A SPEECH CODER FOR AERONAUTICAL TELECOMMUNICATIONS

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Abstract

British Telecom International (BTI) has introduced an aeronautical telephone service known as the Skyphone. This service uses 9.6 kbit/s speech codecs because of the limited satellite power available. Because of the need to achieve operation of aeronautical telephones between any aircraft and any ground station in any country, the technical specifications have been agreed internationally. After extensive testing of the various speech codecs proposed for the service by companies from different countries, the codec designed at British Telecom Research Laboratories (BTRL) was found to be the best and is to be used both for international telephony and for air traffic control.

This paper gives a brief description of the Skyphone and the factors which influence the speech codec design and then describes the codec designed by BTRL for the service.

1 INTRODUCTION

With around 1000 wide-bodied jets in service, several million busy executives traverse the world every year. For these executives, accustomed to mobile phones and constant contact with their offices, it is unacceptable to be out of touch for the duration of long international flights.

As a first step to meeting this market need and providing improved communications between airlines and their aircraft, a consortium comprising British Telecom, British Airways and Racal Decca was formed to demonstrate in-flight telephone facilities. Following technical trials during 1988, which successfully demonstrated telephony via satellite to an aircraft, a trial service was started in February 1989, providing two telephony channels to a British Airways Boeing 747 flying trans-Atlantic routes. The service is being marketed by British Telecom International (BTI) and is called the Skyphone service. For this service, British Telecom is providing the earth station equipment at Goonhilly, the links through to the international telephone network and billing facilities. Racal Decca has provided the aircraft equipment required to set up calls, including the special aircraft antennas. The system uses a satellite of the INMARSAT organization which currently provides satellite communications mainly for the maritime community.

Later in 1989, the trial service will be replaced by an augmented fully automatic operational Skyphone service conforming to the specifications agreed internationally. With the introduction of this operational service an agreement between British Telecom and the telecommunications administrations of Norway and Singapore will come into effect. This agreement will extend the Skyphone service to enable truly worldwide service to be offered, with telephony traffic from flights over the Pacific, Indian and Atlantic oceans.

Digital transmission is to be used between the aircraft and the BTI international exchanges. Low bit-rate speech coding has to be used because of the very limited satellite power available. Following an international competition to select the voice coding system for global use, the BTRL speech codec has now been chosen by INMARSAT and has been recommended to the Airlines Electronic Engineering Committee by its specialist sub-group.

The next section describes the features of the aeronautical system which affect the design of the speech codecs. The British Telecom Research Laboratories (BTRL) speech coding algorithm for the Skyphone service is presented in section 3. Section 4 describes the performance of the codec, giving test results and discussing operational experience of its behaviour in the Skyphone system. Conclusions are presented in section 5.

2 SPEECH CODEC DESIGN FOR SKYPHONE

The main elements of the aeronautical satellite system are illustrated in figure 1. The operating bit-rate for the speech codec was selected as 9.6 kbit/s because no lower bit-rate codec examined by INMARSAT and BTI gave acceptable speech quality. Higher bit-rate codecs, while yielding better speech quality, were ruled out because of the limited satellite power available.

The data multiplexer of figure 1 is used to multiplex the output data stream from the speech encoder and some signalling information. The 10.368 kbit/s bit-rate of this combined data stream is doubled when forward error correction (FEC) is applied. The half rate FEC unit consists of a convolution encoder of constraint length seven: the corresponding decoder is an 8-level soft decision Viterbi decoder [1]. To combat the multi-path fading characteristic of the aeronautical transmission path, interleaving is applied to preserve the FEC coding gain by spreading burst errors throughout a transmission frame. A number of interleaved blocks are collected together and a 'unique word' and a few dummy bits are added to form a half second RF frame which is transmitted at 21 kbit/s. The RF demodulator synchronizes itself with respect to the unique word. Loss of frame synchronization is assumed to have occurred after the demodulator unique word match is below a prescribed threshold for two consecutive frames. A form of Offset Quadrature Phase Shift Keying, (O-QPSK), known as Aviation - QPSK (A-QPSK) [2, 3], is used to modulate the 21 kbit/s signal onto the radio frequency carrier. The aircraft antenna for the trial service will be an electronically steerable high power antenna.

The bit stream presented to the speech decoder after errors have been introduced on the radio path and reduced by the channel coding has an average bit error rate of between 1 in 100 and 1 in 1000. There will also be bursts of errors when the error correction cannot cope. The speech codec must be
able to suppress the effects of all of these. Particular attention was paid to the different types of transmission error in the development of the BTRL codec for the Skyphone system.

