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A Model of an Integrated Unified Communication Network Using Public Switched Telephone Network Gateways and Cisco Unified Communication Manager Server – A Case of University of Nairobi, Kenya.

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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. This project identifies the challenges of having a parallel VoIP and PSTN systems, proposes a solution on how to integrate MGCP, H.323 and SIP gateways with CUCM and designs a model to determine its applicability. Unified communication refers to a method of transmission of voice data over IP networks and traditional circuit PSTN (Public Switched Telephone Networks).In this paper the researcher discusses about UC, about the design issues of Gateways, how to measure efficiency, reliability, latency and packet losses at the sender and receiver side and also inter-arrival delay and delay variation (Jitter) ,determining how they are measured using round trip delays for IP-SLA (Service level agreement).In this paper, two telephone networks namely traditional Public-Switched Telephone Network (PSTN) and Voice over Internet Protocol (VoIP) are extensively examined to come up with Unified Communication by means of measurements on our University voice networks, modeling and actual experiments. Results show that the unified communication is more efficient in terms of performance in the current technology.*

General Terms: *Unified Communication, UC.*
Keywords: *Cisco Unified Communication Manager (CUCM), Public Switched Telephone Network (PSTN), Media Gateway Control Protocol (MGCP), Session Initiation Protocol (SIP), and H.323.*

I. Introduction

To place external calls, Cisco Unified Communications Manager (CUCM) deployments need a connection to the public switched telephone networks (PSTN). PSTN connections are provided through gateways, which connect traditional time-division multiplexing (TDM) telephony interfaces (digital T1/E1 or analog FXO ports) and VoIP domains. Gateways can be integrated in Cisco Unified Communications Manager by using protocols such as Media Gateway Control Protocol (MGCP), H.323, and Session Initiation Protocol (SIP) for signaling on VoIP call 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.

II. Literature Review

Here my main concern is overview of PSTN and CUCM; and thereafter presenting the unified communication network. 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 support multiple gateway protocols. In this research, gateway protocol selection will be based on the capabilities of the communication network available at the UoN. (H.323, SCCP, SIP and MGCP). 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.

III. Methodology

The goal was to come up with a selection of the study area advised by the need to identify a communication domain that could reasonably be used for integration model for unified communication networks. That has a domain where there is a lot of interaction/comm. within and without the institution either manual or automated and where different platforms of VOIP and PABX legacy phones interact.

1.1 The System Architecture

It consists of CUCM that is used to configure voice gateways (MGCP, SIP, and H.323), PSTN from the service providers through the PABX to connect the legacy/analog phones and VOIP networks using internet for connection. The proposed model will therefore ensure integration of both PSTN and VOIP is realized for a full unified communication network.

The system delivers fully integrated communications, converging voice, video, and data over a single network infrastructure using standards-based protocols. It delivers unparalleled performance and capabilities to address current and emerging communications needs in the enterprise environment, as illustrated by the network topology in the figures below.

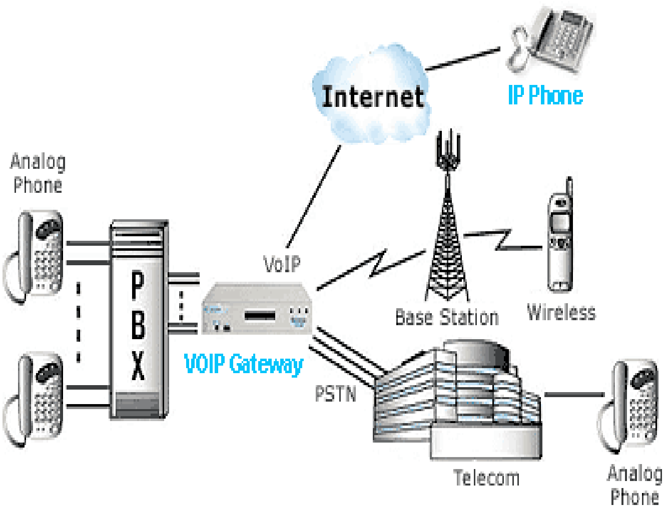


Fig 1. The model Design

This is how the system will be staged:

