# Suprima Family
# Jitter Buffer Settings

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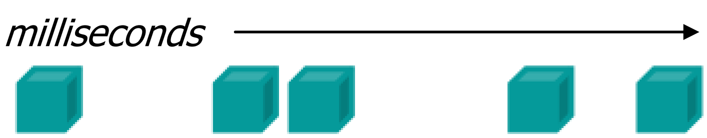
## Audio Over IP

Most of the digital audio communications over the last years have been achieved on synchronous Networks, where a common clock is provided at both ends of the connection, so that it is possible to synchronize encoders and decoders.

Nowadays IP networks are increasingly taking over the transport of the media due myriads of reason like cost efficiency, means of access almost universally available, stationary as well as mobile applications, very high thru-put for new media applications, and so on.

Homogeneous delayed audio packets



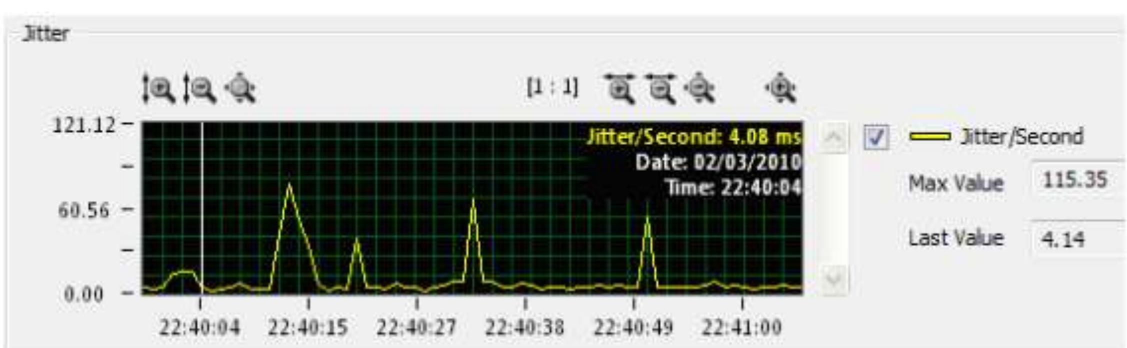

Variable delay means "jitter"

However IP networks are not synchronous anymore. This means the decoders must conciliate bursts of incoming media data with a precisely and continuous audio output. This is achieved buffering the ingress data before giving out the decoded audio signal. A receiver buffer, jitter buffer, provides this adaptation.

## The Jitter Buffer

For a perfect adaptation the *jitter buffer* should be set at least slightly larger than the largest jitter (ms) expected. Insufficient buffering would lead to audible audio interruptions.

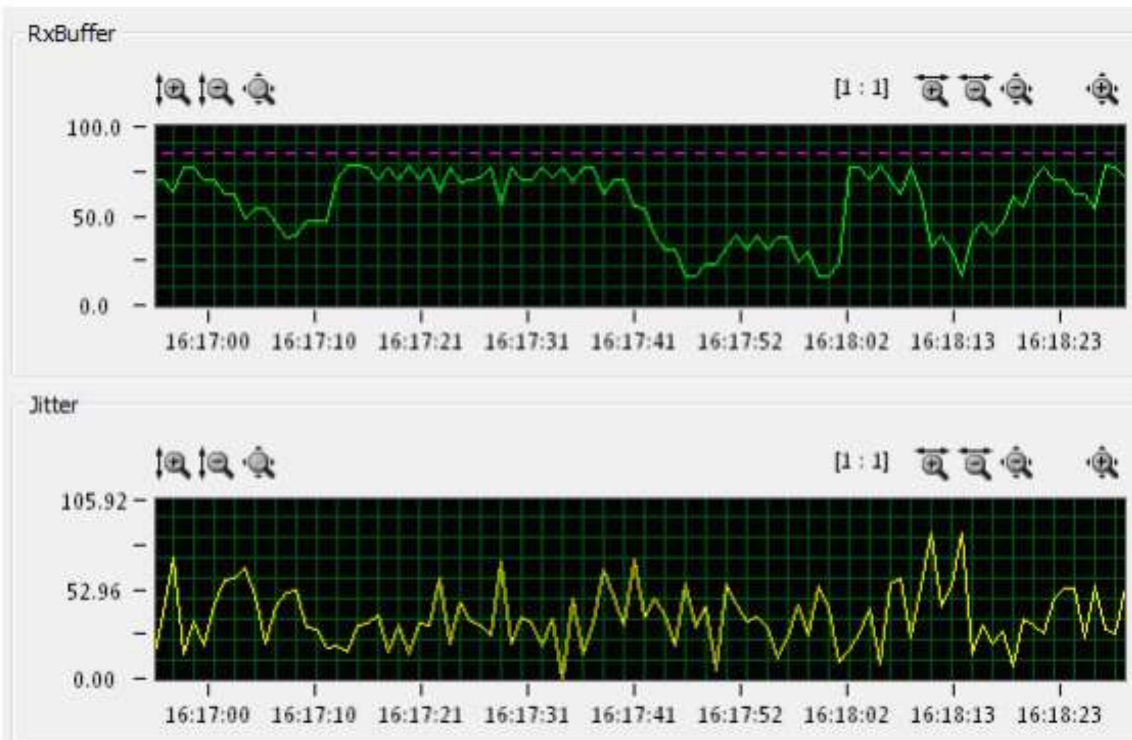
Given the following jitter example a 120~1000ms buffer should work it out.



