UNIVERSITY OF NAIROBI

SCHOOL OF COMPUTING AND INFORMATICS

A MODEL OF AN INTEGRATED UNIFIED COMMUNICATION NETWORK USING PUBLIC SWITCHED TELEPHONE NETWORK GATEWAYS AND CISCO UNIFIED COMMUNICATION MANAGER SERVER

BY

OCHOLA J. OMONDI GODFREY

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SUPERVISOR: Dr. Abade Elisha

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DECLARATION

This project, as presented in this report, is my original work and has not been presented for any other institutional award.

Signature:……………………………… **Date:………………………**

Ochola J. Omondi Godfrey

This project has been submitted in partial fulfillment of the requirements of the degree of Master of Science in Information Technology Management with my approval as the University Supervisor.

Signature:……………………………… **Date:………………………**

Dr. Elisha Abade

School of Computing and Informatics

DEDICATION

To the love and support in my life, Lydia.

ACKNOWLEDGEMENT

Thanks to my family, friends and colleagues for their support. In particular much thanks to Dr. Elisha Abade – my Supervisor, the panelists and fellow researchers and the Cisco team, who are exceptional at what they did and prepared this thesis achievable.

ABSTRACT

Communication requirements are constantly evolving in recent times demanding flexibility and response mechanism in controlling deployments need for everyday connections. It is a necessity in business today and a well designed unified communication networks facilitate smooth flow of information. An effective and efficient communication network gives an enterprise competitive advantage and reduces operational costs. In my study area, we have in place installed PSTN – PABX and VOIP – CUCM that run in parallel and investment complete in both set ups. It's therefore very necessary for integration so that we have a unified communication network.

This project identifies the challenges of having a parallel VoIP and PSTN systems, proposes a solution on integration of MGCP, H.323 and SIP gateways with CUCM and designs a model to determine its applicability.

Unified communication will be used to transmit voice services in the internet environment and legacy closed Public Switched Telephone Networks. My main discussion is about UC, Gateways design issues, how to measure efficiency, reliability, latency and packet mismatch from the transmitter and receiving end and also delay and Jitter, and determination of measurements using delays within the Internet Protocol - SLA (Service level agreement).

I have looked at telephone systems namely legacy Public-Switched Telephone Network
and Internet Protocol telephony that are extensively examined to come up with Unified Communication by measuring of our University of Nairobi voice networks/services, this is achieved various models and real experiments. Resulting solutions show that the unified communication is more efficient in terms of performance in the current technology.

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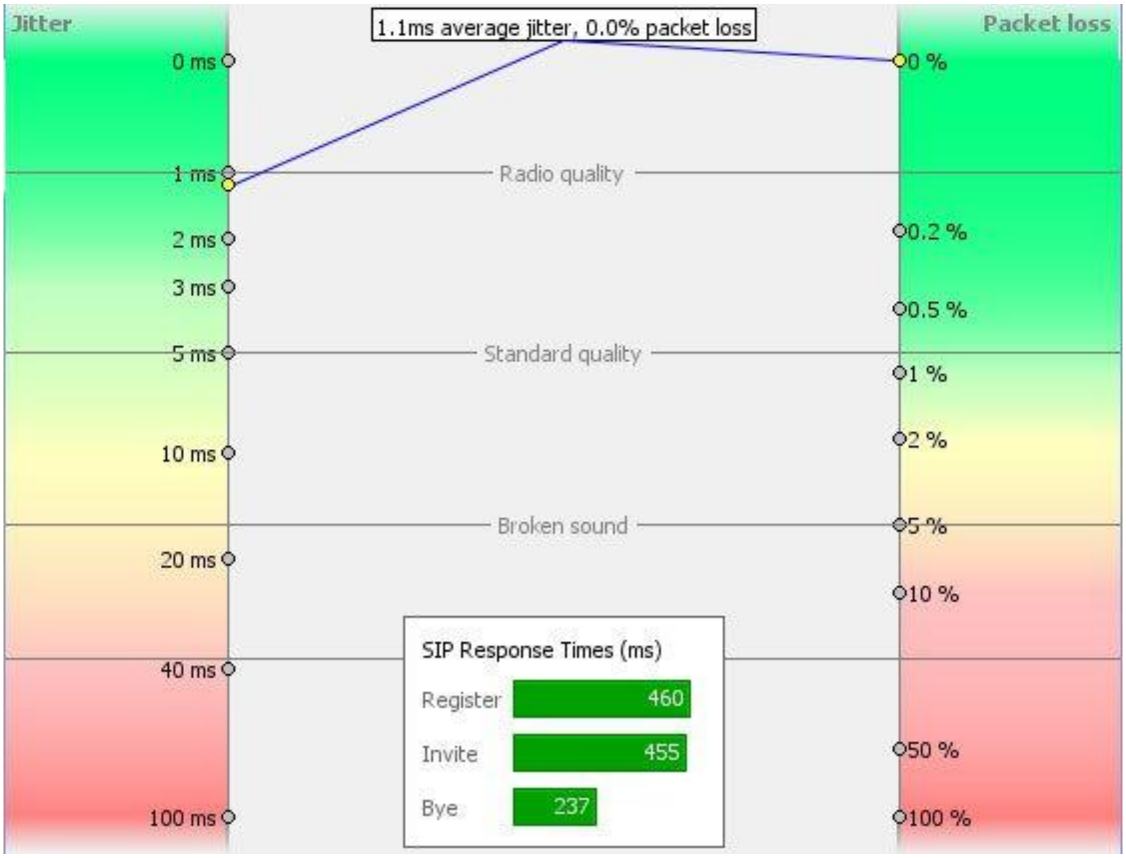
Jitter
1.1ms average jitter, 0.0% packet loss
Packet loss
0 ms
1 ms
2 ms
3 ms
5 ms
10 ms
20 ms
40 ms
100 ms
Radio quality
Standard quality
Broken sound
0 %
0.2 %
0.5 %
1 %
2 %
5 %
10 %
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CHAPTER ONE

INTRODUCTION

1.1 Background to Research problem

Main objective of this study is to analyze and design an integrated unified communication network using Public Switched Telephone Networks (PSTN) gateways and Cisco Unified Communication Manager (CUCM) server.

Communication requirements are constantly evolving in recent times demanding flexibility and response mechanism in controlling deployments need for everyday connections. It is a necessity in business today and a well designed unified communication networks facilitate smooth flow of information. An effective and efficient communication network gives an enterprise competitive advantage and reduces operational costs.

To set outside calls, required is a link to the public network. The Public network is throughout long-established time-division multiplexing telephony interfaces and Voice over IP domain; incorporated using access to Voice over IP label legs.

Unified communication systems/networks involve the convergence of voice and data networks to drive radical changes in the development and delivery of products for organizations. Unified communication will involve blending of PSTN, VoIP, converged IP services including PC-based distance learning solutions, video conferencing and streaming, security surveillance, data and unified messaging amongst others.

PSTN connect legacy telephone systems through Telkom (K) telephone exchanges such as plain old telephone service (POTS) and ISDN. VoIP is a technology whereby telephone signals are digitized via dedicated circuits and transmitted as packets; and uses CUCM server as IP PABX system.

The research develops and validates a model of implementing PSTN gateways in CUCM in the University of Nairobi; and unification can greatly improve the performance of communication network fields.

UoN provides communication solutions in all its campuses with the widest range of voice and data services; with Telkom (K) ltd being the sole provider of wired and wireless services and Cisco for the IP telephony. It uses CUCM and PSTN platforms that serve intra-campuses needs. The legacy phones and VoIP networks link offices and telephone exchanges in all campuses.

In the UoN, we have different types of IP phones ranging from Cisco 7970, 7961, 7960 , 7945, 7911, 7821, 7860, 6921, etc. The prices inclusive of power adaptors range from KES 65,000 to 20,000. The phase one and phase two of the VoIP implementation has already consumed at least KES 50million. The IP telephony does not incur any monthly bills/charges except for the provision of internet services; this is because it is run over the internet. Once internet is up, making IP to IP calls are exclusively free of charge; it's only charged when calls are made to other networks.

The PSTN involves use of PABXs and legacy phones. Initial cost involves purchase of PABXs which is configured to provide ports for the CO lines and extension lines. PABX ranges from KES 15,000 to 10million depending on usage and infrastructure. Legacy phones range from KES 1,000 to 20,000. They also incur recurrent monthly bills that must be paid for the said services.

The infrastructure for the PSTN and CUCM in UoN is already in place; and they both run parallel to one another. Therefore need for integrated unified communication networks to move from CAPEX to OPEX.

1.2 Statement of the problem

It aims at development of: "A unified and optimized communication networks"

Translation of diverse types of signaling and media requires access. VoIP phones use a TDM interface to communicate using Voice gateways that are used anytime (for example, the switching networks, legacy PABX, analog telephone, facsimile, surveillance and monitoring etc). IP telephony Real-Time transfer procedure medium packets are converted en route for Time Division Multiple - ADC or DAC signals using the gateway router.

We have in place installed PSTN – PABX and VOIP – CUCM that run in parallel and investment complete in both set ups. It's therefore very necessary for integration so that we have a unified communication network.

1.3 Research Objectives

To develop a comprehensive PSTN in CUCM Unified communication network and to investigate factors influencing adoption of unified communication system in Public organizations in Kenya.

1.4 Specific objectives

i. To investigate and analyze the model to determine its applicability.

ii. To evaluate and research on the gateways types that will interact with CUCM and differences described to come up with unified communication.

iii. To propose a solution on integrating MGCP, H.323 and SIP gateways with CUCM.

1.5 The Study Area

The University of Nairobi is an establishment of higher learning; and has contributed significantly to the growth and improvement of Kenya.

We have in place PABX that has both ISDN and POTS in all the campuses; and a CUCM (Subscriber & Publisher) server installed. The CUCM server is accessed in all campuses through routers and switches. Recent trends in Kenyan telecommunication industry have focused on development of communication systems and applications. Challenge will be in determining implementing the technology properly, including providing for maximum optimal performance and essential security measures enabled.

This research will take place in the ICTC department within Central Administration; the main section of integration being at Data Communication for the services of internal and external communication.

1.6 Significance of the study

The major principal is to propose and validate a process model for integration of Unified Communication Networks using PSTN gateways and CUCM; with the main emphasis of the project being analysis and design of integration.

This study will be beneficial for Communication Officers as it will ensure the following:

i. Conversion of TDM voice to VoIP, DTMF Relay, and vice versa.

ii. Controlling voice connections using routers and allowing controlled dialling configuration by MGCP. Whereby the PRI (T1/E1), which is the backbone of the function that are transparently passing to Cisco Unified Communication Manager using Q.931.

iii. Provision of flexible and easy way to connecting VoIP calls to the PSTN through H.323 gateways.

iv. Call-routing configuration by using SIP trunk configuration.

The ladder and philosophy concerned in originating Voice over IP telephone calls are similar to long-established digital telephony.

Because IP telephony know-how provides bandwidth efficiency and low costs; businesses are migrating from long-established PABX to Voice over IP systems and lastly to Unified Communication.

The unified communication networks will thus allow voice, data, message and conferencing running in a single network; and significantly reducing infrastructure and operational costs.

Key Benefits:

This research work will benefit both the users and the University management as a more and useful unified communication network in the entire University and its satellite campuses.

Gateways will definitely broaden the existence cycle of conventional equipment by "*VoIP enabling*" it. Whereby analog trunk lines are replaced with SIP trunks and routing to traffic subsets over IP telephony to a secluded PABX or access - a method branded as toll bypass.

The Gateways will therefore enhance elasticity, instead of moving from a legacy PABX to a digital PBX in one step. Companies can stage and migrate using bridge between the two systems in the gateway.

Unified Communication Digital format will better be controlled. This is done by compressing it, routing it, converting it to a new better format, and so on; also it is more noise tolerant. Transport Communication Protocol/Internet Protocol - TCP/IP networks are made of packets of IP containing a header (to control communication) and a payload to transport data.

1.7 Document Layout

Chapter 1 gives an introduction to the project including the locale, statement, objectives, significance and methodology. It also gives an overview of the study area and provides a summary of document layout.

Chapter 2 is the Literature review. Key concepts used throughout the document are introduced together with their meaning within the context of the project. A few methodologies proposed by other authors are reviewed.

Chapter 3 outlines the research methodologies. It gives details of how the study area was selected, steps taken in the project, methods used for process application respondents.

Chapter 4 is where the proposed design of the unified communication network tests, results and discussions are made.

Chapter 5 is where the research conclusions are made. Limitations of the research are outlined and based on these; recommendations for further work in the research area are made.

CHAPTER TWO

LITERATURE REVIEW

2.1 Key Concepts

Here, literature related to and consistent with research objective is reviewed. It begins by reviewing overview of PSTN and CUCM; and thereafter presenting the existing literature and models related to the research area.

Telecommunications is a technology that has been around for years and still being adopted by businesses worldwide:

"The Voice over IP telephony services will raise to about 400,000 U.S. households by the end of 2005, and to 12.1 million households by 2009", (Jupiter Research).

"Businesses that comprise already implemented Voice over IP have saved up to 90% on their stretched expanse and intercontinental phone bills" New York Times.

The important goal for this research is provision of knowledge and skills in unified communications, and deploying CUCM and PSTN products as shown below with the following attributed factors: lower costs, greater efficiency, higher reliability, and supporting innovation. The main goal is to enhance communication within the University, (ICT investment at UoN).

"A latest way of system amalgamation that focuses on both interior and exterior applications by means of homogeneous middleware frameworks plus object oriented technology." - Irani, Themistocleous, & Love, (2003)

For the purposes of this project, we adopted the definition advanced by (EAI Industry Consortium, 2004), but added an extra qualification "in a manner that ensures smooth and effective exchange of information among the applications". Therefore:

"*Procedure of integrating numerous applications that were independently urbanized, may use mismatched expertise, and remain independently managed in a manner that ensures smooth and effective exchange of information among the applications*"

2.2 Public Switched Telephone Network (PSTN) Overview

The public switching “contributes to the assortment of unified telephone networks, mutually mercantile and organizations. It uses either the Plain Old Telephone Service (POTS) and/or Integrated Service Digital Networks (ISDN). Resulting to switching of handset networks to wholly digitized networks.

Represents communication system available to the public and private domain to allowing users to be integrated to all contact procedure. Public telephone networks within UoN are benchmark incorporated systems of broadcasting and switching facilities, signalling processors, and associated operations support systems allowing phone call devices to converse with apiece erstwhile when they are operated.

