channels have their flow stopped at the source side (operation stop at source binders).

4.2. Performance Considerations

In order to assess the channel performance two channels (audio and video) with one source and three sinks were established. In the audio channel the application object samples audio at 8 KHz and 8 bits/sample, while in the video channel the application object samples video frames of 640x480 pixels (16 bits/pixel) at rate of 30 frames/s.

Typical values for QoS obtained for the audio channel are shown in Table 1. Notice that the application is not able to transfer the demanded bandwidth (8 kbytes/s for 8 KHz 8 bit audio and 88.2 Kbytes/s for 44.1 KHz 15 bit audio). This is explained by the lack of a real-time scheduler at the node and a synchronous or asynchronous network.

<table>
<thead>
<tr>
<th>Type</th>
<th>Bandwidth (Kbytes/s)</th>
<th>Delay (ms)</th>
<th>Jitter (ms)</th>
<th>PER (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>7.89</td>
<td>0.3</td>
<td>0.03</td>
<td>0</td>
</tr>
<tr>
<td>A</td>
<td>7.38</td>
<td>7.8</td>
<td>0.04</td>
<td>0</td>
</tr>
<tr>
<td>B</td>
<td>39.27</td>
<td>8.7</td>
<td>0.06</td>
<td>24</td>
</tr>
<tr>
<td>B</td>
<td>41.29</td>
<td>9.1</td>
<td>0.05</td>
<td>10</td>
</tr>
</tbody>
</table>

Table 1: Typical QoS measurements for an audio channel. Audio type A has sampling rate of 8 KHz and precision of 8 bits; audio type B has sampling rate of 44.1 KHz and precision of 16 bits.

The overhead introduced by the channel can not be perceived when compared with the same application build above the plain socket interface. In fact, compared with MBONE tools such as NV (videoconference tool) and VAT (audioconference tool), audio and video transported through ODP channels perform equally as well. We claim that tools build over our multithreaded implementation of ODP channels will perform much better in multiprocessed workstations where true parallelism is achieved when manipulating multiple media flows.

5. Quality of Service Issues

We developed a very elaborated architecture for quality of service negotiation and management based on mobile agents technology [11]. References [12, 13] report such architecture, while references [14, 15] provide a comprehensive survey on QoS for distributed multimedia applications. In this paper we present a simplified scheme based on fixed objects.

During the establishment of a channel, some parameters related to quality of service are passed to the factory. At the moment we have no way to guarantee that these parameters will be honored by the network. At most we can monitor these parameters and take some corrective action when a strong degradation of QoS is detected.

Our current implementation of ODP channels handles the following QoS parameters:

- network delay: the elapsed time between the submission of a RTP packet to the network and its receiving at the sink side;
- delay jitter: the variation of network delay;
- bandwidth: byte flow (bytes/s) from source to sink(s);
- packet error rate (PER): the percentage of packets carrying media segments discarded by the network.

These parameters are computed at the sink side only. Jitter and PER are computed using the information carried in the RTP header. Bandwidth is computed simply by counting the amount of bytes received during a fixed interval. Delay is computed by sending an echo request to the UDP port number 7 (echo port) at the source socket and measuring the round trip time. A weighted-average scheme is employed in order smooth delays observed during a short network connection.

A global data structure storing these parameters is updated when a certain amount of RTP packets are received. This same structure is accessed by the binder when a QoS report is requested. We implemented a