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Load Balancing of Video Traffic in Wireless Mesh Networks with Xorencoding

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ABSTRACT: Wireless mesh networks are rapidly evolving. Wireless mesh networks plays a vital support in people's life worldwide. The capacity of wireless mesh networks to integrate with other networks made it so popular. But it fails to guarantee the quality of service. As the technology advances, the utility increases widely thereby the quality degrades. This paper proposes an approach called XOR Encoding which will helpful in increasing the throughput of WMN with the help of ViL Bas which is selective video load balancing solution for delivering videos in WMN's. ViL Bas identifies the network congestion and keep a track of it and reroute traffic around congested node. RTOC is an application-independent architecture that takes into consideration the characteristics of real-time traffic and provides an efficient framework to optimize multimedia application such as bandwidth, delay, delay variation, and loss. The performance of RTOC was evaluated using simulated video streaming traffic. The results demonstrated that RTOC can improve the real-time performance metrics such as the packet delivery ratio, end-to-end delay, jitter, and throughput by considerable margins. This will help high quality video on demand services in case of broadcasting. The application of this technique in includes Cable TV, IP TV etc....

KEYWORDS: WMN,RTOC,QoE,XOR Encoding

I.INTRODUCTION

Wireless Mesh Network is an emerging technology which connects the world. Wireless Mesh Network is comprised of radio nodes. As the name indicates the nodes are arranged in mesh topology. Mesh cloud is also made up of radio nodes, but when all the radio nodes work together as a single network it is called so. Wireless Mesh Network can be either to work as a centralized network or decentralized network. Wireless Mesh Network is used in US military forces, satellite constellation, public safety, city wide wireless coverage etc., the important constraint of Wireless Mesh Network is shared bandwidth and interference, number of nodes and their location. The topology of Wireless Mesh Network remains stable. The main desirable options of Wireless Mesh Network are reliability, redundancy, rapid deployment, cost effective, resilient and extensible. Dynamic protocols (ad-hoc on demand distance vector, Babel etc.,) or auto configuration protocols (proactive protocol, ad-hoc configuration protocol) can be used. In addition to user data it is imperative to protect the control data (routing, monitoring, etc.).If the control data is unprotected, it will be relatively easy for an attacker to disable a WMN. The principle of Wireless Mesh Network is similar to wired network. The application of Wireless Mesh Network includes broadband internet access, indoor WLAN coverage and mobile access connectivity. In spite of the advantages Wireless Mesh Network also faces challenges such as uneven distribution of traffic, degraded quality of service, channel assignment problem.
To overcome the issue of load balancing ViLBas was introduced. It is a selective load balancing algorithm for delivering videos. This prevents packet drops. Rerouting is carried out at each congested node. Since the network is arranged in mesh topology, the data can reach the destination through an alternate path until the congested node is not the destination node. Rerouting at all the nodes will result in overloading and hence the most congested node is chosen and load is balanced selectively. The congested node is identified with the help of queue occupancy maintained at each nodes of network. Threshold is maintained at each queue, once when a threshold is reached the notification of congestion is sent and rerouting is carried out. To maintain the bandwidth smart-streaming was included which tracks the segment time and delivers at a constant rate with the help of round trip time. VilBas[1] Faced problem of reduced network throughput which is resolved by XOR encoding technique. It is a paradigm in which intermediate nodes are

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allowed to create new packets by combining (XOR ing) the incoming packets which provides the possibility to maximize network throughput and reduce number of transmissions.

