

#### Feb 2009

## Broadband Quality of Service Experience (QoSE) Indicators<sup>1</sup>

Price is not the only dimension that is of interest to customers and regulators. Quality of Service Experience (QoSE) is integrally connected to price: an increase in quality is an invisible decrease in price and vice versa.

Broadband quality can be evaluated through speed tests. Test sites provide a variety of information about the speed of a link. Careful design and implementation of tests can shed light on the exact segment where inadequate capacity constrains speed. Carefully implemented tests can also be the basis for Service Level Agreements (SLAs) between operators and users and for regulatory action.

In the present tests, the methodology has been developed in collaboration with a team headed by Professor Timothy Gonsalves of IIT Madras. The following dimensions of quality have been measured for two networks in Bangladesh (Dhaka) three in India (Chennai and New Delhi) and four networks in Sri Lanka (Colombo).

Referred to as the "actual amount of useful data sent on a transmission"<sup>2</sup>. Defined by the ITU as "an amount of user Throughput (kbps) information transferred in a period of time" (ITU-T X.641 (97), 6.3.3.16), more commonly referred to as download or upload speeds. A key advertised metrics in broadband services is the download speed. It defines how much information a user can received from a local or international server. Upload speed defines the speed in which the user can send information to local or international servers. It plays a significant role in responsiveness and real-time applications like VOIP (Voice Over Internet Protocol). Throughput, or download and upload speeds, varies depending on the location of the server that holds the content. If the location is local, such as an ISP server, the throughput may be higher than it would be if the location is international. Therefore the testing has included throughput for both local (ISP) and international (vahoo.com) servers. "Latency refers to delays when voice packets transverse the network"<sup>3</sup>. It is measured in milliseconds by using the Round Latency (ms) Trip Time (RTT). This is significant in systems that require two-way interactive communication, such as voice telephony, or ACK/NAK [acknowledge/not acknowledge] data systems where the round-trip time directly affects the throughput rate, such as the Transmission Control Protocol (TCP). The ITU definition states that "Latency means transmission delay for FEC (Forwarding Equivalence Class) encoding, decoding, interleaving and de-interleaving" (ITU-T G.972 (04), 3025).

**Jitter (ms)** "Jitter is uneven latency and packet loss"<sup>4</sup>. It is the variation of end-to-end delay from one packet to the next within the same packet stream/connection/flow. Jitter is more relevant for real-time traffic like VOIP. Ideally the figure should be low.

1



#### Feb 2009

E.g. Radio quality voice requires less than 1 ms Jitter, toll-quality voice requires less than 20 ms jitter, normal VoIP requires jitter to be less than 30 ms. Beyond 30 ms, VoIP performance will degrade.<sup>5</sup> Also defined by ITU as "Short-term non-cumulative variations of the significant instants of a digital signal from their ideal positions in time" (ITU-T G.701 (93), 2024).

Packet Loss (%) Number of packets (as a percentage) that does not reach the destination. Degradation can result in noticeable performance loss with streaming technologies, VOIP and video conferencing. ITU states that "In general, IP-based networks do not guarantee delivery of packets. Packets will be dropped under peak loads and during periods of congestion. NOTE – In case of multimedia services, when a late packet finally arrives, it will be considered lost" (ITU-T H.360 (04), 5.3.2.2).



## Feb 2009

Results of QoSE testing<sup>6</sup> (Chennai, Colombo, Dhaka and New Delhi)