The architecture is as shown in the figure below:

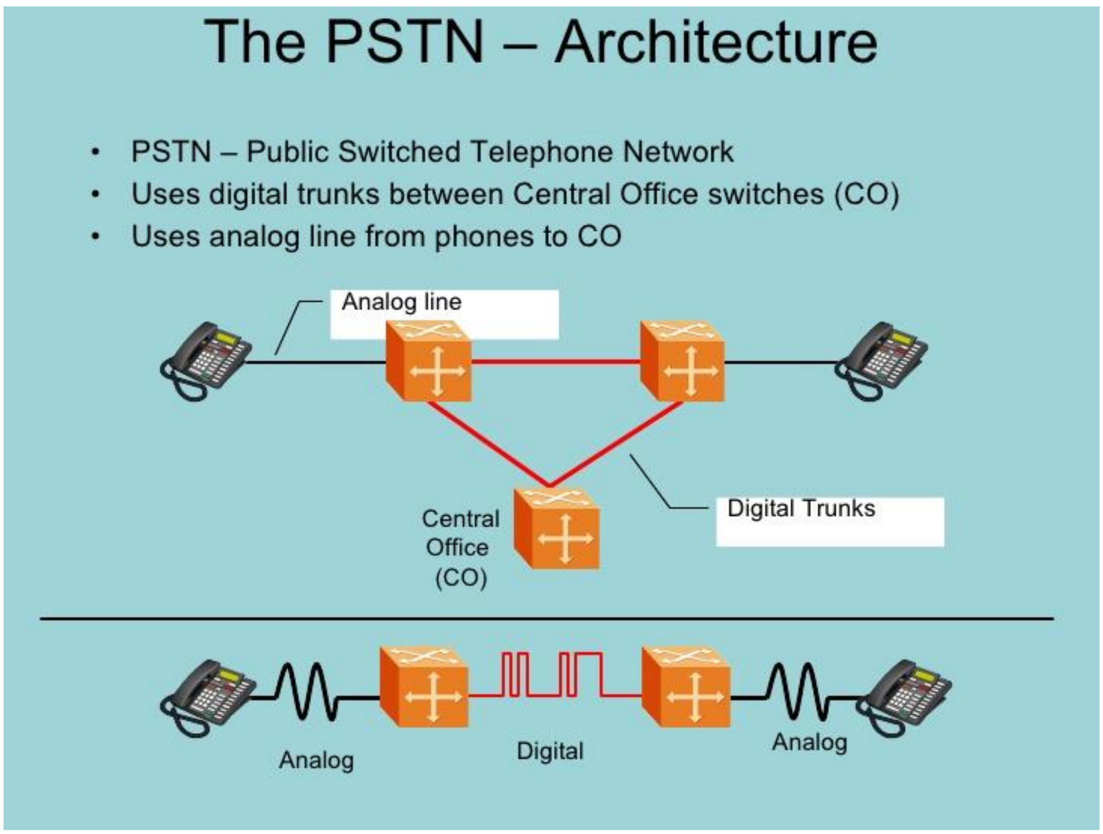


Figure 2-1 PSTN Architecture

In relative to the IP technology, the switching networks act a great deal of the Internet's long-distance transportation. We pay for access/service using established infrastructure and models. Internet users will therefore avoid paying usage tolls to anyone except ISPs.

“Its Architecture describes a variety of apparatus and interconnecting services so as to make available phone facility to the public. At the core of Public Switching are digital switches. The term "switch “basically means cross-connecting a phone line with a lot of other handset lines and switching since one link to another. Switches bounded by the Public Switching transmit control messages to each other, frequently throughout detach control-signalling network called signalling system number 7 and provides reliable communications to its subscribers”.

“The swing from plain tone of voice communication to prosperous content interactions (video and image) over internet, even in terms of the simple voice communication, voice carried by packets over mobile networks and the internet has seen dramatic increase while the voice traffic over conventional PSTN has dropped in recent years”, Lee & Knight, (2005).

2.3 CISCO Unified Communication Manager Overview

Cisco Unified Communication Manager is an Internet based communication systems incorporating tone of voice/ capture/and information, and mobility applications. Thus a more successful, protected connection along with transforming method talking using new technology. Unified Communication represents a statement prototype swing same as discovery of the wire. Unified Communication removes stops of successful telecommunication. Uses variability enabling in sequence distribution to create knowledge and value.

“Cisco Unified Communication Manager enables network infrastructure, security, mobility, network management products, lifecycle services, flexible deployment, and third party communication applications. It results in creating additional valuable message face-to-face conversation. More useful communication leads to a reduced time to market and agile transformation of business processes through collaboration”.

2.3.1 Cisco UC Solution Components

The Unified Communication policy comprises much traffic inside a solitary set of connections structure. Unified Communication equipment is capable of administering voice/video/data passage types and interfacing with all standards-based network protocols. Cisco IP Connections represents a new way of delivering Unified Communication functionality to venture clients.

The figure below illustrates a typical layers used in the Unified Communication voice infrastructure sculptor and layers in components.

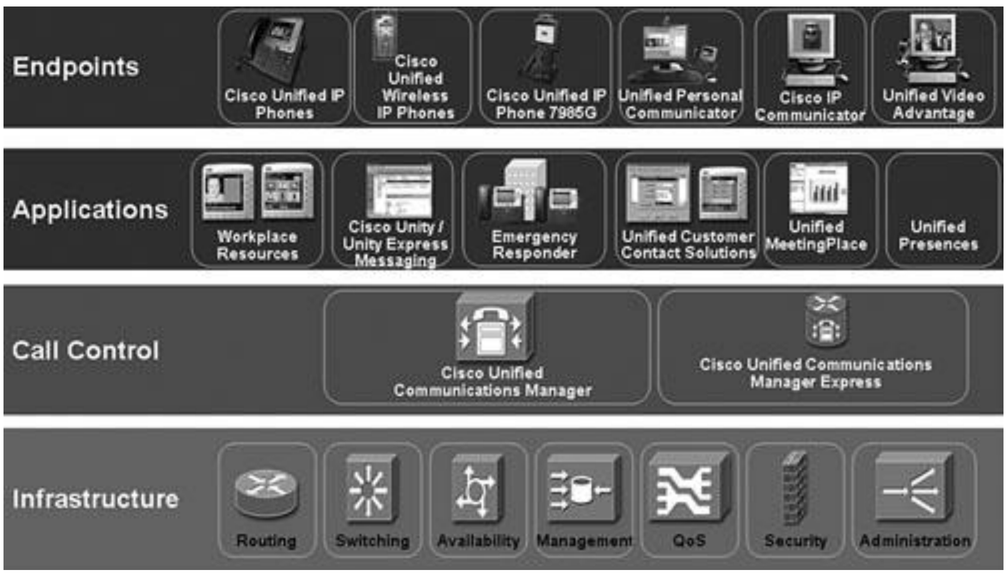


Figure 2-2 Cisco Unified Communications Solution Components

Mechanism of the distinctive layers is defined as follows:

Infrastructure layer: This includes typical components “routers, switches, as well as tone of voice access”. It carries “information, tone, and capture amid the entire set-up procedure.

Call control layer: provision of identify dispensation, control, and management control of panel arrangement. “The Call direct layer is shown by a Call Manager, Cisco UC Manager Express, or Cisco UC Manager Business Edition (CMBE)”.

Applications layer: Applications are autonomous from call-control functions and the physical voice-processing infrastructure. Applications include "Voice mail, integrated messaging, and unified messaging, Contact centers, Cisco Unified Meeting Place and Meeting Place Express, Cisco Emergency Responder (ER), Cisco Unified Presence server, and Standard procedure which are integrated through IP, that allows the applications to reside anywhere within the network".

Endpoints layer: Ensures the consumer receives all applications, and end machine be VoIP Phone, a Personal Computer using a IP communicator receiver, or a connections client or video workstation. The Unified Communication provides multiprotocol support for Skinny Client Control Protocol (SCCP), H.323, MGCP, and SIP.

2.3.2 CUCM Functions

Cisco UC Manager extends venture telephony facial appearance plus functions to packet telephony network devices; and includes all the types of equipment used to communicate with other devices including all IP telephony types.

Objectives and aims provided by Cisco UC Manager include:

Call processing: Away by which complete process to originate, route, and terminate calls, together with several bill plus numerical compilation of processes".

Signaling /and device control: this is done by the Cisco Unified Communication Manager for signal links to be streamed connections.

Dial plan administration: Sets up Cisco UC Manager used to carry out call map reading. The Unified Communication procedure administration is accountable for digit study of all calls, and scaling.

Phone feature administration: these are services extended by Cisco UC Manager".

Directory services: Customer validation is performed in the vicinity or aligned .

Programming interface to external applications: uses services .

Backup and restore tools: Cisco UC Manager provides a Disaster Recovery System (DRS) to support and reestablish the Unified Communication Manager configuration database. The DRS scheme also supports “call details, management and Analysis and Reporting.

2.3.3 CUCM Signaling and Media Paths

For Cisco IP Phones to communicate CUCM uses SIP or SCCP for call arrangement and rip down and for complementary service errands. After a call is fixed, medium swap over occurs straight involving the Cisco internet Phones across the network to carry the audio.

Example: Basic IP Telephony Call (placing a call from phone A to point B)

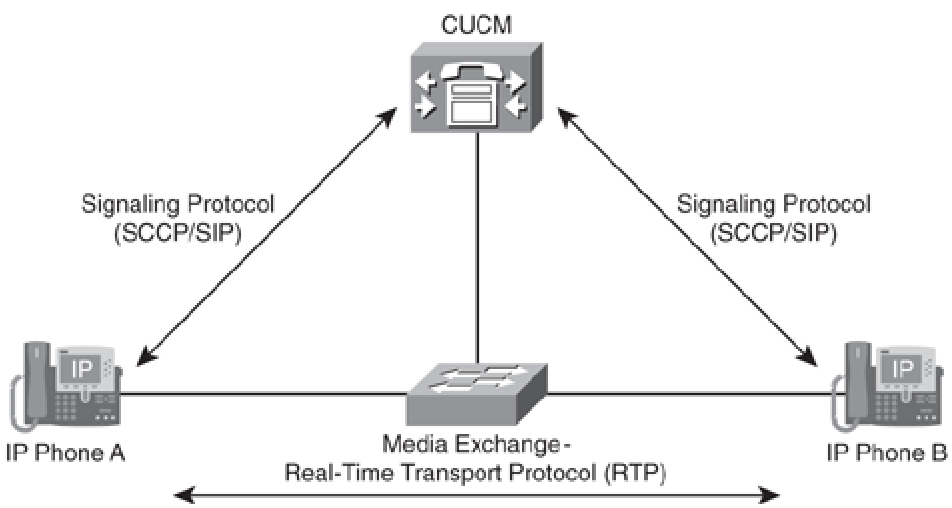


Figure 2-3 CUCM Signaling and Media Paths

2.3.4 CUCM Hardware, Software, and Clustering

Network appliance use Cisco UC Manager Release 6.0 that is a “bunged scheme to facilitate and supports only Cisco-authorized application plus utilities”. Simplification of the setting up and improvement for Goals of the appliance model.

The features of The CUCM appliance include:

Complete hardware and software solution.

Appliance OS providing ease of installation and upgrade, while also on condition that security and reliability are enhanced.

We then be able to advance Cisco UC Manager Servers whilst carry on to progression of calls.

Management parameters variety output via a valuable boundary for provision of information in order to approve supervision of application, such as “NetIQ Vivinet Manager, HP Open View, plus incorporated investigate PROGNOSIS”.

Application will be operated through or devoid of plug n play.

Clustering shore-up of servers in support of the reason of joblessness and load allocation.

2.3.5 CUCM Cluster

This mode will allow the set-up to balance to numerous thousands of endpoints, provided that idleness and breakdown not affecting administration.

This is as shown below:

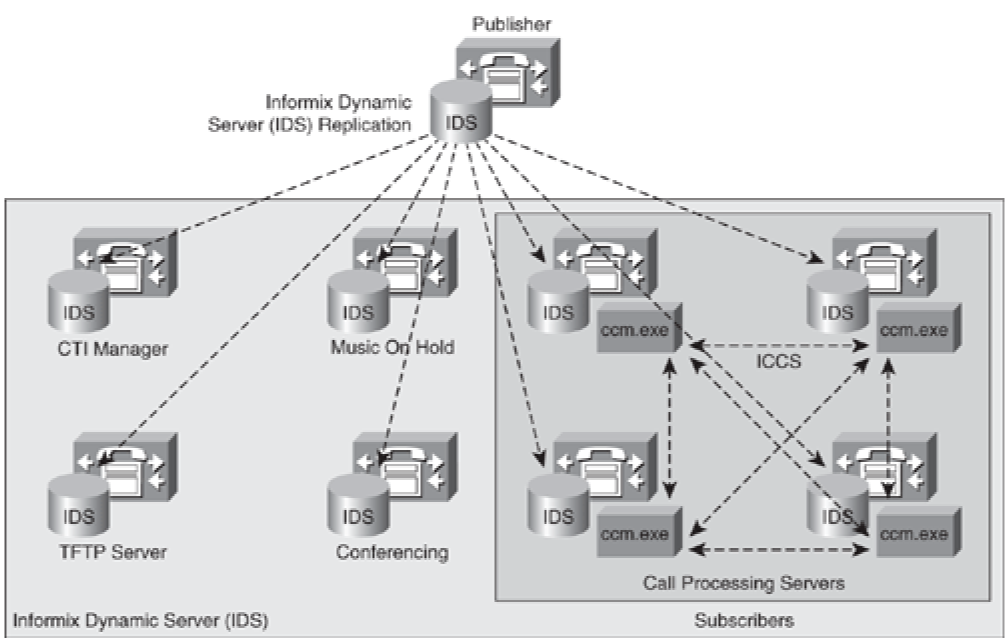


Figure 2-4 Cisco UC Manager Cluster

“Appliance setting is stored in the database. The folder is the storage area for examination and maintenance. The database replicates virtually all arrangement in sequence in a hub-and-spoke topology (one publisher, many subscribers)”.

2.3.6 Cisco UC Operating System

The Unified Communication Manager O/S will be based on Red Hat Linux. O/S and application updates are given by Cisco during patching that are digitally signed by Cisco. Uncorrelated software plus applications cannot be uploaded or installed into the scheme.

We have static configuration data and user facing features as described:

Static configuration data – used for Read/write access

User-facing features (UFF) – include features such as “Call Forward All (CFA), Message Waiting Indication (MWI), Privacy, Enable/Disable, Do Not Disturb, Enable/Disable,Extension Mobility.

2.4 System Architecture

In the system architecture we find an integration of communications, “converged voice/video/ and data over one network as shown below.

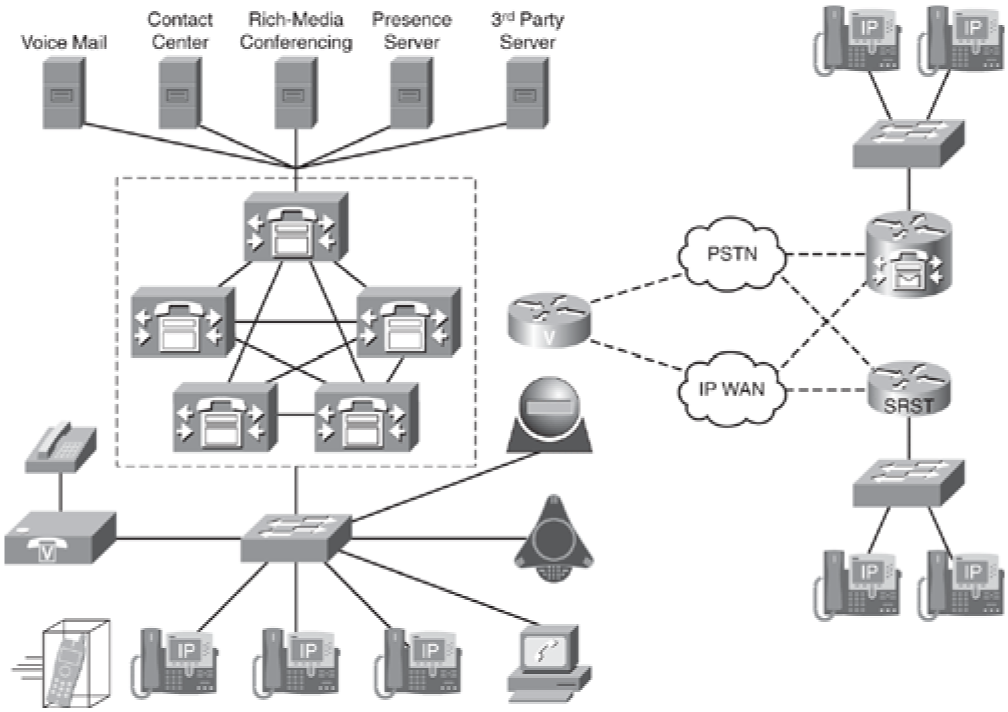


Figure 2-4 A Cisco UC Network for the enterprise environment.

