Chapter 2

Literature Survey

2.1 Introduction

This dissertation contributes protocols and schemes for provisioning QoS in multi-hop WiFi-based long distance networks. In this context, this chapter provides a comprehensive survey on various works done in the field of QoS provisioning in multi-hop wireless networks. Tuning of scheduling and routing protocols are considered as highly effective way of provisioning QoS. Therefore, due emphasis has been given on MAC protocols and routing schemes while doing this survey. This survey will provide a strong foundation to appreciate the different protocols and schemes developed throughout this dissertation. The rest of this chapter is organized as follows. Section **??** introduces the architecture of multi-hop WiLD networks. Various applications envisaged to run over WiLD networks along with their QoS requirements are discussed in Section **??**. Section **??** presents various techniques and schemes which can be used for provisioning QoS in multi-hop WiLD networks. Considering the role of MAC protocols in provisioning of QoS, this section discusses the MAC protocols for multi-hop WiLD networks. Further, it provides a brief survey on dynamic bandwidth allocation schemes which can be useful in QoS provisioning over such networks. Packet level scheduling schemes which provide a fine-grained QoS are also presented in this section. This section further discusses the various multi-path routing schemes proposed for supporting QoS in multi-hop wireless networks. Finally, Section **??** concludes this chapter.

2.2 WiFi-based Long Distance (WiLD) Networks

Although WiFi was originally designed to support Wireless Local Area Networks (WLANs) for short-range communication, it has become very popular in extending Internet connectivity to the remote underserved areas using long distance links in recent time [?]. The widespread standard for WLANs, IEEE 802.11b/g/n operating in the 2.4 GHz frequency band is one of the most popular wireless standards. The license-free operations in the ISM band and varieties of low cost IEEE 802.11 hardware commodities make WiFi an attractive and economically feasible communication technology for rural use [?]. Various research works such as in [?, ?, ?] have also established the viability of WiFi as a practical solution for long distance communication.

Many WiLD mesh networks including research test beds are deployed in different corners of the world. Few notable real life WiLD networks deployed are Digital Gangetic Plains (DGP) [?] in Uttar Pradesh, India, Aravind Network for Telemedicine [?] in Tamil Nadu, India, Long Distance Network in Amazonian Jungle of Peru [?] for telemedicine and telephony, and Akshaya Network [?] for e-governance in Kerala, India. Several research test beds have also been set up in recent times. MIT's Roofnet [?], QuRiNet [?] at the Quail Ridge Reserve in Napa County, California, FRACTEL [?] at IIT Bombay, India, Hop-Scotch [?] of Scottish Highland, UK and VillageNet [?] are some of the important WiLD network research test beds which are working towards network performance enhancement, providing support for various envisaged applications particularly real-time applications such as e-learning, e-governance, telemedicine, and telephony.

The multi-hop WiLD network is a gateway-based converge cast wireless mesh network. These networks are typically used to extend Internet connectivity from some points having high bandwidth connectivity to the rural dispersed areas located far away from such points. The node interfaced with the high speed connectivity acts as a gateway to the WiLD network. Gateway node is entrusted with the responsibility of routing packets to and from the WiLD network domain. Based on the functionality, the other nodes of WiLD networks can be classified into two types: wireless router and wireless client. The wireless routers function as relay nodes for forwarding traffic over multiple hops. The wireless routers form the multi-hop backbone for providing Internet connectivity to the wireless clients. On the other hand, the wireless clients are the end-points in such networks which are entrusted with the functionality of end user connectivity. Internet connectivity can be shared among the rural users by using any local area network technology.

WiFi-based backhaul networks are comprised of long distance point-to-point (P2P) and point-to-multipoint (P2MP) links enabled by high-gain directional antennas. Such links usually ranges from a few kilometers up to a few tens of kilometers to cover long distances. The gateway node in WiLD network is ordinarily located in the district headquarters or some points which are connected to high speed Internet. Such rural network backbone consists of multiple intermediate nodes usually installed on high-raise towers of about 25-50 meters height in order to achieve line of sight (LoS) between the communicating radios. The installation cost involved in erecting tall towers significantly attributes to the overall deployment cost of those networks. The nodes are connected by long distance P2P wireless links with high gain (23-27dBi) directional antennas and beam-width of about 8^0-15^0. For parabolic grid antenna having beam width about 15^0, an angular separation of 30^0-45^0 works well [?]. The longest successful WiFi link so far covers 382kms of distance which used 30dBi antenna in a favorable topography of Venezuela [?]. The link achieved a total of 6$Mbps$ bidirectional throughput by implementing TDMA protocol at the nodes. Nodes in the network are equipped with multiple radio interfaces which enables multi-hop transmission of traffic over such links. Each radio employed in a node of WiLD network is capable of both transmitting and receiving but in a half duplex manner. Wireless clients are connected to the gateway through intermediate nodes which in turn provide connectivity to the local access points using P2MP links. A sector antenna used in a P2MP wireless link typically has a gain of about 17-19dBi and a beam-width of about 30^0-90^0 [?].

A typical WiLD network architecture is depicted in Figure **??** which is redrawn from the architecture proposed by FRACTEL [?]. The architecture assumes that most of the villages can be reached from their district headquarters by a few number of hops; a single hop distance typically being about 1-40km long [?]. The local-access links extend connectivity from this point to multiple nearby locations. Such nearby locations might include individual buildings such as schools, health centers, community centers and residential buildings.

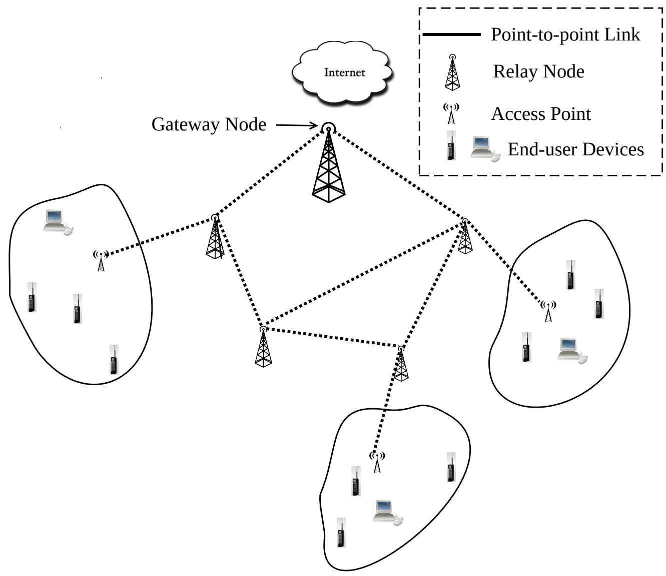

Figure 2-1: WiFi-based Long Distance Network Architecture

2.3 Envisaged Rural Applications and their QoS Requirements

Application of ICT can be productive in supporting sustainable development in the fields of public administration, business, education and training, health, employment, environment, agriculture and science. To minimize the gap between the urban and rural communities, ICT infrastructures need to be extended to the rural areas equally so that the rural people are also involved homogeneously in such development. A few prospective network applications which are expected to facilitate in extending services and facilities to the rural areas are discussed in the following subsections:

Electronic Learning (E-learning)

E-learning refers to the utilization of information systems and ICT in educational services. ICTs can contribute to achieving universal education worldwide through delivery of education and training of teachers, and offering improved conditions for lifelong learning. Increased and improved education through ICTs can be provided to the people who cannot afford it otherwise. The ultimate goal of e-learning is to bring the learning to the learners, not to bring the learners to learning. Various applications and processes such as web-based learning, computer-based learning, virtual classrooms, and digital collaboration are some of the examples of e-learning methodologies. According to Dias et al. [?], if efficiently used, ICT can play a key role in supporting immediate needs of the (online) learning environments; however many educational institutions need to incorporate technologies. E-learning is widely accepted in developed countries whereas due to lack of proper ICT infrastructure, the developing or underdeveloped countries are not able acquire e-learning benefits in full swing.

Internet Protocol Telephony (IP Telephony)

IP telephony is a term used for the technologies that use the Internet Protocol's packet switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the Public Switched Telephone Network (PSTN). Over the Internet, calls travel as packets of data on shared lines avoiding the tolls of the PSTN. IP telephony needs to deliver the voice, fax, or video packets in a dependable flow to the user. Voice over IP (VoIP) is the family of technology that allows IP networks to be used for voice applications such as telephony, voice instant messaging and teleconferencing. It defines a way to carry voice calls over an IP network including the digitization and packetization of the voice streams [?]. VoIP can work towards offering cheaper calls to make business further easier and contribute towards the development of rural regions.

E-health care and e-medicine

The last decade has seen a radical transformation of healthcare using ICT, particularly in emerging economies. The term e-health has been used to refer to the use of ICT in delivering healthcare services [?]. World Health Organization (WHO) defines e-health as the cost-effective and secure use of ICTs in support of health and health-related fields, including health-care services, health surveillance, health literature, and health education, knowledge and research [?]. A wide variety of e-health services exists, including health information networks, Electronic Health Record (EHR), telemedicine services, wearable and portable systems which communicate, health portals, and many other ICT-based tools assisting disease prevention, diagnosis, treatment, health monitoring, and lifestyle management. M-health (Mobile Health) is another related term which refers to the use of "mobile computing, medical sensor, and communications technologies for health care" [?]. Real-time access to electronic information on new technologies and treatments will empower the patients and make them responsible for their own health [?].