3 THE BTRL SKYPHONE SPEECH CODEC

3.1 Introduction
To achieve the spectral efficiency required for the Skyphone service low bit-rate speech encoding is required. A multipulse-excited LPC algorithm [4] was selected by BTRL as it is robust to background noise and transmission errors.

A schematic of a multipulse-excited LPC encoder is illustrated in figure 2. The encoding process involves splitting the incoming speech into frames of duration to derive the short-term and long-term predictor parameters and an analysis-by-synthesis process to obtain the pulse positions and amplitudes. The short-term predictor models short-term correlations in the speech signal. The long-term predictor takes advantage of long-term correlations in speech, primarily arising from pitch-related correlations in voiced speech.

The excitation analysis-by-synthesis procedure involves a search for the pulse positions and amplitudes which minimize the mean squared error between a block of the input speech and the speech obtained from the synthesizer in the local decoder. The derivation of the excitation may be carried out for the entire LPC block length or the LPC frame may be split into several sub-blocks and the excitation derived for each sub-block separately. To find all the pulse positions and amplitudes simultaneously for a sub-block requires the solution of non-linear equations and is thus very complex. Sub-optimal methods have therefore been developed to derive the pulse positions and amplitudes sequentially. A multipulse-excited LPC decoder, corresponding to the encoder of figure 2, is presented in figure 3. The decoding process is very straightforward, requiring the formation of the excitation signal and the application of this excitation to the long-term and short-term predictors to give the synthesized speech output.

3.2 Key codec parameters
The key parameters of the Skyphone speech codec are as follows:

- 10th order LPC autocorrelation analysis
- 32 ms Hamming window with 37.5% overlap
- 20 ms short-term predictor update period
- Single tap long-term predictor updated every 20 ms
- Excitation frame size of 4 ms
- Three pulses per excitation frame

3.3 Error protection
The Skyphone codec was designed to cater for two different error conditions:

- the expected random bit error rate of about 1 in 1000
- burst errors caused by breaks in the radio link due either to momentary misalignment of the aircraft antenna or poor propagation conditions.

It was found that to maintain the speech quality of the error-free case for the random errors required forward error correction to be applied only to the reflection coefficient information bits. With only a few hundred bit/s of the available 9.6 kbit/s data rate used for error protection, the codec maintains the required voice quality for a random bit error rate of 1 in 1000. During the worst propagation conditions the RF demodulator cannot maintain synchronization: when this occurs the demodulator activates a codec squelch control signal. Unfortunately the squelch control signal cannot be activated until two unique word matches are below a prescribed threshold. Thus with a radio system frame size of half a second, up to one second of random data will be passed to the codec before the squelch signal operates. The codec has to detect this situation, and other severe burst error conditions, and mute the codec output signal. Some of the transmission bits in the 9.6 kbit/s data stream have therefore been reserved for burst error detection. When a burst error is detected that lasts for up to one speech frame, 20 ms, it has been found that it is better, perceptually, to repeat the previous speech frame than mute the codec. If however burst errors are present for several consecutive frames then the codec output is muted until no more burst errors are detected. Figure 4 shows the burst error distribution measured at the codec input during some recent flight trials. This distribution does not take account of the times when the squelch control signal was muted until no more burst errors are detected. Figures 5 and 6 show the burst error distribution measured at the codec output for several hundred speech frames occurred.

4 TEST RESULTS

4.1 Subjective Test Results
An introduction to the rationale and the procedures adopted for the subjective testing of low bit-rate speech codecs is presented in [4].

Two sets of subjective tests have been carried out: the first tests were to select the speech codec for the BTI Skyphone trial service and the second to select a speech codec as the international standard for aeronautical satellite telecommunications.

4.1.1 Tests to select the speech codec for the Skyphone trial service
As part of the procedure for selecting the codec for the Skyphone service, four codecs (one from Japan, one from the USA and two from the UK) were evaluated by a series of subjective tests. The subjective tests were designed to cover a range of input levels, listening levels and conditions representative of the operating environment of the Skyphone service.

The subjective test report [5] gave a strong recommendation that the BTRL speech codec was the most suitable for the Skyphone service in terms of speech quality.

4.1.2 Tests to select the speech codec to be the international standard
The Airlines Electronic Engineering Committee (AEEC), which is responsible for standardizing all the communications equipment carried on commercial aircraft, was informed of the results of the first set of subjective tests to select the speech codec for the BTI Skyphone service. This committee then commissioned its own tests to select a codec as an international standard. These tests were very similar to those described above but with the addition of conversation tests. Four codecs were selected by the AEEC for testing (two codecs from the USA, one from Japan and the BTRL codec).
Both English and Japanese were used in the subjective listening tests performed on the four selected codecs. Subjective test laboratories in the USA, Japan and England were used for the tests. The report on these subjective tests recommended that the AEEC adopt the BTRL speech codec for aeronautical satellite telephony, as it out-performed all the other codecs in the realistic operating conditions tested. (The results of these tests are presented in [6]).

