IP Telephony: Review and Implementation

by

Agul Kaul

This project report is submitted to the Department of Electrical Engineering and Computer Science and the Faculty of the Graduate School of the University of Kansas in partial fulfillment of the requirements for the degree of Master of Science

Dr. Joseph Evans

Dr. Victor Frost

Dr. Weichao Wang

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1 ACKNOWLEDGEMENTS

The opportunity to work on the Telephony lab implementation project has been a great learning experience. This VOIP equipment was ordered to be set up for use in the EECS Networking lab.

I have been assisting with setting up the lab so that future undergraduate and graduate students at EECS can be exposed to technologies in a functional VOIP implementation. This exposes students to the new and quickly growing technology in contemporary voice telecommunications.

I would like to thank Dr. Joseph Evans for giving me this great learning opportunity with the implementation of a functional IP Telephony Laboratory.

I would also like to thank Benjamin Ewy who assisted me with ideas and configurations to help integrate the data and the VOIP segments of the laboratory.

2 INTRODUCTION

2.1 Motivation

Voice over Internet Protocol (VOIP) is an upcoming technology and has great potential for future voice communication services. There are many technical aspects of VOIP that make it cutting edge technology in this communication era.

One aspect of VOIP from a corporate standpoint is the cost reduction with operations by implementing a VOIP network. A key feature that makes VOIP an attractive technology to adopt is the idea of a converged network - One single network infrastructure that carries voice, data and video over the Internet Protocol using existing network equipment.

Another advantage of implementing a VOIP network for corporations today is saving costs in long distance calls. Corporations do this by setting up site to site calling over long distances to go over WAN links instead of calling over a long distance carrier.

Cisco Systems, a San Jose based company has been a leader in the market for a variety of products that deal with networking systems. They have introduced a converged network solution called AVVID (Architecture for Voice Video and Integrated Data) which includes a widely used solution for IP Telephony.

Apart from IP Telephony, the AVVID solution includes features like unified messaging (email, voice and fax messaging in a single box), video telephony, IP Call Center applications, enterprise wide application integration and much more. The solution is designed to do this by transmitting data, voice and video over the single network infrastructure using standards based protocols. This framework can be enabled by the wide range of hardware and software based Cisco products.

2.2 Background

The number of corporations that implement the AVVID system is increasing everyday not only because of the complete IP Telephony solution, but also because of its support for inter-operability with other vendor systems. The solution offers various applications that are available as a bundle and provides as many features as legacy systems and more. The AVVID IP Telephony solution has come to become one of the most commonly implemented VOIP solutions for small, medium and large businesses.

2.3 Project goals

My main goal for the project was to understand the different components that constitute an IP Telephony solution. A big part of the project was also to understand the standards that are involved in a VOIP network. Specifically, my implementation goals with the Cisco IP Telephony were:

- To be able to set up a fully functional VOIP lab using MGCP (Media Gateway Control Protocol).

- To be able to set up a fully functional VOIP lab using the H.323 protocol.

A fully functional VOIP lab would include the ability to make VOIP phone calls in the following scenarios:

- Calling from an IP phone to another IP phone and vice versa.

- Calling from an IP phone to a telephone in the KU PBX network and vice versa.

- Calling from an IP phone to a local Lawrence telephone number and vice versa.

3 CISCO AVVID FOR IP TELEPHONY

3.1 The IP Communications Solution

There has been a noted shift in the communications market trends towards Voice over IP. There has been discussion about a shift into the VOIP market for a quite some time, and seems like there is rapid adoption of this technology not only in the communication industry and regular corporations, but also by vendors who have historically used time-division multiplexing (TDM) infrastructures. [1]

The Cisco IP Communication solution comprises of the following selected solutions:

<u>IP Telephony</u>

This solution includes both hardware and software packaged as a solution for transmitting voice communications over a network using the standard Internet Protocol. The hardware and software products concurrently form the basis of call processing agents, IP phones, video devices and other voice applications that provide various services to the Telephony deployment. [1]

Unified Communications

The unified communications solution delivers features for unified messaging over email, fax, and voice all managed from one single mail box. This would include features of receiving messages through all those mediums, and/or through each of those mediums or custom build for each user's requirement. This also includes a full featured advanced voice mail system which Cisco calls "Intelligent messaging" that improves communications across an organization. [1]

Video Telephony and Rich media conferencing

The Cisco Video Telephony solution is a way to provide real time video over IP the same way with the IP Telephony solution. The hardware and some of the software however would be different in the case of video, but it is as simple a concept as video of both parties over a phone call. The rich media conferencing is an extension of the video, voice communication solutions with conferencing features built into it for a virtual meeting environment. [1]

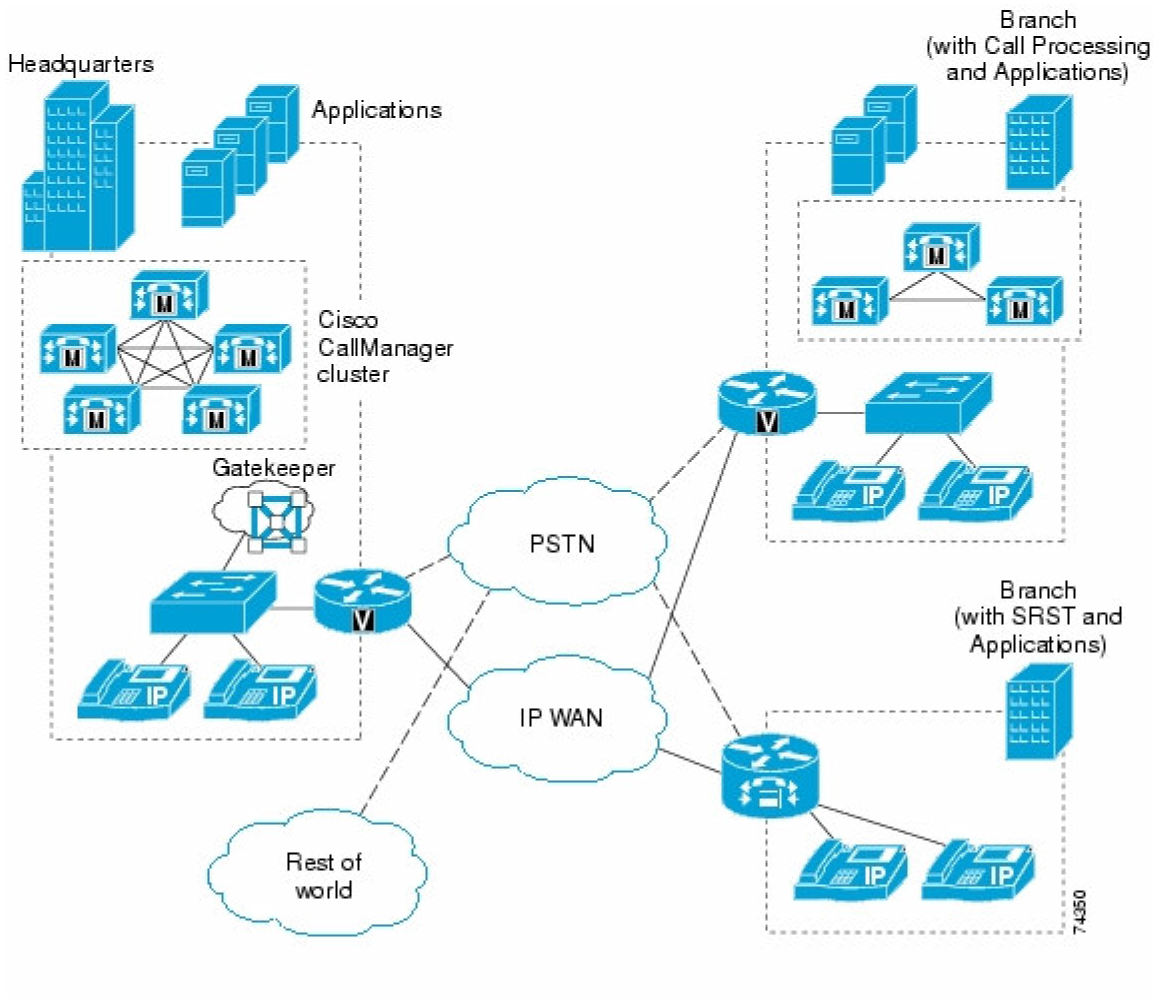
Customer Contact and Call center features

The customer contact solution includes provisions in an IP Telephony system to be able to integrate call center type features and provide monitoring and upkeep mechanisms for management to run reports and verify different parameters to measure and make possible improvements on their call center performance. This solution is deployed on a solution call the Internet Protocol Customer Contact (IPCC). [1]

3.2 IP Telephony Deployment Components

This section introduces the various components of a typical IP telephony solution. Figure 3-1 is a depiction of components in a telephony solution, although depending on the needs of a business, main or branch sites may or may not contain all the depicted elements.

Figure 3-1 Components of an IP Telephony deployment [1]



The main network infrastructure in a solution includes PSTN gateways, analog phone support and digital signal processor farms which include Cisco voice gateways, Cisco routers, and Catalyst switches equipped with the appropriate Cisco IOS platforms. [1] The above mentioned network elements support endpoints as clients like the IP phones (desk phones, video enabled endpoints and wireless IP phones) or fax machines and also provide interfaces to support integration of legacy systems. [1]

In the applications realm, the Cisco Call Manager software could be hosted on various kinds of server hardware. The Call Manager acts as the core call processor as a standalone call processor, a distributed one or in a cluster. In either case , the call manager provides call processing to the main or remote sites. Different call processing scenarios are discussed in following section. Other than core call processing applications, servers in an IP Telephony deployment could also provide services made available by the Cisco IP Communication solution each of the deployment models.

Another important aspect of a typical IP Telephony deployment is Quality of Service. In a converged network, there is much concern about how different media can be delivered through the same medium, and how this affects service. For example, the WAN links between each remote site would be a carrier for data and voice (or video). This brings up concerns of whether voice (or video in certain cases) would be affected if simultaneous data transfers occur over the WAN link.

According to Cisco Systems, “*Voice, as a class of IP network traffic, has strict requirements concerning packet loss, delay, and delay variation (also known as jitter). To meet these requirements for voice traffic, the Cisco IP Telephony solution includes Quality of Service (QoS) features such as classification, queuing, traffic shaping, compressed Real-Time Transport Protocol (cRTP), and Transmission Control Protocol (TCP) header compression*.” [1]

3.3 IP Telephony Deployment Models

There are different models that a corporation can choose from with a deployment based on the needs and requirements of the business. Many different parameters have to be taken in consideration before opting for any of the proposed models with the Cisco IP Telephony Solution.

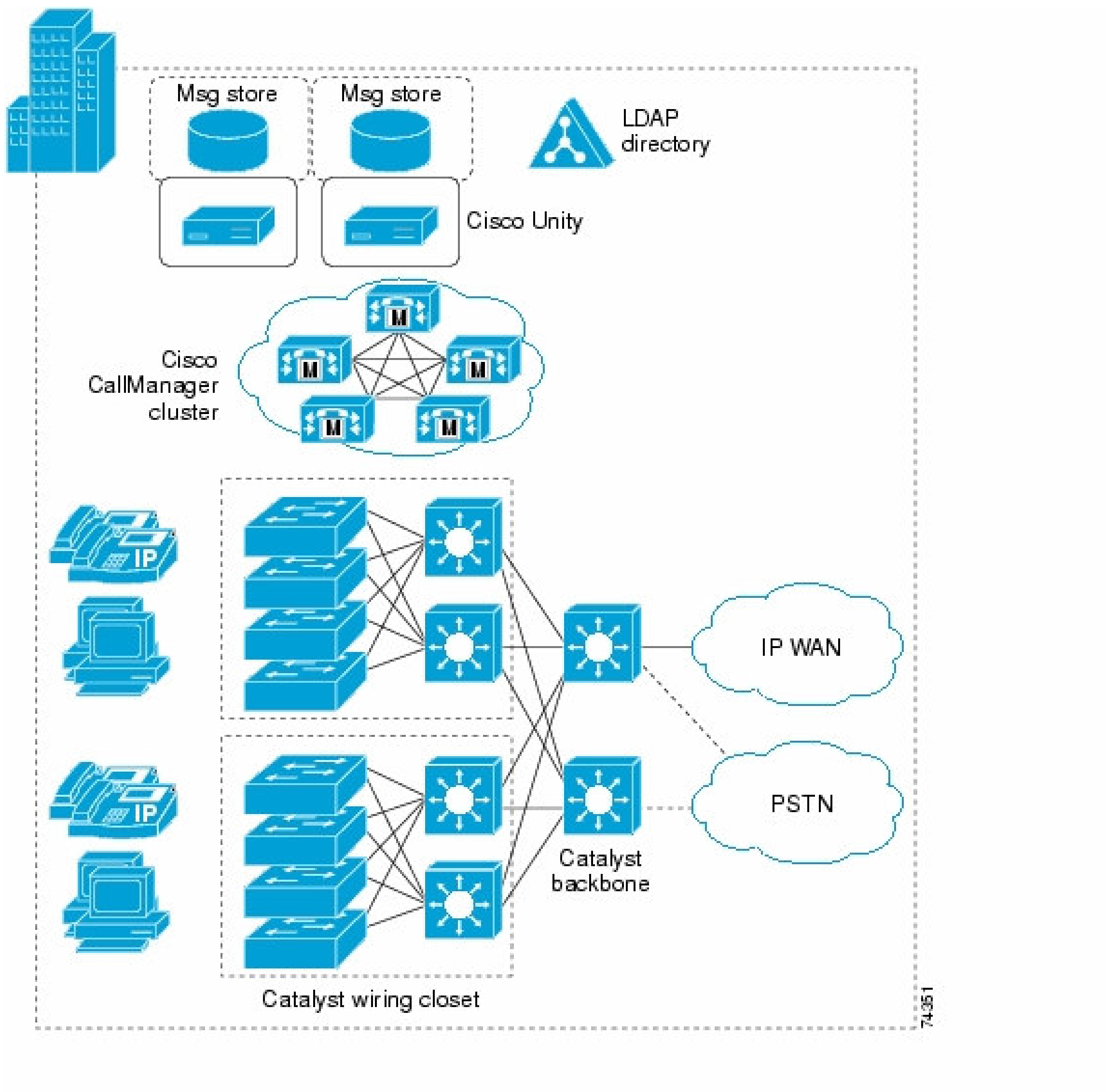
This section introduces some of the models that are generally implemented by corporations. Each of them differ from one another based on the size of the business, the type of business and the degree of dependence of business needs on IP telephony.

This section will discusses the different kinds of deployment models such as the single site model and multi-site WAN models with centralized or distributed processing. There will also be a discussion on a model that is designed for Survivable Remote Site Telephony (SRST).

3.3.1 Single Site Deployment Model

A single site deployment model consists of only one IP Telephony site where all the call processing happens at this main site. There are no VOIP connections going out of a WAN link to provide voice communications to any kind of a remote site.

Figure 3-2 Single Site Deployment Model [2]



In this model, all IP traffic resides and interacts within the LAN or Metropolitan Area Network (MAN) [2] at the main central "office". All calls within the main site will go over IP within the LAN/MAN and any calls that need to be routed outside this LAN/MAN would have to go through the PSTN. It is also possible for the data to travel through a WAN link, but in a single site IP Telephony deployment, there is no voice traffic that would traverse this data-only WAN link. [2]

In figure 3-2, the main call processing is done by a single Call Manager, or with a cluster of Call Managers. [2] A cluster of servers would be needed in the case that there are a large number of call agents. For smaller sites, a single site deployment would require one call processing server.

Outbound calls would have to go over the PSTN link. In figure 3-2, the connection to the PSTN is through the Layer 3 Core Catalyst switches. These two catalyst switches are categorized as the core layer of a deployment. Normally, categorization of network elements in a tiered model consists of the core, distribution and access layers.

The Cisco Unity servers in figure 3-2 are the elements that host voice mail applications in a Cisco IP Telephony deployment. [2] This voice mail may also integrate with an LDAP directory or any third party directory service for information for voicemail boxes on the system.

