Bandwidth, Latency, Routing and Video Conferences

Video conferencing makes business communication easier and more effective, but only when the online experience is smooth and free of delays. The choppy motion and lack of synchronization of some conferences stand in contrast to the synchronization of high-quality conferences.

What is the difference between a smooth video conference and a choppy, jittery video conference? The answer boils down to three factors:

1. Bandwidth – the rate in bits per second at which the network connection can move data in the conference (i.e., how wide is the road from A to B?)
2. Latency – the delay in milliseconds in the network connection used by the conference (i.e., how much other traffic is on the road between A and B?)
3. Routing – the network route that data packets follow during the conference (i.e., how many turns does the road take between A and B?)

When all of these factors are optimal, the video conference runs smoothly, but all three factors are subject to variables such as other network traffic and the status of equipment between the participant and the conference. Typically, low bandwidth, high latency and inefficient routing can lead to low-quality conferences.

Technical issues

Every video conference faces several technical realities:

1. When a network connection is slow (i.e., shows high latency, low bandwidth or poor routing), the computer sends fewer data packets across it.
2. There is only one way out of the computer, and it's serial.
3. Performance bottlenecks in the PC itself can affect video conference quality.
4. When packets of video conference data drop, the quality of the conference decreases, and the process of resending the packets is not always error-free.

Nefsis deals with each of these realities in unique ways to ensure high-quality video, voice and data in every conference.

Latency, bandwidth and routing

Latency, bandwidth and routing are directly related to quality in the video conferencing experience. When latency is low (e.g., 10-20ms), the connection between the computers will accept more packets of data at a time, so the computer can send more and the quality of the video conference will improve. Conversely, when latency is high (e.g., 120-180 ms), the network connection accepts fewer packets at a time and quality suffers.

The computer's capacity for buffering data before sending it depends on the TCP window size. By increasing and decreasing the TCP window size, computers mutually ensure that each can transmit data at the optimal speed for the other.

However, even optimal window size cannot overcome extreme latency if there is insufficient bandwidth or routing problems. Network congestion causes the buffer to fill, so the video conferencing application has no choice but to drop some data packets and send fewer across the network, resulting in lower quality.

> *Most applications simply rely on the operating system to control the TCP window size, but as part of its Real-Time Routing Engine, Nefsis can control this setting. To deal with latency, low bandwidth and routing delays, Nefsis automatically optimizes data flow and compensates for delays in the network.*

Serial connection
By default, computers send data over the network serially. At any given time, multiple processes are sending packets and multiple processes are waiting for the packets at the other end, but as described above, some processes require more time than others before their packets are ready to go to the network. On any network, packets sent in 1-2-3-4-5-6 order may arrive in a different sequence, like 1-4-2-5-3-6. This is not much of a problem in bursty, asynchronous communication like e-mail or Web browsing, but out-of-sequence packets can result in poor quality in the sustained, synchronous context of video conferencing.

Still, because networks are set up to move data serially, it is common for video conferencing products to follow that model and push their packets to the network serially, favoring throughput over conference quality. This presents another problem, because the TCP window can take only so many packets, and handing them off serially can lead to a buffer overflow. To prevent this overflow, most video conferencing products throttle back and send fewer packets, but that leads to the quality and performance problems described above.

Serial processing requires that the source computer drop packets to maintain synchronization or delay them to control the amount of data going to the network; otherwise, there is the risk of overflowing buffers. The common remedy is to reduce the amount of data going to the network, but that lowers the quality of the online meeting.

> *Whereas most products maintain a serial connection during an online meeting, Nefsis maintains a parallel connection process at the socket layer. The parallel communication engine identifies packets at the source computer and reorders them at destination to achieve the highest bandwidth possible. Nefsis also controls the amount of data sent to the network based on available bandwidth.*

PC-centric factors
At the ends of the video conference, participants' computers play a role in the quality of the online meeting. Factors such as the camera, the device driver used with the camera, operating system updates, and software (especially anti-malware) running in the background can affect the machine's ability to process video conferencing data, both sending and receiving.

