

nous performance of the other musicians) perceptible to a musician.

Due to such a tight delay bound, NMP applicability is restricted to some specific scenarios. The distance between each collaborating client and the server is inevitably limited by the signal propagation speed and link capacity and is further reduced by delays introduced in processing at routers in the networks and at the end systems (the NMP client and server). Since we do not assume a QoS-supported Internet, our optimization strategy is to minimize the end system delays in order to increase the remaining budget for network delay that directly affects the application boundary of an NMP session.

Closer inspection of the processing and data paths in the end systems reveals numerous delay sources that for simplicity can be divided into buffering and computational latencies. The processing scheme in the client's sound card defines the application's buffering granularity and is a major component of end-system delay. To assure continuous recording and playback, sound cards preserve internal hardware buffers before the digital-to-analog converter and after the analog-to-digital converter.

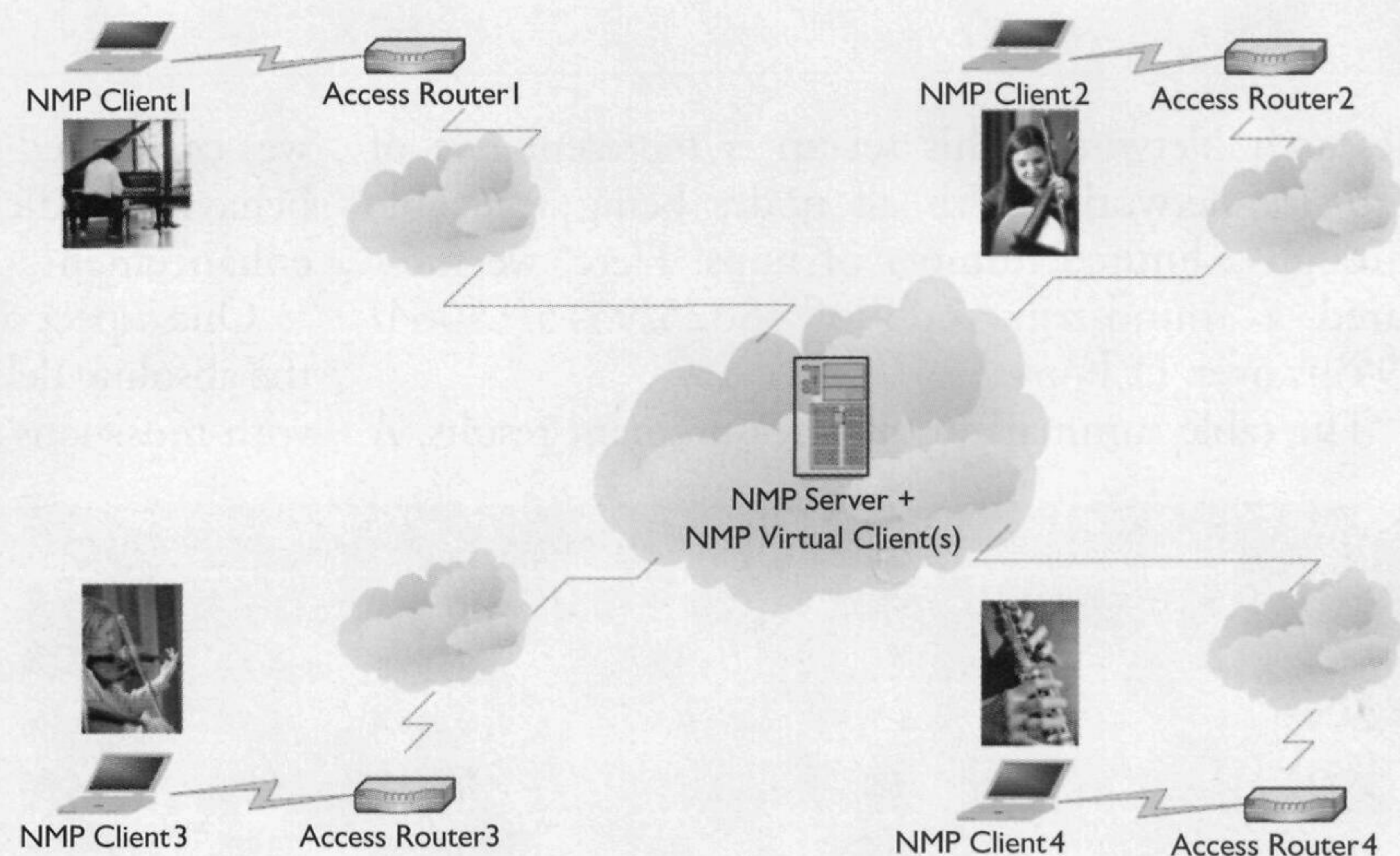
While the converters operate on the front buffer set, the operating system processes the back buffer set. However, the OS is not able to process audio data immediately after the sound card has flipped the front and back buffers, since the sound card's interrupt service routine must be scheduled first. Therefore, at least one additional packet must be buffered by the OS in each direction.

Hence, the sound card (plus the OS) introduce a minimum buffering delay of four buffer units at the client. The size of these units is constant within a session and has a predefined size that is set according to the sampling rate being used. We use a sampling rate of 48kHz with 128 samples per buffer unit. The settings result in a packet buffer delay of 2.667msec. It defines the atomic buffering unit for the application's buffer dimensioning (and is denoted as  $t\phi$  here).

At the server, under realistic network conditions, received audio packets must first be buffered before they are mixed in order to compensate for network jitter; ideal network conditions are assumed for estimating the application boundary. Since in this case de-jitter buffering is not used, the overall latency at

the server is only the computational overhead.

Compared to buffering delay, processing delay due to computational overhead is a magnitude lower and can be ignored. We measured a total computational overhead for an audio packet of  $50\mu\text{s}$ – $180\mu\text{s}$ . We thus



Scenarios involving network-centric music performance.

approximated the total end-system delay under ideal network conditions to the buffering delay at the client, or  $t\phi = 10.7\text{msec}$ , allowing  $30\text{msec} - 10.7\text{msec} = 19.3\text{msec}$  for network delay.

#### UNDER DIFFERENT NETWORK CONDITIONS

The network delay budget of 19.3msec is used to guide the delay jitter compensation at the end systems. This compensation is typically done by configuring the de-jitter buffers at the server and client with granularity of  $t\phi = 2.667\text{msec}$ . More buffers increase the overall tolerable delay but have a better chance of smoothing out the jitters at the network. Trading delay for loss enables an NMP system to operate under different network conditions and to match different user expectations.

Two network set-ups described in the following paragraphs demonstrate that NMP is operable within the defined delay bound. The NMP sessions are formed by three NMP clients connected to a server. Packet loss rate is measured for a given number of deployed buffers in intervals of 1,000 packets for a total period of 20 minutes. Late packets are treated as losses due to the real-time semantics of the application.

The first NMP set-up is for a LAN; the machines are directly connected through a fast Ethernet switch, with a measured round-trip time of (min/avg/max/dev =  $272/310/2630/193\mu\text{s}$ ). The second set-up is for a WAN spanning a one-way distance of about 300km where all the computers are part of the German