E-governance

E-governance refers to the application of ICT to transform the efficiency, effectiveness, transparency and accountability of exchange of information and transaction between governments, between government agencies, between government and citizens, and between Government and businesses. E-governance can transform citizen service, provide access to information to empower citizens, enable their participation in government and enhance citizen economic and social opportunities so that they can make better lives, for themselves and for the next generation. Good governance, which is a key to development can be enabled by e-governance, if appropriately implemented [?].

E-agriculture

Agriculture can serve as an important engine for economic growth in developing countries [?]. Despite the importance of agriculture for development, agricultural production has lagged far behind particularly in developed countries over the past few decades. One possible reason of stagnating growth is the underutilization of improved agricultural technologies including ICT [?]. E-agriculture is an emerging field focusing on the enhancement of agricultural and rural development through improved information and communication processes. ICT can revolutionize the farming sector and can benefit all farmers, including small land holders, marginalized, and poor farmers. E-agriculture is one of the action lines identified in the declaration and plan of action of the WSIS, 2014 [?]. The main phases of the agriculture industry, i.e., crop cultivation, water management, fertilizer application, pest management, harvesting, post-harvest handling, transport of food products, packaging, food preservation, food processing, quality management, food safety, food storage, and food marketing need information and knowledge to manage them efficiently. Risks in agriculture can be mitigated by timely action and through the application of best practices. Sharing of agriculture related information like type of latest crops, sell rate of different agricultural products in nearby markets, seeds, pesticides, and fertilizers to be used in different situations will be easier using ICT than before. In this process, the farmers are expected to be greatly benefited.

Other Common Applications

ICT has greatly impacted and enhanced global socialization and interactions. In fact, ICT has taken over nearly every aspect of our daily lives from commerce (buying and selling) to leisure and even culture. Today mobile phones, desktop computers, hand-held devices have become a central part of our culture and society. These technologies play a vital role in our day to day operations. ICT can also facilitate the students in getting results online, submitting application for job and admission online. Common applications using Internet like e-mail, e-commerce, Internet banking, social networking, and web browsing are also expected to be used by the peoples living in rural regions.

Smooth running of various real-time applications demand some quality assurances from the underlying networks. To support a full motion video, the data rate below some threshold is not acceptable. Regardless of the type of application, ITU-T Rec. G.114 recommends to not exceed a one-way delay of $400ms$ for general network planning. However, it is desirable to keep the delays seen by user applications as low as possible [?]. The amount of delay which cannot be perceived by the users in real-time conversation is $150ms$ [?, ?, ?]. In interactive communication, desirable jitter value should be lesser than $100ms$. Internet banking applications need to be extremely secure and reliable as they deal with financial transactions. Banking transactions should be completed before server time out occurs and hence such applications are strict delay sensitive. Depending upon the type of QoS required by different services, the services or network traffic generated by those services may be categorized into different discrete classes and priorities can accordingly be assigned to the applications envisaged to run in the networks deployed in rural areas. QoS requirements of some important applications discussed above has been summarized in Table ??.

Different real-time applications can be broadly construed as streaming and interactive audio/video. For various applications, their QoS requirements and the standard codecs used are discussed in the following subsections:

Table 2.1: QoS Requirements of Rural Applications

Service Type	**Application Type**	**Expected QoS Requirements**			
		Required Throughput	**Small Delay**	**Small Jitter**	**Packet Loss**
VoIP/Tele-consultation	Voice Communication	Low (strict)	Yes (strict)	Yes	Low
Tele-diagnosis	E-health	Medium/High	Yes	No	Low
Video Streaming/ Video Consultation	Entertainment/E-health/ e-learning	High	No	No	Moderate
Tele-monitoring/ Soil Testing	E-agriculture	Low	No	No	Low (strict)
Tele-education/ Online Classes	E-health/E-learning	High	No	No	Moderate
Access to EHR/ Best Effort Traffic	E-health/Internet browsing/E-mail/ E-agriculture etc.	Low/High	No	No	Nil

Multimedia Streaming

Audio on Demand (AoD) enables users to listen to sound particularly music. AoD uses standards such as MPEG-1 and MPEG-2 with different data rates. MPEG-1 was primarily designed for storing video data (moving pictures) and its associated audio data on digital storage media. It is a compatible lossy audio/video format which also introduced mp3 audio format. MPEG-2 employing Advanced Audio Coding (AAC) provides CD quality mp2 audio stream. Video on Demand (VoD) system supports the MPEG family of protocols, namely MPEG-1, MPEG-2, and MPEG-4 [?]. MPEG-2 substantially reduces the bandwidth required to transmit a high-quality digital video signal, and it optimizes the trade-offs between resolution and the required transmission bandwidth. MPEG-4 is an international standard for coding of audiovisual objects that provides technologies for the manipulation, storage and communication of multimedia objects. Throughput requirement of video streaming basically depends on the codec used. The expected delay should be less than 5 *Seconds* [?]. Jitter requirement of VoD application depends on the video application's buffering capabilities. Packet loss rate should not exceed 5% [?].

Video Conferencing

ITU Telecommunication Standardization Sector (ITU-T) recommended video conferencing protocol stack over packet based Internet, H.323 implements audio codecs such as G.711, G.722, G.723.1, and G.729. Different audio codecs used in videoconferencing applications along with their QoS requirements are shown in Table **??**. Audio transmission does not demand very high data rate, however delay as well as jitter are very sensitive. The required guaranteed priority bandwidth per call depends on many factors like sampling rate and the VoIP codec used. Callers usually notice round trip voice delays of 250ms or more. ITU-T G.114 recommends a maximum of a 150ms one-way latency as other public network can be available as a part of the path. Maximum tolerable jitter is less than 30ms. Up to 1% packet loss is acceptable in this kind of applications.

Table 2.2: Different Audio Codecs used in H.323 and their QoS Requirements

Coding Standard	**Expected QoS Requirements** [?, ?, ?, ?]			
	Data Rate ($Kbps$)	**Max. Delay** (ms)	**Max. Jitter** (ms)	**Loss Rate** (%)
G.711	64	150	< 30	< 1
G.726	16/24/32/40	150	< 30	< 1
G.728	16	150	< 30	< 1
G.729	8	150	< 30	< 1
G.723.1	5.3/6.3	150	< 30	< 1
G.729.1	8-32	150	< 30	< 1

H.323 also implements H.261, H.263 and H.264 video codecs. Video transmission requires more bandwidth than audio whereas delay and jitter, acceptable packet loss requirements are same as audio. Video conferencing requires over-provisioning of minimum-priority bandwidth guarantee to the size of the videoconferencing session plus 20 percent [?]. The QoS requirements for video transmission in videoconferencing in presented in Table **??**.

Table 2.3: Different Video Codecs used in H.323 and their QoS Requirements

Coding Standard	**Expected QoS Requirements** [?, ?, ?]			
	Data Rate ($Kbps$)	**Max. Delay** (ms)	**Max. Jitter** (ms)	**Loss Rate** (%)
H.320	64-1920	150	< 30	< 1
H.323	64X	150	< 30	< 1
H.324	< 64	150	< 300	< 1

Voice over IP (VoIP)

VoIP also uses the same codec as audio conferencing, i.e., G.711, G.722, G.723.1, and G.729. Although VoIP can tolerate upto 1% packet loss by using packet loss concealment algorithm (to minimize effect of packet loss); to ensure good voice quality, VoIP networks are typically designed for very close to 0% packet loss [?]. Acceptable delay and jitter in case of VoIP should be below 150ms and 30ms respectively.

Data Applications

The conventional data applications such as HTTP, FTP, TELNET and E-mail are based on file transfer mechanism. Although the applications like E-mail and HTTP do not require very high bandwidth, but they cannot tolerate any packet loss and transmission error. A bit relaxed delay is acceptable to such applications whereas jitter is not having much importance. The QoS requirements for this type of services are given in Table ??.

Table 2.4: QoS Requirements for FTP, E-mail & Web Browsing

Application	**Expected QoS Requirements** [?, ?]				
	Response Time	**Data Rate**	**Delay**	**Jitter**	**Loss Rate**
FTP	2-5 Sec.	High	Medium	N/A	Zero
E-mail	2-5 Sec.	Low	Low	N/A	Zero
Web Browsing	2-5 Sec.	Low	Low	N/A	Zero

2.4 QoS Provisioning in Multi-hop WiLD Networks

The promising future of mesh networks has brought various applications into the picture. Real-time applications such as e-learning, e-governance, tele-medicine, disaster relief, and emergency response systems are also expected to be running over WiLD networks. As these kinds of applications demand certain levels of quality for their successful operation, QoS provisioning in multi-hop WiLD network has become a growing need. QoS requirements of different applications may vary substantially with parameters such as throughput, delay, jitter, packet loss, packet error rate, reliability, etc. Provisioning of QoS for real-time applications like voice and video over multi-hop WiLD networks is considered to be highly challenging. Running heterogeneous traffic with stringent delay and bandwidth demands simultaneously over such network makes the problem even more complicated. Since throughput and delay has greater impact in most of the real-time transmissions, we consider both throughput and delay as QoS parameters in this work.

