VoIP call routing Optimization in WLAN and Cellular Integrated Environment

A Dissertation submitted to the

Jawaharlal Nehru University, New Delhi,

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Master of Technology

in

Computer Science and Technology

by

Kuldeep

Under the Supervision
of

Prof. P.C. Saxena

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CERTIFICATE

This is to certify that the dissertation entitled **"VoIP call routing Optimization in WLAN and Cellular Integrated Environment"** being submitted by Mr. **Kuldeep** to the School of Computer and Systems Sciences, **Jawaharlal Nehru University**, New Delhi, in partial fulfillment of the requirements for the award of the degree of **Master of Technology in Computer Science and Technology**, is a record of bonafide work carried out by him under the supervision of **Prof. P.C. Saxena**.

This work has not been submitted in part or full to any university or institution for the award of any degree or diploma.

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DECLARATION

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The matter embodied in the dissertation has not been submitted for the award of any other degree or diploma.

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To God and my Parents

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ABBREVIATIONS

IMTS	Improved Mobile Telephone System
AMPS	Advanced Mobile Phone System
GSM	Global system for mobile communications
GPRS	General Packet Radio Services
TDMA	Time Division Multiple Access
FDMA	Frequency Division Multiple Access
CDMA	Code Division Multiple Access
UMTS	Universal Mobile Telecommunications Systems
MS	Mobile Station
BS	Base Station
BSC	Base Station Controller
BTS	Base Transceiver System
MSC	Mobile Switching Center
PSTN	Public Switched Telephone Network
HLR	Home Location Register
VLR	Visitor Location Register
EIC	Equipment Identity Register
AUC	Authentication Center
RSS	Received Signal Strength
FCC	Forward Control Channel
BCC	Backward Control Channel
VoIP	Voice over Internet Protocol
SS7	Signaling System 7
SIP	Session Initiation Protocol
RTP	Real Time Protocol
WLAN	Wireless Local Area Network
WLAN NIC	Wireless LAN Network Interface Card
VoWLAN	Voice over WLAN
PBX	Private Branch Exchange
MAC	Medium Access Control
CCK	Complementary Code Modulation

BSS	Basic Service Set
ESS	Extended Service Set
AP	Access Point
DCF	Distributed Coordination Function
PCF	Point Coordination Function
PC	Point Coordination
CDMA/CA	Carrier Sense Multiple Access with Collision Avoidance
WMA	Wireless Mobile Station
QoS	Quality of Service
WGSN	WLAN based GPRS Support Node
RNC	Radio Network Controller
UTRAN	UMTS Terrestrial Radio Access Network
GGSN	Gateway GPRS Support Node
PDN	Packet Data Network
SGSN	Serving GPRS Support Node
DHCP	Dynamic Host Configuration Protocol
AAA	Authentication, Authorization and Accounting
SIM	Subscriber Identity Module
IMSI	International Mobile Subscriber Identity
CSP	Call Server with Push
URI	Uniform Resource Identifier
UA	User Agent

ABSTRACT

The increasing popularity of VoIP and WLAN enabled devices has triggered interest in the integration of Cellular and Voice over WLAN (VoWLAN). In this integrated environment, users are equipped with dual-mode mobiles with two interfaces, one for cellular network and other interface for WLAN. For providing seamless service support to users, here, we worked towards reducing the upward vertical handoff latency of VoIP calls from Wireless LAN (IEEE standard 802.11) to Cellular networks which optimize the routing of VoIP call for which the vertical handoff is done in the integrated environment.

The proposed vertical handoff algorithm works in the direction of faster vertical handoffs, thus reducing the power consumption by dual-mode mobile station. Due to fast vertical handover, the voice quality doesn't degrade and the VoIP call dropping probability reduces significantly. The algorithm works by modifying some functioning of Cellular/VoIP gateway and CSP components in the integrated environment.

CHAPTER 1

INTRODUCTION TO CELLULAR AND VOIP

1.1 History of Cellular Radio Networks

Wireless communication started when G. Marconi developed the first wireless telegraph system in 1896. Later in 1946, the first car-based telephone was set up, which used a single channel system. In the 60's the system was improved which used a two-channel system, called improved mobile telephone system (IMTS). IMTS was having limited frequencies so it could not support many users. This problem was solved by the cellular concept, developed in the 1960's and 1970's. In mid 90's, the cellular communications industry has witnessed explosive growth. More users can be supported in such a cellular radio system. It was implemented for the first time in the advanced mobile phone system (AMPS). AMPS is a part of first generation cellular radio systems.

Second generation systems use digital radio transmission for the traffic and the frequency channels are simultaneously divided among several users (either by code or time division). System capacity was further enhanced by use of hierarchical cell structures—in which the service area is covered by macrocells, microcells, and picocells. In the USA two standards are used for second generation systems - IS-95 (CDMA) and IS-136 (D-AMPS). Europe consolidated on one system called global system for mobile communications (GSM). Japan uses a system called personal digital cellular (PDC).

"Generation 2.5" systems implemented some advanced technologies in the 2G networks. Those advanced technologies included high-speed circuit-switched data (HSCSD), General Packet Radio Services (GPRS), and Enhanced Data Rates for Global Evolution (EDGE).

"Third Generation" systems makes use of WCDMA technique which is more efficient than FDMA and TDMA, which are used by first and second generation systems. 3rd generation systems are called Universal Mobile Telecommunications Systems (UMTS).

The exact specifications for the "4th Generation" systems have not been specified, but the recent trend is to integrate various interface techniques. In 4th generation networks, the different networks (GSM/CDMA/UMTS/WLAN) are integrated into IP-based networks.

1.2 Cellular Radio Networks

1.2.1 Fundamentals of Cellular Systems

Cellular concept was a major breakthrough in mobile radio networks. The idea was to replace the single, high powered transmitter (large cell) with many low power transmitters (small cells), with each small cell providing coverage to the small portion of the entire service region.

The size and shape of the cell and amount of resources allocated to each cell dictate the overall performance of the system. The shape of the cell can be circular, hexagonal, square or triangular, with base station at the center of the cell providing radio coverage within the cell. But the actual shape depends on many parameters such as contour of terrain, height of transmitting antenna, presence of hills, valleys, tall buildings and whether conditions. So the actual shape may be a zigzag shape. For simulation and other purposes, using hexagonal as the cell shape, it is possible to divide a larger are into non-overlapping sub-areas of the same shape, so cell shape is taken as hexagonal, illustrated in Fig. 1.

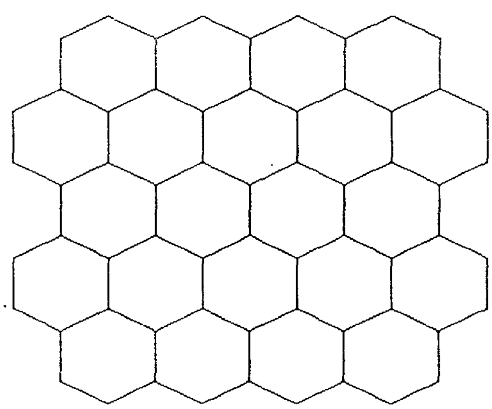

Fig.1: Cellular Network with hexagonal cell size.

To increase the effectiveness of the cellular system, the network implements the bandwidth sharing mechanisms between the users within a cell. The bandwidth sharing within the cells uses the multiplexing techniques like frequency division multiple access

(FDMA), time division multiple access (TDMA), and code division multiple access (CDMA).

In FDMA, the allocated frequency band to a BS is divided among a number of channels, and one channel is allocated by the BS to each active user. In TDMA, one such channel is shared between several users by the time-sharing means. The BS assigns time slots for different users and each user uses the channel in a round robin fashion, but the time sharing is invisible to the users. The CDMA technique shares the channels by use of orthogonal codes. CDMA uses tight power control and synchronization within a cell to provide multiple access to frequency channels. In this technique, one unique code is assigned to each user by the BS. All the users can use the channel simultaneously by encrypting the voice signals with the code. BS uses same code to decrypt the signals from different users and sends them accordingly.

1.2.2 Frequency Reuse

In any radio network, the number of simultaneous calls that may occur depends on the available frequency spectrum and the number of channels. Since cellular networks have limited bandwidth, so these networks rely on allocation and reuse of channels throughout their coverage region.

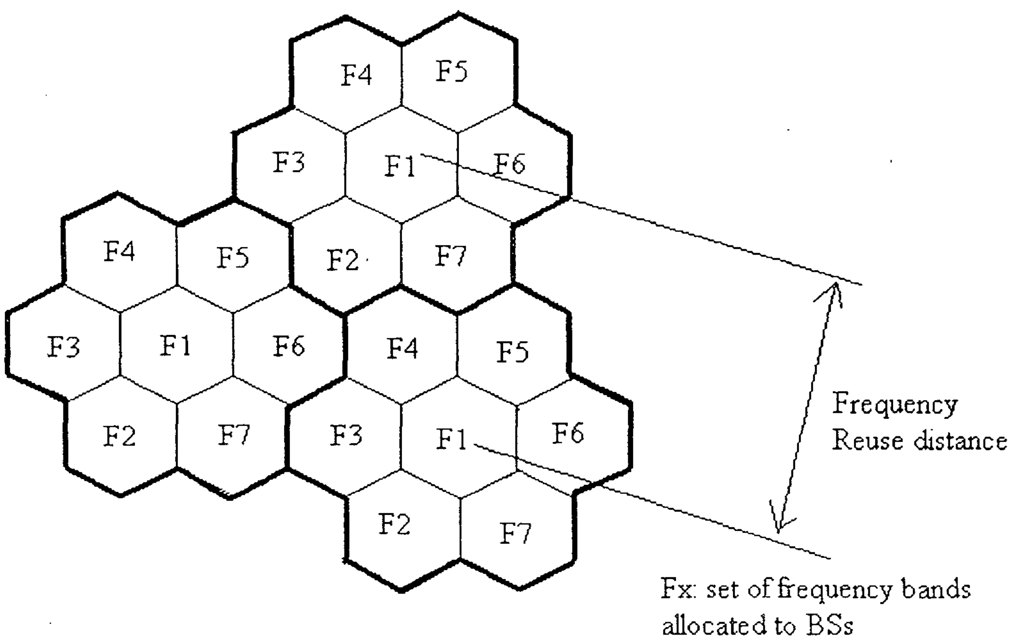

Fig. 2: Frequency Reuse with cluster size 7.

Every base station (BS) is allocated a group of radio channels and antennas of base stations are designed to achieve the proper radio coverage within the boundary of its cell. By limiting the coverage area to within the cell boundaries, the same group of channels are used to cover different cells that are separated from one another so that their signal strength do not interfere with each other. This process is called frequency reuse. Figure 2 illustrates the concept of frequency reuse with a cluster of seven cells and three such clusters with no overlapping area. The distance between the two cells, which are using the same channel, is known as the "reuse distance".

