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

Document #75
March 1986

SOURCE: CHAIRMAN OF THE SPECIALISTS GROUP ON CODING FOR VISUAL TELEPHONY

TITLE : EXTRACTS FROM THE REPORT OF THE WORKING PARTY XV/1 MEETING HELD IN GENEVA (24, 25 and 27 February 1986)

I. Question 4/XV: Visual Telephone Service

2. Discussion and agreements

2.1 Report of the activities of the Specialists Group on Coding for Visual Telephony

Chairman of the Specialist Group reported on their activities covering the two meetings held in September 1985 and January 1986, as detailed in ~~Temporary Document 7~~ Annex 1. These activities focused on the ~~384 × n~~ n × 384 kbit/s (n = 1 - 5) codec, in particular, basic parameters, coding algorithm and frame structure. Working Party XV/1 approved this report thus encouraging further progress in the future.

Working Party XV/1 wishes to advise Study Group XVII that the Recommendation for the nx 384 Kbit/s codecs is expected to be finalized by the end of this study period. Thus, Study Group XVII is requested to provide all the necessary means including nx 384 kbit/s switched connections for the smooth expansion of the video conferencing services.

Note to Secretariat : This section should be sent to SG XVII.

2.2 Amendments to existing Recommendations

2.3 Video conferencing system parameters

Parameters necessary for the construction of a compatible video conferencing system were identified.

During the discussion, the point was raised as to whether G.721 could be applied for video conferencing.

The meeting was of the opinion that for video conferencing a higher qualify for audio is required and although there may be progress in the voice coding technology in the future, it was decided not to have G.721 in the list of parameters.

Moreover, a small group chaired by Mr.P.Bryan (AT&T,USA) was charged with the task of identifying essential versus optional for each parameter.

The list of essential/optional functions and parameters for video conferencing together with the guidelines for selecting essential and optional functions are contained in Annex 7 to this report.

2.4 Multipoint video conferencing

D.87(U.K.), D.88(Netherlands,BT,France,Italy) and D.95(NTT)/XV were addressed to the subject of multipoint video conferencing.

It was considered useful to draft an initial paper dealing with the multi-point network configurations and the functional requirements of the multi-point control units(MCU) by merging the information contained in these contributions so as to make it serve as a basis for further studies.

The paper is annexed as Annex 8 to this report.

During the discussion it was pointed out that the use of such transmission media with long delay as satellites might impose burden on the control functions and consideration should be given to this fact when studying the network configuration and the functions for the MCUs.

In the contributions two abbreviations were used for the multipoint control unit, ie, MCU and MPU, the latter being used in the text of Question 5/XV. However, the meeting found it more common to use MCU at present and decided on the exclusive usage of MCU.

2.5 Reservation systems

2.6 Higher bit rate codecs

2.7 Review of the report of task group on broadband aspects of ISDN

II. Question 5/XV: Multifacility Services

3. Draft List of Recommendations

The case for an infrastructure approach covering all audiovisual services was presented as in Document TD.6. This was endorsed by WP XV/1 and the substance of the document is appended to this Report as Annex 1*. The importance was highlighted of establishing, at an early stage, the principles of interconnecting terminals of different types, i.e. intended mainly for different audiovisual services. A start had been made in this direction by the rapporteurs' liaison, for example by proposing that all 64/56/48 kbit/s wideband audio codecs should also be capable of 64/56 kbit/s PCM operation. Further work would be needed to determines compatibilities between still-moving-picture codecs.

After some refinement the List of Recommendations proposed in TD.17 was endorsed as given in Annex 2; the headings of contents of the Recommendations swerve as explanatory indications only, and require further study.

It was strongly suggested that the complete set of Recommendations should form a new Series Y, rather than be dispersed through many series having a totally different compass.

Some items in the list could already be identified as achieved or in progress:

110: TD.2 recognized the need for different teleconferencing services, and a draft covering Recommendation has begun (TD.18);

221: D.88/XV contains elements for the frame structure Recommendation, as do also D.72/XV and TD.10;

241: TD.10 can be used as a starting point;

331: will cross-reference G.711, G.721, G.72X;

332: the Recommendation H.120 is relevant; moreover, the video codec specialist group of Q.4/XV will provide Recommendations suitable for 320 kbit/s and 64 kbit/s;

210: H.110 applies to videoconference; generalization has begun (D.87 and 95/XV).

4. Frame structures

At an early stage in the work of SG XVIII on coding of 7 kHz bandwidth audio into 64 kbit/s, other groups concerned with audiovisual service development (notably audio- and telematic-teleconferencing) had requested development of the algorithm if possible to operate also at 48/56 kbit/s, in order to leave capacity within a 64 kbit/s clear channel for lower bit rate meeting aids and control/indication signals. When it became apparent that

* Note to Secretariat - Annex 1 reproduces TD.6

such an objective was realistic, some organizations set about the development of a frame structure which could be used in general on such 64 kbit/s paths, and which in particular would adopt G.72X in its 48/56 kbit/s form for wideband speech. This frame structure is now presented in D.88/XV, and its use for 384 kbit/s videoconference transmission is proposed in D.72/XV.

