = Single Last Digit of POD Number* **PODS 20-22: X = BOTH Digits of POD Number***	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Table 19 – FAC Codes

Branch 1	DN
LD Code	9558
International Code	9889

Section 28 Configuration Tasks

Infrastructure

1. Ensure that the link between the HQ/BR2 routers and appropriate switches are configured as dot1Q trunks. Give the voice sub-interface the appropriate IP address from **Table 1**. Check connectivity between all sites and to CallManager/Unity.

2. Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 2**. Use **Table 3** for HQ site 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3. Configure all phone ports such that they bypass the spanning-tree listening and learning states.

4. Set the clock on the BR2 router to be an authoritative time source.

5. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for all Devices. For Branch 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

CallManager Basics

6. Configure CallManager and register devices manually using CDP information. Assign directory number to devices based on **Tables 5** and **6**.

7. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

8. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

9. Assume a Publisher exists (**Table 1**) and every phone is registered in a CCM Group first to a Subscriber, and then to the Publisher. Configure the keepalive interval between any IP Phone and the Publisher server CallManager to be set to 40 seconds.

10. Configure Calling Restriction using the **Table** below.

Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

11. Set the phones that reside in the POD (all 79XX not physically accessible to you) to auto-answer. Phones must only auto-answer after 5 seconds.

CallManager Express Basics

12. Register phones at BR2 based on **Table 7**.

13. Create a shared line on Branch 2 phones 1 and 2 with DN 3010. The first call coming into the shared line must ring both phones. When a second call comes into the shared line it must ring the unused phone while at the same time displaying 'call waiting' on the first phone. Ensure that the shared lines have the second channel enabled.

14. BR2 should use the same class of restriction as the other sites.

Gateways

NOTE:

Please note that clocking will be provided by the network for all locations.

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15. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

16. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed. Secure all MGCP signaling and RTP stream packets from being decoded by a sniffer of any sort.

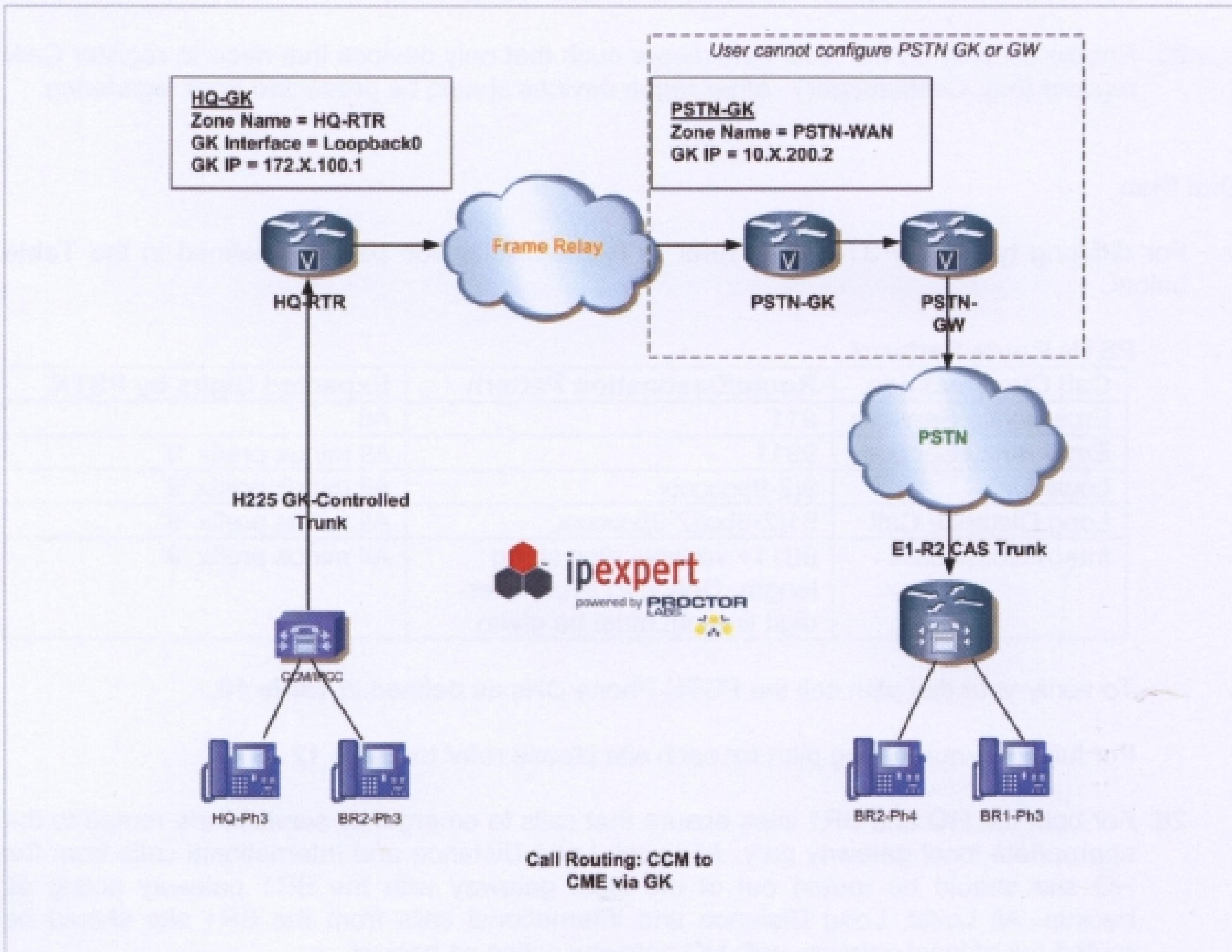
17. Configure the HQ-RTR as an IPIPGW for any possible dial-peer scenario to come.

18. Configure BR2 as an H323 gateway based on the information in **Table 8**.

19. Configure the E1 R2 such that calls to the PSTN are set up 3 seconds quicker (where possible) than the default. The PSTN will always send 10 digits inbound.

20. Unity will later need to be able to accept in-bound faxes at the mailbox of IncomingFax@voip.lab. Unity will also be your DNS server. Configure the BR1 GW to intercept these faxes from the PSTN PRI as an On-Ramp gateway and email all incoming faxes to this mailbox. Refer to **Table 17** of this section for FaxDN ranges. Unity will also need to be able to send out-bound faxes to the PSTN. Configure the BR1 GW to accept these faxes via email from Unity and then to send them out through the PSTN PRI as an Off-Ramp gateway. Refer to **Table 18** of this section for PSTN FaxDN ranges. All of the files you will need are in BR1 Flash memory.

Gatekeeper



21. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    Register CCM with this zone.
    [use loopback interface for local zone]

    Local zone= VGK
    domain name = ipexpert.com
    Register IPIPGW on loopback interface with this zone.

    Remote zone= PSTN-WAN
    domain name= ipexpert.com
    ip address= 10.X.200.2 [X=Last digit of POD number]

22. Configure GK CAC to allow one G711 call plus one G729 call.

23. Calls should attempt to use G711. If the call fails due to there not being enough bandwidth over the WAN provisioned, then the G729 codec should be selected.

24. Both media and signaling packets from CallManager to the remote zone should be sourced from the loopback interface of the HQ router.

25. Enable security on the local gatekeeper such that only devices that need to register CAN register (e.g. CallManager) - other rogue devices should be prevented from registering.

Dial Plan

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan call the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

26. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

27. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

28. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out of the local gatekeeper with the appropriate local gateway acting as backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. **NOTE:** The gatekeeper is expecting the full E164 number, i.e. international prefix '011' plus 10 digits.

29. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway. You must use 4 different methods of digit manipulation to send calls to the PSTN (e.g. one method is by using the 'forward-digits' command inside the POTS dial-peer).

30. Calls originating from BR2 to CallManager (both HQ and BR1) should use VoIP and PSTN as backup. If you decide to register the CME to the gatekeeper use your local HQ-RTR gatekeeper as opposed to the PSTN-WAN gatekeeper. 4-digit dialing must be preserved.

31. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits. At BR2, you are not allowed to use translation rules or the 'num-exp' command.

Media Resources

32. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

33. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

34. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

35. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

36. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should use the IOS resources and the 6608 resources as backup.

37. Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source.

38. Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You cannot use the transcoder to achieve this task.

39. Ensure that the HQ and BR1 phones receive multicast music on hold.

40. Configure music on hold on the BR2 CME.

41. Create a meet-me conference with DN=1900. Only HQ Phone 3 should be able to initiate the conference - other devices should be able to join/initiate this conference using DN=1901. Set the maximum number of participants of a single Meet-me conference to 6 conferencees.

High Availability

42. Configure AAR such that calls between HQ and BR1 will be rerouted over the PSTN when there is not enough bandwidth over the WAN. You must preserve 10 digit Calling Number display. The text "*Network Congestion, Rerouting*!!" must be displayed on the phone when AAR is being used.

43. Configure SRST at BR1 so that in the event of a WAN failure users can still make/receive calls from the PSTN. Use the voice sub-interface as the source address. All phones registered must have the second channel enabled on their lines.

44. Configure the SCCP heartbeat timer to 20 seconds and ensure the clock displayed on the phone is using the 12 hour display. Also ensure that the inter-digit timeout is 7 seconds.

45. Secure SRST such that phones with existing LSCs will continue to communicate in a secure fashion when in fallback mode.

46. Configure Music on Hold for all phones in SRST fallback.

47. Configure SRST such that the phones will only re-register back to CallManager after normal service has been resumed for 5 minutes. (normal service is defined as the WAN is operational and CallManager is running).

48. Calls to HQ and BR2 must be preserved using 4 digit dialing.

49. Ensure that the same Class of restriction is preserved when phones are in SRST fallback.

50. Configure Class of Restriction such that no PSTN caller can dial BR1 phone 2.

Unity

51. Integrate CallManager with Unity with the following information:

    - Voice Mail Pilot = 1600
    - Voice Mail Ports = 1600-1603
    - MWI On/Off = 1999/1998

52. Configure the BR2 router to support the CUE module using information from **Table 16**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate user's extensions and mailboxes.