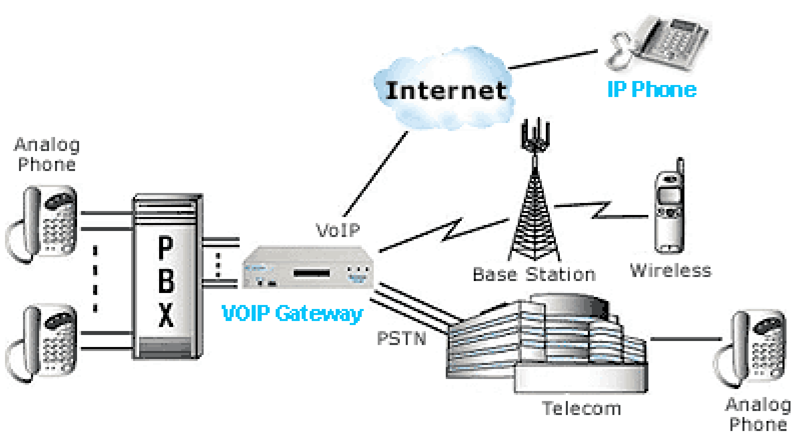


Figure 2-4 B Gateways between VoIP and PSTN to realize Unified Communication

The system architecture will be staged as shown below:

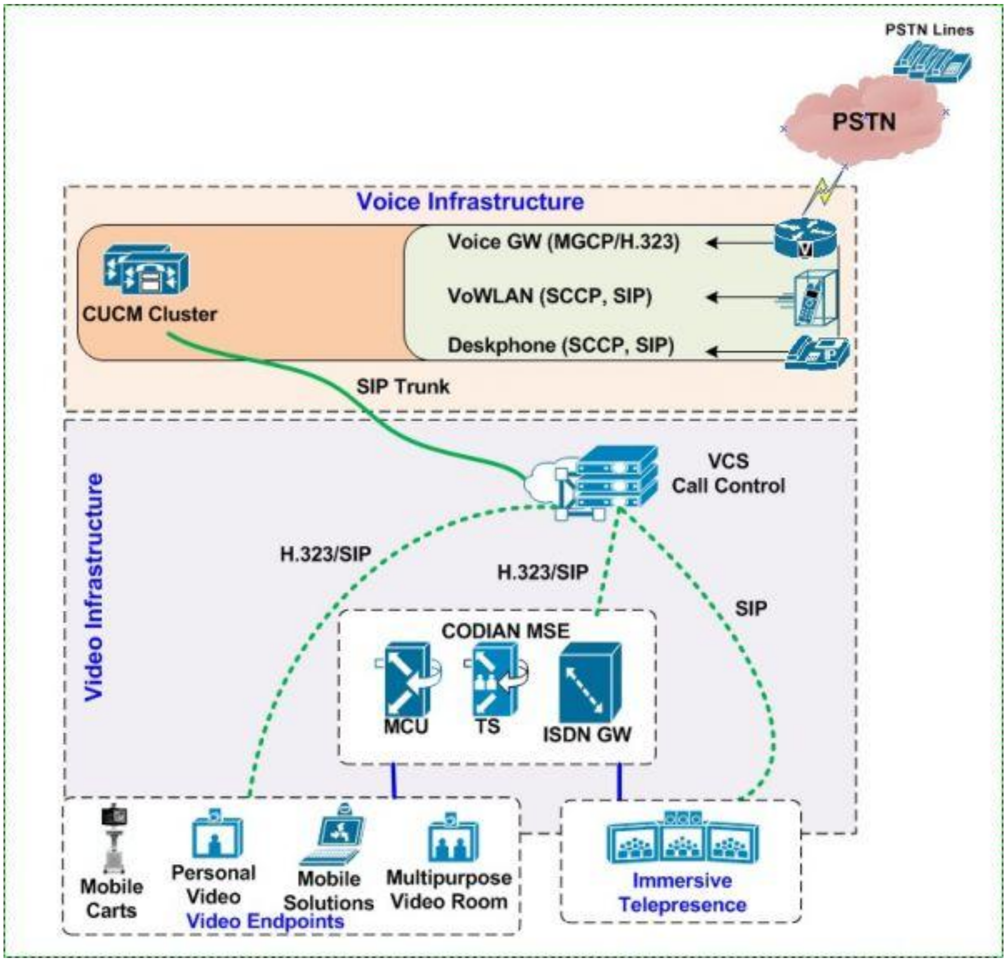


Figure 2-5 Deployment platform of the gateway

2.5 PSTN Gateways and Protocol

Connecting IP telephony system to the Public switching network requires an FXO gateway; that will be functioning as a conduit amid internet set-up and the Switching Telephone Network. Depending on where the voice passage originating from, an FXO gateway translating the voice traffic into the proper form to reach its target network (IP or PSTN). For example:

Condition that voice traffic originates from the Public Switching Network the Voice over IP Gateway will convert the analog voice signal into a digital signal.

Condition that voice passage is originating commencing internet set-up the Voice over IP gateway will decompress the digital packets into a digital pointer that is then transformed into an analog signal to be sent transversely the PSTN.

The below diagram shows a fundamental FXO access used to set-up IP telephony system.

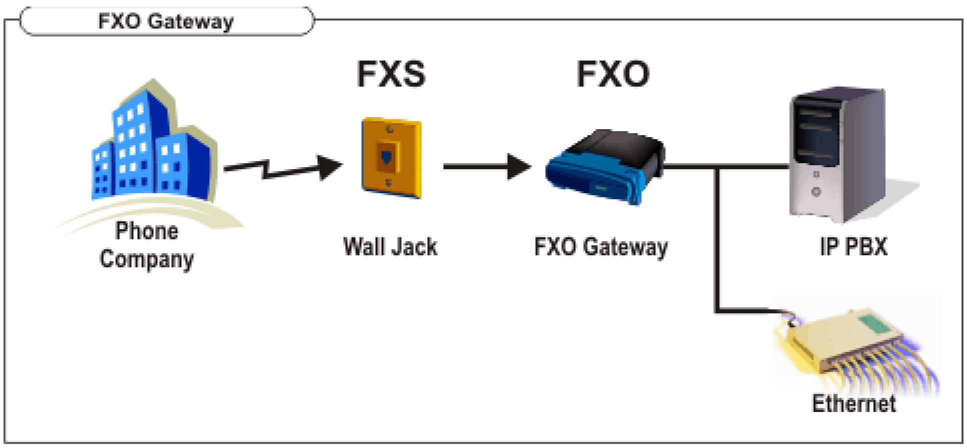


Figure 2-6 Basic FXO Bridge for a Voice over IP system

This is defined as "an Internet telephony access used to construct a connection linking the worlds of inheritance telephony and the Voice over IP. Gateways are characteristically used to fix inheritance telephone systems (PABXs or ACDs) with Voice over IP resources, or to fix up to date Voice over IP phone systems with legacy phone lines.

Additional uses for VoIP gateways comprise theatrical migrations, where the opportunity acts as a connection flanked by the PSTN, a bequest PABX and a new IP PABX. In this case, the PSTN trunks are linked to one boundary on the access. Another interface connects to the trunk port on the legacy PBX. The new IP PABX is integrated over a Voice over IP protocol (generally SIP). The gateway directs some incoming calls to the legacy PABX and others to the IP PABX. It also passes calls between the two PABXs. This allows some department or other subdivisions of the company to remain on the legacy system while others move to the IP system".

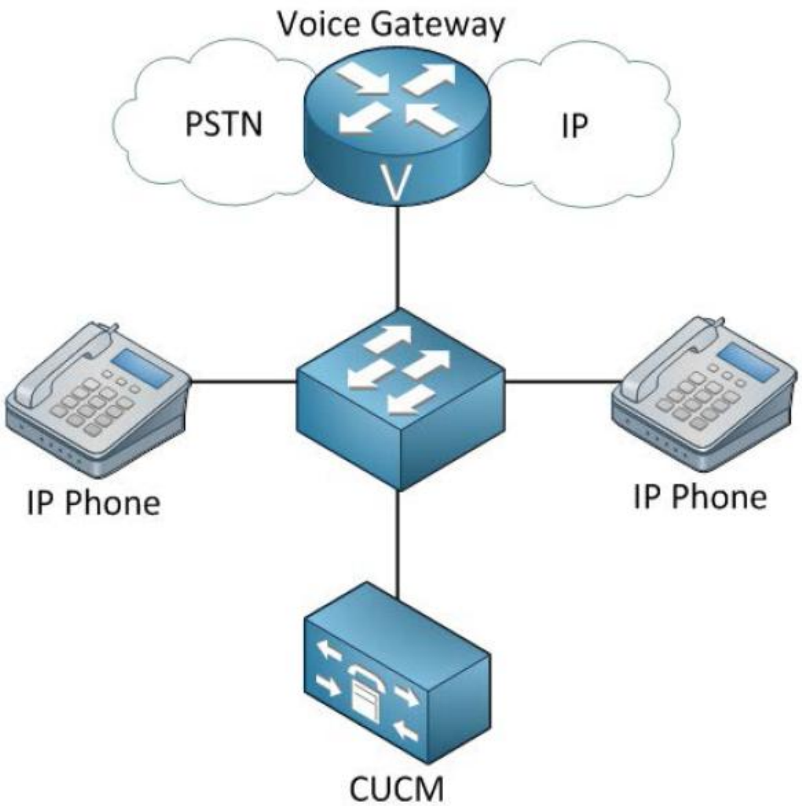


Figure 2-7 Voice Gateways

Only an individual call is able to be full of life at any one time using “Cisco” admission analog and Cisco admission digital trunk gateways:

1. Cisco access analog gateways

Station (Analog) gateways – this connects CUCM to POTS analog phones, fax machines, voicemail and IVR systems and connected through “FXS ports (supply 2-pair (4 pin) – Tip and Ring finished by RJ-11) to generate dial tone and battery”.

Access analog trunk gateways – bond UC Manager to trunks (PSTN central office (CO) or PABX) for the correlation to a conventional PABX.

2. Digital trunk(Cisco access) gateways

“Connects Cisco UC Manager to various TDM destinations through much higher capacity for the unified communication

2.6 Selecting the Right Integration Solution

In (TechTarget, 2009), three broad factors are considered:

“Features of Applications. Packaged applications from different vendors sometimes come with standard adapters that can be used. In condition applications are home-based, profoundly personalized, or bequest central processing unit systems, then incorporation will liable necessitate building an adapter from scratch.

Types of Transactions - long-running vs. short running transactions. If the applications you are trying to integrate support long-running transactions, then a message-queuing solution that can send messages to a queue and then forward them to the destination application may be the preferred choice. If the applications support short-term connections, where the customer relevance sends a appeal to the server applications and waits for a response, then a Remote Procedure Call (RPC)-based solution may be preferred.

Types of Data: Differences in the data formats and object models of different data bases may necessitate technologies that provide a quantity of kind of change expertise to deduce statistics from dissimilar systems”.

2.7 Integration Challenges

Integrating business applications is however not easy. To start with, for full application-to-application integration to take place, the number of inter-application connections increases exponentially with the addition of an extra application. Total number of connections required can be given as per the formula below:

$$\mathbf{T = N\,(N - 1) / 2} \qquad \text{Eq} \quad (1)$$

Whereby

T represents total number of connections;

N represents total number of applications.

Clearly, it may not be prudent to implement a point-to-point integration solution especially in A situation where the number of applications is involved is large.

Other challenges that may be encountered are:-

The target applications are implemented using different technologies and architectures, different database solutions and computer languages and may even have been built with different operating system environments in mind.

Most applications are constantly evolving, due to the dynamic nature of Information Technology and also because of evolving user needs. This complicates implementation and maintenance of application integration.

2.9 Review of Critical Literature

In this regard, the researcher has identified the gaps that still exist on the design of Unified Communication whereas there are most gateways devices that sustain numerous access protocols. In this research, access protocol collection will be based on the capabilities of the communication network available at the UoN and will be as follows:

H.323 –"Peer to peer calls processing protocol and configured; most of the configs will be completed in Cisco IOS dominion contour edge on the gateway of voice.

MGCP – Client/server model through CUCM Administration. CUCM is the server while MGCP is the client (TDM interface on the router) and simplifications of the configuration of the gateways of voice from end to end in a centralized admin throughout the Call Manager GUI. "It works in such a way that when the gateway is reorganizing, the TFTP server pushes arrangement documents to the IOS gateway, and loads any indispensable for GUI management performance

SIP – Peer to Peer protocol by IETF and perform media negotiations using SDP. It establishes, maintains, and terminates calls between two or more endpoints using requests and responses. The call manager supports SIP as trunk and line side support. The trunk in SIP is used in CUCM to direct calls to the gateway.

SCCP – Client/server model communicating between IP and CUCM. The Cisco VoIP telephones and tone of voice gateways obtain their design for MGCP.

2.10 Summary/Conclusion

Telecommunication services are indispensable because they are the driving forces of economic growth and integration; and the expected "Unified Communication" may offer a solution to the problem.

CHAPTER THREE

RESEARCH METHODOLOGY

3.1 Introduction

Deal with descriptions of methods functional in carrying out the research study and how it is organized. The objectives of this project were: To develop a comprehensive PSTN in CUCM Unified communication network and to investigate factors influencing adoption of unified communication system in Public institutions in Kenya – University of Nairobi. It deals with the process, principle and procedure of identifying a problem and systematically seeking facts that will help one find a solution to the identified problem; and methods of analysis or techniques to be used to analyze the model.

The project was broken down into:

Model development and validation

This is where the proposed process model was developed and validated.

It shows how the integration was designed.

Technical fact finding

This stage was meant to create an informed base on which to build the proposed process model of application integration. The key objectives of this stage were:

To obtain the views of IT professionals on available systems development methodologies and their applicability to application integration projects

To obtain views of IT professionals about the challenges of applying current development methodologies to application integration projects.

3.2 Selection of the Study Area

For the study area, all information processes within the ICTC Department – Communication and Data section of the University of Nairobi were selected. These were processes concerned with facilitation of communication within internal and external clients.

The selection of the study area was advised by the need to identify a communication domain that could reasonably be used for integration model for unified communication networks. That is:-

A domain where there is a lot if interaction within and without the institutions.

A domain where different platforms of VOIP and PABX legacy phones interact.

