CCITT Study Group XV
Working Party XV/1
Specialists Group on Coding for Visual Telephony

Doc #169
November 1986

Source: UK

Title: Some Questions about m*64kbit/s and its Relationship with n*384kbit/s coding.

This contribution contains no proposals, but is intended to stimulate discussion in the general area of video coding at m*64kbit/s. Some questions are raised, and answers given, though it is not claimed that these are complete or definitive.

1. What range of values of m is intended to be applied to m*64kbit/s?

An important issue is whether m=1 is included or not. ISDN networks are emerging which will offer two 64kbit/s channels as standard. Nevertheless, there may be a requirement for video plus sound in a single 64kbit/s.

Even with m=2 there is uncertainty whether the two channels will have equal delay through some networks. If not, there may be problems in using more than one 64kbit/s for video.

Values of m in the range 3 to 5 do not seem to be discussed much, probably because they do not match transmission options.

So, currently it seems that m*64kbit/s could include for example

48kbit/s video + 16kbit/s audio

64kbit/s video + 64kbit/s audio

96kbit/s video + 32kbit/s audio

among a variety of possible combinations

In the remainder of this document the terms "64kbit/s" and "384kbit/s" are used to denote m*64kbit/s and n*384kbit/s. They are not intended to imply m=1 and n=1 only.

2. What is the target for 64kbit/s in a market and technical sense?

Ideally the market-place would like an inexpensive unit giving broadcast quality moving pictures plus wideband audio in 64kbit/s. This is unlikely to be possible for some time, if ever, and current approaches tackle the question from two different ends of the spectrum:

1. The complexity (price) versus quality compromise is biased towards getting the best possible picture quality, resulting in equipment aimed at the business market, initially at the higher end of it.

2. A different compromise is reached where price is the dominant factor. Here, picture quality is reduced to what is thought to be the minimum acceptable. The resulting codecs would certainly be attractive at the lower end of the business market right from the start, and the residential sector also though perhaps not immediately.

So, there may be two 64kbit/s strategies. If so, we need to consider interconnectability between the two of them as well as their relationship to 384kbit/s.

3. Is interconnectability between 64kbit/s codecs and 384kbit/s codecs wanted?

It is difficult to argue against it being a laudable aim. However, the costs incurred by doing it and the resulting quality must be weighed against its desirability. This cannot be done unless the means to achieve it are investigated.

4. What is needed for interconnectability?

Interconnectability can be separated into three topics:

1. Bit transport. This concerns the interconnection of the bit streams from the two codecs and knowing which bits are which at the receiving end. In principle there are no serious problems here. Obviously if a 384kbit/s coder is connected to a 64kbit/s decoder then only a fraction of its output bit-stream will be received. Either the coder must be made aware of this so that it will not insert necessary data in the part which will be discarded, or the coding scheme must be tolerant of the loss of bits. (See 3.1 below). In either case there must be network equipment which transfers the correct portion of the 384kbit/s to a 64kbit/s channel. Similarly data from a 64kbit/s coder must be inserted into the higher rate channel in the place where the 384kbit/s decoder expects to find it.

Since some transfer equipment is necessary, it could in principle also perform any reformatting of the data arising from the adoption of different framing structures.

2. Audio coding. This should be relatively straightforward. The same coding algorithm should be used at both transmission rates, unless there are very strong reasons for not doing so, for example only 16kbit/s available for audio. However, even if different audio bit-rates and coding algorithms are used at 64kbit/s and 384kbit/s, it would be possible to convert between them in PCM form at some point in the network. Alternatively, since the audio is likely to be a minor part of the 384kbit/s codec and since the audio bit-rate in the 64kbit/s device is unlikely to be higher than for the 384kbit/s one, the latter could include both types of audio processing.

3. Video coding. This is the crux of the matter.

5. How might video coding compatibility be achieved?

The exact method will of course depend on the coding schemes that are chosen for each bit-rate. To date, work on 384kbit/s has not considered compatibility with future 64kbit/s to be an over-riding factor. Possible options now are:

1. To urgently consider 64kbit/s and see if it would be possible to have one coding scheme which could be used at 64kbit/s and 384kbit/s (and by implication up to 2Mbit/s). It is also possible to envisage a scheme of a basic coder operating at 64kbit/s with extra channel capacity, when available, being used by auxilary coders to augment the performance. This would give the sort of automatic compatibility that is present in FM stereo radio and colour television.

2. To continue in the current vein with 384kbit/s and subsequently develop one, or more in the light of the first and second questions above, 64kbit/s algorithms under the constraints of compatibility with the 384kbit/s scheme, which involve minimal changes to 384kbit/s codecs.

3. To develop 64kbit/s algorithm(s) without being constrained by compatibility. Interconnectability would then be arranged by:

a gateway device, preferably transcoding at a digital level, but in the worst case at analogue. It must be borne in mind that the closer the interface level approaches direct video the higher the processing delay is likely to be.

Or:

by adopting an additional coding scheme which was a compromise between the 64kbit/s and 384kbit/s methods. This would entail switching of the operation of both types of codec on interconnection calls.

Or:

by having both a 64kbit/s and 384kbit/s unit available at the "384 end", either as two separate units or integrated. In effect this amounts to a dual mode 384kbit/s device.

Or:

By making the 64kbit/s device dual mode. The primary mode would be optimised for operation to another 64kbit/s codec. The secondary mode would be the 384kbit/s algorithm wound down to 64kbit/s.

In reality there are grey areas and the solution may be a combination of some of the above.

6. What about graphics at 64kbit/s?

This will probably be wanted, at least in the "high end" 64kbit/s codec. It could be provided in several ways:

1. If the moving picture mode has sufficient attainable resolution, then it may be used for graphics, in a similar manner to the philosophy adopted at 384kbit/s.

2. If the moving picture mode has reduced resolution, then the moving picture algorithm could be applied sequentially to parts of a higher sampling density picture input.

3. By employing any graphics mode emanating from audiographic teleconferencing standardisation studies.

Conclusions

Free discussion on these and related issues should be held during the time allocated at the Specialists Group meeting. The aim of the group should be to produce an outline recommendation for m*64kbit/s to include such items as picture format, framing structure etc, not all the details of the coding algorithm, but perhaps a fundamental structure suitable for programmable hardware, by the end of the current plenary. A full recommendation could then be produced, perhaps using the accelerated procedure, sometime in 1989/90.

End