Study Group XVIII wishes to use the accelerated procedure for formalization of G.72X, and since there has as yet been no formal indication elsewhere as to the purpose of the 48 and 56 kbit/s alternatives therein, an Annex A containing system aspects is intended. Since the content of such an Annex is the concern of several other Study Groups (as indicated in Figure 2 of TD.3) a liaison meeting of rapporteurs from these Study Groups was set up (Paris, 14-16 January 1986). The work of this body was used as a basis for drafting the above mentioned Annex A (TD.10).

Working Party XV/1 noted that the accelerated procedure would be satisfactory for the audio coding algorithm of G.72X itself, since the method had been thoroughly tested and its quality of performance measured. However, it would be premature to give Annex A the full force of a Recommendation at this time: work is currently in progress in several countries which could verify the complex procedures of Type Identification and Mode Initialization in time for Recommendation by the normal procedure. In view of the adoption by WPXV/1 of the List of Recommendations as in Annex 2 to this Report, it could be suggested to WP XVIII/8 that the difficulty could be overcome by cross-reference (in Annex A to G.72X) to Recommendations 221 and 241 of this list.

Annex 1 (Q.4/XV)

4-XV/1

CCITT
Study Group XV
Geneva, 24 February - 7 March 1986

Temporary Document - E

Question: 4/XV

SOURCE: CHAIRMAN OF SPECIALISTS GROUP ON CODING FOR VISUAL TELEPHONY

TITLE : PROGRESS REPORT ON THE ACTIVITIES OF THE SPECIALISTS GROUP

CONTENTS

1. General

The Specialists Group on Coding for Visual Telephony started its activities in December 1984 and presented the first progress report to the Working Party XV/1 last July (see Annex 1 to COM XV-R 4), stating the conclusion on the Part 3 codec Recommendations (H.120, H.130) and the study on the sub-primary rate codec. The Group decided to concentrate on the standardization on the 384 x n kbit/s codec for the time being.

Since July 1985, we met twice, from 24 to 27 September 1985 in Torino (Italy) and from 21 to 24 January 1986 in Ipswich (U.K.), and reviewed 42 documents including two meeting reports. The list of participants at the two meetings appears at the end of this report.

This document reports major conclusions and the present status of the study.

2. Basic Parameters

Basic parameters concerning the digital video signal format are defined at the input of the video source coder. The following two approaches to ensure direct connectivity between 625/50 and 525/60 regions were compared:

Single approach - To define 'intermediate format' which are common to both 625/50 and 525/60 television systems. Necessary standards conversion is carried out at both the transmitting and receiving sides.

Dual approach - To define two parameters according to the local television systems. Necessary standards conversion is carried out at the receiving side.

After extensive discussion, it was agreed to adopt the single mode approach with parameters shown in Table 1, where the active picture area obtained by a camera with local television standards is covered by the single intermediate format (Notes 1, 2). Pre- and post-processing to/from 360 x 288 intermediate format is carried out in such a way that a 4:3 aspect ratio is maintained throughout.

Note 1: The U.S.A. is of the opinion that the dual mode approach is more practical, however in the interests of proceeding with an international agreement the single intermediate format is acceptable.

Note 2: Italy agreed to the decision but expressed concern over the service degradation caused by an unpreventable interregional connection in which two compatible though non-standard codecs are employed each using the nominal 288 lines but one with 240 active lines 29.97 fields/s and the other with 288 lines 29.97 fields/s.

As for the graphics facility to provide full resolution still pictures in the form of a normal television signal, we reached the following agreements.

(1) Graphic mode parameters are not defined independently of moving picture mode parameters in the 384 x n kbit/s codec.

(2) The coding algorithm should, for all bit rates, inherently permit the full spatial resolution (luminance and chrominance) of the common intermediate format to be realized when the incoming video comprises pictures having little movement.

(3) In the future, higher resolution graphics facility may be defined to utilize a part (e.g. 64 kbit/s) or the whole of the video signal time slot.

3. Coding Algorithm

3.1 Discussion at the Torino meeting

Various algorithms were presented with documents and video tape demonstrations. These gave the participants better understanding on the present state of the art. It was felt that some more time is necessary to get performance improvement for each candidate. After some discussion, the followings were agreed.

(1) Various coding algorithms proposed as candidates for the new generation codec in this meeting are classified into the following two categories;

Category 1: Interframe prediction + further processing
- Prediction/interpolation
- DCT
- Vector quantization

Category 2: Sub-band coding
- Anthropomorfic transform
- Pyramid transform/conditional replenishment

(2) General view of the Group is that future study should be focused on Category 1.

