Quality of Service Comparison Between LTE and WiMAX Networks For Streaming Voice and Video Conferencing Using OPNET v16.0

ENSC 427 - Communication Networks
Spring 2013

Group #9 Project URL:
http://www.sfu.ca/~jpa30/
Hamidreza Haghshenas
Jeff Priest
Filip Zivkovic
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
Why Test Video and Voice?

- High versus Low Throughput
  - Vary the amounts of traffic.

- Both use UDP, which is a simpler protocol than TCP.
  - Thus, less changing variables.
LTE (long-term evolution)

- Increased capacity and speed via new DSP techniques
- Reduced latency
- IP-based network
- Available in December of 2009
- Packet-switching protocol vs circuit-switching in GSM
WiMAX (Worldwide Interoperability for Microwave Access)

- Wireless communications standard
- Provides wireless broadband access
- Alternative to Cable / DSL
- Easy to deploy in remote locations
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
Streaming Multimedia

- Applies to telecommunication networks (as opposed to TV / Radio / etc…)
- Internet television (VoD over IP)
- Real-time text updates (stock tickers, closed captioning, etc…)
- Live streaming (Conferences, gaming, presentations, etc…)
- VoIP (proprietary vs. session-based)
SIP (Session Initiation Protocol)

- Application layer signaling protocol
- Communication sessions for voice and video calls over IP (TCP / UDP / etc…)
- Video conferencing, IM, File Transfers, Online Games
- Similar to HTTP (request / response model)
- URI: `sip:username:password@host:port`
RTP and RTCP

- Real-time Transport [Control] Protocol
- RTP carries the data and RTCP gives asynchronous connection metrics (QoS)
- End-to-end transfer of stream data
- Typically runs over UDP
- Makes use of SIP and RTSP (and SDP) to set up the connection between endpoints
RTSP

- Real-time Transport Streaming Protocol
- Establishes and controls media sessions between end points (similar to SIP)
- Typically uses RTP (and RTCP) as the actual transport medium
- Contains directives such as PLAY, PAUSE, RECORD, TEARDOWN, etc…
- Skype, YouTube, QuickTime, etc…
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
LTE Network Topology
WiMAX Network Topology
Choosing Application Attribute Parameters for Video Conferencing

- Frame Interval Time Information: 10 frame/sec
- Constant frame size
- Interactive multimedia service
- Discrete Traffic

For Voice Conferencing:

- 50% background traffic.
- Global System for Mobile Communications (GSM), full data rate, silence filled with noise at receiver.
Choosing Profile Attribute Parameters for Video Conferencing

- New session starts every 10 sec.
- Endlessly created.
- Each runs in parallel.

Choosing Profile Attribute Parameters for VoIP

- New session starts every 1 second.
User Equipment Modulation Scheme

- 1/2 data redundant.
- QPSK (Quadrature Phase Shift Keying) with 1/2 coding rate
Other Attributes

- Maximum Transmission power of 0.5 Watts, which suitable for mobile devices.
- User Datagram Protocol (UDP) was used for both video and voice conferencing.
- Random seed: 127.
- 20 MHz Frequency Division Duplexing
  - Good for symmetric receive and transmit traffic, as opposed to time division duplexing.
Requirements

- Video Conferencing
  - About 140 ms end-to-end delay is acceptable
  - Some packet loss is acceptable.

- Voice over IP
  - About 140 ms end-to-end delay is acceptable
  - < 0.5 ms jitter
Affect of frame per packet on VoIP latency and bandwidth

<table>
<thead>
<tr>
<th>G.729 Samples per Frame</th>
<th>IP/RTP/UDP Header</th>
<th>Bandwidth Consumed</th>
<th>Latency*</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default (two samples per frame)</td>
<td>40 bytes</td>
<td>24,000 bps</td>
<td>25 ms</td>
</tr>
<tr>
<td>Satellite (four samples per frame)</td>
<td>40 bytes</td>
<td>16,000 bps</td>
<td>45 ms</td>
</tr>
<tr>
<td>Low Latency (one sample per frame)</td>
<td>40 bytes</td>
<td>40,000 bps</td>
<td>15 ms</td>
</tr>
</tbody>
</table>

* Compression and packetization delay only

Packets: Network Layer

Frames: Physical Layer
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
Simulation Results

- **WiMAX Video Results**

**Small Mystery:**

One would expect to see increasing delay or an increased packet-loss.

We don’t see this.
Simulation Results

- LTE Video Results (low traffic)
Uplink is the Bottleneck:
Does this mean that WiMAX outperformed LTE??!
Does this mean that WiMAX outperformed LTE??!

Not quite.
LTE doesn’t perform well for low levels of traffic in our simulation. It is not a setup delay, which we originally thought.
Simulation Results

- LTE Video and WiMAX Video (high traffic)

LTE is designed for higher data rates.
Simulation Results: Voice

- WiMAX voice results 1, 2, and 4 packet per frame

- 15ms, 25ms, and 45ms latencies were predicted for 1, 2, and 4 packets per frame (but parameters were not specified).
Simulation Results: Voice

- Poor LTE voice results transmitting 2 packets per frame at low data rates
Simulation Results

- WiMAX and LTE voice results 10 packet per frame
VoIP for Two Packets Per Frame in LTE Network:
Brief Overview

- Introduction to LTE and WiMAX
- Streaming Multimedia and Streaming Protocols
- Implementation Details
- Simulations and Results
- Conclusions
- Q&A
- References
Conclusion

- LTE generally demonstrated higher throughputs, however, WiMAX gave us the lowest possible delay.

- The LTE model has more complex behavior, and is specialized for high data rates.

- Throughput is proportional to packets per frame sent using VoIP.

- Delay is proportional to packets per frame sent.
Final Comments:

- WiMAX Model was much better documented by OPNET.

- Further testing is required to understand the details of the LTE model.

- We did a lot of tests for the time we were given with the licenses.
Questions?
References