Solutions to Performance Problems in VOIP over 802.11 Wireless LAN

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Presented By
Syed Zaidi
Outline

- Introduction
- VOIP background
- Problems faced in 802.11
  - Low VOIP capacity in 802.11
  - Interference between VOIP and TCP
- Solutions
- Simulations
- Conclusion
- Observations
Introduction

VOIP in an 802.11 Network

[Diagram showing a network setup with APs, Voice Gateways, and S1, S2, ..., Sn connected to BSS 1 and BSS 2 through the Internet.]
VOIP Background

VOIP Call

User A

Invite

OK

Audio

User B

Signaling traffic
SIP

Audio/Media traffic
RTP

Audio
VOIP Background

RTP

- Real-time transport protocol.
- Built on UDP.
- Sequence Numbering.
- Time Stamping.
- Sent at a continues rate every 20ms.
Low system capacity in WLAN network for VOIP calls.

Interference between VOIP traffic and data traffic.
Too Much Overhead

The diagram illustrates the structure and overhead of IP, UDP, RTP, and payload packets. The overhead includes frame control, duration, addresses, sequence control, and a payload CRC. The diagram also shows the timing for channel busy, data, and next frame packets, with backoff and SIFS intervals.
Too Much Overhead

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>DIFS</td>
<td>50 µsec</td>
</tr>
<tr>
<td>SIFS</td>
<td>10 µsec</td>
</tr>
<tr>
<td>Slot Time</td>
<td>20 µsec</td>
</tr>
<tr>
<td>CWmin</td>
<td>32</td>
</tr>
<tr>
<td>CWmax</td>
<td>1023</td>
</tr>
<tr>
<td>Data Rate</td>
<td>1, 2, 5.5, 11 Mbps</td>
</tr>
<tr>
<td>Basic Rate</td>
<td>2 Mbps</td>
</tr>
<tr>
<td>PHY header*</td>
<td>192 µsec</td>
</tr>
<tr>
<td>MAC header</td>
<td>34 bytes</td>
</tr>
<tr>
<td>ACK*</td>
<td>248 µsec</td>
</tr>
</tbody>
</table>

22 µs to transmit the payload
29 µs to transmit the header
802.11 MAC\PHY layer additional 800 µs

Solution is to use Voice Multiplex-Multicast
Voice Multiplex-Multicast (M-M)

- Multiplex packets from various streams into one stream to reduce overhead.
- Replace RTP, UDP and IP header with a mini-header.
- Each payload is preceded with this mini-header representing an ID.
- The AP broadcasts this packet and ID is translated into RTP, UDP IP header at the receiver.
Voice Multiplex-Multicast (M-M)
Issues

- Broadcast issue.
- Security.
- Power Saving Mode of AP.
Voice Capacity Analysis

- Original VOIP

\[
\frac{1}{T_{avg}} = \text{number of streams} \times \text{number of packets sent by one stream in one second.}
\]

\[
OH_{\text{hdr}} = H_{RTP} + H_{UDP} + H_{IP} + H_{MAC}
\]

\[
OH_{\text{sender}} = \text{DIFS} + \text{averageCW} + \text{PHY}
\]

\[
OH_{\text{receiver}} = \text{SIFS} + \text{ACK}
\]

\[
T_{\text{down}} = T_{\text{up}} = \frac{(\text{Payload} + OH_{\text{hdr}}) \times 8}{\text{dataRate}} + OH_{\text{sender}} + OH_{\text{receiver}}
\]

\[
T_{avg} = \frac{T_{\text{down}} + T_{\text{up}}}{2}
\]

\[
n = 11.2
\]
Voice Capacity Analysis

- **M-M VOIP**
  Mini-header is used to compress the IP/UDP/RTP header into 2 bytes.