(3) Since the Group is planning to carry out international field trials early 1987 to get a recommendation during this study period, we are required to specify the new generation codec early 1986. Hence contributions on coding schemes different from those of Category 1 should be presented not after the next meeting.

The Specialists Group also agreed on a number of topics which should guarantee an appropriate comparison and evaluation of source coding results.

(1) Picture material
The three sequences shown in Table 2 have been selected.

(2) Initial conditions
Frame No. 0 is mid-gray (127), where applicable the buffer is empty for frame No. 0. Simulation is started from frame No. 1. After simulation the first 8 frames of intermediate format are discarded to avoid initial transients.

(3) The Specialists Group agreed to allow around 200 msec for the total delay (encoder-decoder). The size of the buffer is still open for further consideration.

(4) Assuming that some overhead is necessary, the bitrate for simulations will be 300 kbit/s. Bitrate should be determined based on the number of bits at the output of variable length coder, but not based on entropy estimation.

(5) The simulation will be carried out in closed loop mode.

(6) Pre- and post-processing may be included in the demonstrations of simulation results provided they are described in detail.

(7) Demonstration of simulation results should include the 3 agreed sequences in color.

3.2 <u>Discussion at the Ipswich meeting</u>

After presentation of the documents, extensive questions and answers were carried out for clarification. Various demonstrations of source coding simulation results were also given.

As there were no proposals on Category 2 coding algorithms, the Specialists Group concentrates future study on Category 1 coding algorithms: 'interframe prediction + further processing'.

Since there were no proposals for coding algorithms based on 'prediction/interpolation', the Specialists Group will evaluate and compose Transform, VQ or a combination of these approaches in the future (see Figure 2). The meeting agreed to make the decision on the coding algorithm and the generic structure at the next meeting.

On other aspects of the video source coding, the following agreements were obtained.

(1) Compatibility with the 64 x m kbit/s codec
Since the possible need for compatibility between services such as videophone, audiographic conferencing using 64 x m kbit/s codecs and video conferencing using 384 x n kbit/s codecs is foreseen in the future, this matter has been discussed. Various opinions were expressed. The meeting agreed that compatibility between these services mentioned is desirable. For the moment it has been agreed to concentrate on 384 x n kbit/s codecs bearing in mind the desirable compatibility with 64 x m kbit/s codecs. It is requested that each coding algorithm proposal will contain some comments on this point.

(2) Motion compensation
It has been agreed that the motion compensation part of the coder has to be treated as an option for manufacturing, though that part of the decoder is mandatory. There is, however, a general feeling that motion compensation will be a necessity at 384 kbit/s. This is not clear for the values $n > 1$ of 384 x n kbit/s. As for the blocksize for motion compensation, the meeting expressed a general view that it should be the same as the blocksize for coding in the Transform-based scheme.

(3) DCT transform
It has been agreed to use the classical DCT for further simulation studies whereas further considerations of more easily implementable transform are required as soon as hardware has to be defined.

(4) Test material
It has been agreed to add a graphical test sequence consisting of two scenes to Table 1 in order to evaluate the performance of a coding scheme for a variety of input pictures. U. K. will provide this new test sequence.

3.3 Future study on coding algorithm

Toward the decision at the next meeting, the Specialists Group will further study video source coding algorithm on the following items.

1) Improvements of Transform-based and VQ-based algorithms
2) Blocksize
3) Adaptive quantization schemes
4) Easily implementable transform
5) Methods to process color difference signals
6) Motion compensated field interpolation
7) Transmission error resilience

4. Frame Structure

The Specialists Group supported the basic frame structure for 384 x n kbit/s videoconferencing codec proposed by France, which consists of one timeslot of audio and Service Channel and five or more timeslots of moving video, one of them capable of being dynamically allocated to data (Figure 3).

It should be noted that the same frame structure is also applied for mode switching of wideband speech coding which is being studied by Study Group XVIII for recommending G.72X.
It was confirmed that the Specialists Group is responsible to define 'Transfer Rate' attribute in 'BAS' and also some bits in the Application Channel related to videoconferencing codec-to-codec control.

5. Transmission Aspects

Time Slot Integrity and one's density restriction problems were discussed. The Specialists Group confirmed that the new generation sub-primary rate codecs be designed assuming the usage of transparent transmission channel. Measures to the restriction will be dealt with as optional exceptions.

In order to promote the videoconferencing service, it is recognized that switched HO services should be recommended. This request will be forwarded from this Group to SGXVIII through SGXV.

6. Future Work Plan

Subsequent to obtaining the fundamental coding algorithm and generic structure for the 384 x n kbit/s codec , the activities of the Specialists Group will take the following steps toward completing the recommendation at the end of this CCITT study period.

1) Specifications for prototype hardware
2) Construction of prototype hardware and parameter optimization
3) Compatibility checks at laboratories
4) International field trial
5) Drafting of Recommendations

More details will be discussed and decided in a step-by-step way. The relatively short period for hardware construction was pointed out and the full usage of available time by the end of this study period was suggested.

