Session III: New ETSI Model on Wideband Speech and Noise Transmission Quality – Phase I

IP transmission simulation

TELEFÓNICA I+D / UNIVERSIDAD DE VALLADOLID
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01 Performance parameters

Delay, jitter and packet loss

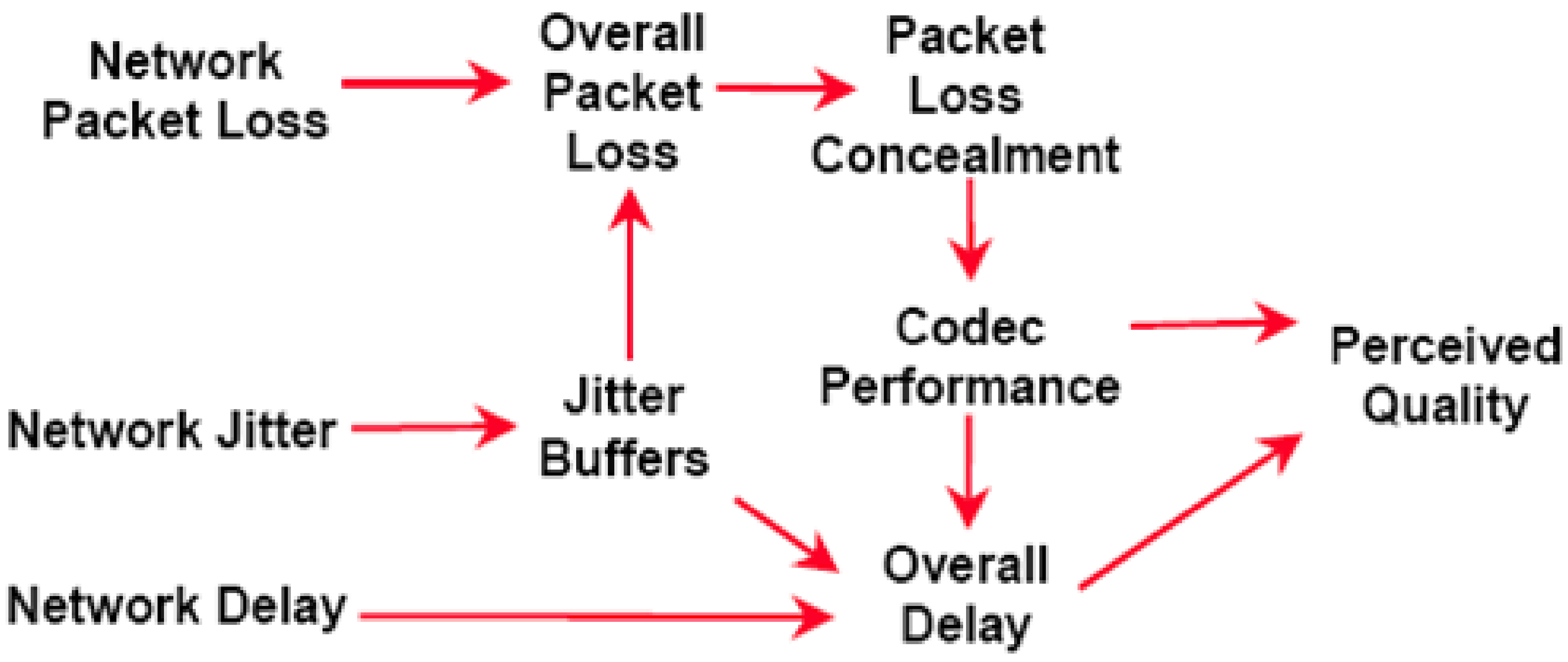
Lots of conditions and parameters that can influence on speech quality

- Delay: amount of time it takes for a signal to reach a destination
  - Very direct impact on user satisfaction
  - ITU: <150ms (preferred) ; 400 ms (limit)
  - Codec delay + packetization delay + output queuing delay + serialization delay + network delay + network switching delay + propagation delay + de-jitter delay
- Jitter: variation of delay
  - Services intolerant of delay variation take solutions to reduce it by means of buffering (de-jitter buffers) → increases delay
  - ITU: <1ms (audio applications after de-jitter buffer); <30 ms (no buffer)
- Packet loss: percentage of data packets which are lost
  - Very direct effect
  - ITU: < 3% (audio applications)