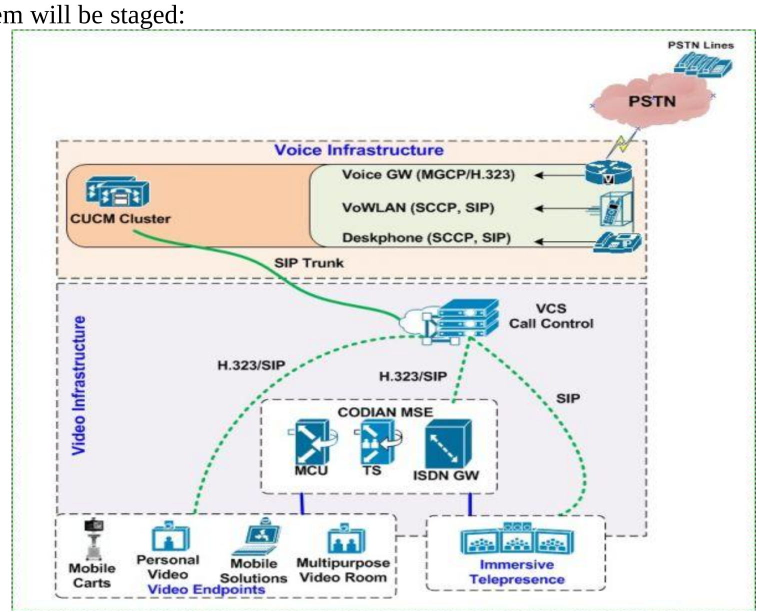


Fig 2. Deployment platform of the gateway

3.2 Integrating MGCP gateways with CUCM

MGCP is a plain-text protocol that call-control devices use to manage the IP telephony gateways. It is a client/server protocol that allows the call agent (CA) to take control of a specific gateway endpoint (port).It has the advantage of centralized gateway administration in the CUCM. CUCM controls the state of each port on the gateway endpoint. The gateway can be controlled per endpoint (TDM port) level but H.323 and SIP cannot. MGCP: Centralized dial plan configuration and Centralized gateway configuration hence it will be easy to implement in a large SP network.

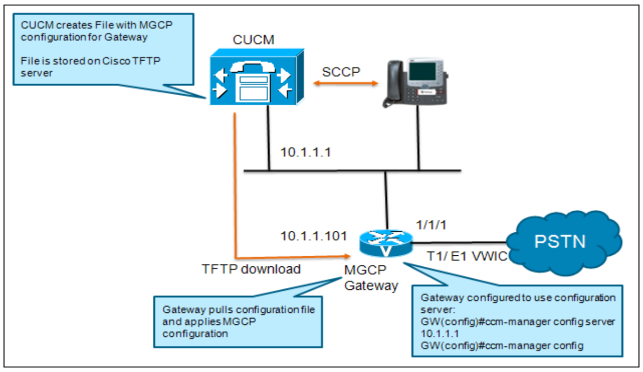


Fig 4. MGCP configurations sever communication

3.3 Integrating H.323 gateways with CUCM

When calls are made from IP Phone to the PSTN, the dial plan configuration on the CUCM must re-direct the call to the H.323 gateway and in turn the call will be routed to the PSTN carrier. This is configured at Main Campus-Law-of-the-Sea.
Step1: In CUCM Administration Page, choose Device > Gateway
Step2: Click Add New and select Gateway Type > H.323 Gateway
Step4**:** Enter the Device Information as shown here, Device Name, Description and Device Pool
Step5: Scroll down and enter the Call Routing Information - Inbound calls & outbound calls
Step6: Click > Save and Apply Config
Configure Basic Cisco IOS H.323 functionality (interface loopback0, IP address and H323 gateway VoIP interface and VoIP dial peer config.

