VOICE NETWORK FOR AVIATION DATA NETWORKS

A Thesis by

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I have examined the final copy of this thesis for form and content, and recommend that it be accepted in partial fulfillment of the requirements for the degree of Master of Science with a major in Electrical and Computer Engineering.

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Ravi Pendse, Committee Chair

We have read this thesis and recommend its acceptance:

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DEDICATION

To my parents and my family who have supported me throughout my life. Without their help, it would not have been possible to finish my thesis.

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ABSTRACT

Airline companies are always looking for various services they can offer to differentiate them from others and increase there market share. On the other hand passengers in the aircraft always want to be in touch with the rest of the world through phone or internet to make there time more productive. The drive of gaining market share by airline companies and advances in wireless and internet technology has made voice communication possible from the aircraft. Present technologies offer cell phone access or allow voice communication over internet through satellite link. The major disadvantage they pose is the interference of cell phone transmission with navigational system of aircraft and the cost of implementation. Present research work attempts to address the above problem by designing an IP based network which is capable of connecting travelers from aircraft to the rest of the world.

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LIST OF ACRONYMS

CoA .... Care of Address

CME .... Call Manager Express

DSL .... Digital Subscriber Line

DCS .... Disengage Confirm Signal

DTMF .... Dual Tone Multiple Frequency

DHCP .... Dynamic Host Control Protocol

FA .... Foreign Agent

GRE .... Generic Routing Encapsulation

GSM .... Group System for Mobile Communication

HA .... Home Agent

IOS .... Internet Operating Software

MGCP .... Media Gateway Control Protocol

MGC .... Media Gateway Controller

MN .... Mobile Note

PCS .... Personal Communication System

PSTN .... Public Switch Telephone Network

PCM .... Pulse Code Modulation

RTP .... Real Time Protocol

RCM .... Registration Confirm Message

RRM .... Registration Request Message

LIST OF ACRONYMS (Continued)

SCCP...Skinny Client Control Protocol

SDP ....Session Descriptor Protocol

SIP...Session Initiation Protocol

TCP .... Transmission Control Protocol

TFTP .... Trivial File Transfer Protocol

UDP...User Data Protocol

VoIP ....Voice over Internet Protocol

WAN...Wide Area Network

WLAN..Wireless Local Area Network

CHAPTER 1

INTRODUCTION

1.1 Background

Public Switched Telephone Network (PSTN) is the earliest form of voice traffic engineering and is presently the most broadly interconnected communications system in the world. This network system uses circuit switching technology (Davidson, 2000) to communicate between the two ends, as a result of this when a call is made a dedicated bandwidth is reserved for that call which results in a poor utilization of resources. The other major disadvantage of a traditional PSTN infrastructure is that it is designed specifically to support only voice communications.

The Defense Advance Research Project Agency (DARPA) around mid-1970 developed a packet switch network called ARPANET (Arpanet, 2000) which used conventional point-to-point leased lines. The goal of this technology was to create a shared network having multiple paths from source to destination with no central point of failure. This technology later became the base of the present day communication infrastructure called the Internet. The technology utilizes the resources more efficiently by transferring data in form of packets and sharing a single path among multiple users.

Because of the advantages Internet had to offer a lot of development efforts were put in this field. As a result of these efforts the internet technology now has become the most widely deployed protocol in the world. Moreover, the latest developments in the networking field are making internet technology more reliable and fast by increasing its bandwidth and speed respectively. As the internet is becoming increasingly more reliable there has been considerable interest in transporting voice over these data networks. Although internet was not intended for transporting voice, the keen interest has led to the development of a whole new technology called as Voice over Internet Protocol (VoIP) (Jonathan, 2006).

VoIP can be defined as routing of voice traffic over a shared medium which can be the internet or any IP based network. This technology presently has become extremely attractive because of many advantages it has to offer. One of the key advantages of the technology is monitory savings as voice uses the same network as data. Thus enterprises do not have to maintain a separate PSTN for voice calls and an IP based network for data communication, secondly this technology also offer's long-distance calls at a relatively lower price compared to the normal PSTN call, maintaining the same type of reliability and quality of service offered by PSTN technology.

Today's fast paced world has become increasingly mobile requiring instant access to information any were and any time. Wireless technology helps addressing this problem by allowing users to be connected while they are on move. After development of ARPANET, Defense Advance Research Project Agency (DARPA) explored the possibility of sending packet over radio signals. This research lead to the development of first wireless data network called ALOHANET (Abramson, 1985). The technology allowed people in different locations to access the main computer systems through the use of radio signals thus proving that it is possible to connect by using air as a medium. This concept sparked great interest in networking industry which lead to the expansion in the use of wireless technology in a way that today most wireless technologies such as satellite-based networks, cellular networks, wireless local area network (WLAN) became an integral part of communication infrastructure.

Researchers have expanded the use of wireless technologies and developed new techniques in the area of aviation data networks which allows passengers on an aircraft to access internet. One of the basic techniques used to provide this service is to place a network operations center in a central location and direct traffic from all the aircrafts to this location via satellite connection. Boeing is the first company to use this approach, and deploy commercially its own technology called as Connexion (Boeing, 2005) which provides internet access on aircrafts. As bandwidth of the satellite links improve, internet technology for aircrafts are becoming more and more stable. Taking advantage of this technologist in airline companies are researching the possibility of passing voice over these networks.

One technology used to deliver voice traffic from the aircraft to the ground is designed in such a way that it allows the use of cell phone and other wireless devices on aircrafts. As these devices send and receive the voice signals through radio waves they pose a risk of interference with the navigational system of the aircraft. Few other network designs tend to take advantage of the present internet infrastructure by allowing signaling and voice traffic to go through the internet with the help of a satellite link. However the biggest drawback of this network design is the cost of the calls made.

The present research work attempts to develop an IP based network which not only allows passengers to access internet but also gives them the freedom to talk through IP/soft IP phones to the rest of the world at a much lower cost. This is achieved through mobile IP (Perkins, 1996) technology integrated with Cisco voice servers and routers which provides seamless wide area network (WAN) connectivity to the aircraft through satellite link to the company's intranet. Thus all the voice traffic instead of passing through the service provider network is passed through the WAN link to the company's intranet network and then passed to the PSTN network. As this network design does not rely on radio waves it is much safer and does not pose any risk on the aircrafts navigational system. This design also eliminates the use of cell phones thus reducing the cost of implementation further.

In this research work the author intends to design an efficient network as well as investigate the affect of mobility on a) call setup time between the IP phones present in the aircraft and the ones on the ground b) quality of voice on this network and c) to determine what kind of codec is best suited for this network.

1.2 Thesis Organization

The thesis is organized into six chapters. Chapter 2 presents an overview of the literature that was reviewed for design. Chapter 3 focuses on how aircraft data networks work and how mobility protocol helps in providing seamless connectivity to these networks. In Chapter 4 the author explains the basic proposed voice network design for aircrafts and also discusses about the various network topologies (keeping the proposed model as base) considered to develop an efficient voice network. Chapter 5 presents the test bed and experimental setup which was employed for this research and explains the results obtained for this experiment. Chapter 6 concludes by explaining the results and future work that can be done to enhance the network even further.

CHAPTER 2

LITERATURE REVIEW

In this chapter author reviews relevant topics, books & papers that helped to lay the foundation for this research work. The chapter is divided into three sections. First section gives a comprehensive summary of different kinds of VoIP protocols and the issues around them. The second section explains about the software's such as Cisco Call Manager and Cisco Call Manger Express which are used to control the voice signaling over the enterprise voice network. The third section discusses about the various wireless networks and how this technology has helped in developing a new field in the networking world called as aircraft data networks.

2.1 Voice Communication

The emergence of high speed Internet technology, complex voice supporting protocols has encouraged internet technologist to implement and develop a technology which allows voice traffic and video traffic to go through the already existing data IP network. This section explains the different advancements accomplished towards developing a VoIP network and relate the relevance of current research in this field.

2.1.1 Traditional Voice Network

Majority of the present day voice communication network still use traditional PSTN network to switch calls from a source to destination. In the PSTN technology analog voice signals are first encoded into digital stream of 1s and 0s by using Nyquist theorem and are passed through a pair of copper wires known as local loop. This loop connects the home telephone to the Central Office (CO). CO decides through the dual tone multi frequency (DTMF) signaling were the call is destined to and sends the data to a different central office which is connected to the local loop of the destination through a trunk link. Although the PSTN networks are efficient and do a good job in switching voice calls , many businesses are changing to a new network, where voice is an application on top of a already wide spread data network. This transition is taking place because of following advantages VoIP has to offer over PSTN network.

- The technologies such as Ethernet which were capable of taking only data packets are now able to handle voice packets effectively. As a result of this organizations can have only one infrastructure for both voice and data thus reducing the cost.
- The circuit-switched network calls require a permanent 64-kbps dedicated circuit between the two telephones. Whether the caller or the person called is talking or not, the 64-kbps connection cannot be used by any other party. This means that the telephone company cannot use this bandwidth for any other purpose thus billing the parties for consuming its resources. In comparison, the VoIP protocol allows the bandwidth to be shared thereby reducing the cost of the call.
- In analog line one cannot have data access (Internet access), phone access, and video access across a 56-kbps modem. High-speed broadband access, such as digital subscriber line (DSL), cable or wireless is needed to enable this convergence. As the bandwidth issue for the IP network is resolved it is now possible to converge all these technologies.

2.2 Voice over Internet Protocol (VoIP)

VoIP is a packet switching technology which allows voice information to be passed over the present internet data infrastructure. In order to send the voice information on the data network, first voice signals which are in an analog form has to be converted into packets. This is done through coder-decoder (codec). The codec converts audio signal into a compressed packet form for transmission and then decodes the compressed packet back into an audio signal for replay. More detail information about the working is discussed in the upcoming sections.

2.2.1 Working of Voice over Internet Protocol (VoIP)

Since voice signal is analog in nature it has to be first converted into digital signal .This is achieved by sampling the voice signal using Nyquist theorem and once the signal is sampled it is quantized through different quantization technique to form a digital signal. The digitized signals are transmitted over the packet switch network in a efficient manner, thus enhancing the speech quality.

The components used for sending the digitized signals in an efficient manner are codecs. The codecs not only encodes the digitized signal at the transmitting side but also decodes the encoded traffic at the receiving end. There are basically three coding schemes which can be used to encode the digital data

- Waveform codec
- Source codec
- Hybrid codec

2.2.1.1 Waveform Codec

This is the most simplistic form of codec types. Here the codec reconstructs input signal without making much assumption about the type of input signal. As a result of this the input wave form is re-created regardless of whether the input is speech, music or other noise. Example of this kind of codec is a) Pulse Code Modulation (PCM) codec specified in ITU-T recommendation G.711 (ITU-T Recommendation G.711, 1988) b) Adaptable Pulse Code Modulation (AD-PCM) which is specified in ITU-T recommendation G.726 (ITU-T Recommendation G.726, 1990)

2.2.1.2 Source Codec

This type of codec is used for specific input type such as human voice. While constructing an input signal these codec's try to replicate the physical process of sound created by the body**.** During the speech conversation the excitation signals heard and produced by lungs and vocal cord are filtered by the vocal tract. These excitation signals and un-voiced sounds are grouped into frames and analyzed by the codec. The codec then further encodes it in a single bit indicating whether it's an excitation signal (or) an un-voiced excitation and sends it to the destination. The destination re-constructs the original signal frame-by-frame by passing the excitation signal through the filter for that frame.

2.2.1.3 Hybrid Codec

These codecs are the most complex form of codec but they provide better speech quality by using a combination of source modeling & waveform analysis. The codec generates the excitation signal in the same way as the source codec but encodes the excitation signal in the more sophisticated manner. As a result of this they have higher delay associated with it. They are three different ways in which it encodes the excitation signal.

- **MPE:** Here the excitation signals are encoded as a series of non-zero pulses which vary in position and amplitude (Davidson, 2000).
- **RPE:** In this type of model also the signals encoded as a series of non-zero pulses but here only the amplitude of pulse is specified (Davidson, 2000).
- **CELP:** This model uses a completely different approach. It maintains codec books were all the combination of pulse amplitude and positions are represented as index values. The sender sends the index value of the excitation signal which is the best match in the code book. The receiver which shares the same code book compares that index value in the

code book and extracts the excitation signal. This approach is very efficient as it allows sharing a lot of information in very few bits. (Davidson 2000)

Once the analog signal is converted into packets they are sent trough the IP network infrastructure to the destination. But in order to send the voice packets to the destination the network has to know about the destination and connect to the destination before hand. In order to solve this issue different kinds of protocols have been developed

2.2.2 Voice over Internet Protocol (VoIP) Protocols

In voice world two types of VoIP protocols exist:

- Protocols that provide the call control and signaling
- Protocols that carry voice payload (RTP, RTCP, UDP, and IP)

The job of the call-control and signaling protocols is to set up and tear down the connection between two or more endpoints in a network. The following is a list of the call-control and signaling protocols commonly found in voice network:

- H.323 (peer-to-peer model)
- Session Initiation Protocol (SIP)
- Media Gateway Control Protocol (MGCP)
- Skinny Client Control Protocol (SCCP)

2.2.2.1 H.323 Protocol

H.323 (Alexander, 2005) is an umbrella that includes the protocols which provide audio-visual communication sessions on any packet network. These protocols support four different kinds of components, which when networked together, provide the point-to-point and point-to-multipoint multimedia-communication services. These components are

- **Terminals**: An H.323 terminal can be a computer or a stand alone device which is capable of running multimedia application.
- **Gateway:** It is a device which is used to provide connectivity between a H.323 network and a non-H.323 network. Commonly these devices are routers.
- **Gatekeepers:** They are the focal point of voice calls in the VoIP network. These devices not only provide services such as addressing, authorization and authentication of terminals and gateway's but also manages the bandwidth of the network
- **Multipoint control units:** These units basically provide support for conference of three or more H.323 terminals.