A domain where there is a lot of communication among both internal and external business entities; either manual or automated.

3.3 Integration Design Model

The design is based architecture and built using integration design, **coding** (the software codes necessary to actualize any changes to affected systems and for the integration are developed) and **testing**(this is a full regression test on all affected applications as well as integration tests on all interfaces). It was developed based on analysis of requirements, systems restructuring and enhanced technologies.

Assembling was done as shown below:

3.3.1 Gateway Integration with CUCM

The gateway (voice gateway) will be translating between various types of signaling and media. Voice gateways used to facilitate a Cisco Voice over IP Phone to communicate with Time Division Multiple (TDM) edge. An illustration is PSTN, legacy PBX, Analog phones, facsimile and so on. Here, we will be discussing the following Gateway integration:

3.3.1.1 Integrating MGCP gateways with CUCM

MGCP is a protocol comprising of plain-text that callers use to control devices used to manage the IP telephony gateways.

The MGCP uses both client/and server protocol for allowing the identify representative (CA) to manage a precise entry protocol endpoint (port).

It is more useful as a centralized voice gateway administered in the Call Manager. The Call Manager will be controlling states of all ports on the gateways endpoint.

The gateway can be controlled per endpoint (TDM port) level but H.323 and SIP cannot.

Integrating the MGCP requires a centralized dialing plan to be configured and Centralized voice gateway configurations to ensure trouble free implementation in a large SP arrangement.

MGCP Access shore up:

"MGCP shore up in Call Manager needs extensive assortment of ADC/DAC interface that is worn on many Cisco routers and switches platform. Call Manager will push the Cisco IOS MGCP access arrangement to the opening of configured parameters.

Call Manager Chains backhaul of "Q.931"; which will be supported on ISDN voice ports. The backhaul then allow Call Manager to be processing the "Q.931" messages from the ISDN circuit system. The access router will then encapsulate the "Q.931" signals to access over the Transport Control Protocol port.

MGCP Call flow:

The call flow of communication is used between the Call Manager and the Gateway.

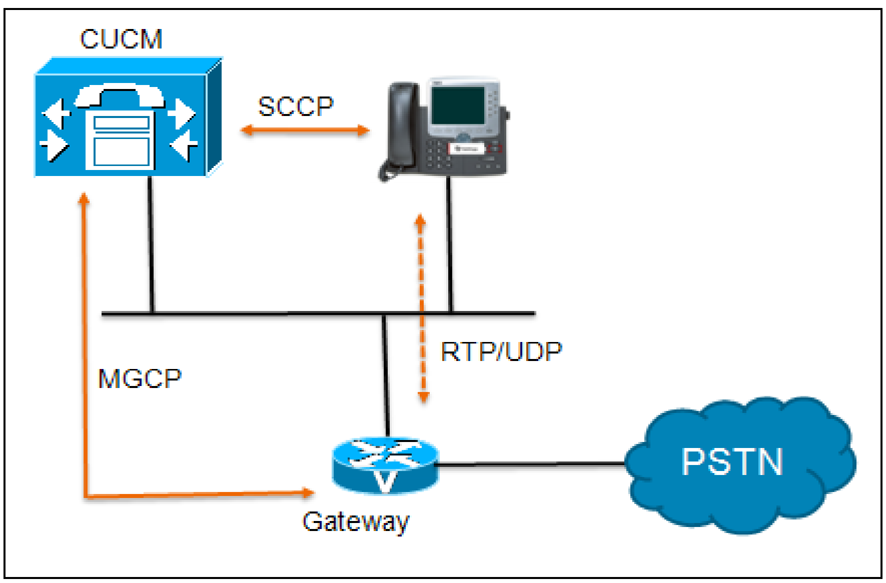
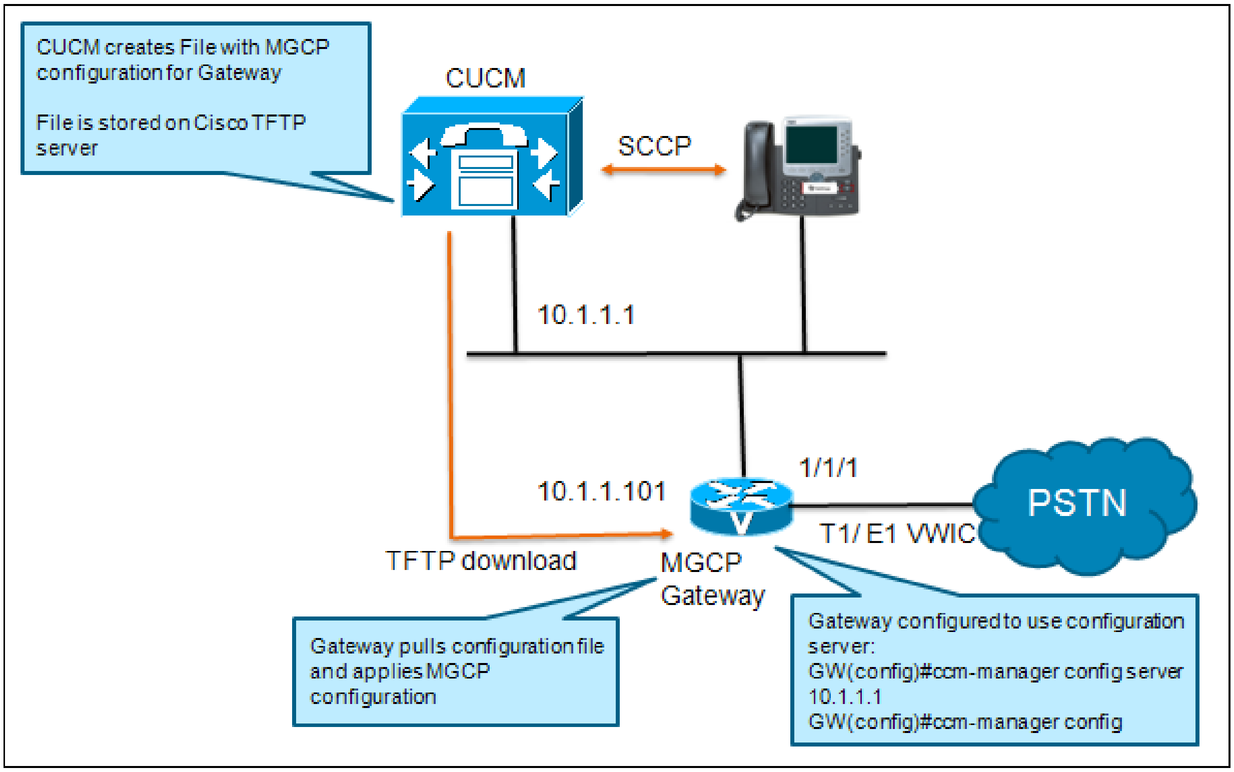


Figure 2-8 MGCP call flow diagram

MGCP outline of Configuration Server:

"Server for Configuration attribute allows all "MGCP" administration of configurations to be in the course of the GUI in Call Manager.

Figure 2-9 MGCP configurations sever call flow:



Q.931 Backhaul:

The backhaul - Q.931 is defined as "dependable transport layer TCP/IP correlation stuck between Call Manager and Cisco MGCP gateway. Sequence will be carrying the unrefined Q.931 signaling to Call Manager being processed natively. The MGCP Gateway will be responsible to terminate Q.921 Link Access signaling in Layer 2, but and setups and call tear down signal backhaul sent to Call Manager to be processed".

1.1) MGCP Gateway Configuration: CUCM

"Affix MGCP entry to Cisco UC Manager

Step1: In Call Manager admin, select tool > Gateway.

Step2: Click the insert new.

Step3: Select MGCP access or router name

Gateway Type: Cisco XXXX

Step4: Click Next.

Step5: Select MGCP since the procedure drop-down menu then next"

MGCP Gateway Configuration
"Sample MGCP gateway configuration displayed on the CUCM Page: *Select the type of gateway you would like to add:* *Gateway Type Cisco xxxx* *Protocol * - options are MGCP / SCCP* *In this case for MGCP gateway, select MGCP as the Protocol"*
Detailed MGCP Gateway Configuration
Sample MGCP gateway configuration display from the CUCM Administration Page: *"Gateway :* *Product: Cisco xxxx* *Protocol: MGCP* *Domain Name * : UoN-1* *Description: UoN-1 headquarters MGCP gateway* *Cisco Unified Communications Manager Group*: Default*

Configured slots, VICs and Endpoints *Module in Slot 0* *Module in Slot 1* *(To add modules refer the next section "Endpoint module configuration"* *artifact explicit configuration layout*
MGCP Endpoint configuration
Sample MGCP Endpoint configuration, from the CUCM administration page *"appliance in sequence:* *artifact: Cisco MGCP E1 Port* *Device Pool* Default* *Location Hub_ None* *Enable : PSTN Access* *border in sequence* *PRI Protocol Type* PRI UoN* *QSIG alternative* No changes* *Protocol region* consumer* *control assortment order* Bottom up* *PCM Type* A-law"*

1.2) MGCP Gateway Configuration: Cisco IOS Configuration

As in:

call-manager configuration

call-manager configuration server 192.168.1.300".

Configuring MGCP Gateway Registration
“regulator E1 0/3/0 *frame crc4* *line code hdb3* *pry-group timeslots 1-31 service mgcp* *call-manager mgcp* *call-manager music-on-hold”* *“!* *“mgcp* *“no mgcp package-capability fxr-packageno mgcp timer receive-rtcp* *“mgcp sdp simple* *“mgcp rtp payload-type g726r16 static* *!”*

1.3) MGCP Gateway: Registration Verification

Verifying MGCP Gateway Registration
“Router# show ccm-manager *MGCP domain name : Router* *Primary Registered 10.16.240.124* *Current active Call Manager: 10.16.240.124* *Backhaul/Redundant link port: 2428* *Failover Interval: 30 seconds* *Keep alive Interval: 15 seconds*

PRI Backhaul Link info *Link Protocol: TCP* *Remote Port Number: 2428* *Remote IP Address: 10.16.240.124* *Current Link State: OPEN* *Statistics:* *Packets received: 32* *Receive failures: 0* *Packets transmitted: 32* *Xmit failures: 0* *FAX mode: Cisco"*

1.4) Verifying MGCP Endpoint Registration

Verifying Endpoint Registration
"Router# show mgcp endpoints

1.5) Fractional T1/E1 Configuration on MGCP Gateway

Here much configuration analysis, "not all instance correlation" consumed. Specifying the integer of serviceable "B channels" in Cisco UC Call Manager by localizing the CUCM examination constraint; Change "B-Channel" safeguarding mode. Configuration encompassing "B channels" in service mode. The Control service mode locale has no cause on the XML arrangement folder that is to be established on the MGCP configuration. And necessitate to subsist performed to only specify the quantity of lively time – slots.

New MGCP package capabilities/commands to facilitate requirements for the new versions of "Cisco IOS" and Call Manager for proper communication purposes. Most often use the automatic configuration commands or else if requirement for a fractional T1 / E1 is needed.

1.6) Fractional T1/E1 Configuration on CUCM

Since CUCM Administration, select "*System > examination Parameter after that select the Cisco Call Manager; then configuration page top to access essential to change the B - Channel Maintenance mode parameter"*.

The considerations allow Cisco UC in altering each of the B-Channel repairs particular state for the "era - slots on crossing point".

The selection order for fractional T1 interface has four active channels:

Xxxx yyyy yyyy yyyy yyyy yyyy

1.7) Verifying T1/E1 Controller Status

"Router # show controller e1

E1 status should be UP.

E1 should be Channelized E1 – balanced, 120 ohms

No Alarms Detected".

Verifying T1/E1 Controller status
"UoN-1 ## show controller e1

1.8) Verifying T1/E1 Interface status

Operation check is easily done by "MGCP" restricted "T1/E1 ISDN PRI" boundary by means of the

"# illustrates isdn-status domination and checking" the Layer1 plus 2 statuses as demonstrated here.

Verifying T1/E1 Interface status
“Router# show isdn status

3.3.1.2 Integrating H.323 gateways with Cisco UC Manager

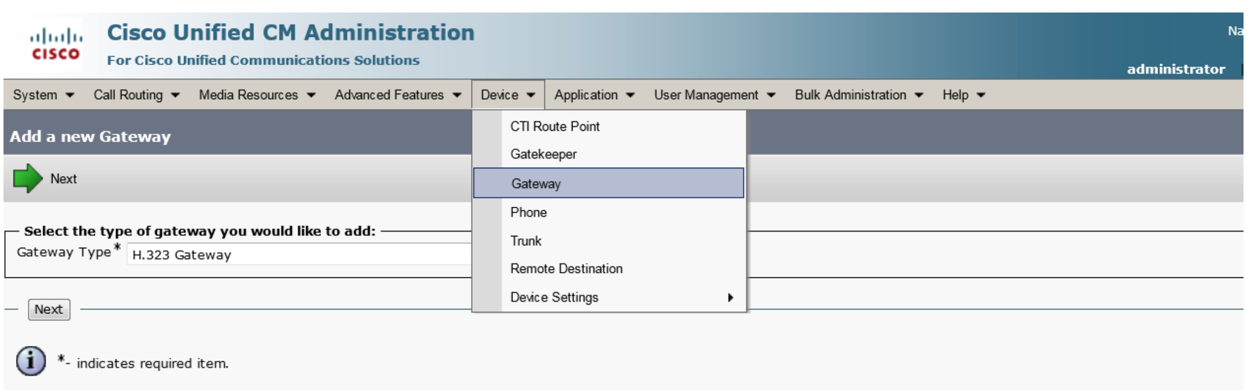
Integrating the “H.323” access with Cisco UC Manager will allow registering of devices with Cisco UC Manager for making and receiving calls from the Public Switched Telephone Network service provider network. The signaling information is transformed by Call Manager phones. Whereby, the call manager acts as a proxy between the VoIP IP Phones and Gateways. Cisco IP Phone and H.323 gateway use Rtp voice media stream where the DSP on the gateway exchanged the RTP media stream in to a TDM format required.

For any call from VoIP phones to PSTN rerouting to the H.323 is very necessary to enable a full connection across the network.