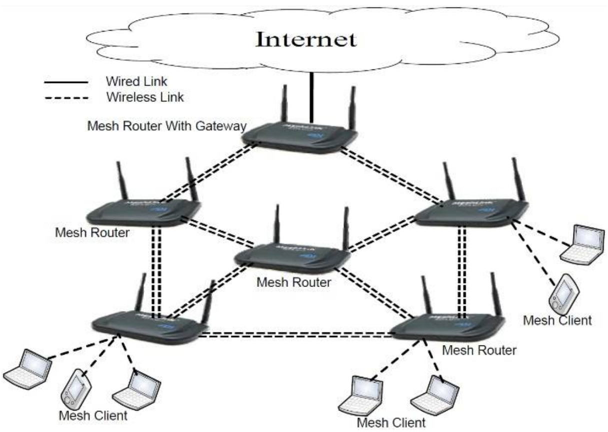

Fig 1:Wireless Mesh Networks

This Paper is standardized as Section II VILBAS Algorithm, Section III explains about existing System, Section IV We will analyze the results of simulations and Section V Conclusion

II.VILBAS ALGORITHM

VILBAS load balancing mechanism avoided the congestion of data packets but it faced problem of overhead and delay. To overcome this problem we integrated a new technique XOR encoding which reduce the data rate so that the data sized reduced.

A. *Load Balancing using VILBAS Algorithm*

VILBAS [3]employs distributed monitoring of network traffic, identifies the node most affected by congestion and prevents imminent packet drops by re-routing the video flows around the congested node.

1. The decision to re-route is taken per flow and is event triggered by a congested node in distributed manner.
2. The traffic is split into different classes, where delay sensitive applications have the highest priority.
3. Queue occupancy levels at each network interface are used to detect congestion Pro-actively.
4. Selective load-balancing is performed such as to increase user QoE levels.

The VILBAS algorithm has some demerits that it has a large delay and overhead. We integrate a new XOR based encoding and decoding which help us to overcome the problems.

III. PROPOSED SYSYTEM-XOR ENCODING OF VIDEO PACKETS

Network coding[3] for multimedia transmission over wireless networks can be improved by improving throughput, delay and bandwidth. A Good Encoding topology is necessary as in video transmissions as we always required an efficient usage of network bandwidth while transmitting video over leased lines etc. A Video stream which takes more bandwidth will consume more network space and thereby increase the complexity and cost for video transmission. XOR[4] operations have been widely used to implement network coding. The reason is that the complexity of XOR-based network coding in wireless networks is relatively low. RTOC(Real-Time Opportunistic Coding), a new XOR-based opportunistic network coding architecture for real-time data transmission. Unlike the existing network coding approaches for multimedia transmission over wireless networks, RTOC[6] is an application-independent architecture

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that takes into consideration the characteristics of real-time traffic and hence offers an efficient framework for applying various techniques to satisfy different multimedia application requirements such as bandwidth, delay, delay variation, and loss.

A. Real-Time Opportunistic Coding

1. RTOC Design Principles

The first system architecture for wireless network coding was presented by COPE (Coding Opportunistically). The two main principles of COPE are opportunistic listening and opportunistic coding. With opportunistic listening, as the medium is broadcasting the nodes will overhear video packets broadcasted in network and store the packets for a short time. With opportunistic coding, the node looks for opportunities to encode maximum number of packets, ensuring that the recipients can decode their packets. This can be illustrated by the example in Fig. 2. In this example, node B has 4 packets in its output queue P1, P2, P3 and P4. Each packet needs to be forwarded to one of node B's neighbours (A, C, and D). Based on its local information (i.e.; overheard or exchanged information), node B knows which packets each neighbour has.

When node B is ready to transmit a packet, it consider first packet from its packet queue and starts the coding operation (i.e.; searching for other packets to be encoded with the selected packet).Node B has multiple coding options, as shown in Fig. 2. In Case 1, node B can XOR P1 with P2 and broadcast the result. In this case, node C can reconstruct P2 since it already has P1 and needs P2, but node A cannot reconstruct any of them since it has neither P1 nor P2 in its buffer. In Case 2, node B can XOR P1 with P3. In this case, nodes A and C can decode their original packets, which is better than Case 1. However, the best coding decision is Case 3, where node B can XOR P1, P3, and P4. In this case, nodes A, C, and D can reconstruct their original packets.