As bandwidth is one of the scarcest resources in wireless networks, requirement-based allocation of bandwidth among different applications or nodes is obligatory as QoS requirements vary from application to application. Unreliable lossy links in wireless networks not only hinders the maximum achievable throughput of a network but also affects its energy constraints due to retransmissions and broadcasting [?]. The multi-hop traffic forwarding in wireless mesh networks are often accompanied by scheduling delay which greatly impacts the QoS parameters such as latency and jitter. QoS-aware routing protocols are used to establish a path from source to destination which meets the QoS needs of certain traffic. The routing protocols which include path computation, path selection, path recovery, and path optimization algorithms need to be stable and robust against heterogeneous network environments and traffic variation.

E.800 Terms and definitions for QoS and network performance [?] defines QoS as "the collective effect of service performance which determines the degree of satisfaction of a user of the service". QoS refers to a set of qualitative and quantitative traffic characteristics which describes a traffic flow in support of a specific application. QoS is a set of service requirements such as throughput, latency, jitter, and packet loss to be met by the network while transporting a particular flow of data [?]. However, these parameters may vary from application to application, and can also be combination of more than one. Among these parameters, throughput and delay are considered to be the most important QoS parameters for real-time applications. Generally considered characteristics are throughput, service interval, packet size, delay, jitter, security, priority, constant bit rate, availability, etc.

A QoS model is a mechanism for provisioning of QoS resources to meet the QoS requirements. IETF has developed two of the main QoS provisioning models in the Internet: (i) Integrated Services (IntServ) [?], and (ii) Differentiated Services (DiffServ) [?]. IntServ maintains per-flow reservation states at QoS network entities aiming at a greater level of accuracy and a finer level of granularity. It uses Resource Reservation Protocol (RSVP) to explicitly signal the QoS needs of an application's traffic along the devices in the end-to-end path through the network. If every intermediate network device along the path can reserve the necessary bandwidth, the originating application can begin transmitting. On the other hand, DiffServ is a packet level QoS mechanism which relies on aggregation and differential treatment of traffic classes. It provides much better scaling compared to IntServ with low accuracy [?].

The following two sections discuss how QoS can be provisioned at MAC and routing layer of WiLD networks.

2.4.1 QoS at MAC Layer

MAC protocol is responsible for actual scheduling of traffic on the air. Provisioning of QoS guarantees via MAC layer has huge potential and a prolific field of research. While the existing IEEE 802.11a/b/g/n standards do not provide any QoS support, different alternatives such as 802.11e, 802.16, and 802.11s have addressed the QoS requirements of different real-time applications to some extent. Contention-based protocols usually support real-time communication by tuning some of the parameters of CSMA/CA such as the initial backoff and contention window. Due to the randomised backoff approach, CSMA/CA may not be suitable for real-time applications requiring some level of guaranteed performance. The problem of scheduling transmissions in TDMA-based mesh networks is an active and stimulating area of research [?, ?, ?]. Contention-free TDMA protocols are better candidates for providing predictable performance guarantees as it allocates a certain transmission opportunities to the nodes. QoS provisioning schemes for multi-hop WiLD network must address the issues born due to the architectural considerations of WiLD networks. In an interference prone WiLD network, scheduling of links transmission keeping the QoS requirements of the applications in mind can facilitate some level of QoS guarantees. Packets belonging to certain priority classes can also be provided with some priority while scheduling them for transmission.

Link level and packet level scheduling schemes with reference to multi-hop WiLD network are discussed in the following subsections.

Link Level Scheduling

multi-hop WiLD networks employing point-to-point directional antennas pose unique link level scheduling challenges. Nodes connecting multiple neighbours use multiple radios for creating links to each one of them. Number of usable channels is one of the important considerations in such architecture. Use of single channel makes scheduling task more difficult in multi-hop scenario. Links transmission needs to be scheduled such that interference possibility is as minimum as possible.

A detail survey on link scheduling in multi-hop WiLD network is presented in subsection **??**.

In provisioning QoS to various real-time applications, many dynamic bandwidth allocation schemes are found in the literature of wireless sensor network (WSN). Since the architecture of multi-hop WiLD network resembles with WSN, the dynamic bandwidth allocation schemes proposed for WSN are also included in the survey. Some of the relevant dynamic bandwidth allocation schemes are discussed in subsection **??**.

Packet Level Scheduling

Packet scheduling is the process of assigning packets to appropriate shared resources to achieve some performance guarantee. In providing QoS to some applications, packet scheduling scheme is heavily used. In the context of multi-hop WiLD networks, once the TDMA slots are assigned by the link level scheduler, the nodes may utilise packet scheduling to prioritise some of the available packets transmission. In this process, packet scheduling can provide granular QoS to some applications. Some relevant packet level scheduling schemes for TDMA based networks are discussed in the subsubsection **??**.

2.4.1.1 MAC Protocols for WiLD Networks and their QoS Support

Although 802.11 offers a few non-overlapping channels, most of the WiLD network set up uses a single channel for creating backhaul network. Single channel assumption allows the other channels to be used for local access networks restricting the radio frequency pollution problem to occur. Internet connectivity reaches the rural end-user nodes via multiple WiLD links. In supporting multi-hop communication using directional WiLD links, multiple radios are employed at various non-terminal nodes. Use of more than one radio in a node, and all of them using the same channel does not allow independent transmission and reception at different radios. Existence of multiple radios at a node has been depicted in Figure **??**. The node A can either simultaneously transmit to the nodes C, D, and G or receive from all of them simultaneously.

Raman et al. in [**?**, **?**] studied on the possibility of performing independent receive/transmit operations by multiple radios available at a node. They reported that the long distance links on a node with high gain directional antennas suffer

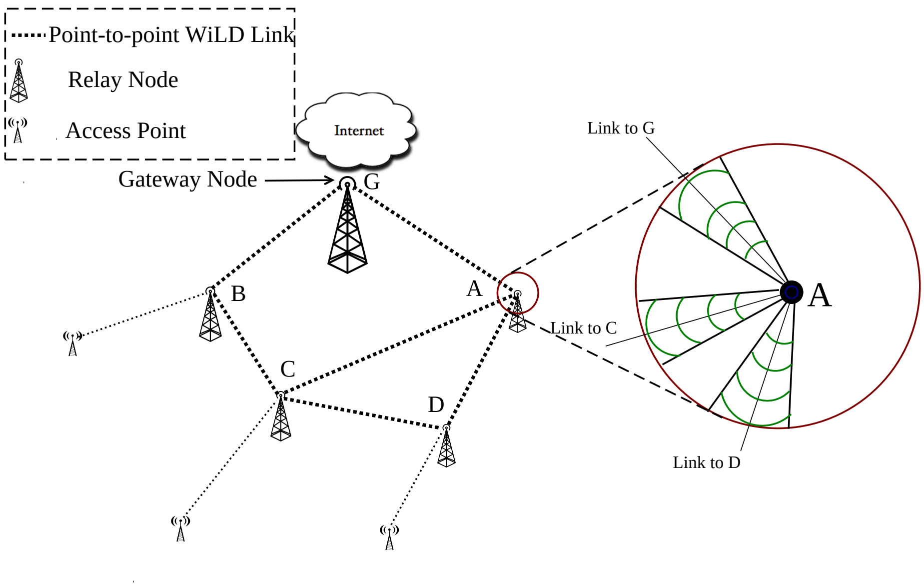

Figure 2-2: Multiradio Operation at a Single Node

from side-lobe interference and cannot operate independently in the same channel. However, with proper design, simultaneous transmit (SynTx) and simultaneous receive (SynRx) operations in all the radios are possible but mix of transmit and receive operations (Mix-Tx-Rx) by the radios are prone to interference. They termed this SynTx and SynRx operation as Simultaneous Synchronous Operation (SynOp). SynTx, SynRx and Mix-Tx-Rx operations are explained in Figure **??** in detail.

MAC protocol plays a key role in optimally utilizing the shared transmission medium which directly impacts on overall network performance. It solves the main sources of energy waste problems such as collision, idle listening, overhearing, and control packet overhead. The wireless channel access mechanisms can broadly be classified as contention-based, contention-free, and hybrid. The first two types are the dominant channel access mechanisms in the literature of WMN. Most of the reported protocols found in these two categories are based on either Carrier Sense Multiple Access (CSMA) or Time Division Multiple Access (TDMA). A carrier sensing MAC such as CSMA/CA cannot perform SynOp in multi-radio configuration as radios can hear each others transmission causing other radios to backoff [**?**]. However, this is not the case with TDMA-based MAC protocols.

