RTT TECHNOLOGY TOPIC
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Handset Software Testing

In last month's Hot Topic, we discussed some specific aspects of handset hardware testing, how and why radio hardware performance determines network and system stability.

In this month's Topic, we consider some of the software tasks and tests that will be required to make multi-media handsets work with multi-service networks.

Software functionality within a handset can be divided into baseband processor tasks and (in multi-media phones) the tasks undertaken by the media or applications processor.

The partitioning of hardware and software is dependent on a number of factors including cost, power budget/processor overheads and processor delay and delay variability. This in turn depends on the maturity of the technology.

In GSM phones in 1991, the only baseband function performed by the DSP was the voice encoder/ decoder. More or less all the other tasks including channel coding, were performed in ASICS.

Over the next ten years, the DSP gradually took over almost all of the baseband functions of the phone. This included the baseband tasks such as DC offset compensation, the digital filtering needed for direct conversion receivers and the digital waveform shaping needed for IQ modulators, translational loop architectures and (for EDGE transceiver design), polar modulation techniques.

History is now repeating itself. In a UMTS handset, the voice encoding/decoding is done in the DSP though some of the media processing source encoding/decoding tasks are passed on to hardware co-processors to meet power budget and delay constraints. Channel coding/symbol level encoding/decoding is done in the DSP though tasks such as turbo coding are passed on to hardware co-processors. Chip level processing is presently done in ASICS. It is reasonable to assume that these hardware tasks will move across into the software domain over the next three to five years but only as power budgets and delay budgets come down. The introduction of HSDPA will place particular demands on the handset with the turbo coder taking almost as much processor power as the RRC filter load.

So the 'software radio' is in practice a software radio with a (large) number of hardware accelerators needed to meet power, delay and delay variability budgets.

This suggests that we need to look at software testing not as a separate topic but as part of an overall test strategy to define how well a multi-media phone will perform when accessing a multi-service network.

The starting point is not to look at the handsets themselves but to look at the network.

When signing an infrastructure contract, network infrastructure vendors and network operators will have normally agreed on, or at least discussed, the key performance indices (KPI's) that will need to be achieved for a certain percentage of the area to be covered by the radio system.

This will typically include blocked call rates, call success rates, dropped call rates and possibly, also, agreed voice quality metrics.

The achievement of these KPI's is very dependent on the mix of handsets used in the network.

The bar graphs below are test results from the five UK mobile operators (four GSM and the one publicly available 3G network) and show the mean opinion voice quality score results for 8 different mobile handsets.

Note the significant difference from handset to handset. The worst performing handset in the test was also one of the most expensive. Note that the uplink quality was generally worse than the downlink quality though the worst performing handset actually performed better on the uplink than the downlink suggesting that it had a receiver sensitivity problem.

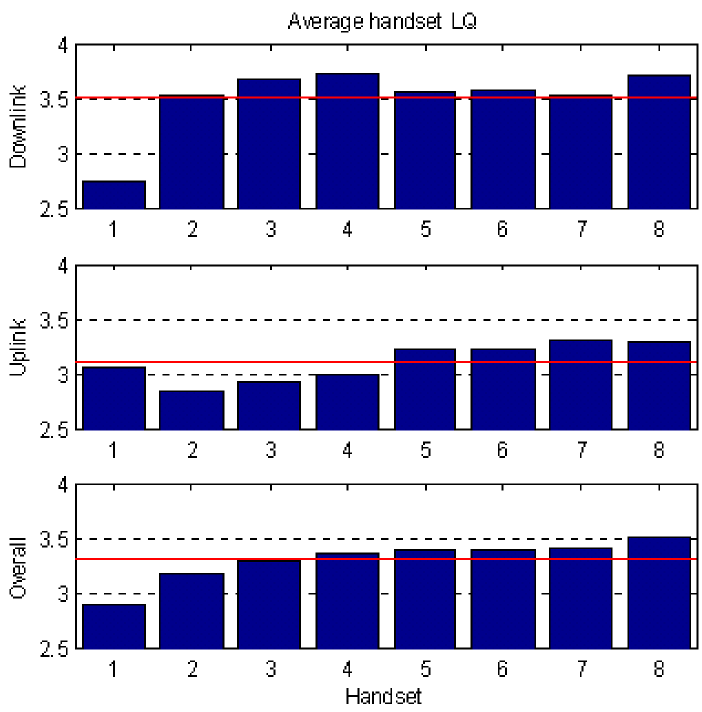

Figure 1 With thanks to [Psytechnics]

The bar graphs below show the worst case performance for the 8 handsets defined as the minimum quality during any 6 second measurement period. The point here is that measurement quality can vary significantly during a call. Not shown but significant is the fact that some handsets dropped calls much more frequently at busy times during the day.