\[
T_{down} = \left[ (\text{Payload} + 2) \times n + H_{\text{UDP}} + H_{\text{IP}} + H_{\text{MAC}} \right] \times 8 / \text{dataRate} + OH_{\text{sender}}
\]

\[
T_{avg} = \frac{T_{down} + n \times T_{up}}{n + 1}
\]

\[
1 / T_{avg} = (n + 1) \times N_p
\]

\[
n = 21.2
\]
Voice Capacity Analysis

Two type of sources

- Constant-bit-rate (CBR)
- Variable-bit-rate (VBR)

In VBR average traffic is 45% of CBR.
Original VOIP with VBR source can have 26 sessions.
M-M VOIP with VBR source can have 50 sessions.
Simulation

- ns-2 simulation
- Payload size GSM 6.10 codec
- Increase the number of streams until we reach 1% packet loss

<table>
<thead>
<tr>
<th>Different Schemes</th>
<th>CBR</th>
<th>VBR</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Analysis</td>
<td>Simulation</td>
</tr>
<tr>
<td>Original VoIP</td>
<td>11.2</td>
<td>12</td>
</tr>
<tr>
<td>Multiplex-Multicast Scheme</td>
<td>21.2</td>
<td>22</td>
</tr>
</tbody>
</table>
Simulation Observations

- Adding 13\textsuperscript{th} session to CBR for original VOIP causes 6\% Packet loss.

- VBR for M-M scheme number of sessions is lower than expected.

- VBR traffic is bursty causing packet collisions.

- Multicast frames will be dropped after first collision.
802.11 Packets

**Unicast Packet**
- Channel Busy
- DIFS
- Backoff
- PHY
- MAC
- PAYLOAD
- Next Frame
- DIFS
- Backoff
- ACK
- SIFS

**Multicast Packet**
- Channel Busy
- DIFS
- Backoff
- PHY
- MAC
- PAYLOAD
- DIFS
- Backoff
- Next Frame

**MAC-layer multicast priority**
- MIFS
- PHY
- MAC
- PAYLOAD
- DIFS
- Backoff
- Next Frame
Delay Performance

- To provide good quality minimizing the delay is important.

- Delay jitter is the variation in delay.

- Delay of simple VOIP is AP + Receiver Station.

- Delay of M-M is AP + MUX + Receiver.

- We want to keep less than 1% packets with delay of 30ms.
Delay Performance (CBR)

Original VOIP with 12 sessions

- Average delay 2.5ms
- Jitter delay 1.4ms

- Average delay 1.2ms
- Delay jitter 1ms
M-M VOIP with 22 sessions
Average delay 0.9ms
Jitter delay 0.2ms

Average delay 2ms
Delay jitter 1.5ms
Delay Analysis (VBR)

Original VOIP with 25 sessions
Average delay 3.6ms
Jitter delay 5.9ms

Average delay 1.4ms
Delay jitter 1.3ms
Delay Analysis (VBR)

M-M VOIP with 36 sessions

Average delay 1.1ms
Jitter delay 0.7ms

Average delay 0.9ms
Delay jitter 0.7ms
# Delay Analysis

## Access Delay Distribution for Ordinary VoIP

<table>
<thead>
<tr>
<th></th>
<th>Access delay for the AP (Local delay for downlink VoIP packets)</th>
<th>Access delay for the station (Local delay for uplink VoIP packets)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CBR(12)</td>
<td>VBR(25)</td>
</tr>
<tr>
<td>$\Pr [A \leq 10, ms]$</td>
<td>1</td>
<td>0.900</td>
</tr>
<tr>
<td>$\Pr [A \leq 30, ms]$</td>
<td>1</td>
<td>0.990</td>
</tr>
<tr>
<td>$\Pr [A \leq 50, ms]$</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>
Delay Analysis

• Delay at the MUX is upper bound limited by 20ms.
• We are going to consider MUX delay to be distributed from 0 to 20 ms