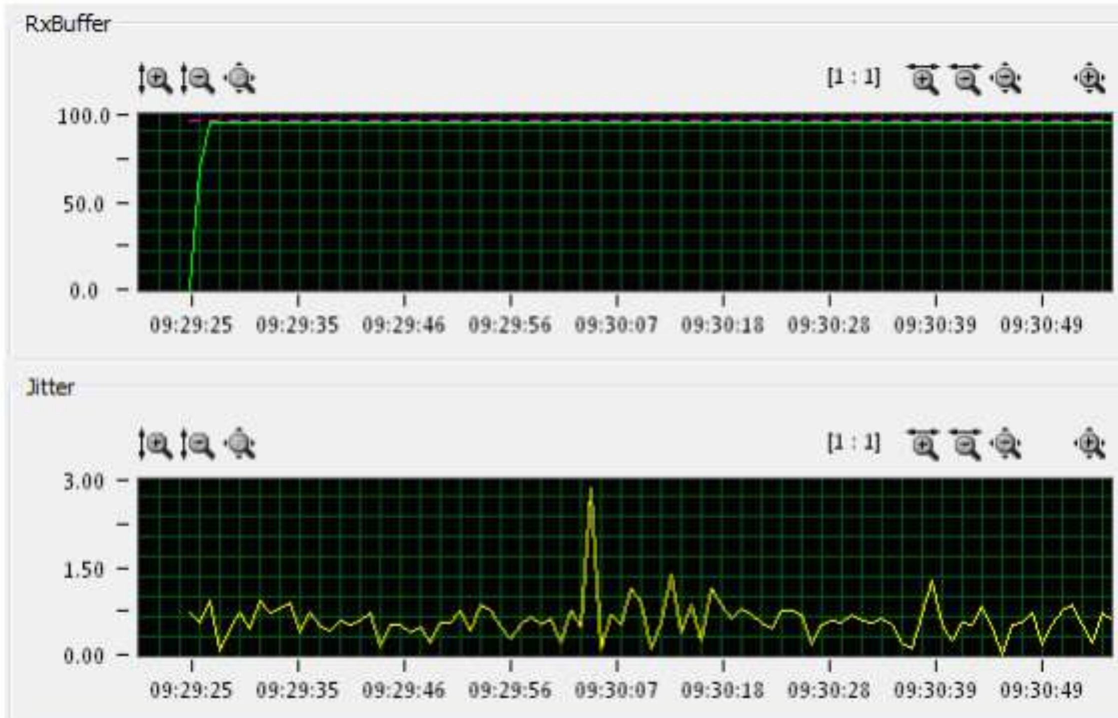
## Low Delay Streaming

Caution is advised for Low Delay Streaming applications. Buffering implies introduction of some amount of delay. Therefore the optimal Jitter Buffer definition is to be just enough for 99.9% of the time.

In the figure below it is depicted the jitter buffer adjustment <RXbuffer> for an optimal low delay application. The filling level of the buffer varies between the minimum and maximum, but without underflowing.



The following example depicts a 1000ms buffer; which is perfect for a STL application, but is oversized for a low delay application.

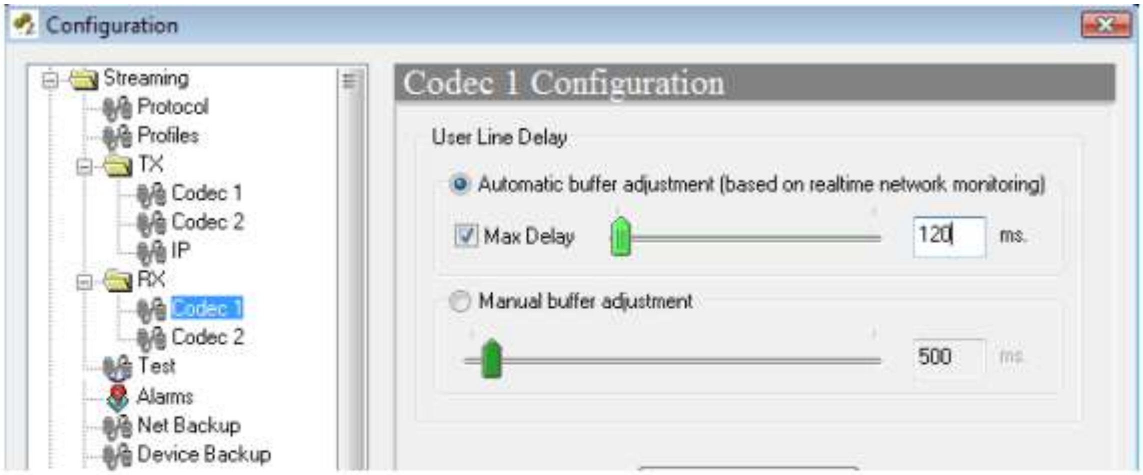



## Automated Buffer Configuration

The Suprima Family of codecs provides an automated buffer configuration. An adapting buffer increases the initial depth until the best fit is met for the ongoing IP stream.

This process aims to have the lowest possible jitter buffer. The user is not required to gain experience of the current network performance for 15 ~ 30 minutes forehand.

Optionally the user can let the buffer shift up to a reasonable limit for the low delay application. Please refer to this method on the configuration menu.
Since Suprima Family IP codecs offer the <double codec feature>, the automated buffer configuration is independent, as well for each streaming connection.




Although jitter is completely random for a short period of observation, its performance usually meets a pattern over a medium and long term. This is: the last jitter peak will happen in few seconds /minutes again.


Due the fact that peaks are followed by peaks, it hardly makes sense to shorten the jitter buffer after each jitter peak. A continuous variation of the audio delay is obviously more disturbing for the listener. Therefore the automated buffer is defined as a learning mechanism for every new initiated IP stream.