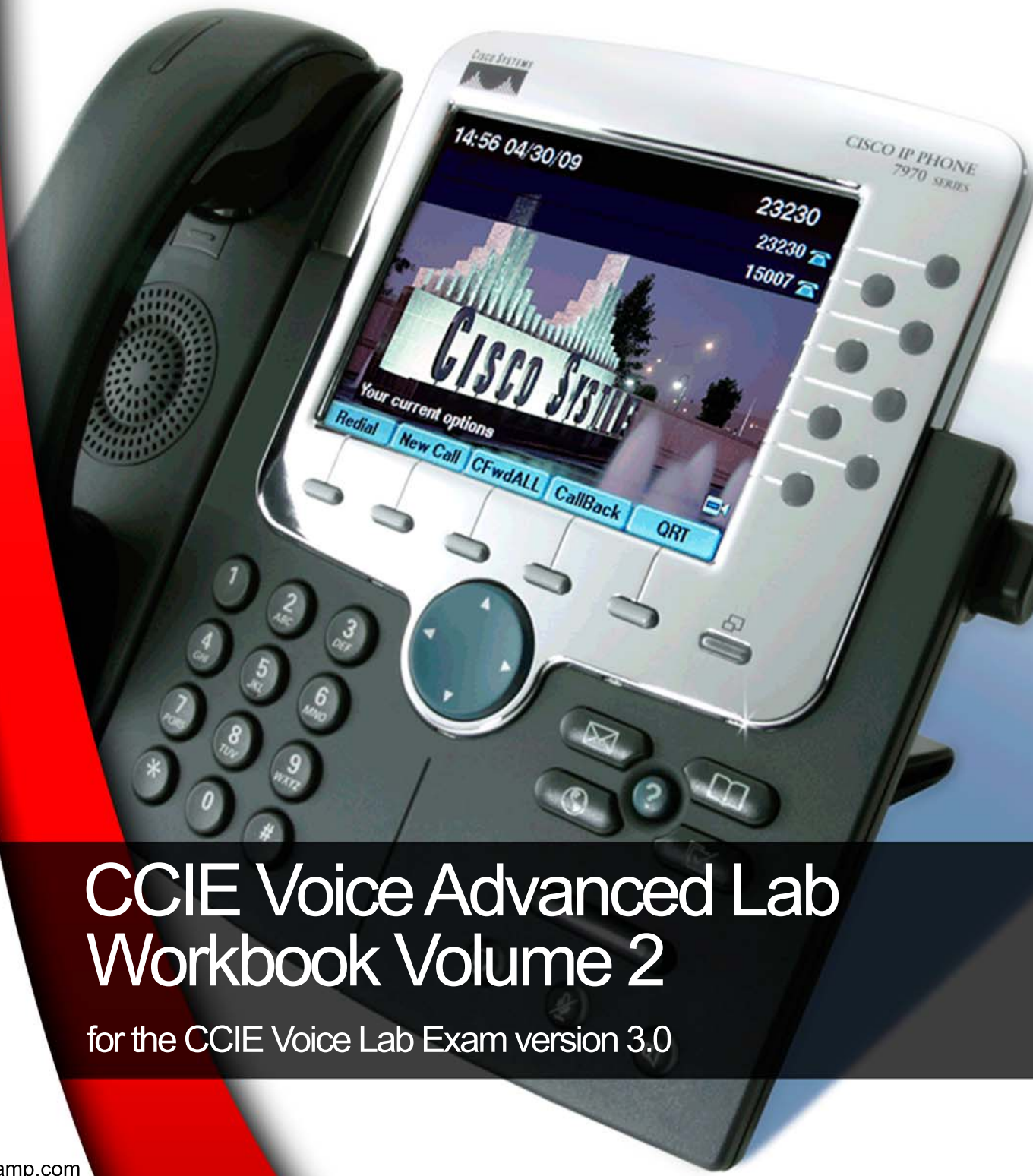

CCIE Voice Advanced Lab Workbook Volume 2

for the CCIE Voice Lab Exam version 3.0

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CCBOOTCAMP™

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Table of Contents:

A note from the Author:

Thank you for choosing CCBOOTCAMP as your partner in your journey to attain your CCIE Voice certification. Before beginning this advanced lab workbook, you should have already completed CCBOOTCAMP's CCIE Voice Technology Lab Workbook and the Advanced Lab Workbook Volume 1.