As for hardware construction, several countries expressed their interest to participate in compatibility check. In order to facilitate compatibility check, a necessity of having the same test pattern generator among concerned organizations was stressed. The Specialists Group solicited a volunteer to provide such test equipment, and a U.K. member offered to do so. When international field trial is carried out, the Specialists Group would seek INTELSAT's cooperation for the free use of the space segment.

Annex 1

Table 1. Basic parameters for the new generation 384 x n kbit/s CODEC

Items	Parameters
1. Reference point	Point B in Fig. 1/Annex 1 to COM XV-R 4
2. Baseband signals and their levels	Y, R-Y, B-Y, as defined in CCIR Rec. 601
3. Number of pels per line	Y: 360 (Note 1) R-Y: 180 B-Y: 180
4. Number of lines per field	Y: 288 (Note 2) R-Y: 144 B-Y: 144
5. Field frequency	Y, R-Y, B-Y: 29.97 Hz
6. Interlace	Y, R-Y, B-Y: 1:1
7. Sampling structure	Y, R-Y, B-Y: orthogonal, positioning of R-Y and B-Y samples share the same block boundaries with Y samples as shown in Fig.1

Note 1: Active line duration is approximately 53 us.

Note 2: Active field duration is approximately 18.4 ms (for 625/50 systems) and approximately 15.2 ms (for 525/60 systems).

Note 3: The common intermediate format defines the maximum attainable spatial and temporal resolution in the codec. Effective resolution may eventually be reduced by some coding operating modes.

Note 4: The common intermediate format is a logical specification to ensure compatibility among codecs. Hence, it might not appear at the physical interface points in the codec.

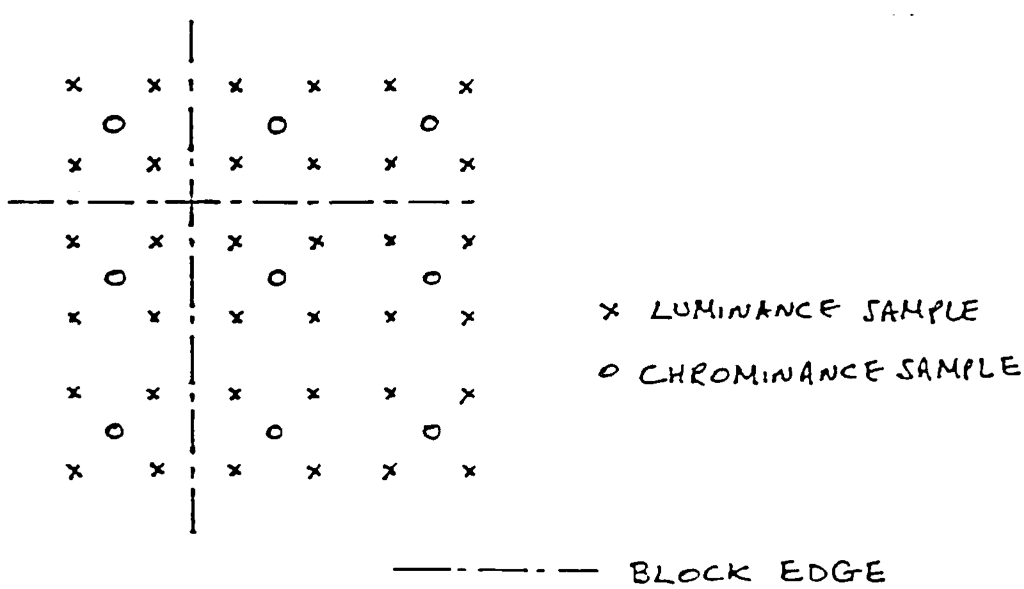
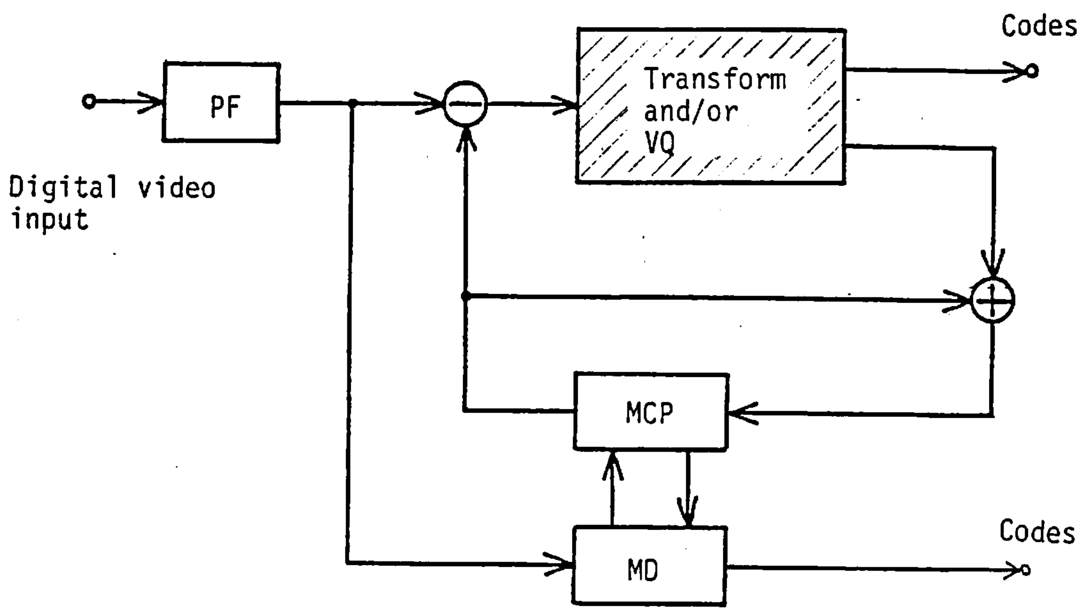