As stated above, that H.323 is an umbrella of protocols listed below are the protocols under the umbrella.

- RAS (H.225) signaling
- Call Control/Call Setup (H.225)
- Media Control and Transport (H.245) signaling

H.225 RAS Signaling

It is a signaling protocol mainly used between the gateways and the gatekeepers. Here all the gateways in the network first register with the gatekeeper by informing the gatekeeper about their IP addresses. After the registration process admission process takes over the control. In this process the gatekeeper checks for the available bandwidth in the network and accordingly gives a confirmation or rejection of an admission request. Lastly with the registration admission status (RAS) channel gatekeeper can check the status of the end points that is whether they are on-hook or off-hook.

H.225 Call Control & Call setup signaling

Once the gatekeeper agrees on admitting the call through registration admission status (RAS) channel H.225 (Alexander, 2005) call control signaling helps to setup the connection between the H.323 endpoints. The setup creates a reliable Transmission Control Protocol (TCP) call control channel .This channel can be created through the gatekeeper or directly between the end devices.

H.245 Signaling

After the H.225 channel setup it is the H.245 (Alexander, 2005) signaling protocol that handles end-to-end control messages between H.323 end points. These control messages conveys the information such as encryption, flow control, jitter management, preference requests which are the essential parameters used for multimedia communication between the H.323 end devices.

2.2.2.2 Session Initiation Protocol (SIP) Overview

Session Initiation Protocol (SIP) (Ubiquity, 2004) is also used during setting up and tearing down of voice or video calls. One of the most important feature of SIP is that it does not define the type of session that is being established but is only involved in the signaling portion of a communication session. The session is basically defined by session description protocol (Ubiquity, 2004) which provides information on IP port and codec used. As a result of this the protocol can be used for many applications and services such as video and web conferencing. It utilizes four major components to deliver messages embedded with the session descriptor protocol (SDP) to establish a session. These components are user agents, registrar servers, proxy servers and redirect servers

The user agents are end devices such as cell phones, personal computers etc. which initiates the message and are used to create and manage the sessions. These messages are responded by the user agent server. These servers are like a database that contains the location of all user agents within a domain. Their main job is to retrieve and send participants IP addresses and other important information to the proxy server. The proxy servers when receive a query from the user agent, they query the registration server for the information as stated above. After receiving the information from the server the proxy server forwards the invitation directly to the user agent if it is in the same domain or sends to a redirection server which in turn sends the request to another proxy server located in a different domain.

2.2.2.3 Media Gateway Control Protocol (MGCP)

This protocol acts like a bridge between the circuit switch network and the packet switch network. The protocol accomplishes this by converting the media signals between the circuit switch and packet switch networks so that they are compatible with each other. Unlike H.323 protocol which is a peer-to-peer this is more like a client-server protocol. It is based on the concept of a media gateway controller (MGC) (Anderson, 2003) were the media gateway controller acts like a call signaling and media control signaling source for end devices which are basically slaves to the gateway controller. In the protocol terminology the controller is called as a call agent and the end devices which are called as the media gateway can be different types of devices such as VoIP gateways, network access servers etc.

The protocol functions by converting and sending digital signals from the PSTN network to the call manager (brain of the voice network) (Alexander, 2005) in such a way that the call manager understands the signal .After this the call manager directs the gateway to either establish, modify, and tear down media connections. The manager sends this notification through UDP (Postel, 1980) .One of the major disadvantageous of this protocol is that it is not designed to handle multimedia calls.

2.2.2.4 Skinny Client Control Protocol (SCCP)

Skinny Client Control Protocol (SCCP) is a Cisco proprietary protocol based on client/server model. The call signals which are originated from the client are transmitted through TCP connection which is maintained to the server by using port number 2000. After the appropriate connection is established between the end-points the voice packets are passed through different protocols such as Real Time Protocol (RTP) which work on UDP protocol (Postel, 1980).

This protocol is used between the Cisco call manager & the IP phones mainly because it is very light on the system and use very less central processing unit power. When the client makes a call it maintains a TCP connection with the call manager which in turn handles the call establishment for the client. Once the call is established the end stations use connectionless communication protocols such as UDP for audio transmissions. However sending the voice packets over the infrastructure initially designed for data networks poses some drawbacks.

2.2.3 Issues with Voice Networks

The switching technology was initially developed for transmitting only data information in the form of packet. Since the data information is not real time sensitive these networks were designed to provide best effort services. On the contrary voice traffic has to be real-time, thus sending voice traffic over the networks designed traditionally for data networks pose following issues.

2.2.3.1 Delay/Latency

Delay is defined as the amount of time it takes for the speech to travel from source to the destination. It can be sub characterized as follows:

- **Propagation delay**: The time taken by the packet to propagate through the medium. This delay alone does not make much difference to the voice quality as human ears are imperceptible to it. But the delay in conjunction with other delays can drastically affect the sound quality.
- **Packetisezation delay**: This delay is generally caused by the digital signal processors (DSP) while forming a speech sample. For example, the G.729 codec (ITU-T Recommendation G.729, 1996) takes 10 ms to generate a speech sample.
- **Queuing delay**: It is defined as the amount of time the packet takes to come out of the output queue of the interface. This delay generally occurs when more packets are sent out of the interface than it can handle at a given interval.

The sum of all these delays may be greater than 150 ms in heavily congested traffic and some quality of service is needed to reduce it to acceptable level.

2.2.3.2 Jitter

Jitter (Davidson, 2000) is defined as the variation in the inter-arrival time between the packets. When the sender sends the voice packet at regular interval it is expected that the voice packets should reach the receiver with the same interval between them, but due to delay in the network it is possible that the packets do not arrive at that same regular interval at the receiving station, this makes the voice information choppy thus reducing the quality of voice drastically.

One way to solve the jitter problem is to use a play out buffer (Davidson, 2000). This buffer, buffers the packet that arrive early to compensate for the delay of the slowest packet. The play out buffer must be large enough to accommodate the inter-arrival time of the packets as a result buffers are used which dynamically increase or decrease there buffer size based on the inter arrival delay variation of the last few packets.

2.2.3.3 Echo

Echo (Davidson, 2000) is a phenomenon where the speaker hears his or her voice from the receiver while talking. This is commonly seen when there is a mismatch in impedance from four-wire switch conversation to two wire local loop. In normal PSTN network the echoes are not annoying because of low circuit delays but in the voice networks the echo become more intense due to the additional delay introduced in the VoIP network. One of the ways to solve this problem is by introducing echo cancellers either in the digital signal processor or implements it as software on the device.

2.2.3.4 Packet loss

In IP network packet loss mainly is due to congestion in the network. The packet loss for the data transmission is critical because once the data is lost it cannot be retrieved as a result reliable protocol such as Transmission Control Protocol (TCP) (Addison, 1993) are used to send the data packets.

Dropping a few packets of uncompressed speech will not affect the voice quality as voice is very redundant in nature. Because of this nature voice traffic is passed over the network through unreliable protocol such as UDP (Addison, 1993).

The advancement in internet technology and multimedia protocols have led these issues to slowly fade away. As a result more and more enterprise users are using their IP network as a carrier for voice calls, video conferencing and other real time applications. Implementing VoIP will require a central system that will not only handle the calls from different end devices but also maintain the call control. The most commonly used central system in the industry is a Cisco Call Manger (Alexander, 2005).

2.3 Voice Controlling Software

One of the most widely used software in the voice industry to control voice calls is the Cisco Call Manger (Alexander, 2005). The software functions like a telephone operator who places a call on behalf of many different end points in the network. It acts like a soft switch by handling the signaling of the calls originating within the network and also handling calls that originate or terminate outside the network.

2.3.1 Basic Call Manager Operation

Call mangers are the central brains for the voice network and are placed in the form of the clusters. Clustering not only provides scalability but also provides reliability to the voice network. Depending on the size of an enterprise, a voice network cluster can support 2 to 8 call mangers in its domain. The clusters can further support 7,500 to 30,000 VoIP end devices.

When an end device wants to place a call it has to first consult with a designated call manger in its cluster. The call manager which serves this call in turn provides the information to the end devices by consulting a database server. A database server, Microsoft SQL 2000 (Haselden, 2006) contains all the information such as route plan service, parameters etc. The call manager cluster and database server go hand in hand.

As stated above for the end device to make a call it has to first register itself with the call manager. The steps involved during the registration process are shown in the Figure 1.

- An IP address is requested when IP phone comes up and is connected to the network.
- The dynamic host control protocol (DHCP) server (Brown, 2003) responds with the IP address and an IP address of the trivial file transfer protocol (TFTP) server.

- After receiving the TFTP server (Habraken, 1999) address the IP phone requests a configuration information file from the server. This configuration file consists of a list of all call managers in the cluster and the primary call manager address.
- Once this information is received the IP phone sends registration request to the primary call manager. The call manager through the registration request checks the version of the firmware present on the IP phone. If the firmware version is supported by the call manager it lets the registration process further otherwise it loads the new firmware version on the phone through TFTP server and then allows the registration process.

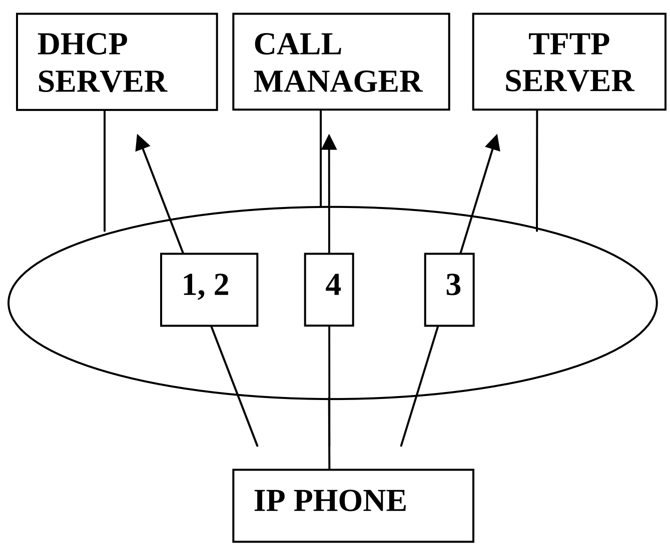

Figure 1 - IP Phone Registration Process

After this registration the end devices like IP phones or software smart phones can talk to the call manger by using any one of these three major protocols such as H.323, Skinny Client Control Protocol (SCCP) (Alexander, 2005) and Session Initiation Protocol (SIP). The call manager also communicates with the gatekeeper and the gateway to maintain connectivity with the PSTN.

2.3.2 Communication of Call Manger with Gatekeeper

Gatekeeper (Danelle, 2005) in the voice network is not only used to authorize and authenticate the terminals and gateways it is also used to manage the bandwidth of the network. When gatekeeper is initialized in the network call manager registers the end devices with the gatekeeper. This is done through the registration request. When the end device comes up it broadcasts the registration request message on the network. After seeing this request gatekeeper responds with the registration confirm message (RCF) or registration reject request message (RRJ). After registration end devices maintains continuous connectivity with the gatekeeper by sending a light weight registration request (RRQ) as a registration keep alive.

When the end device makes a call the destination number is analyzed by the call manager and sent to the assigned gatekeeper. The gatekeeper checks the destination number and decides where the call has to be directed as it has the directory information in its database .When the conversation over the admitted call is over each end device notifies the gatekeeper about the call termination by sending a disengage request (DR) message. This message is accepted by the gatekeeper by sending a disengage confirm signal (DCF).

2.3.3 Communication of Call Manger with Voice Gateway

Gateways (commonly known as gateway routers) provide connectivity between an IP based (most commonly H.323 network) and a PSTN based network by converting digital signals to IP signals and vice versa. As IP phones are controlled by the call manager in the network similarly voice gateways (as shown in figure 2) are also controlled by the call managers through various signaling protocols such as H.323, Media Gateway Control Protocol (MGCP) and Session Initiation Protocol (SIP).

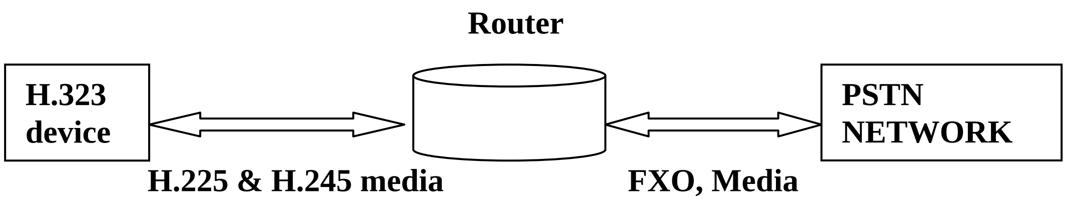

Figure 2 - Protocols used by the gateway

Call managers are designed for large enterprise network as it is capable of processing calls for many IP phones. Small enterprise networks are reluctant in deploying call mangers as the solution is not cost effective for them. To solve this problem a lighter version of call manager called as Cisco Call Manager Express has been developed by Cisco Systems.

2.3.4 Cisco Call Manager Express

Cisco Call Manager Express is a router running on Cisco internet operating software (IOS) software that deliver most of the core telephony features which are required in the small enterprise enviourment. It also provides many advanced features such as music on hold, call parking, call forwarding etc which are not available with traditional telephony solutions.