The advanced practice labs contained in this workbook are designed to test your speed and knowledge in voice technology and help you identify the areas where self study is needed most. This approach will help you attain the knowledge needed to not only pass the CCIE Voice lab exam, but also to become an expert in applying voice technology.

In this second advanced lab workbook volume you will be tested in areas of dial plan, plus dialing, number globalization, and more. Also included, is a lab designed specifically to test speed and help you gauge if you're ready for the actual lab.

You will need to have a complete and in-depth understanding of the topics contained in this workbook in order to pass the lab exam.

I wish you the best!

Chris Fortner - CCIE #18065
CCIE Voice Program Manager

General Information:

You may notice that some of the sections within the lab repeat. This is done on purpose as it is meant to make you practice the basics over and over again. This will help your speed and accuracy when taking the lab.

CCIE Voice Blueprint 3.0:

Listed below is the current published 3.0 hardware and software blueprint from Cisco Systems. The CCBOOTCAMP remote racks features a hardware, where relevant, and software match based on the published blueprint.

Lab Equipment:

- Cisco MCS-7845 Media Convergence Servers
- Cisco 3825 Series Integrated Services Routers (ISR)
- Cisco 2821 Series Integrated Services Routers (ISR)
- ISR Modules and Interface Cards
    - VWIC2-1MFT-T1/E1
    - PVDM2
    - HWIC-4ESW-POE
    - NME-CUE
- Cisco Catalyst 3750 Series Switches
- IP Phones and Soft Clients

Software Versions:

Any major software release which has been generally available for six months is eligible for testing in the CCIE Voice Lab Exam.

- Cisco Unified Communications Manager 7.0
- Cisco Unified Communications Manager Express 7.0
- Cisco Unified Contact Center Express 7.0
- Cisco Unified Presence 7.0
- Cisco Unity Connection 7.0
- All routers use IOS version 12.4T Train.
- Cisco Catalyst 3750 Series Switches uses 12.2 Main Train
- Network Interfaces
  - Fast Ethernet
  - Frame Relay
- Telephony Interfaces
  - T1
  - E1

Pre Configuration:

The pre configuration files for the voice racks at CCBOOTCAMP can be downloaded from the link provided below and will contain the base information for the start of all labs.

http://www.ccbootcamp.com/download/!Voice/Files-For-NLIs-CCIE-Voice-Tech-Workbook/V3/configs-v2.zip

VPN Access Information:

In order to access the voice racks at CCBOOTCAMP you will first need to download the Cisco VPN client from www.cisco.com. Once installed you can then download the appropriate VPN profile from the link provided below for the rack you have scheduled.

http://www.ccbootcamp.com/download/!Voice/voice-rack-vpn-profiles/

To reset your rack:

There is a Windows XP workstation in each rack with the IP address of 10.1.200.26 which can be accessed via RDP. The username is "enablemode" with a password of "enableme". On the desktop is an internet explorer icon named "Reset" which when accessed will take to the reset website.

Tips for the initial setup:

- Adjust the T302 timer in the CallManager service parameters to "4000" so your calls complete quickly.
- Define some useful shortcuts that may save precious time with aliases. Some common commands: "show call active voice brief", "show gatekeeper call status" and any command you feel might be used heavily throughout your lab.
- Think ahead and try not to touch any page more than once or twice.