Long distance wireless links are highly unreliable due to the factors such as signal fading and interference which limits the overall network performance. Multiradio operation in a single node of WiLD network directly impacts the end-to-end performance. Transmission of multiple radios needs to be meticulously scheduled for the entire WiLD network so that interference is as minimum as possible. End-to-end throughput and delay performances are constrained by SynOp operation of multiple radios in multi-hop WiLD networks.

CSMA/CA based MAC Protocols for WiLD Networks

CSMA/CA is a channel access mechanism which was originally designed to resolve contention in indoor conditions. In CSMA mechanism, nodes contend for the shared channel for a specified time before transmission of data, thus ensuring that the channel is free. Only after making sure that the channel is free, a node starts its transmission. If the channel is sensed busy, the node defers its transmission until it becomes idle. Collision avoidance mechanism is used to improve the performance of the CSMA method by attempting to share the channel somewhat fairly among all the transmitting nodes available within the collision domain.

Since the standard CSMA/CA channel access mechanism was not designed for long distance operation, real WiLD links set up by using off-the-shelf WiFi hardware show abysmal end-to-end performance [?, ?, ?, ?, ?]. The expected performance of such networks in supporting various real-time applications such as e-learning and tele-medicine cannot be delivered by the existing MAC protocols unless redesigned or tuned properly. The key reasons for this detrimental performance are highlighted in [?, ?] as (i) high probability of packet loss, (ii) inefficient acknowledgement mechanism, and (3) possible interference. Since the signal propagation time increases in long distance links, the probability of packet loss due to collision also proportionally increases [?]. Increased propagation delay with the increase of link distance makes the sender to wait for a longer time to receive the acknowledgement packet. This acknowledgement mechanism decreases the channel utilization significantly. Due to large side lobes, the adjacent WiLD links operating in the same channel are vulnerable to interference.

TDMA-based MAC Protocols for WiLD Networks

TDMA channel access method is used for sharing medium among the nodes available within a common transmission range. It allows multiple stations to share the same frequency channel by dividing the transmission time into discrete slots. In TDMA mechanism, the channel is bound by a superframe structure which is comprised of a number of time slots allocated by a coordinator node. The consecutive time slots are separated by small period of time, called guard period. The guard period is used to ensure non-overlapping transmissions among the stations. Two stations with synchronization time less than the guard time may successfully communicate without suffering any collision. The time slots are allocated among the contending nodes according to their traffic requirements, i.e., a node gets a time slot whenever it has some data to send. Since TDMA is a scheduling-based MAC scheme, nodes may turn off their radios during idle times to conserve energy. Proper functioning of this MAC scheme requires all the nodes to be synchronized in time.

On the contrary to CSMA/CA, TDMA permits several users to share a channel by dividing the time into discrete time slots. It saves the unnecessary overhead of contention for gaining access to the shared medium. Based on the TDMA schedule generated, each node gets a particular share of non-overlapping time to transmit and thus TDMA based MAC protocols are collision-free. The main task in generating a TDMA schedule is to allocate non-overlapping time slots to each station depending on the topology, packet generation rates of a node, traffic priorities, etc. TDMA-based MAC protocol enhances WiLD network performance by facilitating simultaneous transmissions without any interference. These advantages make TDMA based MAC protocols more suitable for multi-hop WiLD networks.

Now, we will discuss the existing TDMA-based MAC protocols proposed for WiLD networks. It starts with discussing some of the traditional TDMA-based MAC protocols which form the basis of the existing MAC protocols for WiLD networks. In the later part of this subsection, the existing TDMA-based MAC protocols for WiLD networks have been thoroughly examined and a comparative study on those protocols is presented.

Many TDMA-based MAC protocols for WiLD networks are proposed recently covering different scenarios such as considered network architecture, TDMA frame generation and dissemination process, etc. However, SoftMAC [?], MadMAC [?], FreeMAC [?], and Overlay MAC Layer (OML) [?] are found to be providing the initial platform for the development of TDMA-based MAC protocols for WiLD networks.

SoftMAC [?] developed a software system that allows researchers to use inexpensive, commodity wireless network cards to experiment, easily construct and deploy experimental dynamic MAC layers on Linux systems. MadMAC [?] extended the idea of SoftMAC and implemented a single-hop TDMA system between two nodes with tight time synchronization. Several design challenges were addressed to maintain the persistent slot structure and continuous packet transmission. FreeMAC [?] leveraged the methodology adopted in SoftMAC and implemented a single-hop TDMA system with strict timing requirements and provided a multi-channel approach. Overlay MAC Layer (OML) [?] is designed on the top of the 802.11 MAC layer using Click Modular Router [?] combining the power of changing the MAC layer and the ease of modifying the higher layers. It focuses on TDMA slot allocation according to a Weighted Fair Queuing (WFQ) policy to improve the fairness of 802.11. OML uses loosely synchronized clocks to divide the time into equal sized slots and then employs a distributed algorithm to allocate these slots among the competing nodes.

In the remaining part of this subsection, we discuss some of the important TDMA-based MAC protocols which specifically address the issues of WiLD networks.

2P

2P protocol [?] was the first to propose a TDMA-based MAC scheme for WiLD networks. This protocol relies on using off-the-shelf 802.11 hardware so that such low cost solutions can be affordable for rural areas. It considers the use of multi-radio operation in a single tower and demonstrates a simultaneous Synchronous Operation (SynOp) of Transmit (Tx) and Receive (Rx) in bipartite topology. A single channel is assumed to be used by all the radios leaving the other independent channels for local use. 2P operates by switching mode of each node between the two phases Synchronous Receive (SynRx) and Synchronous Transmit (SynTx). When a node switches from SynRx to SynTx, all of its neighbours have to switch from SynTx to SynRx, and vice versa. It keeps a transmission link active in either of the directions all the time. To initiate this switching activity, the protocol uses a special synchronization packet called marker packet which acts as token. A node possessing the marker packet can only transmit for a given duration of time. When the allotted time is over, the maker packet is passed from one end of a WiLD link to the other, so that at any instant of time exactly one end of the link is in transmitting and the other is in receiving mode. In this process, the nodes of a WiLD network get loosely synchronized. When there is no data from the IP layer, 2P sends dummy filler bytes instead just to maintain synchronization between the nodes.

WiLDNet

WiLDNet [?] is basically built upon 2P [?]. It highlighted some of the limitations of 2P and proposed some additional changes in order to further improve link utilization and to make the system more robust in handling high and variable packet loss in WiLD links. WiLDNet uses an adaptive loss recovery mechanism that uses a combination of Forward Error Correction (FEC) and bulk acknowledgment to reduce loss rate and hence increase end-to-end throughput. The bulk acknowledgement mechanism is implemented by replacing the standard 802.11 Stop-and-Wait protocol by sliding window based flow control. This protocol argues that due to ripple effect of marker packet loss in multi-hop WiLD network, link utilization in 2P is significantly affected. They proposed an implicit loose time synchronization mechanism to solve this problem. WiLDnet features application-based parameter configuration such as time slot period, FEC, number of retries, etc. All the modifications proposed in this scheme have been implemented as a *shim layer* above the driver using Click Modular Router [?].

JazzyMac

JazzyMac [?] is a distributed MAC protocol which aims to provide some level of QoS to the real-time applications by employing a dynamic time slot allocation scheme. Through variable length transmission slot allocation, each node can adapt the length of their transmission slots in accordance with their changing traffic demands. The protocol is specifically designed to allow neighbour nodes to proceed with parallel independent transmissions which contribute to enhanced throughput. To achieve single hop synchronization, it uses tokens similar to the marker packet used by 2P. In the line of 2P [?] and WiLDNet [?], JazzyMac also assumes single channel operation. The interesting part of this scheduling protocol is that it does not require the topology to be bipartite as in the case of 2P. Hence, this protocol is applicable to any arbitrary topology and each node can use purely local information for slot adaptation.

In case of unidirectional flows, use of dynamic time slot may not provide any specific advantage over networks. In such cases, the amount of traffic forwarded by the previous hop will also need equal time in the subsequent hops to forward them successfully towards the destination node. However, the use of variable length slots may provide advantages in some localized regions of the network.

JaldiMAC

JaldiMAC [?] is a TDMA-based MAC protocol designed for WiLD networks employing single hop point-to-multipoint transmission links. This protocol basically provides support for differentiated service classes to provision QoS. It allows dynamic traffic patterns with varying symmetry ratios to adapt with the asymmetry of Internet traffic. JaldiMAC proposes a centralized ply-based packet scheduling algorithm. The algorithm broadly classifies traffic into two different categories: latency sensitive class and bandwidth class which correspond to delay and bandwidth sensitive traffic respectively. The central node first schedules the traffic belonging to latency sensitive class meeting their maximum delay bounds one after another in the gaps left by any previous schedules. Once the scheduling of latency sensitive traffic is over, the remaining gaps are assigned to traffic of bandwidth class. JaldiMAC guarantees per-session fairness, provides loose QoS guarantees for latency sensitive traffic without compromising fairness. This protocol also uses bulk acknowledgement mechanism to increase efficiency of the network. Further, it handles error correction to hide packet loss from overlaying TCP traffic.