As shown in Figure 3 Cisco Call Manager Express (Danelle, 2005) is compatible with the IP phones and works with the gatekeeper and gateway devices in the same way as call manager does**.**

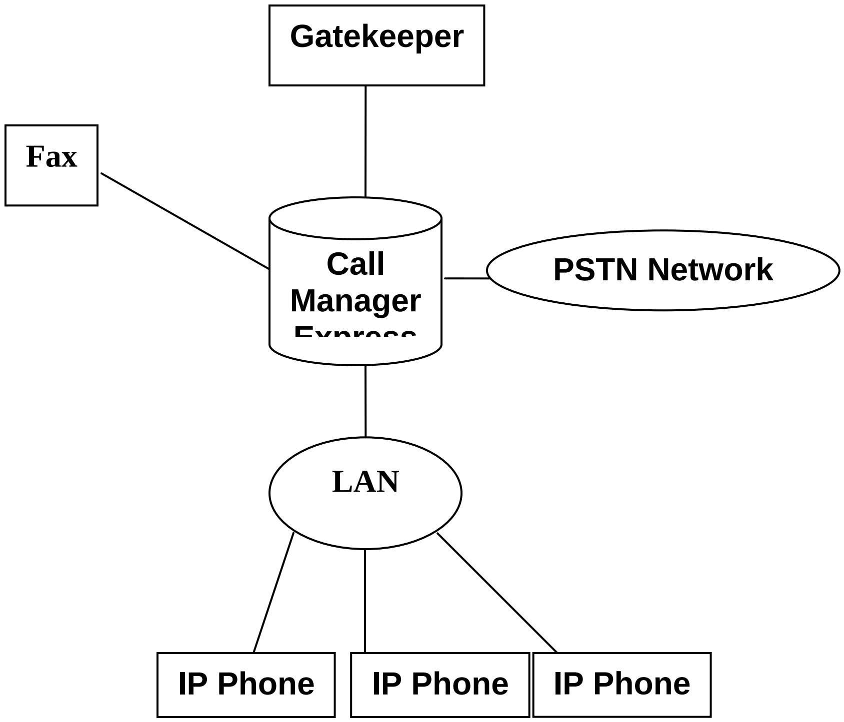

Figure 3 - Integration of call manager express

2.4 Wireless Networking

The exponential growth in wireless field has made the world a smaller and increasingly mobile place. Because of this, traditional ways of networking like wired networking is proving to be inadequate as it is not being able to meet the challenges posed by new collective lifestyle. As an example a user who has to be connected to a network by physical cable is always confined to that place thus restricting his/her movements dramatically. On the other hand wireless connectivity poses no such restriction and allows users more freedom. This makes the wireless technologies popular and is being adopted by more and more users. Some of the widely used wireless technologies are explained below:

2.4.1 802.11 Standard

In today's traditional wired networking, Ethernet technology is the most predominant one, but in order to expand the wireless technology, wireless manufacturers are being pushed for open standard. In June 1997 Institute of Electrical and Electronics Engineering (IEEE) released the 802.11 (Gast, 2005) standard for WLAN.

This standard provides internet connectivity not only to fixed nodes but also provides to the mobile nodes. The connectivity is maintained through WLAN by using radio frequency which is used to transmit and receive the data packets over air. Depending on the type of frequency used the 802.11 standard is further divided into three sub-standards which are 802.11a (Gast, 2005), 802.11b (Gast, 2005) and 802.11g (Gast, 2005). Wireless networks consist of four major physical components, these are summarized as follows:

- **Stations:** These are devices equipped with wireless network interface cards. Most common stations seen today are laptops or handheld computers.
- **Access points:** These devices are used as a bridge between the wired and the wireless medium. The bridging is done by converting the wired frames in such a way that they are compatible with the wireless medium.
- **Wireless medium**: In order to move frames from station to station, the standard uses radio frequency (RF) as a wireless medium.
- **Distribution system:** Single access point coverage is very limited because of this many access points have to be deployed in the network to increase the coverage area. Moreover all these access points have to communicate with each other to track the movements of mobile stations. This is achieved by the distribution system logic which forwards the frame to their destination.

2.4.2 Satellite Networks

Previously satellite networks were primarily used by military. But today they are an essential part of our daily life. They are used for voice calls, weather reports, television transmission etc. To implement data or voice packet networks over satellite one must employ special techniques to reduce the delay caused by the 96,000 mile round-trip. This is because most of the packets switch networks work on TCP (Addison, 1993).This protocol which was designed for terrestrial networks works by sending packets of data, and then waiting for acknowledgments. This acknowledgment in turn signals the sender to transmit more data. When acknowledgments are returned slowly, protocol slows the speed at which data is being sent in order to avoid overloading a network. When implemented over the satellite network it does not know that a satellite is involved and operates as if the satellite latency is in fact network congestion and sends all packets over a satellite network at a slow-start rate.

To increase efficiency, techniques like TCP spoofing (Ishac, 2001) are used. In this technique special equipment is implemented at the carrier's main satellite hub site which appears to transmit as if it were the remote location, when the spoofing equipment receives internet traffic destined for a remote satellite location, it acknowledges receipt of the packet immediately on behalf of the remote site so that more data packets will follow immediately thus increasing the efficiency.

The ultimate objective of wireless data communication is to use radio waves to connect end users in areas ranging from a few feet for desktop's, to hundreds of feet for local area networks . For wireless networks to be a truly viable technology it needs to be ubiquitous providing seamless integration with existing wired networks and robust enough to support and transfer data in the range of 10 to 100 Mbps.

2.4.3 Air Craft Data Networks

The increased development in wireless local area network infrastructure and satellite communication has made voice over a wireless network a viable option. The same network now can support both prioritized voice and data traffic thus giving telephones, computers and personal digital accessories (PDA) access to the internet from any point .Taking advantage of this broadband wireless development, steps are being taken by many airline companies to develop a network in a way that it provides passengers in the aircraft access to the internet and also allows them to talk either through cell phones or IP phones.

Technologies such as Pico Cell Technology (Cell phones, 2005), Wireless Cabin System (WCP, 2001), Air Cell have developed a network which allow passengers to use their cell phones as a medium. However using cell phones has a major drawback as it can interfere with the aircrafts navigational system as they use the same under lying technology (radio waves) which are used by navigational system of the aircraft. Other technologies such as Connexion (Boeing, 2005) are trying to provide the same feature through IP based networks. The advantage of using this technology is that there is no interference with the aircrafts navigational system. However, the cost per call is expensive as this technology uses various service providers to route the voice traffic.

The above stated technologies use satellite link to communicate with the base station on the ground network. Using the satellite link has few drawbacks of its own on voice over internet protocol networks. The first major disadvantage voice traffic faces over this link is high latency and jitter due to long round trip propagation delay. Secondly, limited bandwidth of the satellite link is a huge disadvantage on quality of voice. However concepts like jitter buffer (Textronix, 2003) are being developed which are enhancing the quality of voice over satellite links.

In this chapter the author discussed about VoIP technology challenges, devices which process the calls, improvements in wireless and satellite technologies and how these technologies have helped in developing aircraft data networks. Keeping this discussion as the basis, in the following chapter the author explains in detail about the aviation data network, the current technologies and protocols used which help aircrafts to be mobile and connected to the IP network.

CHAPTER 3

AIRCRAFT DATA NETWORK

In this chapter the author explains the technical aspects of aircraft communication and discusses various available solutions in the market that provide voice services in aircrafts. The chapter is divided into three sections. The first section covers the basic method by which aircrafts communicate with the ground station to attain internet connectivity. The second section explains different protocols that can be used to maintain an efficient communication. The final section discusses the drawbacks of clubbing cellular and internet technologies and explains various voice solutions present in the market that try to address the problem.

3.1 Aircraft Communication

Wireless networking (Ganz, 2003) at present is one of the most rapidly emerging technologies. It provides the user with seamless network connectivity. One application domain where this technology is playing an important role is aviation data networks. In this domain wireless networking helps the aircrafts to be continuously connected to the base station. A common architecture which is followed to provide these services is explained in Figure 4.

In this model the aircraft can be considered as an internal network where wireless access point is used to provide connectivity to passengers and crew members. The satellite link is used to connect the aircrafts internal IP network to the ground station which is then connected to the internet service provider or cellular services acting like a gateway. As satellite connection is one of the integral component to maintain a seamless connectivity between the aircraft and ground station different protocols and techniques have been developed to provide uninterrupted IP connectivity over the satellite connection. The next section discusses in detail about these protocols

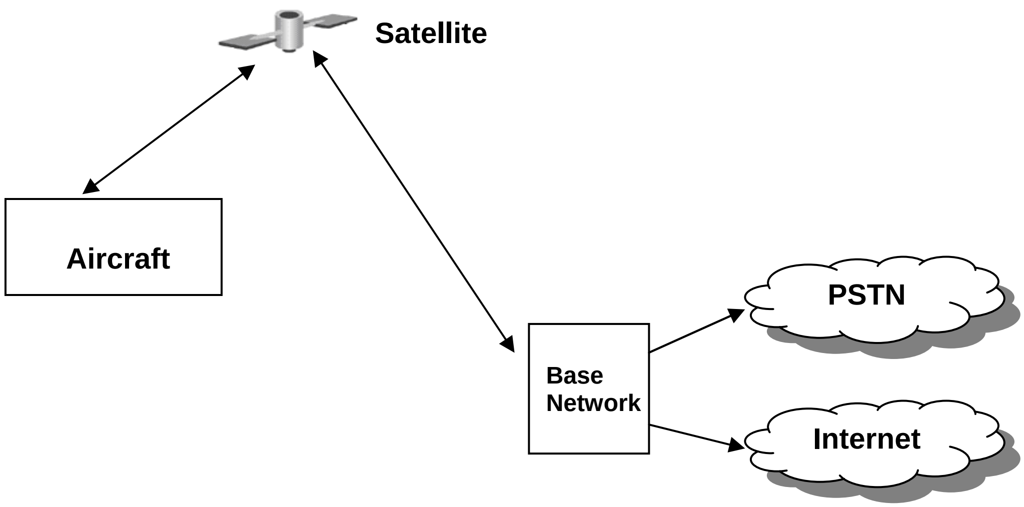

Figure 4 Basic aircraft data network

3.2 Mobility protocol

The traditional IP behaves awkwardly when dealing with the host moving from one network to the other. The reason for this behavior is that the Internet Protocol version 4 (IPv4) uses hierarchical routing and assumes that the node attached to the internet is fixed. If the node is mobile, normal routing mechanism fail to connect the node to the internet as the IP address used on the nodes interface cannot indicate the physical location. In the mobile environment the mobile node has to be connected seamlessly to the internet irrespective of its position. One way of achieving this is through a protocol called Mobile IP.

3.2.1 Mobile IP

Mobile IP (Perkins, 1996) is a dynamic routing protocol which allows a node to move across various IP subnets and access links all the time, while maintaining continuous connection with the network or the internet. This protocol sits below the TCP/IP stack and provides transparent mobility support to the higher layer protocols.

There are four primary architectural entities used by Mobile IP (Perkins, 1996) protocol which provide seamless and secure connectivity to the mobile users. They are as follows.

- **Mobile Node (MN):** It is an IP device (end user) running a Mobile IP client stack which changes its point of attachment from one network to the other.
- **Home Agent (HA):** It is a router in the home network whose job is to send the data traffic (destined for the mobile node) to the MN. This is done by maintaining current location information of the MN when it moves to a foreign domain. Home agent also runs the Mobile IP protocol software.
- **Foreign Agent (FA):** It is a router on the foreign network which connects the MN to its home network.
- **Mobile router (MR):** It acts like a gateway for the mobile network. It consists of a roaming interface through which it is connected to the rest of the world and a non roaming interface which connects the internal entities of the network. All internal entities in the network are configured to talk to the external world through the Mobile Router (MR).

3.2.1.1 Protocol Operation

Mobile IP protocol operation is divided into three phases which are agent discovery, registration and tunneling. In the agent discovery phase HA and the FA periodically advertise their services on the network when configured with Mobile IP protocol. These advertisements carry extensions which describes whether the agent is a HA or a FA, type of services it will provide and the CoA. MN which is capable of listening to these advertisements determines whether it is in its home domain or has moved to a foreign domain. This process of detection can be accomplished by the MN in one of the two ways.

In the first method movement detection depends on the lifetime expiry method. Here if the MN does not receive any new agent advertisement by the time the lifetime expires it detects that it has moved to the foreign domain (Perkins, 1996).

In second method all the mobility agents like HA, FA etc broadcast their advertisements which include their prefix extensions. Upon receiving the advertisement with a different prefix MN realizes that it has moved to a different network.

When the MN detects that it has moved to the foreign network it acquires Care of Address (CoA) in one of the two ways.

- **Acquiring care of address from the Foreign Agent (FA)**: In this mode CoA is the address provided by the FA. Mobile node notifies the HA about its current location through this CoA. Same address is also used by other MN in the same foreign domain to notify there respective position.
- **Running with a Co-located care of address:** It is an address assigned to the roaming interface by the MN itself or it can be obtained by the dynamic host control protocol (DHCP) server. This address can be used only by a single node and represents the position of that MN in the foreign network.

The next step is to inform the HA about its current location. This is done in the registration phase were MN sends a registration request which contains the CoA and some other information which the node learns from the foreign agent’s advertisement. The request is sent either directly to the HA (if the mobile node has co-located care of address) or is sent through the FA.

If the registration request is sent through the FA, it checks the life time field to make sure that it does not exceed its limitation and also checks the encapsulation field to make sure that it can support the requested encapsulation technique (explained in detail in the following section). After the inspection if the foreign agent decides the request to be valid it forwards it to the HA.

The HA then checks the validity of the registration request and also makes sure that the request has come from the authenticated MN. If the registration request is valid the HA associates the MN IP address with the Care of Address. The association between the MN, the CoA and the lifetime together is called as binding. After this binding process a registration reply is sent to the MN through the FA. The FA checks the registration replies and updates its visitor list by adding MN to it and establishes a tunnel between HA and FA.