The steps are as shown:

“Step1: In Cisco UC Manager Administration, select Device > Gateway

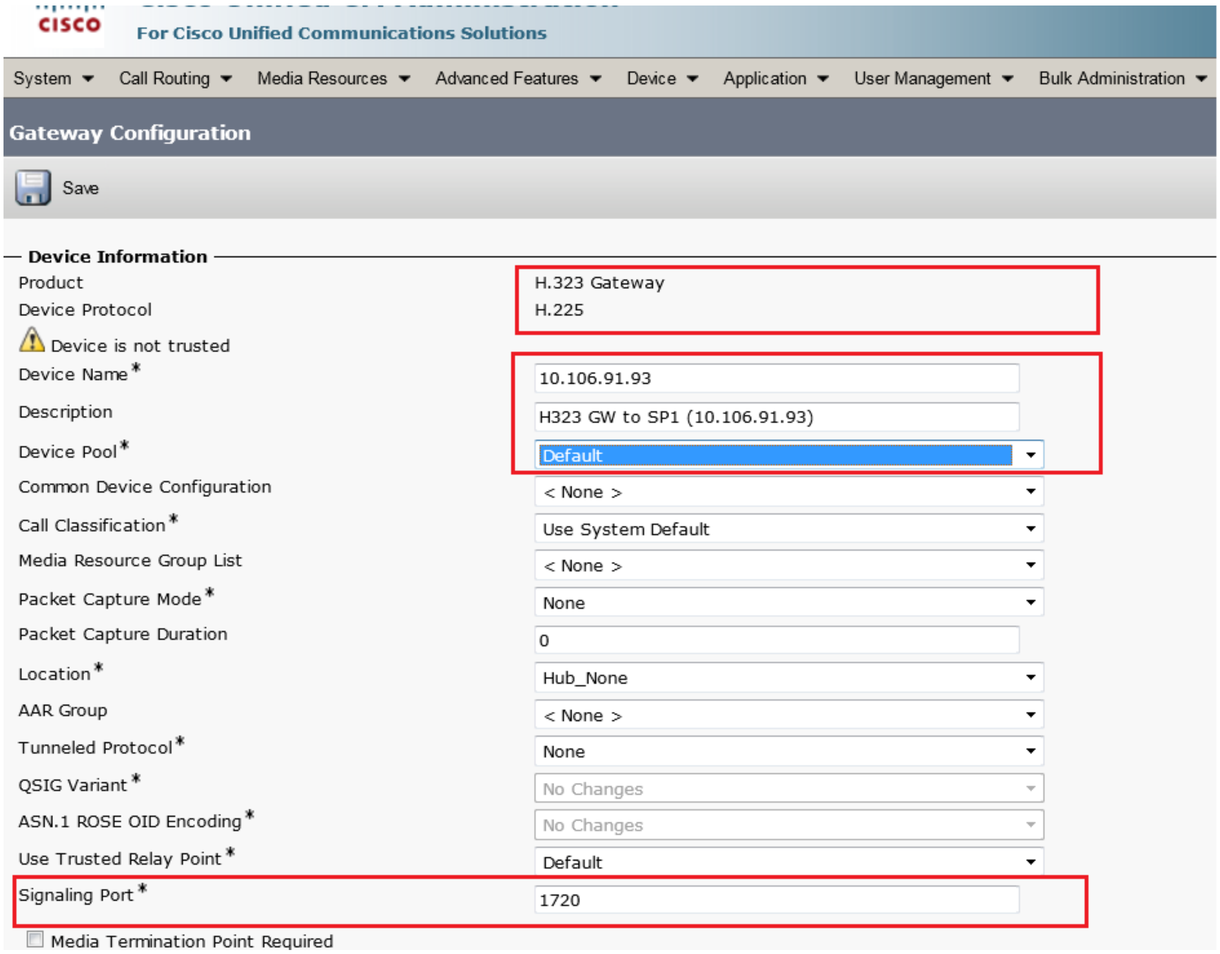
Step2: Add New and choose Gateway Type > H.323 Gateway



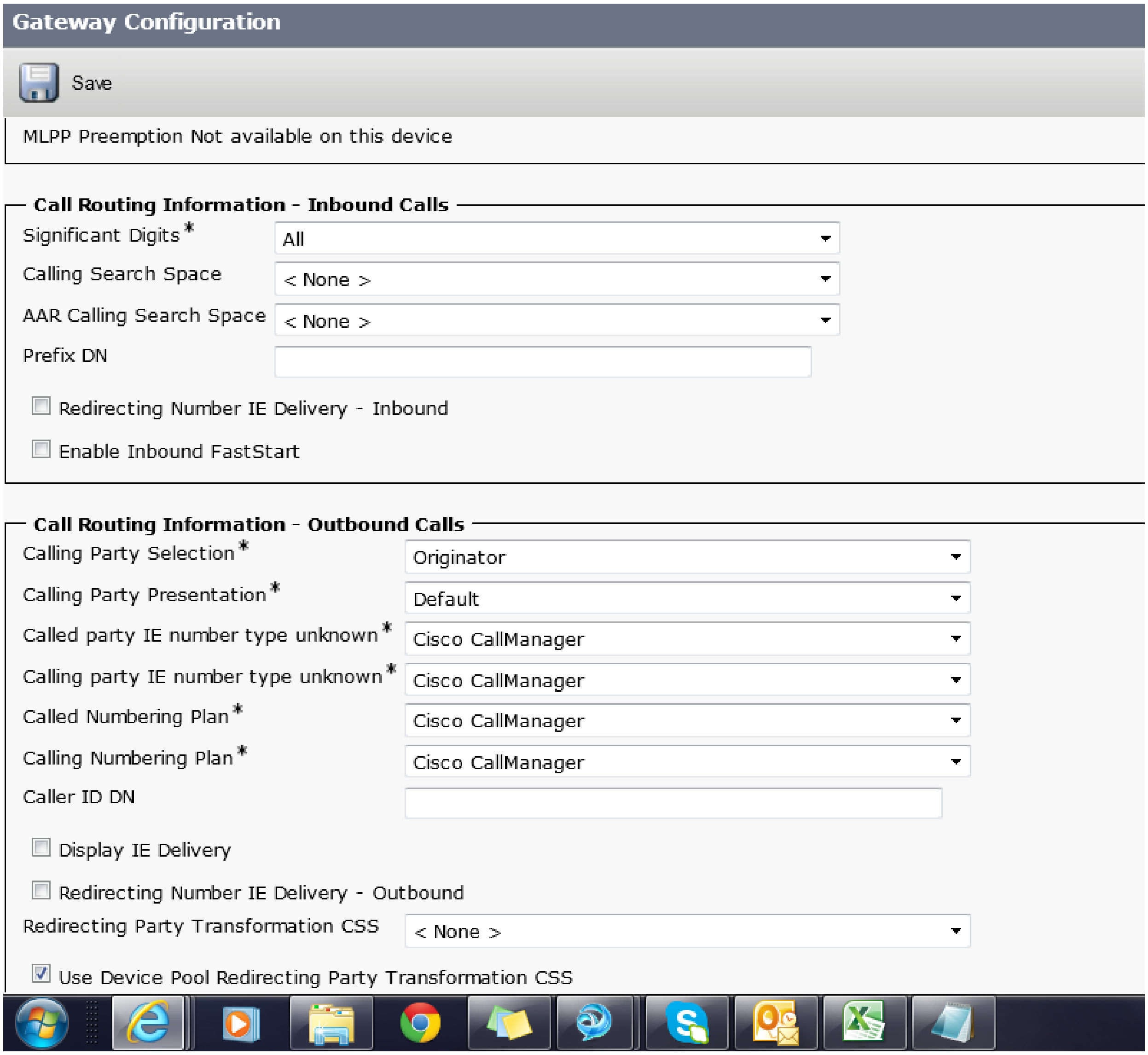
Step3: Click Next

Step4: ***:*** *Enter the Device Information: Name, Description and Device Pool*

Put the H.323 Gateway IP address or name in the Description Field”.



Step5: Scroll then enter the Call Route Info - Inbound calls & Outbound calls as given below,



Step6: Save and/or Apply Config

2. Configuring the basic Cisco IOS H.323 functionality

"!

Interface LoopBack0

ip address 10.10.0.91 255.255.254.0

H323-gateway VoIP interface

H323-gateway VoIP bind scraddr 10.10.0.91

!

! --- The gateway then accepts inbound call from the specific IP address.

! H.323 bind command gives IP address for every communication with the access using Cisco UC Manager i.e. inbound PSTN call routing.

VOIP Dial -Peer Configuration

!

Dial-peer voice 1 VoIP

Destination-pattern 2...

No vad

Ip quos dscp cs3 signaling

Codec g711ulaw

Session target ipv4:10.106.91.80

!"

Routing calls from the H.323 gateway to Cisco UC Manager, we configure one VOIP dial-peer.

The H.323 is the default signal protocol for CISCO IOS VOIP Dial-peer; and no configuration will be needed to implement the inbound call routing functionality.

Keep alive configuration.

Uses H.225 keep alive to monitor the TCP session between Call Manager and H.323 gateway. Active calls being dropped is avoided when phone call malfunction amid "h.323" gateway and Call Manager is realized, Preserving of TDM to VoIP calls is done globally by configuring H.225 command.

"*Voice service VoIP*

h323

No h225 timeout keep alive

!"

3.3.1.3 Integrating SIP gateways with CUCM

SIP Gateways will be integrated with Cisco UC Manager by using SIP Trunks. SIP gateway is translates between different types of signaling and media as shown below.

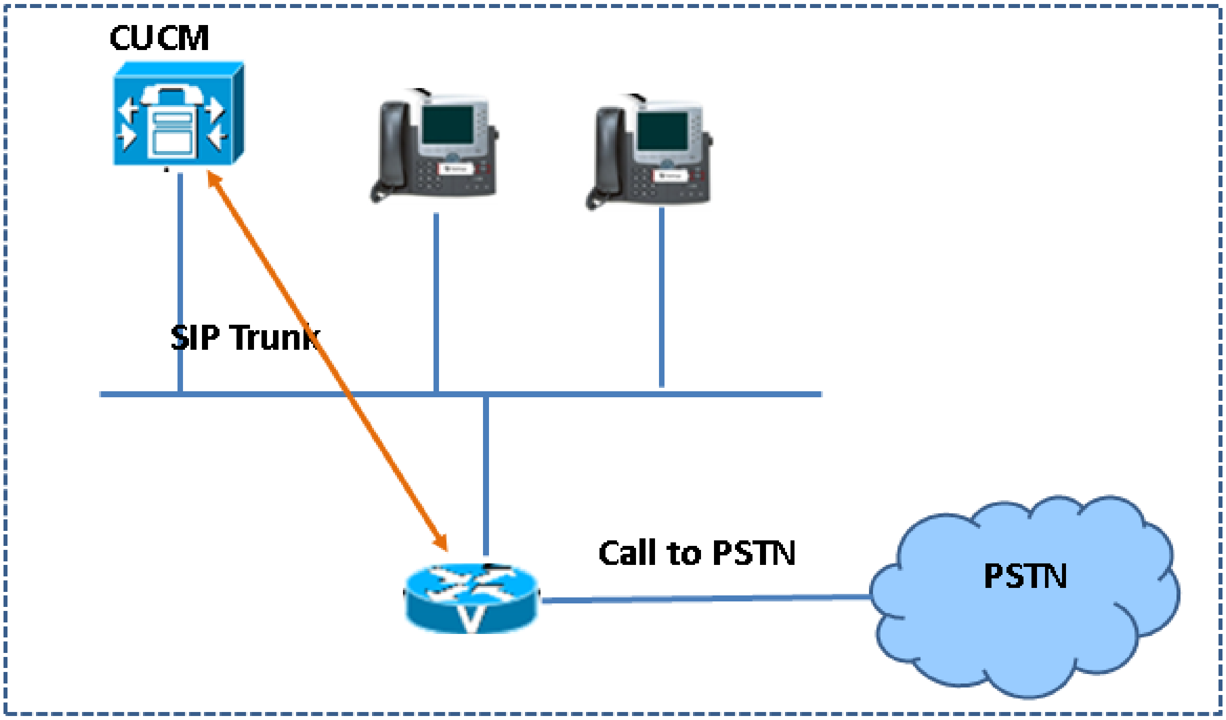


Figure 2-10 SIP Trunks

Configuring a SIP gateway in Call Manager:

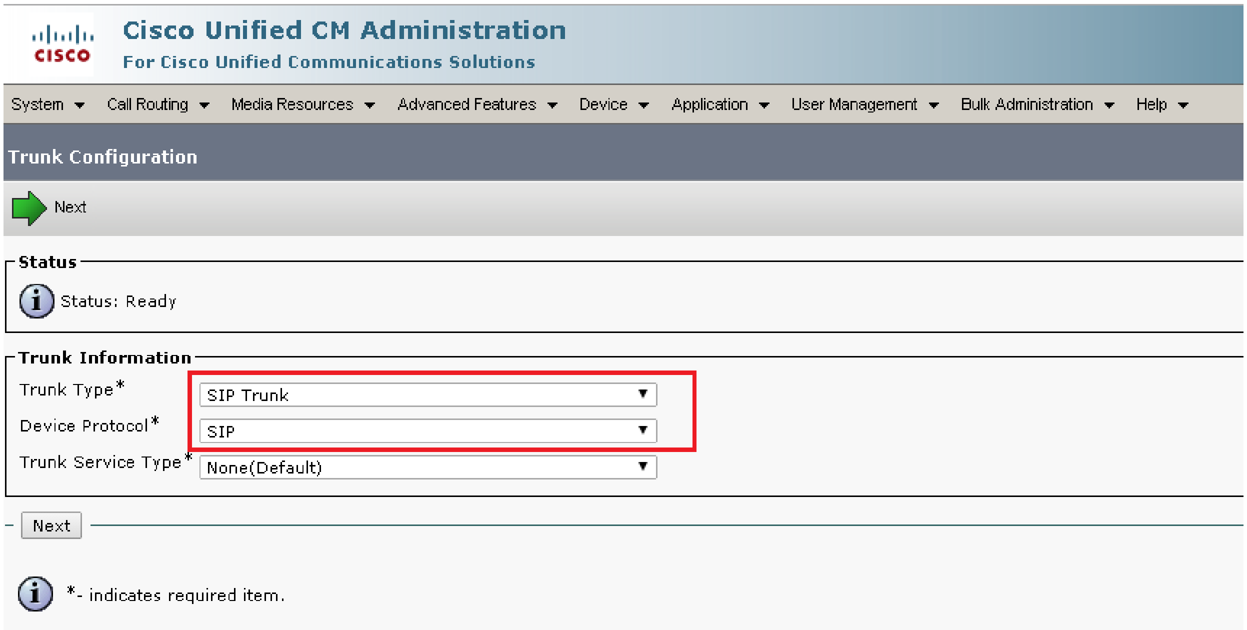
1. Configure a SIP Gateway in CUCM

2. Configure SIP Gateway on the Cisco IOS router

Configuring SIP access functionality on the Cisco IOS router:

1. *Constitute "Cisco IOS SIP" functionality.*
2. *Constitute "Cisco IOS" call-routing information.*
3. *Constitute the "SIP" user agent parameters.*

Enter CUCM and add SIP trunk as shown below:



Enter the Trunk configuration: Device Name, Description and Device Pool; and configure SIP trunk parameters using system IP addresses.

Configure Basic Cisco IOS SIP Functionality:

Enable TCP

Use IP address of 172.20.100.125 as a supply IP address while communicating with Call Manager.

Interface loopback0

Ip address 172.20.100.125 255.255.254.0

!

The "dtmf-relay" dominion specify exploit of 1st priority in in-band configuration:

!

Dial-peer voice Voice over IP

Destination-pattern 2…

Session procedure sipv2

Session objective ipv4:10.1.1.1

Codec g711ulaw

Dtmf-relay rtp-nte sip-notify

!

SIP Trunk: MTP Allocation Configuration

Enable the Media Termination Point (MTP) statically for both VoIP and PSTN calls and select a codec from MTP most preferably from SIP trunk. The G.711 is audio codec that communicates between Call Manager and SIP gateway run over the high- speed LAN at gigabits measured speeds. VoIP calls use an audio codec (G.729) since the TDM makes use of pulse code modulation techniques, with the voice feed set to 64 kbps. The codec is manually provisioned on the Cisco gateway router.

