

# STATISTICAL TIME DIVISION MULTIPLEXING ARCHITECTURES AND DESIGN

A2

15 mV

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Asadullah Shaikh  
Muniba Shaikh  
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Dini Oktarina Dwi Handayani  
Zoya Shah

200mV

20mV



0.1 500ns

IIUM Press  
INTERNATIONAL ISLAMIC UNIVERSITY MALAYSIA

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## Editors

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Published by:

IIUM Press

International Islamic University Malaysia

First Edition, 2011

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Perpustakaan Negara Malaysia

Cataloguing-in-Publication Data

**Asadullah Shah**

**Statistical Time Division Multiplexing Architecture and Design / Asadullah Shah  
... [et al.].**

ISBN: 978-967-418-190-1

Member of Majlis Penerbitan Ilmiah Malaysia – MAPIM  
(Malaysian Scholarly Publishing Council)

Printed by:

**IIUM PRINTING SDN. BHD.**  
No.1, Jalan Industri Batu Caves 1/3  
Taman Perindustrian Batu Caves  
Batu Caves Centre Point  
68100 Batu Caves  
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## 14. Packet loss and Frame Substitution Techniques

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### 14.0 Abstract

In packetized speech communication the packet loss occurs. The packets can also be forced to drop in situations when traffic load increase at the multiplexing end. The dropped packets can be substituted at the demultiplexing end. To do so frame substitution techniques can be utilised to mitigate the effects of lost packets. In this chapter packet loss (dropping) and frame substitution techniques are explained in detail.

### 14.1 Method

When a frame discarding occurs in speech coded at low bit rate coding, the output signal for that frame cannot be reconstructed by zero-amplitude stuffing or waveform substitution techniques. This is because the low bit rate coding techniques code signal sequentially and involves recursive filter memories. These memories get corrupted when a frame is discarded or lost. If the adaptation of these filter memories is substituted by the zero-stuffing or freeze-out for that particular frame. Then the decoder stops its adaptation and decodes dummy code words. However, the encoder computes the coded signal continuously whether the packet, loss occurs or not, so the internal variables in the encoder and decoder have different values once a frame is discarded. Therefore, large distortion can be produced owing to mistracking operations between the encoder and the decoder and the synchronous operation of encoder and decoder is disturbed. Although the mistracking operation subsides within a certain time period, but severe degradation at the beginning of the mistracking operation always occur. The substitution techniques are not applicable to lost frame recovery for low bit rate coded speech. LFR reported at random frame loss performs well at, 3% loss rate for low bit rate coders. At higher frame loss rates, speech quality suffer severe degradation. The substitution of a speech frame results in two forms of degradation at the decoder:

1. Frame reconstruction error: This is localised degradation suffered by the reconstruction frame at the output of the codecs. An effective lost speech frame reconstruction technique must seek to minimise this error objectively and or subjectivity.
2. Error propagation: Low bit rate speech coders incorporate recursive filters in which the memories becomes corrupted when a frame is lost. The corrupted memories tend to