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    Finally setup mailboxes for all 3 phones at BR2.

    Also ensure that when Callers leave Voicemail, that their callerID will be read to the recipient

53. Configure Unity to be an Inbound and Outbound Fax Server. Create the necessary VM box for BR1 Phone 2 and ensure that Faxes coming in from the BR1 Gateway arrive in the appropriate user's mailbox (**Table 17**). Ensure that Faxes going out to the PSTN number in **Table 18** from the BR1 Gateway arrive and are viewable.

54. Phone 1 at HQ should be configured with a Unity subscriber account with DN=1001. You must create this subscriber account using the Bulk Import Tool and a CSV file. Record a subscriber greeting.

55. Phone 3 at HQ should use the corresponding subscriber account of Phone 1 (i.e. extension '1003' will use the voicemail account for '1001'). Ensure the Phone 1 subscriber greeting is heard when Phone 3 is forwarded to voicemail and that MWI lights up both phones.

56. Create a Unity subscriber account for BR1 phone 1 with DN = 22001. Record a subscriber greeting. When Call Forward occurs from extension '2001' the correct subscriber greeting must be heard and MWI must be working correctly. You may NOT use Alternate Extension or Alternate MWI to achieve this task.

57. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

58. Configure CallManager such that an incoming call from the PSTN at the Branch 1 site destined for a BR1 phone which is forwarded to voicemail (either Divert all or CFNA/CFB) should be re-routed out of the PSTN (AAR) to Voicemail when bandwidth is not available over the WAN.

IPCC Express

59. Configure Auto-attendant and ICD with the following information:

- AA Route Point = 1710
- Script = aa.aef
- CTI Ports = 1711, 1712

- ICD Route Point = 1700
- Script = icd.aef
- CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10.
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9.
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each.
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine).
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ.

60. Modify the "icd.aef" script with the following information (you may wish to rename it to avoid loosing the original script:

    - You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'.

    - If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds.

    - Regardless of whether the call is routed to a specififc agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue').

61. Configure IPCC Phone Agent for both phones and also ensure that they only have to press the services button once (i.e. that they don't have to provide User/Ext/Pin information on the phone when logging in).

CallManager Features

62. Configure Attendant Console with Pilot number = '1550'. Add HQ phone 3 and BR1 phone 3 into the huntgroup. Enable Circular Hunting within the AC huntgroup. Also ensure that if there are up to 20 callers simultaneously calling 1550, and both IP Phones are presently taking calls, that callers will not be dropped but any number greater that 20 callers will be dropped and not sent to VM. Also, a call should not be dropped no matter how long it remains waiting.

63. Configure calls such that a HQ user must enter a forced auth code with a level of at least 20 or better in order to be allowed to dial an LD number and one of 30 or better in order to be allowed to dial an International number. See **Table 19** for Auth Codes.

64. Restrict internal callers from BR1 only, so that they may not see CNAM information only regarding who they are calling or who is calling them, however ensure that BR1 Phone 3 **can** see all CNAM information.

65. Enable support for a VTA camera on HQ Phone 3.

66. Configure IPMA with the following information: BR1 phone 3 will be used as the manager phone and HQ phone 3 will be used as the assistant phone.

Username: br1phn3	**Username: Assistant**
Primary Line: 2003	Primary Line: 1003
SD to Intercom	Proxy Line: 1560
	Incoming Intercom

QOS

67. Assume the single cable solution is used; configure the switches to trust the Layer 2 QoS classification of the IP phones but not the attached PC.

68. Configure the Catalyst 6500 to mark all VOIP control traffic from the CallManager to the appropriate L3 setting.

69. Configure the Catalyst 6500 to move VOIP control traffic to the 2nd queue and 1st threshold.

70. Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

71. Configure FRTS and apply to the PVC between HQ and BR2, however do so in such a way that the traffic shaper only engages when Voice traffic is present on the link. For this task you may assume that the FR port speed is 768kbps and that the CIR provided by the carrier is 384. A proper LFI mechanism should be engaged at all times and should be relevant to the CIR not the Port speed.

72. Assume that CUE does not mark any of its traffic with correct DSCP bits. Ensure that in the router, this traffic is marked correctly as soon as it comes from CUE and ensure that it follows the same standards of marking set from CCM regarding voice and call control.

73. Configure Low Latency Queuing (LLQ) for the HQ and Branch 2. Allocate 33% of the total bandwidth to Media and 2% as one of the CBWFQ for the control traffic.

74. Configure LLQ between the HQ and Branch 1 sites. Allocate 256Kbps for media and 8Kbps for signaling. Assume the speed of the PVC between HQ and Branch 1 is 1544 Kbps.

75. Use the Catalyst 6500 policer to police all control traffic originating from CallManager to 32 Kbps- the exceed action should be to drop excess control traffic.

Fax

76. Configure Fax Passthrough throughout the network.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 29: Multiprotocol Challenge L

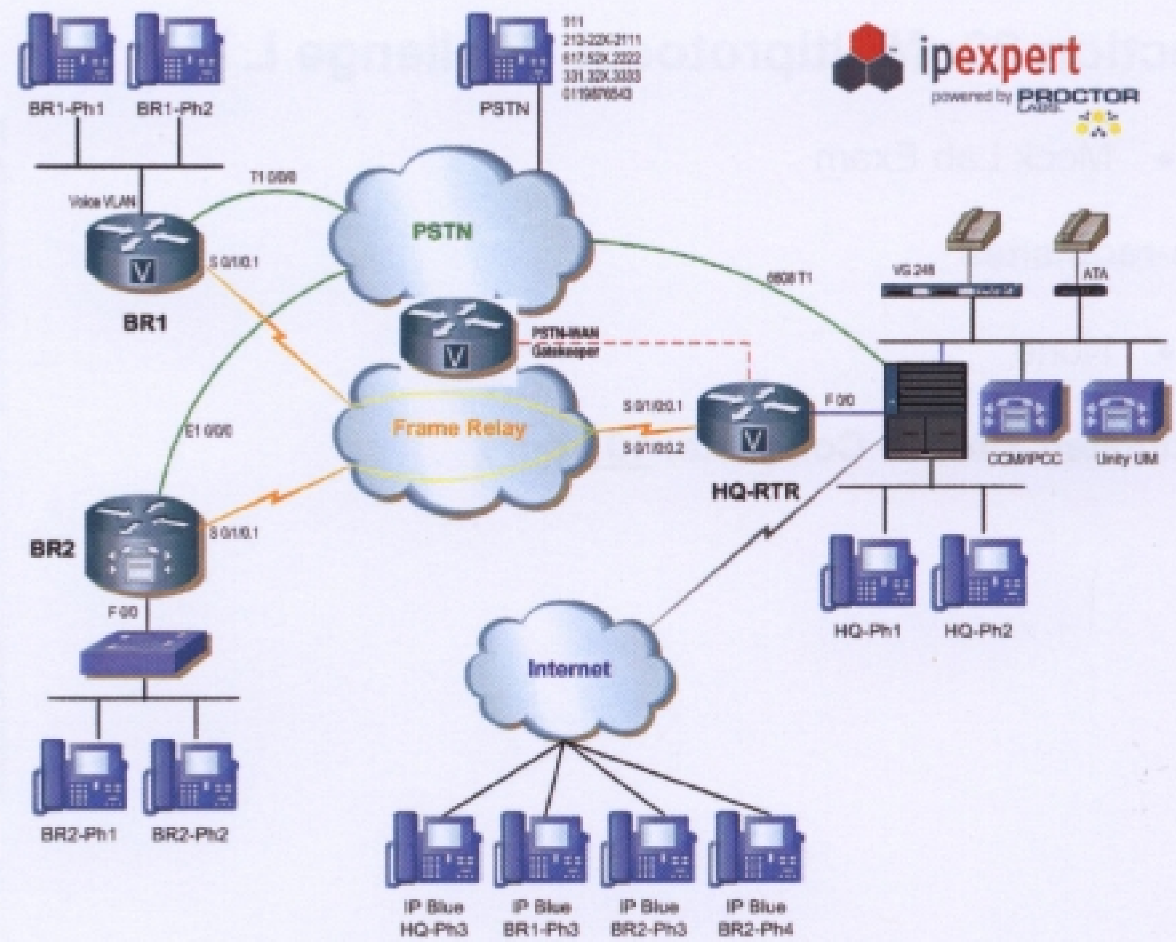
- Mock Lab Exam

Pre-requisites

- None

Estimated Time to Complete: <u>10 hours</u>

Section 29 Topology



Section 29 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number		
PODS 20-22: X=Both Digits of POD Number		
***Server is not present on network – but configure everything as if it were*		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23**	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1...
Branch 1*	617-52X-2...
Branch 2*	011-331-32X-3...
*** X=Last Digit of POD number including pods 20-23**	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
PODS 11-19: X = Single Last Digit of POD Number ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Section 29 Configuration Tasks

Infrastructure

1. Ensure that the link between the HQ / BR2 routers and appropriate switches are configured as dot1Q trunks. Give the voice sub-interface the appropriate ip address from **Table 1**. Check connectivity between all sites and to CallManager/Unity.

2. Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 2**. Use **Table 3** for HQ site 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3. Set the clock on the HQ router to poll and set its time from the PSTN-WAN router (See **Table 1** for IP address). Configure the BR1 and BR2 routers to be able to poll and update their times from the HQ router loopback interface.

4. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for all Devices. For Branch 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

CallManager Basics

5. Configure CallManager and register devices manually using CDP information. Assign directory number to devices based on **Tables 5** and **6**. Also Assume a Publisher exists (**Table 1**) and every phone is registered in a CCM Group first to a Subscriber, and then to the Publisher. Configure the keepalive interval between any IP Phone and the Publisher server CallManager to be set to 40 seconds.

6. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

7. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

8. Ensure that when a user presses the 'Services' or 'Directories' button the error message is not displayed.

9. Configure Calling Restriction using the **Table** below.

Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

10. Set the phones that reside in the POD (all 79XX not physically accessible to you) to auto-answer. Phones must only auto-answer after 5 seconds.

CallManager Express

11. Register phones at BR2 based on **Table 7**.

12. Create a shared line on Branch 2 phones 1 and 2 with DN 3010. The first call coming into the shared line must ring both phones. When a second call comes into the shared line it must ring both phones again but this time ringing in as a Call Waiting call on the active phone.

13. BR2 should use the same class of restriction as the other sites.

14. International calls should be blocked from all phones Mon-Fri outside office hours. Office hours are 9am-5pm. Allow phone 3 at BR2 to be able to enter a code in order to make International calls after hours, and make phone 2 to never be restricted for after hours international calls.

15. A call from another site supporting only G729 may ring into a BR2 site CME phone, and that phone may be busy and have set to forward busy calls into CUE VM. If this is the case, ensure that G729 call into CUE will not fail.

16. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 & 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member. Now create an incoming AutoAttendant at the DN of 3000. Also Create a Basic ACD using the support team hunt group you just created (The necessary TCL scripts are already loaded in BR2 router's flash memory). Have the AA script automatically hand-off the callers into the support ACD hunt group when a user presses 2. Allow no more than 20 callers in the Q at any one point. Play a prompt for the user every 30 seconds to let them know that all agents are busy. Allow the users to dial-by-extension by pressing 4. Ensure that the Q is collecting statistics and view them as part of your troubleshooting.

Gateways

NOTE:

- Please note that clocking will be provided by the network for all locations.
- Top-Down B channel selection should be used.
- Only the first 3 B channels are active.
- You are not allowed to use External Number Mask on lines.

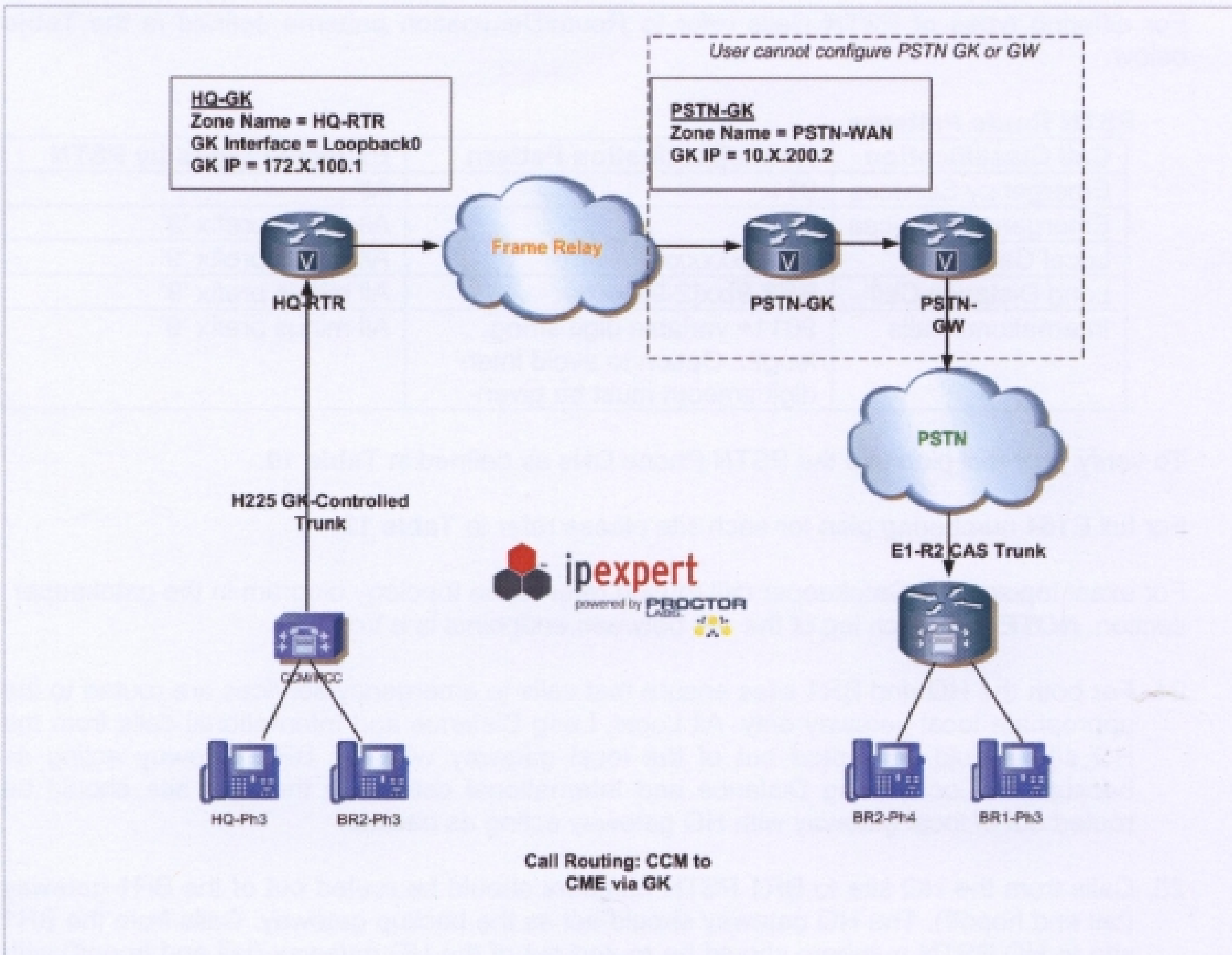
17. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

18. Configure BR1 as an MGCP gateway, based on information in **Table 8**. Calling Number (10 digits) and Calling Name should be displayed. Secure the signaling and RTP transport traffic (RTP if allowed by phone) back and forth to this gateway.

19. Configure the BR1 Router as an IPIPGW.

20. Configure BR2 as an H323 gateway based on the information in **Table 8**.

Gatekeeper



21. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= CCM-GK
    domain name = ipexpert.com
    [use loopback interface for local zone]
    Register CallManager to this zone.

    Local zone= VGK
    domain name = ipexpert.com
    Register the IPIPGW to this zone.

    Remote zone= PSTN-WAN
    domain name= ipexpert.com
    ip address= 10.X.200.2 [X=Last digit of POD number]

22. Configure CAC such that one G711 call is allowed through the gatekeeper to all zones.

23. Enable security on the local gatekeeper such that only devices that need to register CAN register (e.g. Both CallManagers, IPIPGW) - other rogue devices should be prevented from registering

Dial Plan

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

For exact topology of Gatekeeper call routing refer to the topology diagram in the gatekeeper section. **NOTE:** Not each leg of the call between endpoints is a VoIP leg.

24. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

25. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

26. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out through the local gatekeeper to the PSTN gatekeeper, with the appropriate local gateway acting as backup. Users must be able to dial the BR2 site using extension (4 digit) dialing. **NOTE:** The gatekeeper is expecting the full E164 number, i.e. international prefix '011' plus 10 digits. All signaling traffic for these calls to the PSTN GK should be sourced out of the BR1 Loopback interface; however RTP should not be terminated on BR1 at all.

27. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway.

28. Calls originating from BR2 to CallManager (both HQ and BR1) should use H323 signaling and arrive at the CCM via SIP signaling. If the WAN were to go down, then calls must automatically reroute via the PSTN. 4-digit dialing must be preserved always. Also, you must use the minimum amount of dial-peers possible and you may not use a translation-rule or num-exp to accomplish this task.

29. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

Media Resources

30. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

31. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

32. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

33. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

34. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should load share the IOS resources and 6608 resources.

35. Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source. You may not configure music sources on line, device or device pool to accomplish this task.

36. Configure the CallManager such that HQ phones will receive music using the G711 codec and BR1 phones will receive music using the G729 codec. You must use the transcoder to achieve this task.

37. Ensure that the HQ and BR1 phones receive multicast music on hold.

38. Configure music on hold on the BR2 CME.

39. Create a meet-me conference with DN=1900.

High Availability

40. Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring simultaneously when this DN is called. You may not use a shared line to accomplish this task. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 11** for a time schedule. Also, all members of this HuntGroup should get the message in their VM box and see their MWI for any message left for 1005. Finally Create a new DN of 1010. When this DN is called, all calls should be directed to the DN of 1005. The call should ring this DN for approximately 2 rings. If for any reason this DN does not answer, the call should then be forwarded to VM if the call originated from an internal number, however if the call originated from an external number, the call should then be forwarded to the PSTN phone at '221-1111.

High Availability

41. Configure SRST at BR1 so that in the event of a WAN failure users can still make/receive calls from the PSTN. Use the voice sub-interface as the source address.

42. Ensure that if BR1 gateway goes into fallback mode, that it has already created its own certificate, forwarded that certificate to CallManager and that CallManager in turn has installed that certificate into the IP Phones at BR1, so that signaling and media continue to be encrypted during any fallback occurrence.

43. Configure SRST such that one 3-party conference is allowed.

44. Configure Music on Hold for all phones in SRST fallback.

45. Configure SRST such that the phones will only re-register back to CallManager after normal service has been resumed for 5 minutes. (Normal service is defined as the WAN is operational and CallManager is running).

46. Calls to HQ and BR2 must be preserved using 4 digit dialing.

47. Ensure that the same Class of restriction is preserved when phones are in SRST fallback.

Unity

48. Integrate CallManager with Unity with the following information:

    - Voice Mail Pilot = 1600
    - Voice Mail Ports = 1600-1603
    - MWI On/Off = 1999/1998

49. Configure the BR2 router to support the CUE module using information from **Table 16**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate user's extensions and mailboxes.

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    Finally setup mailboxes for all 3 phones at BR2 and ensure that they hear the CallerID of the person leaving the message when they go to retrieve messages.