<table>
<thead>
<tr>
<th>Access delay for the AP plus MUX delay in the MUX (Local delay for the downlink VoIP packet)</th>
<th>Access delay for the station (Local delay for the uplink VoIP packet)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CBR(22)</td>
</tr>
<tr>
<td>Pr[M + A ≤ 0.01s]</td>
<td>0.455</td>
</tr>
<tr>
<td>Pr[M + A ≤ 0.02s]</td>
<td>0.955</td>
</tr>
<tr>
<td>Pr[M + A ≤ 0.03s]</td>
<td>1</td>
</tr>
</tbody>
</table>
Coexistence with TCP

![Diagram showing network components and data flows related to coexistence with TCP. The diagram includes nodes labeled FTP Server, FTP Client, AP, Voice Gateway, S1, ..., Sn, and VoIP, with data flows indicated by arrows.]
Coexistence with TCP

- TCP interferes with downlink FIFO queue.
- TCP ACK interferes with clients sending VOIP streams.
Coexistence with TCP

6 VOIP sessions with one TCP flow

**Original VOIP**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access delay / jitter of the AP (ms)</td>
<td>83.9 / 15.6</td>
</tr>
<tr>
<td>Access delay / jitter of the station (ms)</td>
<td>2.3 / 3.0</td>
</tr>
<tr>
<td>VoIP downlink packet loss rate</td>
<td>1.0 %</td>
</tr>
<tr>
<td>VoIP uplink packet loss rate</td>
<td>0</td>
</tr>
<tr>
<td>TCP throughput (Mbps)</td>
<td>2.55</td>
</tr>
</tbody>
</table>

**Original VOIP with Priority Queuing**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Access delay / jitter of the AP (ms)</td>
<td>3.0 / 1.5</td>
</tr>
<tr>
<td>Access delay / jitter of the station (ms)</td>
<td>2.6 / 2.2</td>
</tr>
<tr>
<td>VoIP downlink packet loss rate</td>
<td>0.01 %</td>
</tr>
<tr>
<td>VoIP uplink packet loss rate</td>
<td>0</td>
</tr>
<tr>
<td>TCP throughput (Mbps)</td>
<td>2.55</td>
</tr>
</tbody>
</table>
## Coexistence with TCP

### 6 M-M VOIP sessions with one TCP flow

<table>
<thead>
<tr>
<th></th>
<th>Access delay / jitter of the AP (ms)</th>
<th>Access delay / jitter of the station (ms)</th>
<th>VoIP downlink loss rate</th>
<th>VoIP uplink loss rate</th>
<th>TCP throughput (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>M-M</td>
<td>42.7 / 19.2</td>
<td>4.5 / 6.2</td>
<td>10.8 %</td>
<td>0</td>
<td>3.46</td>
</tr>
<tr>
<td>M-M + PQ</td>
<td>4.3 / 2.4</td>
<td>4.7 / 6.2</td>
<td>12.2 %</td>
<td>0</td>
<td>3.49</td>
</tr>
<tr>
<td>M-M + MMP</td>
<td>17.2 / 14.5</td>
<td>4.4 / 5.2</td>
<td>0</td>
<td>0</td>
<td>3.47</td>
</tr>
<tr>
<td>M-M + PQ + MMP</td>
<td>2.7 / 2.1</td>
<td>4.6 / 5.8</td>
<td>0</td>
<td>0</td>
<td>3.47</td>
</tr>
</tbody>
</table>
## Coexistence with TCP