1.4 Integrating SIP gateways with CUCM

SIP Gateways are integrated with CUCM by using SIP Trunks provisioned from CUCM. A gateway is a device that can translate between different types of signaling and media. In CUCM, configure SIP gateway, SIP on Cisco IOS router functionality and call routing info. It's integrated with CUCM by using SIP Trunks provisioned from CUCM, and translates between different types of signaling and media. Enter CUCM and add SIP trunk as shown below:

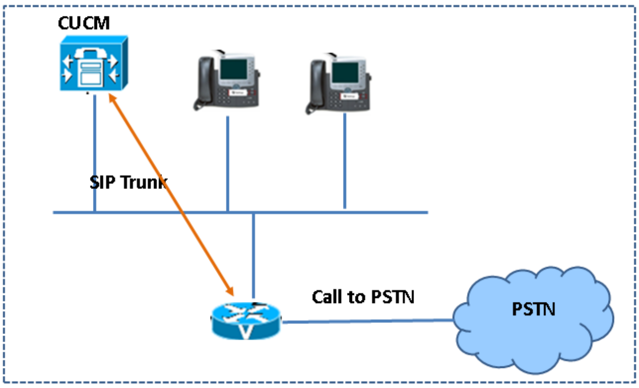


Fig 3. SIP Trunks

IV. Results and Analysis

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. Figure.6 shows the traffic sent and received with the voice Application in VOIP systems. For simulation purposes the traffic received is consistent with the traffic sent and packets are lost when the numbers of calls get too high. As the number of VoIP clients and calls increases, the voice jitter, delay variation, and ETE delay become significant factors to the calls quality simulation, packets are lost when the numbers of call gets too high.