Even though RTOC adopts COPE's basic principles of opportunistic listening and opportunistic coding, the two systems are primarily different. COPE was designed for regular non-real-time traffic and hence the reliability of data transfer was granted higher priority than the timely delivery of data. Whereas RTOC, being specifically designed for real-time traffic, gives higher priority to the packet delivery time and inter-packet delay by minimizing the overhead associated with the

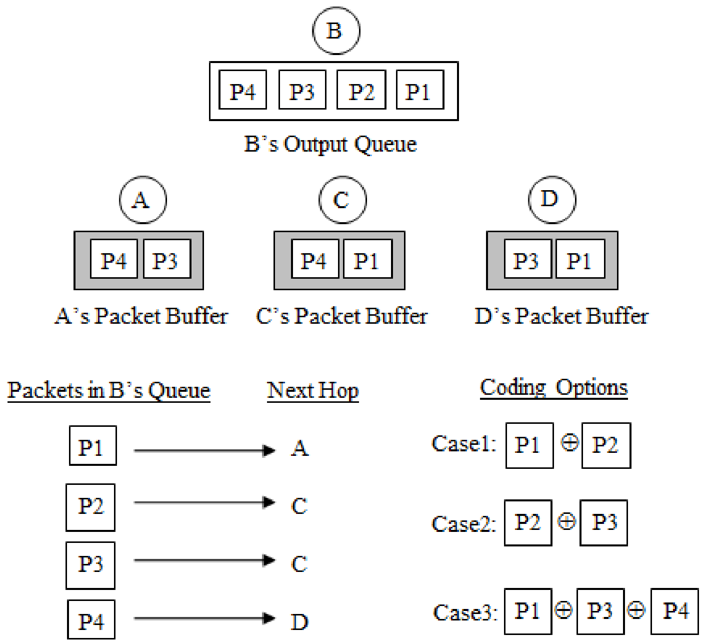

Fig. 2 XOR Mechanism

encoding and decoding processes, as explained .Thus, the design principles used in RTOC cope with the performance characteristics required for real-time transmission.

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2. RTOC Packet Coding Algorithm

The whole system will works based on an algorithm. RPCA uses one virtual FIFO queue for each neighbor node, whereeach queue contains the packets destined to that neighbor node. When a node is ready to senda packet *p* to the next hop node *N*, it picks the head of the output queue *Q* (*N*) and for each of neighbor node *i*, it checks (using the stored information about what each neighbor node has inits repository) whether encoding *p* with the head of the virtual queue *Q* (*i*), *p*, would benefitany of the neighbor nodes. If it does, then *p*is added to the list of native packets to beencoded. The final encoded packet is then transmitted over the wireless channel. On the other hand, if no encoding is found useful, then *p* is sent without encoding. The more packets are encoded, the less the total number of transmissions and hence the less the number of packet collisions and transmission overhead. If the packets to be encoded have different sizes, smaller packets are padded with zeros. This way, no searching for the same-length packets necessary and no packet reordering would occur, which minimizes the packet encoding time and makes it suitable for real-time traffic. RPCA is shown below.

Pick packet p at the head of the output queue

Natives={p}
Nexthops={nexthop(p)}
For Neighbour i=1 to N do
Pick Packet pi,the head of virtual queue Q(i)
If ᵘn €Nexthops ∪ {i},(n can decode p ⊕ pi) then
p= p ⊕ pi
Natives=Natives ∪{ pi}
Nexthops= Nexthops ∪ {i}
end if
end for
return p

IV. RESULT AND DISCUSSION

Simulations can be done using network simulator-2, we give input as MPEG-4 video traces which will be simulated in NS-2 environment and output parameters are monitored using Xgraph tool. The mobile Ad hoc network scenario consist of 50 nodes and we use AODV protocol first and then include VILBAS algorithm after that we will go for XOR algorithm.The improvements in three main parameters are considered while considering XORVILBAS.