Conditions on each participant's LAN also come to bear on how the online meeting performs. Proxy servers and firewalls, if too restrictive or improperly configured, may create bottlenecks.

> *Nefsis' Virtual Conferencing Server (VCS) identifies any high-latency connections in an online meeting and automatically compensates for the latency in the flow of conference data to the participant's computer.*

Dropped packets
Inevitably, some packets drop in the transmission of video conferencing data. This manifests itself in missing regions of the on-screen image, distortion in audio or video, and blank spaces in the session.

Because most traditional video conferencing products use UDP, which does not provide for resending dropped packets, poor network conditions can severely impact quality. Web conferencing products take advantage of common network protocols for resending packets, but in greatly degraded network conditions, some of these packets still remain irretrievable.

> *Nefsis' VCS uses the parallel communication engine to act as a broker by moving data packets among conference participants. The VCS also acts as a real-time router to request re-send of dropped packets, receive them, and forward them to all participants' machines, which reorder and reassemble them. The*

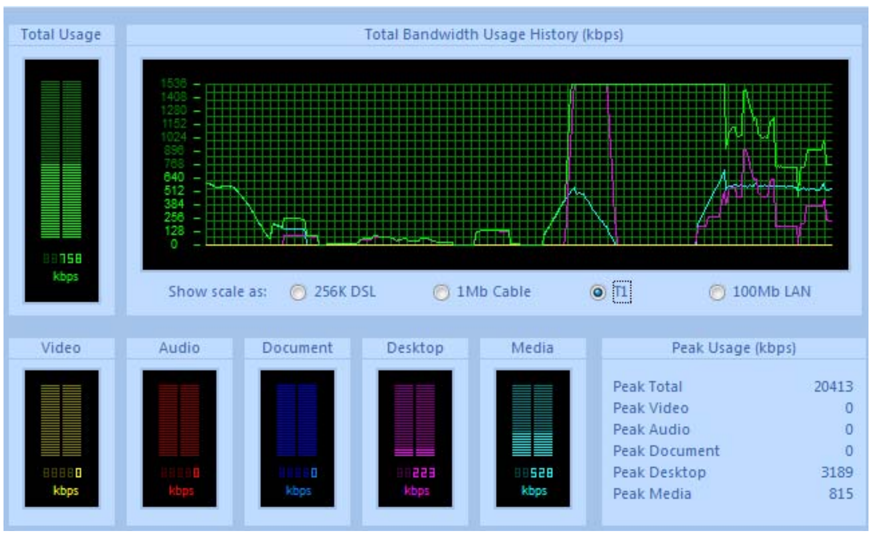
VCS does this not only on the current conference, but for all conferences taking place on the server at any time.

Nefsis tools

Because the network plays such a prominent role in video conference quality, Nefsis has developed tools for gauging its effects.

