Performance Evaluation of *WiFiRe* using OPNET

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Goal

- Building WiFiRe model in OPNET.
- Finding minimum slot length to support VoIP and Video services.
- Finding maximum number of users system can support.
- Analyzing queuing delay, end to end MAC delay and throughput for different types of flows with different slot lengths.

WiFiRe overview

Protocol Phases

- Network entry and initialization
    - Ranging
    - Registration
- Connection management and Data transport
    - Connection creation using DSA and DSC
    - Connection release using DSD

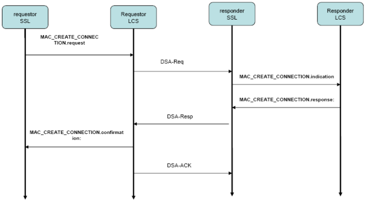


Figure: Connection Setup

MAC Services

- UGS: Designed to support real-time flows, such as VoIP.
- rtPS: Designed to support real-time flows that generate variable size data packets on a periodic basis, such as MPEG video.
- nrtPS: Designed to support non real-time flows those require variable size data grant slots on a regular basis, such as high bandwidth FTP.
- Best Effort: The intent of the Best Effort (BE) service is to provide efficient service to best effort traffic. These flows served in contention slots.

WiFiRe model block diagram

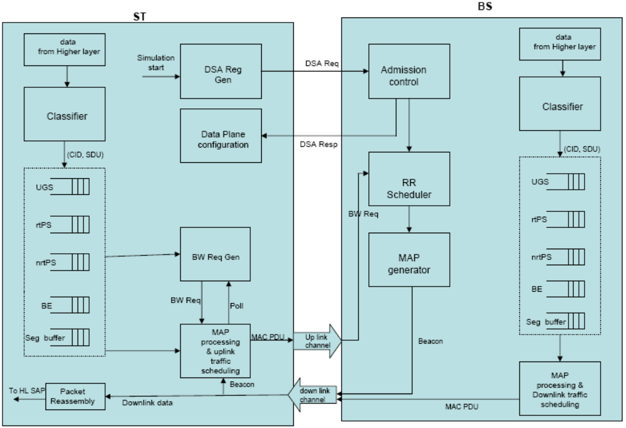


Figure: WiFiRe model block diagram

WiFiRe model modules

- Classifier
- DSA Req Generator
- Admission Control
- Round Robin slot Scheduler
- MAP processing and traffic scheduling
- Packet segmentation and Reassembly

Process Models

- Common MAC process model
- BS control child process
- ST control child process

Common MAC process functionalities

- Packet classification
- constructing MAC frames
- segmentation and reassembly .
- Beacon processing

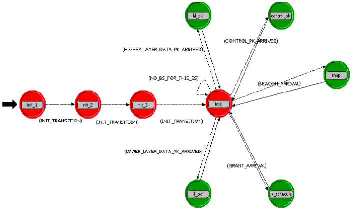


Figure: Common MAC process

BS control child process functionalities

- Admission control
- Scheduling admitted connections and MAP generation
- Polling and Bandwidth grant
- Activation of admitted service flows

ST control child process functionalities

- DSA-Req generation
- DSA-Resp process

Classifier

- Set of matching criteria's
- Protocol specific criteria, ex: destination IP, Source IP etc.
- Finds CID using service class name and destination MAC address.

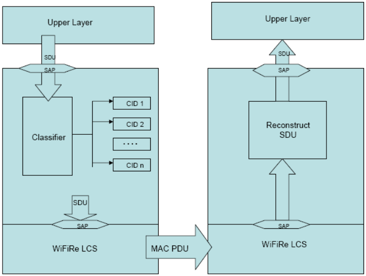


Figure: IP classifier

Classifier setup

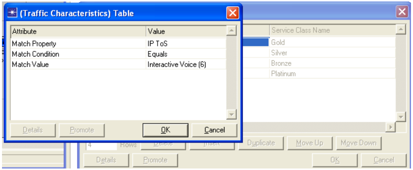


Figure: IP classifier setup in opnet WiFiRe model

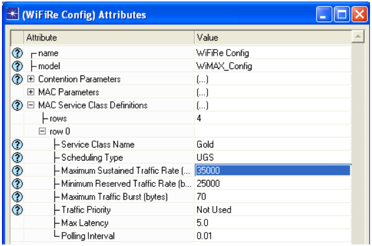


Figure: UGS parameters

Connection Admission Control

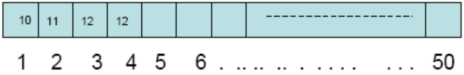
- Admit UGS with maximum sustained traffic rate
- Admit rtPS and nrtPS with minimum reserved rate.
- If it is rtPS or nrtPS flow
  Then Add Polling overhead.
- Add WiFiRe MAC overhead and PHY overhead to requested rate.
- If available bandwidth is more than requested bandwidth, then BS will admit the flow and sends response.

