

Performance of VoIP in WLANs

Sameer Soni
VJTI, Mumbai

Dr B.B. Meshram
VJTI, Mumbai

ABSTRACT:

The WLAN is the rapidly growing technology used for communication today. Integration of VoIP is the current area of research that is very lucrative as it would reduce the communication cost to a very large extent. Presently we have high speed data networks available but, they have poor performance for voice facilities integration. Here in this paper, I propose a scheme for integrating Voice over WLAN, by integrating DRR Scheduling scheme and Multiplexing the data and voice packets for achieving, best effort delivery, reduced packet loss and best scheduling criteria.

KEYWORDS: *VoIP, WLAN, Inter-operability*

1 INTRODUCTION

The Wireless terminals (hand-held, truck mount, stationary, pen-based, laptops) send out data through the radio. Radio frequency is the wireless transmission of data by digital radio signals at a particular frequency. Some type of identification accompanies the data so the receiving computer knows which terminal sent the data.

A wireless system is simply an extension to wired system. Placing an access point on wired system allows wireless Ethernet connection to places where wired devices may not be able to access. In simple terms, an access point receives wireless Ethernet data from wireless devices and places that information on a wired Ethernet cable.

Wireless means transferring the signals without wires using radio waves, infra red etc. Wireless technologies differ in a variety of dimensions Bandwidth, Distance, Frequency, License and Power. In wireless networks there is unlimited mobility; we can access the network services from almost anywhere. In wired networks we have the restriction of using the services in fixed area. The demand of wireless is increasing very fast as everybody wants to use the broadband services anywhere and anytime.

Recently, with the dramatically increasing in usage of wireless technologies, there has been a tremendous growth in the proposal and deployment of network systems and applications. The rapid growth of wireless internet causes a demand for high-speed access to the World Wide Web.

Broadband wireless access system are well known high data rate wireless MAN in which data rate is in mega bits per second which is its significant feature Wireless broadband access standard could supply the missing link for the "last mile" connection in wireless metropolitan area networks. Wireless broadband access is set up like cellular systems, using base stations that service a radius of several miles/kilometers. Base stations do not necessarily have to

reside on a tower. More often than not, the base station antenna will be located on a rooftop of a tall building or

other elevated structure such as a grain silo or water tower. A customer premise unit, similar to a satellite TV setup, is all it takes to connect the base station to a customer. The signal is then routed via standard Ethernet cable either directly to a single computer, or to an 802.11 hot spot or a wired Ethernet.

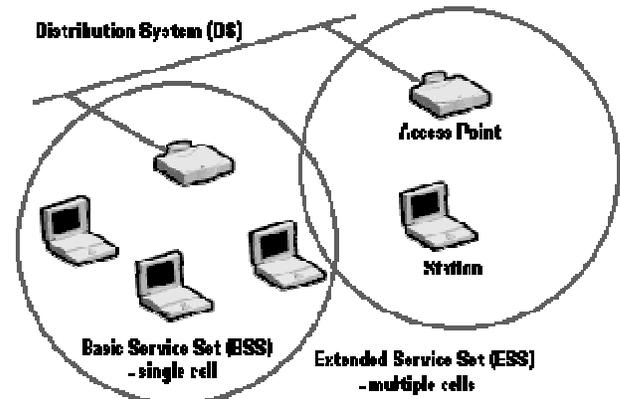


Figure 1: General Organization of 802.11

2. VoIP

Voice over Internet Protocol (Voice over IP, VoIP) is one of a family of internet technologies, communication protocols, and transmission technologies for delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms frequently encountered and often used synonymously with VoIP are *IP telephony*, *Internet telephony*, *voice over broadband (VoBB)*, *broadband telephony*, and *broadband phone*.

Internet telephony refers to communications services—Voice, fax, SMS, and/or voice-messaging applications—that are transported via the Internet, rather than the public switched telephone network (PSTN). The steps involved in originating a VoIP telephone call are signaling and media channel setup, digitization of the analog voice signal, encoding, packetization, and transmission as Internet Protocol (IP) packets over a packet-switched network. On the receiving side, similar steps (usually in the reverse order) such as reception of the IP packets, decoding of the packets and digital-to-analog conversion reproduce the original voice stream. Even though IP Telephony and VoIP are terms that are used interchangeably, they are actually different; IP telephony has to do with digital telephony systems that use IP protocols for voice communication while VoIP is actually a subset of IP Telephony. VoIP is a

technology used by IP telephony as a means of transporting phone calls.