Since all the IP voice is used within the LAN/MAN, transcoding resources are not required. There is no translation of codec's for incoming voice. Hence, in a single site environment the only codec used is the standard G.711 pulse code modulation [3] on all end points. This frees up the DSP resources for other activities such as conferencing and media termination points. [2]

The dialing plan on a single site deployment is easier [2] than any of the other deployments because the only dialing plans needed would be plans for calling within the site and to be able to connect to the Telco to get out the PSTN. There would have to be configurations for long distance and international calling, but it would not be necessary to create dial plans to other VOIP sites with dissimilar phone number nomenclature.

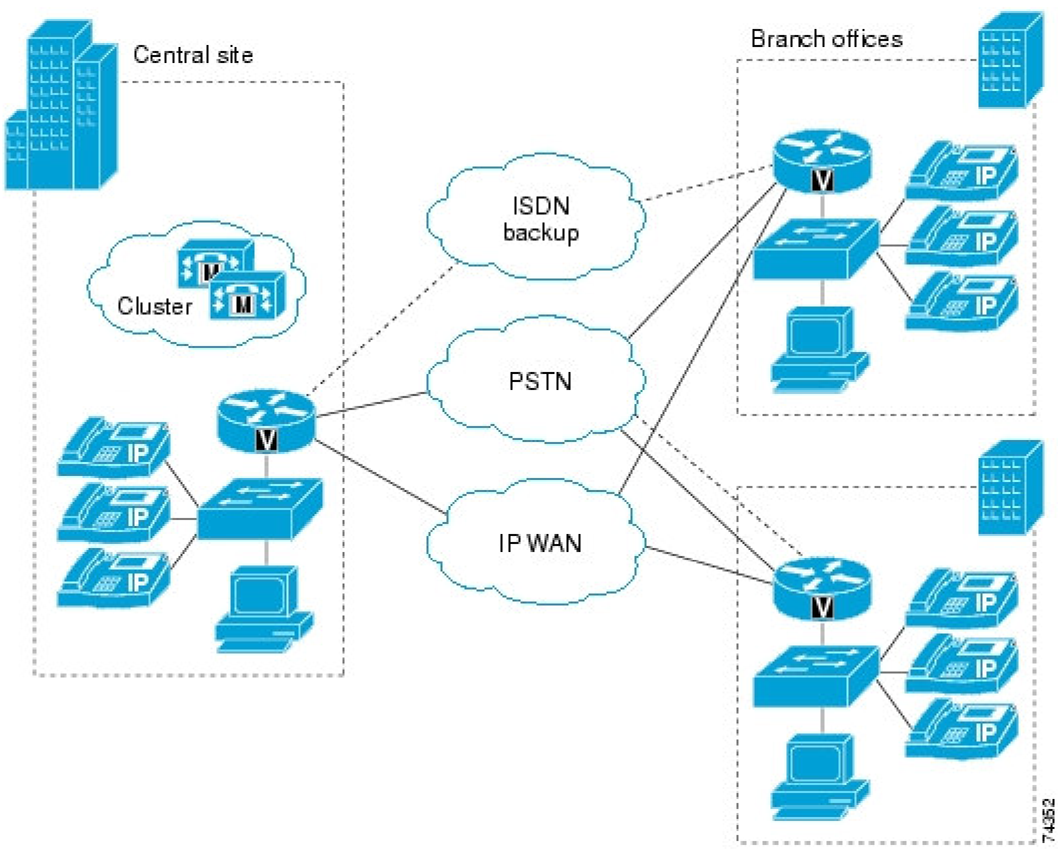
As per Cisco Systems, it is recommended that MGCP be used at the gateways for the PSTN unless specific H.323 functionality is required. [2] In this project implementation, a single site deployment was configured using both protocols.

The single site deployment is common among small to medium sized businesses that do not have branch offices at different locations. Single sites are normally easier to deploy and maintain.

3.3.2 Multi-Site WAN Deployment Model

A multi-site deployment is implemented when there is more than just a main office that needs IP telephony. One main site or multiple remote sites may be involved in the deployment. This section discusses some of the features of a multi-site WAN deployment model.

Figure 3-3 Multi-site WAN deployment Model [2]



In figure 3-3, all call processing occurs at the main site. All call processing for the branch sites are done over the IP WAN cloud. The IP WAN also carries all the call control signaling functions between the branch sites. [2] The call processing is done by a call manager cluster. However, a cluster is not necessary if there are fewer call agents. Since voice media traverses across the WAN, careful design and configuration is required with Quality of Service (QOS). QOS has to be configured on the edge routers to make sure that the voice is given priority over the data packets, such that delay or jitter does not cause voice quality to deteriorate. [2]

According to figure 3-3, each of the branch sites has a connection to the PSTN. The call processing for the branch sites occurs at the main Call Manager cluster. Calls from the branch site go out through their gateways after the Call Manager processing. The Call Manager processes the PSTN call, sends the call information over the WAN link the branch and forwards it out though that branch gateway.

The IP WAN connection can be to be transported over Frame Relay, Asynchronous Transfer Mode (ATM), Multi-protocol Label Switching (MPLS) and over a Virtual Private Network (VPN) connection. [2] By using this deployment model, the remote sites can bypass toll charges since it routes calls through the main site gateway. For example, all the sites could all be in different cities and for the sites to call each other would be classified as a long distance call. Since all voice communications between sites go over the WAN, there are no toll charges applied through the Telco.

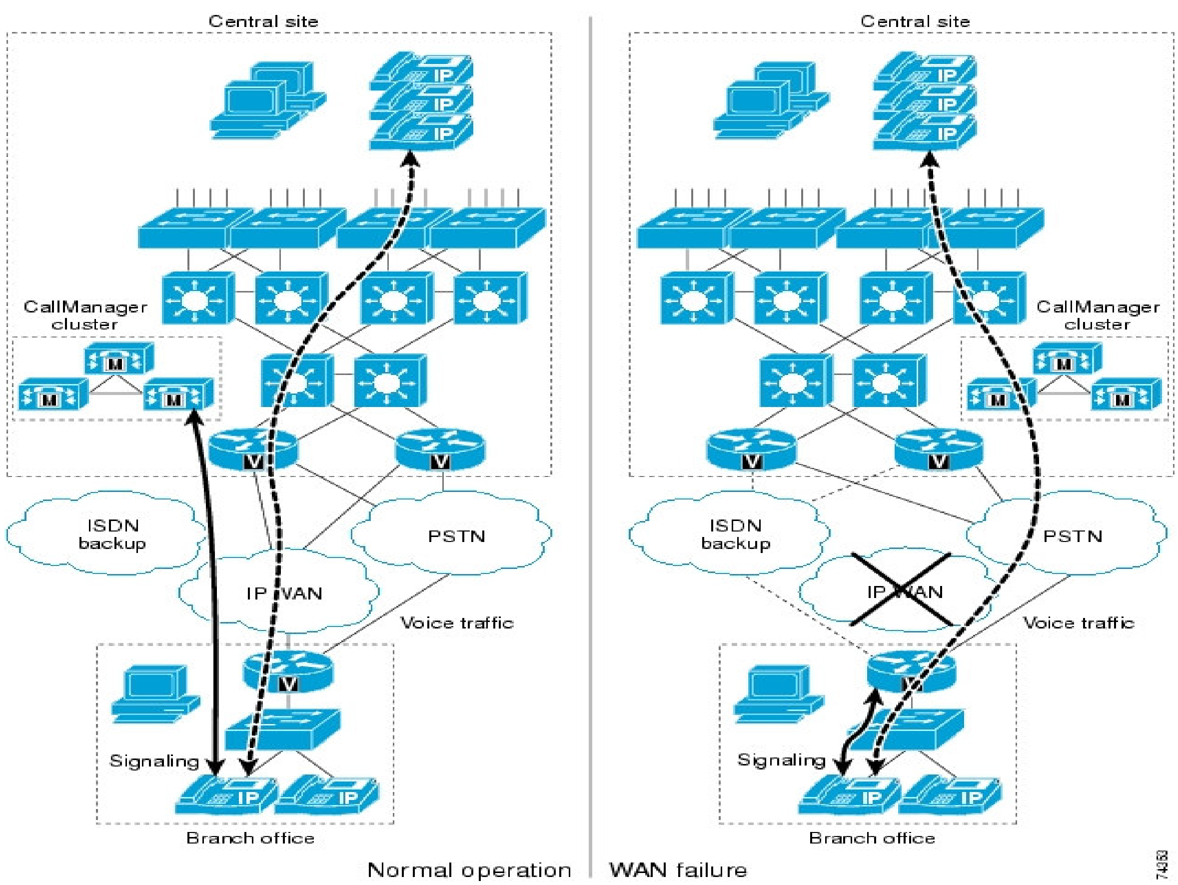
If the WAN link goes down, the back-up ISDN connection can be configured so that it would dial the Telco to provide a temporary connection. This temporary connection would now act as a WAN link until the original WAN link has been fixed. [2] The next section discusses the details of Survivable Remote Site Telephony (SRST).

In some cases, a business might require a distributed call processing model. For example, when one particular site has a larger call volume compared to the main or other branch sites, another call manager or a call manager express (router call processor) can be implemented for call processing [2]. This reduces the load on the main call manager cluster and also simplifies the call processing for the site with a large call volume.

3.3.2.1 Remote Site Survivability

When deploying a multi-site WAN model with centralized call processing, a network has to be designed with features that ensure the high availability at the remote sites. Considerations should be made for redundant WAN links at the branch router for voice and ISDN back up for high availability of voice at the remote site. According to Cisco Systems, "Survivable Remote Site Telephony (SRST), provides high availability for voice services only, by providing a subset of the call processing capabilities within the remote office router and enhancing the IP phones with the ability to "re-home" to the call processing functions in the local router if a WAN failure is detected." [2]

Figure 3-4 SRST Scenarios [2]



An SRST configuration is depicted in figure 3-4. In “Normal Operation”, all WAN links are up and functioning. The phones get signaling options from the call manager cluster at the main site. At this point the branch site router does not provide call signaling. When the WAN link is lost, the router at the branch office is triggered into SRST mode. This provides certain amount of call processing to the IP phones at the branch. This router also provided call signaling and registration options and communicates with IP phones on the main site through the PSTN channel. Although there are no advanced call manager features at the branch office, they are still equipped with basic calling features. [2] After the WAN connectivity is re-established, the phones re-register with the call manager and the gateway router shuts out of SRST mode and all functionality returns to the initial “Normal operation”. [2]

4 IMPLEMENTATION

4.1 Overview of the Implementation

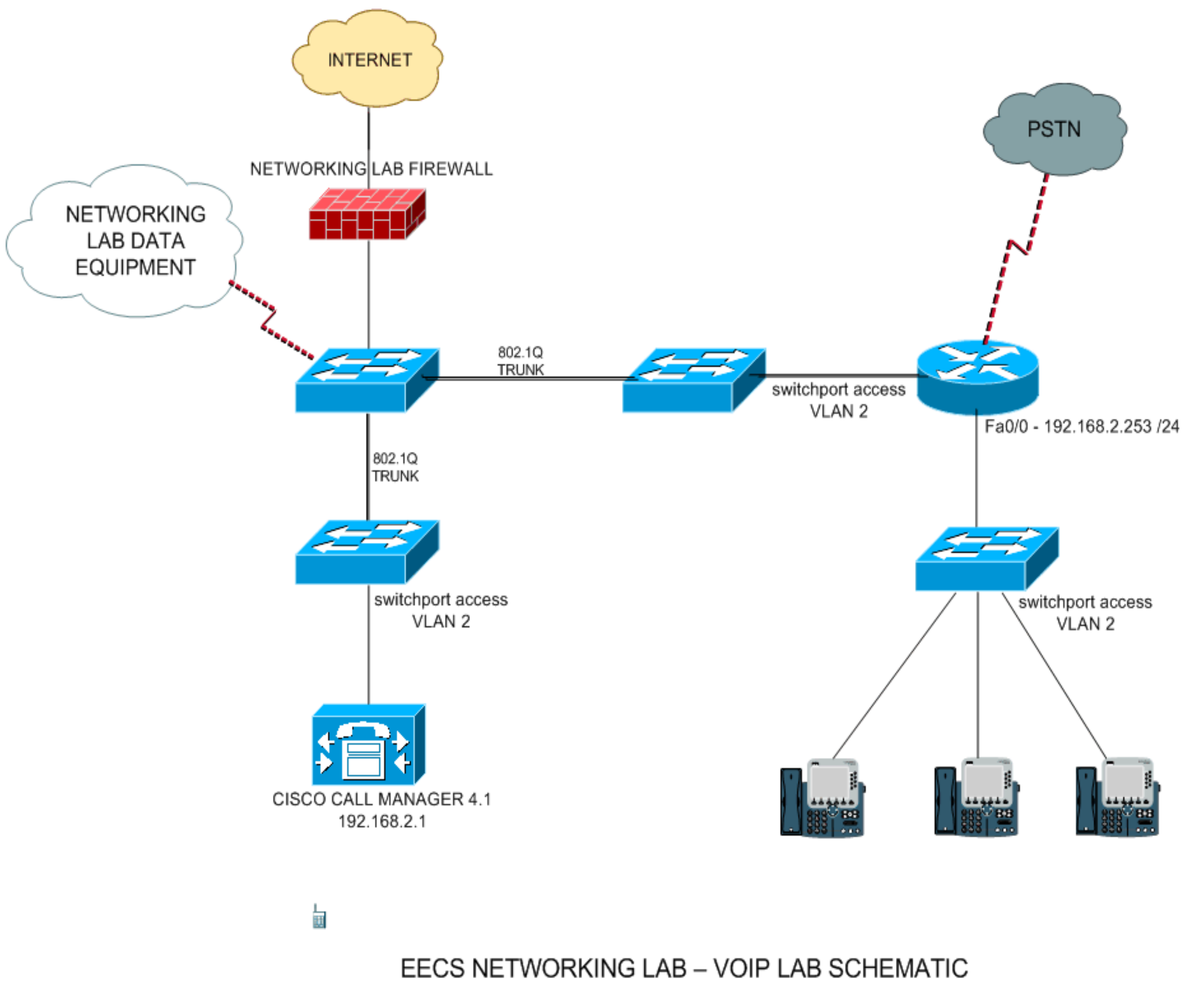
In regard to the deployment models, the implementation at the EECS Networking lab was a single site deployment with one centralized call processing agent. This lab also had one connection to the PSTN via an FXO port. There was only one data-only internet connection through the lab firewall. The VOIP lab connected to the outside PSTN through a KU Mini-PBX in the Eaton building.

In this section, the following topics relating to the implementation will be discussed: basic network schematic including the IP addressing scheme, various network elements used, implementation of IP Telephony with both H.323 and MGCP, configurations on the call processing agent and gateways. The end of this section will include different outputs from both the Call Manager and the gateway.

4.1.1 Infrastructure and Schematic

Figure 4-1 is a schematic of how the network was set up for IP Telephony.

Figure 4-1 Networking Lab VOIP Schematic



Although the Call Manager was physically in a different area than where all the phones were, 802.1Q trunks propagated the VLAN 2 (VOICE VLAN) information to that part of the network. The gateway router had a connection to the PSTN through its FXO Interface card. The following section will explain more about VLANs and the IP Addressing schemes.

VLAN Information and IP Addressing

Two different VLANS [5] were implemented to separate the Voice information to propagate on a different subnet than the data information. All the IP phones and all the voice equipment were a part of VLAN 2. VLAN 2 was the Voice VLAN and VLAN 1 (native) had been assigned to the Data VLAN. The Voice VLAN 2 subnet ranges in the 192.168.2.0/24 subnet and the Data VLAN 1 subnet ranges in the 192.168.1.0/24 subnet.

The Cisco 2801 Routers Ethernet interface had been set with an IP address of 192.168.2.253/24. This connected down to a Cisco 2950 POE switch which had its IP address set on the 192.168.2.201/24 network. This switch fed power to each of the IP Phones that were plugged into this switch. The Cisco Call Manager was on same Voice VLAN and had an IP address of 192.168.2.1/24. All the phones were configured to pull Dynamic Host configuration (DHCP) addresses within the voice VLAN range.