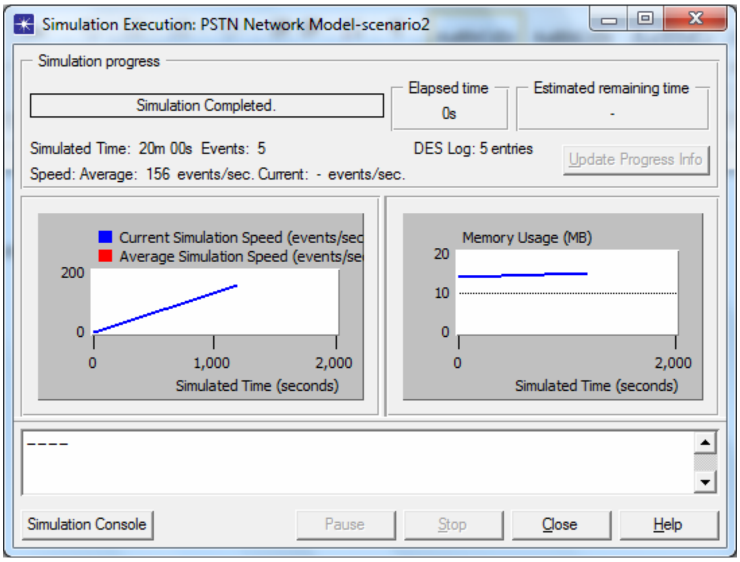


Fig 5 Simulation Time and Memory usage of PSTN network

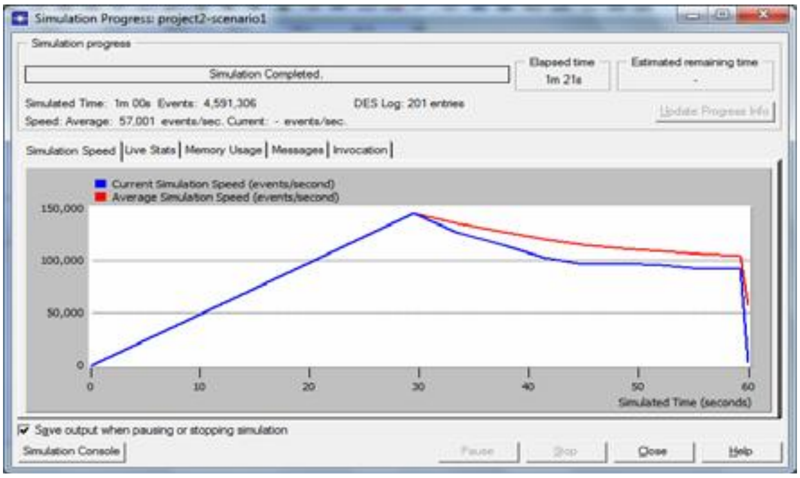


Fig 6 Simulation Time and Memory usage of PSTN campus network

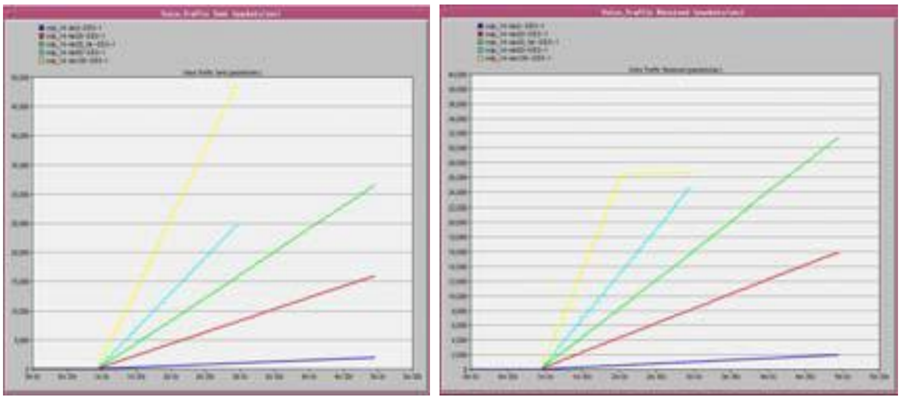


Fig 7. Traffic Sent and Traffic Received in VOIP Systems

4.2 On SIP VOIP test

Offers VoIP-oriented testing tool which can offer valuable information about the way your broadband connection handles simulated VoIP traffic. The one click test provides an easy-to-read graph rating and summary of the connection's performance. The result offers the usual metrics of speed, jitter, and latency, but also gives you a Mean Opinion Score (MOS).

Graph 1. On Sip VOIP Test

4.3 Unified Communication End-to-End Delay Measurement

To measure packet losses at the sender and receiver side and also inter-arrival delay and delay variation (Jitter), how the receiver and sender sends reports about the sender, how one-way end-to-end delay is measured, and explains the functionalities of static and dynamic jitter buffers and describe how Cisco routers are used to measure packet losses and round trip delays for IP-SLA (Service level agreement).

4.3.1 Time Stamp Calculations

Time stamp calculations basically depend upon the type of application we use for voice only.

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

Table1. Sender 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

Table 2. Receiver Timestamps

Calculation at the Receiver Side

When a receiver receives the packet, the timestamp of the received packet is according to equation below.

$$\text{Receiver timestamp } Rec(i) = \frac{Rec(i) \text{ in times of time unit x sampling frequency}}{1000}$$

Where, Rec(i) in terms of time unit can be calculated by equation below, sampling frequency of voice is 8000Samples/Sec

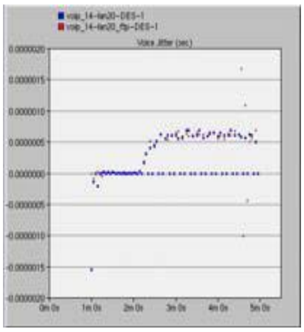
Rec(i) in times of time unit = (arrival time of packet (I) – arrival time of packet(i-1)) + Rec(i-1) in times of time unit.