Fig. 1 Positioning of luminance and chrominance samples

Annex 1

Table 2 Test Sequences

Name	Source	Format	Length
1. "Splitscreen-Trevor"	Europe	4:2:2	5.5 sec
2. "Miss America"	U.S.A.	Intermediate format	5 sec
3. "Checked jacket"	Japan	4:2:2	2 sec



PF : Pre-filter
MCP : Motion-compensated predictor
MD : Motion vector detector
VQ : Vector quantization

Fig. 2 Generic configuration of 384 x n kbit/s codec

Annex 1

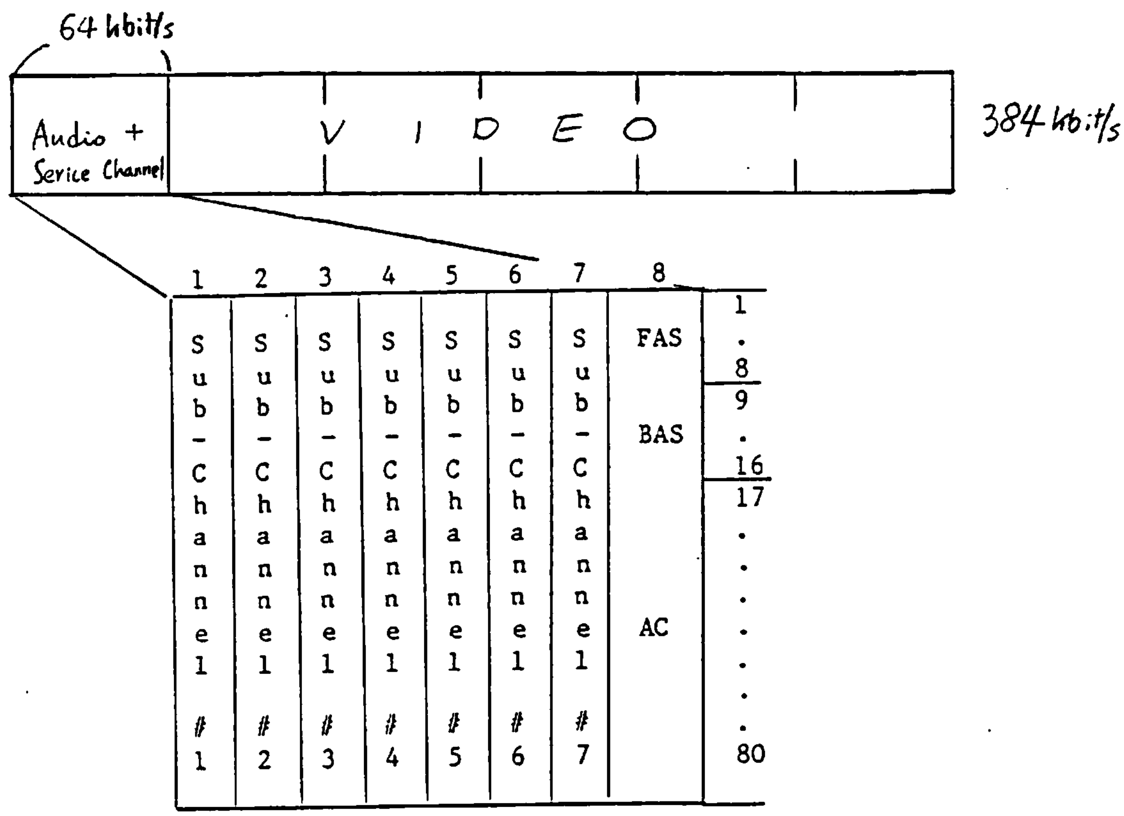


Figure 1. Frame structure

FAS : Frame Alignment Signal (note 1)
BAS : Bitrate Allocation Signal
AC : Application Channel

Note 1 : The block termed as FAS contains also other information than for frame alignment purposes.