50. Create a General Delivery Mailbox in CUE for the Support Queue but give it the extension of 3215. Ensure that any phone in the office can access this mailbox by pressing "9" after they sign into their VM box. Finally, modify the Support Queue (not the hunt group) so that if agents are unavailable they will go to this GDM. Also ensure that all BR2 phones see if there is a message waiting in the GDM.

51. Create a Distribution List in CUE that allows users to be able to forward important messages to extension 3250 and all phones will receive the message in their own mailbox. The GDM must also receive the message in its box. You may NOT directly select phone extensions when creating this List.

52. Network CUE with Unity. Allow messages to be seamlessly forwarded back and forth.

53. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

54. Configure a subscriber mailbox for phone 2 at each location. Record the subscriber's greeting to say "I am not in the office right now. If you need to reach me press 3 to be transferred to my cell phone otherwise please leave a message". Have callers transferred from Voicemail to the External Numbers listed in **Table 10**.

IPCC Express

55. Configure Auto-attendant and ICD with the following information:

- AA Route Point = 1710
- Script = aa.aef
- CTI Ports = 1711, 1712

- ICD Route Point = 1700
- Script = icd.aef
- CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10.
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9.
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each.
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine).

The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ

56. Modify the "icd.aef" script with the following information (you may wish to rename it to avoid loosing the original script):

- You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'.

- If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds.

- You must also determine what time the call is coming into the script – and if between the hours of 8 AM to 5 PM local server time – then you must determine if any agents are logged into the CSQ and if they are, continue processing the call in the script (do not worry about logged in agents for agent based routing here). If no agents are logged into the CSQ or if you determine that the time is outside of the specified hours, then the call must be routed to VM. You must route the call to the VM subscriber box of 1580 (create this mailbox), however you may **not** use any forwarding device in CCM (e.g. CTI Port, CTI RP, Phone, etc ...) **or** any Call Routing Rules in Unity to get this call to the correct mailbox.

- Regardless of whether the call is routed to a specififc agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue').

CallManager Features

57. Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2 will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that **no** user interaction is required on the IP Phone.

58. Configure Extension Mobility such that a Device Profile with DN = '1551' is assigned to the following user.

- UserID='em'
- Password='adgjm'
- PIN='12345'

This user must be allowed to log into any device in the BR1 site. The user should be allowed to log into the ICD and become an agent.

59. Configure IPMA with the following information: BR1 phone 3 will be used as the manager phone and HQ phone 3 will be used as the assistant phone.

Username: br1phn3	**Username: Assistant**
Primary Line: 2003	Primary Line: 1003
SD to Intercom	Proxy Line: 1560
	Incoming Intercom

The IPMA Assistant Console can be installed on the CallManager.

Make sure that the IPMA can intercept the calls to the manager's primary extension.

Configure a Unity account for the manager. Make sure that on the manager's phone MWI still works correctly.

QOS

60. Assume the single cable solution is used; configure the switches to trust the Layer 2 QoS classification of the IP phones but not the attached PC.

61. Configure the Catalyst 6500 to mark all VOIP control traffic from the CallManager to the appropriate L3 setting.

62. Configure the Catalyst 6500 to move VOIP control traffic to the 2nd queue and 1st threshold.

63. Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

64. The link speed between HQ and Branch 2 is 256 Kbps. Employ the well-known technique to minimize serialization delay. There is no need to do this on the link between HQ and Branch 1 since its link speed is 1544kbps.

65. From HQ Site to Branch 2, reserve 33% of the bandwidth for voice RTP traffic on a priority queue and reserve 5% minimum guaranteed bandwidth for voice control traffic.

66. From Branch 2 to HQ Site, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 12kbps for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queuing.

67. From the HQ site to the Branch 1 site (PVC speed is 1544 Kbps), reserve 25% of bandwidth for voice RTP traffic on a high priority queue and 5 % of bandwidth for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queuing.

Fax

68. Configure Fax Passthrough throughout the network.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 30: Multiprotocol Challenge M

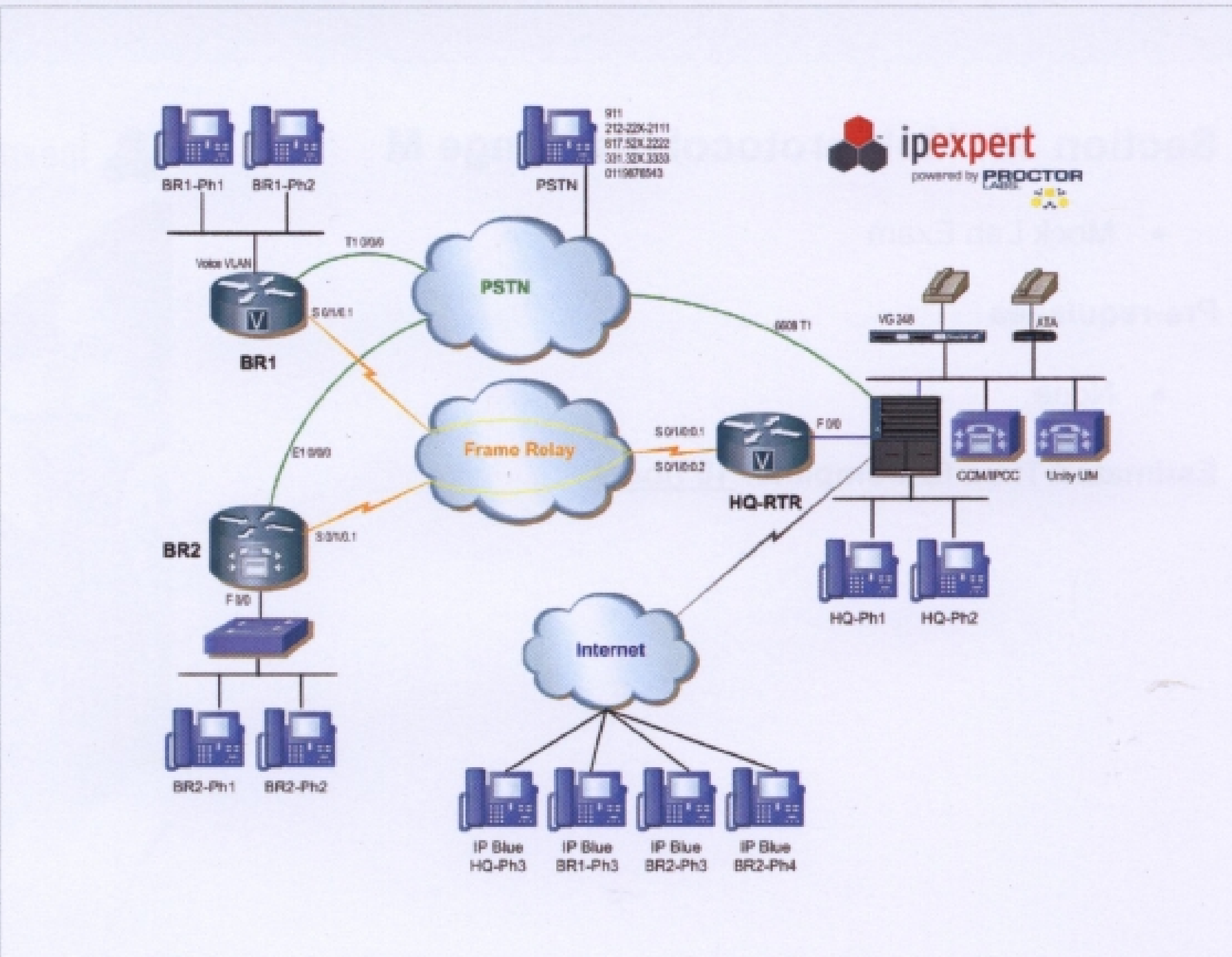
- Mock Lab Exam

Pre-requisites

- None

Estimated Time to Complete: **<u>10 hours</u>**

Section 30 Topology



Section 30 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number *PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23**	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
*** X=Last Digit of POD number including pods 20-23**	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
***PODS 11-19: X = Single Last Digit of POD Number** ***PODS 20-22: X = BOTH Digits of POD Number**	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Table 19 – CMC Codes

Branch 1	DN
PSTN Phone number for BR1	701
All other LD numbers	555

Section 30 Configuration Tasks

Infrastructure

1. Ensure that the link between the HQ / BR2 routers and appropriate switches are configured as dot1Q trunks. Give the voice sub-interface the appropriate ip address from **Table 1**. Check connectivity between all sites and to CallManager/Unity.

2. Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 2**. Use **Table 3** for HQ site 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3. Set the hardware clock on the HQ router (use EST as the time zone which is 5 hours behind GMT and setup Daylight Savings Time as well as EDT). Configure the HQ-RTR to become an authoritative time source which distributes the time via NTP. Configure the BR1 and BR2 routers to synchronize their clock with the HQ-RTR loopback interface.

4. For the Headquarter site, use Microsoft DHCP server to allocate IP address for all Devices. For Branch 1 and 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

CallManager Basics

5. Assign directory number to devices based on **Tables 5** and **6**. Register HQ and BR1 devices manually to the CallManager.

6. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

7. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

8. Ensure that when a user presses the 'Services' or 'Directories' button the error message is not displayed.

9. Configure Calling Restriction using the **Table** below. Restrict internal callers from BR1 only, so that they may not see CNAM information only regarding who they are calling or who is calling them, however ensure that BR1 Phone 3 **can** see all CNAM information.

Calling Restriction

	Phone1 at each site	**All other phones at each site**
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

10. Set the phones that reside in the POD (all 79XX not physically accessible to you) to auto-answer. Phones must only auto-answer after 5 seconds.

CallManager Express

11. Register phones at BR2 based on **Table 7**.

12. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 and 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member.

13. BR2 should use the same class of restriction as the other sites.

14. Configure CME such that outbound calls to 1 900 XXX-XXXX from Phone 1 and 2 are blocked.

15. Configure CME such that outbound calls from Phone 2 do not forward caller ID. All other phones from CME should forward caller IDs. However allow that if a user dials *67 and then a pattern (PSTN or Internal), that caller ID will not show up on the receiving phone.