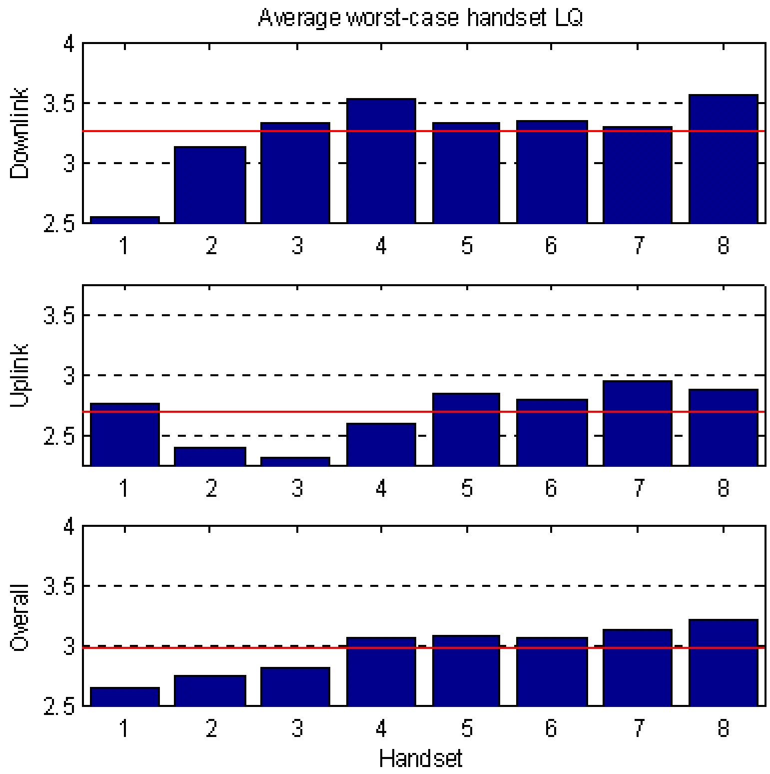

Figure 2: With thanks to [Psytechnics]

The reason for these large differences may be just due to differences in radio performance. The fact that the worst performing phone is also one of the most expensive means that it will also probably have the smallest form factor which makes it harder to produce good receive sensitivity.

It highlights the fact that the RF performance of the phone is still a dominant factor in achieving consistent end to end quality.

The above results were for phones supported on traditional circuit switch connections.

Now consider the impact of introducing Release 6 based IP voice, IP audio and IP video.

The idea of IP voice is that a packet stream from a software based AMR narrow band or wideband encoder will be multiplexed on to a common packet channel and delivered into the IP RAN for software switching over the IP core.

The IP packet stream will carry additional stuffer bit and header overheads and will be subject to delay and delay variability as it passes through the network. This will make it harder to deliver consistent good quality voice.

The same applies for IP audio and IP video, both of which have similar sensitivity to IP overheads and end to end delay variability metrics.

So the software testing of handsets will need to include considerably more than just straightforward functionality testing (does the baseband protocol stack work with the network). To be useful, software testing has to simulate the user experience across a wide (almost infinite) range of end to end channel conditions and comprehend voice, audio and video quality,. This in turn requires the industry to agree on a set of standardised audio and video quality metrics building on existing ITU voice quality measurement methodologies.

Once this is in place, it at least becomes feasible to contemplate the reasonably consistent delivery of IP voice and video over wide area wireless networks.

Acknowledgement: We would like to thank Psytechnics for providing us with the bar graphs used in this Hot Topic. Additional background on Psytechnics voice quality test methodologies can be found on their web site www.psytechnics.com

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