Fig. 3 Frame structure for 384 x n kbit/s codec (in case of n = 1)

Annex 1

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(Torino, September 24 - 27, 1985)

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Annex I

LIST OF PARTICIPANTS
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Annex 7

(Q.4/XV)

Question : 4/XV

SOURCE : DRAFTING GROUP

TITLE : ANNEX 7 TO THE PRELIMINARY REPLY TO Question 4/XV - Functional aspects of video teleconferencing for n x 384 kbit/s service

Remembering that both essential and optional functions are to be specified in the Recommendation, the following guidelines are set for selecting essential and optional functions.

- Essential functions are the minimum required functions for a video teleconference.

- A proposed optional functions should be evaluated as follows.

  1) If an optional function does not require significant transmission capacity and can be accomodated when the option is not implemented without significantly increasing complexity of the essential system, then it should be included.

  2) If an optional function does require significant transmission capacity or will significantly increase the complexity of the essential system when the option is not implemented, then tradeoffs should be assessed before it is included. It may be that a function can be included for higher bitrates, but not for lower bit rates.

- Each proposed value of n for n x 384 kbit/s video teleconferencing should be considered when assessing functions, particularly, any function that may be rejected at a lower bit rate because it required excessive transmission capacity, should be considered at higher bit rates.

This Table lists functions that will be specified in Recommendations for video teleconferencing systems. The Table is not intended to be final. Its structure should be modified as the work on Questions 4/XV and 5/XV progresses. Also, it is believed that this framework should be used for other purposes, such as :

- establishing the priority of the standards work on the system specification

- developing these aspects for other bit rates

- harmonizing with the evolving aspects of ISDN.

The Table lists functional requirements and then indicates whether they are essential, optional, or under study for video teleconferencing systems. It is intended that both essential and optional functions be completely specified to ensure interoperability. Under study items may be included after further study.

Essential functions are the minimum required functions for a video teleconference. Optional functions will be specified, but it will be the manufacturers and users decision to decide which options to implement and use. This does not mean that the options are not important only that different users may consider different options to be important.

TABLE

1.0	Video functions		
1.1	Full motion (nx384-64 kbit/s) 64 kbit/s is for audio, control and a data channel		- Essential
1.2	Video standard		
	1.2.1	NTSC	- Essential } according to local standards
	1.2.2	PAL / SECAM	- Essential } according to local standards
	1.2.3	RGB - 525 line	- Optional
	1.2.4	RGB - 625 line	- Optional
	1.2.5	MAC (HDTV) CCIR Rec. 604, T101	- Under study
1.3	Picture composition options		
	1.3.1	Single picture	- Essential
	1.3.2	Split screen	- Optional
	1.3.3	Four pictures	- Optional
1.4	Field/Frame interleaving		- Under study
1.5	Graphic~~al~~ mode		
	1.5.1	Standard TV resolution	- Optional
	1.5.2	Pointer	- Optional
	1.5.3	Graphics camera indicator	- Optional
1.6	Remote camera selection		- Under study
2.0	Audio functions		
2.1	Audio standard		
	2.1.1	G.72x	- Essential
2.2	Audio add-on (separately bridged)		
	2.2.1	Point-to-point add-on	- Optional
	2.2.2	Broadcast add-on	- Optional
	2.2.3	Interactive multipoint add-on	- Optional
2.3	Remote control of person speaking ~~KTS~~		- Optional
2.4	Echo feedback control (there are various technical solutions)		- Essential
2.5	Audio source		
	2.5.1	Monaural	- Essential
	2.5.2	Stereo	- Optional
	2.5.3	Bilingual	- Optional

3.0 Data functions

3.1 Data channels

3.1.1 n x 64 kbit/s channels - derived from video capacity - Optional

3.1.2 Capacity derived from audio channel - Optional

3.2 Visual aid functions

3.2.1 Telewriter/cursor - Optional

3.2.2 Facsimile - Optional

3.2.3 Mixed mode-teletext - Optional

3.2.4 Videotex - Optional

3.2.5 Still picture - Optional

3.2.6 Data file - Optional

3.2.7 Personal computer - Optional

4.0 Multipoint functions (more than two locations) - Optional

4.1 One sender

4.1.1 Broadcast, fixed sender - Essential for multipoint

4.1.2 Broadcast, alternating sender - Under study

4.2 Two senders

4.2.1 Two way interactive with broadcast of one or both ends to other locations - Under study

4.3 Multiple senders

4.3.1 Continuous presence - all participants displayed simultaneously - Optional for multipoint

4.3.2 Switched video - switched video can be achieved by audio level, by host or control, and/or by user selection. The methods have to be studied - Essential for multipoint

5.0 Encryption functions

5.1 Encryption standard - Optional

5.2 Key administration

5.2.1 Key transmitted over service channel - Optional

5.2.2 Key Passed separately - Optional

6.0 Maintenance functions *)

6.1	Digital loop back toward network	- Essential
6.2	Digital loop back toward codec	- Essential
6.3	Analogue loop back toward network	- Essential
6.4	Test signal generation	- Optional

7.0 Other functions - These items are essential ~~by~~ but still under study

7.1	Refresh due to transmission error
7.2	No input to coder indication
7.3	Codec facility indicator
7.4	Frame memory clear

*) Administrations are asked to provide information concerning the maintenance of video teleconferencing systems and the relation between the maintenance of the network and the maintenance of the video teleconferencing system. Further, this point should be brought to the attention of SG IV, SG XVIII and WP XV/2.

main body ^.