### 11 M-M VOIP sessions with one TCP flow

<table>
<thead>
<tr>
<th></th>
<th>Access delay / jitter of the AP (ms)</th>
<th>Access delay / jitter of the station (ms)</th>
<th>VoIP downlink loss rate</th>
<th>VoIP uplink loss rate</th>
<th>TCP throughput (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>M-M</td>
<td>32.5 / 25.8</td>
<td>6.6 / 10.2</td>
<td>15.6 %</td>
<td>0</td>
<td>2.55</td>
</tr>
<tr>
<td>M-M + PQ</td>
<td>4.5 / 3.2</td>
<td>6.7 / 13.5</td>
<td>12.0 %</td>
<td>0</td>
<td>2.54</td>
</tr>
<tr>
<td>M-M + MMP</td>
<td>20.3 / 21.7</td>
<td>8.9 / 20.8</td>
<td>0.2 %</td>
<td>0</td>
<td>2.54</td>
</tr>
<tr>
<td>M-M + PQ+MMP</td>
<td>2.9 / 2.7</td>
<td>5.8 / 7.2</td>
<td>0</td>
<td>0</td>
<td>2.54</td>
</tr>
</tbody>
</table>
Transmission Errors

- In a real world there are transmission errors.

### Multicast Packet Loss Rate for Different Distances and Data Rates

<table>
<thead>
<tr>
<th>Distance (m)</th>
<th>Multicast frames at 2 Mbps</th>
<th>Multicast frames at 11 Mbps</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0</td>
<td>0.17%</td>
</tr>
<tr>
<td>5</td>
<td>0</td>
<td>0.15%</td>
</tr>
<tr>
<td>10</td>
<td>0</td>
<td>0.17%</td>
</tr>
<tr>
<td>20</td>
<td>0.02%</td>
<td>0.23%</td>
</tr>
</tbody>
</table>
802.11e

Implements QoS by having multiple queues for different types of packets.

- EDCA0: One queue for all the traffic, same parameter setting as in DCF
- EDCA1: CWmin[voice] = CWmin[data] = 31
- EDCA2: CWmin[voice] = 31, CWmin[data] = 63
- EDCA3: CWmin[voice] = 31, CWmin[data] = 127
## Performance of Different Parameter Settings for One VoIP + One TCP

<table>
<thead>
<tr>
<th></th>
<th>Access delay / jitter of the AP (ms)</th>
<th>Access delay / jitter of the station (ms)</th>
<th>TCP throughput (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDCA0</td>
<td>23.26 / 15.46</td>
<td>1.98 / 1.47</td>
<td>3.45</td>
</tr>
<tr>
<td>EDCA1</td>
<td>2.72 / 2.12</td>
<td>2.84 / 2.06</td>
<td>3.45</td>
</tr>
<tr>
<td>EDCA2</td>
<td>2.21 / 1.54</td>
<td>2.23 / 1.41</td>
<td>3.07</td>
</tr>
<tr>
<td>EDCA3</td>
<td>1.99 / 1.15</td>
<td>1.94 / 1.16</td>
<td>2.43</td>
</tr>
</tbody>
</table>

## Performance of Different Parameter Settings for Six VoIP + One TCP

<table>
<thead>
<tr>
<th></th>
<th>Access delay / jitter of the AP (ms)</th>
<th>Access delay / jitter of the station (ms)</th>
<th>TCP throughput (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDCA0</td>
<td>56.15 / 26.62</td>
<td>4.12 / 2.65</td>
<td>2.19</td>
</tr>
<tr>
<td>EDCA1</td>
<td>14.58 / 6.43</td>
<td>4.89 / 4.17</td>
<td>2.44</td>
</tr>
<tr>
<td>EDCA2</td>
<td>10.82 / 3.02</td>
<td>4.29 / 2.94</td>
<td>2.16</td>
</tr>
<tr>
<td>EDCA3</td>
<td>9.23 / 2.03</td>
<td>3.86 / 2.56</td>
<td>1.71</td>
</tr>
</tbody>
</table>
Conclusion

- M-M improves the VOIP capacity.
- M-M requires no MAC change in the Client station.
- M-M doesn’t increase delay above 30ms.
- With Both Original VOIP and M-M VOIP quality is unacceptable with TCP.
- Priority queue can solve TCP interference problem.
Observations

- Didn't discuss the effect of mobility causing variable bandwidth.
- Need to look at scenario with majority of TCP and few VOIP users.
- Power utilization at the client stations.
- Only works between AP and AP.