16. Configure the inter-digit timeout to be 7 seconds on CME.

17. Configure phones 2 and 3 at BR2 so that they can intercom each other and have an immediate 2-way conversation – security should be in place so that no one else can dial their respective intercom numbers. Use whatever DNs you wish for this. Also, if another intercom call happened to be present on phone 1 when phone 2 places the intercom call – the first call should be automatically put on hold.

Gateways

NOTE:

Please note that clocking will be provided by the network for all locations.

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18. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.

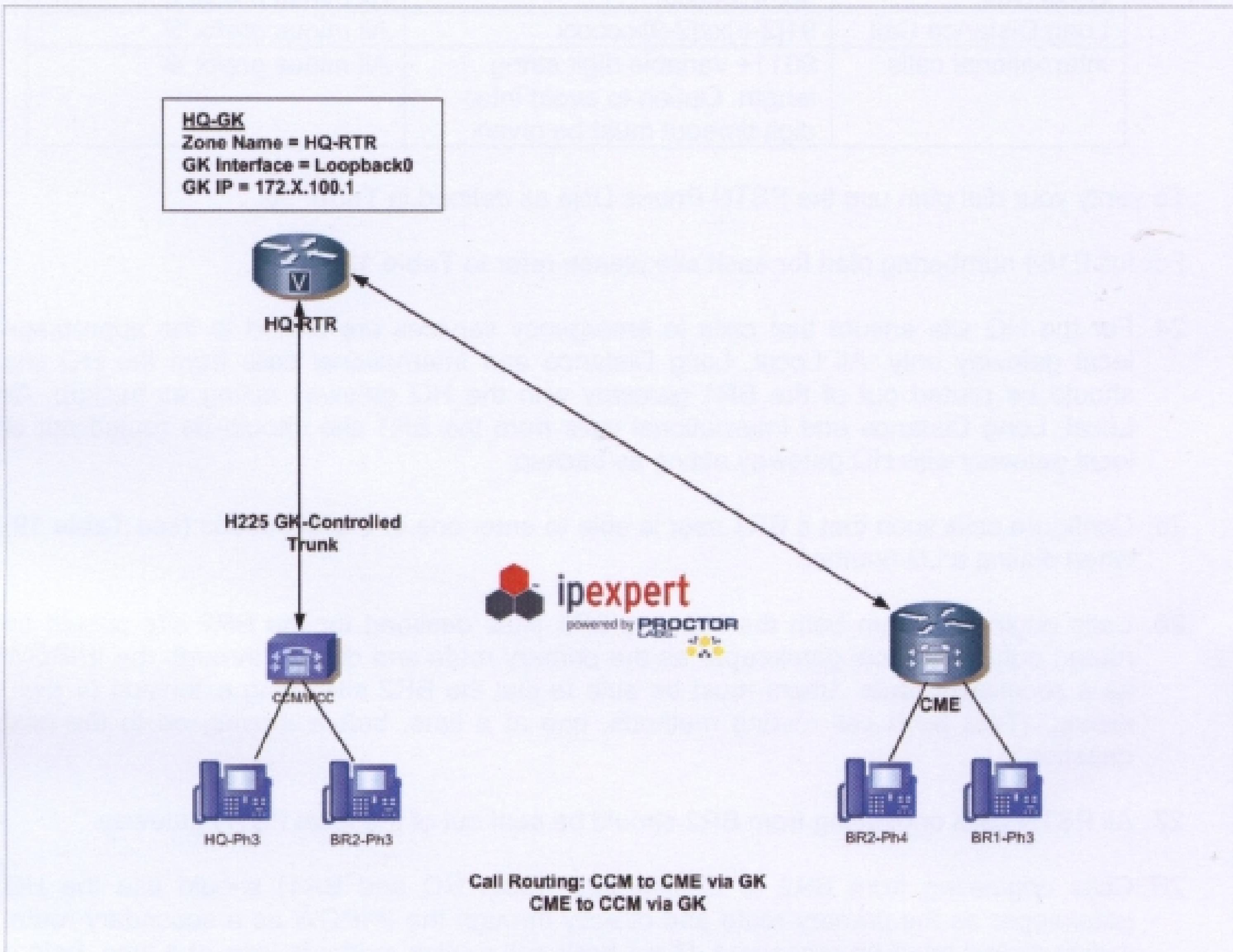
19. Configure BR1 as an MGCP gateway, based on information in **Table 8**. Calling Number (10 digits) and Calling Name should be displayed. Secure the signaling and RTP transport traffic (RTP if allowed by phone) back and forth to this gateway.

20. Configure the HQ-RTR as an IPIPGW. Calls will be coming into it from CCM via H323 using G711ulaw and then be routed out to the BR2 CME via SIP using G729 (see **Table 9** for DNs at CME). Calls will also be coming from CME via H323 using G729 and routed to CCM via SIP using G711ulaw (see **Table 10** for DNs at CCM). Also ensure when calls are coming from SIP to H323 that RFC 2833 is properly stripped. Ensure that if calls are coming from H323 to SIP, that RFC 2833 is used for the SIP side.

21. Unity needs to be able to accept in-bound faxes at the mailbox of IncomingFax@voip.lab. Unity will also be your DNS server. Configure the BR1 GW to intercept these faxes from the PSTN PRI as an On-Ramp gateway and email all incoming faxes to this mailbox. Refer to **Table 17** of this section for FaxDN ranges. Unity also needs to be able to send out-bound faxes to the PSTN. Configure the BR1 GW to accept these faxes via email from Unity and then to send them out through the PSTN PRI as an Off-Ramp gateway. Refer to **Table 18** of this section for PSTN FaxDN ranges. All the files you will need are in BR1 Flash memory.

22. Configure BR2 as an H323 gateway based on the information in **Table 8**. Configure the E1 R2 such that calls to the PSTN are set up 3 seconds quicker (where possible) than the default. You may assume the maximum length digit string is 11 digits.

Gatekeeper



23. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= HQ-RTR
    domain name = ipexpert.com
    [use loopback interface for local zone]

    Allow 2 G729 calls through the gatekeeper. All calls involving the gatekeeper must use G729.

Register CCM to the Gatekeeper. Register CME to the gatekeeper. You **must not** register any DNs or E164 numbers already configured on CME to the GK. You also **may not** use any 'default-technology' clauses in the GK configuration. The PSTN-WAN remote gatekeeper in the PSTN backbone is **not** allowed anywhere in this lab.

Dial Plan

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

24. For the HQ site ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the BR1 gateway with the HQ gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of local gateway with HQ gateway acting as backup.

25. Configure calls such that a BR1 user is able to enter one of 2 client codes (see **Table 19**) when dialing a LD number.

26. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out of the local gatekeeper as the primary route and directly through the IPIPGW as a secondary route. Users must be able to dial the BR2 site using extension (4 digit) dialing. (**Test both** call routing methods, one at a time, before moving on to the next question.)

27. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway.

28. Calls originating from BR2 to CallManager (both HQ and BR1) should use the HQ gatekeeper as the primary route and directly through the IPIPGW as a secondary route. 4-digit dialing must be preserved. (**Test both** call routing methods, one at a time, before moving on to the next question.)

29. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

Media Resources

30. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

31. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

32. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

33. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

34. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should use the IOS resources first and then 6608 resources.

35. Configure a Music on Hold server on the Publisher CallManager.

36. All sites must receive music using the G711 codec.

37. Ensure that the HQ and BR1 phones receive multicast music on hold. The IOS gateway must be the multicast source for music played to the BR1 phones and the CCM MOH server must be the multicast source for music played to the HQ phones.

38. Configure music on hold on the BR2 CME.

39. Create a meet-me conference with DN=1900. Nobody should be able to dial this number directly other than HQ Phone 1. The maximum number of participants of a single Meet-me conference should be 6 conferencees.

Unity

40. Integrate CallManager with Unity with the following information:

    - Voice Mail Pilot = 1600
    - Voice Mail Ports = 1600-1603
    - MWI On/Off = 1999/1998

41. Integrate CME into Unity Express with the following information (**Table 16**):

    - Voice Mail Pilot = 3600
    - MWI On/Off = 3999/3998
    - AA DN = 3100
    - TUI = 3200

    Also ensure that when users listen to their voicemail messages, that they hear the ANI announced of the phone who left them the message

42. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

43. Phones 2 and 3 at all locations should be configured with a VM account.

44. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

45. Configure Unity to be an Inbound and Outbound Fax Server. Create the necessary VM box for BR1 Phone 2 and ensure that Faxes coming in from the BR1 (**Table 17**). Ensure that Faxes going out to the PSTN number in **Table 18** from the BR1 Gateway arrive and are viewable.

IPCC Express

46. Configure an Auto-Attendant and ICD application with the following information:

- AA Route Point = 1710
- Script = aa.aef
- CTI Ports = 1711, 1712

- ICD Route Point = 1700
- Script = icd.aef
- CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10.
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9.
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each.
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine).
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ).

CallManager Features

47. If a call rings into HQ Phone 3, that user must have the option of sending that call to VM without exhausting the CallFwdNoAn timer.

48. Configure Attendant Console with Pilot number = '1550'. Add HQ phone 3 and BR1 phone 3 into the huntgroup. Enable Circular Hunting within the AC huntgroup. Also ensure that if there are up to 20 callers simultaneously calling 1550, and both IP Phones are presently taking calls, that callers will not be dropped but any number greater that 20 callers will be dropped and not sent to VM. Also, a call should not be dropped no matter how long it remains waiting

49. Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2 will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that **no** user interaction is required on the IP Phone.

50. Create a DN of 1005 for Tech Support. Make HQ phone 3 and BR1 Phone 3 ring simultaneously when this DN is called. You may not use a shared line to accomplish this task. DN 1005 should forward directly to VM anytime outside of normal business hours with no user intervention needed. Check **Table 11** for a time schedule. Also, all members of this HuntGroup should get the message in their VM box and see their MWI for any message left for 1005. Finally create a new DN of 1010. When this DN is called, all calls should be directed to the DN of 1005. The call should ring this DN for approximately 2 rings. If for any reason this DN does not answer, the call should then be forwarded to VM if the call originated from an internal number, however if the call originated from an external number, the call should then be forwarded to the PSTN phone at '221-1111'.