Round Robin slot scheduling

- Allocate Polling slots to rtPS and nrtPS flows
- Allocate slots to UGS flows.
- For rtPS and nrtPS
  - If requested bandwidth is less than admitted then
  - Allocate requested number of slots
  - otherwise, allocate admitted number of slots
- Allocate remaining slots to BE flows
- Mark remaining slots as contention slots

Beacon processing and Traffic Scheduling

- Scan through DL-MAP and UL-MAP
- If CID in particular slot is my CID, then
- extract one segment from connection segment buffer, if there are any pending packets in segmentation buffer.
- otherwise, extract new packet from connection packet buffer.
- Then Construct MAC PDU and schedule it for transmission.



Goals

- Building WiFiRe model in OPNET.
- Finding minimum slot length to support VoIP and Video services.
- Finding maximum number of users system can support.
- Analyzing queuing delay, end to end MAC delay and throughput for different types of flows with different slot lengths.

S.No	Parameter	value
1	Frequency Channel	2.4 GHz
2	Bandwidth	22 MHz
3	Data Rate	11 Mbps
4	Frame Duration	10 ms
5	Symbols per frame	220000
6	Symbol duration	.045μs
7	Scheduling Algorithm	Round Robin

Table: Common parameters for all experiments

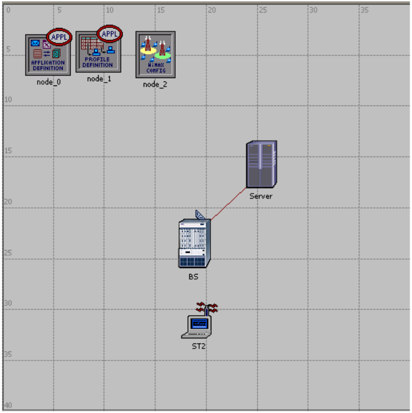
VoIP user Scenario

UGS Service flow parameters

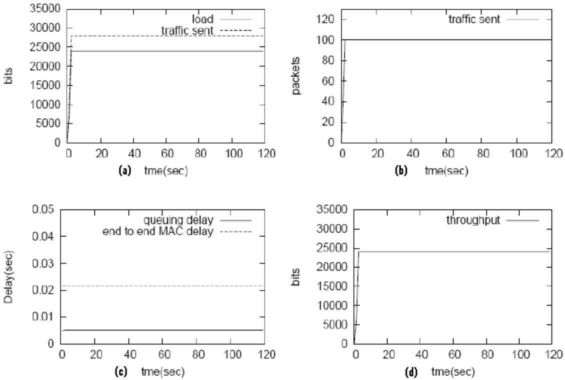
S.No	Parameter	value
1	Service Class Name	UGS
2	Maximum traffic rate	24 Kbps
3	Minimum traffic rate	24 Kbps
4	Max latency	4 seconds
5	Polling Interval	NA

Table: UGS service flow parameters

Simulation Setup

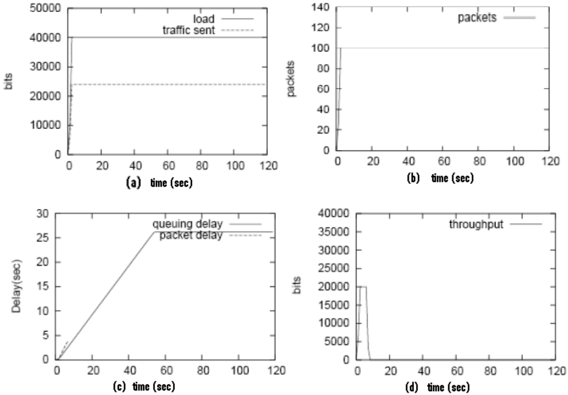


UGS statistics with slot size 32 μs and VoIP application with G 729 codec, 20 ms sampling rate and 8 kbps coding rate.