Traditional Mobile IP protocol supports two type of encapsulation techniques a) IP-in-IP encapsulation b) minimal encapsulation. In IP-in-IP encapsulation type an additional IP header is encapsulated over the original packet which is destined for MN. This kind of tunneling is seen when a single node (acting like a mobile node) moves from home network to the foreign network. As shown in the Figure 5 when PC 4 moves from its home network to the foreign network a tunnel is formed between the home agent and the foreign agent which is an IP-in-IP encapsulation of packet (Perkins, May 1996).

Data packets which are destined to the MN are first routed to the home network. HA in the home network checks the destination address of the packet, and if it’s for a node in foreign domain it tunnels it to the FA. The HA forms the tunnel by encapsulating the original IP header with a new IP header containing the CoA as its destination address. Figure 6 shows the IP-in-IP encapsulation packet.

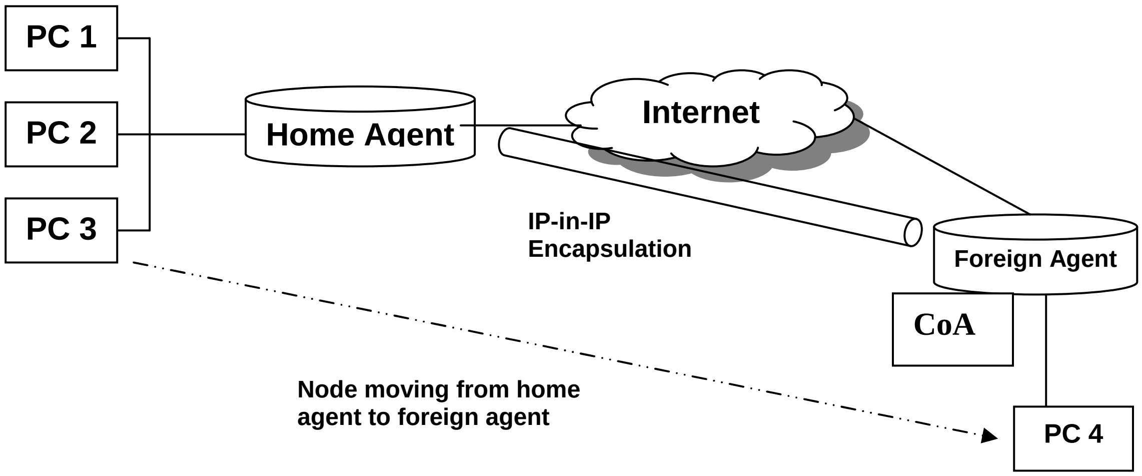

Figure 5 Formation of tunnel during hand-off

Care of Address (CoA)	**Destination IP Address**	**Corresponding Node (CN) IP Address**	**Load**	**HA IP Add.**

Figure 6 IP-in-IP encapsulation

When the FA receives the packet it de-capsulate the first IP header and forwards the remaining IP packet to the MN. However this tunneling mechanism does not work well if the whole network is mobile. This is because when mobile network moves to the foreign domain it is not allowed to participate in routing as a result foreign agent fails to learn the topology of the network which leads to loss of connectivity. One way to resolve the issue is by using double encapsulation technique.

In this technique (explained with the help of Figure 7) a packet which is destined to a node present in foreign network is always passed through the HA. HA encapsulates the original packet with the destination address of the mobile router. However in order to reach the mobile router, HA again encapsulates the packet with an IP header destined to the FA. Figure 8 shows the double encapsulated packet from the HA to the mobile router.

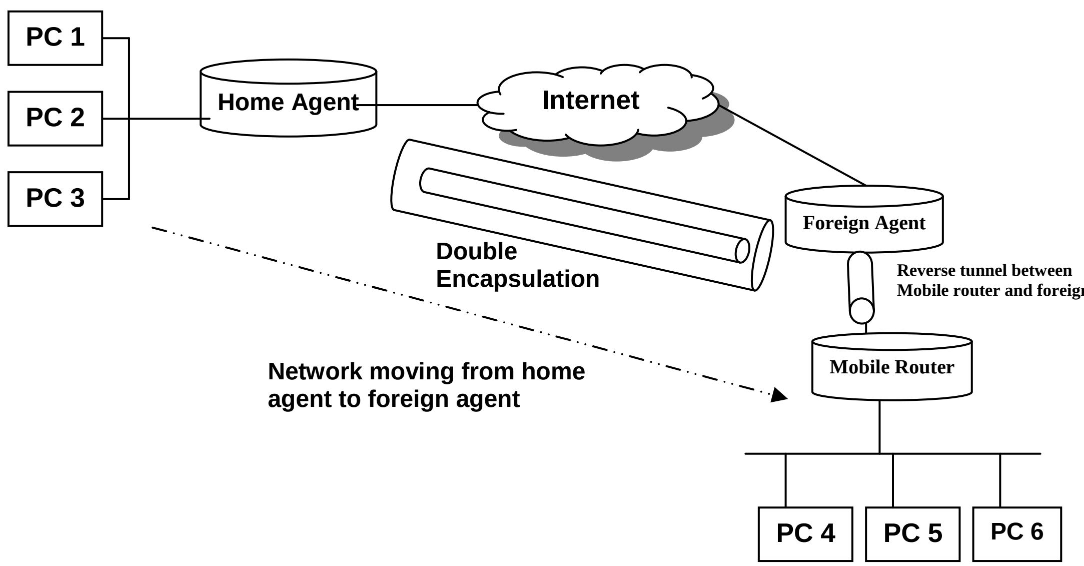

Figure 7 Formation of dual tunnel while hand-off in mobile router

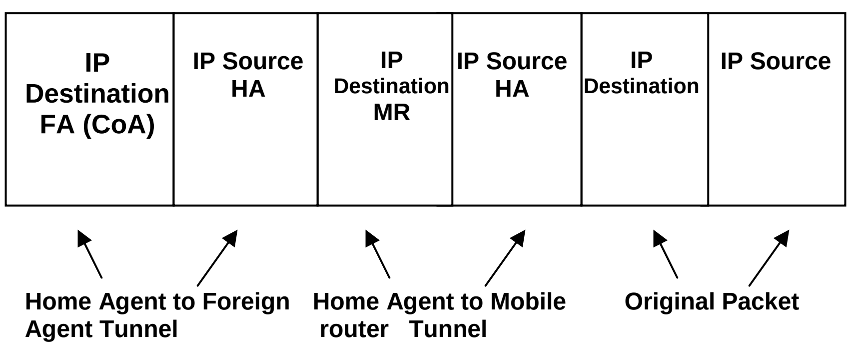

Figure 8 Dual encapsulated Packet

When the encapsulated packet reaches the FA, the FA de-capsulates the outer packet and sends the packet to the mobile router. Mobile router again de-capsulate the inner IP header and routes the packet to the respective node.

In the second encapsulation technique which is minimal encapsulation the IP header of the original datagram is modified and the minimal forwarding header is inserted into the original IP header. This is shown in Figure 9.

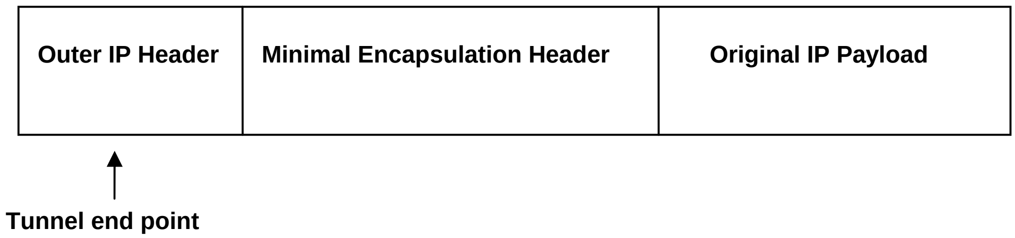

Figure 9 Minimal encapsulation

In the reverse way when MN tries to send the packet to the Corresponding Node (CN) it first sends the traffic to the mobile router. Mobile router then sends the packet to the corresponding node in one of the three ways.

- **Triangular Routing**: In this technique mobile agents use standard IP routing mechanism to deliver the datagram's in the reverse path. As a result of this the packets originated from mobile node or a mobile entity in the mobile network is sent to FA. The Foreign Agent determines the destination by using configured IP routing mechanism and routes the packet to destined node.
- **Reverse tunneling (Perkins, 1998):** In this technique mobile router reverse the packet to the FA who passes the packet to the HA and in turn to the CN. However for the packet to be reversed tunneled the mobile router first creates a default route to the FA and sends all the traffic through the FA. This type of communication is used when there is a private addressing scheme behind the mobile router (Raab, 2005).
- **Cisco Foreign Agent (FA) optimize routing technique (Raab, 2005):** In this technique FA directly sends the traffic to the CN and vice versa. This kind of routing is only

possible if the CN is connected to the same FA as the mobile router. In order to accomplish this FA eavesdrops on the registration process done by the mobile router with the HA. As a result of this the FA gains knowledge of all the networks connected to the mobile router

3.2.2 Issues with Mobile IP Protocol

3.2.2.1 Quality of Service (QoS)

QoS in a deployment is very important because the same network infrastructure is being used to send real-time and non real-time data. Some data streams like real-time traffic requires guaranteed delivery with low end-to-end transmission delay across the network where as other data streams are not bounded to these rules. To ensure that both kinds of traffic demands meet the quality of service it has to be made sure that the resources are made available for specific traffic type.

In Mobile IP environment when the mobile network is in the home network, the service is provided by the home network devices. But when the MN or mobile network moves to the foreign domain the issue of quality of the service becomes complicated because of two reasons. Firstly, the foreign network might not have enough bandwidth to support the mobile network traffic and secondly the IP-in-IP tunnels established by the MN or mobile router to communicate with the home network does not transport the quality of service information of the inner IP header to the outer IP header. As a result, the packets with higher IP priority will also be treated as normal packets in the intermediate domain connecting the home network and the foreign network To overcome the issues typically all routers in the network are configured with traffic shaping policies which helps them maintain quality of service in their own traffic streams and to the streams from the mobile network.

The modified Mobile IP protocol (MIP-LR) proposed by Ravi Jain (Jain, 1998) tries to address the quality of service issue in Mobile IP in a similar manner as discussed above. In MIR (Gwendal, 2000) author proposed a new technique which offers statistical bandwidth guarantees to the mobile users. There are a few other authors (Yi, 2004) who have also proposed theories which integrate quality of service with Mobile IP.

3.2.2.2 Handoff Issue in Mobile IP Protocol

One of the major disadvantages of Mobile IP protocol is that the mobile network has to register and authenticate whenever it moves from one network domain to the other. The process of registration and authentication depends upon the hand off mechanism and the movement detection algorithm used. But during the process of registration MN or mobile network is completely cut off from the home network which results in loss of data.

To resolve this issue different types of protocols have been designed. In (Srikant, 2004) this paper author proposes low-latency hand off scheme which expedites link-layer handoff detection and speeds up network-layer hand-off by replaying cached Foreign Agent (FA) advertisements. In (Ali, 2003) the author Ali Diab proposes a new technique which uses local authentication approach with the new FA and is independent of the re-authentication with the HA which helps MN to quickly resume the transmission in an up and downlink after an handoff . The other approach is the POLIMAD (Stefan, 2003) algorithm which uses link layer parameters for Mobile IP handoff decisions and reduces the hand off time drastically.

Taking advantage of the mobile and wireless technology airline companies who are always looking to provide different services to the passengers today are interested in providing both internet (for data communication) and cellular connectivity (for voice communication) on aircrafts. But combining these two technologies brings the issues like interference and interoperability.

3.3 Problem with Cellular Technology on Aircrafts

To provide both of these services, an aircraft must have receiver (access points) for receiving both kinds of wireless signals and should be able to transmit traffic from both systems to the ground via the satellite link. This causes interference which might hamper the functionality of navigational and communications systems needed for operation of the aircraft. In addition the personal electronic devices like cell phones cause aircraft navigational system to malfunction because of the radiation these devices emit during conversation. There are two kinds of radiation emitted by these devices.

- **Intentional Radiation**: These radiations are related to transmitted data in the allocated frequency bands of the PED. Intentional transmissions are not of a concern because their frequency bands are limited and do not overlap with the frequencies of airline systems.
- **Spurious Emissions:** These are unintentional radiation which contributes to the radio frequency (RF) noise level. If the power level of a spurious radiation is high it could interfere with aircraft operations.

The other major drawback of this type of implementation is the low interest shown by passengers. According to the survey done by the airline companies and other research institutions it is found that majority of people (passengers) not only favor the cell phone restrictions on commercial aircraft but instead prefer to have internet access over cell phone availability. United Airlines conducted a survey and found that 80 percent of its passengers preferred wireless Internet access over cell phone availability (Cellphones, 2005). Similarly a survey done on frequent air travelers by Lauer Research for the Association of Flight Attendants (AFA) and the

National Consumers League found that 63 percent opposed the lifting of cell phone restrictions on commercial aircrafts.

3.3.1 Related Research Work

In order to integrate both the technologies and reduce interference between the wireless devices and navigational systems on the planes several techniques have been proposed and developed.

3.3.1.1 Pico Cell Technology

Pico cell technology (Cell phones, 2005) allows users to use cell phones and other wireless communications devices on aircraft basically by lowering the power setting which are used by electronic devices to make a call. All wireless signals are collected by a smoke detector-sized transmitter and sent to the ground-based cellular network through satellite-based network. This technology even allows the crew to control the call flow by disabling incoming or outgoing calls.

This technology helps in reducing the interference problem but does not completely eradicate the problem, moreover the present transmitters used are not capable of handling many calls and installing the transmitter itself has its own compatibility problems. The other major non-technical disadvantage about this technology is licensing of frequency from the Government.

3.3.1.2 The Wireless Cabin System

As an aircraft will have several mobile users with different carrier services, this system tries to provide seam less connectivity to all users by accumulating all the signals and sending it to ground through satellite link. Wireless cabin model (WCP, 2001) architecture consist of three segments

- Several wireless access domains in the aircraft cabin.
- A satellite segment for interconnection of the cabin with the terrestrial telecom networks
- An air com service provider segment supporting the integrated cabin services.