Cisco's Nondisclosure Agreement:

This book was written in high respect for the CCIE certification and is not intended to violate Cisco's Nondisclosure Agreement. We will not answer questions regarding things that relate to the actual LAB exam and will not take part in any activities that stand against the CCIE NDA in any regards. Please visit the link below for the complete NDA specification.

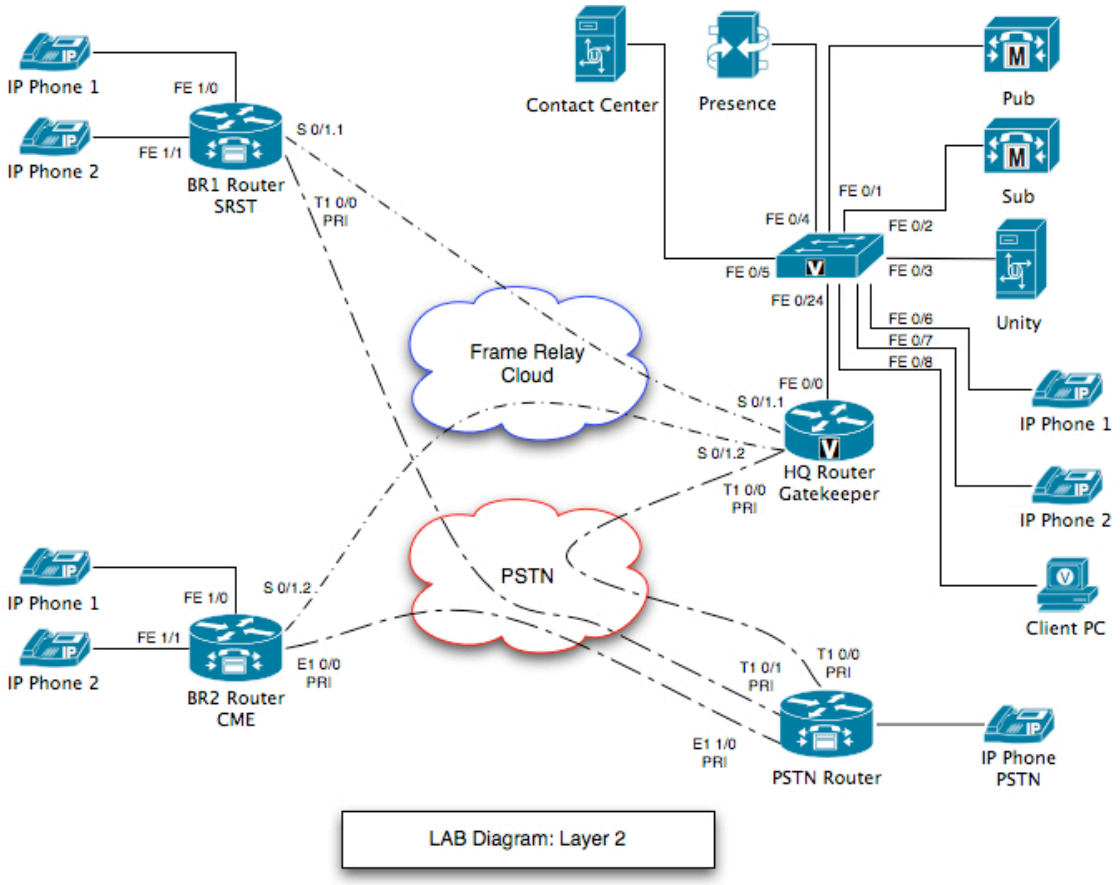
http://www.cisco.com/web/learning/le3/ccie/exam/violation_rules.html

Read Before Start:

Here are some simple rules to follow during your lab which should be taken into consideration before you configure any devices.

1. All devices should register to the subscriber and then the publisher.
2. All usernames and passwords for devices are “admin” and “cisco5796” unless specified in your lab.
3. All devices should pull NTP from the HQ1 router and should be set to the appropriate time zone.
4. Credit in the real lab will only be given for working solutions. This should be taken into consideration throughout your lab.