There are three types of VoIP tools that are commonly used; IP Phones, Software VoIP and Mobile and Integrated VoIP. The IP Phones are the most institutionally established but still the least obvious of the VoIP tools. Of all the software VoIP tools that exist, Skype is probably the most easily identifiable. The use of software VoIP has increased during the global recession as many persons, looking for ways to cut costs have turned to these tools for free or inexpensive calling or video conferencing applications. Software VoIP can be further broken down into three classes or subcategories; Web Calling, Voice and Video Instant Messaging and Web Conferencing. Mobile and Integrated VoIP is just another example of the adaptability of VoIP. VoIP is available on many smartphones and internet devices so even the users of portable devices that are not phones can still make calls or send SMS text messages over 3G or WIFI.

Motivations of VoIP

- Consumer market
- PSTN and mobile network providers
- Corporate use

Benefits of VoIP

- Operational Cost
- Flexibility
- Cost Reduction
- Simplification
- Consolidation
- Advanced Applications

3 VoIP & WLAN

Currently, the voice and data traffic are treated separately by the GSM (Global Special Mobile) and GPRS (General Packet Radio Service) wireless access technologies.

The Release.1 definition of 3rd Generation (3G) systems also kept this separation in the following sense: The wireless voice traffic is routed through the circuit-switched cellular infrastructure, whereas data traffic is routed through the packet-switched (e.g., GPRS) network.

Voice over IP has been implemented in various ways using both proprietary and open protocols and standards. Examples of technologies used to implement Voice over IP include:

- H.323
- IP Multimedia Subsystem (IMS)
- Media Gateway Control Protocol (MGCP)
- Session Initiation Protocol (SIP)
- Real-time Transport Protocol (RTP)
- Session Description Protocol (SDP)

Major Challenges

- 1. Quality of service
- 2. Susceptibility to power failure
- 3. Emergency calls

- 4. Lack of redundancy
- 5 Number portability
- 6 PSTN integration
- 7 Security
- 8 Securing VoIP
- 9 Compatibility with traditional analog telephone sets
- 10 Fax handling
- 11 Handset devices

WLANs Progress

IEEE adapted Standard 802.11 that standardised a WLAN technology in 1997, later revised in 1999.

IEEE Standard 802.11b. This is the technology that has widest deployment today (End 2005). The theoretical capacity of 802.11b is 11 Mbit/s, and the radio frequency band used is 2.4 GHz. The standard specifies the use of Complementary Code Keying (CCK) Modulation.

802.11a - A high bit-rate alternative to this standard is specified in IEEE Standard 802.11a. The theoretical capacity of IEEE 802.11a is 54 Mbit/s. The devices operate in the 5 GHz frequency band using Orthogonal Frequency Division Multiplexing (OFDM) technology.

802.11g - A high bit-rate alternative based on transmission on the 2.4 GHz radio frequency band is defined in IEEE Standard 802.11g. The OFDM technology is also specified in IEEE 802.11g, and the maximum theoretical bit-rate is 54 Mbit/s. For backward compatibility with 802.11b devices, the standard also specifies support of CCK modulation.

In a WLAN network the actual throughput will depend on,

- The radio transmission environment,
- The characteristics of the information to be transmitted (i.e. packet size),
- The number of transmitters active.

The actual throughput will therefore be lower than the maximum rate indicated.

Managing communication in a mixed b/g environment

The principle is to use short messages sent in 802.11b format,

- Request to Send (RTS)/Clear to Send (CTS) messages,
- CTS-to-shelf mechanism where only a Clear to Send message is sent to clear the air.

IEEE has recently initiated work on a new high-speed WLAN standard identified as IEEE Standard 802.11n. It will probably work in the 2.4 GHz frequency band. The transmission capacity is expected to be 2-4 times that of 802.11a/g.