Some of the drawbacks of the system are that the technology only concentrates on the cellular services and does not take into consideration the internet services. This can be a major disadvantage as most of passengers prefer Internet technology to be present rather than the cellular service. Another technical hitch is that to communicate with base stations on the ground, phones must transmit at their maximum power, which can cause interference not only with other devices in the cabin but also may malfunction the navigational system of the aircrafts.

3.3.1.3 Air Cell Technology

The Air Cell (Rocky, 2003) technology mainly functions with a network of 134 cells on the ground, which are composed of many wireless partners, including Alltel, Western Wireless and U.S. Cellular. As the aircraft passes over these cells, the phones will connect with the most powerful one, and the towers will hand off to nearby towers as the aircraft continues to move.

This technology allows the passengers to use their cell phones on aircrafts which brings with it three major disadvantages. The first and the foremost is the interference of cell phone radio waves and ground station radio waves with aircrafts navigational system. Second major disadvantage of the technology is the cost of the call. Since the cell phone connects to any wireless provider as discussed above the user has to pay to the other service provider including its own for the call. Last but not the least is the extra infrastructure cost which has to be put in place to reduce the interference that the network can cause with aircraft navigational system.

3.3.1.4 Connexion WI-FI Technology

This technology (Boeing, 2005) makes it possible to provide different kinds of services like delivering real-time traffic and high-speed connectivity to airline passengers. The key to making this technology work is the antenna connected to the plane which acts like a gateway to the internal network (stub area). It is designed so that it can track satellites efficiently which then allows it to seamlessly connect to the home network on the ground.

As with any new technology there are many advantages and few disadvantages. This technology also suffers from some disadvantages such as security issues, filtering and bandwidth limitation. The other problem with this technology is the cost of a call. The cost per call is high because to send the voice packets to the destination address satellite bandwidth maintained by other service provider is needed.

In this chapter the potential problem with the current technologies were discussed. The following chapter explains in detail about the various network topologies considered (keeping the proposed design as the base) to develop an efficient voice network for aircrafts and also highlights the potential issues these designs face in implementation.

CHAPTER 4

PROPOSED NETWORK

Keeping the considerations discussed in the last chapter author here attempts to design a network which integrates both the voice and data communication as a part of one network. This network not only allows passengers on the aircraft to access internet but also allows them to make a voice call at a much lower cost. The other advantage of the proposed network is that since it does not use cell phones for making calls it completely eliminates interoperability issues generated between the cell phones and the navigational system on the aircrafts. The present chapter is divided in to two sections. The first section explains the general network design and the second section discusses different kind of scenarios that can be made possible with this general design.

4.1 General Proposed Model

As shown in the Figure 10 the proposed network design is divided in to three major segments:

- Mobile Node Segment (aircraft cabin).
- Mobile Segment (satellite communication)
- Home Segment (base station)

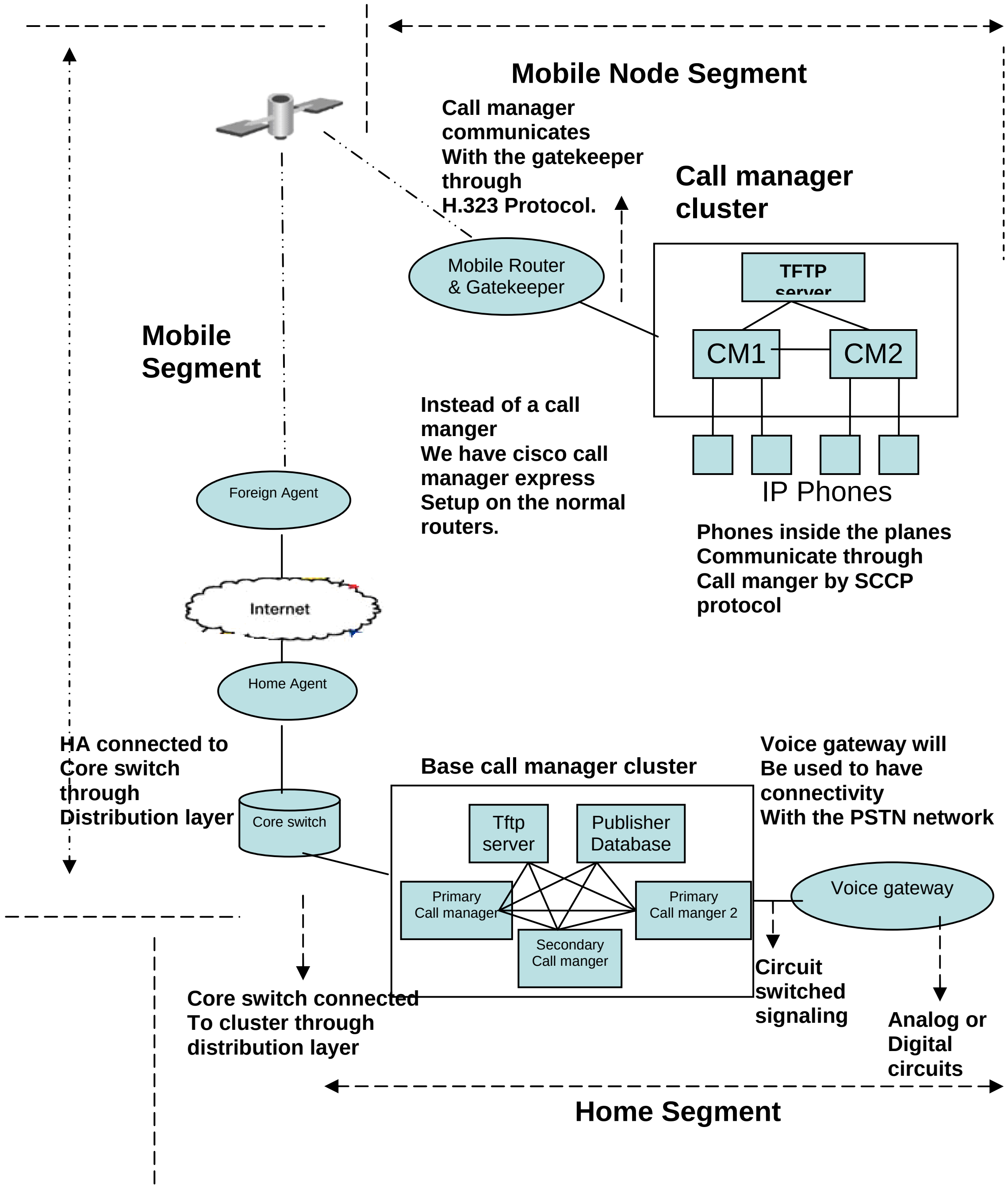

Figure 10 Proposed model for voice communication on the aircraft

4.1.1 Mobile Node Segment

This segment mainly deals with the internal network on the aircraft. As per the proposed solution it is assumed that each seat on the aircraft will be equipped with an IP phone or a software IP phone. These IP phones with the help of a switch will be connected to the call manger cluster. The cluster will control voice calls and act as a DHCP, TFTP & database servers which are an integral part for the call manager voice network.

This cluster will then be connected to a different router which will act like a gateway. The main job of this router is to maintain a seamless connectivity with the home network by acting like a mobile router. It can also act like a gatekeeper, by supporting registration admission status protocol (RAS) message set which are used for call admission control, bandwidth allocation, and dial pattern resolution (call routing).

The IP phones in the aircraft when activated will request services like device configuration call acceptance, call termination from the call manager cluster either by using SCCP or H.323 protocol. However the most efficient protocol which can be used for communication is the skinny protocol as it is lightweight and very simple.

The cluster on the other end is connected to the gateway/gatekeeper router. First to communicate with the gateway/gatekeeper router it has to register it self, registration can be done manually by assigning the router address. After the registration process the communication between the registered cluster and gatekeeper is carried out through the H.323 protocol.

Generally in this segment when the end device tries to make a call depending on the protocol used the signaling part is handled by the call manager cluster and the gateway/gatekeeper and once the signaling path is established then the end devices stream media directly between them through the available network. Moreover as the node segment is gatekeeper-routed model all calls that a given cluster doesn't know how to route locally queries the gatekeeper, which in turn has a centralized configuration containing the route and transfers the call to the base station through the satellite link.

4.1.2 Mobile Segment

This segment makes sure that the aircraft is always connected to the base station irrespective of its position. This task is accomplished with the help of three components. The first component is the mobile router. It sits on the aircraft and acts like a gateway and is also configured to work as a mobile router. When the aircraft is in the home domain the mobile router acts as a normal gateway router transferring the calls (those not destined to the internal network) to the base station gatekeeper. But when the aircraft is on the move the mobile router senses that it is away from its domain and registers it self to the home network with the help of a satellite link and intermediate router acting as FA Foreign agent is also a router which is connected through the internet to the home network gateway router**.** The FA periodically broadcast its presence through advertisements to the aircraft via satellite link. When the wireless interface of the mobile router detects this advertisement it assumes to be in a different domain and sends a registration request to the home network gateway router about its location. This registration request in turn is carried by the FA to the home network. The HA (third component) which is the gateway for the home network sees the registration request of the mobile router and updates its database with the present location of the aircraft. Now when the traffic is destined to the mobile router it is first encapsulated with the mobile routers address and re-encapsulated with the foreign agents address and sent to the FA which in turn is sent by the FA to the mobile router through the satellite link**.**

Similarly when the mobile router wants to setup a call or send the voice data to the home agent network it reverse tunnels the traffic to the associated FA by encapsulating the packets to the HA and sending it through the satellite link.

4.1.3 Home Segment

This is a segment which allows the IP phones in the node segment to be connected to the rest of the world. In this segment the main cluster not only handles the calls from the aircrafts but also handles the whole enterprise voice traffic. As a result of this the cluster is comprised of two or more Cisco call managers, a separate DHCP server and a TFTP server. The IP phones of the enterprise are connected to the call manager and talks in the same way as the IP phones communicate with its call manager in the node segment. The other end of the call manager cluster is connected to the HA which also acts like a gatekeeper. The main function of the gatekeeper here is to route the calls destined from the aircraft to the main call manager cluster.

The calls from the aircraft will be destined for the home network and to the phones connected to the PSTN. In order to transfer the call to this network the call manager cluster is connected to the voice gateway. The voice gateway can be a H.323 device or any other router which can run media gateway control protocol (MGCP). Generally MGCP capable routers are used as the call manager is capable of controlling the gateway, thus maintaining the centralize control of the network.

Similar to the node segment the call manger in home segment can communicate with the IP phones connected to it either through skinny client control protocol (SCCP) or H.323 protocol. In the proposed solution the call manager cluster communicates with the IP phones through SCCP as it uses minimal bandwidth and to the voice gateway through MGCP. The advantage of using MGCP protocol is that as it works on a master/slave protocol concept the call control intelligence is retained by the call manager as it acts like a master.

In the section above author discusses in detail about the hardware architecture and protocols used. The following section explores in depth the various network topologies considered (keeping the proposed design as the base) to develop an efficient voice network for aircrafts and also highlights the reason why this network design (even though functional) cannot be applied in the commercial market.

4.2 Proposed Methodology

4.2.1 Network Design 1

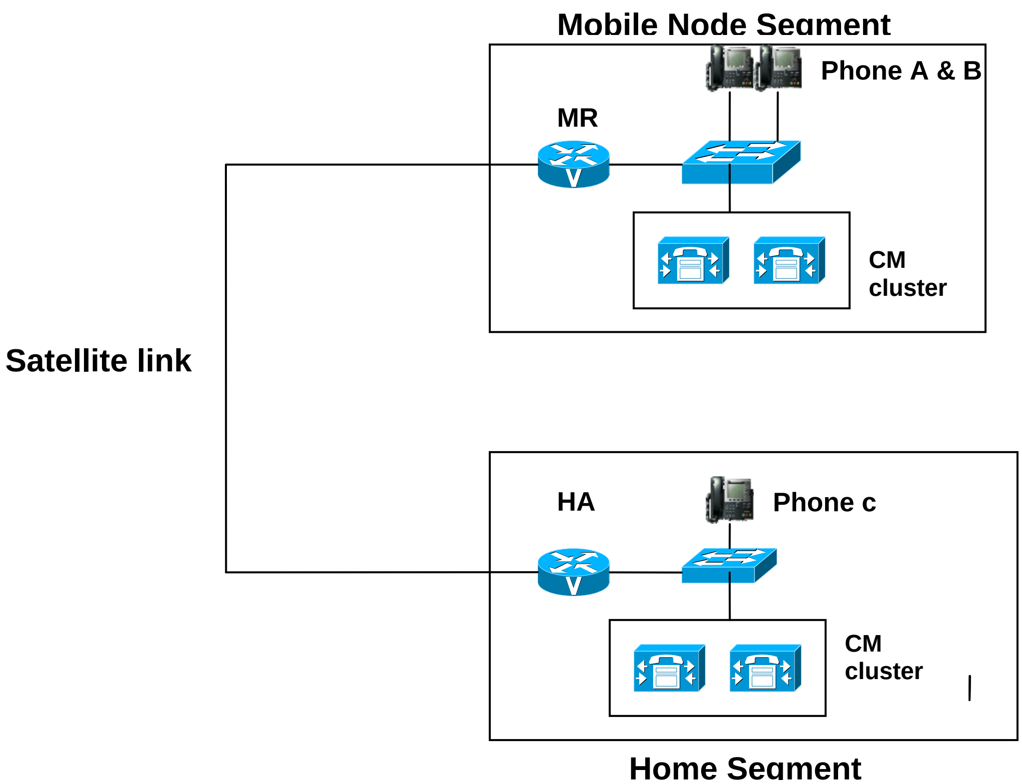

Figure 11 Network design 1

As shown in the Figure 11 call manager cluster is deployed in the mobile node segment (aircraft) as well as the home segment of the proposed network to deliver the voice traffic from the aircraft to the base station. The main reason for deploying a call manager cluster in each segment is to reduce the signaling and voice traffic over the satellite link which has limited bandwidth. This can be verified from the scenarios explained below.