A. Parameters under Considerations

Packet loss:The PDR depends mainly on the transmission mode used (i.e.; whether reliable or unreliable) and the packet error rate caused by packet collisions. Packet collisions are caused by simultaneous transmissions on the communication channel by two or more nodes contending for channel access. Hence, as the number of active nodes on the channel increases, the probability of packet collision increases. Therefore, the use of network coding has a two-fold effect on the PDR. That is, as network coding combines more than one packet per transmission, the overall average number of transmissions per node decreases, which in turn reduces both the contention for channel access and the probability of packet collision. On the other hand, each packet collision that involves an encoded packet results in losing more than one original packet, especially if an unreliable transmission mode is used. If we can obtain an optimum PDR value it will increase the clarity of video and reduce freezing of video.

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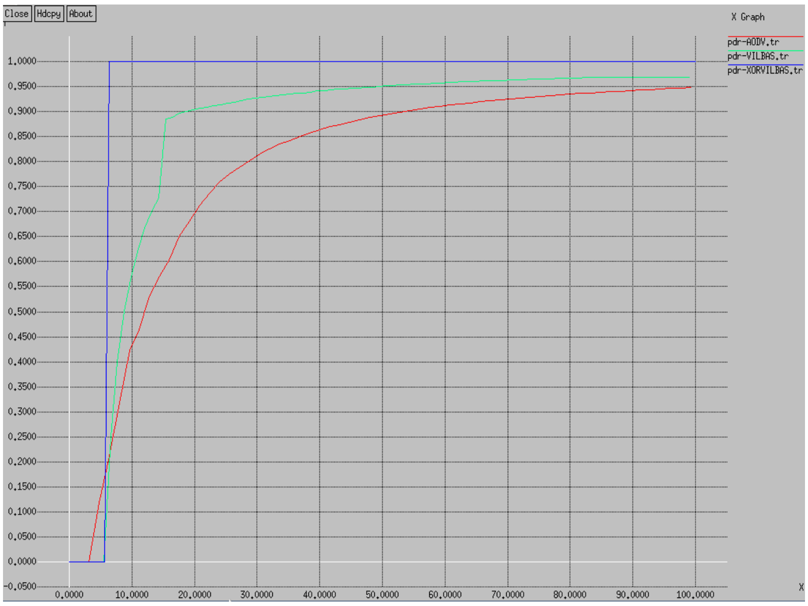

Fig. 3 PDR comparison with AODV, VILBAS and XORVILBAS

From the above figure 3 we can see that PDR is increased and reaches and optimum value .Blue line indicates PDR with XOR VILBAS

Throughput: When used in the context of communication networks, such as Ethernet or packet radio, throughput or network throughput is the rate of successful message delivery over a communication channel. The data these messages belong to may be delivered over a physical or logical link, or it can pass through a certain network node. Throughput is usually measured in bits per second (bit/s or bps), and sometimes in data packets per second (p/s or pps) or data packets per time slot. In case of video transmission networks we need to achieve maximum throughput for