4.1.2 Equipment

This section lists the equipment that was used in the implementation of the IP Telephony deployment project. All equipment in the lab were Cisco products. These products are often implemented by corporations in the industry as their primary voice communication medium.

Call Processing:

The core call processing agent was hosted on an IBM MCS-7845-I server. The software used for the call processing was the Cisco Call Manager 4.1 which ran on a Windows 2000 platform.

Router/Gateway:

The core hardware component for this install was the Cisco 2801 Router. This router included two different interface cards that enabled it for Voice communication. These cards included the four port FXO (Foreign Exchange Office) card and the four port FXS (Foreign Exchange Station) card. The router also needed an IP voice enabled software image.

Access Switches:

Configuration tasks were performed on some Cisco switches that were involved in this project, mainly to pass VLAN information through 802.1Q trunks .[5] One particular Cisco 2950 Power over Ethernet switch (POE) was used to provide power and access to the Cisco IP phones in the lab. This switch also had basic LAN Quality of Service (QOS) configurations for IP voice packets. The switch needed a particular IP Voice enabled software image for support of QOS commands.

IP Phones:

The IP phones in the lab are Cisco 7970 phone sets. These phones have a 2 port switch built into them, so that we can connect PCs to them instead of having two different connections to the LAN at the desk. The switch port that the switches are connected to should be configured as a trunk port so that it can pass both Data and Voice VLAN information to the phone for the PCs Data access. The phone also pulls power from the POE switch, therefore does not need a power adaptor, although is equipped with a power adapter in the case of a switch that is not POE.

POTS Line:

A Plain Old Telephone Service (POTS) line was provisioned to lab for gateway access to an analog connection via a KU PBX. This provided access to a legacy connection, which allowed the implementation and testing of VOIP to analog voice communication and vise versa. This line was the connection from the internal VOIP network to the PSTN (Public Switched Telephone Network) via the FXO interface card on the Voice enabled router.

4.2 Basic Configuration

4.2.1 Basic Router Configuration

All the IP phones were to be assigned DHCP addresses. Therefore, a DHCP scope had to be declared. [6] Once this scope was configured, IP addresses would be assigned from within that scope.

DHCP could either be configured on the Cisco Call Manager, or on the Cisco 2801 router. In this implementation, the DHCP pool was configured` on the router.

The DHCP pool on the 2801 Router was called CISCOVOICE. The 2801 Router was configured with a host-name of “VoiceRouter”.

```
!
hostname VoiceRouter
!
ip dhcp excluded-address 192.168.2.1 192.168.2.20
ip dhcp excluded-address 192.168.2.200 192.168.2.255
!
ip dhcp pool CISCOVOICE
  network 192.168.2.0 255.255.255.0
  option 150 ip 192.168.2.1
!
```

The scope of this DHCP pool was the 192.168.2.0/24 subnet, but a few IP addresses were excluded. The commands configured before the IP DHCP Pool excluded the following ranges of IP addresses: 192.168.2.1 – 192.168.2.20 and 192.168.2.200 – 192.168.2.255

An important part of the DHCP pool configuration was the “option 150 IP 192.168.2.1” statement. Option 150 in a DHCP scope tells a client the IP address of a TFTP server. [7] So on this CISCOVOICE DHCP pool, the TFTP server was the Call Manager. When a phone got its IP address from the DHCP server (router), it got information about where to get its phone load and phone configuration via TFTP (from the Call Manager). [7]

```
!
interface FastEthernet0/1
 description VOICE VLAN
 ip address 192.168.2.253 255.255.255.0
 duplex auto
 speed auto
!
```

The above commands show the configuration of the IP address of the VOICE VLAN default gateway. The router was configured with a 192.168.2.253/24 IP address and the VOICE VLAN was configured on FastEthernet0/1 of our VoiceRouter.

4.2.2 Basic Call Manager Configuration

After installing the Cisco Call Manager on the server, the Call Manager services had to be activated to run in the background. For example, since a separate TFTP server was not available, the "Cisco TFTP Service" on the Call Manager was activated. One of the other functions that needed to be activated on the Call

Manager was the Cisco Music on hold (MOH) Audio Translator that converts audio files so that they can be used as music on hold resources. The Cisco RIS Data collector also needed to be activated. This service gets real time information such as IP addresses and distributes this information as needed. [8]
The main service that needed to be activated is the "Cisco Call Manager" Service. This service provides call processing, signaling and call control functions. The server with the Cisco Call Manager service activated needed the Cisco Database Lay Monitor and the Cisco RIS data collector services activated for proper functioning. [8]

When the dial plans on the call manager were configured, a choice about the protocol needed to be made before the configuration process. Depending on the choice of protocol (either H.323 or MGCP), the configurations on the Call Manager would differ. This will be discussed in the following sections specific to each protocol.

There were some common configuration tasks on the Call Manager that did not change for the two different protocols. These configurations included configuration of Class of service and Dial Plan parameters. Partitions and Calling Search Spaces were the two Class of Service (CoS) components in the Call Manager, [16] where as Route Patterns were a part of a dial plan configuration.

A Partition was always assigned to anything that has a patter, such as a directory number or a route pattern, while a Calling Space were assigned to devices such as phones and gateways.

The following figures show the configurations of Partitions, Calling Search spaces and Route Patterns.

Figure 4-2-1 List of Partitions

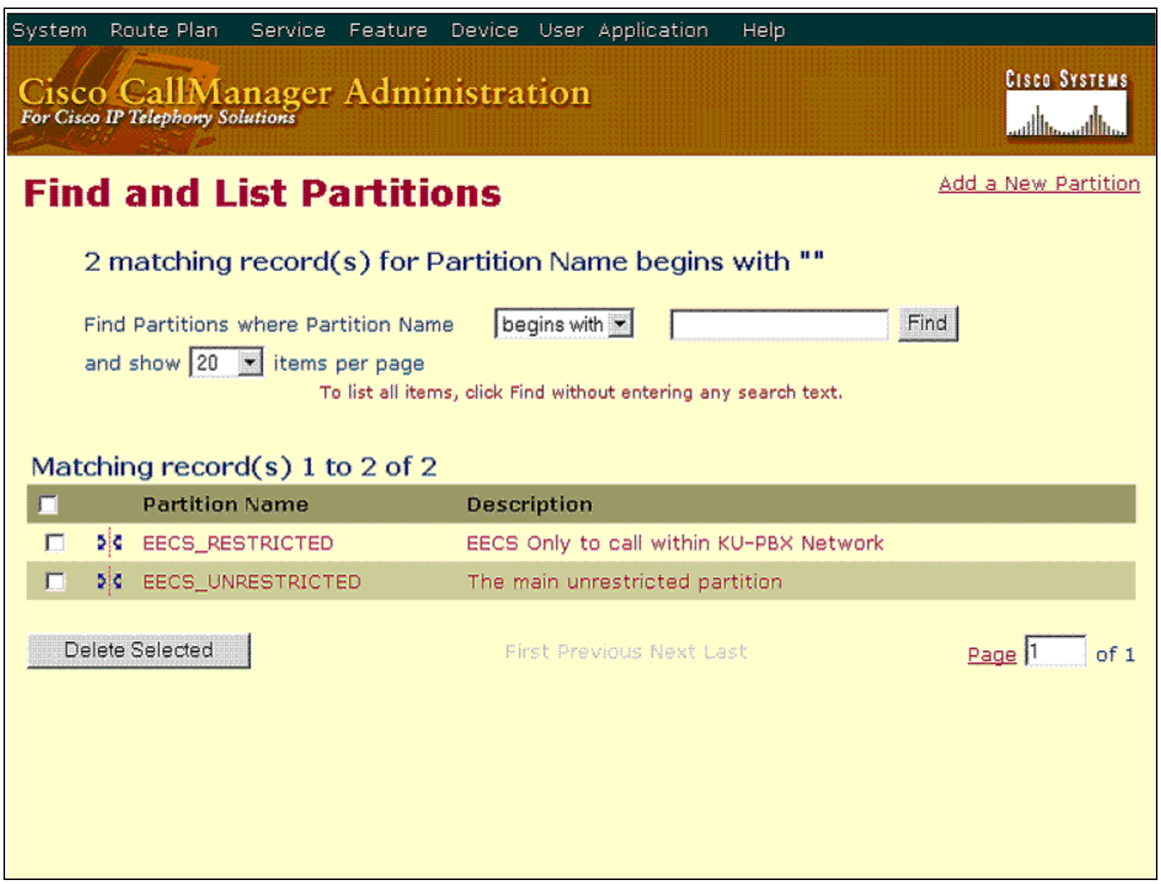


Figure 4-2-1 shows the two different partitions configured which were called EECS_RESTRICTED and EECS_UNRESTRICTED in the Call Manager.

Figure 4-2-2 Partition Configuration

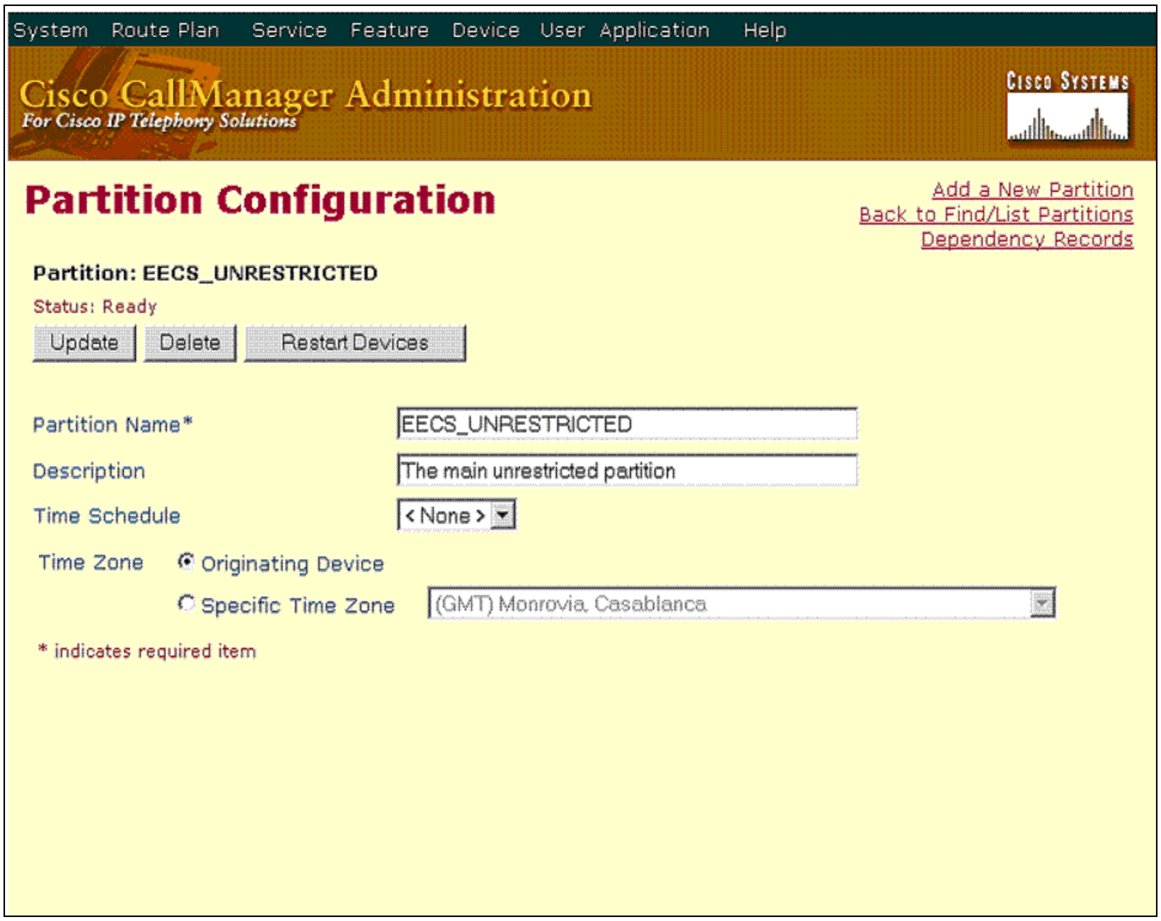
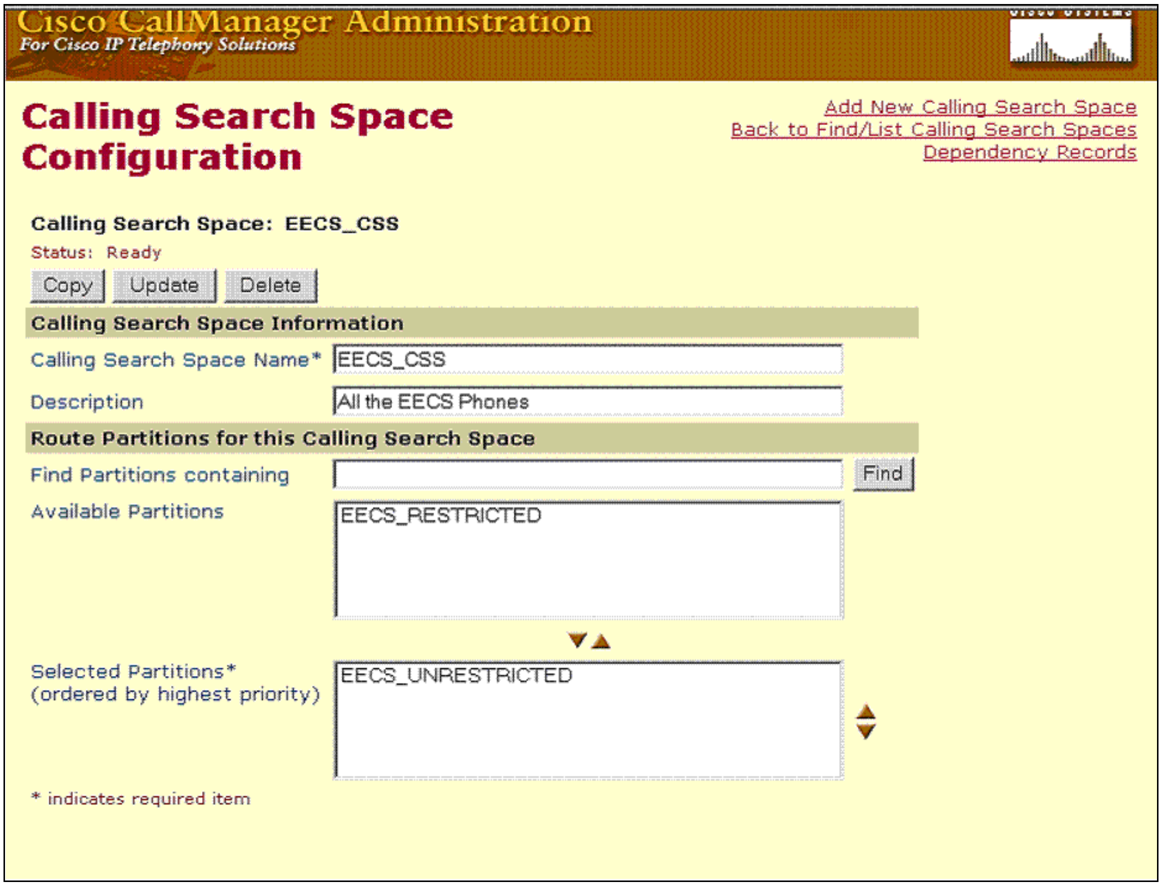


Figure 4-2-2 shows the configuration of the EECS_UNRESTRICTED partition. This partition would be applied on different elements, for example a calling search space or a route pattern.

Figure 4-2-3 Calling Search Space Configurations



In figure 4-2-3, a Calling Search Space called EECS_CSS was configured to include the EECS_UNRESTRICTED partition. This Calling Search space can in turn be applied to phones or gateways.

The EECS_UNRESTRICTED partition was added into the EECS_CSS Calling Search Space. EECS_RESTRICTED was another partition that was configured for test purposes but was not used for the deployment.