3.4 Data Collection and Analysis

The key methods of data collection used were questionnaires and interviews; both formal and informal. Due to the different target groups and purpose of the study in the different stages, different questionnaires were used at each of the stages. Specific respondents were from:

Technical Fact Finding

In this stage, the key interaction group comprised of Telecommunication/IT professionals with advanced experience in the systems lifecycle, available methodologies and wherever possible experience in application integration projects. The respondents are shown in Appendix A1.1.

Model Development and Validation

After model construction was complete, 5 experts were invited to validate the model. This process Involved critiquing both the process used for model construction and the product based on the respondent's experience. When selecting the reviewers at this stage, the intention was to get experts with the following combination of characteristics:

Expert level of Telecommunication/IT knowledge

Expert knowledge of systems development methodologies

In addition, to the above the expert reviewers had one or more of the following:

Vast experience in large systems development projects

Past involvement in management of large IT projects a large bit of which involved application integration

Past involvement in management of big institutions or organizations.

This section also outlines how the project data was collected and implemented. It gives the number of legacy phones using PSTN and IP telephony using CUCM. The data is given in the tables below:

Table 3-1 UoN phones' data

NO	CAMPUS/COLLEGE	LEGACY PHONES	VOIPS	TOTAL
1	MAIN	800	400	1200
2	CBPS	400	150	550
3	CHS	150	100	250
4	CAVS	200	50	250
5	CEES	200	30	230
6	SOB	100	35	135
7	SOL	60	35	95
8	KSC	80	20	100
9	SOD	50	15	65
10	IAGAS	20	02	22
11	KISUMU	05	06	11
12	MOMBASA	10	08	18
13	EMCS	30	20	50
14	PENSION	05	04	09
16	UHS	40	25	65
17	UoN Towers	-	-	-
	Total	**2,150**	**900**	**3,050**

The process of having an integrated unified communication networks using PSTN gateways and CUCM server in the entire University of Nairobi campuses/colleges is very possible. The first phase can be tested; whereby we use what is already available to provide a unified communication system.

Key points synthesized from the research include:

Gateways are necessary mechanisms in an IP telephony background to provide functions because translation as of “TDM” tone of voice to Internet Telephony and “DTMF” relay

Cisco UC Manager is “MGCP call negotiator; thus organize accent interface. It gives a central configured and the functions to evidently pass Q.931 communication to Call Manager”.

An “H.323” gateway enables a flexible and easy way to connect VoIP calls to the legacy systems. Application of call routing configs will be applied on both the gateway and CUCM.

SIP gateways uses SIP trunk configuration to be implemented in Cisco UC Manager; and call routing configured on both the gateway and CUCM.

From the table above, the legacy phones are 2,150 compared to VoIP which are only 900. With integration of VoIP phones and legacy phones, the internal connection for provision of affordable and effective communication will be realized in the entire University; whereby a total of 3,050 customers will be connected to integrated unified communication.

3.5 Design model and tools

The research applied the following tools to represent the data and interactions of the entire configuration: GUI in CUCM, XML file into Cisco IOS commands, config-server commands with TFTP server IP addresses, CUCM proxy between IP phones, gateway to instructs call setup and teardown events, loopback/FHRP interface on routers, and coding

Scripting Language - PHP (Hypertext Processor)

This Language is appropriate for this project due to extensive use of remote execution. PHP is purely a server side programming language that is able to make system calls.
Mostly used for trouble-free preservation and prompt improvement time.

MySQL

This is a vigorous and scalable relational catalog supervision arrangement. It’s a multi-user, multithreaded database server that uses SQL to interact with and manipulate data. MySQL features include:

Multithreading capabilities enables the database to perform multiple tasks concurrently and allowing the server to process client requests efficiently.

It increases the good organization of retrieving precise and essential information by use of a inquiry and has the capacity to hold large databases

CHAPTER FOUR

RESULTS AND DISCUSSIONS

4.1 Testing and Results

A number of tests are conducted to verify that the requirements that will be identified during analysis will have been identified and implemented. We will be able to integrate existing telephone systems with IP telephony, and thereafter perform tests and show results in a summarized table. The test aims to show how much the technology supports to implement the task, the complexity of the design, the skill and knowledge base needed to implement the system, and the timing and performance constraint.

The requirements for the design model construction involved the following:

- CUCM server
- MGCP, H.323 and SIP gateways installed in the system
- PSTN with an installed PABX using analog/legacy phones.

The above requirements were installed in a laboratory and a call initiated to determine the functionality of unified communication network involving CUCM and PSTN gateways.

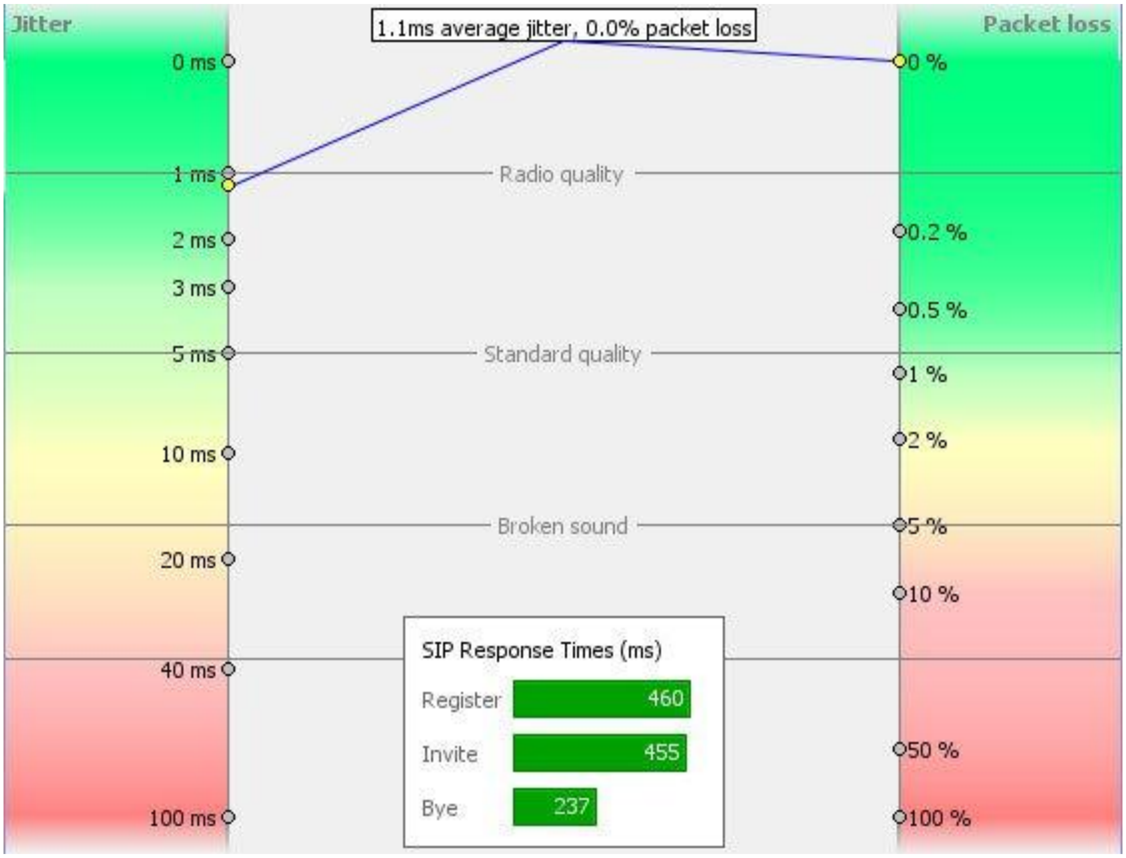
The research work achieved its expectations since the results showed the unified communication network is workable in my study area – University of Nairobi.

Calls from either side of VOIP and analog/legacy phones could be integrated into one.

To trust this unified communication method to business user's technology, I used the most reliable speed test tool called **On SIP VOIP test**.

On SIP offers VoIP-oriented tests. The testing tool offers valuable information concerning broad-band correlation that handles computer generated Voice over IP traffic. The experiment provides mobile chart evaluation and abstract of the connection's presentation. The result will be based on the metrics of speed, jitter, and latency, combined with a Mean Opinion Score which is based on individual participation from correlated excellence tests and gives a glimpse of how users could experience call quality over Unified Communication association.

Graph 4.1: On Sip VOIP Test



Results are as follows:

4.1.1 Unified Communication End-to-End Delay Measurement

Here measurement of VoIP and PSTN integration using end-to end simulation is done. The main design issues involves "real-time transport/real time control protocol", measurement of packet losses from transmitter to recipient side , inter-arrival delay and delay variation as well as Jitter. Reports from the sender and receiver will be shown, and how end-to-end delay is calculated.

4.1.1.1Time Stamp Calculations

Point in time trample calculation principally depend upon the type of application we use for voice only.

Timestamp Calculation at the Sender Side

For the experimentation purpose, the researcher uses 10 RTP packets. The audio that is used is fixed-rate the sampling period dictates the time stamp clock i.e. for each increment by the sample period the time stamp would likely to increment by one. If an output device provide an audio application that has blocks covering 160 sampling period .the sender timestamp would increased by 160 ,regardless of the block has transmitted or dropped.

Table 4.1 **Sender Timestamps.**

Sequence No. i	Si (Timestamp)	Si (ms)
1	**0**	**01:02:02:00**
2	**160**	**01:02:02:20**
3	**320**	**01:02:02:40**
4	**480**	**01:02:02:60**
5	**640**	**01:02:02:80**
6	**800**	**01:02:02:100**
7	**960**	**01:02:02:120**
8	**1120**	**01:02:02:140**
9	**1280**	**01:02:02:160**
10	**1440**	**01:02:02:180**

Calculation at the Receiver Side

When a receiver receives the packet, the timestamp of the received packet is according to eq (2)

$$\text{recipient timestamp } Rec(i) = \frac{Rec(i) \text{ in times of time unit x sampling frequency}}{1000} \qquad \text{Eq} \quad (2)$$

Where, Rec(i) represents time unit calculated by eq (3) with a sampling frequency of 8000Samples/Sec

$$Rec(i) \text{ in time unit } = (\text{arrival time of packet (I)} - \text{arrival time of packet(i-1))} + Rec(i-1) \text{ in times of time unit} \qquad \text{Eq} \quad (3)$$

Table 4.2 **Receiver Timestamps.**

Seq.	Arrival Time	Rec(i) (ms)	RecTS(i) (timestamp)
1	01:02:01:20	0	0
2	01:02:01:43	23	184
3	01:02:01:63	43	344
4	01:02:01:84	64	512
5	01:02:01:109	89	712
6	01:02:01:130	110	880
7	01:02:01:150	130	1040
8	01:02:01:170	150	1200
9	X	X	X
10	01:02:01:230	210	1680

The recipient calculates the timestamp only when the packet is received. The former timestamp is zero for the first packet because it acts as a reference. In below table the packet nine doesn't arrive at the receiver therefore no calculation is done for the packet.

4.1.1.2. Inter-Arrival Jitter Calculation Method

Exact statement there is no warranty that the packet sent, will arrive in order and delivered on time. If the packet is not received at the receiver on time/order over the network; it means delay occur that create a delay gap. The inter-arrival jitter (J) is defined by variation of delay in the network that is perceived by the receiver for each packet. Every packet received has a timestamp that informs the receiver that the time data is received in the packet should be played back. The difference in the "transit relative time" D (which is relative time discrepancy involving RTP timestamp of the packet and the receiver clock at the time when the packet arrives,) D (x, y) for the two packets x and y can be calculated by equation.

"The calculation of inter-arrival jitter for the packet 1 and 2
are shown below
Sender timestamp for packet 1, S1=: 0
Sender timestamp for packet 2, S2=0+160= 160 timestamp =160/8=20ms
Rec (1) in terms of time unit = 0 ms
Rec (2) in terms of time unit = (43-20) +0=23 ms Now the receiver timestamp for packet 1, RecTS(1) =
0
*Receiver timestamp for packet 2, RecTS(2) = 23 * 8000/1000 = 184*
Delay: D(1,2) = (184-0)-(160-0) = 24timestamp =24/8Khz=3ms
Jitter: J(2) = 0+(24-0)/16 = 1.5 timestamps =1.5/8Khz =0.188 ms
In similar fashion we can calculate the other inter- arrival jitter values as show in table 4.3".

Table 4.3 Inter-Arrival Jitter.

Seq. No, i	Si (Timestamp)	Si (ms)	Arrival Time	Rec (i) (ms)	RecTS(i) (Timestamps)	D(i-1,i) timestamp	D(i-1,i) Ms	J(i) Timestamps	Ji (ms)
1	0	0	01:02:01:20	0	0	0	0	0	0
2	160	20	01:02:01:43	23	184	24	3	1.5	0.188
3	320	40	1:02:01:63	43	344	0	0	1.406	0.176
4	480	60	01:02:01:84	64	512	8	1	1.818	0.227
5	640	80	01:02:01:109	89	712	140	17.5	4.204	0.526
6	800	100	01:02:01:130	110	880	8	1	4.442	0.555
7	960	120	01:02:01:150	130	1040	0	0	4.164	0.521
8	1120	140	01:02:01:170	150	1200	0	0	3.904	0.488
9	1280	160	X	X	X	X	X	X	X
10	1440	180	01:02:01:230	210	1680	160	20	13.66	1.708

4.1.2 Quality of Service Parameters

Measured through queuing delay using synchronize timepiece between dispatcher and recipient. This is achieved using Network Time Protocol (NTP); and also to reduce the delay using the Priority Queue algorithm.

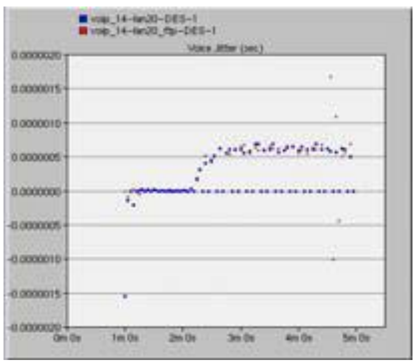
This prevent packet traffic for a voice connection that is not being delayed / dropped from other low priority traffic. It delivers better network service by providing the following features such as: The delay of the VoIP packet can be measured, having a synchronizing clock interface between the sender (TX) and receiver (RX). One way delay variation is defined as the difference in One-way delay experienced by two packets of the same length, where the definition of one way delay variation among any two packets a and b (a ≠ b) is the variance of their One-way delay.

Parameters to be measured include:

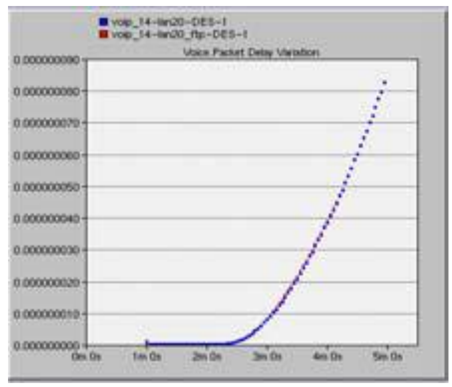
1. **Packet loss** - malfunction of one or more packets to reach respective purpose in entire set of connections which is acknowledged as packet loss. The packet failure depend a lot on packet size; and are related to end-to-end delay and its presence is felt almost in every network. When packet size becomes higher, the delay becomes shorter.
2. **Latency** –average point in time it takes for a packet to travel from the source to destination. A caller on the phone from the source and his/her destination is the listener at the other end.