JaldiMAC is implemented in two layers. The first layer defines the high-level behavior of the JaldiMAC protocol which is referred to as JaldiTDMA. This layer is responsible for tasks such as building frame headers, calculating the TDMA schedule, error control, and station addressing. The second, i.e., the lower layer is responsible for configuring the chip hardware and physical layer settings as well as providing an interface for the higher layer to inject packets over the air. JaldiTDMA is implemented using the Click Modular Router in user space, while Jaldi9k is a Linux kernel module. For each packet transmission, a switch from user space to kernel space and vice-versa is needed which adds some overhead to the system.

LiT MAC

LiT MAC [?] is TDMA-based MAC protocol which maintains μs level time synchronization among the nodes in the network over multiple hops. It also incorporates spatial reuse and dynamic routing to improve network performance. This centralized light-weight TDMA-based MAC protocol is implemented over 802.15.4 based platform by using soft-state mechanisms to maintain schedule-state, network-state and flow-state. The scheduling in LiT MAC employs a centralized scheme which is responsible for allocating slots to a flow meeting the end-to-end delay bound for real-time traffic. The scheduler first finds the path between the source and destination nodes. The path is computed by shortest-hop metric over the network connectivity graph. Admission of new flow is taken on the basis of the interference graph and the data schedule for the ongoing flows. A centralized routing mechanism is also employed in LiT MAC. A node sends its neighbourhood information to the root as a part of node join request or topology update message. The routing module at the root uses those neighbourhood information to decide the node's forwarding tables. To improve network efficiency in the presence of wireless channel errors, this scheme enables hop-by-hop acknowledgement for control slots.

LiT MAC performance has been evaluated over 802.11 based multi-hop networks in [?]. Foundation of this work is reported in [?] which implements a simple TDMA MAC. LiT MAC integrates a stability based routing scheme for wireless mesh networks. A node wanting to start a new data flow, conveys its request to the root using the contention slots. The root, after getting the request, registers the flow and allocates time slot in the TDMA schedule for it. The centralized scheduler calculates the number of TDMA slots for each node which is computed as the total number slots available in the TDMA frame divided by the number of active nodes available in the network.

Multi-channel MAC

The MAC protocol proposed by Dutta et al. [?] provides a new channel assignment mechanism for WiLD networks. This scheme lifts the SynOp restriction [?] of 2P by using multiple channels[1]. Thereby, it enables continuous full-duplex data transfer on every link in the network. The use of multiple channels eliminates cross-link interference and thus do not require tight synchronization among the nodes. Considering any link in the network as made up of two direct edges, the assignment mechanism assigns non-interfering IEEE 802.11 channels in such a way that the set of channels assigned to the outgoing links is disjoint from set of channels assigned to incoming links of a node.

After discussing the state of art TDMA-based MAC protocols for WiLD networks, a comparison of the key protocols has been presented in Table ??.

[1]We call this protocol as Multi-channel MAC

Table 2.5: Comparison of TDMA-based MAC Protocols for WiLD Networks

Criteria	2P [?]	WiLDNet [?]	JazzyMac [?]	JaldiMAC [?]	LiT MAC [?]	Multi-channel MAC [?]
Implemented at	MAC Layer	Click Modular Router	MAC Layer	Click Modular Router	MAC and Routing Layer	MAC Layer
Single Channel	Yes	Yes	Yes	Yes	No (Multi)	No (Multi)
Time Synchronization	Loose	Loose	Loose	Loose	Tight	Loose
Multi-hop Time Synchronization	No	No	No	No	Yes	N/A
Dynamic Size Transmission Slot	No	No	Yes	Yes	No	N/A
TDMA Slot Size Allocation	Static	Dynamic	Dynamic	Dynamic	Static	N/A
Addressing QoS Issues	No	Partial	Partial	Yes	Partial	No

2.4.1.2 Dynamic Bandwidth Allocation Schemes in Multi-hop WiLD Networks

In multi-hop WiLD network architecture, the bandwidth of a link towards the gateway node is shared by the children links of the node. Due the SynOp constraint, a node can either receive from all of its neighbours or transmit to all of them simultaneously. However, if the nodes are allowed to transmit in their full capacity, congestion is bound to occur at the intermediate nodes. To avoid congestion, children nodes cannot be allowed to transmit in their full capacity even if they can do so. As a result, with the increase in the number of levels of the tree topology, the nodes available towards the bottom of the tree are restricted with very limited transmission bandwidth. In dynamic traffic situation, a node with sufficient real-time traffic in hand may not get adequate transmission opportunity whereas some other may not have any data to transmit or continue transmitting best-effort traffic. Schemes providing static transmission opportunity to all the nodes do not map well in dynamic traffic conditions. Therefore, the TDMA slots need to be dynamically allocated based on the QoS demands of the nodes. The architecture of multi-hop WiLD network resembles with WSN. Gateway-based multi-hop WiLD network is used as backbone network and hence traffic is expected to keep the links saturated most of the time. On the other hand, conventional Wireless Sensor Networks (WSN) carry sensory information to the sink node in a periodic fashion. Moreover, the traffic flow is mostly unidirectional in WSN whereas multi-hop WiLD network carry traffic in both the directions. Few relevant dynamic bandwidth allocation schemes available in the literature are discussed in this subsection.

Queue-MAC

Queue-MAC [?] is a hybrid CSMA/TDMA MAC protocol proposed for multi-hop WSN. This protocol addresses the issue of burst network traffic maintaining a limited duty cycle. It uses a customised superframe structure composed of beacon frame, variable TDMA period, fixed CSMA period and inactive period. The parent node acquires the load of each children from the packets received from them. The packets carry the load information through a special field called queue indicators. In normal or light load, transmission is carried out during the fixed CSMA period of the superframe. As the traffic load increases, the active period is accordingly extended by adding more TDMA slots to increase the bandwidth. This scheme combines the strengths of CSMA and TDMA while offsetting their weaknesses.

Funneling-MAC

Funneling-MAC [?] is a localised, sink-oriented hybrid MAC protocol. It recognises that the traffic density towards the sink node is more and as a result of that overall network performance degrades. This fact is termed as funneling effect. To address funneling effect problem, this protocol proposes a hybrid of CSMA/CA and TDMA protocol. A localized TDMA is used in the funneling region and CSMA/CA protocol is implemented for the rest of the network. The local TDMA controlled by the sink node provides additional transmission opportunity to the nodes closer to it. The sink node monitors data traffic and keeps track of incoming traffic rate for each path. The traffic rate is calculated by monitoring the number of incoming packets in one superframe per path. To maintain the traffic rate information, the sink node uses a path table. Based on the rate of monitored traffic in the table, the sink node allocates time slots to each path. If the traffic rate of a path is k and the number of hops of the path is h, the sink allocates $k \times h$ slots to the path. But, if the traffic rate of a path is less than 1, the sink allocates $1 \times h$ slots to the path.

TreeMAC

In supporting high data rate real-time applications in WSN, TreeMAC [?] proposes a localized TDMA MAC protocol. Here, every node gets a number of time slots proportional to its bandwidth demand. At the time of network initialization, every node runs CSMA to discover its neighbours. After neighbour discovery, a

TDMA approach is adopted to schedule nodes transmission. TreeMAC divides each TDMA cycle into frames and each frame into three slots. Once the children of the sink node are discovered, it assigns schedules to them. Using the parent-children relationship of tree topology, the frame and slot assignment decision between a parent and its children nodes is taken by themselves. Keeping its own required slots, each parent node distributes the remaining frames among its children. This frame assignment is carried out based on relative bandwidth demands of its children. Each node individually calculates slot assignment according to its hop distance to the sink node.

I-MAC

In [?], a hybrid MAC scheme called I-MAC is proposed. I-MAC protocol assigns different levels of priority to different nodes thereby improving the channel utilization. To reduce energy consumption, shorter back-off periods are given to higher priority nodes. It reduces the amount of packet collision as only the nodes having the same priority compete to access the channel. I-MAC is composed of two phases – *a set-up phase* and *a transmission phase*. During the set-up phase neighbour discovery, slot assignment, local framing, and global synchronization is done. The transmission phase use three levels of priority and the priority of each node is set according to its role within the network and to its traffic load. All nodes are allowed to transmit during any slot. However, the owner of the slot gets the first priority. If the owner has no data to transmit, non-owners can compete to use the slot. The chance of getting a slot by a non-owner node depends on its priority level. Thus I-MAC allows sensor nodes that have higher load (higher priority) to get more chances to access the radio resources. Moreover, the lifetime of loaded sensors is prolonged since the prioritization mechanism reduces collision and shortens their listening period.