Figure 4-2-4 Route Pattern List

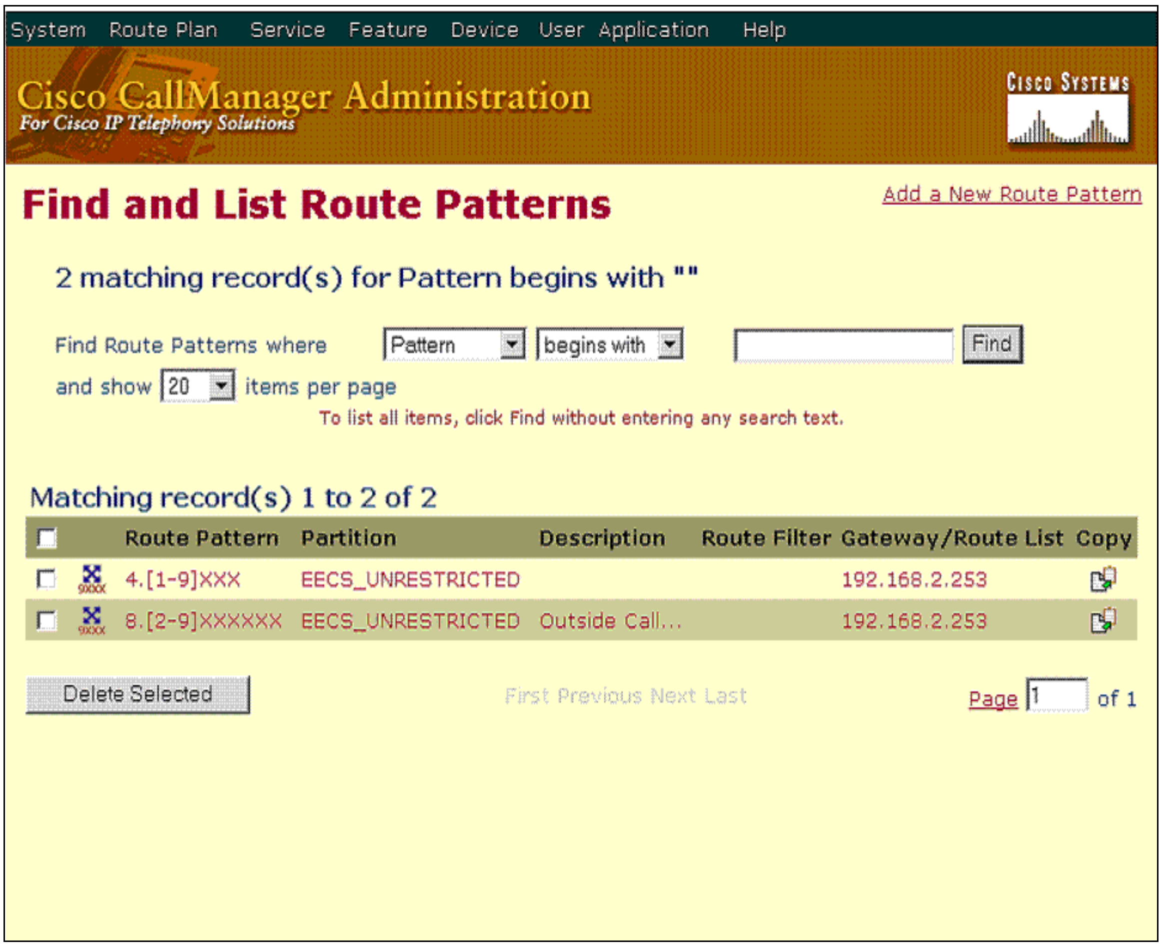


Figure 4-2-4 lists the different Route Patterns that were configured in the Call Manager. As shown, these Route Patterns described the dial plans. There were two different dial plans configured: one for within the KU PBX network 4-digit dialing and another pattern for 7-digit local Lawrence, KS number calling. Both these patterns had the EECS_UNRESTRICTED partition applied.

The next two figures show the details of each of the route patterns.

Figure 4-2-5 Four Digit Route Pattern

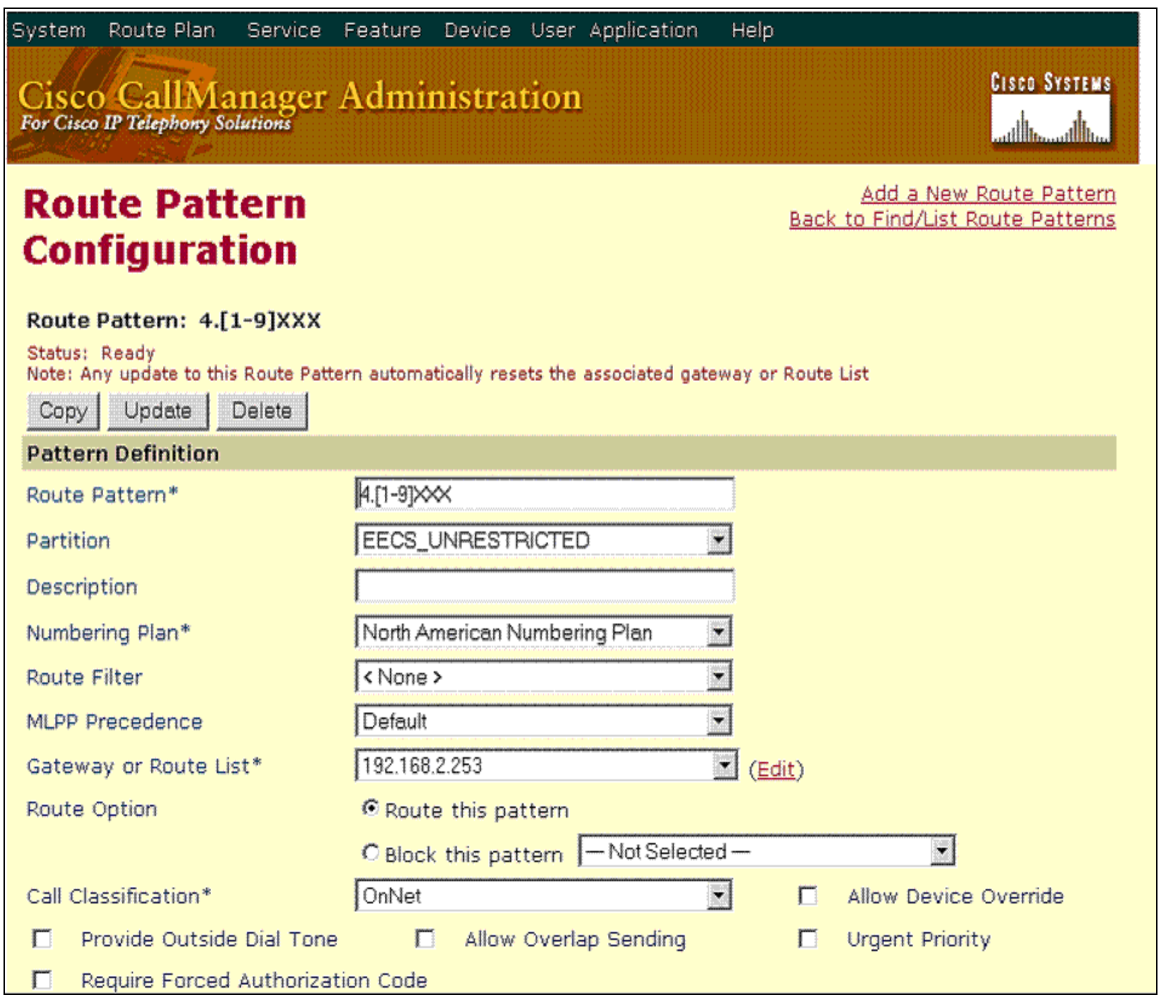
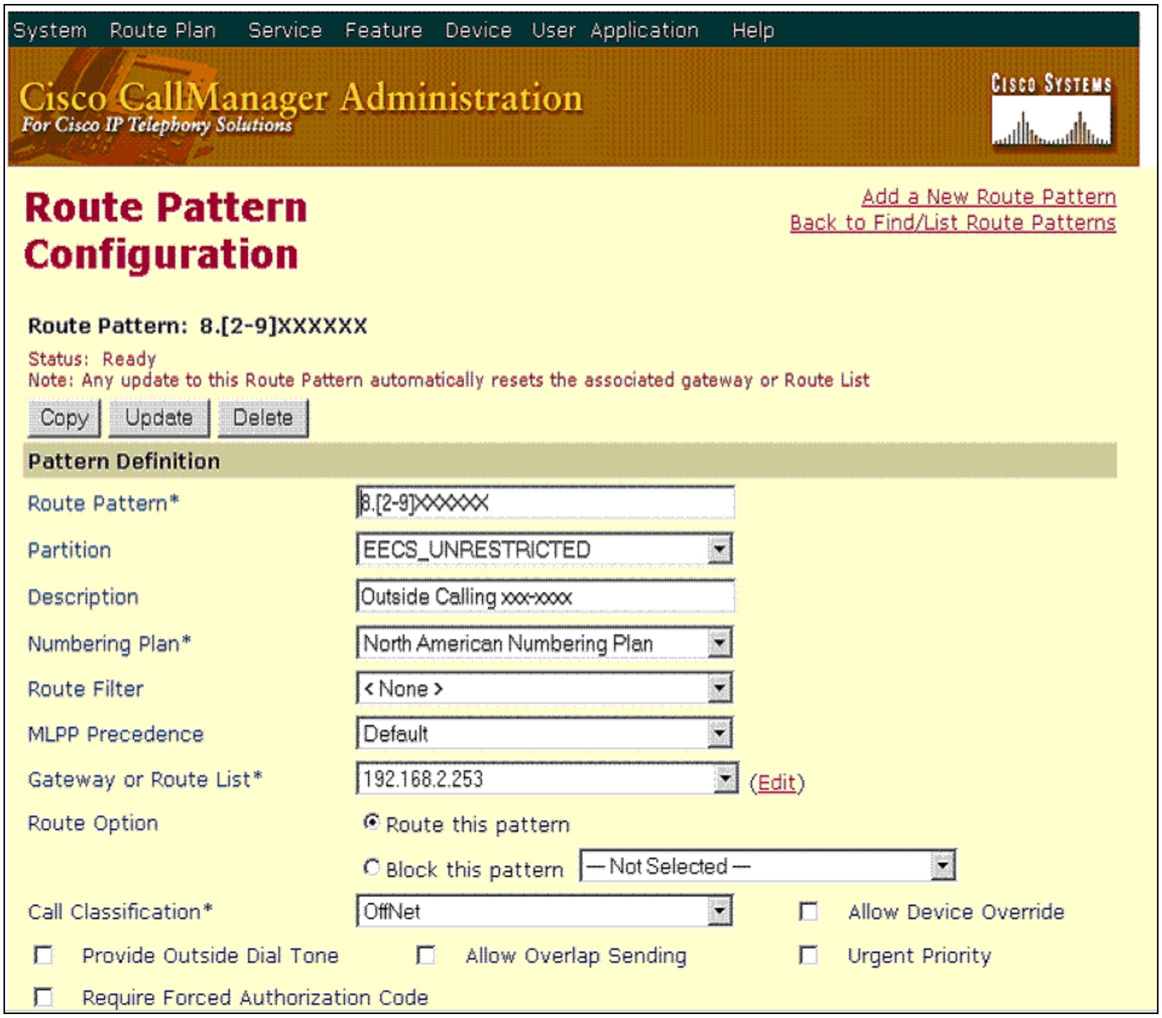


Figure 4-2-5 shows the configuration of the 4-digit dial plans while figure 4-2-6 shows the configuration of the 7-digit dialing. The gateway field in these figures point to the IP address of the gateway router. This was specific to the H.323 gateway configuration. If this was set up as a MGCP gateway, the configuration of the gateway would not be an IP address, but would be the FXO port as configured on the MGCP Gateway. When this Route Pattern was matched, the call was forwarded to the gateway configured in the call manager i.e. the IP address of the gateway or the FXO port depending on the protocol used.

Figure 4-2-6 Seven Digit Route Pattern Configuration



In both these route pattern configurations, the wildcard [x-y] had been used. This wildcard configuration represented a single digit range. For example, 5[3-5] would only match a dialed pattern of 53, 54, and 54. The numbers inside a bracket always represent a range and match only one digit. [8]

4.3 Voice Gateway Configurations

Gateways that are used in IP Telephony must provide support for the following core gateway features that include Dual Tone Multi-frequency (DTMF) relay, supplementary services like hold, transfer and conferencing, and Cisco Call Manager redundancy support. [9]

Both MGCP and H.323 support all of these core gateway features. In the following sections, the configuration of each of these protocols and how they provide these core gateway features in our lab installation will be examined.

This section includes MGCP and H.323 configurations on the Cisco Router and the corresponding configurations for each protocol on the Call Manager.

4.3.1 MGCP Configuration on the Voice Gateway

MGCP is a protocol that is used to control gateways. MGCP does not handle any routing of calls. MGCP depends on the Call Manager for this. This is why the configuration on the router for MGCP is not as detailed as the configuration for H.323. MGCP provides complete support for DTMF-Relay, supplemental services and for Call Manager redundancy.

With an MGCP gateway, there was minimal configuration on the VOICEROUTER. The following configurations were required on VOICEROUTER for MGCP to work with the Call Manager.

```
!
ip domain name ku.edu
!
ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config server 192.168.2.1
ccm-manager config
!
```

The gateway needed to have a domain name configured to be able to register with the Call Manager when using MGCP. The ccm-manager statements were used to specify the protocol and to enable support for Call Manager within MGCP [11]. The "ccm-manager mgcp" specified the protocol in use was MGCP. The ccm-manager config server 192.168.2.1" triggers the router to pull any other configuration needed from the Call Manager. The other ccm-manager commands enabled support for supplementary services to be requested from the Call Manager. [11] The "ccm-manager redundant host <ip-address>" command was used to configure call manager redundancy. This was not configured in this deployment, but is an option in other multiple call processor deployments.

```
!
mgcp
mgcp call-agent 192.168.2.1 2427 service-type mgcp
mgcp dtmf-relay voip codec all mode out-of-band
!
```

This above command line interface (CLI) code contains the configurations on the voice gateway with MGCP specific commands. The statement “mgcp” initiated the MGCP application on the router. The “mgcp call-agent” command specified the call agents IP address, port and the service type. This was the address of the Call Manager and also specified port 2427 for MGCP communication. The “mgcp dtmf-relay” command was used to relay digits through the VOIP network. This command specified the use of all VOIP codec’s and removes DTMF tones from the voice stream. [11]

```
!
voice-port 0/1/0
 no shutdown
!
dial-peer voice 8648060 pots
 service mgcpapp
 port 0/1/0
!
```

The voice-port 0/1/0 was the first port on the VIC4-FXO voice card. The break down of the numbering nomenclature was voice-port *slot/subunit/port* [12]. The dial-peer configurations set up the dial peer to use the MGCP application service. Any call going out on the FXO port, or coming into the FXO port would know that the protocol in use was MGCP. [12]

All these configurations on VoiceRouter needed to have complimentary configurations on the Call Manager so that the gateway can register with Call Manager and also pull configurations from the call processing agent.

4.3.2 MGCP Configuration on Call Manager

When using MGCP as the gateway protocol, the Call manager was the device responsible for most of the call processing. The gateway pointed to the call manager for all the work to be done. All the dial plans for call routing were configured in the call manager. The configuration of a Route pattern in the section “Basic Call Manager configurations” was examined, but that was an example for H.323. This section discusses the configurations tasks completed for adding a MGCP gateway and configuring a route pattern for an MGCP gateway configured with IOS commands pointing to the call manager.

Under the CCMAdmin page (the main page for Call Manager administration), selecting Device > Gateway > Add New Gateway adds a new gateway in Call Manager. Figure 3-3-1 depicts the selection of the chassis of the router to be added as a gateway and the protocol chosen to run on this gateway.