ANNEX 8 (Q.4/XV)

Framework for multipoint videoconferencing system study

1. Definition

In a multipoint videoconferencing where more than two terminals are involved, interconnection of these terminals is performed through one or more Multipoint Control Units (MCUs). An MCU is a piece of equipment located at a node of a digital network which receives several videoconferencing channels from network access ports. Each access port is connected to corresponding videoconferencing terminals or to another MCU. The purpose of the MCU is to permit the transmission of all the signals including coded audio and video information and control/indication information among a number of separated videoconferencing terminals.

2. Referrence Network Configuration

A basic reference network configuration is shown in Figure 1, while the extension to the case with two MCUs included is shown in Figure 2.

In these configurations, application of the following items should be clarified from the viewpoint of quality;

a. Satellite circuit
b. Codec cascade connection

3. Items to be Satandardized

1. Types of multipoint videoconferencing services

1.1 One sender
1.1.1 Broadcast fixed sender
1.1.2 Broadcast alternating sender
1.2 Two senders
~~1.2.1 Two-way interactive (Note: identical with traditional two-way videoconferencing)~~
1.2.1 Two way interactive with broadcast to other participants
1.3 Multiple senders
1.3.1 Continuous presence (all participants displayed simultaneously)
1.3.2 Switched video
a. Based on audio level
b. By host control
c. By user selection

2. Conference system parameters

2.1 Conference form
2.1.1 Chairman control mode
2.1.2 Conferee control mode
2.2 Call establishment
2.2.1 Resevation based mode
2.2.2 Demand mode
2.3 Call set-up mode
2.3.1 Assembly method
2.3.2 Call-up mode

3. Conditions of MCU

3.1 Location of MCU
2.1.1 Midpoint MCU
2.1.2 End point MCU
3.2 Interwoking of MCU
2.2.1 Singlepoint-branch control method (Single MCU)
2.2.2 Multipoint-branch control method (Multiple MCUs)

3.3 Interface condition of MCU
3.3.1 Maximum number of parties
3.3.2 Interface condition between terminal and MCU
3.3.3 Interface condition among MCUs

4. Functions of MCU

4.1 Network access and interface
4.2 Management of framing structure: multiplexing and demultiplexing
4.3 Mixing of audio signals
4.3.1 N x (N-1) mixing
4.3.2 Mixing suppression
a. Audio broadcast selection
b. Speaker and chairman mix and broadcast
4.3.3 Noise suppression
4.4 Processing of video signals
4.4.1 Simultaneous display
a. Composition multiplexing
b. Picture interleaving
4.4.2 Alternating display
a. Automatic switching according to audio level
b. Manual switching according to conferee/chairman request
4.5 Processing of data signals
4.6 Analysis of control messages
4.7 Processing of the subchannels
4.8 Routing of signals to videoconferencing terminals and other MCUs
4.9 Handling of encrypted signals
4.10 Addressing of specific other terminals (optional)
~~4.11 Office Automation facility (optional)~~
4.~~12~~ 11 Operator's console for conference monitoring and control (optional)