1.2.3 Cellular System Architecture

Generally, the infrastructure of mobile cellular networks is composed of a wired, packet-switched, backbone network and a wireless network. A mobile cellular network typically covers a large geographical service area which is partitioned into many small regions called cells. Each cell is served by a base station (BS), which provides the wireless radio coverage in its service area. The mobile stations in each cell are tied to the base station of the corresponding cell by radio links. These radio links employ either time division multiple access (TDMA) or spread-spectrum code division multiple access (CDMA) techniques. The network may include base station controllers (BSC), which manage a group of base stations or a single BS as well as do radio channel management and handles handoffs. The BS includes Base Transceiver System (BTS), which handles the transmission and reception of signals as both tower and antenna are part of the BTS. The BSs and BSCs are connected to PSTN via mobile switching centers (MSCs) through wired links. The home location register (HLR) and visitor location register (VLR) are two sets of pointers which support mobility. For each MS the HLR-VLR pair is stored at the MSC where the MS is registered. The MSC provides switching the call between different BSs (Handoff), location tracking/updating and call delivery functions by use of HLR-VLR pair, an Equipment Identity Register (EIR), and Authentication center (AUC). Figure 3 shows the Cellular radio networks architecture.

The HLR contains user profile information such as initial home location for billing, the types of service subscribed. VLR basically contains information about the visiting MSC by the MS. Figure shows the architecture of mobile cellular networks.

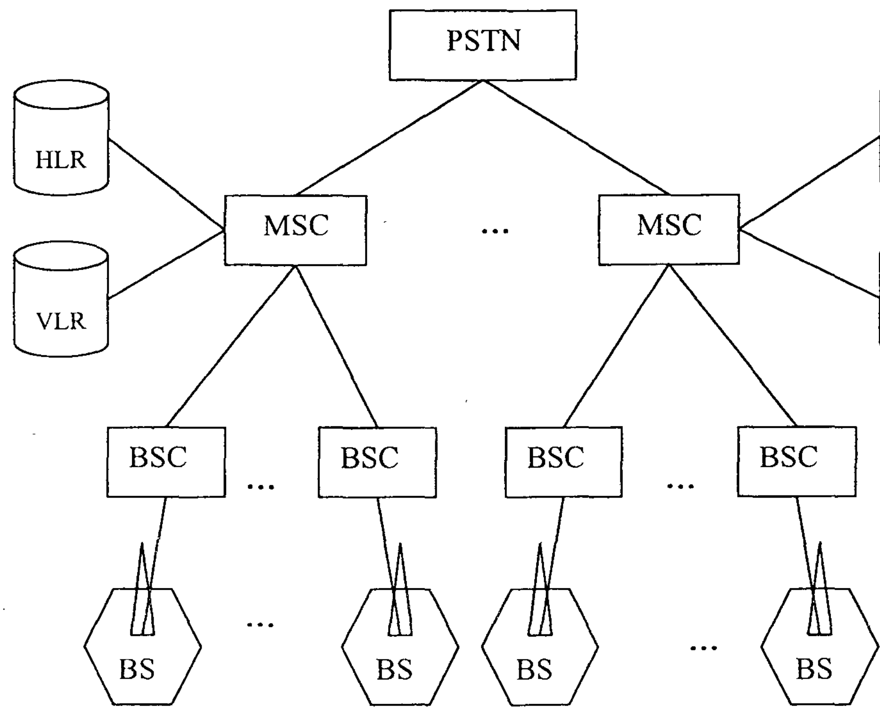

Fig. 3: Cellular network architecture

1.2.4 Handoff

When a MS moves into a different cell while a conversation is in process, it requires a technique called Handoff, in order to provide the call continuity. The new cell has the different frequency range. If there is no pivot channel in the new cell, the call drops due to unavailability of channels for the voice and control signals for the visiting MS. Then depending on the borrowing strategy, the BS can borrow channels from a neighboring cell, and allocate the borrowed channel to the MS for the call to handoff in this cell. In cellular networks, handoff is an important task and it may occur frequently because the cells are being splitting into smaller cells to provide the services to increasing number of users. The handoff process must be imperceptible to the users and performed infrequently as possible. The handoff process involves two steps: Detecting the handoff region and handoff execution. Usually these steps overlap each other. The decision about the handoff to occur or not is taken on the RSS from BSs.

To achieve non-interruptive handoff, the coverage area of two neighboring cells has to be overlapped or at least touch each other. Figure 4 depicts the handoff case where two neighboring cells have a distance of D between their Access Points, and their coverage areas touch each other.

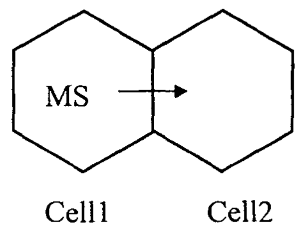

Fig. 4: Handoff when MS moves between cells

When a MS moves from one cell to another, if the MS maintains radio connection with only one BS at a time, then it is hard handoff. Hard handoff is employed by GSM. When the MS creates radio connection with the BS of new cell before breaking its radio connection with the old BS, then the handoff is soft handoff.

1.2.5 Call Setup in cellular Systems

The working of a cellular network can be elucidated on the parameters of one mobile station calling another. Four simplex channels are needed to exchange the synchronization and data between BS and MS, which are shown in Figure 5.

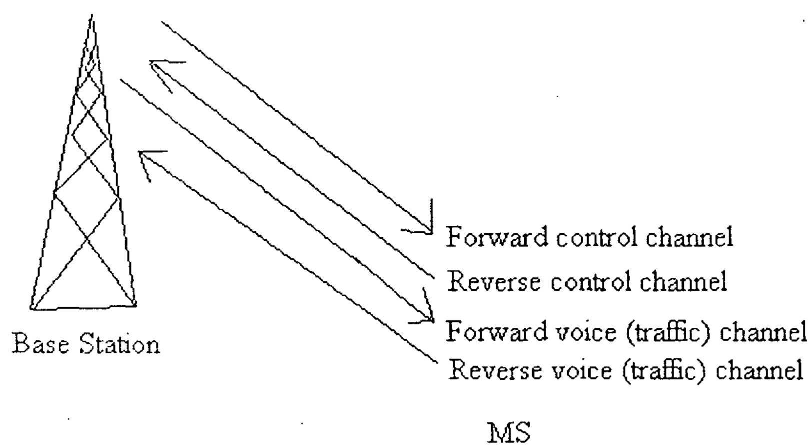

Fig. 5: Four simplex channels between BS and MS

The channels from BS to MS are known as Forward Channels and channels from MS to BS are called Reverse Channels. Control channels are used to exchange control messages between BS and MS, while data/traffic or voice channels are intended for actual data transfer between BS and MS.

The whole process starts with a call-initiation request sent to the corresponding BS by the calling MS on the reverse control channel (RCC). For the validation of the request, it is sent to MSC by the base station. Then through PSTN (Public Switched Telephone Network), the MSC connects to the called party. The process is completed in three steps:

1. The connection of calling MSC with the MSC of the called party.
2. Secondly, instruction to the calling mobile station and corresponding BS by the MSC to switch to voice channels now.
3. Thirdly and finally the calling mobile station is connected to the called station on unused forward and backward voice channels (FVS, BVC).

Forward control channel (FCC) is continually scanned by the mobile stations for paging signals from the BSs. When a MSC receives a request for a connection to a mobile station, immediately, to all base stations in its control, sends a broadcast message containing the number of mobile station which is being called. The corresponding BS which is having the called mobile station in its cell then broadcasted the message on all forward control channels. The corresponding MS acknowledges this paging by identifying itself over the reverse control channel. Then though the BS, the MSC receives the acknowledgement and instruction is sent to switch to an unused voice channels, to the BS and MS. Finally, over the forward voice channel, a data message is transmitted, thereby, instructing the MS to ring. So, extensive control signal processing is needed before the actual call is setup.

1.3 Voice over Internet Protocol

The idea of transporting voice in IP packets (VoIP) was introduced in the mid 90's. VoIP, also called Internet telephony, is a rapidly emerging technology for voice communication that uses the ubiquity of IP-based networks that makes it possible to have a telephone conversation over the Internet or a dedicated (closed) IP network instead of dedicated voice transmission lines. This eliminates the need for circuit switching and the associated bandwidth used for signaling. Instead, a system using packet switching is used. VoIP technology provides a solution for combining voice and data over one network. VoIP is viable and cost-effective alternative to circuit-switched voice network. IP packets carrying voice data are sent over the network only when data needs to be sent, such as when a caller is talking. VoIP traffic does not necessarily have to travel over the public Internet. VoIP applications can also be deployed on private IP networks, such as a company's Intranet (or a telecommunications carrier's IP network).

VoIP technology is useful not only for phones but also as a broad application platform that enables voice interactions on devices such as desktop computers, mobile devices, set-top boxes, gateways, and many devices with applications specific to certain businesses where voice communication is an important feature.

1.3.1 What is VoIP

Voice over IP or VoIP provides the capability to break up your voice into small pieces (known as samples) and place them in an IP packet. The entire process of transmitting a VoIP call can be illustrated as: The analog voice waveform is converted into digital signals, this digitized signals are packetized and sent over the network. The receiver of the VoIP call does the opposite procedure to de-packetize, decompress and finally convert the signal into analog signals that is played back.

1.3.2 Components of VoIP

VoIP is the name of the protocols that are used for Internet telephony. The components of VoIP can be split into four categories:

- Signaling
- Encoding
- Transport
- Gateway Control

Signaling protocols are application layer protocols which are responsible for setting up, maintaining, and tearing down the VoIP call sessions. The analog signals from the human voice are needs to be converted into digital format which is suitable for transmission over the IP network. This process is called Digitization process and is done by Encoding protocols, which are also called Codecs. The Transport protocols are responsible for real-time transport of the voice calls by means of those digitized packets across the IP networks in a manner that produces acceptable voice quality. Gateway control protocols are responsible to convert the IP packets format to some other format because thee voice call might have to be delivered to the PSTN network, as we can make VoIP call from some IP based phone to some PSTN phone. The protocols responsible for this kind of conversion run on the gateway, which connects two different kinds of networks.

SS7 (Signaling System 7), SIP (Session Initiation Protocol) and H.323 are signaling protocols. RTP or Real-time Transport Protocol is a Transport protocol used to provide the transport of audio and video packets over IP network. The Encoding protocols or Codecs used by VoIP are G.711 and G.729, both specified by ITU-T.

SS7 is a set of protocols used for signaling i.e. call setup, maintenance and tear-down of voice calls sessions in PSTN. SS7 is implemented as a packet-switched network and makes use of dedicated link either real or virtual circuits for the signaling. While SS7 provides the call control over the PSTN, SIP and H.323 provide the call control and required signaling over the IP network, so these protocols are used in conjunction with VoIP.

1.3.3 Why VoIP?

VoIP is not a single entity, rather it is a collective term given to protocols, software, and hardware that allow the transmission of voice traffic to traverse packet-switched data networks, instead of Public Switched Telephone Network (PSTN).

VoIP can be compared to PSTN (Public Switched Telephone Network) network because both networks can carry voice, one by using packet-switching and other by means of circuit switching. Although the PSTN is effective and does a good job at what it was built to do (that is, switch voice calls), the PSTN networks have major drawbacks in today's rapidly growing environment to support the business drivers and users. The IP network to transport the VoIP calls uses packet switching, which has advantages over circuit-switching, which is being used by the PSTN network. In circuit-switching, a circuit is maintained for the whole duration of the conversation, thus a large portion of the PSTN network resources is wasted. In contrast, using data networks to deliver voice not only avoids the need for two separate systems for voice and data, but also makes better use of the network's resources. The major problems with the PSTN networks which lead to VoIP are: -

1. PSTN network has the speed limitation of 64kbps
2. PSTN uses analog signaling, which has a lot of impairments.
3. Re-dimensioning of PSTN networks is Expensive
4. Introduction and implementation of new services (like video conferencing, transport of multimedia data) to the PSTN network is not easy.