Figure 4-3-1 Adding an MGCP Gateway

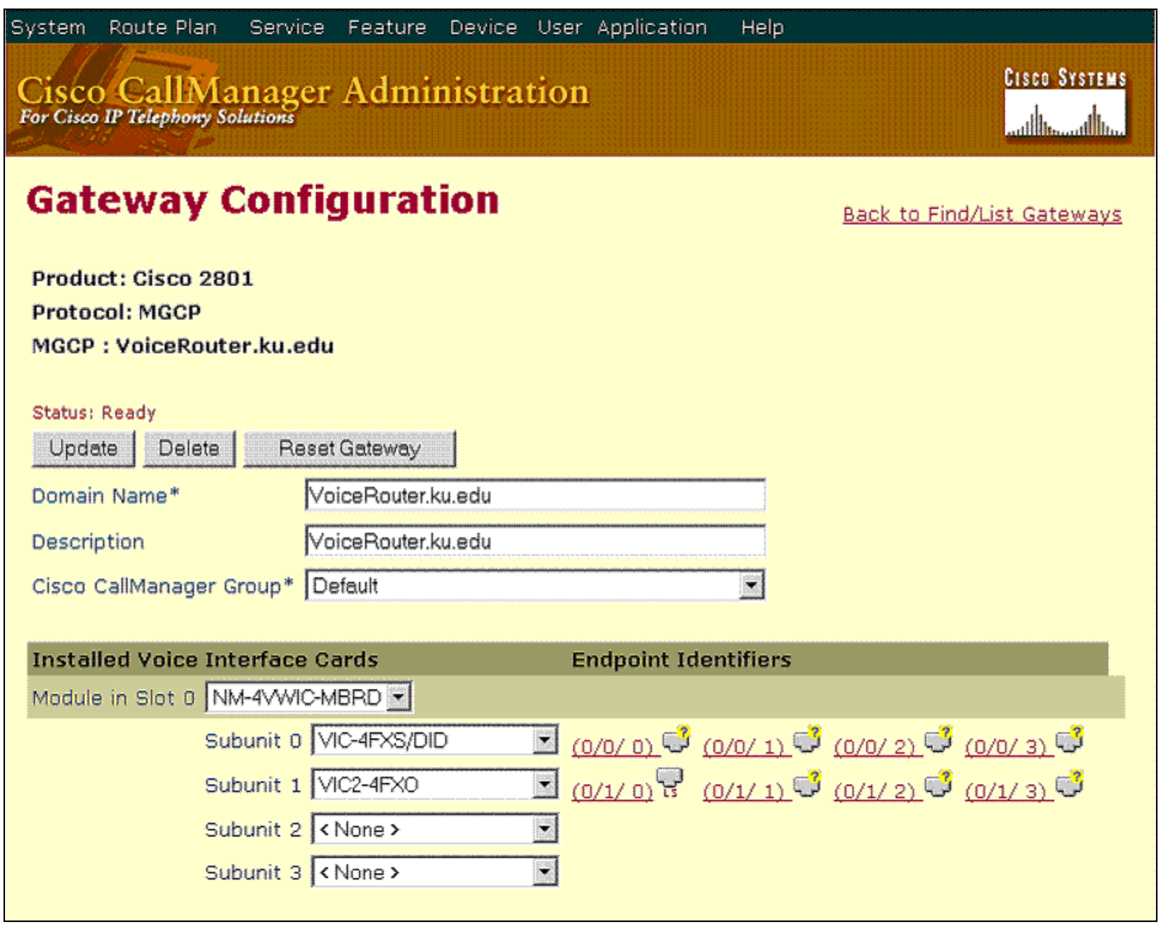
Clicking next prompted into the next configuration screen where details of the voice gateway were filled in. It is very important to have a domain name configured on the IOS router in order for Call Manager registration. The domain name was configured as "ku.edu" on the Cisco router. The hostname of the router was configured as "VoiceRouter". So the domain name as stored in the Call Manager configuration should be listed as **VoiceRouter.ku.edu.** If this was incorrectly configured on the IOS, the voice gateway would not register with Call

Manager. However, if there was no domain configured on the Router, the hostname of the IOS router in the Domain Name field in CCMAdmin could be used.

Figure 4-3-2 Gateway Configuration

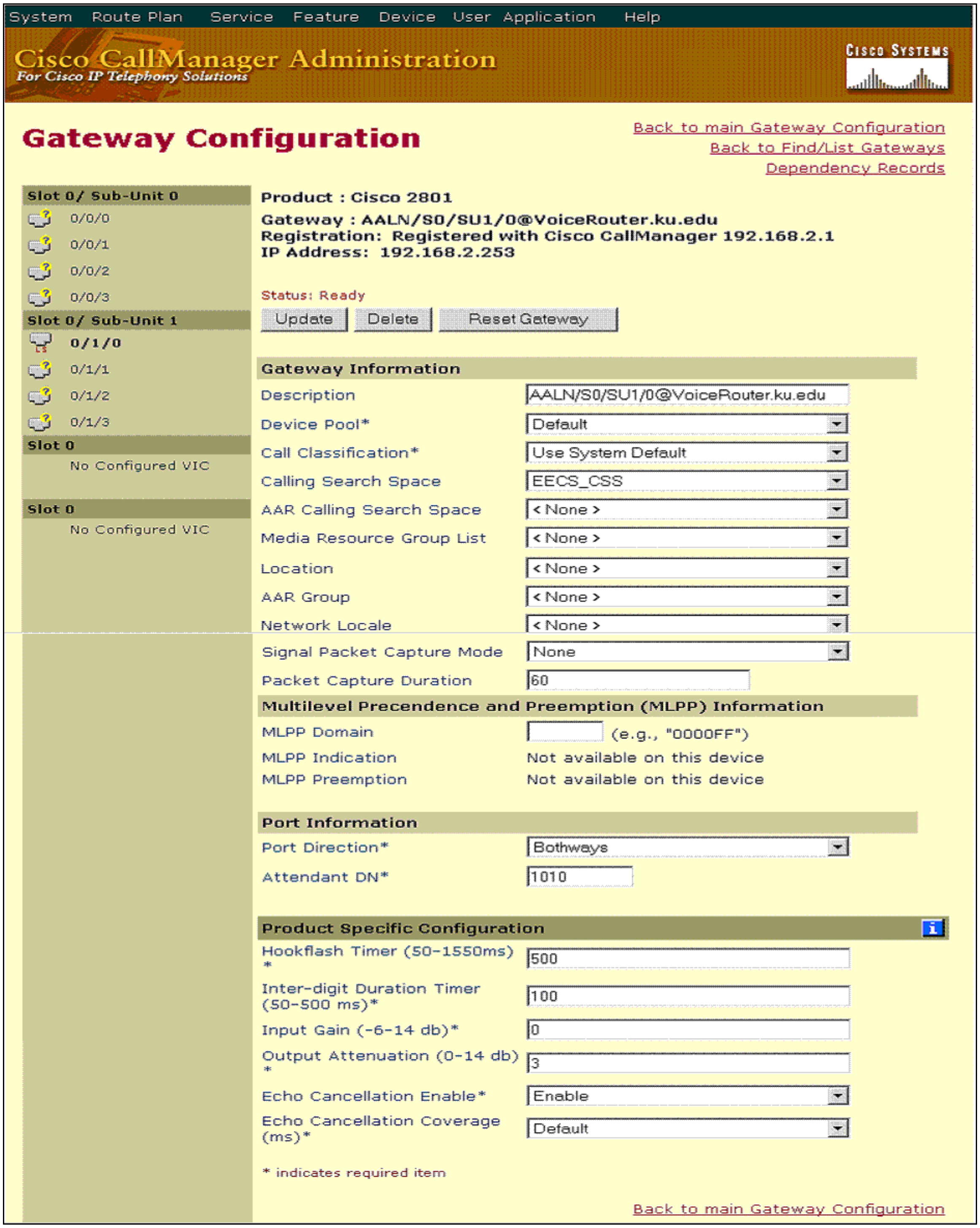
After the name and description was entered, the module type in slot 0 had to be selected. To get this information from the router, the “show diag” can be used on the router. This displays detailed information for the EEPROM, the motherboard, modules in the router, the WIC's (WAN Interface cards), and the VIC (Voice Interface Cards that are present on the router. Figure 4-3-3 shows the choices for the modules, FXO cards and FXS cards that have been selected. The order of the selection was important because the naming nomenclature in the Call manager should exactly match the voice-port naming convention.

Figure 4-3-3 Gateway Configuration - Voice Card Selection



As shown in figure, Slot0/Subunit0 was selected to be the VIC-4FXS/DID card, which corresponded with the slot number that the VIC-4FXS card plugged into the router. Subunit 1 corresponded to the VIC2-4FXO Card that used port 1. Port 0/1/0 was selected and configure as "Loop Start" on the Call Manager. This is shown in figure 4-3-3. Once the gateway was updated and port 0/1/0 selected, the next displayed was a configuration/status page for that particular FXO port, as seen in figure 4-3-4.

Figure 4-3-4 Gateway FXO Port configuration

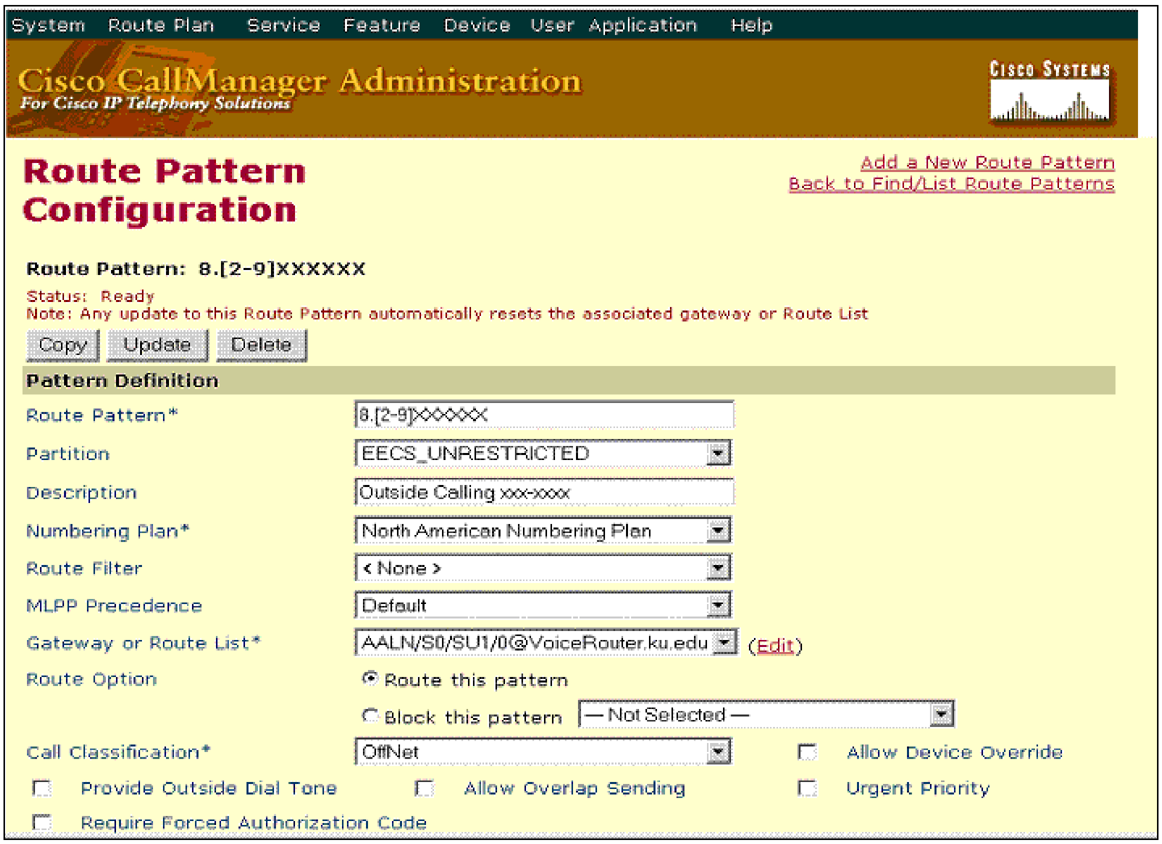


When the gateway was added into the call manager, a Calling Search Space was assigned to the FXO port. Since the EECS_CSS was assigned to the FXO port, any calls coming into the FXO voice-port0/1/0 could only be forwarded to a phone that had its partition listed in the EECS_CSS calling search space. [8] This was governed by the "Attendant DN" Setting tab in the above figure. Any call coming into the FXO port from the PSTN was forwarded to an IP Phone with extension "1010". This IP phone had been configured with a Partition EECS_UNRESTRICTED which was added into the EECS_CSS Calling search space (Basic Call Manager Configuration section). All other settings had been left to the default configuration on this gateway. When the gateway registered in the Call Manager with MGCP, the FXO port to the call manager was recorded as "AALN/S0/SU1/0@VoiceRouter.ku.edu" based on the naming convention described above. The call manager displayed registration information at the top of this page.

As soon as the gateway was registered with the call manager and the IP address of the gateway appeared on the gateway configuration page, the IOS gateway could be considered to be configured correctly. If route patterns were configured appropriately on the Call Manager, outside calling would also be functional, an IP phone that was registered to Call manager could make calls to numbers that matched route patterns. A route pattern configuration was different in the case of

MGCP when compared to H.323. An MGCP configuration is shown in figure 4-3-5.

Figure 4-3-5 MGCP Route Pattern Configuration



In figure 4-3-5, the gateway points to the FXO voice port 0/1/0 of the router, as opposed to an IP address as in H.323. This was the same FXO port that was configured and registered to the Call Manager. With the gateway registered, any IP phone would be able to both receive and make phone calls over the PSTN.

Incoming calls would be forwarded to an IP phone with directory number 1010. Outgoing calls would match the 8[2-9]XXXXXX pattern, or any other pattern that has been configured on the CCMAdmin.

4.3.3 H.323 Configuration on Voice Gateway

H.323 is a gateway protocol which is best described as a suite of protocols. A number of protocols together make up H.323. [15] The specifics of H.323 is beyond the scope of this report, but this section describes the configuration tasks for configuring an IOS gateway for H.323 and the next section describes the corresponding configurations on the Call Manager for H.323.

The configuration tasks for an H.323 gateway are more intensive on the Cisco IOS than on the Call Manager. When compared to MGCP, only a few commands are configured on the IOS gateway and more complex configurations are put on the Call Manager. In MGCP, the Call Manager handles all the dial plans and call routing. In H.323 the gateway does not depend on the call processing agent for call routing, but has its own dial plan. Dial plans on the H.323 gateway are configured using CLI commands on the Cisco 2801.

In addition to the basic configurations on the IOS gateway, it is necessary to configure one of the interfaces on the router to be a gateway interface. [13] In the implementation, the FastEthernet0/1 interface was configured as the gateway interface. This was done by specifying the "h323-gateway voip interface" under the interface.

```
!
interface FastEthernet0/1
 description VOICE VLAN
 ip address 192.168.2.253 255.255.255.0
 duplex auto
 speed auto
 h323-gateway voip interface
 h323-gateway voip bind srcaddr 192.168.2.253
!
```

The next statement "h323-gateway voip bind srcaddr" was a statement used to set the source address as the IP address of the gateway. This was the address to be included as the source address for outgoing H.323 traffic, which included H.225, H.245, and RAS messages. [13]

```
!
voice-port 0/1/0
 description FXO PORT FOR CALLS IN/OUT
 echo-cancel coverage 32
 timing hookflash-out 500
 connection plar 1011
 station-id name FXO-PORT
 station-id number 8648060
!
```

These configuration statements were applied on the FXO Port where calls go to and from the PSTN. The "connection plar" statement designates which IP directory number an incoming call is to be forwarded to. If the PLAR statement was not included in the voice port configuration, the gateway would not know where to direct an incoming call. The echo cancel statement adjusts the echo canceller by 32 milliseconds which are defaults. [12] The timing hook-flash is the duration of the hook-flash in milliseconds. [12] The station-id statements were used for caller-ID purposes. When a call comes in from the PSTN, FXO ports cannot translate the PBX caller ID from the outside. So when a call is coming from the outside, we configure the station-id so that we know that the call is coming from the FXO port.

```
!
dial-peer voice 1 pots
 description FORWARDS TO FXO FOR OUTSIDE CALLS
 destination-pattern 8[2-9]......
 port 0/1/0
 forward-digits all
!
dial-peer voice 2 pots
 description FORWARDS TO FXO FOR KU CALLS
 destination-pattern 4[1-9]...
 port 0/1/0
 forward-digits all
!
!
dial-peer voice 1000 voip
 description DIAL PPER FOR VOIP CALL TO CCM
 destination-pattern 1...
 session target ipv4:192.168.2.1
!
```

The above configurations were the dial peers that the gateway needed to route calls. Dial peers were created for the same patterns that were used earlier in the

MGCP gateway configuration with the difference that it was configured on the gateway in the case of H.323. The same destination patterns are used in MGCP. The patterns used for outside calling are: 8xxxxxx denoted as ‘8........’ This call would be sent out the port 0/1/0 which was the FXO port. All digits were forwarded in this case because of the “forward digits 8” command. However, the dial peer 1000 was a little different than the other dial peers configured on the router. This dial peer was configured for all the other IP Phones that make calls to each other. In this case, the test phones were assigned directory numbers 1010 and 1011. So this dial peer translated any call going to a 1xxx pattern and forwarded it to the call manager. If there are other directory phone numbers in other formats, we would have to configure dial peers for those number patterns.