Utilization Based Scheduling (UBS)

UBS [?] is a utilisation based distributed dynamic scheduling scheme proposed for WMN. UBS assigns a weight to each node which is dynamically adjusted according to the slot uses history and packet queue occupancy of the nodes. The protocol works in distributed manner where each node adjusts its own weight and makes pseudo-random transmission attempts using only the locally available information. It divides the time into several equal intervals called frames. Each node periodically runs the dynamic weight adjustment algorithm. Based on the current weight of a node which shows its demand for transmission slots in the next frame, the dynamic weight adjustment algorithm assigns a dynamic weight value to it. The number of time slots assigned to each node in a single frame is proportional to its weight.

i-Queue-MAC

To deal with burst traffic in data collecting tree-based sensor networks, i-Queue-MAC [?] proposes a hybrid MAC combining CSMA/CA and TDMA protocols. iQueue-MAC runs CSMA in light load and TDMA in high load situation. In high load, it uses the senders queue length to dynamically allocate time slots to them for packet transmission. The whole network is divided into two types of nodes- simple nodes and routers. Each simple node is associated with a router. A router is responsible for collecting the packets of its simple nodes. iQueue-MAC uses the queue length of each node and allocates suitable TDMA transmission slots accordingly. In light load situation, no TDMA slot is allocated. In high load traffic, iQueue-MAC senses the build up of packet queues and dynamically schedules adequate number of slots for packet transmission.

A consolidated comparison of the dynamic bandwidth allocation schemes discussed above is presented in Table **??**.

2.4.1.3 Packet Level Scheduling in TDMA-based Networks

Applications with stringent QoS requirements demand a fine granular QoS support. Link level scheduling provides transmission opportunities to nodes but cannot provide packet level QoS. Traffic scheduling at MAC layer is an efficient scheme to support QoS. Its importance is even intensified in multi-hop WiLD networks using half duplex wireless links. Quite a few numbers of packet scheduling schemes for TDMA are found in the literature. Few relevant schemes are discussed below-

QoS Provisioning for Multi-Service TDMA Mesh Networks

Djukic [?] solves an optimization problem to minimize the number of TDMA slots required for guaranteed QoS traffic in mesh network. It assumes that queueing delay is minimized in the network layer. The optimization technique works as

Table 2.6: Comparison of Dynamic Bandwidth Allocation Schemes

	Funneling-MAC [?]	**Queue-MAC** [?]	**Tree-MAC** [?]	**I-MAC** [?]	**UBS** [?]	**iQueue-MAC** [?]
MAC Protocol	Hybrid (CSMA/CA and TDMA)	Hybrid (CSMA/CA and TDMA)	TDMA	Hybrid (CSMA/CA and TDMA)	S-TDMA	Hybrid (CSMA/CA and TDMA)
multi-hop Consideration	Yes	Yes	Yes	Yes	Yes	Yes
QoS Support	No	No	Yes	No	No	No
Network Topology	Tree	Tree	Tree	Tree	Mesh	Tree
Channel Consideration	Single channel	Multi-channel	Single channel	Single channel	Single channel	Multi-channel
System Initialization	Sink starts beaconing	Parent nodes broadcast beacon to divide time	Runs CSMA to discover its neighbours	Nodes run neighbour discovery operation	node's state detection	Nodes transmit beacon
Adaptive Traffic Handling	Sink assign slots to path from traffic rate	Allocating TDMA slots dynamically	Parent allocates children's frame on demand	Nodes are assigned priority according to their roles	Dynamic slot size based on nodes slot uses history and packet queue occupancy	Variable TDMA period
Traffic Demand Notification	Sink monitors incoming data packet	Queue indicator for node's load	Piggybacks in upstream packets	According to the role of the node, contention window is varied	Weight information in HELLO message	Queue length

follows. First, it performs a linear search for the number of TDMA slots required to support end-to-end QoS. Second, the optimization solves a $\{0, 1\}$-integer program [?] at each iteration of the search. The optimization stops as soon as the number of guaranteed slots with a feasible transmission schedule is found. Third, it finds an order of transmissions in the frame so that the maximum TDMA propagation delay is kept below a threshold level and end-to-end bandwidth is also met. Finally, the order of transmissions is distributed to the nodes so that they can find the transmission schedule using the Bellman-Ford algorithm [?].

A Real-Time Traffic Packet Scheduler

To provisioning QoS for real-time audio and video, [?] presents a packet scheduler for TDMA-based MAC protocol. The packet scheduler works in three phases - *Traffic Classification*, *Packet Selection*, and *MAC Layer Protocol Adjustment*. In traffic classification phase, the scheduler identifies the delay sensitive traffic through the information received directly from the packet bits or by keeping track of the packets and by comparing the observed flow profile. In packet selection phase, the decision whether the packet is to be transmitted or dropped is taken. Those packets whose delay is already higher than the acceptable delay of $150ms$ are dropped, and those packets whose delay is within the acceptable delay limits are transmitted according to their descending experienced delay. In MAC layer protocol adjustment phase, the various parameters of the MAC protocol are adjusted to match the properties of the flow that is being scheduled. The time period, t is set to a value such that it can accommodate $2k$ number of voice packets, where k is an integer. The maximum waiting time of a packet, T is also set. If T is set to a large number the cumulative delay guarantees could not be met and if T is set to a small number, it leads to under-utilization of the resources. Hence the value of T needs to be set to an intermediate value. N corresponds to the session length that makes the value of $T \times N$ equal to the length of the training period for the flow categorizer.

JaldiMAC

JaldiMAC [?] proposes a TDMA-based MAC protocol for point-to-multipoint WiLD networks which adapts to the asymmetry of Internet traffic and provides loose QoS guarantee for latency sensitive traffic without compromising fairness. It supports differentiated service classes. The algorithm classifies traffic into two main categories: *Latency sensitive class* and *Bandwidth class*. *Latency sensitive class* corresponds to delay sensitive traffic which attempts to minimize delay and jitter. *Bandwidth class* corresponds to bandwidth greedy traffic which are not concerned about delay and jitter. The proposed scheme is based on a centralized ply-based packet scheduling algorithm. Ply is a virtual layout of large number of unassigned slots. The scheduler assigns slots of the first ply to the first traffic class that arrives. The second traffic class is placed in the second ply, and the process continues until all the slots in the ply are assigned. In this way, the centralized node gives priority to latency sensitive class traffic over bandwidth class traffic. JaldiMAC uses the available bandwidth more efficiently by taking actual traffic needs into consideration. Each station request bandwidth anticipating its traffic need. Since the acknowledgement of each and every packet sent reduces the efficiency of WiLD network, JaldiMAC employs a bulk acknowledgement scheme using cumulative acknowledgement.

An Integrated Scheduling and Routing Approach

In order to provide QoS support for different traffic classes in WiLD networks, an integrated QoS routing and traffic scheduling scheme is proposed in [?]. It presents a QoS routing algorithm called Maximum Allocation with Reservation-based (MAR) QoS Dynamic Source Routing (MQDSR) which is designed to work over 2P [?] MAC protocol. Dynamic Source Routing (DSR) based QoS route discovery mechanism is used to find a route that satisfies end-to-end QoS requirement of a flow. On receipt of route reply message, every intermediate node reserves bandwidth for the requested flow. To address the traffic scheduling issue at MAC layer, it defines a *service index* for traffic with different QoS requirements and proposes a traffic scheduling algorithm based on the service index. The traffic scheduling algorithm works by dividing the traffic into three classes- high (traffic which are most sensitive to delay), normal (traffic with relaxed delay bounds) and best-effort (traffic which has no limitation on delay). It calculates the service index of each of the traffic classes but gives descending order of priority to high, normal, and best-effort traffic class. Service index is defined as a set (I, B) where I is the service interval and B is the number of packets served during the time I. The value of I is given as $\left\lceil \frac{D}{T_r} \right\rceil$. The value of B is calculated using the expression $\left\lceil R \times T_r \times \frac{I}{P_d} \right\rceil$ where R is the data rate of real-time traffic, D is the maximum tolerable delay, T_r is the round trip time (RTT) of the packets, and P_d is the packet size excluding the header.

Real-time Flow Scheduling (RFS)

Chipara et al. [?] proposes a flow-based scheduling scheme called Real-time Flow Scheduling (RFS) for WSN. RFS supports spatial reuse through a interference-aware transmission scheduling. It is designed for point-to-point real-time flows with arbitrary inter flow interference. RFS divides the problem of real-time flow scheduling into two parts *single flow in isolation* where RFS constructs plans according to which all instances of a flow are delivered and *multiple concurrent flows* where RFS's dynamic scheduler executes multiple flows concurrently based on the previously constructed plans. The method works as follows. First, a source node starts the creation of a new flow by checking whether an existing path can be used for the flow or not. If possible, it uses the existing path for the new flow. If not, the planner initiates the construction of a plan for the new flow. Second, admission control is performed to check if the new flow may be added without other flows missing their deadlines. Then finally the scheduler executes flows based on the plans.

DelayCheck

Gabale et al. [?] addresses the issue of maximizing the number of voice calls supported by capacity-constrained WiLD networks with minimum delay guarantees. DelayCheck is an online centralized scheduling and Call-Admission-Control (CAC) algorithm which effectively schedules constant-bit-rate voice traffic in TDMA-based mesh networks. The protocol discovers route, allocates channels, schedules link for a flow in delay-constrained manner. The DelayCheck algorithm works in three phases – *constructing the auxiliary graph*, *allocating resources*, and *running the allocated resources in polynomial time*. In constructing the auxiliary graph phase, an auxiliary graph G' is constructed from the original graph G. Every path between source and destination on such an auxiliary graph gives a feasible schedule. In allocating resource phase, DelayCheck uses Dijkstras algorithm to find the shortest path in G', which gives a feasible routing path in the original graph and finds a feasible schedule in delay-constrained manner. In the third phase, the path goes through a filtering phase to make sure that the schedule is conflict-free for the given interference graph.