This gives a resultant - 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 contains a timestamp that informs the receiver at which time the received data in the packet should be played back.

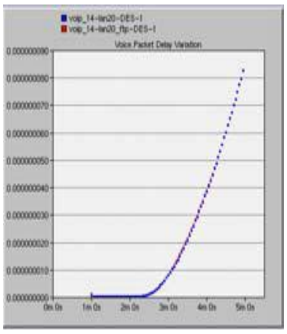
Other Parameters to be measured for analysis include:

1. **Packet loss** - defined as Failure of one or more packets to reach their destination across the network is recognized as packet loss. The packet loss depends a lot on packet size. Packet loss is related to end-to-end delay and is present in almost every network and is extremely noticeable with real time streaming technology such as Skype and online gaming. If packet size becomes higher, the delay becomes shorter. Packet loss can severely degrade the voice quality. Higher packet loss can be reduced by decreasing packet size.
2. **Latency** - the average time it takes for a packet to travel from its source to its destination. A person who is speaking into the phone called the source and the destination is the listener at the other end.
3. **Delay - t**he total transit time for packets in a data stream to arrive at the endpoint and it is inevitable in communication system. Delay time is one of the most important factors in determining the quality of a call. Echoing has been a major problem that is caused by delay.
4. **Jitter -** variation in delay that occurs between two successive packets. It is mainly observed because the packets have to be multiplexed upstream in the network. It occurs when the packets need to be queued. Several types of traffic traverse the IP network.

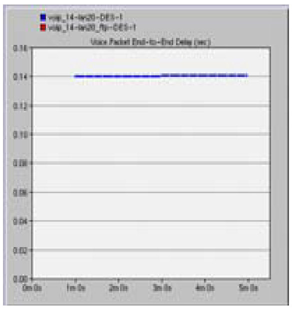
   We assume that there is enough bandwidth to meet the demands of both the Unified Communication connection and FTP traffic. Therefore, the FTP packets are not dropped and the results are shown in Figure.7 below:



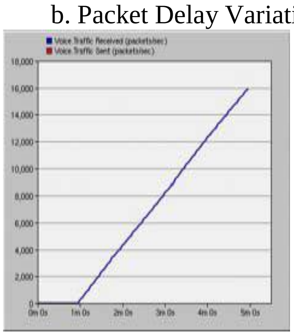
a. Voice Jitter



b. Packet Delay Variation



c. Packet End to End Delay



d. Traffic sent and received

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

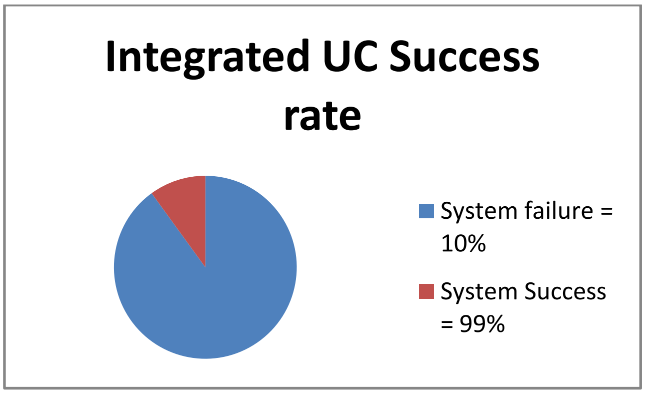
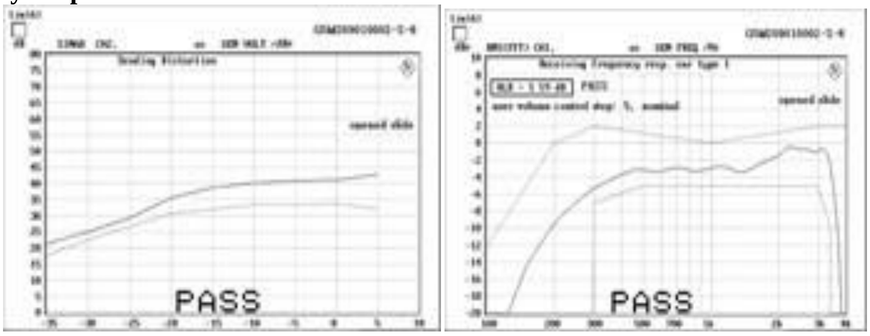


