Audio Streaming Over the non-QoS Internet

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As the web continues to envelope and reshape the world of communications, audio engineers are asking how to use the public Internet for audio streaming. The Telos support team has developed a guide to help you manage the streaming process as simply as currently possible.

1. Know the Bottleneck

In most configurations, especially at remote locations, the uplink direction of the Internet connection is where the bottleneck occurs.

DSL may claim many megabits for Internet connections, but DSL lines are "asymmetrical". This means the download speed is much higher (by a factor of ten or so) than the upload speed. This makes sense for browsing the Internet but not for audio streaming since the upstream direction from a remote site to the broadcast studio is the crucial audio path.

Bottleneck: The uplink direction of low-key DSL lines

2. Know the Problem

Modern audio codecs like the ZephyrIP use fairly low audio bitrates in the range of 50 to 100 kBit/sec. A low-key Internet connection can handle this easily even in the upstream direction – as long as no other upstream traffic is interfering with the audio stream.

TCP uploads (FTP, email, JPG uploads) can instantly occupy all of the available upstream bandwidth. TCP's slow start and congestion avoidance scheme steadily increases throughput until it exceeds available bandwidth and the routers start to drop packets. As soon as packets are lost the TCP connection will reduce its throughput and begin to increase it again in an endless cycle.

Problem: Concurrent upstream TCP traffic

The router drops packets randomly which not only affect the TCP packets but also the UDP/RTP audio packets. As a result, audio dropouts are audible on the receiver side.

Worse than that, before actually dropping packets the router will buffer them up in an effort to deliver them later. This affects TCP as well as UDP/RTP packets in the same

way. A buffered-up audio packet which arrives at the receiver when its playout buffer has already been drained completely is essentially lost. Once again, audio dropouts are audible.

This buffering has been known to cause up to 4 seconds worth of audio delay. If the decoder playout buffer holds this amount of audio no dropout will occur. But of course this amount of end-to-end audio delay is usually unacceptable for two-way audio communication.

“Jitter” is also a factor affecting connections. In IP parlance, “jitter” refers to variations in the time between arriving data packets. Experience has shown that DSL connections tend to have less jitter than cable modem connections as a result of differing network management mechanisms. It seems counter-intuitive, but the fastest connection *may not* give the best performance. Informally, we suggest running this online jitter test prior to settling on a connection; it has proven quite accurate in predicting the success of a given link: http://myspeed.visualware.com/indexvoip.php .

Problem: Packet buffering

3. Solutions

3.1 Own the Link

The most simple solution is to avoid all TCP cross traffic. It is by far better to order an inexpensive, low-key DSL connection exclusively for your ZephyrIP rather than sharing a highspeed connection with many other users.

Solution: Own the Link

3.2 Apply QoS (Quality of Service)

In situations where you have control over the DSL-router (in the studio) try to turn on QoS settings. Modern routers can give priority to protocols (RTP) or IP numbers and ports. If your router does not support any of these features consider buying a new one.

Solution: Use QoS

3.3 Control the Traffic

If you have no control over the router and must share an Internet connection with other users you can introduce semi-QoS by inserting a QoS router between the network devices and the DSL router. This intermediate router must be programmed to give priority to your audio stream so no other users may circumvent your QoS box by connecting directly to the DSL router. Easy to use devices which work as intermediate routers are available off

the shelf. The Hawking HBB1 Broadband Booster is a good example. (www.hawkingtech.com)

Solution: Control all Traffic

3.4 Reduce Delay

Throughput delay is an issue because too-late audio packets are no better than lost packets. The maximum delay that a router can impose on the IP packets depends on its internal buffer size and the speed of the Internet uplink. If you cannot implement any of the above solutions you should order an Internet connection with a high uplink speed. Be aware that if you have to share this link with other users you may still encounter lost packets. But at least the packets won't be delayed as much.

Solution: Order high uplink bandwidth

3.5 Allow Delay

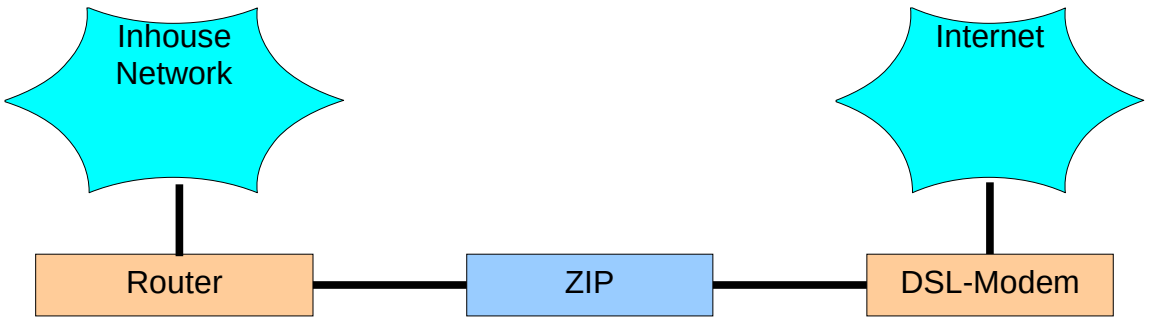
Some audio transmission applications do not require low audio delay. If you are feeding audio from a remote site to the broadcast studio, the talk-back channel from the studio should be used minimally. It is more beneficial to allow a high end-to-end delay and abandon the fast, telephone-like behavior of your audio link even if this hampers the communication with your peer.