# **Fixed Broadband – Throughput (kbps)**<sup>7</sup>

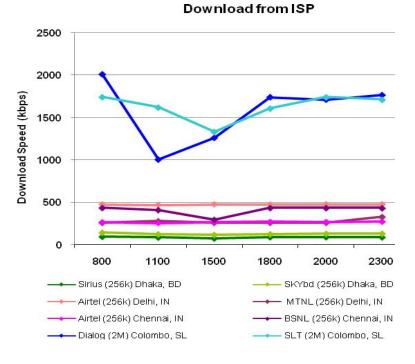


Figure 1<sup>8</sup>

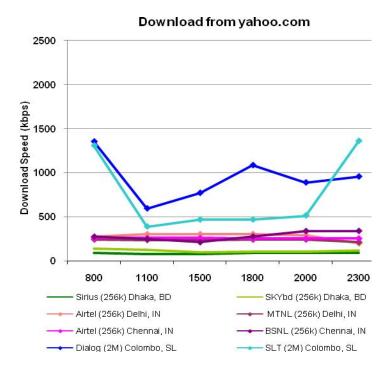
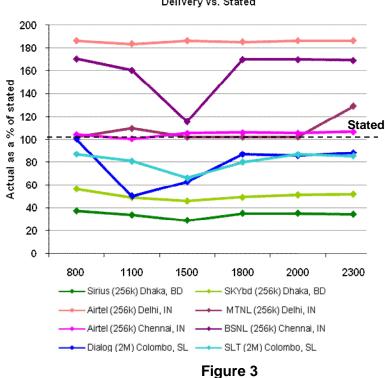


Figure 2



Feb 2009



Download from ISP Delivery vs. Stated

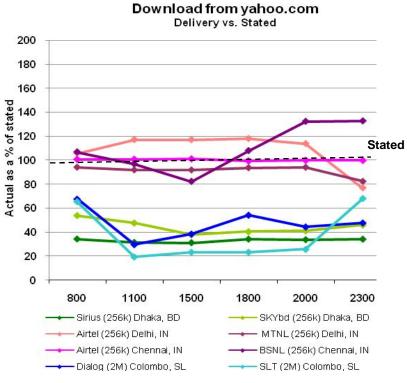
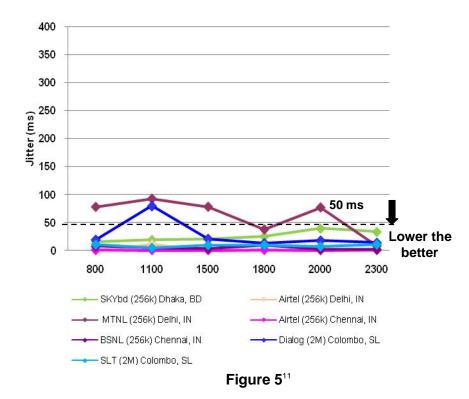