3. **Delay - t**he total transit time for packets sent in a data stream arriving at the endpoint and it's inevitable in communication system. It is an important factor in determining the quality of a call. An example is echoing that is a major problem caused by delay.
4. **Jitter -** dissimilarity in hindrance occurring among two consecutive packets. Jitter will be mostly experimental since the packets encompass a multiplexed upstream in the network. Mostly occurs when the packets need to be queued. Several types of traffic traverse the IP network.

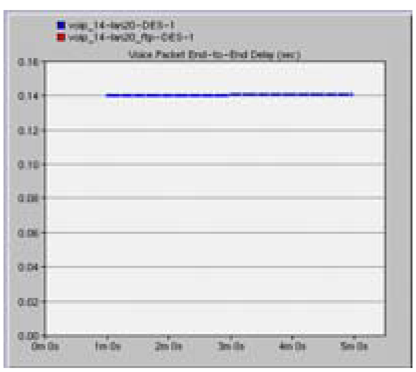
As shown below:



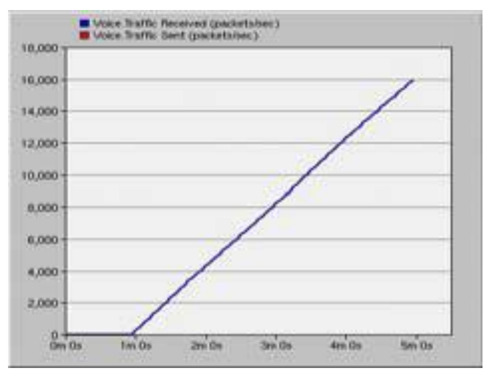
a. Jitter



b. Delay



c. End to End



d.Traffic

The PSTN and VOIP systems are modeled and simulation is done on it. In case of PSTN model of office network of client server model when the traffic is increased quality of voice calls is good. The Simulated time and memory usage taken in case of office network is given in Figure 4.2.

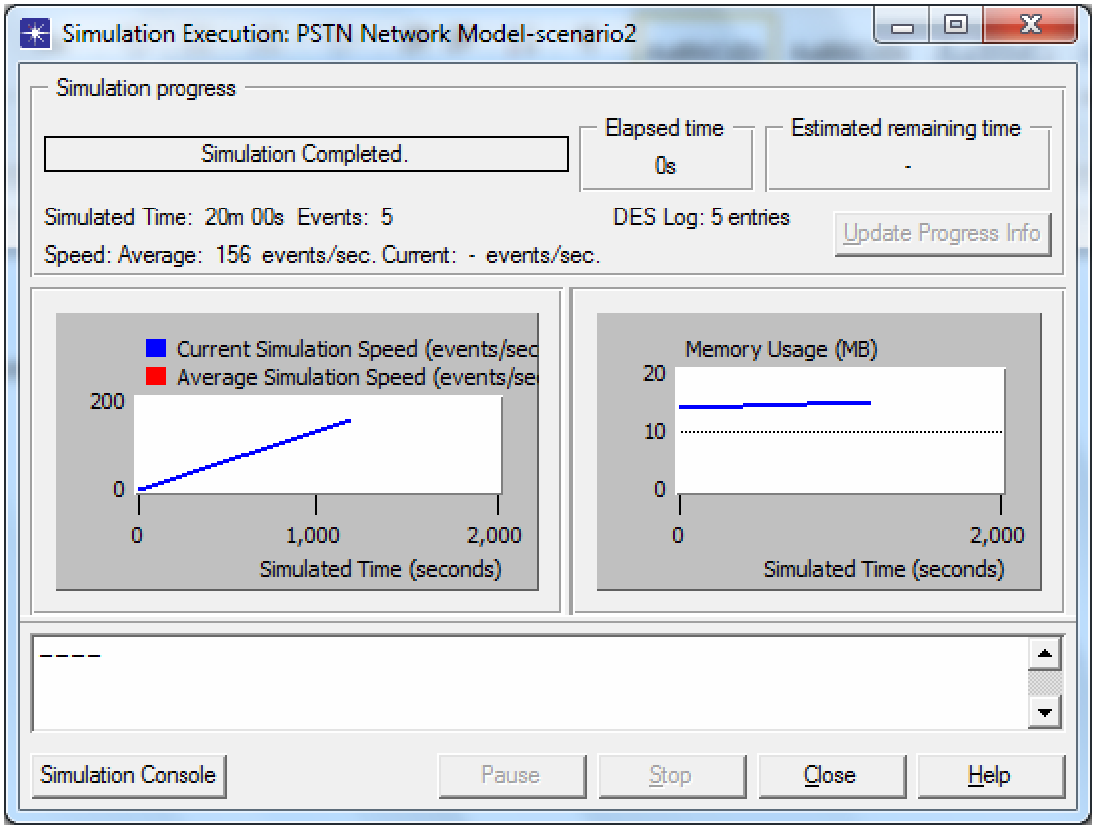


Fig 4.2. Time and Memory Simulation usage of PSTN network.

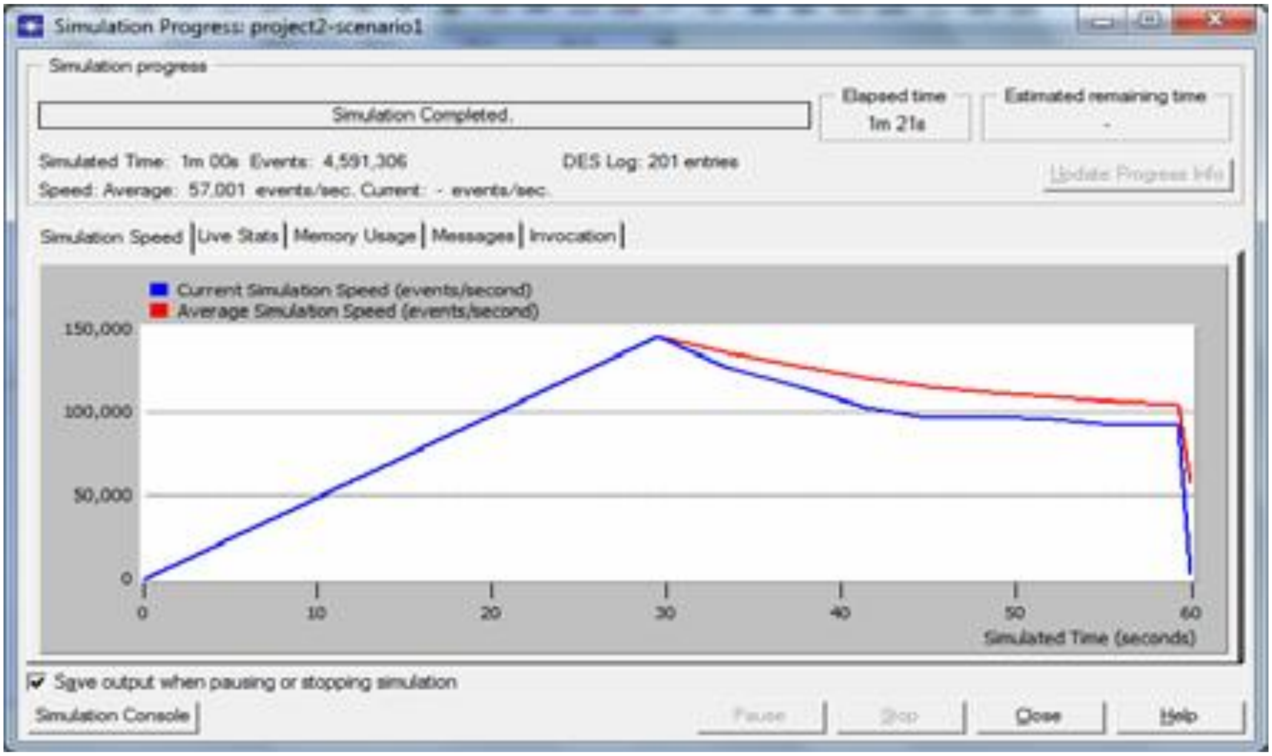
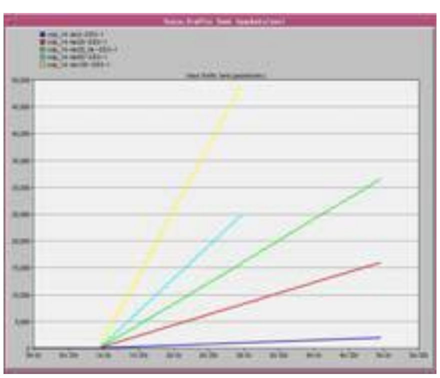


Fig 4.3 Time and Memory Simulation usage of PSTN campus network

From the computer generated tests, the effect of voice signal quality, of main functions of the Public switching Networks and the Voice over IP network; it was found that circuit switching in voice call, while a Voice over IP provides improved packet captured consistently with IP set of connections mechanism. PSTN voice channels have bandwidth and frequency response to fully support a conversational quality voice signal.



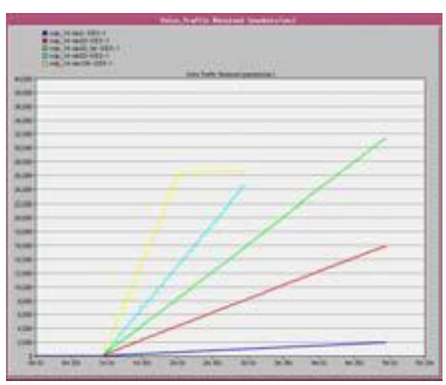


Fig 4.4 Traffic Sent and Traffic Received in VOIP Systems

The above figures show how the traffic transmitted and received with application in unified communication systems behave.. As the VoIP number of clients and calls increases, the jitter, variation, and delay become more significant to the calls quality simulation.

4.1.3 Synchronizing the Timing

Delay is experienced in the network as VoIP packets flow across the network. The nasty nature of the data traffic will make the majority of packets experience a negligible queue delay. This impact is seen in terms of packet traces. Fig. 4.5 (a) shows a one-way network delay typical pattern; which will be measured from a network. Fig. 4.5 (b) is the SSPS corresponding plot monitoring the arrival of packets to harmonize the time.

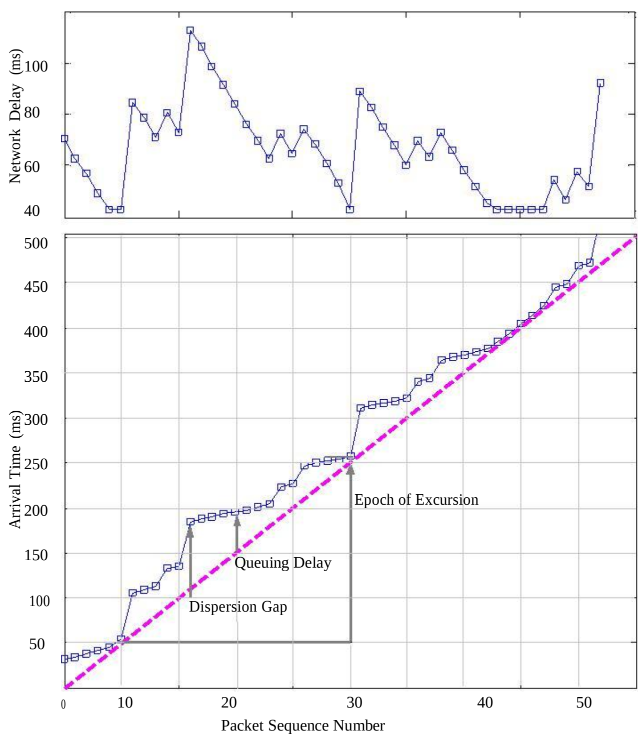


Fig. 4.5. (a) One-way network delay typical pattern. (b) SSPS corresponding plot monitoring the arriving packets to synchronize the timing.

An **epoch of excursion** is found using SSPS in line with the following equation:

$$nD_{pack} - \varepsilon \leq E \leq nD_{pack} + \varepsilon \qquad (4)$$

Whereby a valid epoch of excursion is experienced for a multiple of packetization delays. Our experiments showed There is a margin of error (ε) experienced to allow optimal performance. Without the error margin, it could take a long time to find a complete epoch of excursion.
SSPS find its **dispersion gap** as described below:

$$_D = E - (n - 1)D_{pack} \qquad (5)$$

An Epoch of excursion for the queuing delay (E) is the time at which arrival of packet with a very minute queuing delay and the next arrival packet with the same minute queuing delay. Numbering of packets start at #0 and others #*i*, "where $i = 1, 2, 3, n$. Accordingly, *i* denotes the number of inter-arrival gaps".

Table 3. Summary of the evaluation results of integrating VOIP and PSTN – UC

Number of integrated UC in Experiment	System Failure	System Success
10	1	9

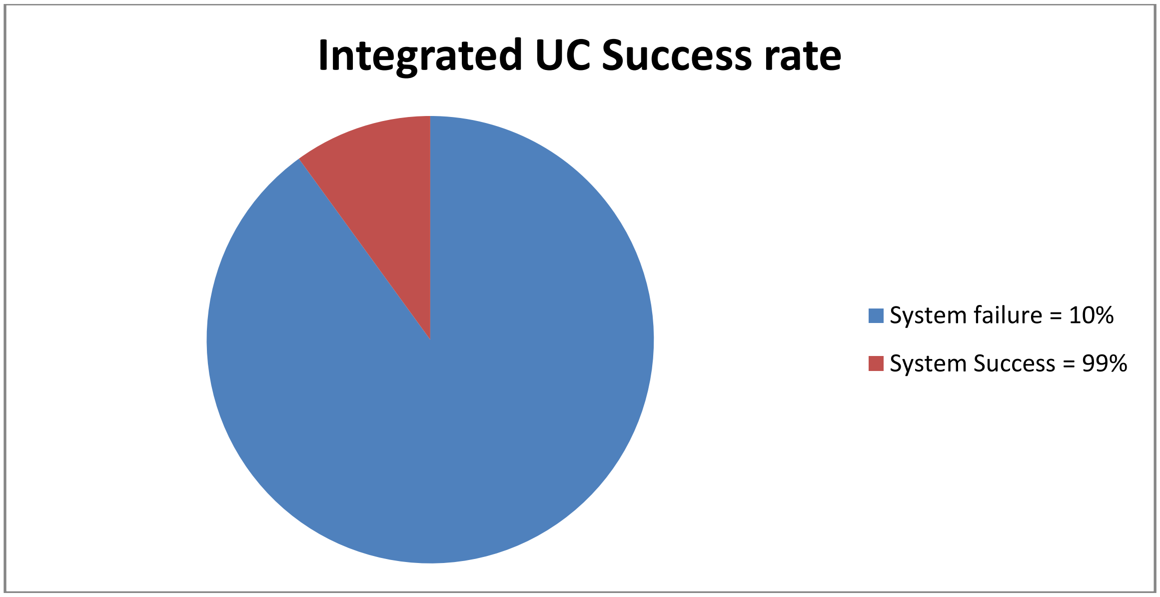


Fig 4.5. Pie Chart showing the success rate of integrating PSTN & VOIP = UC

During the tests carried out in evaluation it was determined reasons why integration may fail to connect VOIP successfully to PSTN phone. These are:-