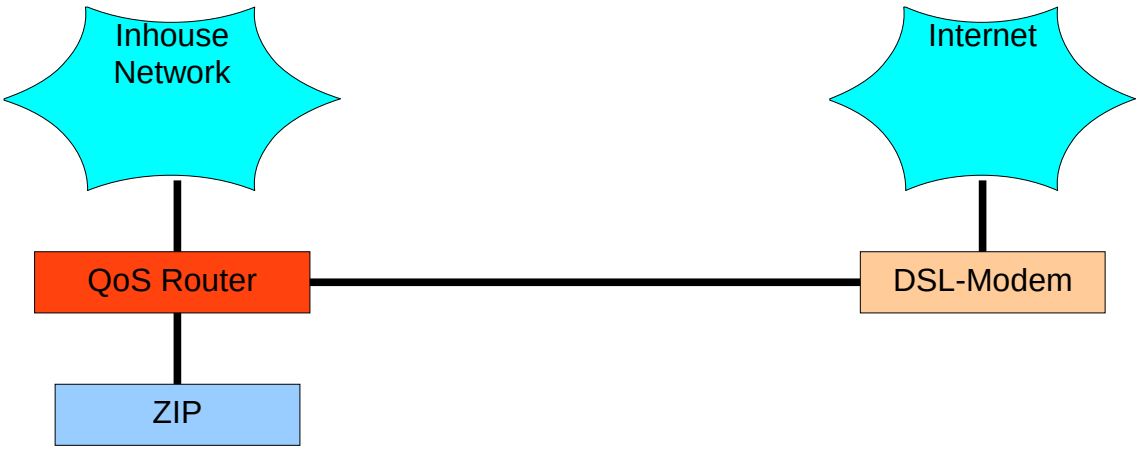
The buffering takes place in the decoder unit which receives the audio. Set its decoder buffer to the maximum which is acceptable for your application.

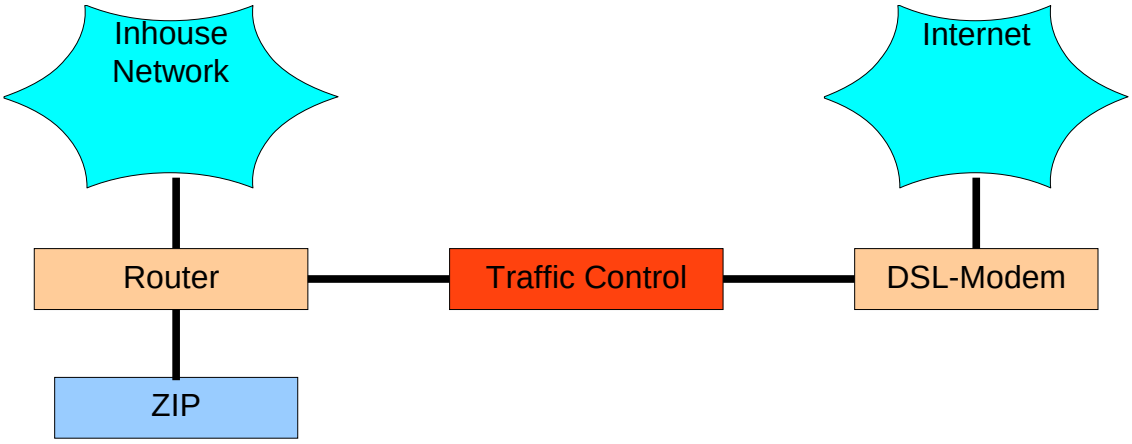
Solution: Increase the decoder buffer

4. Configuration Examples

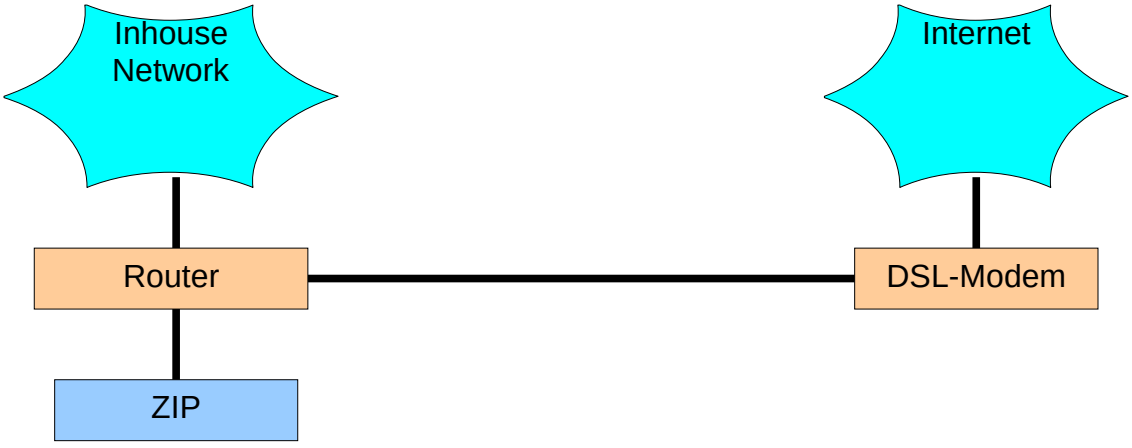
4.1 Good Setups







4.2 Setup Which Won't Work



Happy streaming! If you have further questions, please contact Telos Support at +1.216.241.7225, or support@telos-systems.com .