When IP Phone A calls IP phone B the SCCP signaling from phone A is analyzed by the call manager cluster present in the mobile node segment. As the call manager cluster would know that the called party phone number is also present in the same cluster it will directly notify phone B about the incoming call. Now If IP phone A dials IP phone C or analog phone D which are in the home segment and PSTN respectively, the following steps will be carried out

- The dialed number will first be analyzed by the call manager cluster present in the mobile node segment (aircraft).
- The cluster will know (numbering plan) that this number matches to the phone C or phone D present in the home segment or PSTN network, it will forward the call through the satellite link to the call manager cluster present in the base network which will then complete the call.

Even though the above network works effectively in connecting and delivering the voice traffic, it is not very cost effective. The call manager cluster comprises of a publisher and a subscriber which are basically servers. The lowest capacity server which Cisco offers (MCS 7815-I2) is capable of handling 300 IP phones which is sufficient for mobile node segment. But as the call manager cluster design needs a publisher and a subscriber each aircraft will need at least two of these servers. Each server cost $4000 and the call manager software version 4.2 cost $2000. As a result the deployment of each cluster will cost $12000. Since the cluster has to be present in each aircraft it may not be a cost effective solution.

In order to make this network design cost effective the call manager cluster in the mobile node segment (aircraft) can be replaced by call manager express(Alexander, 2005). Cisco call manager express is a solution embedded in the Cisco internet operating software (IOS) software that provides call processing to IP phones. Since the feature is embedded in software it can be implemented on different kind of routers. For this network design the most efficient router on which call manager express can be run is Cisco 3745 multi service router which roughly cost around $2000 in the market and is capable of handling 200 IP phones.

4.2.2 Network Design 2 (Regions)

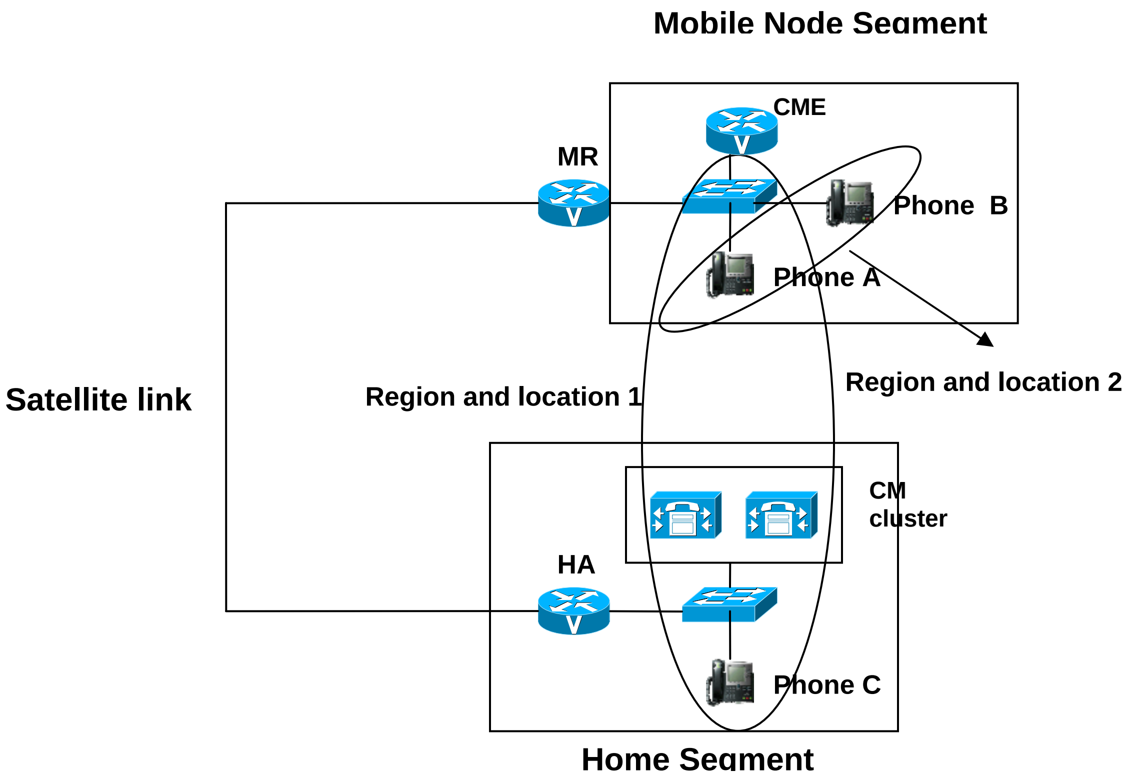

Figure 12 Network design 2

In this design, to make the network cost effective call manager cluster in the node segment is replaced by the call manager express router. As the call manager express has comparable capabilities to that of a call manager cluster, the network design still effectively uses the satellite link's bandwidth by not allowing the internal call signals to pass through the satellite link to the base call manager cluster for digit analysis.

However, the above design fails to provide good quality of voice when the satellite link is over subscribed through voice calls. By default call manager cluster and call manager express are configured to compress voice packets through G.711 codec. As a result when call is made to an IP phone in the base network or analog phone, each voice stream that is generated will use 64 Kbps of bandwidth including overhead. Since the satellite bandwidth is limited it can provide few number of good quality voice calls. If the satellite link is oversubscribed by another call the quality of voice for all the calls established will be degraded as the link will provide less bandwidth to previously established calls in order to squeeze the present call.

To resolve this issue concept of regions (Alexander, 2005) is implemented in the present design. Regions basically allow one to constrain the codecs selected when one device calls another. Thus regions can be used as a way of providing higher voice quality at the expense of network bandwidth for a preferred class of users. In this network design preferred class of users are the passengers calling the base network PSTN. Thus through this concept depending on the destination of the call compression type can be chosen. With respect to the above design when phone A calls phone C or analog phone it is configured to use G.729 codec and when it calls Phone B network uses G.711 codec as the bandwidth is unlimited internally.

The above design does not solve the over subscription problem effectively but surely increases the capacity of the satellite link to provide good quality to large number of calls when compared to the network design 1

To address the oversubscription problem completely in this network design, location (Alexander, 2005) concept along with the regions was used. This concept represents a form of admission control. With each location a specific amount of bandwidth available is specified. When the call is made between the IP phoned in the same location call manager allows users to place an unlimited number of calls. But when the call is made to the other location call manager temporarily deducts the bandwidth associated with the selected codec from the inter location bandwidth remaining. When a user's call terminates, call manager returns the allocated bandwidth to the pool of available bandwidth. As a result when all the bandwidth is allocated no more calls can be made through.

This design not only provides a cost effective implementation but also provides an efficient way to handle the voice calls through satellite link. However this network design may be difficult to be implemented on the aircraft because the location based call admission control mechanism is topologically ignorant it has only one bandwidth counter for all inter location calls meaning that all calls from one location to any other location must traverse only one logical network link which is not possible in the aircrafts deployment enviourment.

To resolve the caveats in the Network Design 1 & 2 gatekeeper concept is introduced in Network Design 3.

4.2.3 Network Design 3

Effectively using the satellite link bandwidth is the key for the proposed network. In this design the concept of gatekeeper is used. Gatekeeper provides call admission control in the same way as location does but it is not implemented as a configuration on call manager or call manager express. It is a feature embedded in the Cisco internet operating software (IOS) software. As a result of this it does not have vulnerability as the location feature has in the above network

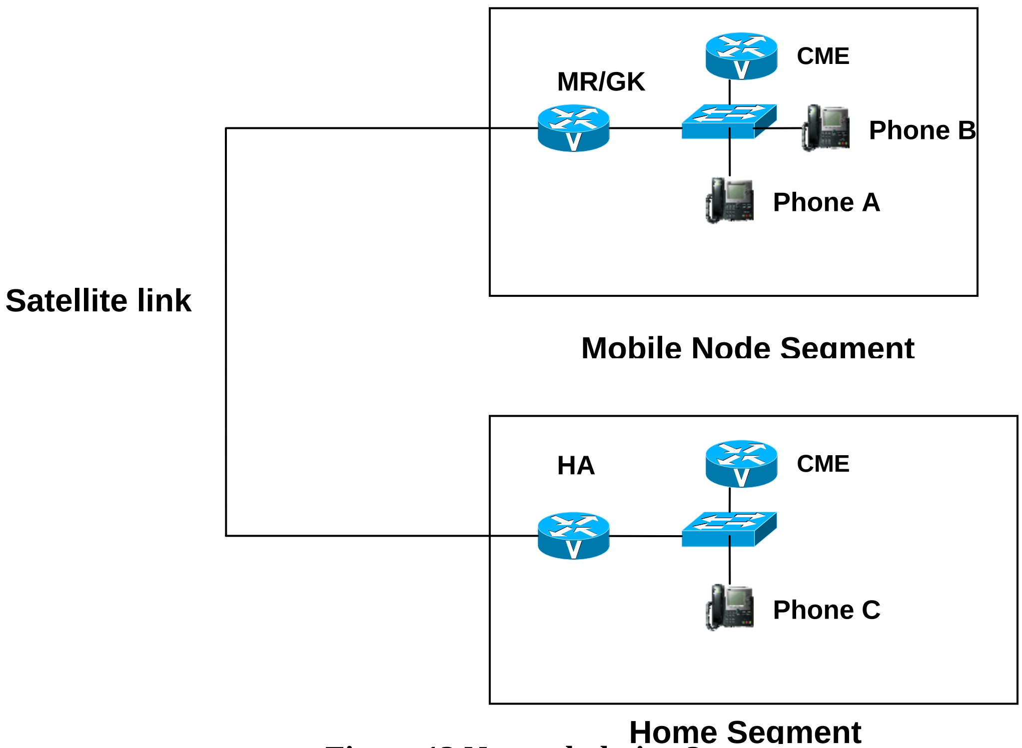

Figure 13 Network design 3

. In this design (Figure 13) the router which is acting as a gateway in mobile node segment is configured to be as a gatekeeper. When the call is made internally that is in the mobile node segment (phone A and phone B) call manager express first analyzes the digits and decides that the dialed number is internal and completes the call. When a call is attempted to the IP phone on the base station (phone C) or PSTN call manager express first asks the gatekeeper if the call can be placed. If there is enough bandwidth, the gatekeeper grants admission. If admission is granted, call setup begins and the call manager on the other side of the call must request admission. If the gatekeeper determines that there is enough bandwidth, admission is granted and the call setup can complete. This allows the voice quality to be maintained. The call that may oversubscribe the satellite link will be rejected by the gatekeeper.

However this design works well only if one aircraft is connected to the base network. This is mainly because for the gatekeeper to work properly the only requirement is that the gatekeeper should be located in such a way that it is able to reach the entire call manager cluster and call manager express in aircrafts through IP path. Secondly gatekeeper has a capability to provide the destination IP address to which the call should be sent. As a result of this having a gatekeeper in a centralized location will totally eradicate the formation of multiple inter cluster trunks thus making the design more efficient. If the gatekeeper is not present an inter cluster trunk must be created between each cluster which is not possible in this design thus a gatekeeper trunk configured in centralized location will carry all the calls destined to any of the call manager clusters.

Next chapter explains in detail about the final network design, the experimental setup and results obtained for different scenarios in this design.

CHAPTER 5

EXPERIMENTAL SETUP & RESULTS DISCUSSION

To study the effect of mobility on call setup time and quality of voice between the mobile network and the base station two scenarios were considered a) Mobile network when it is in the home network and b) Mobile network when it is registered with the foreign network. In both of these scenarios the above parameters were analyzed by changing the codec type and voice traffic rate. The detail of the test bed and the way the experiment was conducted is explained in the following section.

5.1 Equipment Used (Cisco)

For the node segment two 3640 Cisco router's were used one for the call manager express running version 3.1 and the other as a wide area network (WAN) gateway. These two routers along with the Cisco IP phones (Model No. 7960) were connected to each other with the help of a 3560 Cisco switch with inline power capability.

In the Mobile segment the same 3640 Cisco router which was acting as a WAN gateway in the node segment was used as a mobile router. A Cisco 2600 series router with PMOD feature built in its internet operating system (IOS) image acted like a satellite link which connects the node segment to home segment through a 3620 Cisco router acting like a FA.

For the home segment, Cisco 3620 router acted as a HA and windows 2000 server was used as a TFTP server and as a call manager server running 4.0(2a) versions. The router and the server along with the two Cisco IP phones (Model No. 7960) were connected to each other through a 3560 switch with inline power capability. In order to simulate the PSTN network an analog phone was connected to a Cisco 1700 series router running 12.2(17)T IOS image.

Quality of voice was measured through the parameter like end-to-end delay and percentage jitter and perceptual speech quality measurement (PSQM) scores. Parameters like jitter and end-to-end delay were measured through an inbuilt feature present in the Cisco IOS software known as call generator. 3620 router was used for this purpose and the PSQM score was determined with the help of sun machine running this algorithm.

5.2 Scenario 1 Network Setup

In this scenario, as shown in the Figure 14 the mobile segment is considered to be in its home network. As a result of this the mobile router acts as normal IP WAN gateway for the mobile node segment and communicates with the home network in the same way as normal router would communicate with the other router when transferring packet from one network to the other.