Where VoIP provides these key benefits:-

1. Cheap long distance calls because VoIP uses the IP networks for the transport of voice in form of data packets. Since the IP network is ubiquitous and provides services almost everywhere, only cost is of using IP network bandwidth and specially handling the voice data packets as real time packets.
2. IP network transports voice as data in form of IP packets which are in digital form, and thus less susceptible to errors.
3. VoIP uses single connection for all services
4. VoIP uses single network as its infrastructure, the IP network, which is easier to manage with respect to the PSTN network.

However, VoIP calls should be given importance while transmission over the IP network, because voice applications are real time, they are intolerant of lengthy delays, packet losses, out-of-order packets and jitter. All these problems sternly degrade the quality of the voice transmitted to the recipient. IP network sends voice as data packets and unlike circuit-switching it provides no timely delivery of packets and not even in the same order they were sent. So, voice data packets are handled like real-time data, whenever they come should be given priority to utilize the required bandwidth. Figure 6 shows the VoIP network.

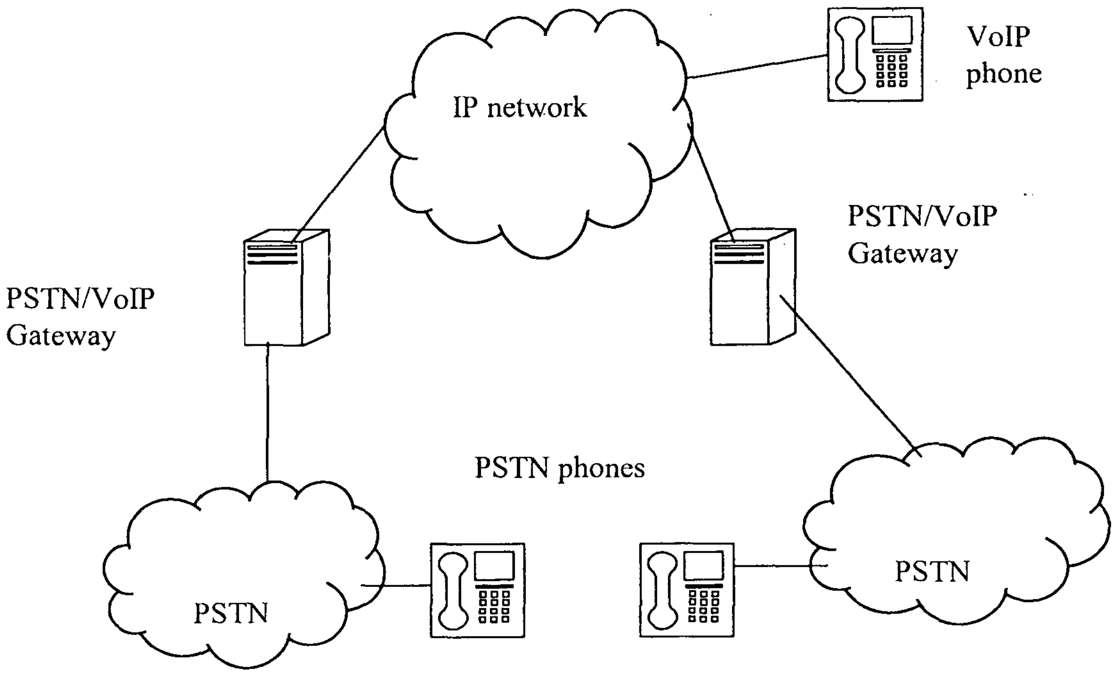

Figure 6: VoIP Network

VoIP can be implemented in several ways. A VoIP application can communicate with a Public Switched Telephone Network (PSTN)-based telephone and vice versa. In this case, some part of the call is routed over the Internet and some part over a dedicated PSTN circuit network Two VoIP applications can communicate directly without accessing the PSTN.

1.3.4 Voice Quality issues

Most data networks devices such as routers and switches are designed to effectively transmit bursty, unpredictable and asynchronous data. As a result there are non-deterministic delays caused by the congestion on the network. But, in case of voice, the traffic is isochronous; it requires timely delivery of a stream of packets to maintain good quality of the voice delivered at the receiver end. Therefore, some method of voice prioritization must be implemented to ensure the timely delivery and good voice quality. To improve the quality of voice, these impairments must be migrated: -

1. **Delay impairments:-** As voice is transported as data packets over IP network, delays may be caused by several reasons like propagation delay, packetization delay, Jitter Buffer delay, and Transport delay.
   a) **The propagation delay** is due to the physical distance between the communicating parties in the call i.e. how long it takes to propagate a signal between them. Over a wireless LAN this kind of delay is not significant because it is proportional to speed of light and can be ignored.
   b) **The Packetization Delay** is due to the time taken to convert analog signals to digital packets and vice-versa. This conversion is done by Codecs. G.711 is a high-speed codec takes about one millisecond to convert analog signals to digital packets, while Low-speed codecs like G.723 takes much longer because they do the compression also to reduce the packet size.
   c) **A Jitter Buffer** is used at the receiver's end and it's function is to smooth the playback of the voice call because VoIP packets do not arrive in sequence. As the packets arrive, they are stored in the Jitter-Buffer memory and then the codec retrieves a packet from this buffer and does its work. The time taken to smooth the playback of audio is the Jitter-Buffer delay.
   d) **Transport delay** is the delay caused by the intermediate networks between the talker and the listener, because every networking device (like router, gateway, and hub) takes some time to transport the packets.
2. **Data Loss impairments:** Data loss impairments are due to the loss of packets in the IP network. Since VoIP packets are transmitted as real-time packets, the re-transmission of lost packets is not feasible, so the lost packets create gaps in the voice. Finally the quality of the audio will be degraded by any lost packet. Some packets arrive too early or too late due to congestion on the network, those packets are also discarded because they may result in unintelligible speech.

CHAPTER 2

VOICE OVER WIRELESS LAN

2.1Introduction to VoWLAN

VoIP over WLAN (VoWLAN) as technology enables IP voice to be sent over an (802.11) WLAN. New types of devices, such as various 802.11 phones have emerged. The proliferation of wireless local area networks (WLAN) in recent years has created a new momentum for research and development in low-range high-bandwidth wireless connections. Voice over Wireless LAN (VoWLAN) is getting great attention from the industry. With the explosion of both VoIP and WLANs, more products are emerging and it is predicted that this technology will be widely deployed in enterprises as well as in homes. VoWLAN is based on two independent technologies:-

- Wireless LAN or IEEE 802.11
- Voice over IP

Wireless Local Area Networks (WLANs) have become very popular in recent years. The Institute of Electrical and Electronics Engineers (IEEE) endorsed the original 802.11 specification as the standard for wireless LANs in 1997. As the IP networks are ubiquitously available, VoIP has become popular. VoWLAN is a natural evolution of Voice over IP.

VoWLAN or Voice over Wireless Local Area Network expands the capability of WLANs or Wireless LANs by allowing voice to be transmitted over the Internet, whereas the WLANs were meant for data connectivity only, by use of Mobile Internet Protocols. So, VoWLAN is the added feature which handles the transmission of voice over the data networks or packet-switched networks.

Basically, VoWLAN systems work in two different ways. One way is to route calls from the wireless mobile station to a VoIP gateway via some access point of the WLAN. The calls are then interpreted between the IP network and the private branch exchange (PBX). Calls from the VoWLAN mobile stations that are made to phones outside the company will to the PSTN (Public Switched Telephone Network) through PBX.

The second method for VoWLAN to work is for software-based phone, also known as softphones, to route calls over the Internet. In this scenario, users can use the softphone on their PDA, cellular phone or laptop to place calls from a location or hotspot that offers a WLAN. The call could be routed anywhere over the Internet, which is almost free; the only cost is to access the IP network's resources. VoWLAN end devices usually connect to a single AP which is similar to BS in cellular network. The AP provides connection to IP network and when VoIP is taking place, the VoWLAN end device will use signaling protocols of VoIP.

Voice over WLAN introduces voice into the packet switching world to be sent over an IEEE 802.11 network (also known as Voice over WiFi). A VoWLAN system works by translating voice calls to IP packets and send these packets over an IEEE 802.11 network. The VoWLAN mobiles having an IEEE 802.11 interface will reassemble those packets and output the audio to the user. On Wireless LAN (an IEEE 802.11 network), data and voice share a common infrastructure. When voice is quiescent, data can utilize the available bandwidth; when voice applications are active, they should be guaranteed the bandwidth required for the transport of VoIP call. The wireless LANs have limitation on the number of simultaneous VoIP calls per Access Point that can be handled. The maximum number of voice call depends on the packetization of the VoIP signals, the geographic distribution of the wireless clients (generally mobiles with WLAN interfaces), and the distance between the wireless clients and the access points of wireless LAN. When the number of simultaneous VoIP calls reaches to maximum per access point, the access point becomes overloaded and the voice call quality degrades. VoIP applications are innately real-time; they tolerate minimal delay in delivery of their packets, and are intolerant of packet loss, out-of-order packets, and jitter. So, the quality of VoIP calls in the Wireless LAN can be measured by packet delay, jitter and loss rate, i.e. how frequently the VoIP packets are lost.

The voice over wireless LAN technology can bring many benefits to the dual mode mobile users, but wireless networks cause some additional difficulties in handling the voice calls, like Latency-induced VoIP performance degradation as user roams. This latency usually occurs because the mobile host moves between different access points of the 802.11 network, and whenever a mobile host attaches to a new access point, re-authentication is required.

2.2 Challenges to VoWLAN

Voice over WLAN (or WiFi) market has a lot of challenges to focus. These are:-

1 Short Battery Life

2 Quality of voice calls

3 WiFi infrastructure

4 Fast handoff

5 Seamless Vertical Roaming

1. **Short Battery Life:** - The radio power interface in the wireless mobile station (WMA) for the wireless connectivity to Access Points of the WLAN consumes a lot of energy, resulting in limited battery life of the device. Today, the mobile devices that support VoWLAN provide only 3 hours of talk time and 48 hours of standby time. So, this is a big problem in mobile devices that support VoWLAN.
2. **Quality of voice calls:** - The quality of voice calls over WiFi is usually poor when voice has to compete with data for the bandwidth allocation on WLAN. Sometimes the call even drops due to lack of bandwidth.
3. **WiFi infrastructure:** - The WiFi network may be overloaded due to more number of simultaneous VoIP calls. So, to handle the simultaneous voice calls, the WiFi network must be having enough Access Points in such a way that one or more access point is not overloaded, while others are almost idle. If there will be more access points then there will be more handovers.
4. **Fast handoff:** - For the security point of view, the wireless mobile station has to re-authenticate whenever it is roaming between access points of the WiFi network, which causes unacceptable delays and sometimes the voice call drops.
5. **Seamless Vertical Roaming:** - Vertical roaming i.e. the handoff between two different types of networks like from 802.11a to 802.11b or from 802.11 networks to Cellular network (GSM/CDMA/UMTS). There are many problems in this kind of handoff because the handoff latency depends on the availability of both

networks, when to handoff, how to ensure the ordered and timely delivery of packets.

