ANNEX 35

(to the Report of SG16)

QUESTIONS: 24/16

SOURCE: ITU-T SG 16 (Geneva, 7-18 February 2000)

TITLE: Parameters for variable bit rate voice codec operation and information on
                   channel models

COMMUNICATION

FROM: 24/16

TO: TIA/TR45; ETSI/SMG11; ETSI/TIPHON

APPROVAL: Approved by SG 16 (Geneva, February 7-18 2000)

FOR: Action

DEADLINE: November 10, 2000

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ITU-T SG16/Q24 is a newly approved project chartered to design a new variable bit rate coder
specifically for use in both 3rd Generation Wireless and VoIP applications. The text of the newly
approved question is attached.

In order to better specify the codec design parameters for the intended applications, Q24 seeks input from ETSI (TIPHON), ETSI-SMG11/3GPP and TIA-TR45/3GPP2. The following is a list of initial coder parameters that need to be addressed:

1. Minimum bit rate

2. Maximum bit rate

3. Average bit rate(s) (there maybe several desired average rates)

4. Resolution of bit rate changes (e.g. size of bit rate increment/decrements)

5. Rate control details (source control, network control)

6. Delay limits (algorithmic, computational)

7. Level of error resilience required

8. Audio bandwidth: narrowband (300-3400 Hz) to wideband (50-7000 Hz)

9. Resilience to background noise

10. Complexity

11. Type of bit stream syntax embedded (e.g. layered or other)

With regard to the above parameter list, Q24 requests input from the respective organisations in their area of expertise (3rd generation wireless and/or IP). Specifically, it is desired to better understand the bounds on the various parameters as they are affected by each application/network.

Lastly, Q24 also would appreciate receiving available information on the channel error models used in 3rd generation wireless and IP, which could then be used in future codec testing activities.

Attachment:

(A) Attach pages 32-33 of TD 89 (P) (the Revised text of Q.24/16)

QUESTION 9/16: Variable bit rate coding of speech signals

(Continuation of Question 24/16)

Background and Justification

The ITU has noted a rapid increase in the use of IP for data and a growing commercial interest in deploying voice over IP as an application. In addition, the worldwide usage of wireless communication systems is rapidly increasing, which includes the possibility of deploying certain IMT-2000 networks as IP-oriented systems. IP traffic is evolving to become the dominant global communication system with both wired and wireless access. The transport of IP voice traffic within and between wired and wireless networks is of special interest.

A key issue therefore is the design of new voice coding technology that can best exploit the properties of these networks to provide efficient use of bandwidth and minimise the need for transcoding. As traditional circuit-switched networks will stay with us for a long time, easy transcoding with them is also an important design aspect.

Hence, new work is proposed to examine the applications, technical issues of IP and wireless networks, and investigate the design of a suitable variable bit rate voice coder.

Examples of bearer networks for VoIP carriage which have been identified as benefiting from a variable bit rate voice coder are:

- Ethernet Networks

- ATM Systems

- Third Generation Wireless Systems, including CDMA and TDMA

Additionally, multimedia applications may benefit from a variable bit rate voice coder.

For the activities of this question, the term variable bit rate (VBR) has been defined as follows. A VBR encoder outputs a bit stream which may have a variable number of bits in successive frames. That is, each frame may contain a different number of bits relative to the last frame. Bit rate may vary, for example, in large predefined increments/decrements or it may vary by as little as one bit resolution. The variability in bit rate may be either network controlled or source controlled according to the input audio signal.

Study Items

What algorithm should be specified for the coding of voice/audio, with specific attention being paid to the applications, and the consequent constraints imposed jointly by the network and application.

Items for study include:

1. Clearly define the VBR parameters for each application

2. Study and definition of applications and performance for variable bit rate voice coding

3. Consideration of network requirements and constraints

4. Definition of performance requirements and system constraints

5. Definition of the test conditions and evaluation procedures to be applied in selecting between candidate algorithms on the basis of subjective performance, as specified by SG 12, and non-voice performance

6. Study of encoding techniques and evaluation of the possibility of selecting a single type of VBR technique capable of operating at different transmission rates with an associated improvement in speech quality as the transmission rate is increased.

7. Selection and specification of procedures to be used in verifying the implementation of selected algorithms.

While studying these work items, specific efforts will be put on the key issues of reducing cost and complexity, minimising transcoding, interworking with other systems, and quality of service.

Specific tasks and deadlines

- Definition of performance requirements and objectives: February 2000

- Approval (Decision) of the Recommendation: November 2001

Relationships

- Other relevant Questions of SG16

- ITU-T SG 2 to identify other potential user applications

- ITU-T SG 11

- ITU-T SG 12 for the performance evaluation of specified algorithms with respect to voice quality

- ITU-T SG13

- ITU-T SG15

- ITU-R TG 8/1 to ensure compatibility with mobile transmission system constraints

- ISO/MPEG

- ETSI (Tiphon)

- TIA

- IETF

- ATM Forum

- 3GPP, 3GPP-2