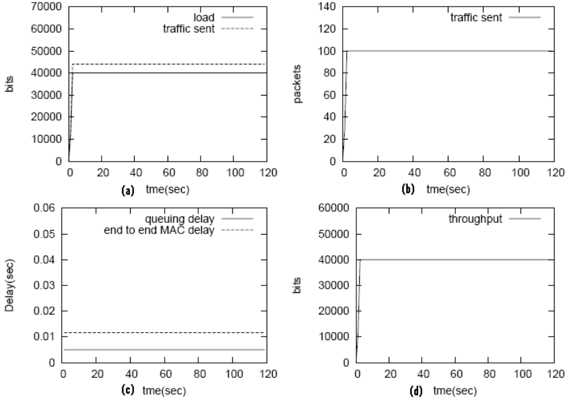
- With above VoIP configuration, load at MAC layer is 24kbps.
- VoIP SDU at MAC layer is 60 bytes with above VoIP configuration.
- This 60 bytes payload is segmented into two packets and served in two frames.
- So service rate is equal to arrival rate, Hence queuing delay and end to end MAC delay are constant shown in Figure (c).
- From this experiment, we observed that, one 32μs slot is suitable for VoIP applications with above configuration.

UGS statistics with slot length 32μs and VoIP application with G729 codec, sampling rate 10 ms and 8 kbps coding rate.



- With above VoIP configuration, load at MAC layer is 40kbps.
- VoIP SDU at MAC layer is 50 bytes, so source MAC will segment it and sends in two frames.
- So ST is sending 50(payload) + 10(MAC header) in 20 ms (two frames), means ST is sending at 24kbps.
- service rate is less than arrival rate, So queuing delay and end to end MAC delay started increasing shown in Figure (c).
- After some time, end to end MAC delay exceeds max latency for all packets, so destination MAC will drop all those packets. So throughput is dropped to zero shown in Figure (d).
- From this experiment, we observed that, one 32μs slot is not suitable for voip applications with above configuration.

UGS statistics with slot size 40 μs and VoIP application with G729 codec, 10 ms sampling rate and 8 kbps coding rate.



- With above VoIP configuration, load at MAC layer is 40kbps.
- In one 40μs slot, ST can send 55bytes in one frame, means ST can send at 44kbps shown in Figure (a)
- So service rate is equal to arrival rate, So queuing delay and end to end MAC delay are constant shown in Figure (c).
- From this experiment, we observed that, one 40μs slot is suitable for voip applications with above configuration.

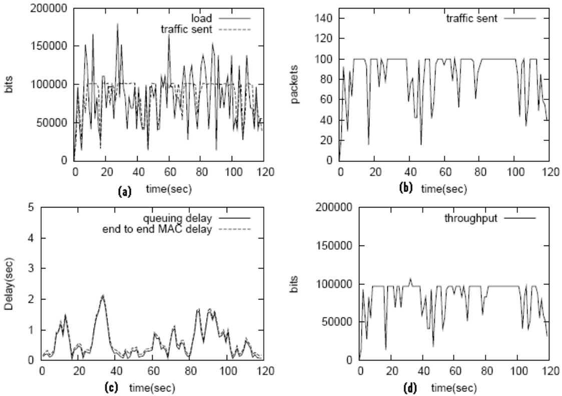
Video user Scenario

rtPS Service flow parameters

S.No	Parameter	value
1	Service Class Name	rtPS
2	Max Sustained traffic rate	100 Kbps
3	Min reserved traffic rate	90 Kbps
4	Max latency	8 Secs
5	Polling Interval	80 msecs

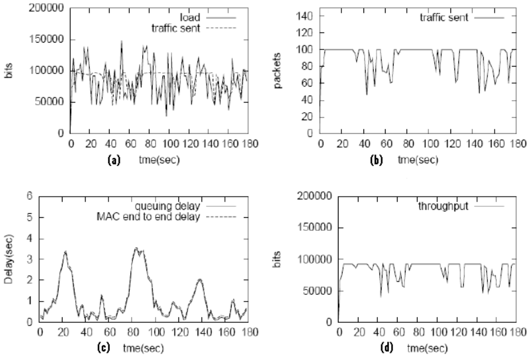
Table: rtPS service flow parameters

rtPS statistics with slot size 32μs.