Although VoWLAN has these problems, its integration with cellular network has bright future.

2.3 Wireless LAN

A wireless local area network (WLAN) is a data transmission system designed to provide location-independent network access between computing devices by using radio waves rather than a cable infrastructure. IEEE has defined the specification for a wireless LAN, called IEEE 802.11, which covers the MAC (Medium Access Control) and physical layers. At present three physical layer options are standardized:

1. IEEE Standard 802.11b
2. IEEE Standard 802.11a
3. IEEE Standard 802.11g

The first version of the standard described transmission modes with a theoretical capacity of 1 Mbps and 2 Mbps data rates at 2.4 GHz. In 1999, the 802.11b standard was approved, which provides data rates up to 11 Mbps with the radio frequency band of 2.4 GHz. This standard uses the Complementary Code Keying (CCK) Modulation. Other additions to IEEE 802.11 include 802.11a with a bit-rate 54 Mbps in the 5 GHz frequency band and 802.11g up to 54 Mbps at 2.4 GHz. The devices in 802.11a standard use Orthogonal Frequency Division Multiplexing (OFDM). The use of 5GHz frequency band in Europe caused interference with some other services. To overcome this problem, a revised version of IEEE802.11a, IEEE 802.11h was developed. The IEEE 802.11a/h systems can only be used indoors. The IEEE 802.11g standard uses OFDM technology and for the support of backward compatibility with 802.11b devices, it also supports CCK modulation.

The IEEE 802.11 standard defines two kinds of services: the basic service set (BSS) and the extended service set (ESS). A basic service set is made of stationary or mobile wireless stations may or may not be controlled by a central base station, known as access point (AP). The BSS without an AP is called ad hoc network. An extended service set (ESS) is made up of two or more BSSs with APs. In ESS, the basic service sets are connected through some wired LAN, which provides access to the other networks by using a gateway. Figure 7 shows an ESS.

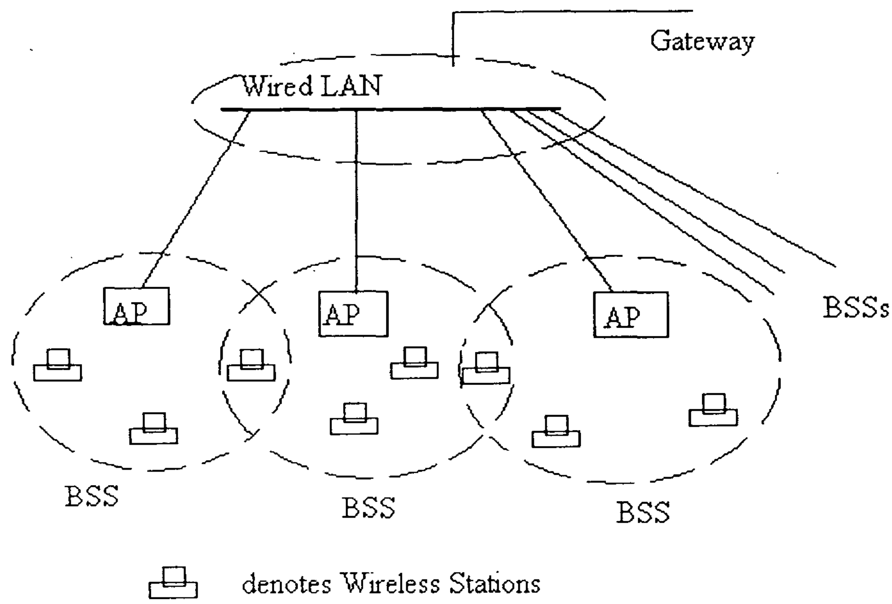

Fig. 7: IEEE Standard 802.11 ESS

2.3.1 IEEE 802.11 medium access protocols

The IEEE 802.11 standard supports two MAC mechanisms, the Distributed Coordination Function (DCF) and the Point Coordination Function (PCF). DCF uses CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance) in order to avoid collisions. DCF is the mandatory 802.11 access control function. Before transmitting, the wireless mobile station senses the radio channels to determine if another station is transmitting on the same radio channel. It can transmit if it senses the radio channel ideal for a period equal to

DIFS + Back off Timer.

Where

DIFS is the inter-frame space which is used to control the medium access.

Back off timer = Random( ) * SlotTime,

Random is a pseudorandom number drawn from a uniform distribution, SlotTime depends on the physical characteristics. The default SlotTime is 20 μs, nut 9 μs is used when all stations of the basic service set support 802.11g.

While the DCF is responsible for asynchronous data services, the PCF offers time-bounded services and can only be used in an infrastructure mode. This method uses a Point Coordinator (PC) which operates at the Access Point to determine which end-point (wireless mobile station) has the right to transmit.

2.3.2 Handoff or Handover in IEEE 802.11 Networks

When a wireless mobile station roams between different Access Points (APs) of the WLAN, it goes through a process, known as Handoff. The wireless mobile station (WMS) needs a wireless connection to some AP of the WLAN to be able to get services and applications of Internet and other such services of the WLAN. So, we say that WMS is attached to some AP. When a WMS changes its wireless connection from one AP to some other neighboring AP, then we say Handoff occurs. Generally handoff occurs when a WMS moves from one AP area to other neighboring AP area. The new AP must be using different radio channel in its area to avoid the interference with its neighboring APs. So, Handoff is better defined as:

> "The process of transfer of wireless mobile station's connectivity from one radio channel to some other radio channel is known as Handoff."

The new Access Point may support different technology of the IEEE standard 802.11 network. For example, the old AP supports 802.11b, while the new AP supports 802.11a only. This type of handoff is called Vertical Handoff, while handoff between the APs supporting the same 802.11 standard is Horizontal Handoff. One important point in handoff is that a wireless mobile station (WMS) can have connection to only one AP i.e. it can not be associated with two APs at the same time, so these Handoffs are Soft-Handoffs. So, the voice packets are delayed or lost during these handoffs. IEEE is currently working on a new standard; IEEE 802.11r, with one goal is to eliminate these problems of perceptible disconnections during handoff.

2.4 VoIP call capacity on IEEE 802.11 network

The CSMA/CA protocol used for medium access control (MAC) in wireless LAN is not optimal for voice calls. An analytical model showed one limitation of the IEEE 802.11a/b distribution coordination function (DCF). It showed that these networks put an upper limit on simultaneous VoIP call by a maximum of six. To address this problem, IEEE established a separate task group (TGe) and defined a set of QoS enhancements to existing MAC standard. Robust Header Compression (ROHC) and Silence Suppression (SS) techniques has been proposed for IEEE 802.11b WLAN, to enhance the capacity of VoIP calls. The upper bound capacity N for IEEE 802.11b wireless network is:

$$N=\left\lfloor \frac{1}{R\left[2*(T_{voice}+SIFS+T_{ack}+DIFS)+\frac{T_{slot}*CW_{\min}}{2}\right]}\right\rfloor$$

Where

R: Number of packets generated by voice codec per second.

T_{voice}: Transmission time of voice data frame. It includes MAC header processing time, RTP/UDP/IP or ROHC header processing time.

SIFS: Short Interframe Space.

T_{ack}: Transmission time of acknowledgement which includes packet loss concealment preamble time, header processing time and actual transmission time of acknowledgement.

DIFS: DCF Interframe Space.

T_{slot}: Slot duration on IEEE 802.11b

CW_{min}: Number of minimum random slots picked during backoff.

CHAPTER 3

VoWLAN and Cellular Integration

3.1 Introduction

In RECENT years, many studies have been conducted to integrate *Wireless LAN* (WLAN) and cellular (such as Universal Mobile Telecommunications System or UMTS, and Global system for Mobile Communication or GSM) networks that extends cellular network services to the WLAN environment. Voice over IP (VoIP) is transporting voice in IP packets and the idea was introduced in mid 90's. Now the VoIP technology is the foundation and enabler for many new service offerings specially targeted to mobile operator's customers. Voice over WLAN (VoWLAN) is a low cost technology for providing mobile communication services due to low cost of WLAN deployment and free spectrum for the areas like Industrial, Scientific and Medical. Figure shows the integrated WLAN and cellular environment.

Cellular networks are basically telecommunication networks which cover wide area and support user mobility and high speed mobile users at low transmission rates. Wireless LAN is data communication network which does not cover a wide area, but it can support a high speed transmission rate with very cheep cost. Due to their different coverage and costs, they impose pros. and cons. on various wireless networks. The integration of these two different networks was required because the number of mobile service uses has been continuously increased, and a majority of users needed a higher bandwidth. But the different characteristics of these integrated networks may degrade the performance and service quality when a mobile user changes its air interface from one network to another, called Handoff. This idea of WLAN and Cellular networks integration is very appealing, because the cost of handling calls placed from a given area using WLAN is a tenth the cost of using cellular. Also degradation of a cellular call inside a building usually happens. Thus WLAN can be used to improve cellular coverage inside buildings.

In the integration of VoWLAN and cellular networks, WLAN serves as an access technology to the cellular system, which scales up the coverage of mobile services. Wireless LANs are found on college campuses, office buildings, and public areas. WLANs are connected to Internet (IP networks) for providing the services like internet surfing, VoIP calls, Video conference and other such services. IEEE has defined the specification for WLAN, called IEEE 802.11, which provides radio access to the Mobile Stations (MS) through 802.11-based access points (APs). Integration between WLAN and Cellular networks is a progressive process. Each step of the process involves a stronger and more complex integration between the two different networks.

The demand for an all-in-one mobile phone with integrated services (like cellular services, internet surfing, VoIP calls, video conferencing, and common billing for all these services) has been accelerated. A cellular/WLAN dual mode mobile has the ability to serve these needs in a single package, however the system suffers from the power consumption problem and service continuity issues between cellular and WLAN systems.

A cellular/VoWLAN system involves a user with a dual mode mobile being able to access VoWLAN in enterprise or hot spot WLANs, and switch to a cellular system without WLAN coverage. A user with dual-mode MS can choose the network interface for making calls based on their personal preferences, network connectivity. Therefore, cellular/VoWLAN dual mode service can reduce the cost of mobile telecommunication services, while also achieving high mobility and wide coverage.

3.2 WLAN and Cellular Integration

The integration between Wireless LAN and Cellular networks (UMTS, GSM) is implemented by using a gateway between the two different networks. Figure 8 shows the inter-connection between UMTS (Universal Mobile Telecommunications System) and WLAN (IEEE standard 802.11) using the WLAN based GPRS Support Node (WGSN) approach. The MS is equipped with two interfaces, one for cellular and other for WLAN. So, the MS can roam between the two networks. The UMTS network consists of two sub-networks. The UMTS Terrestrial Radio Access Network (UTRAN) and Base Stations (BSs). The UTRAN make use of Radio Network Controllers (RNCs), which controls the radio interface between BSs and MSs and uses WCDMA technology. For providing mobility and other services like session management, the UMTS core network (GPRS; Fig 8(4)) makes use of Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN). SGSN is connected to the UTRAN and communicates with the GGSN through an IP-based backbone network. Through an IP-based interface, Gi, GGSN is connected to the external Packet Data Network (PDN). Gr and Gc are two interfaces which provide communication of SGSN and GGSN with the Home Location Register (HLR), respectively.