Lab Topology:



Note: Please note that ports may vary on different racks so please check your connections with "show cdp neighbors".

Lab Dial Plan and Addressing:

Phone	Number	Protocol	Class of Service
HQ Phone 1	1001	SCCP/SIP	International
HQ Phone 2	1002	SCCP/SIP	Long Distance
BR1 Phone 1	2001	SCCP/SIP	International
BR1 Phone 2	2002	SCCP/SIP	Long Distance
BR2 Phone 1	3001	SCCP/SIP	International
BR2 Phone 2	3002	SCCP/SIP	Long Distance

Site	PSTN E.164 Address	PSTN Number
HQ	7029461...	7029465000
BR1	8013332...	8013335000
BR2	442076303...	442076305000
ALL	911/999	911/999

Server	IP Address	Login
Publisher	10.1.200.21	admin:cisco5796
Subscriber	10.1.200.25	admin:cisco5796
Unity Connections	10.1.200.22	admin:cisco5796
Presence	10.1.200.23	admin:cisco5796
Contact Center	10.1.200.24	administrator:enableme

Chapter 1 - Lab 6

Brief Lab Overview:

This lab is designed to test your dial plan knowledge. The focus is + dialing and number globalization.

Basic Campus Design:

1. Ensure VLANS and IP networks in all three sites are configured according to the following two tables:

VLAN	HQ	BR1	BR2
Servers	1	N/A	N/A
Data	20	120	220
Voice	30	130	230

Network	HQ	BR1	BR2
Servers	10.1.200.0/24	N/A	N/A
Data	10.10.20.0/24	10.10.120.0/24	10.10.220.0/24
Voice	10.10.30.0/24	10.10.130.0/24	10.10.230.0/24

2. Make sure you set all IP Phone traffic to use the voice VLAN for all sites. Gateways should use the loopback adapter in all instances.

3. Ensure all devices within the HQ site are set to use Los Angeles as the local time zone. All devices in the BR1 site should be set for New York time zone. The devices within the BR2 site should use Germany (GMT +1) as the local time zone.

Communications Manager:

1. Make sure all HQ, BR1, and BR2 phones are registered according to the dial plan table using SCCP loads. Make sure the phones will display calling names in every scenario presented (internal and PSTN). This does not apply during SRST.

2. Configure directory number 3003 as a second line on both phones in BR2. When a call comes into 3003 it should ring on both phones and be answered by the first user to respond. Make sure the call can be picked up by HQ phone 1.

3. Make sure all phones display the same general information aside from the line specific configuration.

4. Allow BR2 phone 1 to pickup calls on BR2 phone 2 and vice versa by only selecting the call pickup softkey.

5. Both BR2 phone 1 and BR2 phone 2 are members of VIP sales group and are required to take calls dialed to 442076303111. The calls are expected to ring the phones in sequential order. Both phones should ring for 5 seconds and if the call is not answered it should be forwarded to BR2 auto attendant at extension 3000.

Voice Gateways and Signaling:

1. Configure the HQ router as a IOS H.323 gateway using T1 PRI and NI with 3 time slots.

2. Configure the BR1 router as a IOS MGCP gateway using T1 PRI and NI with 3 time slots.

3. Configure the BR2 router as a IOS MGCP gateway using E1 PRI and NET5 with 3 time slots.

Call Routing:

1. In site HQ and BR2 all phones should be able to dial any number. In site BR1 phone 1 should only be able to dial internal, local, and long distance. BR1 phone 2 should be able to dial to internal, local, long distance, and international.

2. Enable plus dialing and globalize all incoming calls.

3. Assume the local calls to HQ and BR1 are 7 digit and local calls to BR2 are 8 digits with the next 2 digits being the region code within the country.