01 Performance parameters

Parameter interaction and dependences

- These parameters are not independent one another

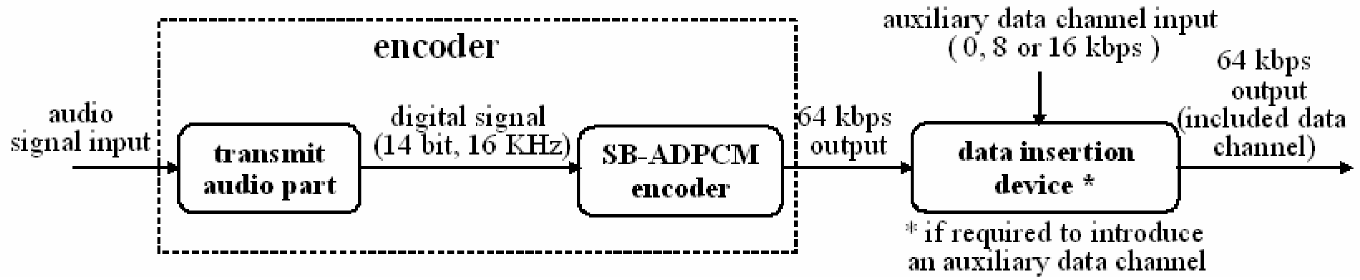


02 Wideband codecs
Overview

- Wider band of frequency (50 Hz to 7000 Hz) compared to traditional Narrowband speech (200 Hz to 3400 Hz)
- Increase intelligibility and naturalness of speech
  - 50 Hz – 200 Hz : increased naturalness, presence and comfort
  - 3400 Hz – 7000 Hz : fricative differentiation and higher intelligibility
- Digitalised at 16 kHz
  - 16-bit integer → 256 kbps
- Speech compression becomes of significant importance

02 Wideband codecs
G.722

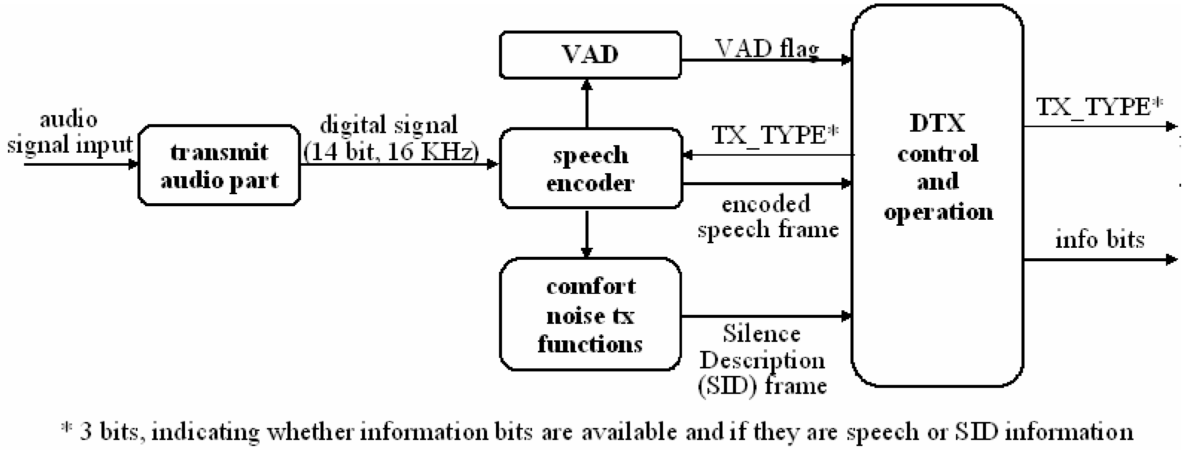
- ITU-T Recommendation
- SB-ADPCM (Sub Band – Adaptive Differential Pulse Code Modulation)
- 3 modes of operation : 64 kbps; 56 kbps (auxiliary data channel 8kbps) and 48 kbps (auxiliary data channel 16 kbps)
- Encoder



02 Wideband codecs

AMR-WB

- 3GPP /ETSI
- Recommendation G.722.2 ITU-T
- ACELP (Algebraic Code Excited Linear Prediction Coder)
- Adaptive codec capable of operating at 9 modes of operation : 6.6 kbps, 8.85 kbps, 12.65 kbps, 14.25 kbps, 15.85 kbps, 18.25 kbps, 19.85 kbps, 23.05 kbps and 23.85 kbps
- Encoder



03 Background noise transmission simulation

Steps

Clean speech sequences recording

Noise-free speech samples (8 samples)

Noisy conditions adding

Speech samples with background noise conditions (80 samples)

Wideband terminal simulation

Speech samples with noise and terminal conditions (720 samples)

Network simulation

Speech sample database (4320 samples)

Database of noise type/wideband terminal/network impairment combinations

03 Background noise transmission simulation

Step 1: Speech sequences

- Recording a representative number of speech sequences without background noise
- Conditions
  - 48 kHz (16 bit) sampling rate
  - Wave format
  - Active speech level equalized to -26 dBov