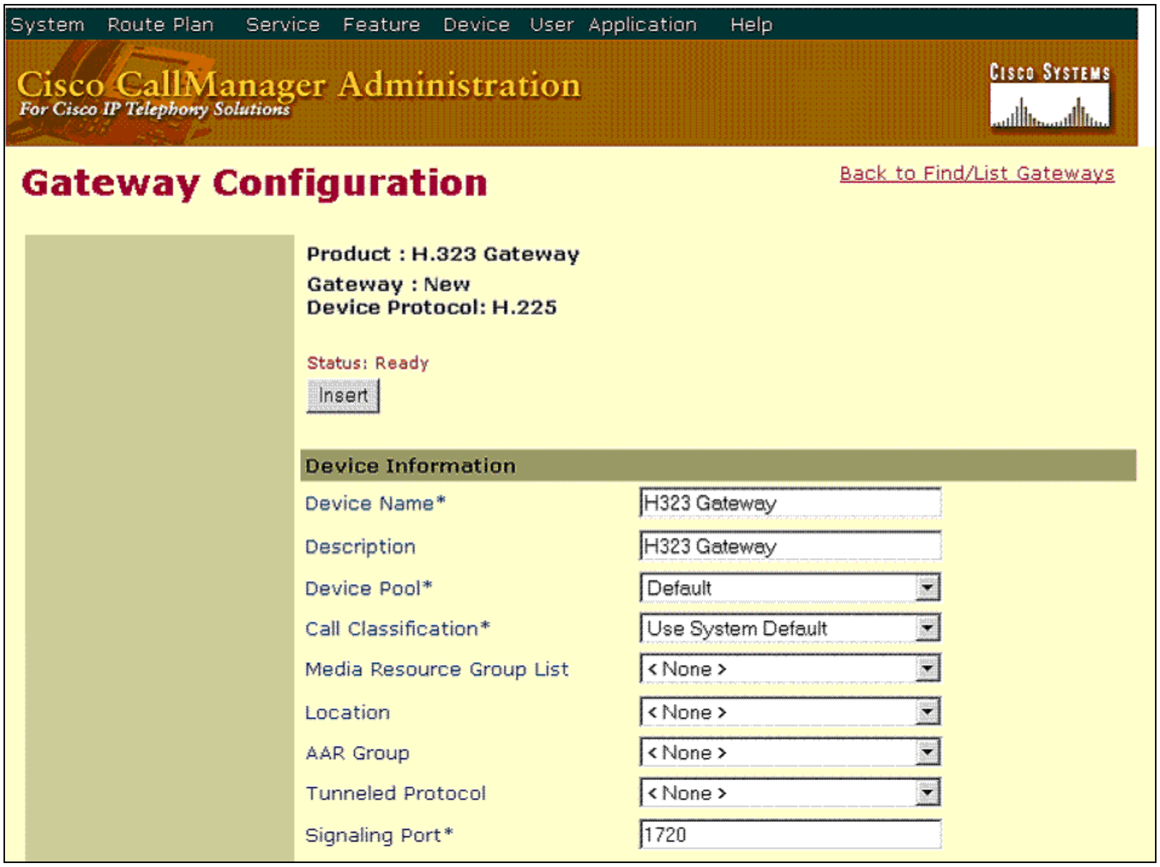
4.3.4 H.323 Configuration on Call Manager

Although the H.323 configuration on the Call manager was supposed to be less intensive since H.323 is configured on the router, there still had to be corresponding configurations on the call manager for the gateways. Only difference between the two protocols’ configuration was that in MGCP there was more work actually done by the call processor, where as in H.323 the work (e.g. dial peers for dial plans) is done on the gateway. The process of inserting a new gateway into call manager was almost the same with both protocols.

Figure 4-3-6 Adding a new H.323 Gateway

Select H.323 Gateway, but not the model of the router as we did in the configuration for MGCP. There was a special gateway type listed within Call Manager. Select "H.323 Gateway" for the type and H.225 was the only choice for the device protocol. Click on next screen takes you to the H.323 gateway configuration screen which is shown in figure 4-3-7.

Figure 4-3-7 Figure for Inserting H.323 Gateway

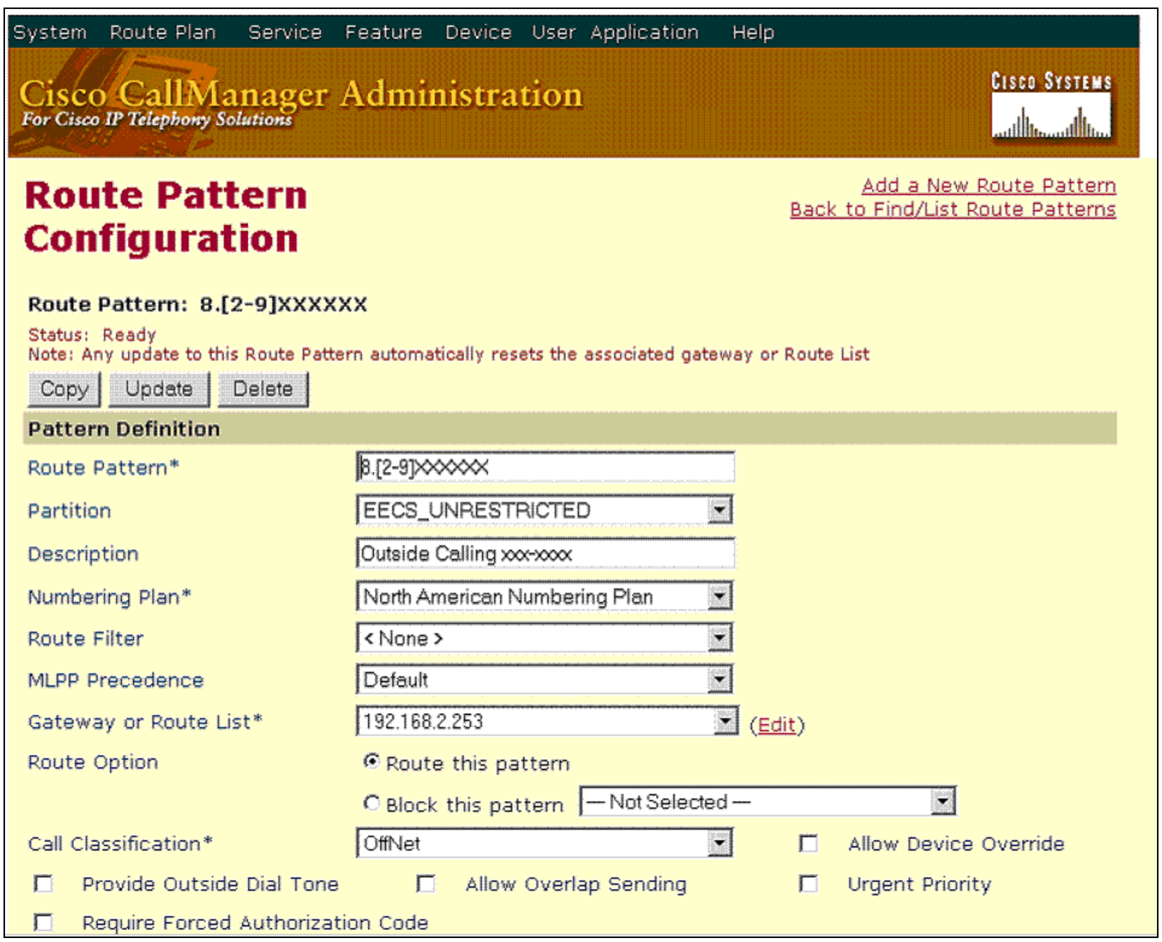


The H.323 gateway configuration was very similar to how MGCP was configured, but as seen in the figure 4-3-8, the gateway name for H.323 would be the actual IP address of the interface that had the "h323 voip bind" statement, which was 192.168.2.253. The device name in the case of H.323 was not as important as the device name in MGCP. MGCP needed a precise device and domain name configured since the Call Manager had to communicate directly with an FXO port. In H.323, the Call Manager did not need such information, so an IP address was sufficient for the communication.

Figure 4-3-8 Gateway configuration for H.323

Figure 4-3-8 shows the H.323 gateway was registered. Although the H.323 gateway "Registration" showed "Unknown", the IP address is displayed on the call manager. The gateway had actually registered in the Call Manager. [13] With the gateway registered with the Call Manager, route patterns need to be configured in the Call Manager. Although dial plans were configured on the H.323 gateway, the Call Manager still needed to know where to forward calls in case it got a sequence of digits needed to match a pattern. Figure 4-3-9 is an example of a route pattern configured on the Call Manager for H.323.

Figure 4-3-9 Route Pattern Configuration for H.323



This pattern was the same route pattern configured on the gateway (8XXXXXX). This route pattern was basically a copy of MGCP route pattern configuration, the only difference being the "gateway" tab on the page. The route pattern points to the IP address of the gateway as opposed to the voice port configuration in MGCP. The call manager when accounted with a dialing pattern for outside calling, it would simply forward the call to the IP address of the gateway.

5 RESULTS AND VERIFICATION

In this section of the report, relevant outputs of show commands from the gateways, outputs from the call processor, registration information and live outputs from the gateways when a call is in progress will be discussed. This will be done for both protocols implemented. Outputs from a dialed number analyzer that provides information about dial patterns for calls blocked or routed will also be included for verification of the dialing plans.

<u>Gateway Show Commands:</u>

The following were relevant outputs from various show commands to depict the correct configuration of the gateway.

```
VoiceRouter#sh mgcp endpoint
aaln/S0/SU1/0@VoiceRouter


VoiceRouter#
```

As per Cisco Systems "An MGCP endpoint is simply any of the voice ports on the designated gateway." [14] In this case, the only voice port configured on the gateway and the call manager was port 0 of the FXO Interface card. This was the only active line out to the PSTN. So a “sh mgcp endpoint” showed only one endpoint. On the router, this port was physically “voice-port0/1/0”, where as the Call Manager recognized it as Slot0/SubUnit1/Port0.

```
VoiceRouter#sh ccm-manager
MGCP Domain Name: VoiceRouter.ku.edu
Priority       Status                Host
============================================================
Primary        Registered            192.168.2.1
First Backup    None
Second Backup   None

Current active Call Manager:    192.168.2.1
Backhaul/Redundant link port:   2428
Failover Interval:             30 seconds
Keepalive Interval:             15 seconds
Last keepalive sent:            23:46:05 UTC Apr 8 2006 (elapsed time: 00:00:09)
Last MGCP traffic time:         23:46:05 UTC Apr 8 2006 (elapsed time: 00:00:09)
Last failover time:             None
Last switchback time:           None
Switchback mode:                Graceful
```

```
MGCP Fallback mode:            Not Selected

Last MGCP Fallback start time:  None

Last MGCP Fallback end time:    None

MGCP Download Tones:           Disabled


Configuration Auto-Download Information

=======================================

Current version-id: {C0A21098-4502-475B-A093-C0EE3704082F}

Last config-downloaded:00:00:00

Current state: Waiting for commands

Configuration Download statistics:

       Download Attempted            : 3

         Download Successful         : 3

         Download Failed             : 0

       Configuration Attempted       : 1

         Configuration Successful    : 1

         Configuration Failed(Parsing): 0

         Configuration Failed(config) : 0

Last config download command: New Registration

Configuration Error History:


FAX mode: cisco

VoiceRouter#
```

This command showed us the Cisco Call Manager (ccm) active in the MGCP configuration. It shows the MGCP domain name that was configured on the Call Manager and also showed the gateway was in registered status. This command provided information about a configuration download from the call manager. This was most likely to be a configuration for a phone or to the gateway that was downloaded from the call manager.

H.225 is a part of the H.323 protocol suite that is responsible for the call control signaling to set up connections between H.323 endpoints. The exchange of H.225 messages is shown in the output below. RAS messages could be displayed by using the “sh h323 gateway” command, but I used only H.225 messages to depict the setup and signaling messages.

In the following code section, “setup”, “set up confirm” and other H.225 messages can be seen. We see the sent and received messages for the setup of H.224 protocol. These were test calls from the PSTN into the gateway and vise versa. This shows the execution of the H.323 gateway when the H.225 messages are exchanged.

```
VoiceRouter#sh h323 gateway h225
H.323 STATISTICS AT 5d23h
 H.225 REQUESTS       SENT    RECEIVED     FAILED
 Setup                     8        2          0
 Setup confirm             2        3          0
 Alert                     0        4          0
 Progress                  2        0          0
 Call proceeding           2        4          0
 Notify                    2        12          0
 Info                      0        0          0
 User Info                 0        5          0
 Facility                  0        0          0
 Release                   8        7          0
 Reject                    0        0          0
 H225 establish timeout    0
 RAS failed                0
 H245 failed               0
H323 gateway internal event queue:
  Events queued 0, free 4, held 4, allocated 4 times, exceed max 0 times
  Max serviced 4, Max serviced exceed 0
  no dest 0, no buffers 0, call flushes 0
CRV statistics
 In use: 0
 Max collisions obtaining crv: 0
VoiceRouter#
```

The next code block was an output for a H.323 call, where only selected output has been displayed. I chose to delete out some of the irrelevant output to make the output seem more understandable. The following was an output of the 'show voice call active' with the H.323 configuration on the gateway.

```
VoiceRouter# sh call active voice
Telephony call-legs: 1
H323 call-legs: 1
Total call-legs: 2

GENERIC:
PeerAddress=1010
PeerId=1000
InfoType=speech
TransmitPackets=307
TransmitBytes=47567
ReceivePackets=2326
ReceiveBytes=372160
VOIP:
RemoteIPAddress=192.168.2.1
RemoteUDPPort=18758
RemoteSignallingIPAddress=192.168.2.1
RemoteSignallingPort=54715
RemoteMediaIPAddress=192.168.2.22
RemoteMediaPort=18758
tx_DtmfRelay=h245-alphanumeric

AnnexE=FALSE
Separate H245 Connection=TRUE
H245 Tunneling=FALSE

SessionProtocol=cisco
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-through
CallerIDBlocked=False
OriginalCallingNumber=1010
OriginalCallingOctet=0x0
OriginalCalledNumber=88653816
TranslatedCallingNumber=1010
TranslatedCalledNumber=88653816
GwReceivedCalledNumber=88653816
GwReceivedCallingNumber=1010
MediaInactiveDetected=no
```

```
DSPIdentifier=0/1:1
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 2

VoiceRouter#
```

The above output showed the active voice calls and also some of the parameters that went into the VOIP call. It specifies the protocols being used and the total number of calls at that particular time.

```
VoiceRouter#sh voice port summary
                              IN       OUT
PORT      CH   SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
========= == ============ ===== ==== ======== ======== ==
0/0/0     --  fxs-ls      up    dorm on-hook  idle     y
0/0/1     --  fxs-ls      up    dorm on-hook  idle     y
0/0/2     --  fxs-ls      up    dorm on-hook  idle     y
0/0/3     --  fxs-ls      up    dorm on-hook  idle     y
0/1/0     --  fxo-ls      up    up   idle     off-hook y
0/1/1     --  fxo-ls      up    dorm idle     on-hook  y
0/1/2     --  fxo-ls      up    dorm idle     on-hook  y
0/1/3     --  fxo-ls      up    dorm idle     on-hook  y

VoiceRouter#
```

The above output was from when an actual active call over the gateways was in progress. These commands are independent of the protocols and only show the state of the port or the active call. In the "sh voice port summary", it shows the status of each of the ports, the signaling type and the admin state. From this output, the only port that matters was port 0/1/0 which is the only FXO port that was in use. While on a call, the out status should show up as off-hook.