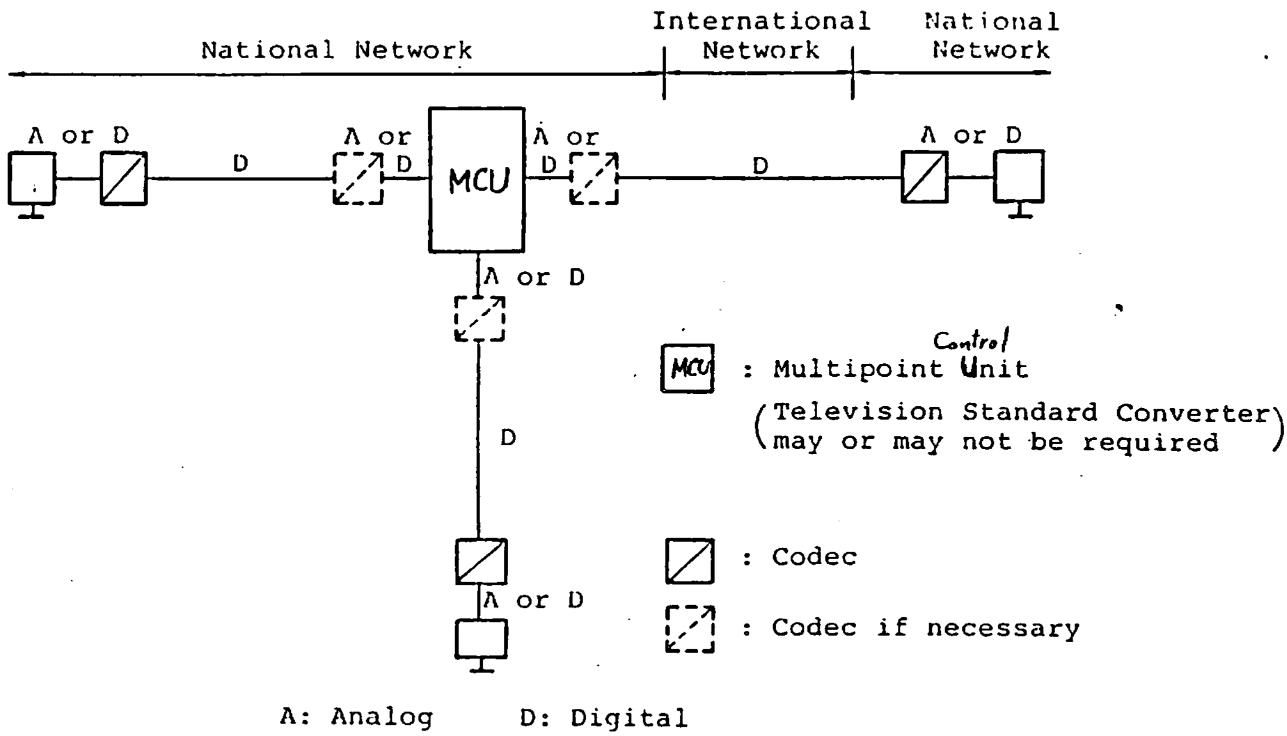

Figure 1 Basic reference configuration for multipoint videoconferencing

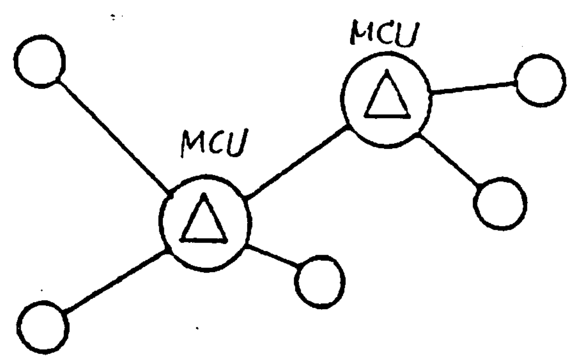

Figure 2 Configuration with two MCUs involved

ANNEX 2 (Q.5/XV)

new series

Summary of proposed Recommendations for audiovisual services

1. **Service definitions**

		new series & name	SG
100	General Recommendation for AV services	* 1.1	I
110	Teleconference services (SG I has begun drafting)	1.2	I
111	Audioconference service	1.3	I
112	Telematic Teleconference service	1.4	I
113	Videoconference service	1.5	I
120	Videophone service	1.6	I
130	... (other AV services)		

2. **Infrastructure** General recommendation for frame structure

200	General Recommendation for AV infrastructure	new	XV
210	Reference network configuration	2.4	XV
220	General recommendation for frame structures		
221	Frame structure for 64 kbit/s AV	2.1	XV
222	Frame structure for 384-2048 kbit/s (cf. H130)	2.5	XV
223..	(Frame structures for higher bit-rates)	2.5	XV
231	Multipoint control for 64 kbit/s AV	2.2	XV/VIII
232	Multipoint control for 384-2048 kbit/s	2.5	XV
233..	(Multipoint control for higher bit rates)	2.5	XV
240	Interworking between 64 kbit/s AV services using the frame structure of Rec. Y.220	new	XV/VIII
241	Interworking between 64 kbit/s AV services to Y.220 and 64 kbit/s audio-only and data-only terminals	2.3	XV/XVIII
242	Interworking between higher bit rate AV and data-only terminals (suggested by SG I Rapporteur)	new	XV/VIII

3. **Facilities**

310	Requirements for teleconferencing	-	
311	Audio requirements	3.1	XII/XV
312	Telematics requirements	3.4	VIII
313	Video requirements	3.2	XV
320	Requirements for Videophone service	-	
321	Audio	new	XII/XV
323	Video	3.3	XV
330	Facility coding	-	
331	Audio codings (of Rec. G.711, 721, 72X...)	3.5	XVIII
332	Video codings (of Recs. H.120, H.12X ...)	3.6	XV

4. **Other Recommendations**

Multipoint call set-up	4.1	XVIII

cross ref. to ~~Annex 1 of [illegible]~~ Appendix III of Annex 1