A brief comparison of the above discussed protocols are presented in Table ??.

Table 2.7: Comparison of TDMA-based Packet Scheduling Schemes

	Slot Level Scheduling [?]	**Real-time Packet Scheduler** [?]	**JaldiMAC** [?]	[?]	**RFS** [?]	**DelayCheck** [?]
Flow/ Packet level QoS	Packet	Packet	Packet	Flow	Flow	Flow
Designed for	Mesh	Mesh	WiLD Mesh	WiLD Mesh	WSN	Mesh
Network Topology	Tree	Mesh	Mesh (Point-to-Multipoint)	Mesh (Point-to-Point)	Tree	Mesh
Channels considered	Single	Single	Single	Single	Single	Multi
Traffic Types	-	2	2	3	-	-
Admission Control	No	No	No	Yes	Yes	Yes
Routing Integration	No	No	No	Yes	No	Yes

2.4.2 QoS at Network Layer

Enhancing the MAC layer with provisions for QoS does not alone placate the scenario of QoS in multi-hop WiLD networks. An effective routing protocol is crucial for adapting real-time traffic flows to the dynamic characteristics of multi-hop WiLD networks. QoS support can be enforced in the network layer through network load balancing, admission control, and topology aware routing. Establishing

QoS-aware path directly facilitates in provisioning of end-to-end QoS. In gateway-based WiLD mesh networks, most of the communications happen through the gateway node. Hence, single path based protocols are likely to end up with congestion within a short span time. Multi-path routing protocols are often used for provisioning QoS in various other networks. Use of multiple paths for transmission facilitates load balancing and can provide better end-to-end throughput and delay performance. Therefore, we focus on multi-path routing schemes pertaining to their QoS support. The important multi-path routing protocols are discussed in the following subsection.

2.4.2.1 Multi-path QoS-aware Routing for Multi-hop WiLD Networks

A large number of QoS routing schemes proposed for WMN in the literature are extension of the basic Mobile Ad-hoc Network (MANET) routing schemes. MANET routing protocols can be divided into reactive (on-demand), proactive (table-driven), and hybrid. Reactive or on-demand routing protocols create a route between a pair of source and destination nodes when the source node actually needs to transmit packets to a destination node. Examples of reactive routing protocols are AODV [?], DSR [?], MCR [?], LBAR [?], etc.

Reactive routing in WMNs and MANETs is different in many respects. In MANET, as there are frequent link breaks caused by the mobility of nodes, flooding-based route discovery provides high network connectivity and relatively low message overhead compared to proactive routing protocols. However, in WMNs, links normally have much longer expected lifetimes due to relatively static nature of nodes. Since the frequency of link failure is much lower in WMNs, flooding-based route discovery is both redundant and very expensive in terms of control message overhead. Therefore, the existing reactive routing protocols are generally not scalable or appropriate for mesh networks [?]. In proactive routing protocols, each node maintains one or more routing tables containing information about every other node in the network. All the nodes periodically update their tables in order to maintain consistent and up-to-date information about the network. When network topology changes, the nodes propagate update messages to all the incumbent nodes of the network. After receiving the routing update information, the nodes modify their routing table contents. These routing protocols differ in the method by which packets are forwarded along routes. Examples of such routing protocol are DSDV [?], WAR [?], OLSR [?], etc. There are two types of proactive routing protocols viz., source routing and hop-by-hop routing. In source routing, the source node discovers route for a flow and puts the entire path of the flow in the packet header. Intermediate nodes only need to relay packets based on the paths given in the packet header. In hop-by-hop routing, every node maintains a routing table that indicates the next hops for the routes to all other nodes in the network. For a packet to reach its destination, it only needs to carry the destination address. Intermediate nodes forward the packet along its path based on the destination address only. Hybrid routing protocols use the features of both proactive and reactive routing techniques. In hybrid routing, the first phase of the routing protocol starts with a proactive approach and then serves the demands from additionally activated nodes through reactive flooding. The reactive approach reduces the control overhead and proactive approach decreases the latency during route discovery process. Routing protocols such as LSRP [?], and ZRP [?] are some of the hybrid type routing protocols.

MANET routing protocols primarily focus on finding an optimal path between a source and destination pair. In WMN, outgoing traffic flow towards the gateway nodes and the incoming traffic moves through the gateway to the end-user nodes. If multiple nodes choose the same optimal path for communication, some intermediate nodes may get congested with packets. Thus, normal routing schemes proposed for MANET are not suitable for WMN. Packets are required to be distributed to balance the network load. Addressing the reliability issue and improving network performance as a result, multi-path routing policy has been extensively investigated in the literature [?, ?, ?, ?, ?, ?]. Although multiple routes are discovered by these protocols, only one path is used to forward the packets at a time. The other discovered paths are used only on failure of the primary path. The problem of congestion in optimal paths still persists. Hence, such multi-path routing protocols cannot be useful in provisioning QoS except improving reliability. As a result, many routing protocols have been proposed to meet QoS challenges focusing on the issues such as minimum bandwidth, end-to-end delay, packet error, and jitter.

The multi-path routes can also be employed for simultaneous data transmission through different paths. The current multi-path routes can be grouped into three categories [?]- (i) Disjoint route: The disjoint routes can be classified into node-disjoint and link-disjoint routes. The paths discovered by node-disjoint multi-path protocol do not share any common node. In link-disjoint multi-path route, the paths discovered may share any node but no links are common. (ii) Maximally disjoint route: The concept is similar to disjoint route with a relaxation that the paths may share a small number of nodes or links. (iii) Hybrid path: the multi-path route can be combination of both (i) and (ii).

A routing protocol computes optimal path based on some routing metric. A routing metric is essentially a value assigned to each route or path, and is used by a routing algorithm to select a subset of routes discovered by the routing protocol. The main objectives of using routing metric are to minimize delay, maximize probability of data delivery, maximize path throughput, maximize network throughput, equal traffic distribution, etc. Various routing metrics such as distance, hop count, delay, expected transmission time (ETT), expected transmission count (ETX), etc., are used in different routing protocols. Customized routing metrics also sometimes used in specific situations. End-to-end path bandwidth and path delay are commonly considered as routing metric in QoS-aware multi-path routing.

Various multi-path routing protocols proposed in the WMN literature are discussed in the following subsection. In supporting QoS, the routing protocols are needed to discover paths which can satisfy certain QoS requirements and other constraints. Therefore, while discussing the multi-path routing protocols, we have focused on their ability to support QoS.

Ad-hoc On-demand Multi-path Distance Vector (AOMDV)

AOMDV [?] is an extension of single path Ad-hoc On-demand Distance Vector (AODV) protocol [?] which was basically proposed for MANET. In case of route failure, AODV needs to discover alternative paths by re-initiating the route discovery process. Due to node mobility, frequency of route failure in MANET is relatively high. Hence, frequent route discovery is needed. To reduce the route discovery frequency, AOMDV discovers and maintains multiple routes for every route discovery. In doing so, AODV makes some new entry to the AODV routing tables, and a new route list table is used to store additional information for each alternate paths including: next hop, last hop, hop count, and expiration timeout. While discovering multiple paths, loop freedom property of AODV is preserved. When multiple alternative paths are available, a new route discovery is needed only when all those paths fail. The route establishment procedure is the same as AODV with an exception that, to form multiple routes, all duplicates of the route request arriving at a node are accepted as each one defines an alternate route. Multiple paths generated by this protocol are guaranteed to be disjoint of alternate paths. This protocol considers both node and link disjointness. The notion of disjointness is limited to one pair of nodes and does not consider disjointness across different node pairs. This disjointness is ensured by distributed computing without using source routing. To preserve connectivity information, each node executing AOMDV can use link-layer feedback or periodic HELLO messages to detect broken links from a node to its immediate neighbours. A node generates or forwards a route error message for a destination when the last path to the destination breaks. In AOMDV, packets forwarded over failed links are re-forwarded over alternate paths

Multi-path Dynamic Source Routing (MP-DSR)

MP-DSR [?] is a DSR-based multi-path routing protocol which addresses the path reliability requirements through the use of multiple QoS-paths. After collecting path reliability requirements of applications, the protocol determines the number of paths needed and the lowest path reliability requirement each path must provide. The protocol computes a set of unicast routes which can satisfy a minimum end-to-end reliability requirement; it then maintains this requirement throughout the life time of transmission. A new routing metric called *end-to-end reliability* is defined which is used to reflect the probability of successful data transmission from the source to the destination node within a given time frame. The path reliability is dependent on the link availabilities of all the links along a path. Link availability is defined as the probability that a link should be available for a given time period. Path reliability is the product of link availabilities of all the independent links of a path.