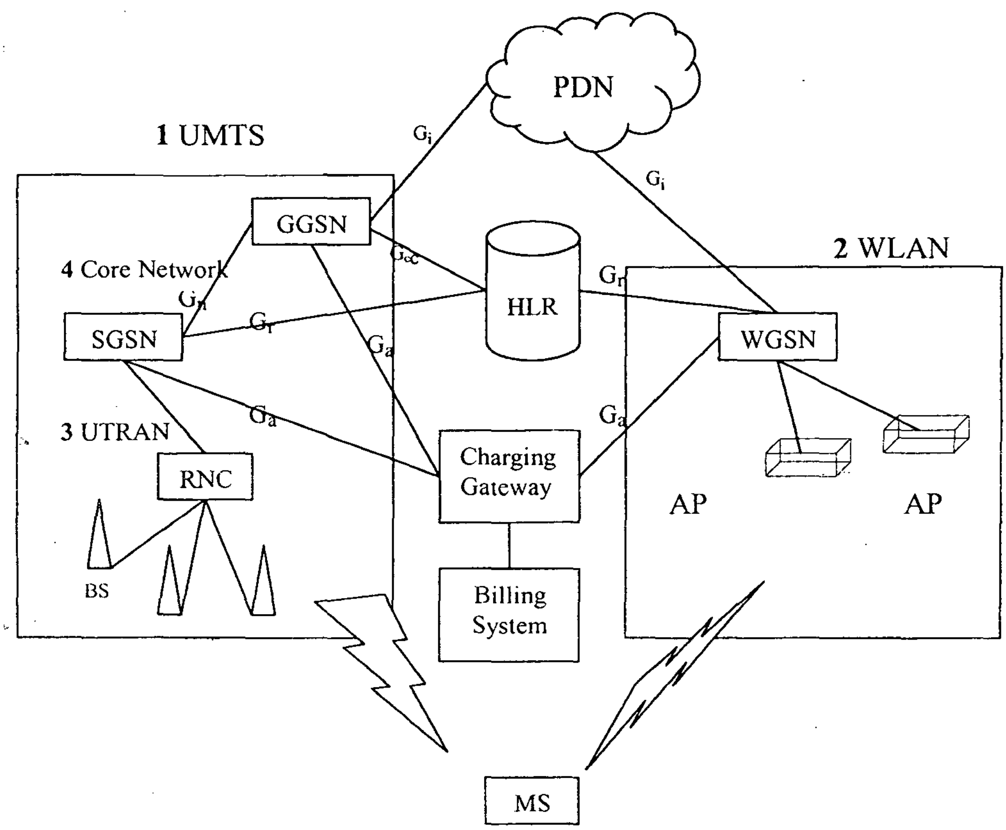

Fig. 8: WLAN and cellular integration: WGSN approach

The WGSN acts as a gateway between the PDN and WLAN nodes (i.e. access points) and obtains the IP address of the MS by using a Dynamic Host Configuration Protocol (DHCP) server and then routes the packets accordingly. To support GPRS/UMTS mobility management, the WGSN node communicates with HLR, which is the master database containing information about all user-related information. For the common billing system for the use of both kinds of services (i.e. WLAN and cellular network services), the WGSN is connected to the Charging Gateway using UMTS protocols. WGSN provides general Internet access and *Voice over IP* (VoIP) services by the use of Session Initiation Protocol (SIP).

3.3 SIM-based Authentication for WLANs

When the dual mode MS attaches itself to some AP of the WLAN, then it needs some access control mechanisms and authentication on the WLAN for billing and service aspects for the user. There are two possible ways to authenticate the subscribers in WLAN:

1. Re-use existing UMTS/GSM authentication mechanisms.
2. Employ an individual AAA (Authentication, Authorization and Accounting) server in WLAN.

Due to secure authentication mechanisms imposed by cellular networks, with well established roaming agreements and well defined service sets, it is favorable to re-use the authentication mechanisms for access control to WLAN. The cellular networks use Subscriber Identity Module (SIM) for the authentication along with International Mobile Subscriber Identity (IMSI), and implements authentication algorithms. All the information related to authentication is stored in HLR of the MS at the MSC where the MS is registered. Therefore, there is no need to use separate WLAN authentication mechanisms and thus the maintenance is simplified because the subscriber profiles for both WLAN and cellular networks are combined and stored in HLR.

3.4 Routing of VoIP call to dual mode mobile

The dual mode mobile can be reached via a single phone (mobile) number and incoming VoIP calls to this number are routed to either cellular system or Wireless LAN depending on the WLAN coverage. In WLAN and cellular integrated environment, the MS typically attempts to access the WLAN first for lower costs and higher bandwidth connection. If WLAN is not available, the MS then tries to access the cellular network. But the system suffers from a serious problem of power consumption caused by WLAN module in the mobile station, so the mobile station turns on the cellular module and turns off the WLAN module.

One solution is to add an extra low power radio access interface in addition to the standard WLAN interface of the mobile. The mobile station can turn off the WLAN interface and only listen to the low power radio signals. If there are packets

(VoIP) destined for the WLAN device (mobile), WLAN access points (APs) can use the low power radio to activate the WLAN interface of the mobile station. Although this approach can reduce the power consumption, it requires the installation of new components on WLAN access points and also on the mobile station (MS).

In dual mode mobiles, the power consumption is very big problem, and to solve this problem, a number of mechanisms have been proposed. By applying these mechanisms, the power consumption problem in the dual mode mobiles is solved, because the dual mode mobile can completely switch off its WLAN interface during the idle period and can only listen to the paging signals from its cellular interface. So by turning off the WLAN interface, the dual mode mobile can't hear or listen to the incoming VoIP call from its WLAN interface which provides low cost, high-speed connection to the rest of the world. The only interface in the dual mode mobile during its idle time which can send or receive signals to the rest of the world is the cellular interface. As the incoming VoIP call has to be received through the WLAN interface, so the dual mode mobile is informed about the incoming call through its cellular interface. Then the dual mode mobile checks if it is in coverage area of some Access Point (AP) of the 802.11 network. If it is, then its WLAN interface is activated and call is received by this interface and the cellular call is ignored, but if it is not in coverage of 802.11 networks, then the call has to be received by the cellular interface.

But, another problem is that for every incoming call, the dual mode mobile attempts to connect to WLAN first, after it got informed by its cellular interface. So, for the normal cellular calls, time is wasted and sometimes the call may even drops. Also, the MS should be able to distinguish a normal cellular call from a VoIP call, otherwise the call setup will be delayed for the normal cellular call because the MS always attempts to connect to WLAN first whether it is in some WLAN's coverage area or not.

3.4.1 VoIP call routing with CSP Approach

The routing of VoIP call to the MS is done with the Call Server with Push (CSP) approach. In this approach, the MS can detect the "originating network" of the calling party and thus the call setup delay to the normal cellular call is avoided. The CSP approach uses a SIP proxy in the IP network, which sets the call to the Cellular/PSTN network through PSTN/IP gateway by implementing a push approach to distinguish the call from cellular call, so it is called Call Server with Push. The PSTN gateway is assigned an SS7 number (say, 987564xxxx), or a group of leased lines with same prefix (say, 987564). The MS has SIP URI (Uniform Resource Identifier) for its WLAN interface because the VoIP call is setup using the SIP protocol. To receive a VoIP call, the SIP User Agent (UA), which is having the SIP URI of the MS, is connected to the CSP. To track the call setup status of the MSs, the CSP maintains a dual-mode MS (DM) table. The MS maintains a PSTN/IP gateway table to distinguish the VoIP call from the normal cellular call. A VoIP call is having the calling party number from one of the SS7 numbers assigned to PSTN Gateway, and this calling party number can be checked in the PSTN Gateway table at MS to find out whether it is a cellular call or a VoIP call. The SS7 number of the PSTN Gateway and the corresponding fully qualified domain name of the CSP (e.g., abc.com) are stored in an entry in the PSTN Gateway table. There may be multiple PSTN gateways connected to CSP (national and international PSTN Gateways), for each gateway, there is an entry in the PSTN Gateway table maintained at the MS. The interworking of cellular and WLAN network with CSP approach for VoIP call routing is illustrated in Fig. 9.

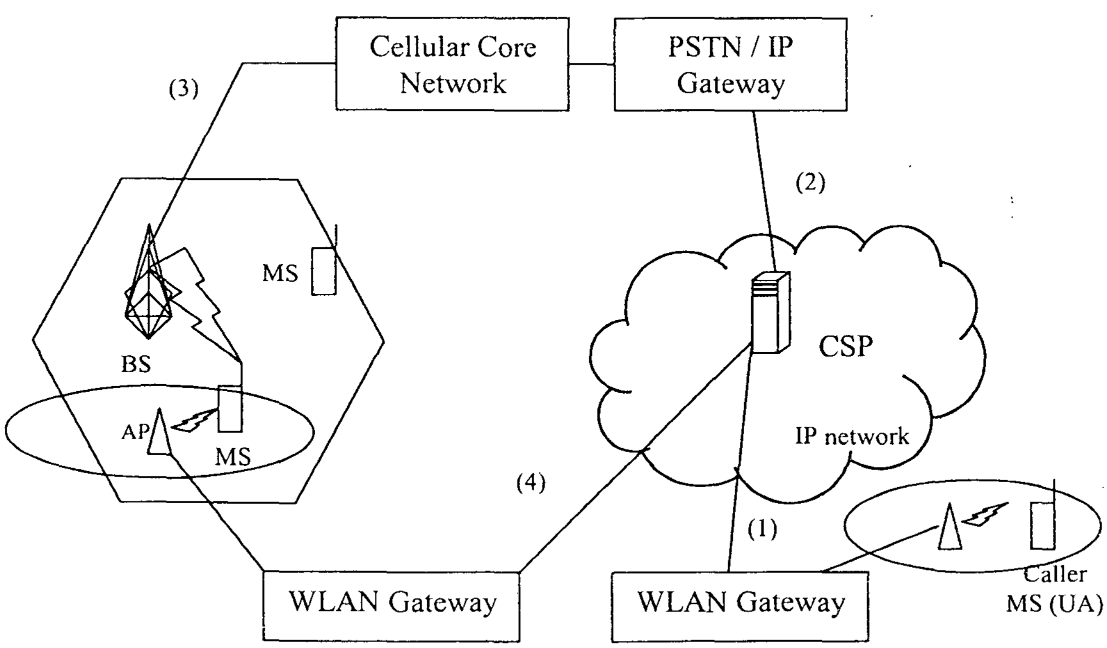

Fig. 9: VoIP call routing

When a dual-mode MS or some IP host (say UA) makes a VoIP call to a MS, the UA sends an INVITE message to the CSP (Figure 1 (1)), which contains the SIP *Universal Resource Identifier* (URI) of the MS, i.e.,

+9871598800@abc.com

and Session Description Protocol (SDP) that provides the RTP information of the MS like, IP address and port number of the UA. Now, if the MS is registered with the CSP, the CSP forwards the INVITE message to MS directly using the SIP call setup procedure. If not, then CSP routes the VoIP call to the PSTN, by forwarding the INVITE message to PSTN Gateway and the CSP also creates a DM table record for the MS to specify that the call is setup to the PSTN (Fig. 1(2)).