CallManager Express Features

51. Create an Auto-Attendant application using CUE. The incoming DN will be 3313213000 or 3000 and the AA should give the option to dial-by-name by pressing 1, dial-by-extension at any time during the prompt, and be transferred into a B-ACD TCL Queue at DN 3500 by pressing 2 and be connected to the Support Hunt Group with Queueing and Statistics.

52. Ensure that any conference call at BR2 in which the conference initiator hangs up – that the conference terminates upon that person's leaving. However Phone 3 should be allowed to hang up or press the 'end-call' softkey and leave a conference but allow it to continue running.

QOS

NOTE:

The PVC speed from HQ to BR1 is 1544 kbps; the PVC speed from HQ to BR2 is 256kbps.

53. Assume the single cable solution is used; configure the switches to trust the Layer 2 QoS classification of the IP phones but not the attached PC.

54. Assume that ATA does not mark voice RTP packets correctly to CoS of 5. Configure the switch port connected to ATA to correct this problem.

55. Assume that CUE does not mark any of its traffic with correct DSCP bits. Ensure that in the router, this traffic is marked correctly as soon as it comes from CUE and ensure that it follows the same standards of marking set from CCM regarding voice and call control.

56. Configure the Catalyst 6500 to move VOIP control traffic to the 2nd queue and 1st threshold.

57. Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

58. Configure the HQ router to set the CoS of voice RTP to 5 and CoS of voice control traffic to 3 on traffic sent to Catalyst 6500 in voice VLAN.

59. Configure FRTS and FRF.12 between HQ Site and Branch 2.

60. From HQ Site to Branch 2, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 8kbps for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queuing.

61. From Branch 2 to HQ Site, reserve 128 kbps for voice RTP traffic on a priority queue and reserve minimum guaranteed bandwidth of 8kbps for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queuing.

62. From the HQ site to the Branch 1 site, reserve 33% of bandwidth for voice RTP traffic on a high priority queue and 5 % of bandwidth for voice control traffic. The remainder of the traffic should be applied to weighted-fair-queuing.

FAX

63. Configure Fax Passthrough throughout the network.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com

Section 31: Multiprotocol Challenge N

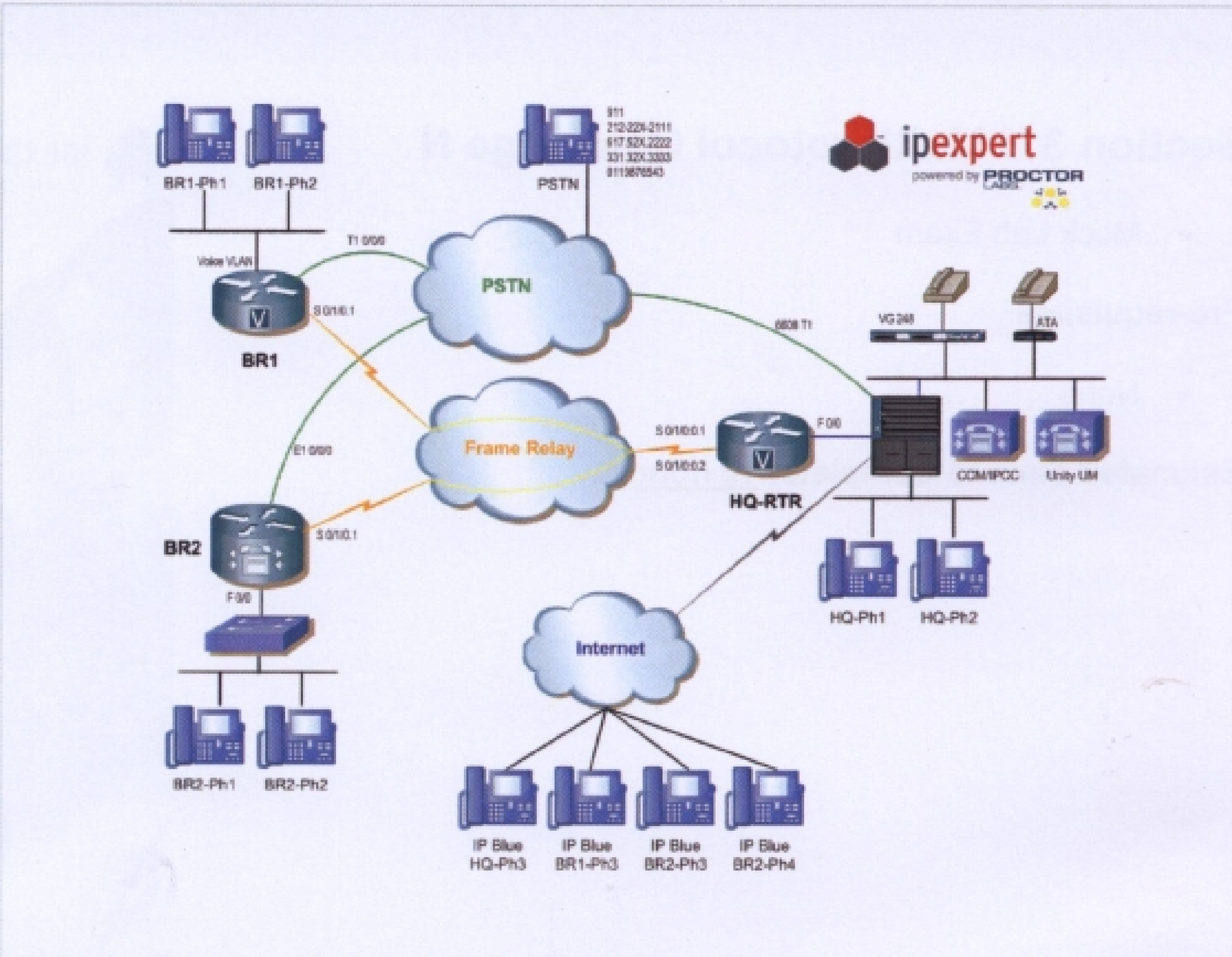
- Mock Lab Exam

Pre-requisites

- None

Estimated Time to Complete: <u>10 hours</u>

Section 31 Topology



Section 31 Tables

Table 1 – LAN IP Addresses

Location	Data Subnet	Voice IP Address
HQ Router*	no ip address	10.X.200.3/24
Branch 1 Router*	no ip address	10.X.201.1/24
Branch 2 Router*	no ip address	10.X.202.1/24
CallManager Pub**	N/A	10.X.200.20
CallManager Sub / IPCC*	N/A	10.X.200.21
Unity / AD*	N/A	10.X.200.22
PSTN-WAN Router	N/A	10.X.200.2
PODS 11-19: X=Last Digit of POD Number ***PODS 20-22: X=Both Digits of POD Number*** *****Server is not present on network – but configure everything as if it were***		

Table 2 – Voice and Data VLAN IDs

	HQ Voice VLAN	BR1 Voice VLAN	BR2 Voice VLAN	Data VLAN
POD11	210	210	210	110
POD12	220	220	220	120
POD13	230	230	230	130
POD14	240	240	240	140
POD15	250	250	250	150
POD16	260	260	260	160
POD17	270	270	270	170
POD18	280	280	280	180
POD19	290	290	290	190
POD20	400	400	400	300
POD21	410	410	410	310
POD22	420	420	420	320

Table 3 – 6500 HQ Device Port Assignment

POD #	6500 port	Purpose
11	2/1	HQ Phone 1
11	2/3	HQ Phone 2 (ATA)
12	2/7	HQ Phone 1
12	2/9	HQ Phone 2 (ATA)
13	2/13	HQ Phone 1
13	2/15	HQ Phone 2 (ATA)
14	2/19	HQ Phone 1
14	2/21	HQ Phone 2 (ATA)
15	2/25	HQ Phone 1
15	2/27	HQ Phone 2 (ATA)
16	2/31	HQ Phone 1
16	2/33	HQ Phone 2 (ATA)
17	2/37	HQ Phone 1
17	2/39	HQ Phone 2 (ATA)
18	2/43	HQ Phone 1
18	2/45	HQ Phone 2 (ATA)
19	3/1	HQ Phone 1
19	3/3	HQ Phone 2 (ATA)
20	3/7	HQ Phone 1
20	3/9	HQ Phone 2 (ATA)
21	3/13	HQ Phone 1
21	3/15	HQ Phone 2 (ATA)
22	3/19	HQ Phone 1
22	3/21	HQ Phone 2 (ATA)
Shared	3/48	VG248 (telco port 1)

Table 4 – CCM / IPCCX / Unity Port Assignment

POD #	6500 port	Purpose
11	2/5	POD11 CCM / IPCCX / Unity
12	2/11	POD12 CCM / IPCCX / Unity
13	2/17	POD13 CCM / IPCCX / Unity
14	2/23	POD14 CCM / IPCCX / Unity
15	2/29	POD15 CCM / IPCCX / Unity
16	2/35	POD16 CCM / IPCCX / Unity
17	2/41	POD17 CCM / IPCCX / Unity
18	2/47	POD18 CCM / IPCCX / Unity
19	3/5	POD19 CCM / IPCCX / Unity
20	3/11	POD20 CCM / IPCCX / Unity
21	3/17	POD21 CCM / IPCCX / Unity
22	3/23	POD22 CCM / IPCCX / Unity

Table 5 – HQ DN Assignment

HQ	DN
Phone1	1001
Phone 2 (ATA)	1002
IP Blue	1003
VG200	1004

Table 6 – Branch 1 DN Assignment

Branch 1	DN
Phone1	2001
Phone2	2002
IP Blue	2003

Table 7 – Branch 2 DN Assignment

Branch 2	DN
Phone1	3001
Phone2	3002
IP Blue	3003

Table 8 – PSTN Information

Site	Signaling	Framing	Linecode	Other info	Timeslots active
6608	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR1	T1 PRI (user)	ESF	B8ZS	Switch type Primary-NI	1-3
BR2	E1 R2	CRC4	HDB3	ITU Q421/semi-compelled	1-3