DCF

DCF (Distribution Coordination Function) is the mandatory 802.11 access control function. The DCF uses CSMA/CA (Carrier Sense Multiple Access with Collision Avoidance). Before transmitting, the end-point senses the radio channels to determine if another end-point is transmitting on the same radio channel.

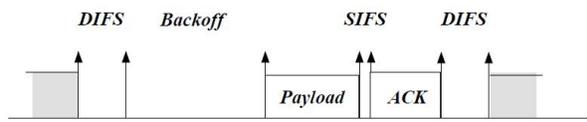


Fig 2: DCF method signal flow

The DCF has no priority mechanism and can be considered as a best effort service. It is found to have poor performance under heavy and unbalanced load conditions

PCF

The PCF (Point Coordination Function) is an optional access method. This method uses a Point Coordinator (PC). The PC shall operate at the Access Point to determine which end-point currently has the right to transmit. The operation is essentially that of polling where the PC is performing the role of polling master.

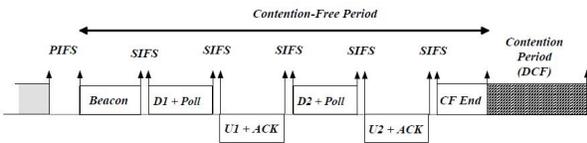


Fig 3: PCF Method Signal Flow

The PCF provides mechanisms for prioritised access to the wireless medium. Two time intervals, CFP (Contention-Free Period) and CP (Contention Period) are defined. A CFP and the following CP forms a superframe. A superframe must include a CP that is long enough to allow at least one MSDU delivery.

During CFP the PCF is used for accessing the medium, while DCF is used during CP.

QoS Extension

The basis of 802.11e is the Hybrid Coordination Function (HCF), which is the enabler for QoS support. HCF has two medium access mechanisms; Enhanced Distributed Channel Access (EDCA) and HCF Controlled Channel Access (HCCA). EDCA is used for contention access, while HCCA is for contention-free access controlled by polling from the access point.



Fig 4: Relation between HCCA and EDCA of 802.11e HCF, and relation to PCF and DCF of legacy 802.11

EDCA

EDCF addresses those two shortcomings of DCF. First, EDCF introduces the possibility of “unfairness”, i.e. that some traffic classes get a higher share of the channel

capacity. Second, the traffic classes are split into separate queues.

HCCA

HCCA is based on polling, similar to PCF.

HCCA might be useful for Voice-over-WLAN traffic, especially when the traffic load on the WLAN is high. It requires an appropriate scheduling mechanism implemented for the polling.

4. REAL TIME COMMUNICATION OVER WLAN

Voice (and real-time interactive multimedia) communication is delay sensitive. For that reason voice packets are sent with short intervals, usually 20 ms. The packets are small, between 60 bytes and 200 bytes, depending on the speech coding algorithm chosen, when using IP v4. On the other hand, typical data packets are 1400-1500 bytes.

There need to be an idle period equal to DFIS + BackoffTimer before a new station can start transmission. The minimum average value of this interval is 360 μs for 802.11b and 101,5 μs for 802.11a. In addition to the payload, each packet also contain PHY headers that are transmitted at a low bitrate, the delay introduced by these is 192 μs when using CCK (802.11b) and 60 μs when using OFDM.

The transmitting time of a 200 bytes voice packet is approx. 150 μs when the link rate is 11 Mbit/s and approx. 30 μs when the link rate is 54 Mbit/s. Compared with the sum of the DCF carrier-sense period and the time used to transmit the PHY headers, the payload transmission may be less than one third of the total time used to transmit a packet.

The consequence is that voice communications over WLAN has a large overhead.

4.1 Concept for Real Time Communication.

It is found that on an average, human voice has a speech activity factor of about 42%. There are pauses between sentences and words with no speech in either direction. Also voice communication is usually half-duplex. i.e., one person is silent while the other speaks.

One can take advantage of these two characteristics to save bandwidth by halting the transmission of cells during these silent periods. This is known as Voice Activity Detection (VAD) or silence suppression.

When the first utterance is detected after a silence period, it has to switch on transmission quickly in order to avoid loss of information that can make it difficult to understand what is said.