The PSTN gateway generates an SS7 Initial Address Message (IAM) which contains the caller ID 987564xxxx, and sends to cellular network (UMTS/GSM). This is illustrated in fig. 1(3). Now the VoIP call is routed as normal cellular call on the cellular network. The BS pages the called MS. The MS now checks its PSTN Gateway table to see if 987564xxxx is found. If not found, then MS reply the paging signal from BS, and receives the cellular call (Fig. 1(5)).

If CSP entry is found, it means the incoming call is VoIP call. The MS retrieves the fully qualified domain name of the CSP (abc.com), and the MS then attempts to access the nearby WLAN. If there is no WLAN accessibility, then the VoIP call is accepted via the cellular interface as a normal cellular call. But, if there is WLAN accessibility then, the MS registers itself to the CSP (Fig. 1(4)) via the AP of the WLAN. Meanwhile the ringing of MS by the cellular network is delayed and as soon as the MS registers and gets the INVITE message from CSP, the MS rings by sending a Ringing message to UA through CSP. The UA then plays an audio ringing tone to the calling party. Finally, the actual conversation starts between the caller and called party. This whole process with message flows is shown in Fig. 10.

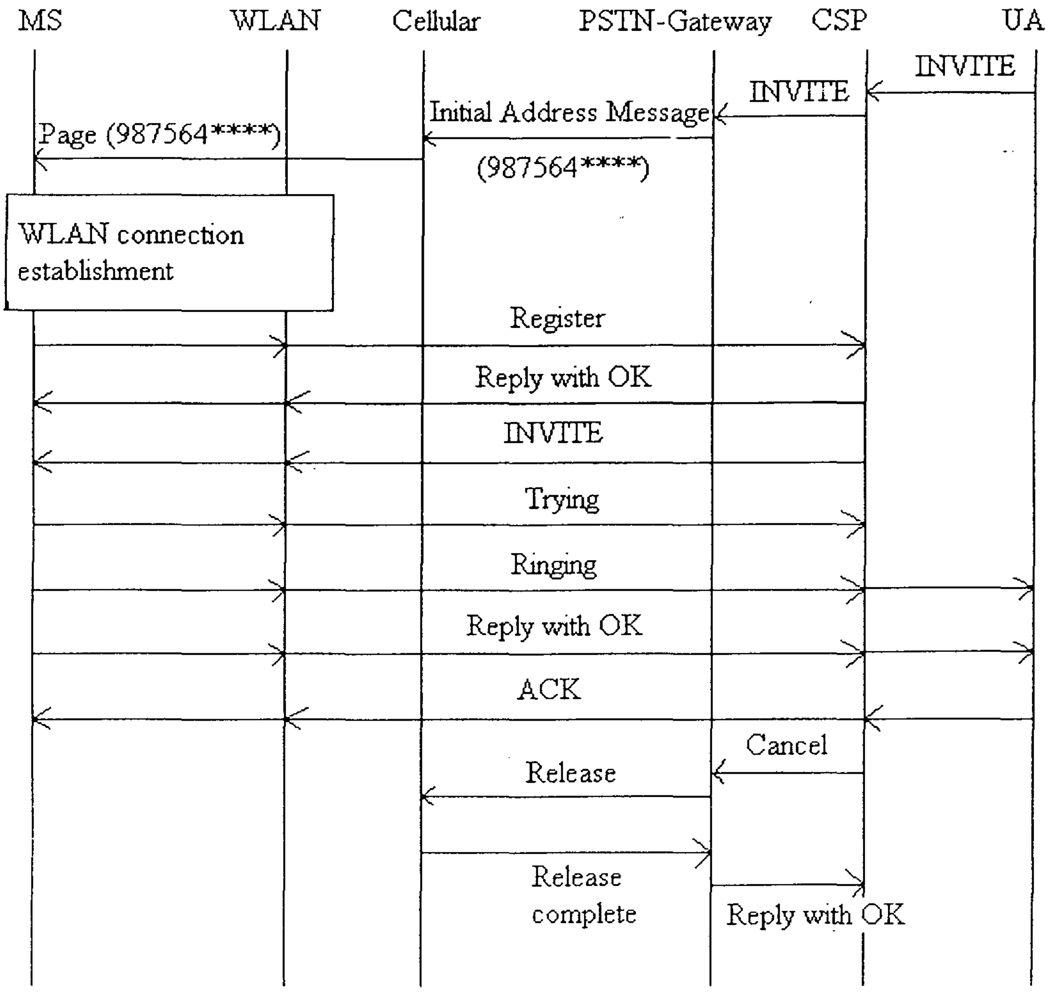

Fig. 10: Message flow for an incoming call to the dual-mode MS.

While the MS is registering it to the CSP via AP of WLAN, and the SIP invite message is not received at the MS within pre-configured time-out value, the MS rings the users to pick up the VoIP call via cellular interface. But still, there is call setup delay and sometimes the call drops. There are two approaches towards minimizing the call-setup delay via the WLAN interface of the MS: -

1. Parallel Fork approach
2. Wake-up and register approach

In parallel fork approach, the MS waits to receive the SIP INVITE message via its WLAN NIC. The CSP sends a SIP INVITE message without getting a response, so it activates the exponential backoffs on SIP INVITE message to a SIP UA. This exponential backoff introduces extra call establishment delays.

The wake-up and register approach was proposed to avoid the delays due to exponential backoffs. In this method, the cellular call is still used to activate the WLAN NIC, the only difference is that the dual-mode MS interacts with CSP to poll SIP INVITE messages instead of waiting for incoming INVITE packets. So, this approach avoids the loss and exponential backoff retransmissions of the SIP INVITE message.

The power consumption of a WLAN NIC is zero for the idle time, it is activated only if there is a VoIP call destined for the MS and accordingly the call is routed via either cellular module or WLAN interface of the dual-mode MS.

While the MS completes the registration process with the CSP (SIP proxy with push mechanism), the incoming VoIP call may lost if its timer expires, and all other outstanding VoIP calls before the WLAN NIC is activated through CSP, are lost and the MS accepts the VoIP calls via its cellular interface as normal cellular call. The timing diagram illustrates this process, where dots in the figure represent the dropping of incoming VoIP call immediately after it arrives at CSP.

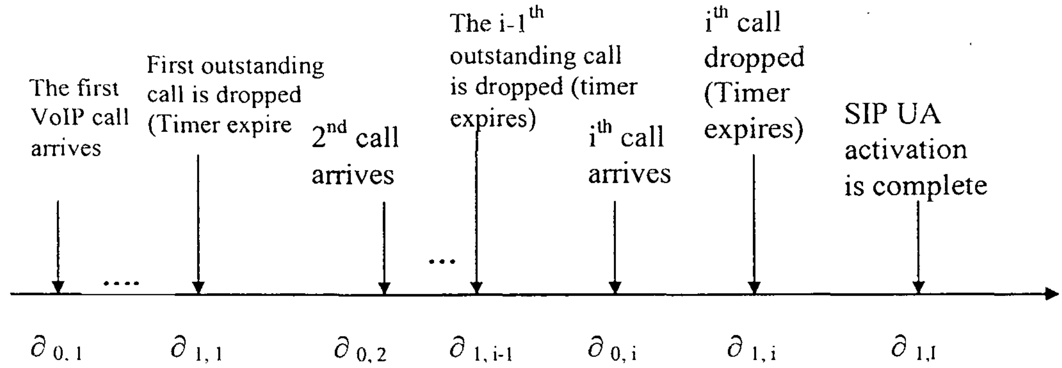

Fig. 11: Timing Diagram for incoming VoIP calls before activation of SIP UA

Suppose the first VoIP call arrives at CSP at time $\partial_{0,1}$. At time ∂, CSP receives the SIP UA activation complete message form the called MS. The i[th] call arrives at time $\partial_{0,i}$. The timer for the i[th] call expires at time $\partial_{1,i}$. So the timeout period is

$$T_{1,1} = - \partial_{0,i}$$

The incoming calls arrive at a rate λ with a Poisson process. The SIP UA activation time is

$$T_{3,1} = \partial - \partial_{0,1}$$

3.5 In-call vertical handoff between VoWLAN and cellular

Seamless roaming between VoWLAN and cellular networks is achieved by "handing over" the VoIP call between the two different networks. The handoff mechanism should be selected that the quality of voice does not degrades. The in-call handoff may be horizontal handoff or vertical handoff. Horizontal handoff occurs when the dual-mode MS currently getting service from some AP and moves to some other AP of the WLAN with the same air interface and this is handled by the handoff mechanism implemented by the SIP and RTP protocols internally within the WLAN network. Vertical handoff occurs when MS changes it access from one type of network to another type of network in order to provide the service continuity. For example, when the MS moves out from the coverage of WLAN and makes a connection with the BS of the cellular network in order to avoid the call drop. So, the MS now transfers the VoIP call from its WLAN interface to the cellular interface, after the handoff is completed the MS switch off its WLAN interface completely to avoid the power consumption problem.

Vertical handoff between different radio interfaces involves a variety of time consuming procedures such as handoff triggering, BS selection; authentication, service negotiation, and IP address acquisition. WLAN signal degradation can be very abrupt and unexpected. In many common situations such as when exiting a WLAN region during an active VoIP call connection, considerable packet loss will typically occur before the connection is established with cellular network.

A handoff process between the networks with different air interfaces is characterized into two main steps: Handoff decision and Handoff execution. Handoff decision is the process of deciding when to perform a handoff. It takes decision about the handoff by measuring the received signal strengths from AP and BS. After the decision to handoff is made, the handoff execution begins. The goal is to minimize the handoff latency, which can be broken into three main components:

1. Detection Period (t_d): It is the time taken by the MS to discover that it is under the coverage of different network to the instant it receives the router advertisement (RA) form the new access router. The AP of WLAN and BS in the cellular network work as access routers for the MS.
2. Address Configuration Interval (t_c): This time interval denotes the time from a MS receives a RA from new network, to the time it takes to activate the corresponding interface.
3. Registration Time with new Network (t_r): It is the time taken to register it at the new network, say at WLAN because the MS has moved to WLAN coverage area.

The handoff latency depends on t_d, t_c and t_r.

$$\text{Handoff latency} = t_d + t_c + t_r$$

The handoff performance can be improved by minimizing the handoff latency i.e. if these time periods i.e. t_d, t_c and t_r are reduced.

The goal of vertical handoff is low handoff latency, power saving and low bandwidth overhead. Also WLAN and cellular networks (UMTS/GSM) are having different frequency and data speed. The vertical handoff can occur in three cases and illustrated in Fig. 12.