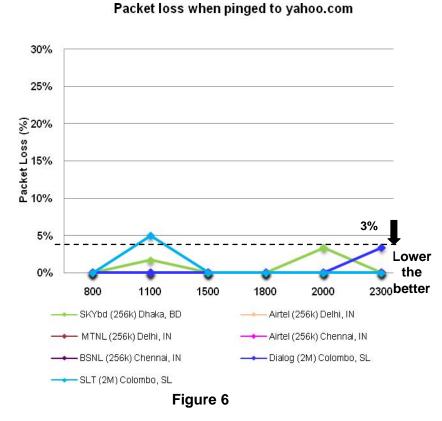
Figure 4



Feb 2009 Fixed Broadband - Jitter<sup>9</sup> and Packet Loss<sup>10</sup>

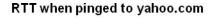


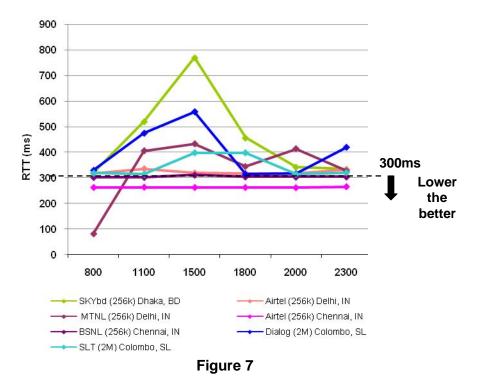






## Feb 2009 Fixed Broadband - Latency<sup>12</sup>

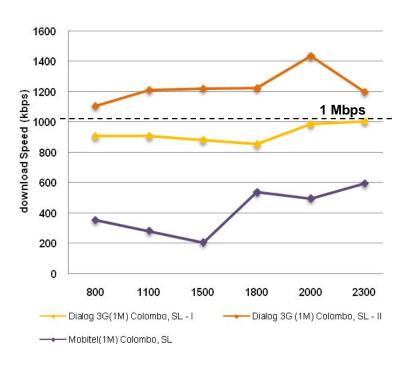




### Quality of Service Experience (QoSE)



## Feb 2009 Mobile Broadband – Throughput (kbps)





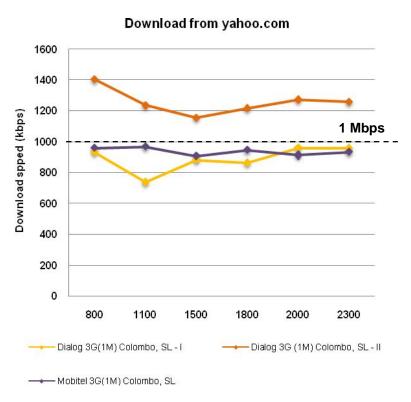


Figure 8<sup>\*+#</sup>



<sup>\*</sup> Dialog 3G(1M) Colombo, SL I – Unlimited Mobile Broadband

<sup>&</sup>lt;sup>+</sup> Dialog 3G(1M) Colombo, SL II – Dialog Mobile Broadband Large

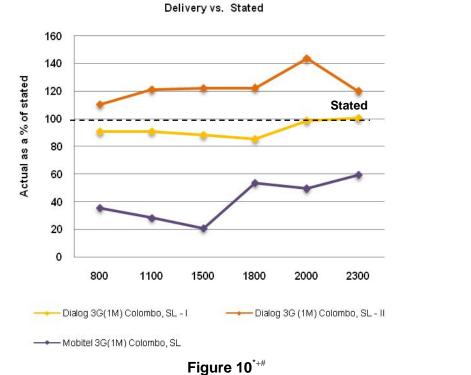
<sup>&</sup>lt;sup>#</sup> Mobitel 3G(1M) Colombo, SL – Zoom 890

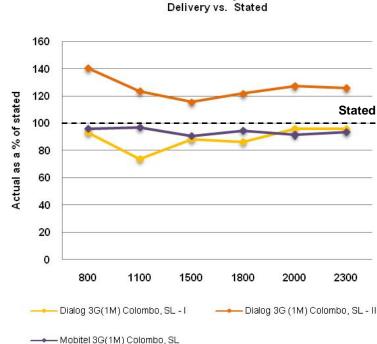


Download from ISP

Quality of Service Experience (QoSE)

Feb 2009





Download from yahoo.com

Figure 11

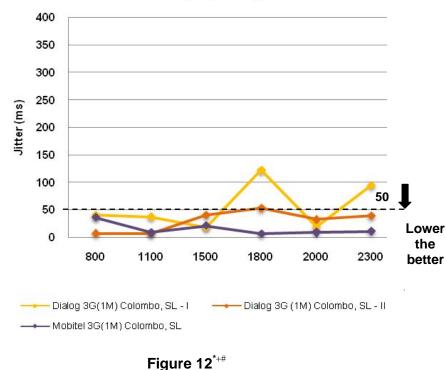
## Dialog 3G(1M) Colombo, SL I – Unlimited Mobile Broadband

<sup>&</sup>lt;sup>+</sup> Dialog 3G(1M) Colombo, SL II – Dialog Mobile Broadband Large

<sup>&</sup>lt;sup>#</sup> Mobitel 3G(1M) Colombo, SL – Zoom 890



Feb 2009 Mobile Broadband - Jitter<sup>13</sup> and Packet Loss<sup>14</sup>



Jitter when pinged to yahoo.com

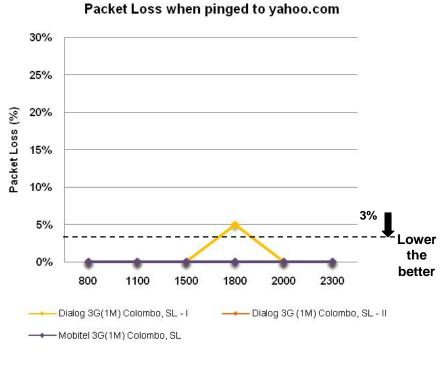


Figure 13

www.lirneasia.net

www. broadbandasia.info

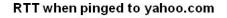
<sup>&</sup>lt;sup>\*</sup> Dialog 3G(1M) Colombo, SL I – Unlimited Mobile Broadband