The Cisco 3640 router acting like a mobile agent does not have any wireless interface module in it and to maintain connectivity to the home agent a serial link was used. The bandwidth of the serial link was configured in such a way that it simulated as a wireless link. The call manager express present in the mobile node segment was configured to manually register IP phone A. In order for the IP phone A to call the Phone B in the home segment the call manager express was configured with a inter cluster trunk pointing towards the call manager and the gatekeeper in the home segment.

In the home segment the call manager was also configured to manually register IP phone B. In order to maintain the audio quality over the IP WAN link between the two segments, gatekeeper concept was introduced. Instead of using one more router as a gatekeeper the Home Agent (HA) router was configured as the H.323 gatekeeper. Without this configuration the audio quality would have degraded with oversubscription. The gatekeeper helps maintain the audio quality by limiting the number of calls. In order for phone B to communicate with the phone A call manager cluster was also configured with a inter cluster trunk pointing to the H.323 gatekeeper and call manager express (CME) in the mobile node segment.

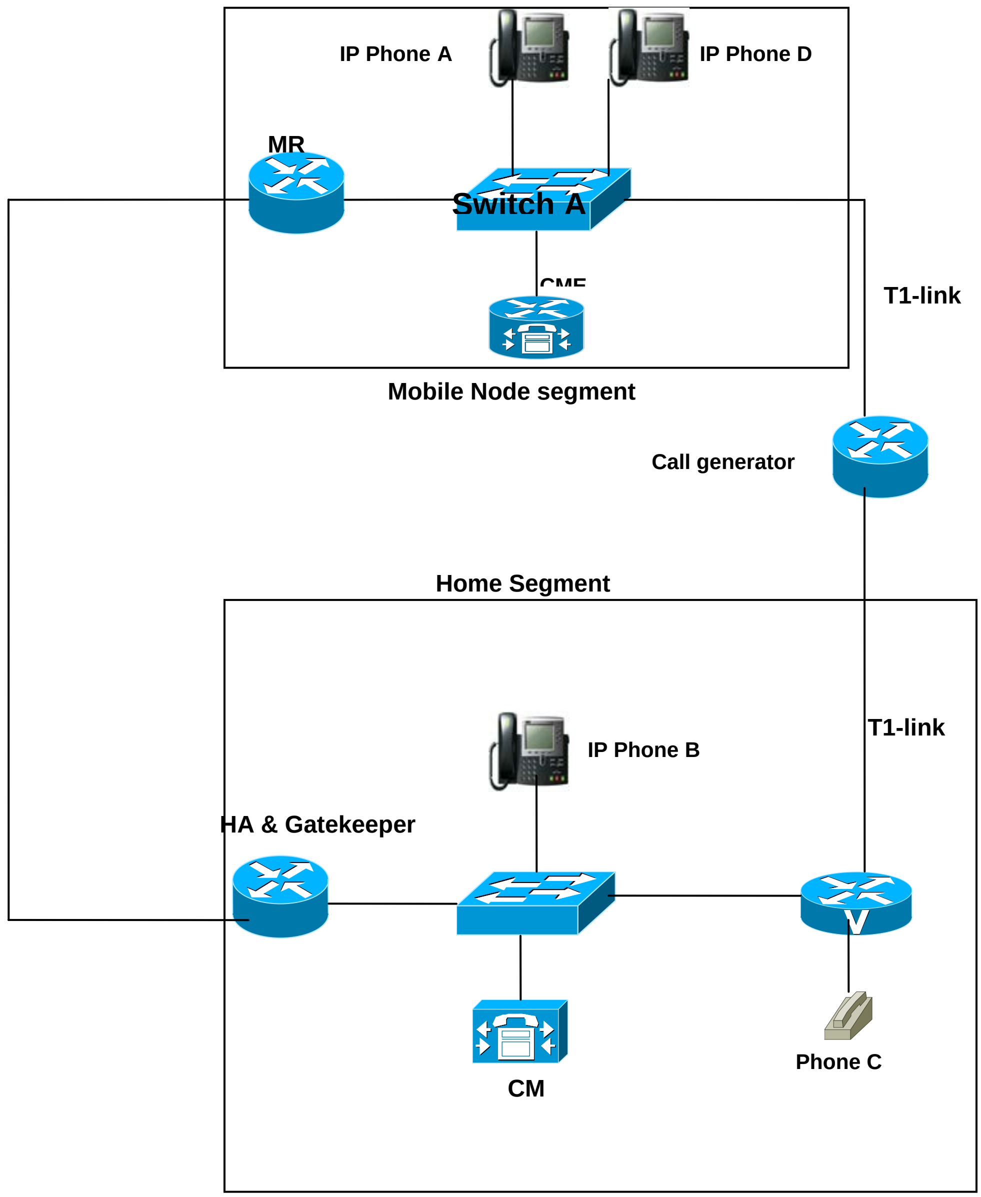

Figure 14 Test bed when mobile node network is in its domain

In order to simulate a call to PSTN from aircraft a Cisco 1700 series router was configured as a H.323 gateway in such a way that when a 9.1 [2-9]xx .[2-9]xx. xxxx number pattern is seen by the gateway it forwards the call to the PSTN network. Moreover the call manager was also configured the same way.

5.2.1 Scenario 1 Simulation Setup & Results

The efficiency of the setup above is tested by measuring the following parameters.

- Call setup time between the phones A,B,C and D
- Type of codecs to be used between the phones.
- End-to-End delay and jitter faced by the voice packets.

5.2.1.1 Call Setup Time Parameter

As the phones communicate with the call manager through the skinny control client protocol (SCCP) protocol one way that the call setup time can be measured is by noting the time of the first “off hook” message sent by the IP phone to the call manager and differentiating it by the time when the call manager or call manager express gives a signal “start media transmission” back to the IP phone connected to its domain. The significance of this signal is that the call manager has negotiated the resources and codec between the two phones and has established a connection between the calling party and the called party phone. After this signal real time protocol (RTP) stream will be generated to send the voice traffic.

In order to check the debug messages the call manager was configured in a way that it could trace all the detailed signaling between the IP phone and intermediate devices. As shown in the table 1 the call setup time between the IP phones in the mobile node segment that is phone A and phone D, IP phone between the mobile node segment and the home segment that is IP phone A and IP phone B and the IP phone between the mobile node segment and the analogue phone that is IP Phone A and phone C was noted.

TABLE 1

CALL SETUP TIME WHEN MOBILE NODE SEGMENT IS IN ITS DOMAIN

Call From	"Off hook" instance	Start media transmission (sec)	Call Setup Time (Differential of off hook instance and Start media trans.) (sec)
Phone A – Phone D	19:31:04.149	19:31:08.508	4.359
Phone A – Phone B	20:38:52.586	20:38:58.430	5.844
Phone A – Phone C	20:56:28.806	20:56:34.166	5.360

As seen in the TABLE 1 the call setup time is minimal between phone A and phone D as they are connected to the same call manager express. When phone A calls to phone B which are on two different call manager cluster domains all the call setup signals pass through the inter cluster trunk present between the two call managers and the gatekeeper. As the mobile node segment is in the home network and communicates directly to gatekeeper without the use of Mobile IP protocol the call setup time between the two IP phones is fairly minimal. Analyzing the call setup time between the phone A and the PSTN Phone C in TABLE 1, it is seen that the time taken for the call setup was much higher when compared to the call setup time between IP phone A and IP phone D. The two main reason for the extra delay is that since PSTN cannot understand the H.323 setup signaling between the call mangers, it has to convert these signals to some other protocol signals such as MGCP so that the PSTN can understand the signaling and setup the call in addition all the signaling has to pass through inter cluster trunk (ICT).

5.2.1.2 Voice Quality Parameter

In order to analyze the quality of voice and the best codec to be used, parameters like end-to-end delay, jitter and PSQM scores for voice packets were analyzed with the help of Cisco call generator tool and PSQM server respectively. The call generator tool generates voice calls, and once a voice call is established between the networks a standard audio file is played across the network. In this scenario the calls were originated from the call generator router and terminated on a different T1 interface of the same router. As the calls are generated by the call generator router the PSQM server which is connected to the call generator analyzes the speech quality and gives a numeric score ranging between 0 to 6.5 (lower number indicating better quality of voice). Rating scale for PSQM is shown in TABLE 2

TABLE 2

VOICE QUALITY RATING SCALE

PSQM Score	Quality of Voice
0	Excellent
Less than 2	Good
2 – 5	Might be acceptable
Greater than 6.5	Bad

For measuring the above parameters the codec type was varied along with the number of voice calls at any given amount of time. One end of the Cisco call generator was connected through the T1 link to call manager express (CME) T1 interface and the other end was connected to the voice gateway T1 interface which is present in the home segment In this practical scenario, when the mobile node segment is in the home network it is assumed that the roaming interface of the mobile router would maintain an 802.11 b wireless connection with the gatekeeper. To simulate the above assumption serial interface is configured for a bandwidth of 54 Mbps.

End-to-End delay and jitter parameters for the network were measured by using different codec's like a) G.729 b) G.723 and c) G.711 and for each codec chosen the calls were increased consecutively.

G.729 codec is configured between the devices and gatekeeper is set for the 11 Mbps allowed bandwidth between the links. Now when the call generator establishes a call through call manager express (CME) to call manager first the H.323 endpoint (call manager express) ask for permission from the gatekeeper to admit calls into the network by giving the information of the called and the calling party. The gatekeeper checks for the amount of bandwidth between the link and if enough bandwidth is available, the gatekeeper admits the call and deducts 20 Kbps (8 Kbps (Payload for the codec) + 12 Kbps (overhead seen by the gatekeeper) = 20 Kbps) of bandwidth from the wide area network (WAN) link.

Keeping the above explanation into account the calls from the call generator were incremented randomly and the statistics collected for the above discussed parameters were recorded.

As seen from the TABLE 4 the quality of voice is good with acceptable delay as the PSQM score is less than 3. The two main reasons for good quality of voice is that as the mobile node network is in its home network there is not much delay faced by the voice packets and there is unlimited bandwidth available which can handle many calls at a time.

TABLE 3

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.729 CODEC AND IS IN HOME NETWORK

Number of Calls	Delay			
	Channel	**Min**	**Max**	**Avg**
1	1 --- 1	86	98	94
	24 --- 24	95	107	101
3	1 --- 3	89	103	93
	24 --- 26	94	108	104
5	1--- 5	92	106	97
	24 --- 28	101	111	106
10	1 --- 10	95	106	102
	24 --- 33	95	115	108
15	1 --- 15	93	104	105
	24 --- 38	105	112	107

TABLE 4

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.729 CODEC AND IS IN HOME NETWORK

Number of Calls	PSQM Scores
1	1.832
3	1.886
5	1.953
10	2.167
15	2.224

In the scenario were G.723 codec was used call manager express and call manager were configured in such a way that when calls were made across them they used G.723 codec. As per the above scenario same configuration was maintained for the gatekeeper and statistics for the same parameter as discussed above were collected which are shown in TABLE 5.

TABLE 5

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.723 CODEC AND IS IN HOME NETWORK

Number of Calls	Delay			
	Channel	Min	Max	Avg
1	1 --- 1	83	97	89
	24 --- 24	80	106	86
3	1 --- 3	86	99	92
	24 --- 26	85	107	101
5	1--- 5	89	96	93
	24 --- 28	92	110	106
10	1 --- 10	95	87	92
	24 --- 33	106	109	105
15	1 --- 15	96	97	96
	24 --- 38	111	112	110

TABLE 6

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.723 CODEC AND IS IN HOME NETWORK

Number of Calls	PSQM Scores
1	1.695
3	1.876
5	1.732
10	1.885
15	1.976

As per the TABLE 6 for G.723 codec also the quality of voice is good as the PSQM score is less than 2. The reason for the quality of voice being good is same as explained for the G.729 codec.

For the G.711 scenario everything was configured same as the above two scenarios except for the codec configuration. Here the call manager and the call manager express were configured to use G.711 codec. The data which was collected for different calls is shown in TABLE 7.

TABLE 7

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.711 CODEC AND IS IN HOME NETWORK

Number of Calls	Delay			
	Channel	Min	Max	Avg
1	1 --- 1	86	108	92
	24 --- 24	95	108	102
3	1 --- 3	91	101	95
	24 --- 26	98	118	107
5	1--- 5	94	109	94
	24 --- 28	96	122	109
10	1 --- 10	93	105	97
	24 --- 33	104	120	112
15	1 --- 15	92	110	101
	24 --- 38	108	124	115

TABLE 8

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.711 CODEC AND IS IN HOME NETWORK

Number of Calls	PSQM Scores
1	1.587
3	1.642
5	1.773
10	2.221
15	2.567

For this codec also the quality of voice is good (as PSQM score is less than 3).

When comparing the statistics of the above scenarios it is seen that although the quality of voice remains almost the same for all the three codec's but the number of calls passed through the gatekeeper maintaining the same quality of voice is highest for G.729 as well as G.723 codec. The reason being is that as compared to all the above codec G.723 and G.729 uses least amount of over head that is 20 Kbps as seen by the gatekeeper.

5.3 Scenario 2 Network Setup

For scenario 2 as shown in the figure 15 the mobile node segment is considered in the foreign network.