- ST can send $((32 \times 10^{-6}) \times (11 \times 10^{6})) \times 3$ bits in one frame with three 32μs data slots.
- So ST can send traffic at $= \frac{((32 \times 10^{-6}) \times (11 \times 10^{6})) \times 3}{10 \times 10^{-3}} = 105$ kbps. This can be observed in Figure (a)
- Queuing delay and end to end MAC delay is not exceeding maximum latency 8 seconds shown in Figure (c).
- So no packets are dropped by destination MAC, Hence throughput equal to load shown in Figure(d).
- From This experiment, we observed that, three 32μs slots are required to support Video applications with above configurations.

rtPS statistics with slot size 45μs.



rtPS statistics with slot size 45μs.

- Figure (c)shows that, queuing delay and end to end MAC delay is increased compared to previous experiment.
- With 45μs, 90 kbps needs two data slots.
- MAC can send $((45 \times 10^{-6}) \times (11 \times 10^{6})) \times 2$ bits in one frame with two data slots. Then traffic sent per second = $\frac{((45\times10^{-6})\times(11\times10^{6}))\times2}{10\times10^{-3}}$ = 99 kbps.
- In previous experiment, with three 32μs ST can send traffic at 105 kbps.
- So service rate in this experiment is less than previous experiment, hence delays are increased.
- Even though ST is requested 90 kbps, BS is allocating 99 kbps. Because BS will allocate bandwidth in terms of slots.

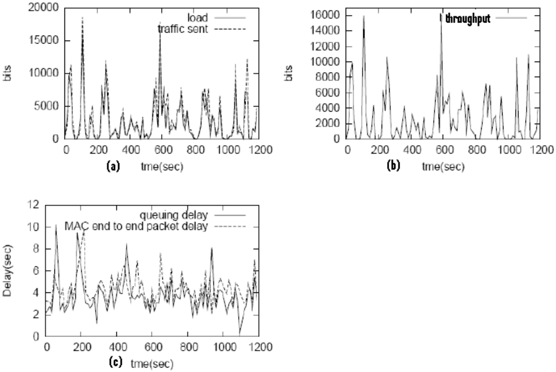
FTP user Scenario

nrtPS Service flow parameters

S.No	Parameter	value
1	Service Class Name	nrtPS
2	Max Sustained traffic rate	20 Kbps
3	Min reserved traffic rate	10 Kbps
4	Max latency	10 Sec
5	Polling Interval	2 sec

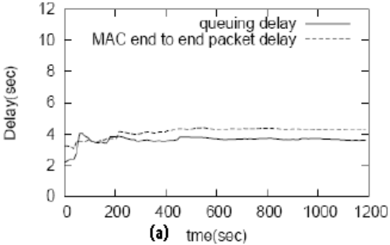
Table: nrtPS service flow parameters

nrtPS statistics with slot size 32 μs and with 2 seconds polling interval.



- ST can send $((32 \times 10^{-6}) \times (11 \times 10^{6}))$ bits in one frame with one 32μs slot.
- So ST can send traffic at = $\frac{((32\times10^{-6})\times(11\times10^{6}))}{10\times10^{-3}}$ = 35 kbps.
- Queuing delay and end to end MAC delay are greater than 2 seconds, because polling interval is 2 seconds shown in Figure (c).
- There are more variations in queuing delay and end to end MAC delay, because of more variations in load.
- end to end MAC delay is not exceeding maximum latency 10 seconds, So no packets dropped by destination MAC, So throughput is equal to load shown in Figure(b).
- From these results, it has been observed that, one 32μs slot is sufficient to support FTP applications.

nrtPS queuing and end to end MAC delay comparison with slot size 32μs for different polling intervals



Conclusion

- Extensive simulation results shown that 32 μs slot length is suitable to provide voice services with G729 codec, with 20 ms sampling rate and 8 kbps coding rate. However rtPS flows need more than one slot.
- 40μs length slot is required to support Voice applications with G729 codec, with 10 ms sampling rate and 8 kbps coding rate
- If slot size is small, then MAC overhead will be increased due to segmentation. If slot size is large, then the packet will be delivered quickly to the destination MAC. Suppose if slot size is equal to SDU size + mac_header_size, then source MAC will deliver entire MAC PDU in one slot. So no additional segmentation overhead is incurred.