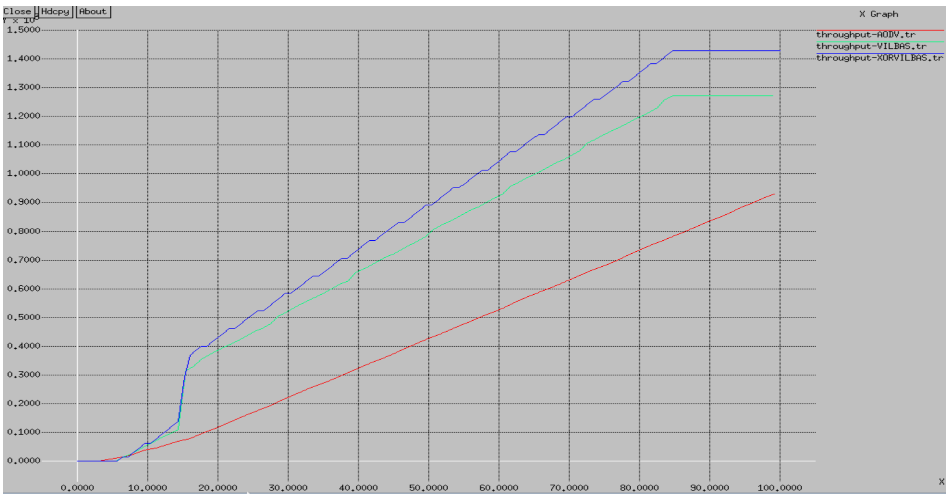

Fig. 4 Throughput comparison with AODV, VILBAS and XORVILBAS

efficient usage of channel. Below shows the throughput status comparing different situations. Blue line indicates throughput with XORVILBAS.

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Delay: Network delay is an important design and performance characteristic of a computer network or telecommunications network. The delay of a network specifies how long it takes for a bit of data to travel across the network from one node or endpoint to another. It is typically measured in multiples or fractions of seconds.

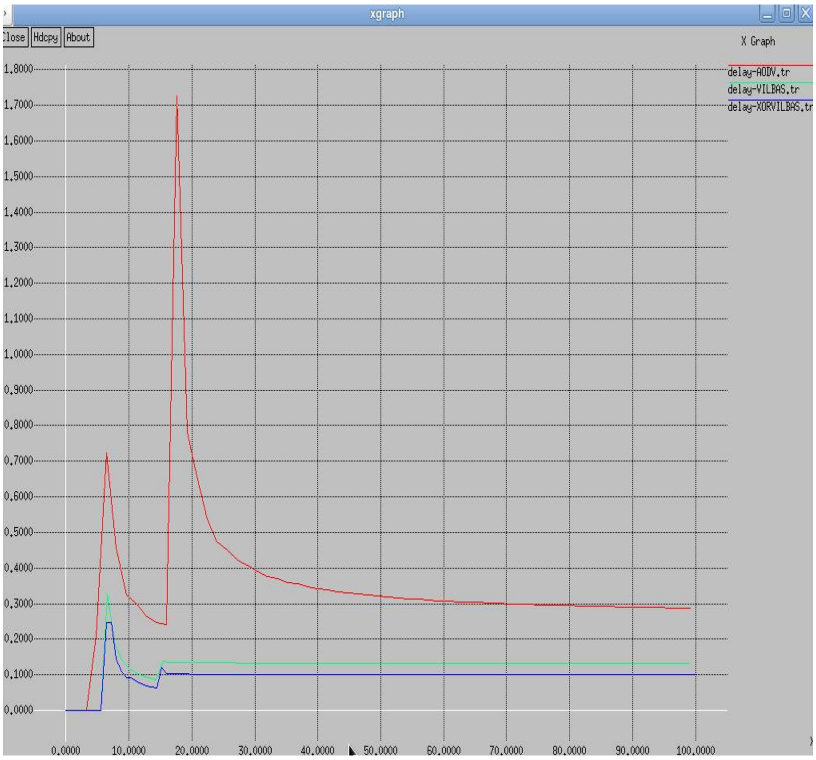

Fig. 5 Delay comparison with AODV, VILBAS and XORVILBAS

Delay may differ slightly, depending on the location of the specific pair of communicating nodes. The packet delivery time relies mainly on the traffic congestion and the overhead associated with the packet transmission over the communication channel. Therefore, asnetwork coding combine packets before transmission, the overall average transmission overhead per original packet is reduced.

VI. CONCLUSION

RTOC, a new XOR-based opportunistic network coding architecture that copes with the performance characteristics required for real-time video transmission. RTOC provides an efficient framework for applying various techniques to satisfy different multimedia application requirements such as bandwidth, delay, delay variation and loss. The simulation results demonstrated that RTOC has a good potential to improve the performance metrics of real-time multimedia transmission such as packet delivery ratio, end-to-end delay, jitter, and throughput. This technology will help maximum usage of network bandwidth in the help of video streaming applications. As a future work we can secure network by adding various security algorithms.

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