Another important parameter is the hangover time, i.e. the period from silence detection until transmission is stopped, The hangover time should be short in order to improve the efficiency, but it should be large enough to ensure that transmission of low-energy parts of the speech are not stopped.

Hums or short utterances like Okay, Yes and so on will reduce the advantage of silence suppression.

5. H.323 SCHEME

H.323 is a recommendation from the ITU Telecommunication Standardization Sector (ITU-T) that defines the protocols to provide audio-visual communication sessions on any packet network. The H.323 standard addresses call signaling and control, multimedia transport and control, and bandwidth control for point-to-point and multi-point conferences.

It is widely implemented by voice and videoconferencing equipment manufacturers, is used within various Internet real-time applications such as GnuGK and NetMeeting and is widely deployed worldwide by service providers and enterprises for both voice and video services over IP networks.

Unpredictability in the air-link conditions, such as rapid fading, shadowing, and intermittent disconnections, affect the radio link such that the frame error rate (FER) can be as high as 10^{-1} .

This causes serious quality degradation to the end users because the packet traffic is very sensitive to the bit error rate (BER).

To overcome the higher BER problem, the wireless air interface protocol standards included the definition of a new protocol layer, called the Radio Link Protocol (RLP), on top of the MAC (medium access control) layer.

The RLP layer provides higher reliability of the air link by using the ARQ (Automatic Repeat request) protocol in conjunction with the cyclic redundancy check (CRC). Thus, the wireless air interface is made to interwork with IP networks to support useful data services.

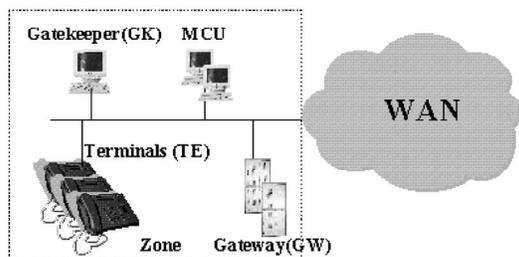


Fig 5: H.323 system components

H.323 protocol components

Terminals, gateways, gatekeepers, and multipoint control units (MCUs).

The terminals are the client endpoints that provide real-time, two-way communications. All H.323 terminals must also support the H.245 protocol, which is used to negotiate channel usage and capabilities.

Three other components are required: 1) H.225 for call signaling and call set-up, 2) a component called Registration/Admission/Status (RAS) protocol used to communicate with a gatekeeper, and 3) support of RTP/RTCP for sequencing audio and video packets.

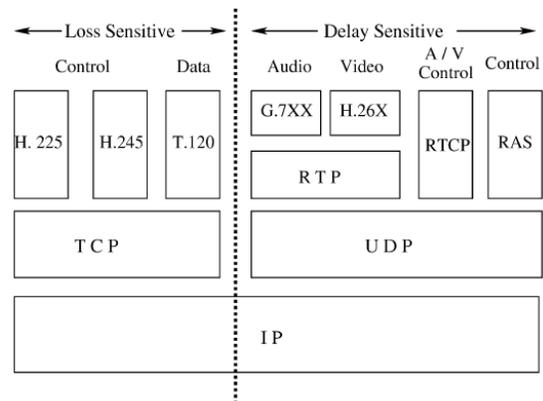


Fig 6: H.323 Protocol Suite

Components-

H.225.0 Registration, Admission and Status (RAS), which is used between an H.323 endpoint and a Gatekeeper to provide address resolution and admission control services.

H.225.0 Call Signaling, which is used between any two H.323 entities in order to establish communication.

H.245 control protocol for multimedia communication, which describes the messages and procedures used for capability exchange, opening and closing logical channels for audio, video and data, control and indications.

Real-time Transport Protocol (RTP), which is used for sending or receiving multimedia information (voice, video, or text) between any two entities.

Many H.323 systems also implement other protocols that are defined in various ITU-T Recommendations to provide supplementary services support or deliver other functionality to the user. Some of those Recommendations are:

H.235 series describes security within H.323, including security for both signaling and media.

H.239 describes dual stream use in videoconferencing, usually one for live video, the other for still images.

H.450 series describes various supplementary services.

The Real-time Transport Protocol (RTP) used in conjunction with H.323 provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video, over a connectionless network [6] with the ability to carry timing information.