- Number of samples
  - 4 speakers (2 male, 2 female), 8 sentences each
  - 2 languages: Czech, French
  - Length of recordings between 24s and 73s
    - Neutral sentences of 2s to 3s separated by pauses
    - Speech activity factor between 30% and 60%

03 Background noise transmission simulation

Step 2: Noisy conditions

- Different background noises need to be recorded for each speech file
  - Cafeteria noise
  - Office room noise
  - Road noise
  - Crossroads
  - Car noise (car hands-free at 130 km/h)

- Two microphone-loudspeaker positions
  - Typical handset microphone position (with loudness ratings adjusted to 7dB)
  - Hands-free microphone position (with loudness ratings adjusted to 11dB)

03 Background noise transmission simulation

Step 3: Noisy signal processing

- The noisy signal must be processed to take into account the influence of the terminal
  - Convolution with impulse response of WideBand (WB) terminals
  - Application of WideBand (WB) Noise-Suppression Algorithm (NSA)
- Signal processing implemented for STF 294
  - Signal speech+noise down-sampled (from 48 kHz to 16kHz) and filtered out using band-pass filters
  - Noise reduction algorithms with the following parameters
    - Parameter 1: with/without noise estimation using VAD
    - Parameter 2: smooth/sharp noise reduction filter
    - Parameter 3: noise reduction level of 9dB/18dB

03 Background noise transmission simulation

Step 4: Network simulation (I)

- Noisy speech samples are simulated being transmitted over a network, adding delay, jitter and packet loss
- Real-time network emulator: NIST Net
- Procedure for simulation
    1. The call generators establish a call
    2. WAV files are encoded into the proper format (WB codec) by the sender
    3. The transport module produces RTP/UDP/IP packets to be transmitted over the packet network
    4. The source call generator sends the IP packets to NIST Net emulator through IP address 1
    5. NIST Net applies the selected network conditions (delay, jitter, and packet loss)
    6. NIST Net sends the result of the emulation to the receiver through IP address 2
    7. The receiver obtains the packet load
    8. The WB information is decoded and recorded into WAV format

03 # Background noise transmission simulation

Step 4: Network simulation (II)

- Parameters which have been varied for the purpose of STF 294
  - Packet loss
  - Delay
  - Jitter
- ITU-T Recommendations
  - One-way speech delay <150 ms (400 ms as an absolute limit)
  - Packet loss <3% for audio communications
  - Jitter should not be more than 20 ms to 50 ms (1ms after de-jigger buffering)
- Conditions emulated

	End-to-end delay (ms)	Jitter (ms)	Packet loss (%)
1	0	0	0
2	150	10	1
3	400	20	3
Delay/Jitter distribution rule : “heavy-tail” Packet loss distribution rule : random			

03 Database description

Speech samples with background noise

Condition description	Number of conditions	Total
Languages	French Czech	2
Speakers	2 males 2 females	4
Noisy background	Cafeteria noise Office room noise Road noise Crossroads Car noise	5
Microphone-loudspeaker positions	Typical handset microphone position (with loudness ratings adjusted to 7 dB) Hands-free microphone position (with loudness ratings adjusted to 11 dB)	2
TOTAL		80 (2*4*5*2)

03 Database description

Noise reduction, coding and network transmission conditions

<table>
<tr><th>Condition description</th><th colspan="3">Number of conditions</th><th>Total</th></tr>
<tr><td rowspan="9">Noise reduction (Flt 135 filter)</td><td colspan="3">No noise estimation</td><td rowspan="9">9</td></tr>
<tr><td rowspan="4">Noise estimation using VAD</td><td rowspan="2">Smooth noise reduction filter</td><td>Noise reduction level of 9 dB</td></tr>
<tr><td>Noise reduction level of 18 dB</td></tr>
<tr><td rowspan="2">Sharp noise reduction filter</td><td>Noise reduction level of 9 dB</td></tr>
<tr><td>Noise reduction level of 18 dB</td></tr>
<tr><td rowspan="4">Continuous noise estimation (no VAD)</td><td rowspan="2">Smooth noise reduction filter</td><td>Noise reduction level of 9 dB</td></tr>
<tr><td>Noise reduction level of 18 dB</td></tr>
<tr><td rowspan="2">Sharp noise reduction filter</td><td>Noise reduction level of 9 dB</td></tr>
<tr><td>Noise reduction level of 18 dB</td></tr>
<tr><td>Coding</td><td colspan="3">G.722
AMR-WB</td><td>2</td></tr>
<tr><td>Network impairments</td><td colspan="3">Delay 0ms ; Jitter 0ms ; Loss 0% (No impairments)
Delay 150ms ; Jitter 10ms ; Loss 1%
Delay 400ms ; Jitter 20ms ; Loss 3%</td><td>3</td></tr>
<tr><td colspan="4">TOTAL</td><td>54
(9*2*3)</td></tr>
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