Table 9 – Loopback IP Address

Location	Loopback0
HQ*	172.X.100.1/24
Branch 1*	172.X.101.1/24
Branch 2*	172.X.102.1/24
***PODS 11-19: X = Single Last Digit of POD Number**	
***PODS 20-22: X = BOTH Digits of POD Number**	

Table 10 – Public Phone DN

Public Phone	DN
Line 1	911
Line 2*	212-22X-1111
Line 3*	617-52X-2222
Line 4*	331-32X-3333
Line 5*	0119876543
*** X=Last Digit of POD number including pods 20-23**	

Table 11– Normal Business Hours

	Mon	Tue	Wed	Thu	Fri	Sat	Sun
Open:	8am-5pm	8am-5pm	8am-5pm	8am-5pm	8am-5pm	10am-12pm	Closed
Exceptions:	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-12:30pm	11:30am-1pm	none	none

Table 12 – Site Numbering Plan

Location	Numbering Plan
HQ*	212-22X-1…
Branch 1*	617-52X-2…
Branch 2*	011-331-32X-3…
*** X=Last Digit of POD number including pods 20-23**	

Table 13 – 6608 Media Resource Port Assignment

	6608 PRI	6608 conf	6608 xcode
POD11	4/1	4/2	4/3
POD12	4/4	4/5	4/6
POD13	4/7	4/8	5/1
POD14	5/2	5/3	5/4
POD15	5/5	5/6	5/7
POD16	5/8	6/1	6/2
POD17	6/3	6/4	6/5
POD18	6/6	6/7	6/8
POD19	7/1	7/2	7/3
POD20	7/4	7/5	7/6
POD21	7/7	7/8	8/1
POD22	8/2	8/3	8/4

Table 14 – WAN Link IP Address

Location	WAN Link IP Address
HQ to Branch1*	162.X.101.1/24
HQ to Branch 2*	162.X.102.1/24
Branch 1 Serial*	162.X.101.2/24
Branch 2 Serial*	162.X.102.2/24
****PODS 11-19: X = Single Last Digit of POD Number*** ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 15 – WAN Connection

Connection	DLCI
HQ-to-BR1 PVC	201
HQ-to-BR2 PVC	202
BR1 Serial	101
BR2 Serial	102

Table 16 – Cisco Unity Express Module (CUE)

BR2 Location	Module IP Address
Service-Engine 0/1	10.X.202.2/24
****PODS 11-19: X = Single Last Digit of POD Number*** ****PODS 20-22: X = BOTH Digits of POD Number***	

Table 17 – Fax DN Assignment

Branch 1	FaxDN
Phone2	2802

Table 18 – PSTN Fax DN Assignment

PSTN	FaxDN
Fax Number	16175212223

Table 19 – CMC Codes

Branch 1	DN
PSTN Phone number for BR1	701
All other LD numbers	555

Table 20 – FAC Codes

Branch 1	DN
LD Code	9558
International Code	9889

Section 31 Configuration Tasks

Infrastructure

1. Ensure that the link between the HQ / BR2 routers and appropriate switches are configured as dot1Q trunks. Give the voice sub-interface the appropriate ip address from **Table 1**. Check connectivity between all sites and to CallManager/Unity.

2. Configure Voice and Data VLAN for all IP Phones including ATA and VG248. VLAN IDs are defined in **Table 2**. Use **Table 3** for HQ site 6500 port assignment. For BR1 and BR2 port allocation you are required to find out port allocation by your own methods.

3. Set the clock on the BR2 router to the correct time.

4. For Headquarter and Branch 1, use Microsoft DHCP server to allocate IP address for all Devices. For Branch 2, use IOS DHCP to allocate IP address for the IP phones. For each voice subnet, allocate the IP address from .50 to .69.

CallManager Basics

5. Configure CallManager and register devices manually using CDP information. Assign directory number to devices based on **Tables 5** and **6**.

6. Configure CallManager with bandwidth values such that Audio calls within the HQ and BR1 sites are allowed to use any of the following codecs: G711, G722, G729, or G728. Audio calls between the two sites must only be allowed to use G729 or G728 codecs. Configure CallManager with values such that Video calls within each site be allowed to use 1 call at the Maximum bandwidth supported by a Cisco VTA camera in wideband mode and between each site with values such that Video calls be allowed to use 1 call at the Minimum bandwidth supported by a Cisco VTA camera in H263 mode. Assign these newly created entities to a corresponding Device Pool.

7. Use CallManager's only CAC mechanism to limit the number of calls over the WAN between the HQ and BR1 to one audio call and one video call using the minimum bandwidth allowable by a VTA camera in H263 mode. Configure CallManager so that this CAC mechanism is set such that the trace files reflect the bandwidth changes.

8. Configure CallManager such that HQ Phones 1 and 2 and BR1 Phones 1 and 2, will encrypt their signaling and media streams using AES128 with any device that will allow such to occur. Configure CCM to install a 1028 bit LSC on each phone. Configure the phones so that when a LSC is to be installed on the phone – that **no** user interaction is required on the IP Phone.

9. Configure Calling Restriction using the **Table** below.

Calling Restriction

	Phone1 at each site	All other phones at each site
Emergency Services	Allowed	Allowed
Local calls	Allowed	Allowed
Long Distance calls	Restricted	Allowed
International	Restricted	Allowed

10. Set the phones that reside in the POD (all 79XX not physically accessible to you) to auto-answer. Phones must only auto-answer after 5 seconds.

CallManager Express

11. Register phones at BR2 based on **Table 7**.
12. BR2 should use the same class of restriction as the other sites.
13. Configure a Call Block rule for all CME phones. Block all 900 and 976 calls. Create a user "IPExpert" and allow that user to override the block using Toll Bar Override. Deactivate the user's login after 25 minutes when the phone becomes idle.
14. Change Phone 2 at BR2 so that the user may not utilize the CallBack feature.
15. Send all IP Phone requests at BR2 for IP XML Services to the CallManager main Services URL.

Gateways

NOTE:

- Please note that clocking will be provided by the network for all locations.
- Only 3 B-channels are active for all PSTN connections

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16. Configure the HQ 6608 T1 PRI gateway based on **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Calling Number (10 digits) and Calling Name should be displayed. For Site Numbering plan use **Table 12**, for 6608 Port Assignment use **Table 13**.
17. Configure BR1 as an MGCP gateway, based on information in **Table 8**. The Telco is selecting B channels using the 'ascending' algorithm. You must configure the T1 PRI to minimize the chances of glare. Only the top 3 B channels are 'lit'. Make sure both media and signaling packets are sourced from the Voice sub-interface. Calling Number (10 digits) and Calling Name should be displayed. Ensure that RTP traffic between the any capable IP Phone and the BR1 MGCP gateway is encrypted with AES128. Also ensure that all MGCP signaling between CallManager and BR1 gateway is encrypted with 3DES 168bit encryption using a SHA hashing algorithm.
18. Configure BR2 as an H323 gateway based on the information in **Table 8**.

Gatekeeper

19. Configure gatekeeper on HQ-RTR with the following information:

    Local zone= CCM-GK
    domain name = ipexpert.com
    [use loopback interface for local zone]
    Register CallManager to this zone.

    Local zone= BR2-GK
    domain name = ipexpert.com
    Register the BR2 CME to this zone.

    Local zone= VGK
    domain name = ipexpert.com
    Register the IPIPGW to this zone.

    You may not use any tech prefix to accomplish this task.

20. Configure CAC such that one G711 call is allowed through the gatekeeper to the remote zone.

21. Enable security on the local gatekeeper such that only devices that need to register CAN register (e.g. CallManager) - other rogue devices should be prevented from registering.

Dial Plan

For differing types of PSTN Calls refer to Route/Destination patterns defined in the **Table** below.

PSTN Route Patterns

Call Classification	Route/Destination Pattern	Expected Digits by PSTN
Emergency Services	911	All
Emergency Services	9911	All minus prefix '9'
Local Call	9[2-9]xxxxxx	All minus prefix '9'
Long Distance Call	91[2-9]xx[2-9]xxxxxx	All minus prefix '9'
International calls	9011+ variable digit string length. Option to avoid inter-digit timeout must be given	All minus prefix '9'

To verify your dial plan use the PSTN Phone DNs as defined in **Table 10**.

For full E164 numbering plan for each site please refer to **Table 12**.

For exact topology of Gatekeeper call routing refer to the topology diagram in the gatekeeper section. **NOTE:** Not each leg of the call between endpoints is a VoIP leg.

22. For both the HQ and BR1 sites ensure that calls to emergency services are routed to the appropriate local gateway only. All Local, Long Distance and International calls from the HQ site should be routed out of the local gateway with the BR1 gateway acting as backup. All Local, Long Distance and International calls from the BR1 site should be routed out of the BR1 gateway with the HQ gateway acting as backup.

23. Calls from the HQ site to BR1 PSTN numbers should be routed out of the BR1 gateway (tail end hopoff). The HQ gateway should act as the backup gateway. Calls from the BR1 site to HQ PSTN numbers should be routed out of the HQ gateway (tail end hopoff) with the BR1 gateway acting as backup.

24. Calls originating from both the HQ and BR1 sites destined for the BR2 site should be routed out of the HQ-RTR gatekeeper while terminating their RTP streams at the IPIPGW on the HQ-RTR, with the appropriate local gateway acting as backup. Users must be able to dial the BR2 site using access-code of 7 and the users' 4 digit extensions. Seeing as we are using a Gatekeeper for our primary route, calls use H323 signaling end-to-end. Calls will leave using codec G711ulaw but arrive at BR2 CME as G729.