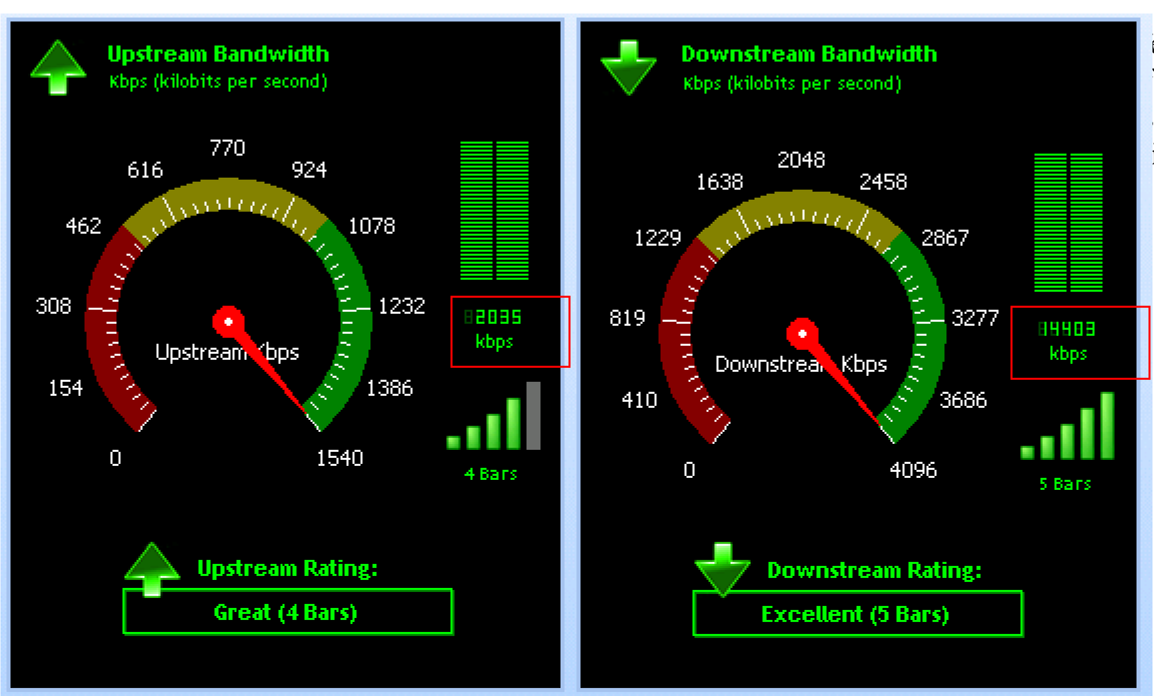
(These tools are for measurement only; they do not allow for configuration or changing settings. For more details on how to use them, see the [Bandwidth Monitor]() and [Network Diagnostics]() sections of the Nefsis Online User Guide.)

Nefsis includes a Bandwidth Monitor utility, as shown in the example below. Presenters and hosts can run the utility to measure total network bandwidth usage during an online meeting. The utility describes the portion of bandwidth dedicated to each of the processes at work in the online meeting: video, audio, document sharing, desktop sharing and media file sharing.



latency-bwith-monitor.png

Nefsis also includes three diagnostic tools for testing network conditions. (Unlike Bandwidth Monitor, these tests pause Nefsis on the computer running them.)