Conclusion

- Increasing slot length will decrease the maximum number of admitted connections.
- The above simulation results also shows the model validness.
- We don't have any conclusive results for multiple STs.

Future extension

- Physical layer effects needs to be added to the model.
- Changing existing static DSA Req procedure such that, ST sends DSA Req when MAC receives first data packet belongs to some service type.
- Dynamic Service Deletion Req needs to be added to the model.
- With present slot scheduling algorithm, there are chances of bandwidth wastage, as bandwidth assignment is in terms of slots. As described above, even though if ST has requested less than one slot per frame, BS will still allocate one slot for it. We can save bandwidth by allocating bandwidth in symbols per frame, like $< CID, number\_of\_symbols\_allocated >$ instead of $< CID, number\_of\_slots >$.

Thank you !

Theoretical Maximum number of rtPS users:

- Requested rate with MAC overhead is,
  $rate \times (\frac{\text{avg packet size+ MAC header size}}{\text{avg packet size}})$
  $= 90 \text{ Kbps} \times \frac{1500+5}{1500} = 90.3 \text{ Kbps}$
- rate in symbols per second is
  $\text{rate} \times \frac{1}{\text{number of bits per symbol} \times \text{coding rate}}$
  $= 90.3 \text{ Kbps} \times \frac{1}{1 \times \frac{1}{2}} = 180.6 \times 10^3 \text{ sps}$
- Number of symbols required per frame = rate sps $\times$ frame duration
  $=180.6 \times 10^3 \times 10 \times 10^{-3} = 1800$
- Number of symbols per slot is, $\frac{\text{slot duration}}{\text{symbol duration}}$
  $= \frac{32}{.045} = 711$ symbols
- Number of symbols needed for WiFiRe PHY overhead (96μs) is,

- $\frac{\text{PHY overhead duration}}{\text{symbol duration}}$
  $= \frac{96}{.045}$ =2133 symbols
- Total number of symbols required is, sum of the PHY overhead and number of symbols requested
  = 2133+1800 = 3933
- Convert symbols per frame into number of slots per frame
- $\frac{\text{number symbols required per frame}}{\text{number of symbols per slot}}$
  $= \frac{3933}{711} = 6$
- Then Maximum number of connections per frame is
- $\frac{\text{total symbols per frame}}{\text{number of symbols requested}} \times$ number of parallel tx
- $\frac{220000}{3933} \times 2 = 111$

Theoretical Maximum number of UGS users:

- Requested rate with MAC overhead is,
  $rate \times \left(\frac{\text{avg packet size+ MAC header size}}{\text{avg packet size}}\right)$
  $= 24 \text{ Kbps} \times \frac{50+5}{50} = 26.4 \text{ Kbps}$
- rate in symbols per second is
  $\text{rate} \times \frac{1}{\text{number of bits per symbol} \times \text{coding rate}}$
  $= 26.4 \text{ Kbps} \times \frac{1}{1 \times \frac{1}{2}} = 52.8 \times 10^3 \text{ sps}$
- Number of symbols required per frame = rate sps $\times$ frame duration
  $=52.8 \times 10^3 \times 10 \times 10^{-3} = 528$
- Number of symbols per slot is, $\frac{\text{slot duration}}{\text{symbol duration}}$
  $= \frac{32}{.045} = 711$ symbols
- Number of symbols needed for WiFiRe PHY overhead (96μs) is,

- $\frac{\text{PHY overhead duration}}{\text{symbol duration}}$
  $= \frac{96}{.045}$ =2133 symbols
- Total number of symbols required is, sum of the PHY overhead and number of symbols requested
  = 2133+528 = 2661
- Convert symbols per frame into number of slots per frame
- $\frac{\text{number symbols required per frame}}{\text{number of symbols per slot}}$
  $= \frac{2661}{711}$ = 4, means one data slot and 3 PHY slots.
- Then Maximum number of connections per frame is
- $\frac{\text{total symbols per frame}}{\text{number of symbols requested}} \times$ number of parallel tx
- $\frac{220000}{2661} \times 2 = 165$