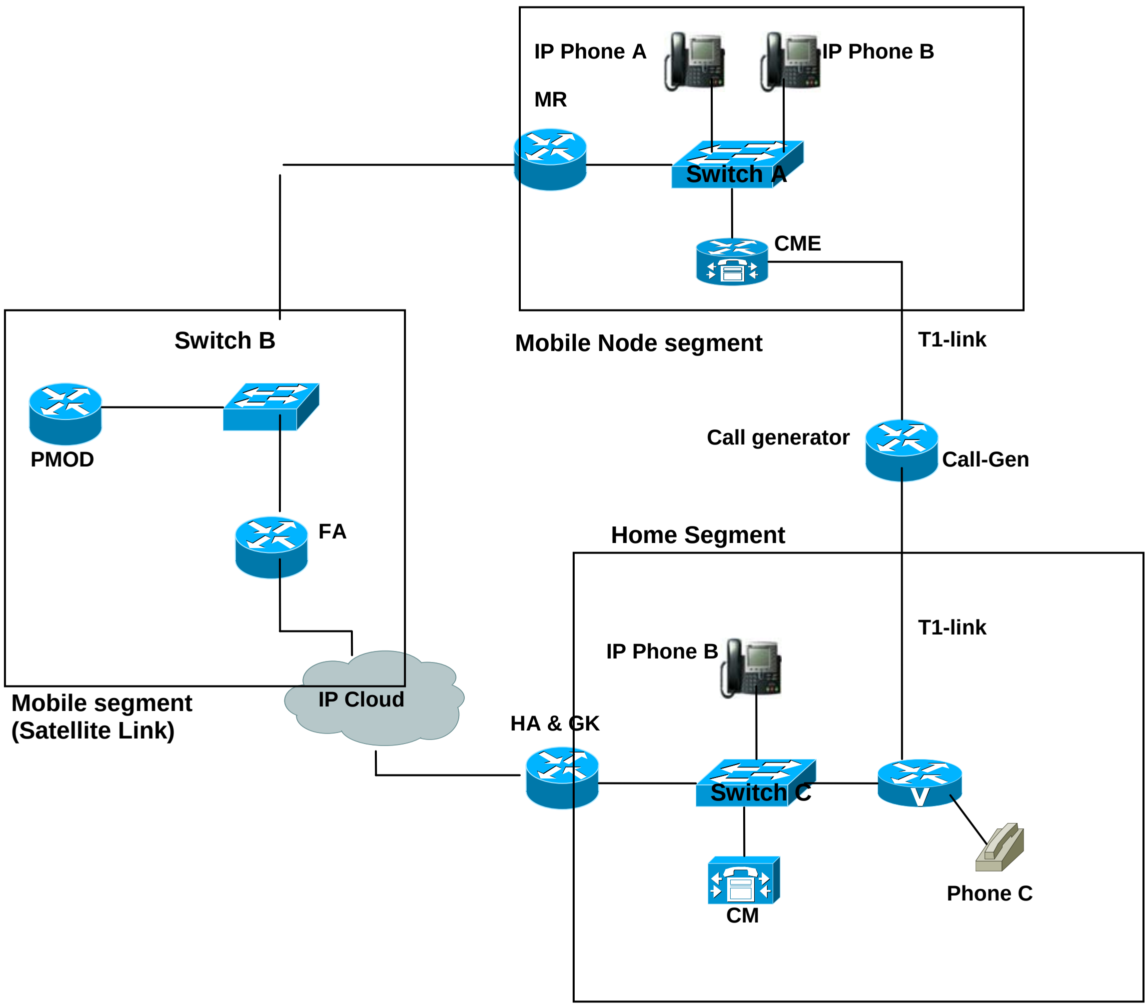

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Figure 15 Test bed when the mobile node network is in foreign domain.

In order to maintain connectivity with the home segment Cisco implementation technique of mobile router was used. In this technique when the mobile router detects it is in a foreign network it forms two tunnels one between the home agent and foreign agent and the other between the home agent and the foreign agent. In the real world situation when aircraft is in the foreign network it would maintain connectivity with the base home segment via the satellite link. In the lab to simulate this environment the roaming interface of mobile router was connected to a switch in the mobile segment which intern connected the foreign agent router and the router which acts like a satellite link. In order to make the other router act like a satellite link P-MOD feature present in the internet operating software (IOS) image was enabled on the Cisco router. By activating this feature the router intern acts like a bridge and tries to simulate the satellite link by introducing the same amount of delay and packet drop in the link as observed by the real satellite link.

Mobility in the scenario 2 was simulated by connecting the mobile router roaming interface into the same virtual local area network (VLAN) of switch B where the foreign agent interface and PMOD router interface is connected. As soon as the mobile router roaming interface was connected to the switch it discovered that it had moved to the foreign network domain because of the advertisements sent by the foreign agent at regular intervals. After this it established connectivity with the home segment by forming a dual IP-in-IP tunnel. But the disadvantage with this approach is that a connection is first broken and another connection is made with the home network. This leads to a short period of time when mobile router is in an isolated state. The internal setup for the mobile node segment and the home segment is same as explained in the scenario 1 section.

5.3.1 Scenario 2 Simulation Steps and Results

In this scenario were mobile node segment is in the foreign domain, it communicates with the base station through the satellite link using Cisco mobile IP protocol. The efficiency of the network is again tested by measuring the same parameters as did in the scenario 1 which are as follows.

- Call Setup time between the phones A,B,C and D
- End-to-End delay and jitter faced by the voice packets.
- Best codec to be used for transferring the voice traffic.

5.3.2 Call Setup Time Parameters

Here also the call setup time was measured by noting the time of the first SCCP "off hook" message sent by the IP phone to the call manager and differentiating it by the time when the call manager or call manager express skinny client control protocol gives a signal "start media transmission" back to the IP phone connected to its domain. To check the debug messages the call manager was configured in the same way as in scenario one so that it could note all the detailed signaling between the devices.

As shown in the TABLE 9 the call setup time between the IP phones in the mobile node segment that is Phone A and Phone D, IP phone between the mobile node segment and the home segment that is IP phone A and IP phone B and the IP phone between the mobile node segment and the analogue phone that is IP Phone A and phone C were noted.

TABLE 9

CALL SETUP TIME WHEN MOBILE NODE SEGMENT IS IN FOREIGN DOMAIN

Call From	Off Hook Instance (sec)	Star media transmission Instance (sec)	Call Setup Time (SEC)
Phone A – Phone D	14:06:54.285	14.06.58.743	4.458
Phone A – Phone B	14:38:25.652	14:38:35.916	10.264
Phone A – Phone C	14:58:12.325	14:58:22:637	10.321

Comparing the call setup time from phone A to phone D with the TABLE 2 we see that the setup time remains almost the same. This is mainly because the signaling is with in the call manager express domain. In comparing the call setup time between the IP phones A and IP phone B with the previous scenario setup time we see that there is large amount of increase in time, this is mainly because when the mobile node segment is in the foreign network it has to build up a dual IP-in-IP encapsulation tunnel to maintain connectivity with the base station. Thus the encapsulation overhead and the large round trip time increases the call setup time. Same condition are faced when Phone A calls Phone C which is in the PSTN thus showing that there is an increase in call setup time.

5.3.3 Voice Quality Parameters

To analyze the quality of voice, and the best codec to be used, same parameters like end-to-end delay, jitter and PSQM score for voice packets were analyzed with the help of Cisco call generator tool and PSQM server. Here the call were generated over the path CME-MR-PMOD-FA-GK-GATEWAY and terminated on a different interface of the call generator. The bandwidth of the satellite link PMOD was configured in such a way so that it acted like a satellite by producing the delay of 600 ms and producing error by dropping one in 10 voice packets.

The end-to-end delay and jitter parameters for the present network was also measured by using same codecs as in scenario 1 such as a)G.729 b)G.723 and c) G.711 and for each codec chosen, the calls were increased randomly. The PSQM score for each of these calls was measured through the PSQM server which was connected to the call generator router in the same way as configured for the scenario 1.

In this scenario when the mobile node segment is in the foreign domain, G.729 codec is configured between the devices and gatekeeper and the bandwidth is set for 250 Kbps which represents the bandwidth offered by a satellite link. Here also when the call generator establishes a call through call manager express to call manager first the H.323 endpoint (call manager express) ask for permission from the gatekeeper to admit calls into the network by giving the information of the called and the calling party. The gatekeeper checks for the amount of bandwidth between the link and if enough bandwidth is available, the gatekeeper admits the call and deducts 20 Kb for that call from the pool of 250 kb.

Keeping the above working into account the calls from the call generator were incremented one by one in the same way as done in the previous scenarios. The statistics which were collected for the parameters discussed are shown in TABLE 10.

TABLE 10

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.729 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	**Delay**			
	Channel	**Min**	**Max**	**Avg**
1	1 --- 1	435	815	543
	24 --- 24	522	782	645
3	1 --- 3	458	739	547
	24 --- 26	557	789	643
5	1--- 5	432	719	554
	24 --- 28	561	838	655
10	1 --- 10	435	744	542
	24 --- 33	551	832	658
12	1 --- 15	427	752	543
	24 --- 38	559	835	656

TABLE 11

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.729 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	PSQM Scores
1	4.258
3	4.248
5	4.321
10	4.467
12	4.612

As seen from the TABLE 11 for 12 calls the quality of voice is acceptable as the PSQM score is less than 5. But when the 13^{th} call is initiated by the call generated it does not go through as the gatekeeper does not allow this call because of insufficient bandwidth. On the contrary if gatekeeper was not used then as the calls increase the traffic on the limited bandwidth link would increase resulting in degrading the quality of voice for all the 12 calls.

To obtain the results for this scenario the call manager and call manager express were configured in such a way that while processing the voice data they used G.723 codec same configuration was maintained for the gatekeeper and statistics for the same parameters were collected which are shown in TABLE 12.

As per the TABLE 13 it is seen that when the bandwidth of the satellite is maintained for 250 Kbps only for 12 calls the quality of voice is acceptable as the PSQM score is less than 5. But here also when the 13^{th} call was initiated it was not completed as the gatekeeper did not allow this call to go through because of the insufficient bandwidth. As per all the above scenario same is true for this scenario also that is if the gatekeeper was not used then as the calls increase the quality of voice of all the calls including the previously established calls would degrade drastically.

TABLE 12

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.723 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	Delay			
	Channel	Min	Max	Avg
1	1 --- 1	419	808	537
	24 --- 24	510	775	639
3	1 --- 3	451	732	541
	24 --- 26	561	792	647
5	1--- 5	438	735	557
	24 --- 28	575	853	661
10	1 --- 10	442	749	546
	24 --- 33	547	828	652
12	1 --- 15	431	756	548
	24 --- 38	556	831	655

TABLE 13

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.723 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	PSQM Scores
1	3.918
3	4.224
5	4.572
10	4.421
12	4.551

For the G.711 scenario also everything was configured in the same way as the above scenarios except for the codec configuration. Here the call manager and the call manager express were configured to use G.711 codec. The data which was collected for different calls is shown below.

TABLE 14

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.711 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	Delay			
	Channel	Min	Max	Avg
1	1 --- 1	429	707	551
	24 --- 24	558	821	652

TABLE 15

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.711 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	PSQM Scores
1	4.326

Here only for one call the quality of voice is acceptable (as PSQM score is less than 4). But after the first call no further calls were executed by the gatekeeper due to the lack of bandwidth between the call manager express and call manager trunk As compared to the G.729 and G.723 codec for G.711 codec the gate keeper allowed only one call because the when the gatekeeper allows a G.711 codec call to pass through it deducted 128 Kbps bandwidth from the pool of 250 Kbps bandwidth.

For experimental purposes when the bandwidth between for the serial link was increased from 250 Kbps to 500Kbps gatekeeper allowed 3 calls to go through which is shown in the TABLE 16.

TABLE 16

DEALY SEEN IN THE VOICE PACKETS WHEN THE MOBILE NODE NETWORK IS CONFIGUED FOR G.711 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	Delay			
	Channel	Min	Max	Avg
1	1 --- 1	437	712	556
	24 --- 24	564	829	658
3	1 --- 3	438	718	553
	24 --- 26	571	834	662

TABLE 17

PSQM SCORES FOR THE CALLS WHEN THE MOBILE NODE NETWORK IS CONFIGURED FOR G.711 CODEC AND IS IN FOREIGN NETWORK

Number of Calls	PSQM Scores
1	4.323
3	4.329

When the data for quality of voice and number of calls is compared while the mobile node segment is in the home network and away from the home network it is seen that the quality of voice is not good but acceptable when the mobile node segment is in the foreign network. This is mainly because of the different kind of delay such as processing delay, packetization delay and transmission delay a voice packet has to suffer when it is transferred from mobile node network to the home network.

In comparing the statistics of the scenarios when the mobile node network is in the foreign domain it is seen that although the quality of voice remains almost the same for all the three codec's when sufficient bandwidth is available, the number of calls passed through the gatekeeper maintaining the same quality of voice varies for different codecs as different codecs consume different bandwidth.

Thus for the proposed network design the best codec that can be used for transferring voice calls is G.729 or G.723 codec.

CHAPTER 6

CONCLUSION AND FUTURE WORK

6.1 Conclusion

Providing passengers the ability to call any part of the world is one of the prime interests of aviation industry today. The existing infrastructure focuses on providing these services through cell phones which pose a considerable amount of risk as they would interfere with the aircrafts navigation system. Currently, efforts are being made to develop an infrastructure which will allow callers to call over the IP network. This research work made an effort to develop a voice network that allows the passengers in the aircraft to talk to their fellow passengers and also allows them to talk to persons on the ground via IP phone at a much lower cost. Secondly as no cell phones are used it completely eliminates the problem of interference.

During the analysis of the quality of voice traffic of the proposed network it has been observed that the quality of voice for the calls when the aircraft is in home domain is excellent but when the aircraft moves into the foreign domain the quality deteriorates. However the PSQM scores show that it is still in an acceptable range. This means that the calling and the called party can have a decent conversation. The other advantage of this network is that, with the sufficient bandwidth provided by the satellite video traffic can also be passed from the aircraft to the ground without modifying the network design.

6.2 Future work

The present thesis focuses on design of an IP network for only one aircraft. Further research can focus on modifying the present IP network to accommodate multiple aircrafts using mobile routers. Secondly, call manager express in the aircraft and the base call manager cluster communicate with each other through inter cluster trunks which are in turn is a transport control protocol (TCP) connection, an attempt can be made to see how performance enhancements proxies (PEP) can play a role in enhancing the call setup time.

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