The protocol RTP/RTCP, along with UDP (unified datagram protocol) has the capability to provide bare-bones real-time services with minimum reliability.

Call Setup in H.323

3 Step Process

- call signaling,
- establishing communication channel for signaling
- establishing media channels.

In the first phase of call signaling, the H.323 client requests permission from the (optional) gatekeeper to communicate with the network. Once the call is admitted, the rest of the

call signaling will proceed according to one of the several call models

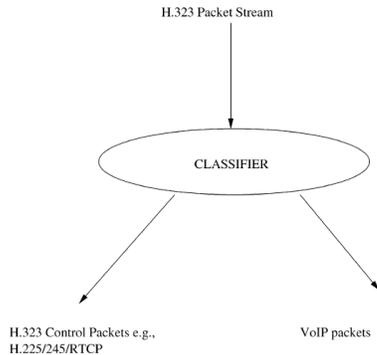


Fig 7: Subflow classes requiring different treatment

6. PROPOSED SCHEME

Multiplexing of data packets along with their scheduling with DRR scheme.

Here the two systems of Improved DRR and Multiplexing are simultaneously employed so as to derive benefits of each of the systems

6.1. DRR Technique

PCF scheme has some advantages over contention-based DCF scheme in that it is possible to guarantee various QoS requirements to real-time applications. But PCF scheme has inherent limitation in the polling scheme. The access point (AP) polls every station (STA) in its polling list in sequence and checks if each STA has packet(s) to transmit.

This will cause unnecessary delay and interruption to other STAs with packets to transmit. So it is not suitable for the transmission of the traffics with different characteristics, especially for the voice traffic, which is sensitive to delay.

Deficit Round Robin (DRR) is a well known round robin scheme and often used.

By improving existing algorithm, the average access delays can be reduced and throughput of VoWLAN can be increased.

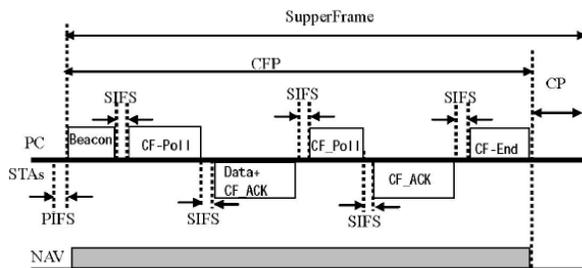


Fig 8: Traditional PCF

Limitation of Basic PCF Polling

- Bandwidth wastage.
- Unnecessary Delays.

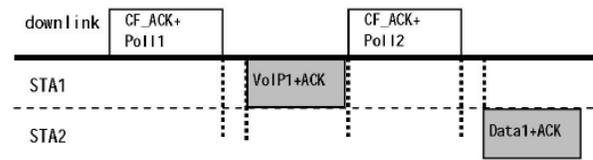


Fig 9: Different STAs transmit the VoIP and data traffic

It adopts a poll-and-response protocol to control the access to the shared wireless medium and eliminate contention among STAs. It makes use of the priority inter-frame space (PIFS) to seize and maintain control of the medium. The period during which the PCF is operated is called the contention-free period (CFP). Once the PC (which resides in AP), has control of the medium, it may start transmitting downlink traffic to STAs. Alternatively, the PC can also send contention-free poll (CF-Poll) frames to those STAs that have requested contention-free services for their uplink traffic. During a CFP, a STA can only transmit after being polled by the PC. If a polled STA has uplink traffic to send, it may transmit one frame for each CF-Poll received. Otherwise, it will respond with a NULL frame, which is a data frame without any payload. Besides, in order to utilize the medium more efficiently during the CFP, it is possible to piggyback both the acknowledgment (CF-Ack) and the CF-Poll onto data frames. During the CFP, the PC sends a frame to a STA and expects the reply frame, either a CF-Ack or a data frame or a NULL frame in response to a CF-Poll, within a short inter-frame space (SIFS) that is shorter than PIFS.