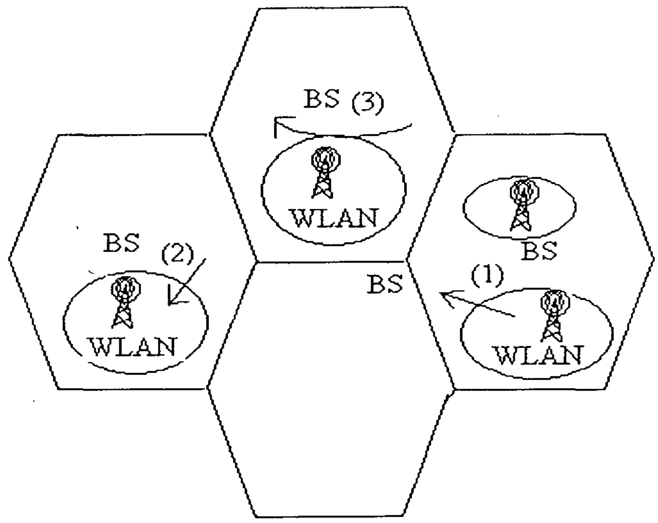

Fig. 12: Vertical Handoff scenarios

1. Mobile Upward (MU) case in which the MS moves from cellular network outside the WLAN into the WLAN. Upward vertical handoff is a handoff from a smaller network with high bandwidth to a larger network with lower bandwidth. In this case, the dual-mode MS user has to initiate the handoff process because the user is talking on the MS via cellular network and the WLAN interface of the MS is completely switched off. The user can activate the WLAN interface of the dual mode MS, which is having a SIP UA, which look for some AP by broadcasting beacon signals from its WLAN NIC.

2. Mobile Downward (MD) case in which the MS serviced in the WLAN region moves out to cellular network. Downward vertical handoff is a handoff from a larger network with low bandwidth to a smaller network with higher bandwidth. In this case, when the dual-mode MS starts receiving the signal strength from the AP below the threshold signal strength, the handoff decision should be taken. The MS can wait for the paging signals from the BS or it may broadcast a signal to the nearby Base Stations on the forward control channel for the allocation of bandwidth for the VoIP call for which vertical handoff to be done

3 Mobile Through (MT) case in which a MS passes over the WLAN.

For the seamless handoff service, the handoff point in the MD is not a critical factor, because the cellular network covers the WLAN region with an overlaid network. The in-call handoff in MD case can be done easily because cellular networks have wide coverage almost everywhere, so the MS can confine the call via its cellular interface and can handoff to WLAN when it receives proper signal strengths, after the activation of WLAN NIC by the user.

WLAN-to-cellular vertical handoff involves time consuming procedures that may significantly disrupt real-time communication, so the handoff point is very critical, because the WLANs have limited coverage area and if the handoff point is not chosen properly the VoIP call may even drop. Once the handoff is decided in the MU case, the cellular call-setup procedure is executed and after the call is setup, the WLAN interface of the MS is switched off in order to save the battery power.

3.5.1 Vertical Handoff Algorithm

The handoff algorithm works by measuring the RSS for the handoff decision. Once handoff is decided, then the actual handoff procedure begins. The handoff occurs only when the necessary QoS for the voice in the destination network is satisfied.

The handoff algorithm makes use of following variables:

λ_{thresh} : It is the Predefined threshold value in the handoff transition region

λ : λ denotes the continuous beacon signals that are received from AP of the WLAN with below λ_{thresh}.

λ_r : It denotes the λ for real-time service (VoIP).

λ_u : It denotes λ for mobile upward case

λ_n : It denotes λ for non-real time service i.e. data services.

The relationship among the variables is

$$\lambda_r << \lambda_n << \lambda_u$$

Handoff Algorithm for MU case: The algorithm assumes that there is VoIP call to the MS via WLAN interface. For VoIP call, the handoff delay must be short in the handoff transition region, so the number of continuous beacon signals should be lower than that of the non-real-time service. The flowchart of the vertical handoff is illustrated in Fig. 13 for the mobile upward case.

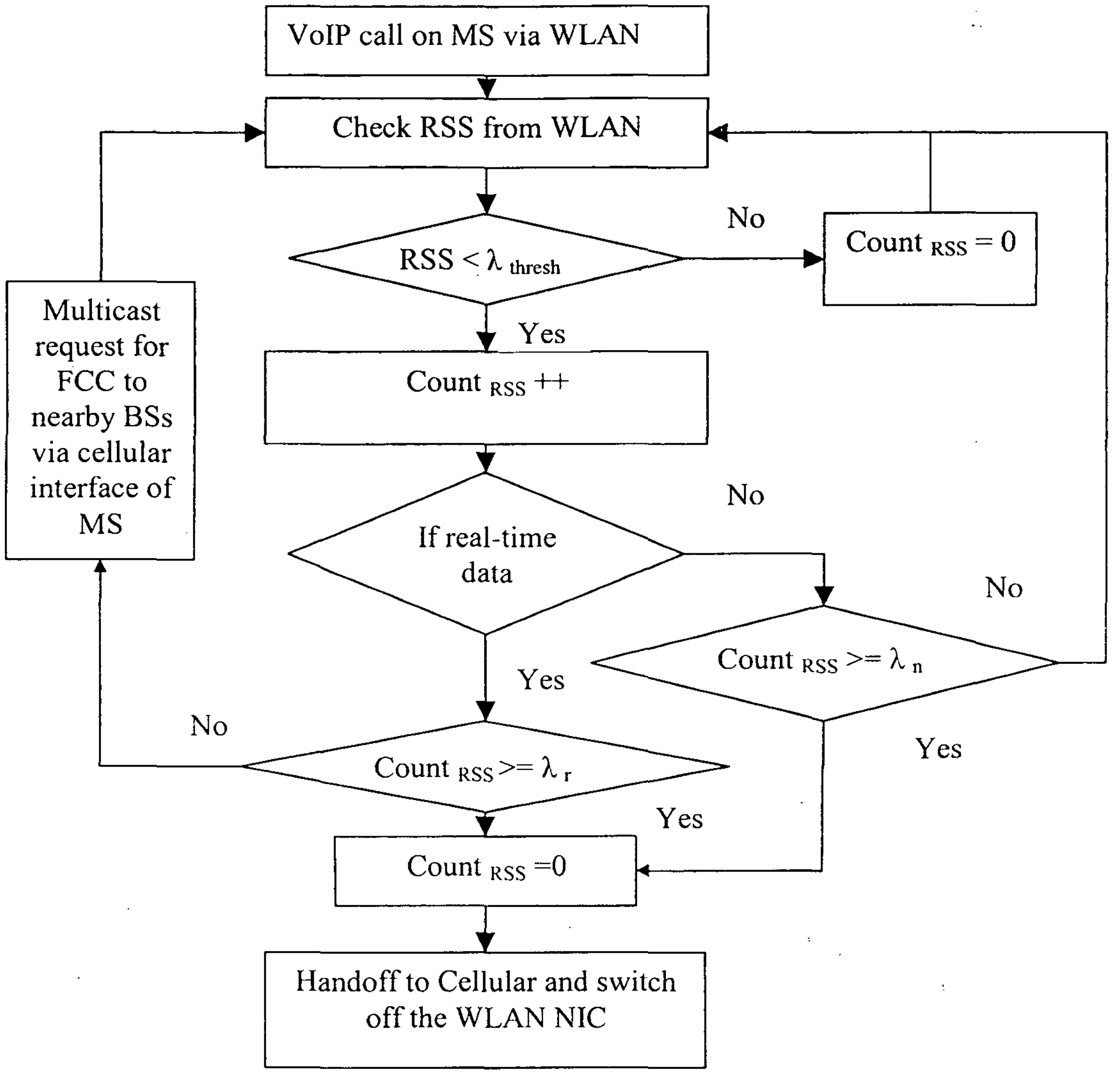

Fig. 13: Upward Vertical Handoff algorithm

Before the actual handoff, the BS allocates a pivot channel to the MS after either the MS has sent the poll request to the BS and requested a pivot channel for the call. Meanwhile, the MS keeps checking the signal strength received from the AP. If the pivot channel is not allocated to the MS due to unavailability, and the RSS from AP reaches less than threshold required for VoIP call, i.e. λ_r , the VoIP call drops.

If the pivot channel is allocated to MS before the RSS becomes less than λ_r, then the VoIP call is successfully transferred to the cellular network. After handoff, now the call is treated as normal cellular call.

CHAPTER 4

VOIP CALL ROUTING OPTIMIZATION WITH VERTICAL HANDOFF

4.1 Problem Definition

For the in-call vertical handoff from Cellular (GSM/CDMA/UMTS) to the wireless LAN, the handoff latency is not critical because the cellular networks are available almost everywhere. So the handoff can be done at any point of time after the dual-mode MS entered in some WLAN region.

For the VoIP in-call handoff from Wireless LAN to Cellular network, the vertical handoff latency should be minimized. When the voice call is handoffed from wireless LAN to cellular network, affordable QoS is needed for seamless and continuous service support at the BS. In the existing algorithms for vertical handoff, the MS broadcasts request for Forward Control Channel (FCC) to the nearby BS to set up the handoff VoIP call. So, it is like to setup a new call and cellular networks like GSM takes in order of about three seconds to set up a call. This cellular call set up time contributes to Handoff latency. So, if the user is moving fast such that the VoIP call quality degrades due to that delay and if the user (MS) is moving fast out of WLAN region, then the call drops. Even if, the handoff of VoIP call is successfully done, due to this handoff latency, the WLAN interface of the MS has to be kept activated for the time that the handoff process takes.

So, for the in-call vertical handoff from WLAN to Cellular network, the handoff latency needs to be minimized. The proposed algorithm works in the direction of minimizing the vertical handoff latency by minimizing the handoff request latency and call establishment latency with cellular network.

4.2 Optimized VoIP call routing with Vertical Handoff

The algorithm implements a scheme that causes the call establishment to be initiated from AP via CSP, instead of dual-mode MS. When the MS goes out of coverage area of WLAN, integrated network needs to set up a cellular call for the VoIP call for which handoff to be done. AP sends the vertical handoff request to CSP; the Call Server with Push (CSP) forwards the call establishment request to the Gateway between

Cellular/PSTN networks and IP network, which then establishes the call (handoffed VoIP call) to the MS as normal cellular call. The MS now come to know that it is a handoff VoIP call by checking the caller's number and it will not try to connect it via its WLAN interface and accepts the incoming handoff VoIP call via its cellular interface. The whole idea is given below:

When the RSS $_{WLAN}$ reaches close to minimum threshold, then there may be two situations:-

1. The MS is moving out of the WLAN region and it is moving out from coverage of one AP to another AP. The handoff in this case may be horizontal or vertical depending on whether the two APs supporting same IEEE 802.11 standard or not.
2. The MS is moving out of WLAN coverage and now it requires a cellular network for the seamless service continuity. The ongoing VoIP call needs to be transferred from WLAN to the cellular network, with proper QoS for voice such that the user is not aware of this type of handoff. So in this case, the handoff request must be delayed until we ensure that it is not AP to AP handoff. If it is not WLAN APs handoff, then immediately the handoff request has to be issued to CSP. The CSP will always connect this call through cellular network as handoff VoIP call, it doesn't matter whether the MS has initiated that call or the MS received that VoIP call.