<sup>&</sup>lt;sup>+</sup> Dialog 3G(1M) Colombo, SL II – Dialog Mobile Broadband Large

<sup>&</sup>lt;sup>#</sup> Mobitel 3G(1M) Colombo, SL – Zoom 890



Feb 2009 Mobile Broadband - Latency<sup>15</sup>



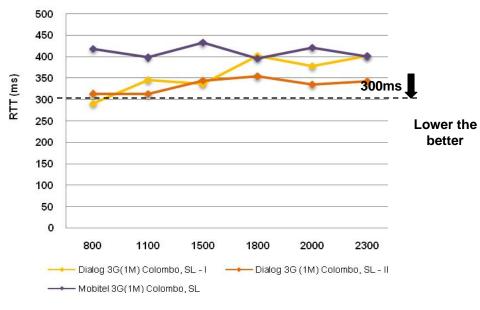


Figure 14<sup>\*+#</sup>

\* Dialog 3G(1M) Colombo, SL I – Unlimited Mobile Broadband

<sup>+</sup> Dialog 3G(1M) Colombo, SL II – Dialog Mobile Broadband Large

<sup>#</sup> Mobitel 3G(1M) Colombo, SL – Zoom 890

### Quality of Service Experience (QoSE)



#### Feb 2009

1 http://www.lirneasia.net/projects/current-projects/2241/.

2 Dodd, A. (2005), "The Essential Guide to Telecommunication" Fourth Edition, Pearson Education, p. 14

3 Dodd, A. (2005), "The Essential Guide to Telecommunication" Fourth Edition, Pearson Education, p. 60

4 Dodd, A. (2005), "The Essential Guide to Telecommunication" Fourth Edition, Pearson Education, p. 60

5 Connection Magazine, http://www.connectionsmagazine.com/articles/5/049.html, CISCO Press Article

6 The connections were tested on:

SLT (Colombo) tested on	: 24 Feb, 2009 & 25 Feb, 2009
Dialog (Colombo) tested on	: 24 Feb, 2009 & 25 Feb, 2009
BSNL(Chennai) tested on	: 22 Feb, 2009 & 24 Feb, 2009
Airtel(Chennai) tested on	: 22 Feb, 2009 & 24 Feb, 2009
MTNL (Delhi) tested on	: 17 Feb, 2009 18 Feb, 2009 & 20 Feb, 2009
Airtel (Delhi) tested on	: 19 Feb, 2009 & 20 Feb, 2009
Sirius (Dhaka) tested on	: 31 Jan, 2009 & 1 Feb, 2009
SKYbd (Dhaka) tested on	: 06 Feb, 2009 & 08 Feb, 2009
Mobitel 3G(Colombo) tested on	: 24 Feb, 2009 & 25 Feb, 2009
Dialog 3G - Unlimited (Colombo) tested on	: 11 Feb, 2009, 12 Feb, 2009 & 13 Feb 2009
Dialog 3G – 1GB Limit (Colombo) tested on	: 24 Feb, 2009 & 25 Feb, 2009

7 The speed at which the subscriber can receive traffic from the ISP server and a commonly used International Server (e.g. yahoo.com). It plays a significant role in responsiveness and real-time applications like VOIP.

8 For Dialog WiMAX (2M) the reading for National domain is taken as the speed for ISP could not be obtained due to unknown technical reason

9 Jitter is the variation of end-to-end delay from one packet to the next within the same packet stream/ connection/ flow. Jitter experienced in packets, more relevant in Real-time traffic like VOIP. Ideally it should be zero.

10 Number of packets (in %) that does not reach the destination. This can result in highly noticeable performance issues with Streaming Technologies, VOIP and Video conferencing.

11 Loss and Delay information not available for Sirius Broadband package

12 Time taken for traffic to reach a particular destination.

13 Jitter is the variation of end-to-end delay from one packet to the next within the same packet stream/ connection/ flow. Jitter experienced in packets, more relevant in Real-time traffic like VOIP. Ideally it should be zero.

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15 Time taken for traffic to reach a particular destination.