Dialed Number Analyzer Outputs

This section contains a few screen shots of the Dialed Number analyzer. These screenshots account for successful calls from IP phone to IP Phone, PSTN to IP Phone, IP Phone to PSTN with both MGCP and H.323.

Figure 5-1 Incoming Call from PSTN

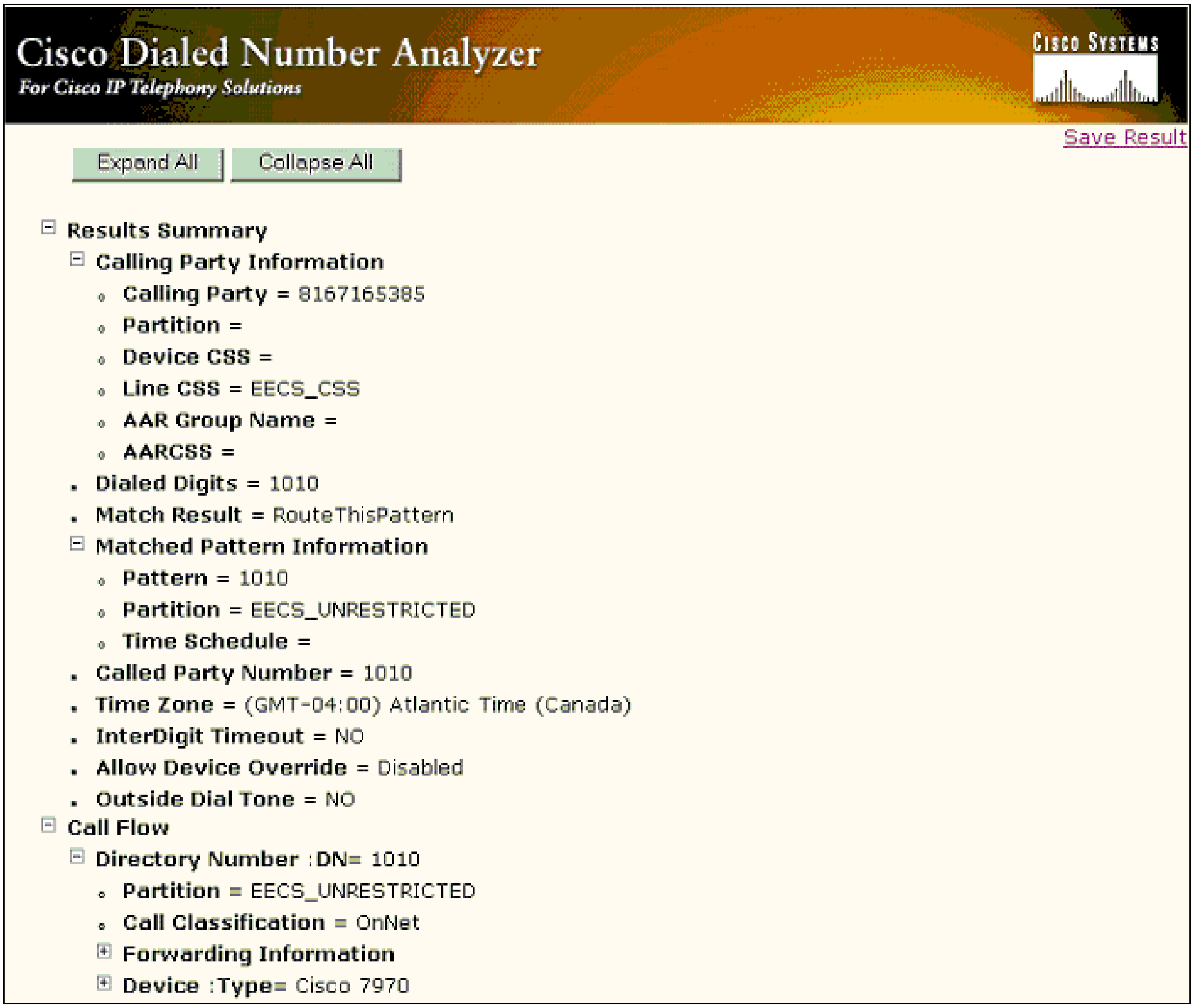


Figure 5-1 shows a successful call that came in from the PSTN with the MGCP configuration. We can deduce this information because the incoming call gets directed to extension 1010, which was configured in the Call Manager under the MGCP gateway configuration. This configuration was such that if any call came into the gateway from the PSTN, the call would automatically be forwarded to DN-1010.

Figure 5-2 Calling from IP Phone to Lawrence Number

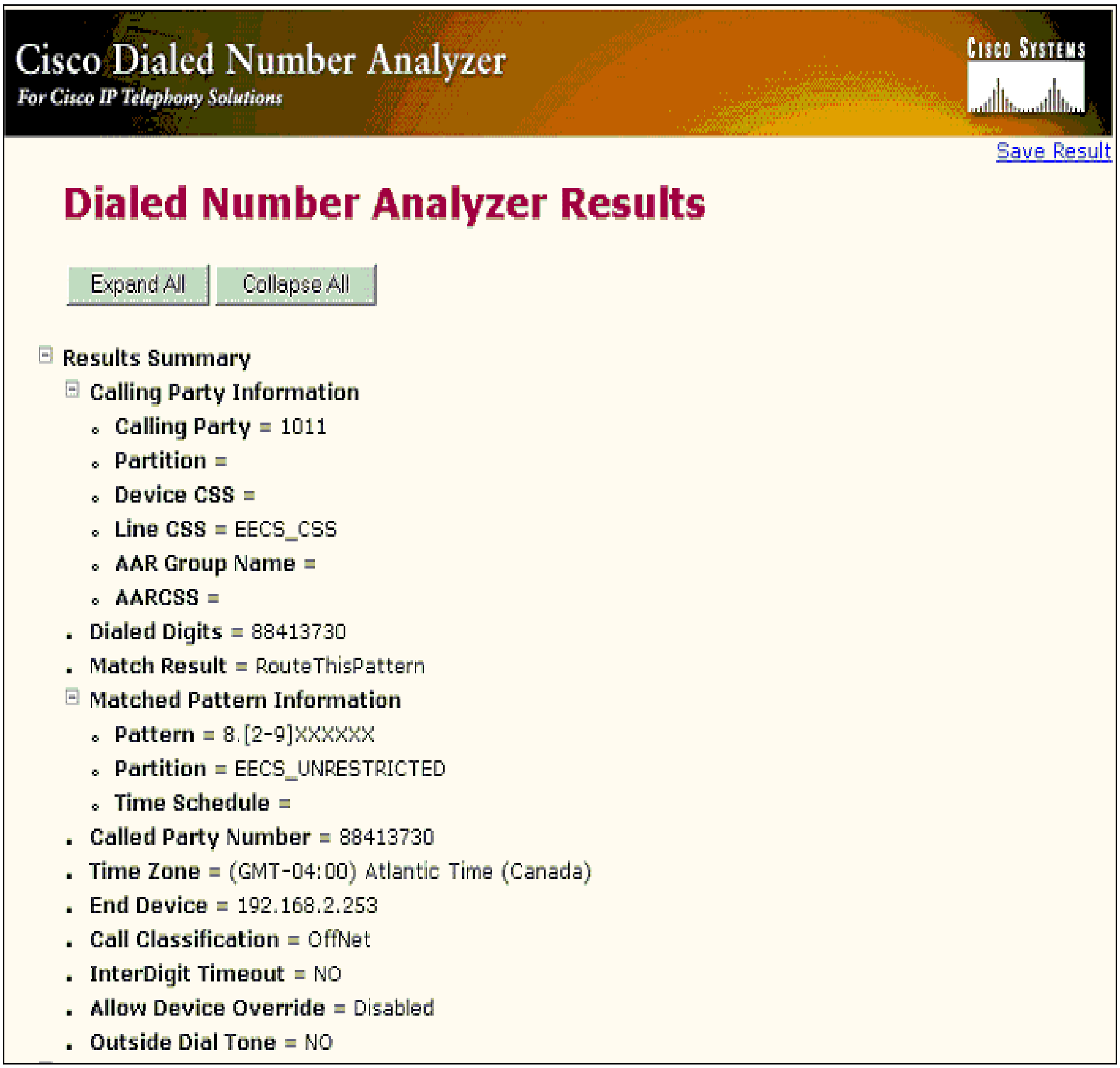


Figure 5-2 depicts a call from an IP phone with DN 1011 to a number dialed as "88413730". These dialed digits matched a route pattern in the Call Manager and was forwarded to 192.168.2.253. This shows that the call was made when the system was configured with H.323. So as long as a pattern was matched in the Call Manager in H.323, it would directly send the call over to the gateway to handle. This again was a successful call from an IP phone to the PSTN in H.323.

Figure 5-3 IP Phone to KU PBX Network

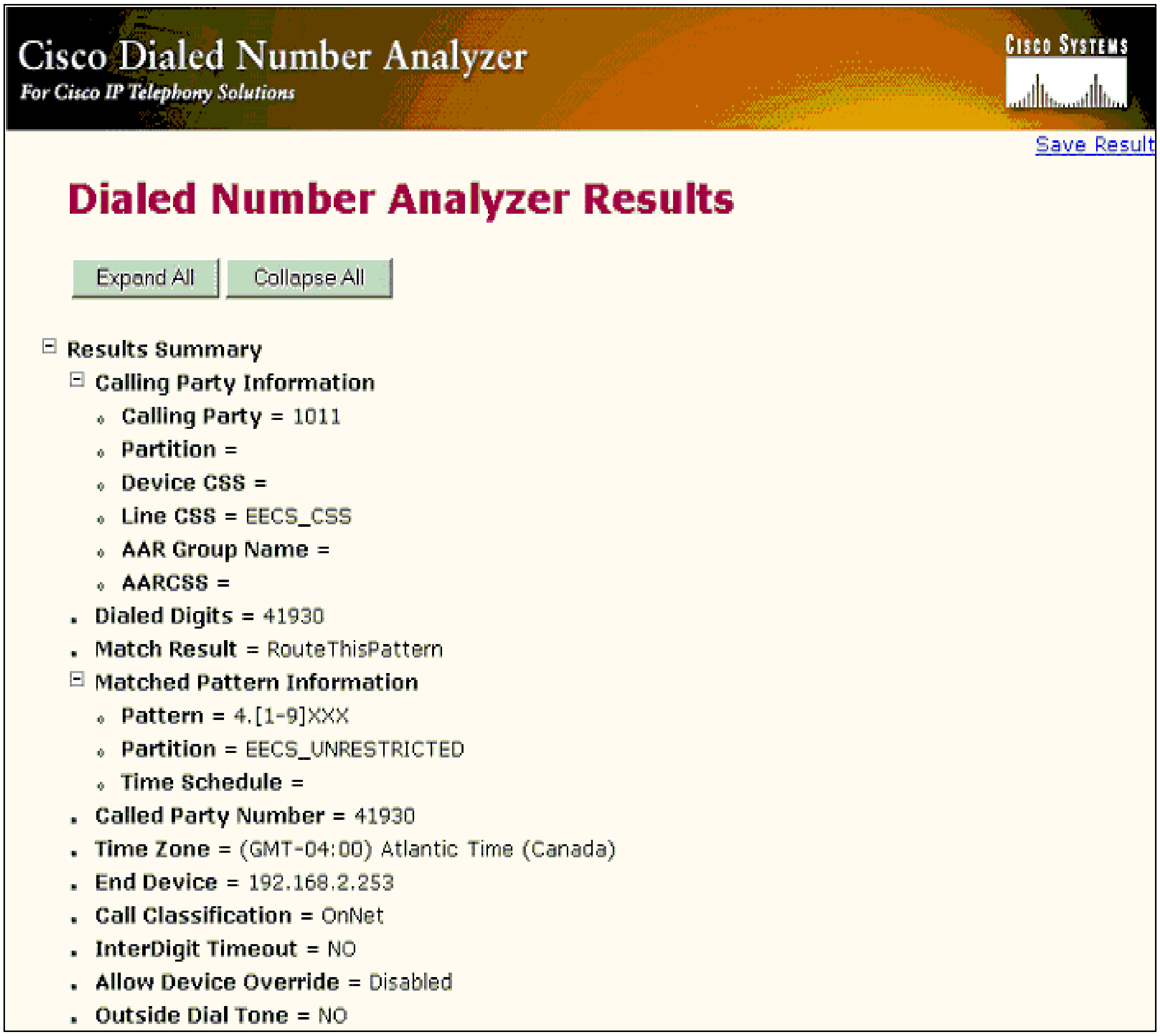


Figure 5-3 is the exact same scenario as the figure 4-2, with only difference being the route pattern matched. This call matched the other outside calling pattern which was set up to call only KU Numbers. This was another successful call using H.323 for KU network calling.