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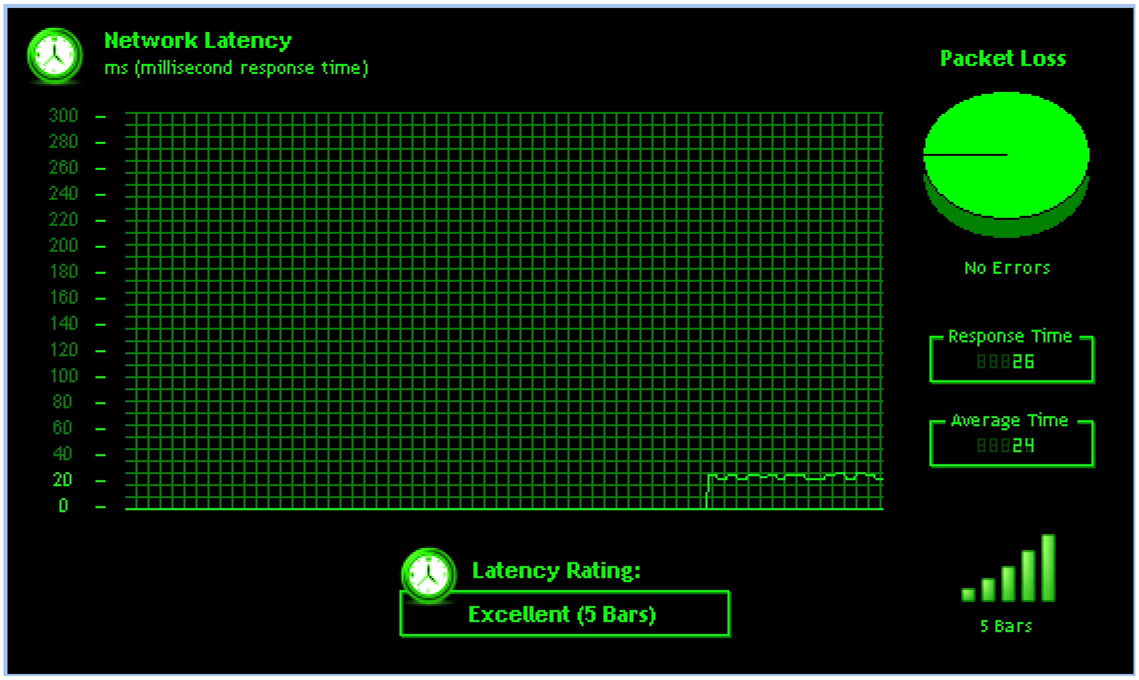
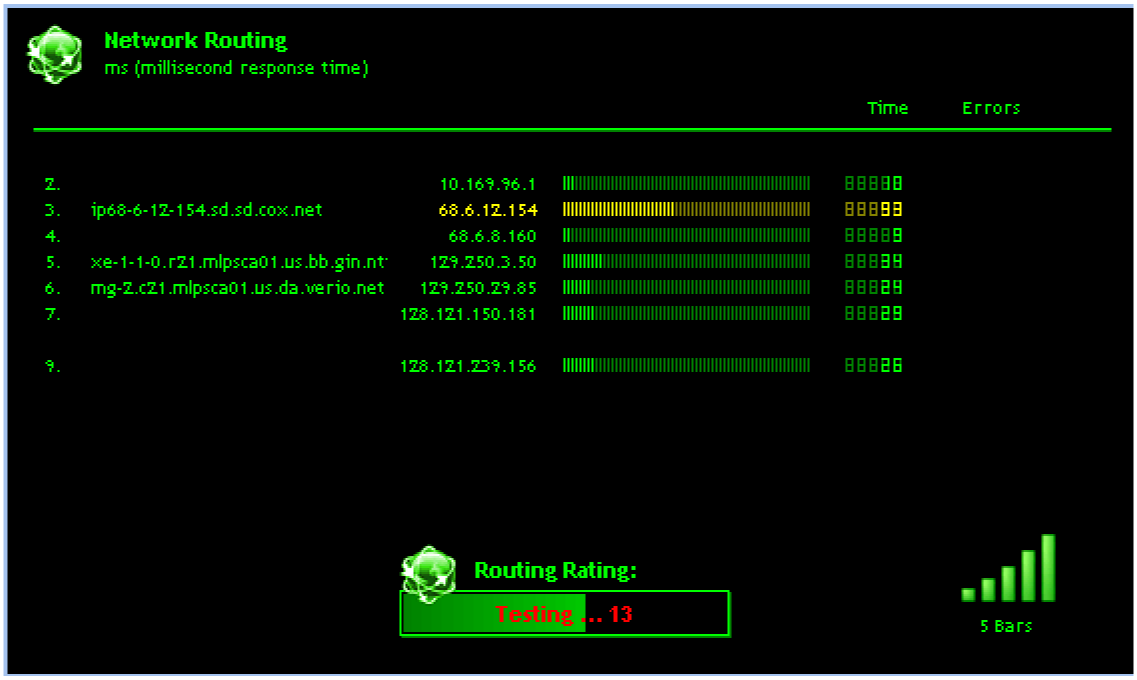
- Latency – Tests the average delay in milliseconds in moving a packet round-trip between the computer and the Nefsis Virtual Conference Server in use for the current conference. The utility reports any packets that may drop during the trip and the average response time for packets, as shown in the example below.



latency-latency-test.png

- Routing – Describes and times the route across the Internet between the computer and the Nefsis Virtual Conference Server. This test can identify bottlenecks and slow connections along the route, as shown in yellow in the example below.



latency-routing-test.png

Conclusion
Nefsis maintains multiple parallel processes for multiple streams of data – VoIP, video, data, image capture, status – going over a single network connection. It also manages the handoff of data to the network to keep the quality of the video conference as high as possible for all participants.