The handoff processing time is reduced because the handoff request is sent to CSP along with the voice data packets as soon as the handoff is decided. To support the VoIP call vertical handoff, the group of leased lines with same prefix (say, 987564xxxx), or SS7 number the PSTN/IP network Gateway is having needs to be modified. Also the Cellular/VoIP Gateway table of dual-mode MS will also needs to be modified. Now the cellular/VoIP gateway table will be having one more attribute for handoff calls with entries 0 and 1, where

0 indicates incoming call is VoIP call.

1 indicates incoming call is handoff VoIP call.

The gateway will also be having a set of these numbers beginning with prefix different from prefix of SS7 numbers. This new set of numbers with same prefix identifies the call as VoIP call for which handoff to be done. So when the call will be established, the MS will check its cellular/VoIP gateway table to find out whether it is cellular call or incoming VoIP call or handoff VoIP call. These modifications at Gateway and dual-mode MS are required to avoid treating the handoff calls as new VoIP call. As soon as the VoIP call is also established via the cellular interface of MS (means that handoff process is over), MS will switch off its WLAN NIC. The CSP will automatically update itself for the routing of incoming and outgoing packets between the two communicating parties. As the handoff latency is reduced, the WLAN NIC will be switched off automatically, the power consumption is also reduced. The handoff is initiated via WLAN interface of the dual-mode MS through CSP to cellular network.

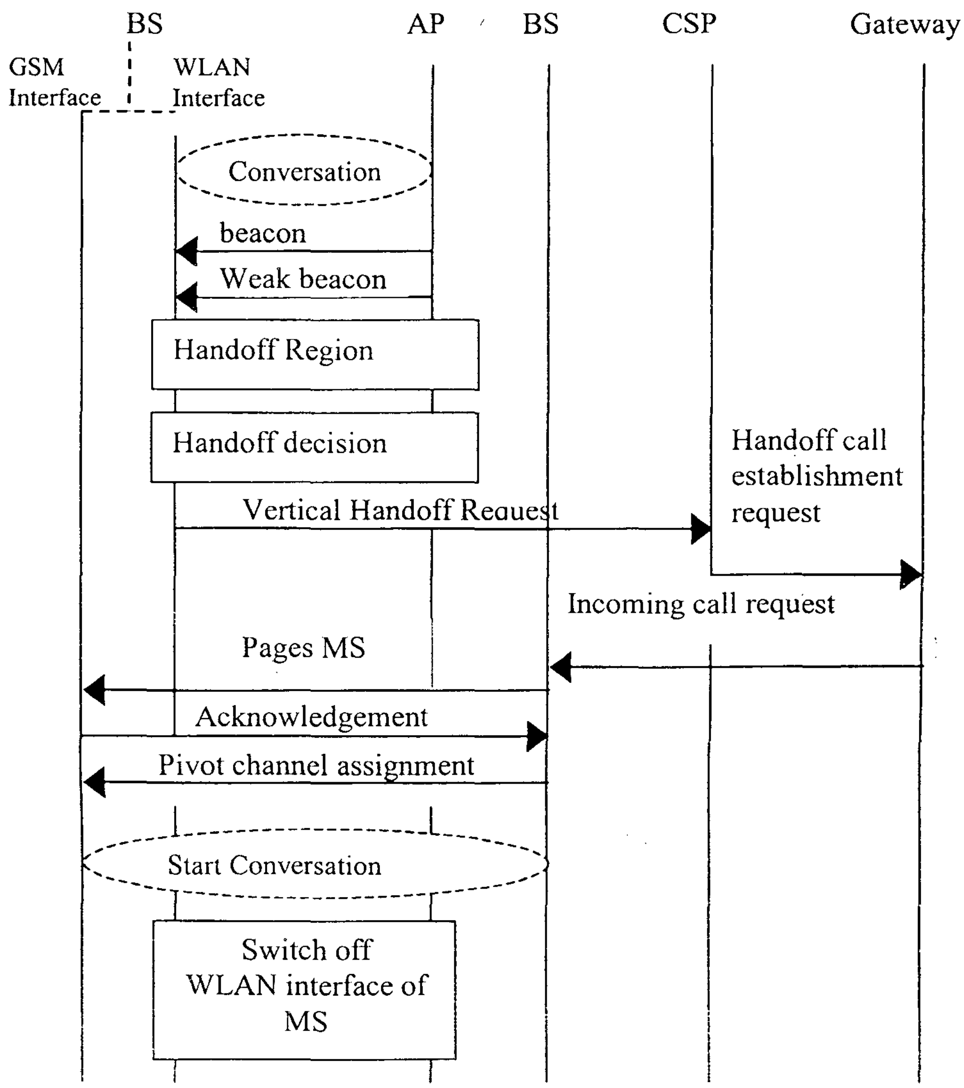

Fig. 14: Message flow for vertical handoff from WLAN to Cellular

So, to implement the vertical in-call handoff from WLAN to Cellular networks, routing of VoIP call is also affected because the structure of cellular/VoIP gateway table at MS is modified and the groups of leased lines with same prefix at Gateway between the PSTN/Cellular and IP networks are also divided for incoming VoIP call and handoff VoIP calls. The message flow for the vertical handoff for ongoing VoIP call from WLAN to Cellular network is shown in Fig. 14.

Procedure: - The procedure assumes there is ongoing VoIP call via the WLAN NIC of dual-mode MS.

1) Handoff Decision: If the received signal strength from AP is close to minimum threshold then decision about handoff is to taken such that there will not be unnecessary handoffs. If the MS receives signals from another AP, with signal strength increasing with time, then it is handoff between different APs. If not, then it is vertical handoff from WLAN to cellular network.
2) Keep sending the voice data packets via WLAN interface and send vertical handoff request message along with the voice packets from AP to CSP.
3) After receiving handoff request from MS via AP, the CSP issues a call establishment request message to gateway. The request contains mobile number of MS and control information for the call, which is to set via cellular, is actually a handoff VoIP call.
4) The Gateway sets the call as normal cellular call to the MS with caller's number chosen from one of numbers assigned to it for handoff VoIP calls.
5) The MS checks the caller's number in its cellular/VoIP gateway table; found that this number is for VoIP calls for which handoff to occur, and sets up the call via its cellular interface without checking for its WLAN interface.
6) As the conversation begins, the WLAN interface of the dual-mode MS is switched off.
7) The CSP updates the routing paths accordingly.

4.3 Pseudo code

Overview of the simulation done for in-call upward vertical handoff with optimized call routing is illustrated through pseudo code given here.

```
main()
 {
        int flag=0;
        c1=establish_new_VoIP_call();
        // c1 is VoIP call  which is established via WLAN interface of dual mode
        //MS.
        If (RSS WLAN > λ thresh + Δ )
         {
                // do conversation.
                //MS continuously receiving beacon signals from AP.
                If (RSS WLAN < λ thresh + Δ )
                 {
                        void handoff_decision(c1);
                 }
         }
 }
void handoff_decision(VoIPcall c1)
 {

                put some timer t1 on c1 &
                If (RSS WLAN AP2 > λ thresh + Δ )
                 {
                        Handoff_WLAN_to_WLAN(c1);
                        // MS moved back in coverage area of same AP.
                 }
                Else
                 {
```

```
                    MU_Vertical_handoff(c1);
                    // vertical handoff from WLAN to Cellular.
                }
    }

MU_Vertical_handoff(VoIPcall c1)
  {
       send_handoff_req_CSP(c1);
       // send vertical handoff request to CSP for call c1.
       Send_handoff_req_BS_via_MS(c1);
  }
Send_handoff_req_BS_via_MS(VoIPcall c1)
  {
      //  multicast  handoff call request to nearby BS with call id c1
       // establish cellular call with call id c1.
   }
```

At CSP

```
Receive_handoff_req(VoIPcall c1)
  {
       // send request to Gateway as handoffed_VoIP call;
       // multicast inbound packets on both routes, i.e. to AP as well as to
       // Gateway with call id as c1.
       // Wait for outbound packets from MS via Gateway, if found, update
       // the route and release the bandwidth acquired by AP for call c1.
  }
```

4.4 Results

A scenario has been taken in which inbound VoIP calls are coming randomly by poison distribution and some calls require MU vertical handoff to get the service continuity. The results obtained using the proposed vertical handoff algorithm for VoIP call from WLAN to Cellular are as follows: Graph illustrates the handoff delay per VoIP call. The delay varies because of network parameters like congestion on network, availability of pivot channels etc.

Average handoff delay = 0.92 sec.

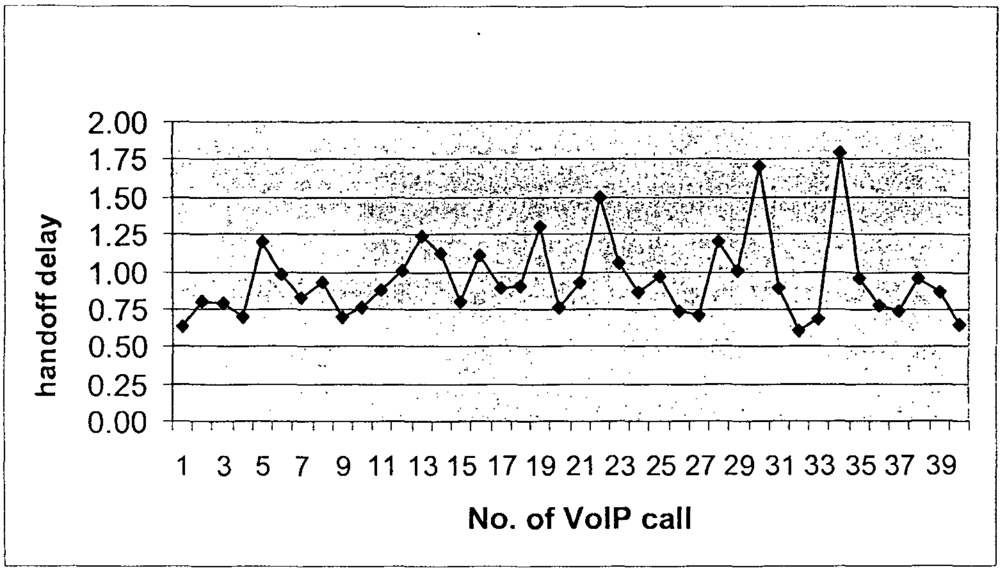

Fig.15: Vertical Handoff delay

CHAPTER 5
CONCLUSION

For providing seamless service between VoWLAN and Cellular networks, the proposed algorithm for vertical handoff from WLAN to Cellular network, initiates the vertical handoff from AP with which the MS is communicating. The AP sends the handoff request along with voice data to the CSP, and informs MS about the vertical handoff request has been sent to CSP. The MS tries to establish a cellular call with the same ID as the VoIP call is having. The CSP forwards the handoff request to cellular/VoIP gateway, which establishes the handoffed VoIP call as normal cellular call with the same call ID. Thus, the handoff latency is reduced. As the vertical handoff from WLAN to cellular is optimized, the inbound call blocking probability on WLAN due to unavailability of resources at Wireless LAN also gets reduced. The handoff time is reduced because AP sends the request for handoff along with voice data to CSP and call establishment on cellular network is achieved by both directions, keeping the call ID same. The packet loss is also reduced and the call dropping probability during the vertical handoff from WLAN to Cellular also reduced to some factor.

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