Ad Hoc Wi Fi

Table of contents

Introduction

Scheduling algorithm and scenario generator

Problem statement

Video & audio encoding

video streaming over networks

problems

types (download,full stream)

environments (multicast,unicast) ,servers/IP , wireless

protocols and methods

wireless protocols

collision avoidance, varying frequences

Omnet++ , MiXiM

802.11g enhancments

application layer contributions (flow control, buffer management)

project implementation architecture overview

- scenario generator & scheduler

- IniGen

- rtp dump,replay and video encoder

- simulation

results

- end2end delay , throughput , loss + drops, snr2per (presentation slides) , rtp statistics

references

appendix

- implementation class diagram

- file formats

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Introduction

The problem of transmitting real time video stream over ad-hoc wireless network poses a special challenge due to the demanding requirement of the data and the special network topology.

The requirement to satisfy for real time video streaming is large throughput of data and minimal and stable end-to-end delay, this characteristics of the data is required since transmission of video with audio requires large bandwidth even with modern encoders due to the growing demand for high quality (e.g High-Definition / Dolby Surround).

The transmission of real time video , for example for usage such as video conference, adds the requirement of minimal end-to-end delay which is required so that the transmission over network would be negligible to create the effect of real-time on both ends.

The medium of an ad-hoc wireless network is mostly challenging due to the fact that though these networks are more common today they still have relatively low bandwidth and transmission distance compared to wide-area networks or cellular base-stations, this requires special handling for multi-hop transmission and rate-control.

The work in this project is part of a research group designed to handle such scenario, this document described the simulator and the flow control which are part of the overall solution to the shown problem.

Network topology

We consider a problem of multiple nodes carrying each a single transceiver able to either listen in one of several known frequencies (channels) or transmit in one channel to neighboring nodes in a known protocol.

In the above network we consider a set of demands for video-streaming which is composed by several triplets which are the sending node (camera node), destination node and the requested rate.

Two nodes transmitting in the same channel interfere with each other and can cause error in packet receive, we can consider two models for such interference:

Graph model: TODO

Real model:

The system should be able to satisfy the requests and to deliver to the destination node what has been send from the source with minimal delay.

Wireless protocol

We consider in this case the wireless protocol 802.11g, this protocol allows a node to transmit packets in one of 8 modulation coding scheme (MCS), each such MCS has different characteristics in terms of the transmission rate as well as signal-to-noise ratio (SNR) to packet-error-rate (PER), as SNR with fixed transmission power becomes a function of distance this leads to different transmission distance for different MCS.

The 802.11g IEEE protocol is designed for Carrier-Sense-Multiple-Access with Collision-Avoidance (CSMA/CA) networks where each each node “listens” to the channel prior to attempting to send a packet such that it doesn’t interfere with existing usage of the medium, if the medium is “busy” then a random delay (commonly exponentially growing) is used until a next attempt to transmit is done.

In order to avoid problems caused by “hidden node’s” which are outside the range of one of the two nodes which desire to communicate special control packets have been added:

* Request-To-Send (RTS) control packet is transmitted by the source whenever it desires to send a new frame
* Clear-To-Send (CTS) packet is being transmitted by the destination node if it isn’t busy and it senses that the channel is not busy
* When the source receives the CTS packet it sends the payload packet to the destination
* When the destination has completed receiving the payload packet it sends to the source an acknowledge (ACK) packet

The above protocol , using a simple global (not per neighbor) state machine is able to safely perform communication in such network of un-synchronized and non-collaborating nodes and can avoid deadlock and collision in the network.

This protocol however causes a severe degradation on actual performance compared to the theoretical rate enabled by the coding, moreover this protocol doesn’t deploy any priority on message transmission and thus doesn’t guarantees fairness or any quality-of-service on both throughput or end-to-end delay.

Scheduling algorithm

In order to accommodate the requirement a scheduling algorithms has been developed by the research group, this algorithm accepts as input the network graph and a set requests {ri}

uests  {ri}k

i=1

Video coding