After learning the number of paths and lowest path reliability required, the source node runs route discovery algorithm to find QoS paths. In the route discovery process, the source node sends Route Request (RREQ) messages to the destination node via its immediate neighbours. The RREQ message contains information regarding reliability requirements, the traversed path, and the accumulated path reliability. After beginning the route discovery process, the destination node collects multiple RREQ messages through various paths. From the path fields stored inside these messages, the destination node uses a path selection algorithm to select the set of disjoint paths that can provide the required end-to-end reliability. Once the destination node completes executing the path selection algorithm, it replies to the source node about the result of its selection by a set of Route Reply (RREP) messages. When all routes fail or when one of the used paths fails, the route maintenance procedure is initiated. On failure of a single route, the protocol examines whether route discovery process should be triggered.

But, in case of all routes failure, the route establishment process is re-initiated immediately.

MP-DSR provides soft QoS guarantees with respect to end-to-end reliability by discovering a set of multiple disjoint paths and transmitting data along these paths. In order to meet the QoS requirements, traffic are distributed along multiple paths if required.

Split Multi-path Routing (SMR)

SMR [?] is a on-demand routing scheme which employs source routing approach. To utilize the network resources efficiently by avoiding congested router from the paths, this algorithm establishes and maintains multiple paths between each source and destination pair. In order to find paths to a given destination, the source node broadcasts a RREQ packet. The RREQ packet contains the source ID and a sequence number that uniquely identifies the packet. When an intermediate node receives a RREQ, if the packet is not a duplicate one, it appends its ID and re-broadcasts the packet. The information of all the nodes along the path followed by a RREQ packet is recorded in it. Unlike other source based routing, the intermediate nodes are not allowed to send RREPs back to the source even when they have route information to the destination. After receiving all the RREQ packets, the destination node calculates two maximally disjoint paths satisfying the given criteria and informs the same to the source node.

On detection of link failure, a Route Error (RERR) packet is sent to the upstream direction of the root to notify those nodes that the link is broken. They can then delete the entry from their routing tables. If only one of the two routes of the session is invalidated, the source uses the remaining valid route to deliver data packets. On link failure, a new route discovery procedure is triggered. The protocol splits the traffic into two available routes using a simple per-packet allocation scheme. However, in this scheme packet may be delivered out of order. Re-sequencing of packet is required at the end of the mesh network.

Multi-path Mesh (MMESH)

Nandiraju et al. in [?] criticizes the existing MANET routing algorithms as multiple routers find the same path as the optimal path to a gateway thus leading to a possibility of deadlock in the chosen path. To address this issue, a multi-path routing protocol called MMESH is proposed in this paper. MMESH effectively discovers multiple paths by redefining the traditional RREQ and RREP mechanisms. The route discovery and route maintenance procedures are divided into two phases: initial network setup phase and maintenance phase. The gateway nodes initiate the network setup phase by broadcasting advertisements of Internet connectivity. The nodes receiving the gateway broadcast set up paths to all the possible gateways. These nodes announce all of their available routes to the gateways in the order of their preference further by broadcasting *connection advertisement* packets. After receiving all the advertisements, a node selects a set of acceptable paths to it and registers with the parent nodes. This procedure is repeated till all the nodes join the network. Each mesh router is given the responsibility to continuously monitor the performance of all the active paths through it. In the event of path status change, a mesh router performs necessary changes in the corresponding routing tables and propagates this information to all the nodes of available in the path via neighbour nodes. This protocol further proposes a load balancing approach to distribute the traffic equally among all the available paths in round robin fashion. Given a set of paths available to a particular destination, a source nodes sends every packet to a different next hop thus uniformly distributing traffic over in the network.

Multi-path Routing Protocol (MRP)

Delay requirements of delay-sensitive traffic over WMNs can effectively be met by multi-path routing protocols. Considering this fact, MRP [?] addresses the problem of traffic optimization over multiple paths to enhance network performance. Formulating the multi-path routing problem using convex optimization methods, this algorithm derives a multi-path protocol through an optimization decomposition. Proposing a interference model for WMNs, MRP defines an interference-aware multiple path selection metric. To explore multiple paths to the destination, a RREQ packet similar to that in AODV protocol is broadcast by the source node. The destination node may receive multiple route requests from different routes. The destination node selects θ-routes with less cumulative metric values. RREP packet is sent to the source along each reverse route.

DAWMNet

DAWMNet [?] is a hybrid multi-path routing algorithm proposed for industrial WMNs which integrates the enhanced Dijkstra's algorithm [?] and Ant Colony Optimization (ACO) algorithm [?, ?]. DAWMNet works in two phases. In the first phase, shortest route is setup from the gateway to each end node by adopting enhanced Dijkstra's algorithm. The enhanced Dijkstra's algorithm uses distance as a routing metric. In the second phase, multiple routes are explored using the chosen route based on ACO algorithm by diffusing pheromone packets. Pheromone diffusion aims to spread the available pheromone information over the network by using periodic "keep-alive" messages. Maintenance of route is carried out through pheromone updating of ACO technique. Pheromone updating aims to control and update pheromone information through route sampling by using ants. In addition to neighbour and graph table, it maintains a table called pheromone table which records the information about the routes to the gateway including the intermediate nodes. On detection of any link failure, a node informs the network manager which in turn informs all other nodes. The route maintenance scheme can respond effectively to some topological changes in a timely manner.

QoS Multi-path Optimized Link State Routing (QoS-MOLSR)

QoS-MOLSR [?] is a OLSR-based multi-path routing protocol designed to support delay-bound real-time multimedia applications over MANET. Multiple paths are established using end-to-end delay as a metric. Just like OLSR, each node sends out HELLO and Topology Control (TC) messages periodically to discover the network topology. After that route is computed using an enhanced Dijktra's algorithm. Dijktra's algorithm is normally used in finding shortest path between a given pair of nodes. However, use of such paths for multimedia transmission may cause network congestion leading to an increment for end-to-end delay. To guarantee QoS for multimedia communication, an improved M-Dijkstra algorithm is proposed in this protocol to calculate multiple disjoint paths with minimum delay from source to other nodes. QoS-MOLSR appends delay information to extend the HELLO messages for enabling routing. When an intermediate node receives a TC message with cumulative delay, it checks whether it has received it for the first time or not. If yes, it creates a new record and save it, otherwise it simply updates its record. When a given delay-bound route is required to be established to a particular destination, the source node can decide the route by using the local information of cumulative delay queue.

Table 2.8: Comparison of Multi-path Routing Protocols

	AOMDV [?]	**MP-DSR** [?]	**SMR** [?]	**MMESH** [?]	**MRP** [?]	**DAWMNet** [?]	**QoS-MOLSR** [?]
Type of Protocol	Reactive	Reactive	Reactive	Hybrid	Hybrid	Hybrid	Hybrid
Multi-gateway Consideration	No	No	No	Yes	No	No	No
Routing Metric	Hop Count	SDR, COR, Error-Ratio	Hop Count	ETT and WCETT	Cumulative Delay	Hop-Count and Distance	End-to-end Delay
Route Discovery Scheme	AODV-based	DSR-based	DSR-based	Gateway-based	AODV-based	Dijkstra Algorithm	Modified-Dijkstra Algorithm
Route Maintenance Scheme	Yes	Yes	Yes	Periodic Neighbour Link Monitoring	No	ACO Algorithm	No
Periodic Message Exchange	Yes	No	No	Yes	Yes	Yes	Yes
QoS Consideration	No	End-to-end Reliability	No	No	Delay	No	Delay
Path Disjointness	No	Yes	Yes	No	No	No	Yes
Bandwidth Aggregation	No	No	No	Yes	No	No	No
Admission Control	No	No	No	No	No	No	No
Interference Model	No	No	No	No	Interference free schedule	No	No

SDR: Success Delivery Rate; COR: Control Overhead Ratio
ETT: Expected Transmission Time; WCETT: Weighted Cumulative Expected Transmission Time
ACO: Ant Colony Optimization

A point-wise comparison of the above discussed protocols are presented in Table ??.

2.5 Conclusion

In this chapter, we have briefly discussed the architecture of multi-hop WiLD networks and different approaches employed in QoS provisioning over such networks. The feasibility of using the standard CSMA/CA and TDMA protocol in multi-hop WiLD networks has been investigated. It has been observed that the TDMA-based protocols perform better than CSMA/CA in WiLD environment. Provisioning of QoS for real-time applications mostly target the MAC and routing layer. In resource-constrained WiLD networks, the MAC protocol should utilize the bandwidth optimally through the scheduling of parallel non-overlapping link transmissions. Packet congestion around the root node poses certain challenges in achieving end-to-end QoS. A QoS-aware dynamic bandwidth allocation scheme can contribute immensely in achieving some level of guaranteed QoS. Giving early and more transmission opportunities to real-time flows through packet scheduling can prioritize certain real-time applications over the others. Similarly, the routing layer can also contribute in enhancing service quality by selecting QoS-aware paths.

With a detailed understanding of the state of the art, the research contributions are presented in the subsequent chapters.