Figure 5-4 IP to Lawrence local calling over MGCP

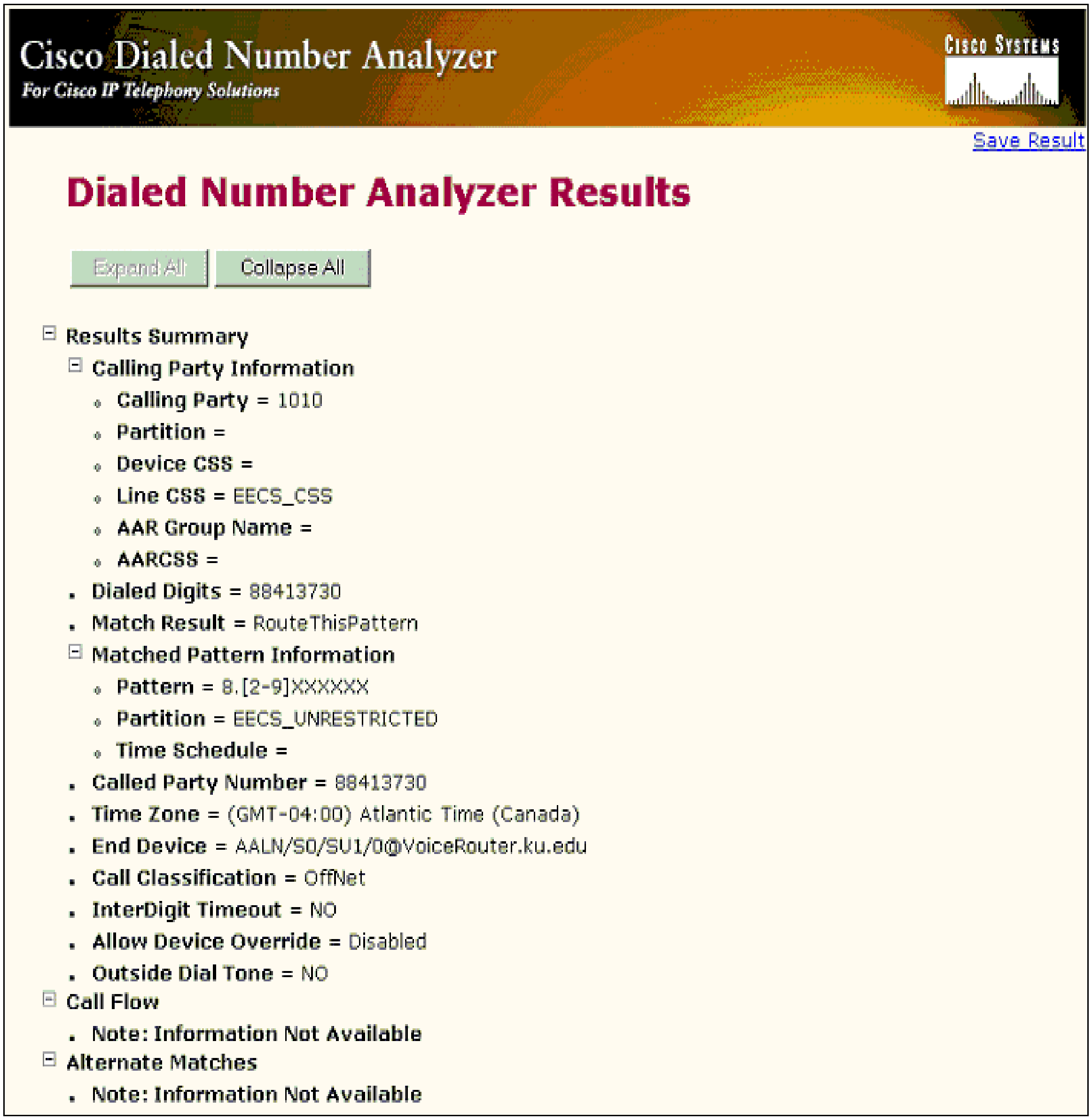
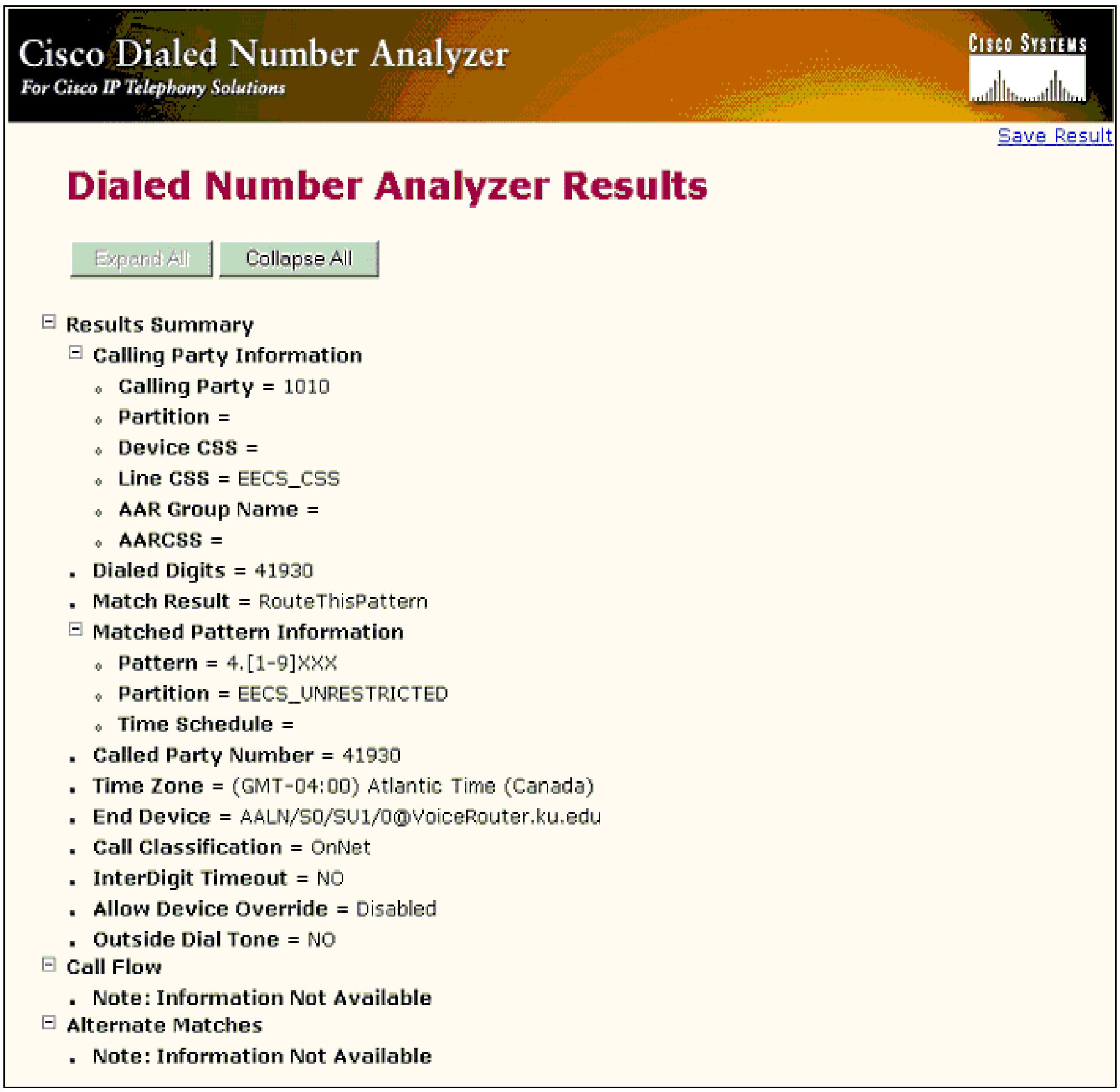


Figure 5-4 is a dialed number output for a call from an IP phone to a local Lawrence number, but only in an MGCP scenario. When the route pattern is matched, it is forwarded to the following MGCP end point –

AALN/S0/SU1/0@VoiceRouter.ku.edu.

This was the same case with figure 5-5 where the protocol used was MGCP.

Figure 5-5 IP to KU network over MGCP



In figure 5-5 the call was forwarded to the MGCP endpoint after a match was detected on the route pattern of 4[1-9] XXXX. The route pattern was for calling an internal KU number. This again was a successful MGCP call from the lab to KU.

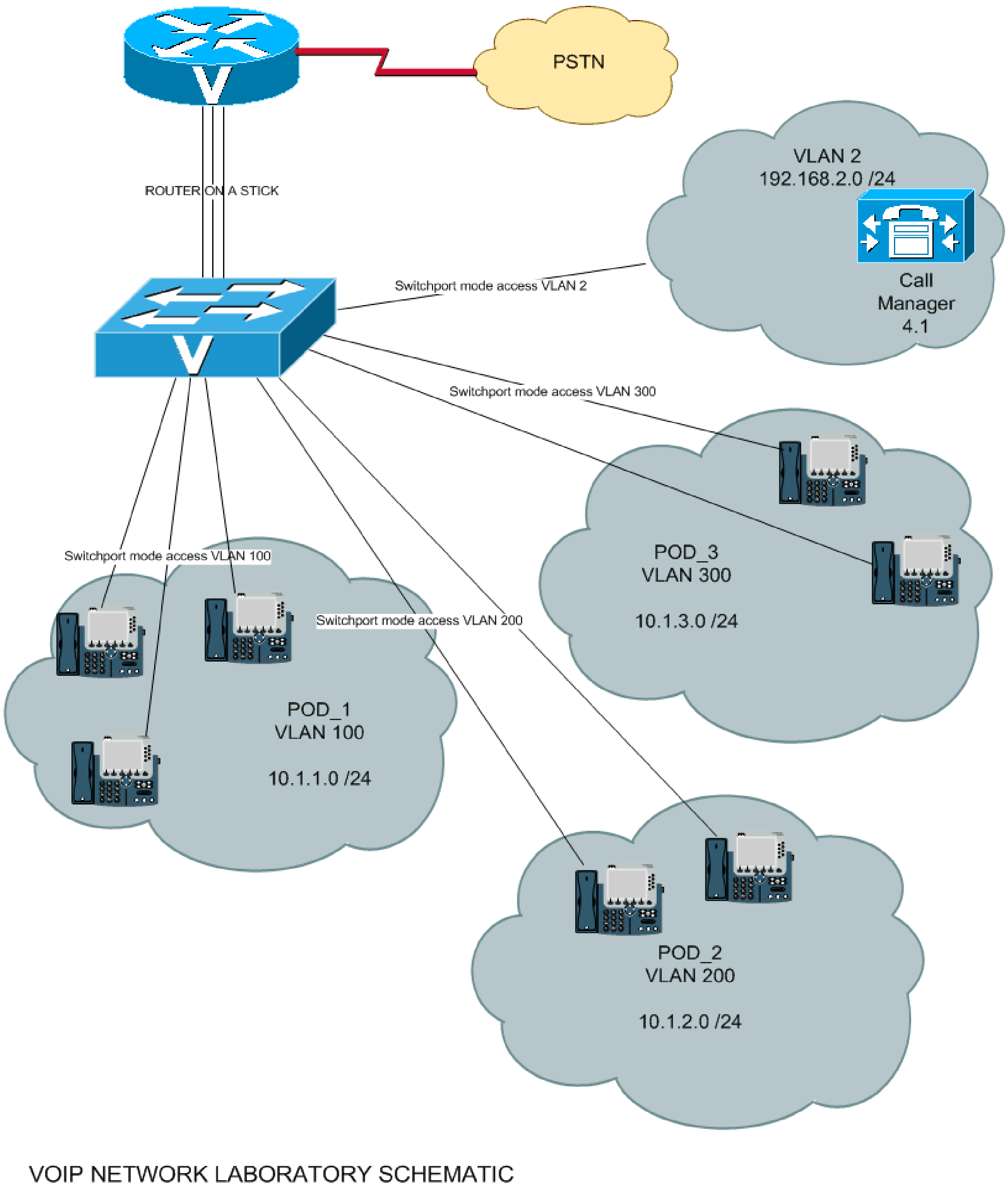
6 LABORATORY ACTIVITY FOR STUDENTS

The following is a lab that was devised as an exercise for students in EECS-745. With this lab, students will get good exposure to configuration tasks that range from the switch configuration to complex Call Manager configurations including partitions, calling search spaces, and route patterns.

Figure 6-1 is a schematic that students can refer to for their lab exercise. In this figure, the voice enabled router was responsible for inter-VLAN routing so that elements in the different VLAN's can access each other. The connection between the router and the switch was labeled as a "router on a stick". This means that there are sub-interfaces configured on the router for each VLAN. Each of these sub-interfaces would be configured with an IP address which acts as a default gateway for each of the VLAN's. The appropriate switch port configuration also has been labeled for each group so that their phones can be added to the appropriate VLAN.

Each of the groups would be broken down into PODS and would get a couple of switch ports to work with. Initially, each POD needs to be set up only to call within the POD and will then work their way into restricting and allowing Inter-POD calling.

Figure 6-1 *Networking Lab Schematic for lab exercise*



SCENARIO:

Each POD is its own VOICE VLAN.

Each member in a POD gets one switch port and one IP Phone

POD_1 - VLAN 100 - 10.1.1.0/24

POD_2 - VLAN 200 - 10.1.2.0/24

POD_3 - VLAN 300 - 10.1.3.0/24

POD_4 - VLAN 400 - 10.1.4.0/24

Call Manager - VLAN 2 - 192.168.2.0/24

OBJECTIVES:

- Understand how to add switch ports (phones) to Voice VLANS
- Understand how to create DHCP Pools for Phone DHCP Addresses
- Get familiar with Call Manager Configurations
- Understand Partitions and Calling Search Spaces
- Understand how calls can be restricted and allowed using Partitions and Calling Search Spaces.
- Understanding Dial Plans and PSTN calling

Each POD configures the following:

- Switch Ports for each POD member
- One DHCP Pool on the router per POD
- One partition and One Calling Search Space per POD
- Dial Plans

Once each member of a POD can call only other members of the same POD then modify calling search space to allow Inter-POD calling.

CONFIGURATION TASKS:

ON SWITCH AND ROUTER

(Replace all “X” with group number)

- Configure switch port to allow VLAN X00 to the phones
- Configure a DHCP Pool on the router called POD_X_DHCP_POOL to give out addresses in the range 10.1.x.20-10.1.x.30
- Plug in the phones and check on phone to see if the phone get IP Address. Ping phone from router.

CALL MANAGER CONFIGURATION:

Log into the CCMAdmin Page.

connect to https://192.168.2.1/ccmadmin

uname: ccmadministrator

passwd: ciscovoice

Create Partition:

1. Go to RoutePlan>Class Of Control>Partition
2. Add New Partition - Enter <<POD_X_PT>><<Description>>
3. Click Insert

Create Calling search Space:

1. RoutePlan > Class of Control > Calling Search Space
2. Name : POD_X_CSS
3. ADD POD_X_PT into "Selected Partitions"
4. Click Insert

Add phones to the Call Manager

1. Device > Phone > Add a new Phone
2. Select the Phone type
3. Enter the MAC Address of the phone (From back of the phone)
4. Enter Description of phone < POD_X_PHONE_1
5. Select Device Pool as Default
6. Select Calling Search Space as POD_X_CSS
7. Select the Default Phone Template (Eg. For Cisco 7980, Select Default 7980 Phone Template)
8. All the other options are left at defaults
9. Click Insert

a. Click on "LINE 1 - Add a new DN" for Line options
b. Enter Directory number as X001 where X is POD/GROUP Number
c. Select Partition as POD_X_PT
d. Select Calling Search Space as POD_X_CSS
e. Click update, and Reset device
f. Go back to main phone configuration phone and Reset phone
g. Phone should reboot and come up with the DN we just configured.

DO THIS CONFIGURATION OVER FOR ALL PHONES IN ONE POD.

Create Dial Plans (Route Patterns)

1. Route Plan > Route/Hunt > Route Pattern
2. Click on Add a new Pattern
3. Each POD creates 2 different patterns :

POD_X_KU_CALLS and POD_X_PSTN_CALLS

4. For the POD_X_KU_CALLS, the pattern is 4[1-9]XXX
5. For POD_X_CALLS, the pattern is 8[1-9]XXXXXX
6. Select POD_X_PT for the partition in each of these patter configurations
7. For the gateway, make sure 192.168.2.1 for H.323 or the VoiceGateway for MGCP
8. Click Insert
9. Reset device and test calls.

VERIFICATION:

AT THIS POINT, POD_X CAN CALL PHONES ON THEIR OWN PODS, BUT NOT PHONES ON OTHER PODS.

CONFIGURATION FOR INTER-POD / INTER-GROUP CALLING.

RESTRICTING AND ALLOWING INTER-POD CALLS.

EXAMPLE:

FOR POD_1 (DN - 1001) to be able to call POD_2 (DN -2001) POD_1_CSS should include the POD_2_PT

1. Go to Route Plan > Class Of Control > Calling Search Space
2. Click Find, click on POD_1_CSS
3. Locate POD_2_PT from Available partitions and Put in selected Partitions.
4. Reset Devices
5. TEST Calling from POD_1 Phone to POD_2 Phones.

DOCUMENTING AND VERIFICATIONS

- Screen shots of registration from the Call Manager.
- Verification by show commands on the routers and the switches.
- Dialed Number Analyzer output

Dialied Number Analyser:

Access to DNA:

https://192.168.2.1/dna

uname: Administrator

passwd: ciscovoice

The Dialed Number analyzer shows what the Call Manager does with a particular phone call. This output can be saved in XML format and shows the Calling Party DN, PT and CSS and also the Called Party DN, PT and CSS. Screenshots of this could be used as documentation, or even the XML outputs could be saved by each POD.

INCLUDE DNA results for the following:

1. Call from POD_1 to POD_1
2. Call from POD_1 to POD_2
3. Call from POD_2 to POD_3
4. Call from POD_X to KU PBX number
5. Call from POD_X to Lawrence Number

7 FUTURE WORK AND CONCLUSION

The implementation of AVVID for IP Telephony involved the core setup of network infrastructure which can be used as a platform for more application rich IP Telephony. There are endless possibilities that can be implemented on the current set up. These applications can be implemented and tested with both an MGCP gateway and with a H.323 gateway. Application rich features that make IP Telephony such a multifaceted and interactive technology can be configured on the Call Manager.

In the report, various industry standard deployment models for Cisco's IP Telephony have been discussed. Based on the hardware resources in the lab, a successful implementation of the IP telephony with both H.323 and MGCP was possible at the EECS Networking Laboratory.

After the implementation of the IP Telephony lab, I was involved in devising exercises that will be used to introduce students to IP Telephony in the EECS-745 "High Performance Integrated Networks" laboratory. These exercises will expose students to some details of how the Call Manager and the gateways are configured and is a great introduction to IP Telephony to EECS students.

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