- Full configuration of voice gateways
- Purchase of licenses from Cisco systems
- Interferences based unavailability of ISDN line
- Availability of internet

Fig 4.7 Conformity of speech transmission for unified Communication

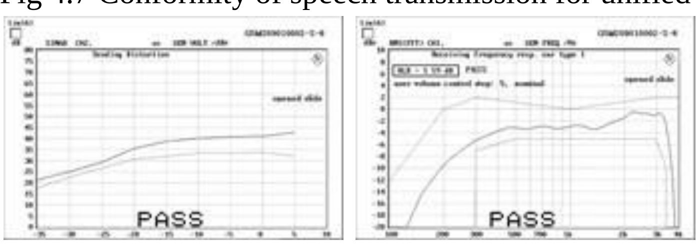
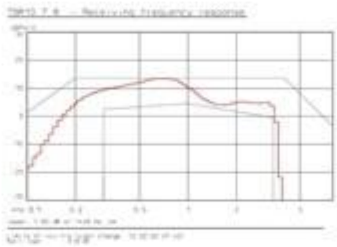
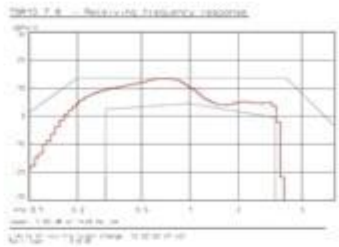
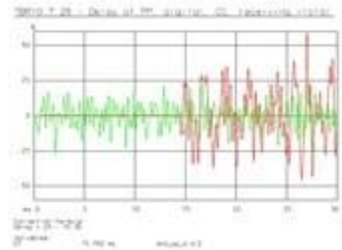


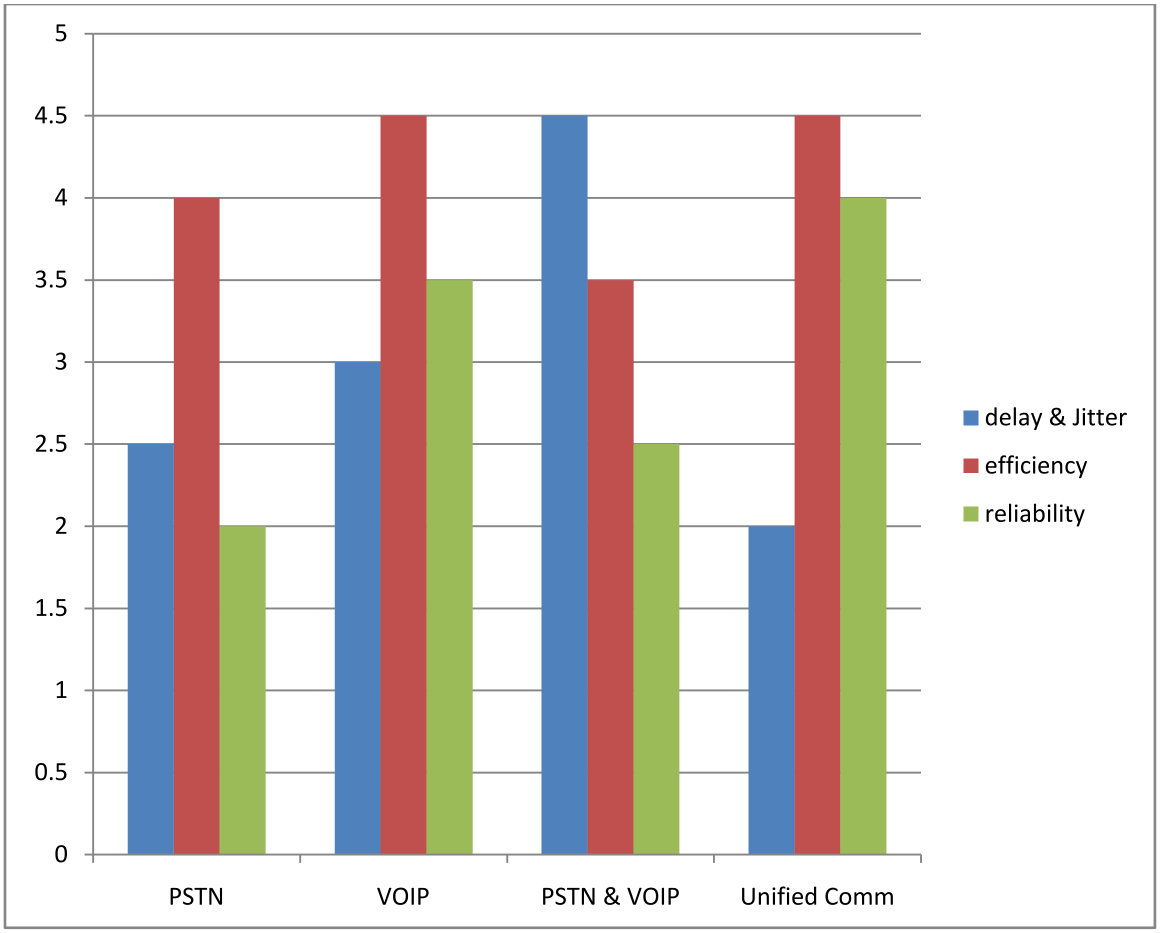
Fig 4.8 The Unified Communication electrical and acoustic characteristics

All the above tests are summed up by having accuracy of various aspects of delay, jitter, efficiency and reliability based on non-functional requirements of unified communication systems as shown in the graph below:

Which compares Rating -vs -Performance

Rating



Performance

Graph 4.2 Performance of various aspects of non-functional requirements

4.1.4 Comparison of Costs

Here I compared the monthly costs on independent PSTN & VOIP, combined PSTN and VOIP systems and a fully integrated Unified Communication network as shown below.

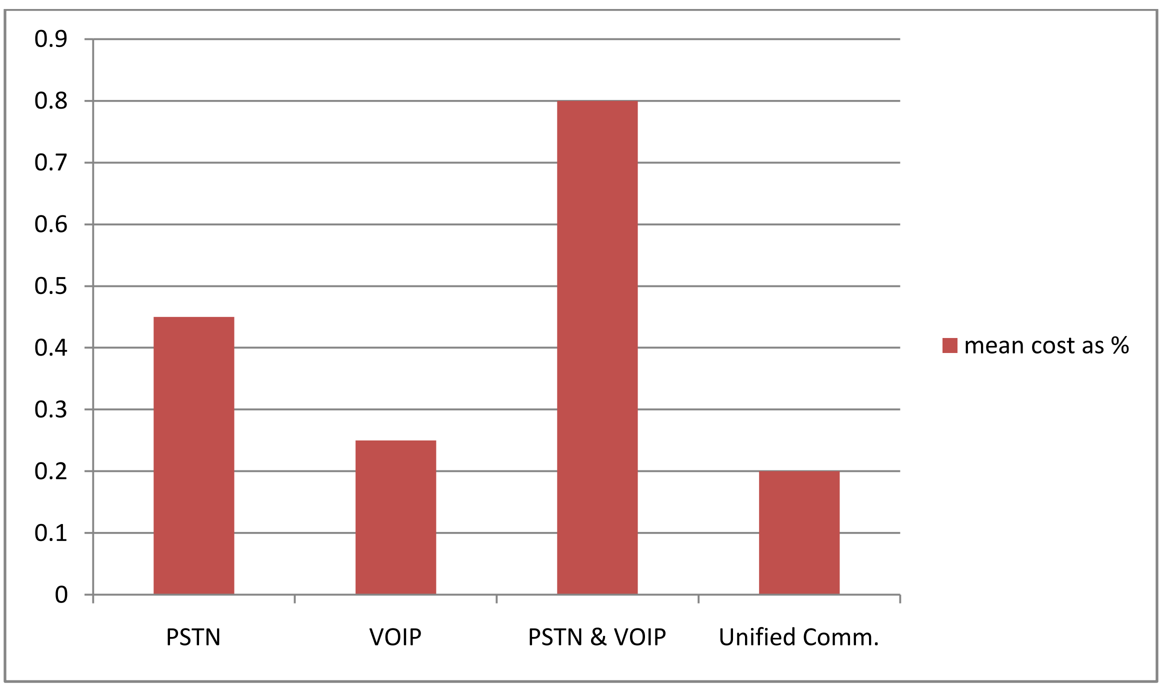
The data is taken from the study area for 6 months starting January 2016 to June 2016 as indicated in the table below: All in times of Kenya Shillings.

MONTH	**JAN**	**Feb**	**Mar**	**Apr**	**May**	**Jun**	**Mean**
PSTN	570000	574000	580000	600000	610000	560000	582000
VOIP	200000	230000	235000	240000	260000	280000	241000
PSTN & VOIP	770000	804000	815000	840000	870000	840000	823000
Unified Comm.	280000	280000	270000	260000	250000	240000	263000

Table 4 Comparison of costs on various systems

The mean is as shown in the graph below:

Percentage – vs. - system



Graph 4.3 Comparison of costs of the systems in the study area

In summary use of Unified communication will have advantage over the others because institution/organizations will saving a lot in terms operational costs and the use available resources influences reduce of expenditures.

4.2 Discussion

This relied on getting the views of domain and technology experts. Respondents are subjected to a real-life situation and evaluating the results. Any gaps or weaknesses can then be addressed in an iterative way.

The integrated voice service tries to combine PSTN and CUCM in order to create a wholesome system that offers enhanced security. It provides the most vital feature most notably the unified communication system that offers the user’s ability to initiate a call from either side.

The integrated unified communication network is a success; given that it is using an available PSTN and a VOIP which is affordable and connectivity once licenses are available.

According to my research objectives, I was able to investigate and analyze the model to determine its applicability, to evaluate and research on the gateways’ types interacting with Cisco UC Manager and describing their differences to come with unified communication, and to propose a solution on methods used to incorporate MGCP, “H.323” and “SIP” gateways with Cisco UC Manager.

The integrated communication strategy and architecture delivers “secure voice mobility application inside an incorporated and intellectual set of connections. The examination supports the employees’ capability to pool resources all moment in time and ubiquitously”.

CHAPTER FIVE

CONCLUSIONS AND RECOMMENDATIONS

It compounds the various findings of the study and proposes recommendations.

5.1 Research Limitations

A number of limitations were noted. First, assembly method techniques are based on the concept of a "method base". This is a repository of parts of a method that perform specific tasks. To effectively use and build a method base, many relevant methods are required. For the case of this project, only three gateways were used to integrate the CUCM.

As noted by one of the expert reviewers, review of methods is a very important undertaking that should involve many experts. These experts should have an obvious interest in the problem domain. Questionnaires are seldom enough. More detailed and technical deliberations are advisable - like seminars and conferences - where a proper critique is guaranteed. This model only relied on questionnaires to validate the model.

In design development, it is important to have a way of measuring applicability or otherwise of the resultant product via measurable metrics.

5.2 Conclusions

The project was indeed a good opportunity to unveil what lies in integrating PSTN and VOIP systems to realize a Unified Communication by using, PSTN, Voice Gateways, VOIP, Internet and Web technology.

The following conclusions were made:

Effective validation of developing an integrated communication network is very critical, if they are to fulfill the purpose for which they are built.

It is essential to have mechanisms of measuring successful applicability of a newly developed design, using actual study areas.

Voice integration enabled easy monitoring and troubleshooting while reducing the cost of ownership and thus increasing employees' efficiency in serving its esteemed customers for now and future needs.

On the basis of the results it was found that Unified Communication systems have enhanced stability and less delay connection than PSTN/VOIP systems connections in parallel.

Voice integration enabled easy monitoring and troubleshooting while reducing the cost of ownership and thus increasing employees' efficiency in serving its esteemed customers for now and future needs.

This research work has been adopted by the University as tender was floated for the provision/migration of analog lines and integration to the existing IP telephony for the University Towers.

5.3 Recommendations for Further Work

The integrated communication strategy and architecture delivers secure mobility of voice applications using incorporated and intellectual network. The service will support the users' needs and capacity to join forces every time and everywhere.

Though selection of the actual technologies to use for systems integration is hazy, will propose a checklist of factors to consider and there is still need for more work in this area.

Provision of metrics to measure the subsequent success or otherwise of the integration initiative is still not clear. This is an area that is important and recommended for further work. It may also be critical to be able to compare the views of process users who may have used other approaches.

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APPENDICES

1A E-QUESTIONNAIRE

1A.1 ICTC Experts

1. Have you ever been involved in an Integration Project? ______________If yes,

   (a). What methodology did you use for analyzing requirement and designing? If Possible, highlight the key steps/activities you went through.

   (b). What challenges did you face?

2. Are you aware of any methodology specifically for integration projects?_____________ If yes, please specify. ____What are Key strengths and weaknesses of the specified methodology?
3. If you were looking for a methodology to help in your systems integration project, What characteristics would you look for in the methodology that conventional? Systems development methodologies do not provide?
4. If you were looking for a methodology to help in your systems integration project, what desirable characteristics would you look for in the methodology that you may also find in conventional systems development methodologies?
5. What application integration technologies (if any) have been applied in your organization or any other you may be aware of? (Please list)
6. Are you aware of the considerations your organization or any other organization may have made in the choice of their particular integration technology?
7. Thank you for your sincere participation.

1A.1.1 Analysis of Responses from ICTC Experts

This questionnaire was sent to 5 IT professionals. Out of this, 5 responses were received. The responses were evaluated and similar ones grouped together.

1. (a). Have you ever been involved in an Integration Project? What methodology did you use for analyzing requirement and designing? If possible, highlight the key steps/activities you went through.

- A fusion of the old waterfall model, prototyping and Rapid Application Development (RAD)
- No specific methodology used

(b). what challenges did you face?

- Users unable to bring out exactly what they want
- Variations in technologies between systems being integrated
- Resistance from users due to existing cultures
- Scope creep due to changes in Requirements
- Very unrealistic timelines
- Communication gaps
- "Internal" interference (vested interests)
- Incomplete requirements due to speed of delivery

2. Are you aware of any methodology specifically for integration projects? If yes, please specify. What are key strengths and weaknesses of the specified methodology?
    - Not aware of any specifically meant for systems integration undertaking
3. If you were looking for a methodology to help in your systems integration project, what characteristics would you look for in conventional systems development?
    - Should provide metrics for measuring the success of integration projects
    - Exhaustive requirements collections to avoid scope creep
    - The ability to identify dearly what is moving from one point to the other including format, urgency and other metrics
    - Allow for change of existing systems to accommodate any required change.
4. If you were looking for a methodology to help in your systems integration project, what desirable characteristics would you look for in the methodology that you may also find in conventional systems development methodologies?
    - Collaboration with stakeholders & developers
    - Implementing functionality based on priority
    - Proper testing
    - Well documented, easy to comprehend and use.
    - Guide integration process in a cost effective manner.

- Harmonization of data formats

5. What application integration technologies (if any) have been applied in your organization or any other you may be aware of? (Please list)
    - RPC and Web services,
    - Direct database connections
    - Infrastructural integration.
6. Are you aware of the considerations your organization or any other organization may have made in the choice of their particular integration technology?
    - Security
    - Reliability
    - Scalability
    - Synchronization requirements
    - Operation standards e.g. XML, EDI ,IATA