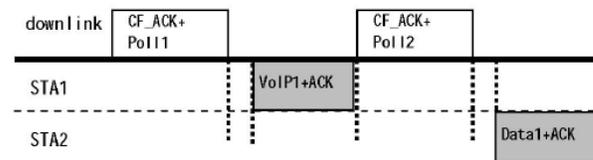


Fig 10: Different STAs transmit VoIP and Data Traffic

In Fig. 2 (above), both of them have packets to transmit. STA1 transmits VoIP packets while STA2 transmits common data packets. According to the basic PCF polling scheme, when STA1 is polled by the AP, it will transmit VoIP packets; but next transmission turn is for STA2, hence STA1's VoIP traffic will be interrupted, and STA1's VoIP packets will be delayed. And this issue will be more serious when the number of STAs increased too much in a WLAN cell, which will lead to the worst performance of voice transmission over WLAN.

Hence the proposed algorithm can improve the QoS of VoIP applications over WLAN by providing lower access delay, even when the number of STAs increased rapidly.

DRR can operate well under variable length load but, in cases of VoIP data, it has poor performance. By improving Scheduling in DRR schemes, VoIP can be very well supported.

Improved DRR Algo

There are two modules in the improved algorithm: Enqueuing module and Dequeuing module. In the Enqueuing module, we use two different queues for different kind of packets: Queue1 for VoIP packets and Queue2 for common data packets; and use two list of active flows: ActiveList1 and ActiveList2 for the sequence number of the different STA in the polling list. ActiveList1 used for the STAs that will send VoIP packets and ActiveList2 for those STAs that will send common data packets. The sequence number of different STA will be stored in the different ActiveList, which will be benefit for the management of a larger number of STAs. Hence the proposed algorithm can improve the QoS of VoIP applications over WLAN by providing lower access delay, even when the number of STAs increased rapidly.

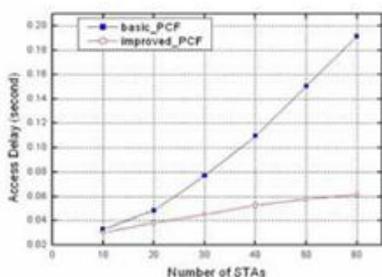


Fig 11: Delay Analysis for different number of STAs

6.2. MUX/DEMUX SCHEME FOR H.323

This scheme proposes a multiplexing scheme for improving the bandwidth efficiency of existing VoIP applications. By installing a multiplexer in an H.323 proxy, voice packets from multiple sources are combined into one IP packet for transmission. A demultiplexer at the receiver-end proxy restores the original voice packets before delivering them to the end-user applications. Results show that the multiplexing scheme can increase bandwidth efficiency by as much as 300%. The multiplexing scheme is fully compatible with existing H.323-compliant VoIP applications and can be readily deployed.

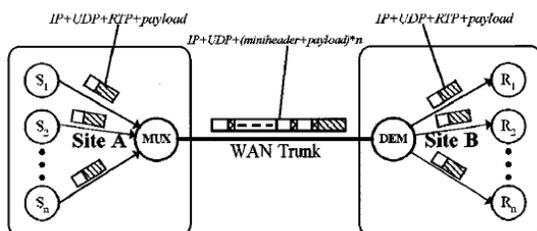


Fig 12: System Architecture

As bandwidth in local networks is abundant, VoIP applications can send each audio frame in a separate RTP packet to minimize packetization delay. For voice traffic to be transported over the same WAN link, they go through a multiplexer (MUX) located at the sender-side network, and a demultiplexer (DEMUX) located at the receiver-side network. The MUX replaces the RTP header of each voice packet with a compressed miniheader and then aggregates multiple packets into a single multiplexed packet. All these chunks share a common UDP/IP header and are then sent to the DEMUX across the WAN link. On the receiver-side network, the DEMUX inspects the miniheaders inside each multiplexed packet, restores the original RTP headers and destination information, and reconstructs the individual voice packets for delivery to their respective receivers.

Connection Procedure

The whole connection procedures are as follows.

- 1) The H.323 initiator registers at MUX A and sends it an *Admission Request* (ARQ) carrying the alias of the H.323 acceptor.
- 2) MUX A searches for the acceptor by broadcasting a *Location Request* (LRQ) to its MUX neighborhood, including MUX B.

