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Title: A few comments on clocking issue
Purpose: Discussion

Summary: Document AVC-1024 gives an overview on clocking issues which need further consideration.This document is intended to provide some information in order to progress on this topic.

1. Timing and clocks for terminals or end points in point to point connections

Timing in terminals depends on network clocks specifications such as frequency range of the sending and received clocks, jitter value, delay in the terminals, price of components and coder, decoder requirements. These coder/decoder requirements might have a strong impact for interworking. For instance, a high performance audio codec such as MPEG codec assumes low jitter of the sampling clock in order to get a high signal to noise ratio whereas G.711 codec doesn't need so tight tolerances. A few cases are described in this point.

I-1 Terminals connected to a synchronous network

In the Narrow Band ISDN, the clock received from the network is used to provide timing to the demultiplexer and to the decoder as well as to the encoder and the multiplexer. This can be made directly (i.e. for PCM) or by using a Phase Lock Loop (PLL) associated to a buffer depending on the requirements for the coders and the decoders.
The network operator synchronise alle the switches and cross-connect equipments to a single clock whose frequency tolerance is better than 10^{-11}. In order to allow some jitter in the transmission path, buffers are implemented in the network. In case of overflow of these buffers, phase shifts occur and bytes are lost on B channels. The jitter figures of the 1544 kbit/s and 2048 kbit/s are specified in the Recommandations G.824 and G.823 respectively. This phase shifts generally lead to impairements in the Videoconference services such as audio clicks and colored blocks which can be partially hidden to the customer by terminal design.

I-2 Terminals connected to an asynchronous network

If the terminals or end points are not intended to work for real time such as audio or video, the implementation is rather straightforward since no special timing is required. It can be assmed that Fcx, Fdy and Fcd,Fdx clocks are different. No clock recovery is needed.
Real time applications such as audio and video are not designed in the same way since the requirements are more severe.

I-3 Audiovisual terminals connected to an asynchronous network

A conventionnel design af an audiovisual terminal assumes that the receiving timing is the "same" that sending timing in order that the decoder processes the reciprocal function of the encoder. In other words, the decoder timing should be aligned to the encoder one with the exception of transmission features such as delay and jitter. So a clock recovery mechanism is one way for timing implementation in the terminals. The provision of Time Stamps in the H.310 and ~~H.320~~ Reommendations was made for this purpose.
For the audio, one could think of another approach by allowing some dropping or inserting of linear samples if the receiving buffer overflows or underflows. For the video, the flying wheel mechanism is often used by software implementations.

These two assumptions are made implicitely in document AVC-1024. For the point to point case, the use of the the combined methods should allow normal operation over a wide range of sending clocks.

I-4 Timing requirements for an audio codec

If no recovery clock mechanism is used, the dropping/insertion rate of linear audio samples depends on the buffer size and on the difference between sending and receiving clocks.A few examples are given for PCM and various cristal frequency tolerances with a receiving buffer size greater than 1 sample:

Cristal tolerance (ppm)	dropping/insertion rate
1	62;5 s
10	6;25 s
50	1.25s

Dropping or inserting more linear samples may be achieved but more signal processing is needed to avoid signal distorsion.

By using the clock recovery mechanism, there is no longer a need to drop or insert audio samples. However, the jitter amplitude may be a bigger constraint. In fact a high performance audio design, such as MPEG audio, relies on jitter amplitudes as low as a few picoseconds at the sampling frequency in order to meet the 90 dB SNR. This means that the jitter rejection of the recovered clock will be very severe since jitter amplitudes are high for ATM networks or LANs.

I-5 Timing requirements for a video codec

Since the requested video SNR is not as high as for the audio, the tolerance for the jitter on the frequency sampling is not so tight. On the receiving side, the flying wheel mechanism allows for large tolerances on jitter.

II Audiovisual terminals in multipoint

Some clock control is defined in H.245 to request terminals to determine master timing. MCUs will request to be master for the clock. Time stamps sent by the MCU will be sent by reference to a very precise clock derived from a synchronous netwok. If all the terminals involved in the conference are using only the clock recovery mechanism for timing at the sending and receiving sides, i.e. the sampling frequency, the audio mixer doesn't need to drop/insert extra linear samples in processing the audio. If one subset of terminals uses drop/insert mechanism without the recovery clock, the audio mixer will have to work in the same way before mixing the audio. The cumulative degradations should be analysed by audio experts.

As far as video is concerned, the H.231/H.243 MCU broadcast the New Speaker picture to all participants except to the souce location. This location receives the Previous Speaker picture. This can be achieved easily if all the terminals use the recovery clock mechanism. Continuous presence multipoint relies on a common timing for video signals in order to merge pictures in a simple way. Other configurations might lead to extra delays due to encoding/decoding process in the MCU or to extra costs if receiving terminals contain one decoder for each participating terminal.

III Gateway for Audiovisual terminals between NB-ISDN, LB-ISDN

Recommendation I.363 and I.580 state that this topic is for further study. Timing and jitter need further considerations for audiovisual terminals in order to provide a consistent set of Recommendations dealing with the network and terminal aspects.

IV Conclusion

This document is pointing out jitter and timing considerations for audiovisual terminals. In particular, audio quality requirements need special attention as far as Signal to Noise ratio. Even if further work need to be achieved, it is shown that these topics have a strong impact on gateways and MCUs which are to be standardised in the new Study Period.