4. Configure the following dialing options for users in HQ:

911 and 9911	Emergency
9+7 and 9+10 digits	Local
9+1+10 digits	Long Distance
9+011+any number of digits	International

- Emergency calls from HQ should use the local HQ gateway and have the correct priority.
- All calls from HQ should use the local HQ gateway.
- Calls dialed from the local phone directories enabled for plus dialing should route via the local HQ gateway and be received correctly by the PSTN.
- Make sure that all calls out of the HQ gateway send the appropriate number type:

  911 = Unknown
  7 and 10 digit = Subscriber (SEND ONLY 7 DIGITS)
  11 digits = National (SEND ONLY 10 DIGITS)
  International = International

5. Configure the following dialing options for users in BR1:

911 and 9911	Emergency
9+7 and 9+10 digits	Local
9+1+10 digits	Long Distance
9+011+any number of digits	International

- Local calls from BR1 should use the local BR1 gateway and then the HQ gateway as a backup.
- Long distance calls from BR1 should use the BR1 gateway as a first choice and then the HQ gateway as a backup.
- All international calls should route out the BR1 gateway.
- Calls dialed from the local phone directories enabled for plus dialing should route via the local BR1 gateway and be received correctly by the PSTN. No call restrictions should apply.
- Make sure that all calls out of the BR1 gateway display the appropriate calling number based on the table below:

  911 = 3332000
  Local 7 and 10 digit = 7 digits
  Long Distance = 1 plus 10 digits
  International = 10 Digits

When BR1 calls route out the HQ gateway all calls should present the main number of the HQ site "7029461000".

6. Configure the following dialing options for users in BR2:

999	Emergency
9+8 digits	Local
9+ region code +8 digits	Long Distance
9+00+any number of digits	International

- All calls should be routed out the local gateway and should not have any fallback.
- Calls to other national numbers will be dialed with region code and then subscriber number. The carrier however requires that the country code be sent. Make sure calls complete correctly.
- The carrier at this site does not look at the number type for any calls today but may in the future. Make sure that all calls exiting the gateway will have the correct number type.

Codec and Call Admission Control:

1. All calls within a site should use the G.711 codec.
2. All calls between sites should use the G.729 codec.
3. Allow three concurrent calls to and from BR1.
4. Allow two concurrent calls to and from BR2.

High Availability Features:

1. Configure BR1 and BR2 as SRST routers.
2. Preserve all PSTN dialing with using 9 as the first digit in the string at both BR1 and BR2.
3. When in SRST mode, the BR1 phone 1 is not allowed to dial international calls.
4. Allow HQ, BR1, and BR2 phones to call each other in times when more than the allowed concurrent calls are in process between the sites.

Media Resources Management:

1. Configure conferencing for the HQ site to use hardware based resources in the HQ gateway.
2. Configure conferencing for the BR1 site to use hardware based resources in the BR1 gateway.
3. Configure transcoding on the appropriate gateway(s) where needed based on defined requirements.
4. Configure MOH for the HQ site based on unicast and streamed from the publisher Call Manager.
5. Configure multicast MOH for BR1 and BR2, based on G729 codec from the subscriber, and streamed from the BR1 and BR2 gateway respectively.
6. Allow 5 simultaneous meet me conferences for the HQ users, in the number range 1201-1205. Every meet me conference should allow a maximum of ten participants.
7. Make sure that when the initiator of an Ad-Hoc conference hangs up, the conference is not allowed to continue.

QoS Features:

1. Configure the frame relay link between HQ and BR1 as if it is 384k in bandwidth, make sure you enable MLP LFI on it.
   - VoIP signaling should have 5% of the line as guarantied bandwidth.
   - VoIP barrier should have 33% of the line as priority bandwidth.
   - All other traffic should be weighted fair queued.

2. Configure the frame relay link between HQ and BR2 as if it is 1200k in bandwidth.
   - VoIP signaling should have 60k of the line reserved bandwidth.
   - VoIP barrier should have 360k of the line priority bandwidth.
   - All other traffic should be weighted fair queued.

3. The HQ gateway should send the signaling and RTP traffic marked as CS3 and EF respectively.