Fig 8 Pie Chart showing the success rate of integrating PSTN & VOIP = UC

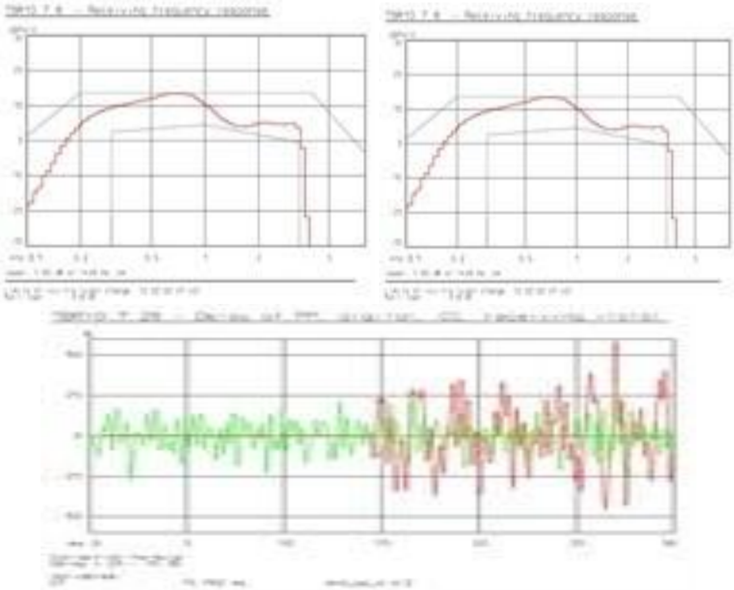
During the tests carried out in evaluation it was determined with reasons why integration may fail to connect VOIP successfully to PSTN phone. These are:-

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

4.3.2 Conformity of speech transmission for unified Communication

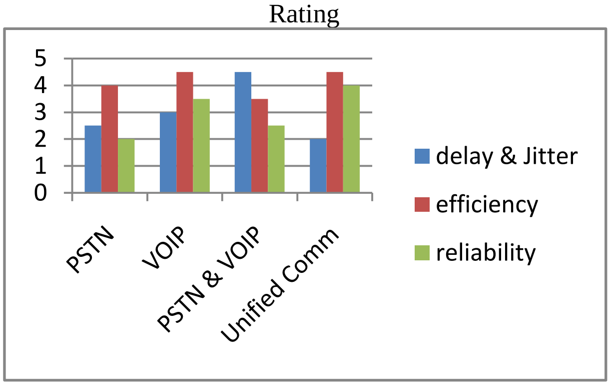


4.3.3 The Unified Communication electrical and acoustic characteristics



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 below: Rating-vs-Performance graph.

Rating



Performance

Graph 2 Performance of various aspects of non-fuctional requirements.

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 types of gateways that can interact with CUCM and describe their differences to come with unified communication, and to propose a solution on how to integrate MGCP, H.323 and SIP gateways with CUCM.

The integrated communication strategy and architecture delivers secure voice mobility applications within an integrated and intelligent network. The service supports the employees' ability to collaborate every time and everywhere.

V. Conclusion

The project was indeed a good opportunity to unveil what lies in Communication by using, PSTN, Voice Gateways, VOIP, Internet and Web technology. 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 UC systems have a more stable 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.

1.5 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 method 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 Further Work

The integrated communication strategy and architecture delivers secure voice mobility applications within an integrated and intelligent network. The service supports the employees' ability to collaborate 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 the process users who may have used other approaches.

Acknowledgment

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