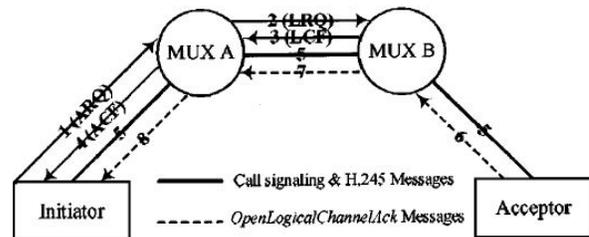


Fig 13: Connection Procedure

- 3) Assuming the called party is currently logging into MUX B, MUX B will reply MUX A with a *Location Confirm* (LCF). For routing the signaling messages through MUX B, its LCF carries its own Call Signaling Channel Transport Address instead of the original called party.
- 4) MUX A sends an *Admission Confirm* (ACF) to the initiator, with the Transport Address set to itself instead of MUX B.
- 5) The endpoints exchange call signaling and H.245 control messages through the MUXs.
- 6) However, when MUX B receives the *OpenLogicalChannelAck* (ACK) message from the acceptor, which contains this endpoint's destination IP address and port number for the RTP stream from the initiator, MUX B stores these two fields in a new entry of its DEMUX table and replaces them in the ACK with its IP address and the receiving port of this multiplexed channel. Meanwhile, a unique CID is assigned to the RTP stream and is embedded into an extra and optional *portNumber* field in

the ACK, which is not used originally. This message is then forwarded to MUX A.

7) When MUX A receives this modified message, it also creates a new entry in its MUX table with the specified CID. After that, it records the DEMUX IP and receiving port in the channel table if needed and replaces them in the ACK with its IP address and receiving port for the RTP source.

8) The initiator receives the ACK from MUX A and starts transmitting voice packets to it with the given IP address and port number. [Step 6)–8) will be repeated in the opposite direction when the initiator feeds back an ACK to the acceptor through the MUXs for establishing an RTP session from the acceptor to initiator.]

After successful connection setup, the first few voice packets from this particular source will not be compressed when it is being multiplexed with the voice chunks from other sources so that the remaining fields in the DEMUX table entry of MUX B can be filled in. After that, MUX A will then compress and multiplex the RTP packets headed for the acceptor by maintaining the CID and the multiplexed channel negotiated during start up.

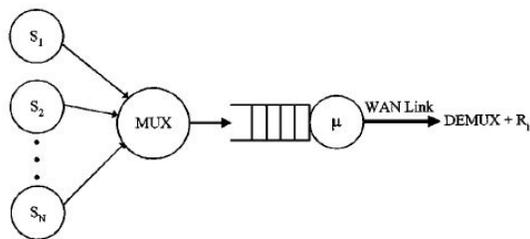


Fig 14: Network Scenario in Simulations

The diagram depicts the network scenario in simulations. The WAN trunk is modeled as a first-in first-out (FIFO) queue with service rate = μ and a fixed buffer size. Incoming packets are discarded when the buffer overflows.

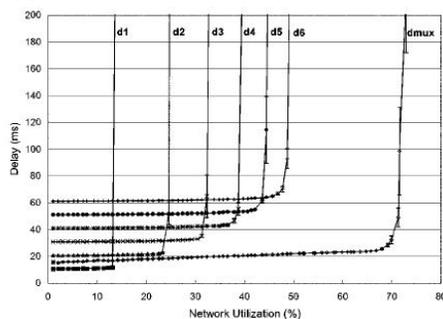


Fig 15: Comparison of overall delay and WAN link utilization between conventional schemes and proposed scheme.

Figure depicts the average overall packet delay, which includes assembly delay, sojourn time at the WAN link and waiting time at MUX if it is used, at different WAN link utilization. We set the WAN trunk bandwidth to 512 Kb/s

and vary the network utilization by increasing the number of simultaneous connections N .

CONCLUSION

Voice integration with the existing data networks is an area of huge demand. It induces many benefits, and the most important is that it will reduce the cost of communication to a very large extent.

The proposed scheme shows best improvements in the existing utilization and it can be well implemented. As it utilises existing networks, so effective integration of voice with WLAN is possible.

WLAN has been into existence since a long time, but due to irregular flow of data, voice could not be used into it. But due to use effective techniques and availability of more bandwidth, the objective can be achieved.

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