Design of Push to Talk Client for Performance Measurements

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Basic Information

• *Thesis written at Celtius Oy*
  - *Celtius is a privately owned company*
  - *Specialized in communication software*
  - *Customers all over Europe and North America*

• **Supervisor: Professor (Pro tem) Jouni Karvo**

• **Instructor: M.Sc. Juhani Malka**
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Push to Talk (PTT)

- A half duplex speech service
- Many existing solutions
  - Conventional Land Mobile Radios (e.g. PMR, VHF)
  - Trunked Radio Systems (e.g. TETRA)
- Push to Talk over Cellular (PoC)
  - Let's do it over a public cellular network
  - Open Mobile Alliance (OMA) is working for an open standard
    - Started in August 2003 from specifications made by an industry consortium formed by Nokia, Siemens, Ericsson and Motorola
    - Candidate Enabler release was supposed to be ready in end of 2004, but it's likely to be released only in February 2005.
  - Some pre-standard / proprietary solutions already available (e.g. Nokia 5140)
OMA PoC uses
- **SIP and SDP** for signalling (e.g. session setup)
- **RTP/RTCP** for media transfer and talk burst arbitration
- **XML Configuration Access Protocol (XCAP)** for group management

Source: OMA PoC architecture document draft version 18.1.2005
PoC session setup

PoC Client A ➔ SIP/IP Core ➔ PoC Server

SIP INVITE ➔ 100 TRYING ➔ SIP INVITE ➔ 100 TRYING ➔ SIP INVITE B

200 OK ➔ TB Granted ➔ 200 OK ➔ TB Granted ➔ 200 OK

SIP ACK ➔ Media (RTP) ➔ SIP ACK ➔ Media (RTP) ➔ RTCP: TB Taken ➔ Media (RTP)
OMA PoC performance requirements

- **Right-to-speak < 2,0 seconds**
  - The duration between the times a user initiates a PoC session and when he receives a “right-to-speak” indication

- **Start-to-speak < 1,6 seconds**
  - The time it takes a user to receive “Start-to-speak” indication after a floor request in established PoC session.

- **End-to-end channel delay ≤ 1,6 seconds**

- **Voice quality MOS ≥ 3 at BER ≤ 2 %**

- **Turnaround-time ≤ 4 seconds**
  - The duration between the times a user quits talking and when he hears a response from another user
Packet-switched cellular networks

- GPRS networks were originally not designed for real-time traffic such as speech
- Low throughput and long delays. No guarantees for the performance
- Cell re-selection may cause outage of 4.5-7.0 seconds
- Improvements coming up with EGPRS and UMTS

Examples of network performance

<table>
<thead>
<tr>
<th>Network</th>
<th>Throughput</th>
<th>RTT</th>
</tr>
</thead>
<tbody>
<tr>
<td>GPRS</td>
<td>40 kbps</td>
<td>700 ms</td>
</tr>
<tr>
<td>EGPRS</td>
<td>150 kbps</td>
<td>400 ms</td>
</tr>
<tr>
<td>UMTS</td>
<td>240 kbps</td>
<td>200 ms</td>
</tr>
</tbody>
</table>
Objectives of the thesis

The objectives of the master thesis were:

- Design a PoC client that can be used in automated performance measurements
- The client should be able to measure the performance parameters stated in the OMA PoC requirement document
- Test PoC performance with the client
Design and Implementation

- **Separate Push to Talk API was designed and the client was implemented on top of it**

- **Implementation done in C++ according to the industry consortium PoC specifications**

- **In future the PoC API may be also implemented according to OMA specifications**

<table>
<thead>
<tr>
<th>Tester client</th>
<th>PoC API</th>
</tr>
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<tbody>
<tr>
<td>RTP and media API</td>
<td>SIP API</td>
</tr>
<tr>
<td>JRTPLIB</td>
<td>Resiprocate</td>
</tr>
</tbody>
</table>

- **Resiprocate SIP and JRTPLIB open-source libraries were used**

- **SIP signalling compression (SigComp) and group management not implemented**
Test setup

- **Ericsson PoC reference test suite servers were used**
- **The tests were performed over a public GPRS network**
- **The server was connected to Internet**
- **The clients were stationary**
Test results

- **Outages of several seconds occurred even though the clients were stationary**
- **Results are just indicative:**
  - Use of SigComp would lower Right-to-Speak
  - Real servers would be in operator's network
  - Many variables that can be optimized

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Requirement</th>
<th>Measured values</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Average</td>
<td>Worst case</td>
</tr>
<tr>
<td>Right-to-speak</td>
<td>2,0 s</td>
<td>2,09 s</td>
<td>2,49 s</td>
</tr>
<tr>
<td>Start-to-speak</td>
<td>1,6 s</td>
<td>0,63 s</td>
<td>1,45 s</td>
</tr>
<tr>
<td>End-to-end delay</td>
<td>1,6 s</td>
<td>1,30 s</td>
<td>4,44 s</td>
</tr>
<tr>
<td>Turnaround time</td>
<td>4,0 s</td>
<td>4,10 s</td>
<td>6,47 s</td>
</tr>
<tr>
<td>Voice quality</td>
<td>MOS 3</td>
<td>MOS 3,1</td>
<td>MOS 2,02</td>
</tr>
<tr>
<td></td>
<td></td>
<td>(PESQ-MOS)</td>
<td></td>
</tr>
</tbody>
</table>
Conclusions

- The client that was designed and implemented can be used for PoC performance measurements
- PoC can work in GPRS network
- The performance of PoC cannot be guaranteed in current cellular networks
  - Long cell-reselection times
  - Network congestion lowers throughput
- Future cellular networks will improve the usability of PoC
Questions or comments?

Thank you!