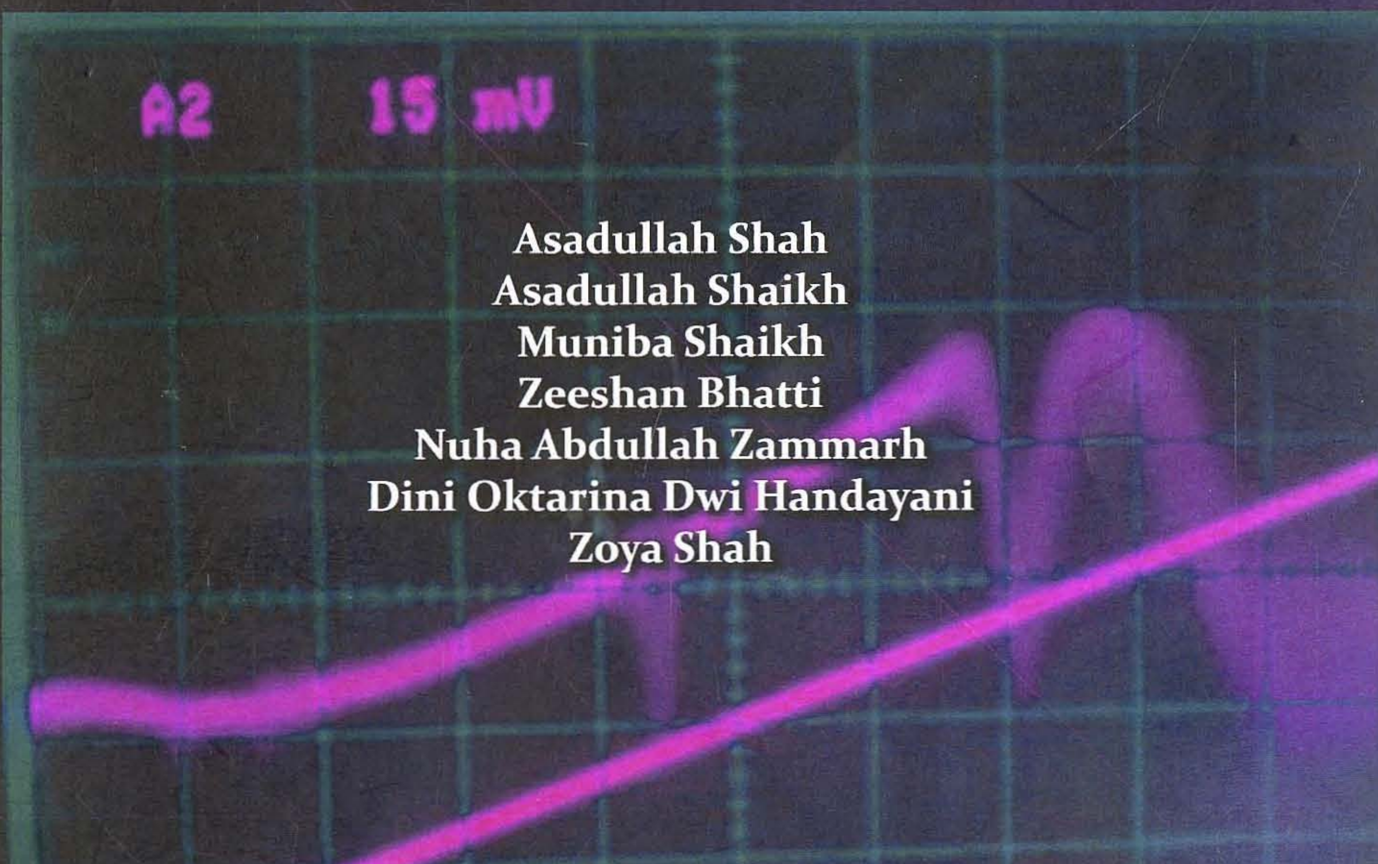
STATISTICAL TIME DIVISION MULTIPLEXING ARCHITECTURES AND DESIGN



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15. Speech Quality in Lossy Multiplexing

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15.0 Abstract

Speech quality in multiplexing suffers due to various factors, including forced packet dropping. If a packet or segment of speech is dropped from any source by selecting users on random bases, and incidentally the randomly selected packet is perceptually sensitive, a single packet drop can cause a significant degradation is the quality both objectively and subjectively. To combat such problem a criterion based packet dropping produce better results. In this chapter speech quality in multiplexing situations both objectively and subjectively is explained.

15.1 Multiplexer simulation

Lossy multiplexer simulation were carried out using 6.4 kb/s PRELP [43] codec with random frame loss of 1...10%. Each 20ms speech segment was coded into a frame of 128 bits. The multiplexer was simulated as a random ON-OFF channel which makes its ON-OFF decision at the beginning of each frame. 'When the channel is ON, speech frames from the encoder are transmitted to the decoder. When it is OFF, the decoder and/or encoder are flagged to indicate that the given frame was not transmitted. On reception of this flag, the decoder goes into a LFR state which employs an algorithm. If the encoder receives this flag, it first resets its memories to the state prior to the encoding of the lost frame. Secondly, it emulates the LFR process that would take place at the decoder for generating a new set of memories. It then continues as normal to encode the following frames. The figure 15.1 shows a speech sequence in which frame n is lost. In comparing (c) and (d) with (a) and/or (b) it can be seen that the effects of reconstruction of frame n is almost completely eradicated for the transmitter-receiver LFR in (d), however the effects of reconstruction is still evident even by the end of frame n + 2. This is again seen in figure, which shows, the frame SNR degradation arising from LFR. Whilst all frames are correctly received, this degradation is 0dB. Immediately a frame is lost (and reconstructed), the degradation rises. As the succeeding frames are correctly received, the degradation slowly decays to 0dB. On average, the period of decay is longer for the receive only than for the transmitter-receiver LFR. Note however that the SNR degradation in the latter case does not actually return to zero. In objective and subjective test however, this low level distortion is more tolerable than the error propagation arising from receiver-only LFR.