25. In the Branch 2 site, use an access-code of 7 and the users' 4 digit extensions to call other sites (HQ and BR1). Primary route must be via an IPIPGW on the BR1 router inbound as a SIP call, outbound to CCM as H323. If the IPIPGW were to fail for some reason, try a direct call to both CallManagers using H323. Finally if this method were to fail then you must send the calls out the local PSTN trunk. Accomplish this with the **minimum** number of dial-peers. Use the local gateway for all other PSTN calls.

26. All PSTN calls originating from BR2 should be sent out of the local PSTN gateway.

27. Enable DID for PSTN users dialing into the CM/CME (i.e. 2-stage dialing must not be used). **NOTE:** For all sites the Telco is sending 10 digits.

Media Resources

28. Configure your assigned 6608 port as a conference bridge. Use **Table 13** for 6608 port assignment.

29. Configure your assigned 6608 port as a transcoder. Use **Table 13** for 6608 port assignment.

30. Configure conference bridge on BR1 router. Use a maximum of 1 session. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

31. Configure transcoder on BR1 router. Use a maximum of 2 sessions. Do not allow any G729 annexes to be configured as supported codecs. Configure only G729r8.

32. HQ should use 6608 resources and BR1 IOS resources as backup. BR1 should use the IOS resources first and then 6608 resources.

33. Configure a Music on Hold server on the Publisher CallManager. Add another music source which can be found on the C: drive of your CallManager. Ensure that the default (Sample) audio source is used when a User presses the 'Hold' button on his/her phone. For all other events that require music to be played use the newly added music source. You may not configure music sources on line, device or device pool to accomplish this task.

34. All sites must receive music using the G711 codec.

35. Ensure that the HQ and BR1 phones receive multicast music on hold.

36. Configure music on hold on the BR2 CME.

37. Create a meet-me conference with DN=1900. Nobody should be able to dial this number directly other than HQ Phone 1. The maximum number of participants of a single Meet-me conference should be 6 conferencees.

High Availability

38. Configure AAR such that calls between HQ and BR1 will be rerouted over the PSTN when there is not enough bandwidth over the WAN. You must preserve 10 digit Calling Number display.

39. Configure SRST at BR1 so that in the event of a WAN failure users can still make/receive calls from the PSTN. Use the voice sub-interface as the source address.

40. Configure the SCCP heartbeat timer to 20 seconds and ensure the clock displayed on the phone is using the 12 hour display. Also ensure that the inter-digit timeout is 7 seconds.

41. Configure SRST such that one 3-party conference is allowed.

42. Configure Music on Hold for all phones in SRST fallback.

43. Configure SRST such that the phones will only re-register back to CallManager after normal service has been resumed for 5 minutes. (Normal service is defined as the WAN is operational and CallManager is running).

44. Ensure that if BR1 gateway goes into fallback mode, that it has already created its own certificate, forwarded that certificate to CallManager and that CallManager in turn has installed that certificate into the IP Phones at BR1, so that signaling and media continue to be encrypted during any fallback occurrence.

45. Ensure that the same Class of restriction is preserved when phones are in SRST fallback.

Unity and Unity Express

46. Integrate CallManager with Unity with the following information:

   - Voice Mail Pilot = 1600
   - Voice Mail Ports = 1600-1603
   - MWI On/Off = 1999/1998

47. Configure the BR2 router to support the CUE module using information from **Table 16**. Setup the basic information needed to work the CUE module including what is needed to access the web-based GUI to manipulate users' extensions and mailboxes:

   - Voice Mail Pilot = 3600
   - MWI On/Off = 3999/3998
   - AA DN = 3100
   - TUI = 3200

48. Configure Unity so that if an IP Phone caller conferences in Ext 1650, that Unity will record the conversation and forward that recording to the mailbox of the IP Phone that initiated the conference.

49. A call from another site supporting only G729 may ring into a BR2 site CME phone, and that phone may be busy and have set to forward busy calls into CUE VM. If this is the case, ensure that G729 call into CUE will not fail.

50. Add subscriber accounts for phone 3 at each location using descriptive names. Ensure MWI is working.

51. Network CUE with Unity. Allow messages to be seamlessly forwarded back and forth.

52. Configure Unity VM Ports to use AES128 encryption for media traversal between themselves and IP Phones or Gateways that support such.

53. When the BR1 site is in SRST voicemail must work in exactly the same way as when the phones are in normal service (apart from MWI). The subscriber greeting must be heard for BR1 phone 1 on Call Forward No Answer (CFNA) and Call Forward Busy when the caller is both internal or external. The CFNA timer must be the same length as when the phones are registered to CallManager.

54. Configure CallManager such that when a call comes into the BR1 phones and is redirected to voicemail, the call is routed out of the PSTN when Locations CAC prevents the call from being sent over the WAN.

IPCC Express

55. Configure Auto-attendant and ICD with the following information:

- AA Route Point = 1710
- Script = aa.aef
- CTI Ports = 1711, 1712

- ICD Route Point = 1700
- Script = icd.aef
- CTI Ports = 1701, 1702

Create the following users that will be used in this task:

- "**jtapi**" to be used for the JTAPI Subsystem.
- "**rmjtapi**" to be used for the ICD Subsystem.
- "**telecaster**" to be used for the ICD Subsystem and enterprise data.
- "**crsadmin**" which must be the designated administrator for IPCC Express.
- **agent1** [assigned device HQ Phone 3 with ICD ext="1003"].
- **agent2** [assigned device BR Phone 2 with ICD ext="2003"]

[all passwords must be 'cisco' and PIN numbers set to '12345' except of course 'telecaster']

For the ICD application you must use skills based routing with the following information:

- CSQ: "GeneralQ"
- Skills: "sales" & "support"
- 'agent1' shall have the skill of 'support' at a proficiency level of 9 and the skill of 'sales' at a proficiency level of 10.
- 'agent2' shall have the skill of 'support' at a proficiency level of 10 and the skill of 'sales' at a proficiency level of 9.
- The CSQ 'GeneralQ' should have both skills assigned to it with a proficiency level of at least 5 or better for each.
- With this stated, the engine should always choose agent1 over agent2 if agent1 is available (you may use whatever method you wish to accomplish this in the engine).
- The engine should also send every agent into an automatic state of 'Work' for 30 seconds before then automatically returning them to a state of 'Ready' (this should happen regardless of whether the call is routed to a specific agent or through the GeneralQ).

56. Modify the "icd.aef" script with the following information: (you may wish to rename it to avoid loosing the original script)

    - You must determine if the call is coming from the PSTN from anywhere in the area code of '617' and if the call is coming from the 617 area code then the call must check to see if 'agent1' is available and route the call directly to that agent without queuing the call, however if agent1 is not available, send the call to the 'GeneralQ'.

    - If the call goes to the GeneralQ, music should be played to the caller while waiting in Q and the caller should also hear the 'QueuePrompt' every 30 seconds.

    - Regardless of whether the call is routed to a specififc agent or through the GeneralQ, the agent receiving the call should be presented with Enterprise Data showing both the ANI of the call and a disposition of how the call was routed (a string showing 'Agent' or 'Queue').

57. Configure IPCC Phone Agent for both phones and also ensure that they only have to press the services button once (i.e. that they don't have to provide User/Ext/Pin information on the phone when logging in).

CallManager and CME Features

58. Restrict internal callers from BR1 only, so that they may not see CNAM information only regarding who they are calling or who is calling them, however ensure that BR1 Phone 3 **can** see all CNAM information.

59. Configure calls such that a HQ user must enter a forced auth code with a level of at least 20 or better in order to be allowed to dial an LD number and one of 30 or better in order to be allowed to dial an International number. See **Table 20** for Auth Codes.

60. Create a circular hunt group for Support with a DN of 3210 at BR2 between phones 1 and 3, and ensure that those phones can login-to and out-of the hunt group in order to receive calls. Allow the call to ring at around 3 times before searching for the next member.

61. Create an incoming AutoAttendant at the DN of 3000. Also Create a Basic ACD using the support team hunt group you just created (The necessary TCL scripts are already loaded in BR2 router's flash memory). Have the AA script automatically hand-off the callers into the support ACD hunt group when a user presses 2. Allow no more than 20 callers in the Q at any one point. Play a prompt for the user every 30 seconds to let them know that all agents are busy. Allow the users to dial-by-extension by pressing 4. Ensure that the Q is collecting statistics and view them as part of your troubleshooting.

QOS

62. Configure the Catalyst 6500 to move VOIP control traffic to the 2nd queue and 1st threshold.

63. Configure the Catalyst 6500 and 3550 to map the CoS to DSCP value.

64. The speed of the PVC between the HQ and Branch 2 is 728 kbps. Configure FRTS and apply to the PVC.

65. Configure LFI between the HQ and Branch 2. Configure such that the serialization delay is appropriate for the link speed. You may not use FRF.12 for this task.

66. Configure Low Latency Queuing (LLQ) for the HQ and Branch 2. Allocate enough bandwidth for 4 G729 calls over the WAN to Media and use the recommended values for control traffic.

67. Configure LLQ between the HQ and Branch 1 sites. Allocate 256Kbps for media and 8Kbps for signaling. Assume the speed of the PVC between HQ and Branch 1 is 800 Kbps.

68. Police all SCCP traffic on the 6500 and 3550 to 32 Kbps- the exceed action should be to remark control traffic to DSCP 10.

Fax

69. Configure Fax Passthrough throughout the network.

Technical Verification and Support

To verify your router configurations please ensure that you have downloaded the latest configurations at www.IPexpert.com. Please visit the following web site for instructions: http://www.ipexpert.com/configs

Support is also available in the following ways:

- Email: support@ipexpert.com
- Telephone (US and Canada): +1.866.225.8064
- Telephone (Outside U.S. & Canada): +1.810.326.1444
- Support Ticket System (Elite Members): http://www.ipexpert.com
- Mailing List: http://www.OnlineStudyList.com
- Online Forum: http://www.CertificationTalk.com