The conversational tests were performed by the UK Civil Aviation Authority (CAA). In these tests, air traffic controllers and pilots engaged in conversations similar to those which occur in air traffic control (ATC). Following these conversations, which were conducted through each of the four codecs under consideration, each air traffic controller was asked to compare the codecs. The UK CAA concluded from their test results that the BTRL speech codec was the most suitable codec for ATC.

The two reports on these subjective tests were presented to the AEEC's Voice Coding Working Group during December 1988. In the light of these results, the working group recommended that the BTRL codec be selected as the International Standard for aeronautical mobile satellite telecommunications. This decision has since been ratified by the AEEC's Satellite Sub-committee which has submitted it for formal adoption at the next plenary session of the AEEC.

4.2 Test Results with Non-Speech Signals

The ability of the BTRL Skyphone codec to pass DTMF signalling tones has been investigated in a comprehensive set of tests. The tests consisted of passing several tens of thousands of DTMF characters through the codec and analysing the received tones to ensure that the transmitted character was received correctly. The codec was tested with no bit errors, bit errors at 1 in 1000 and bit errors at 1 in 500 injected into the transmission path. The results are summarised in table 1 below.

<table>
<thead>
<tr>
<th>Error condition</th>
<th>% characters correctly received</th>
</tr>
</thead>
<tbody>
<tr>
<td>No errors</td>
<td>100.00</td>
</tr>
<tr>
<td>1 in 1000</td>
<td>99.78</td>
</tr>
<tr>
<td>1 in 500</td>
<td>99.36</td>
</tr>
</tbody>
</table>

Table 1 DTMF test results

From table 1 it is clear that the BTRL codec passes DTMF tones very reliably, even at the nominal bit error rate of 1 in 1000.

These laboratory results have been verified by the Jetstream and the public service trials (see section 4.3 below) with numerous calls successfully connected by DTMF dialling through the BTRL codec.

4.3 Jetstream trials and trial public service

During May/June 1988 the Skyphone system was tested using a Racal Avionics executive jet. During the tests many calls were made from the jet to those involved in the airlines industry (including air traffic controllers) in the UK and the USA. The unsolicited comments from those receiving the calls were that the voice quality was very good and that no appreciable difficulty was experienced in conversing over the link.

A trial commercial service was started in February 1989, providing two telephony channels to a British Airways 747 flying trans-Atlantic routes. This service has proved popular with passengers and good usage is being made of it.

As this initial trial service does not provide a separate signalling channel for DTMF dialling, the DTMF tones are transmitted through the speech codec at call initiation. Transmission of these tones through the codec has proved to be very reliable. As most of the other speech codecs examined by BTI were not able to pass DTMF tones, the use of one of these other codecs would have required access to international operators to set up calls during the trial service.

5 CONCLUSIONS

The BTRL codec for the Skyphone service is a robust, 9.6 kbit/s speech codec giving good speech quality. It is also capable of transmitting DTMF signalling tones. The codec is robust in terms of both background noise and transmission errors. Random transmission bit error rates of 1 in 1000 have no effect on speech quality and higher bit error rates of up to 1 in 100 have only a small effect. The effects of error bursts or breaks in transmission up to 20 ms are almost undetectable. The end-to-end codec delay is less than 40 ms with the codec requiring less than 75% of the available processing time on a WE DSP32. In a comprehensive set of subjective tests the BTRL codec was ranked significantly better than the international competitors with which it was compared. It has recently been chosen as an international standard for aeronautical mobile satellite telecommunications.

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7 REFERENCES


4 Southcott C B et al: 'Low bit rate speech coding for practical applications', Br Telecom Technol J, 6, no.2, pp 50-59, 1988

5 Crowe D P: 'Experimental results on 9.6 kbit/s speech codecs for use in the proposed aeronautical satellite service', BT Internal Memorandum 1988.

6 Crowe D P: 'Selection of voice codec for the aeronautical satellite system', European Conf. on Speech Commun & Techn., 1989
Fig 1  Simplified block diagram of aircraft transmission system.

Fig 2  Multipulse excited LPC encoder.

Fig 3  Multipulse excited LPC decoder.

Figure 4  Burst error distribution (measured over several minutes)