World Class Standards

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

IP transmission simulation

ETSI Workshop on Speech and Noise in Wideband Communication

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Performance parameters

Delay, jitter, packet loss

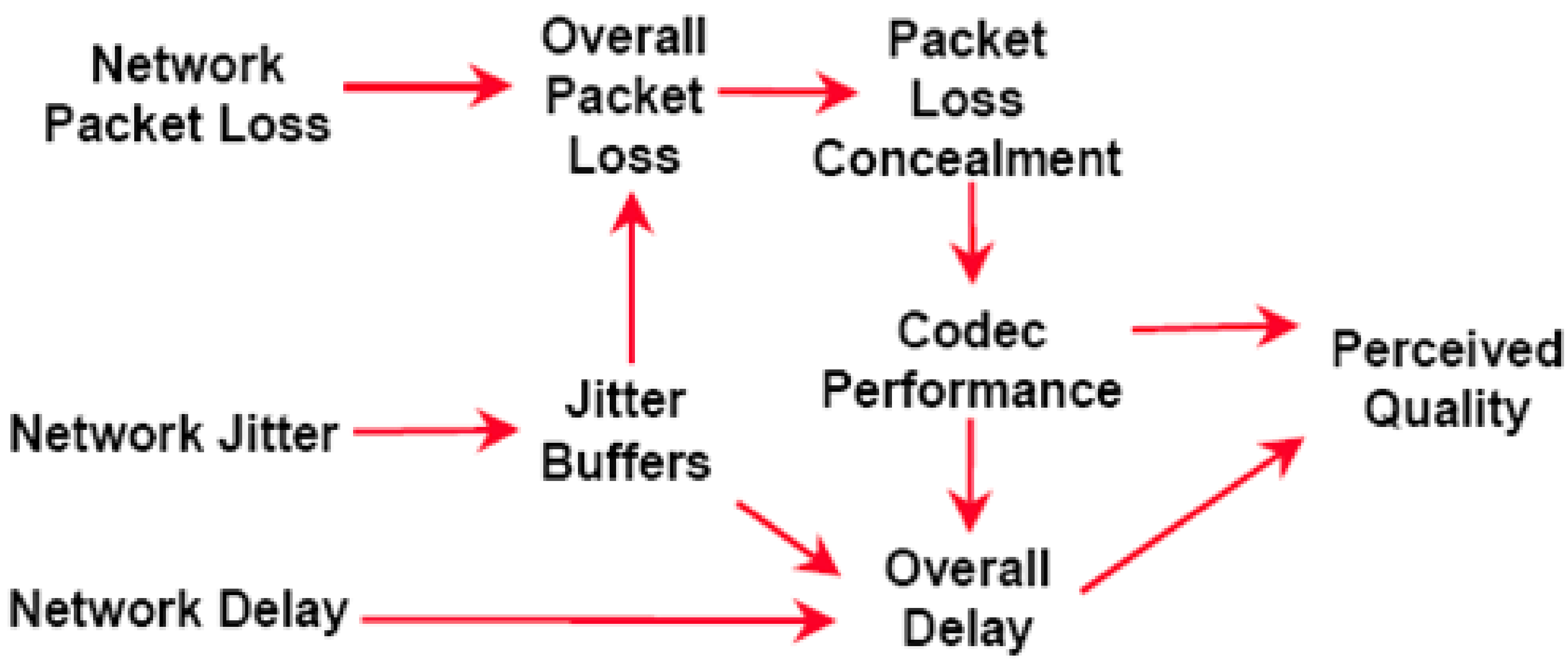
Lots of conditions and parameters that can influence on speech quality

- **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)**

Performance parameters

Parameter interaction and dependences

- **These parameters are not independent one another**



Wideband codecs

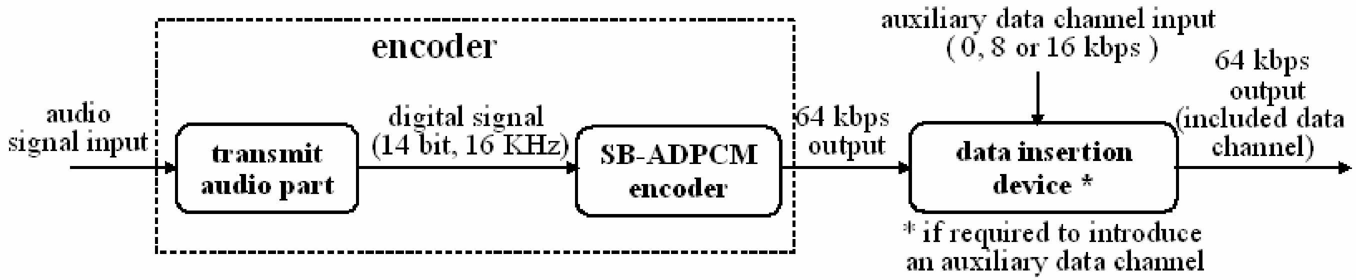
Overview

- 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

Wideband codecs

G.722

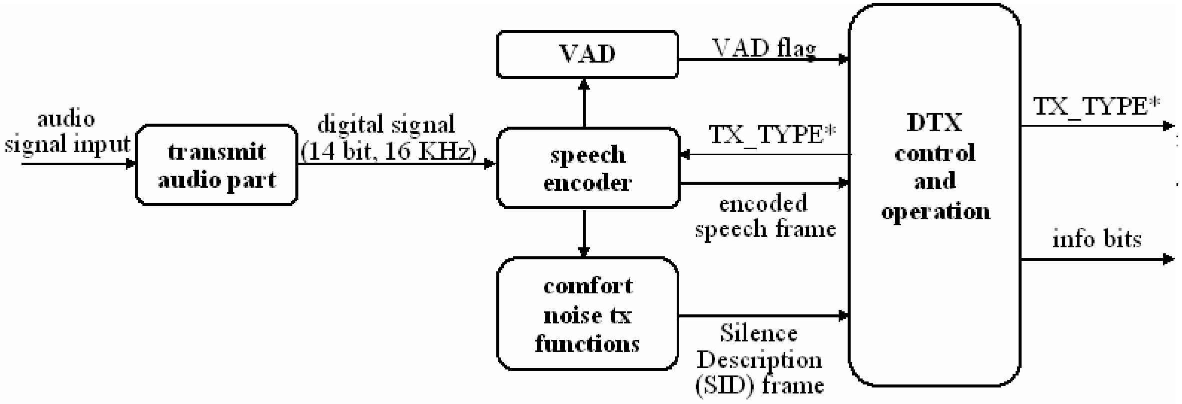
- ITU-T Recommendation
- SB-ADPCM (Sub Band – Adaptive Differential Pulse Code Modulation)
- 3 modes of operation : 64 kbps; 56 kbps (auxiliary data channel 8kbps) and 48 kbps (auxiliary data channel 16 kbps)
- Encoder



Wideband codecs

AMR-WB

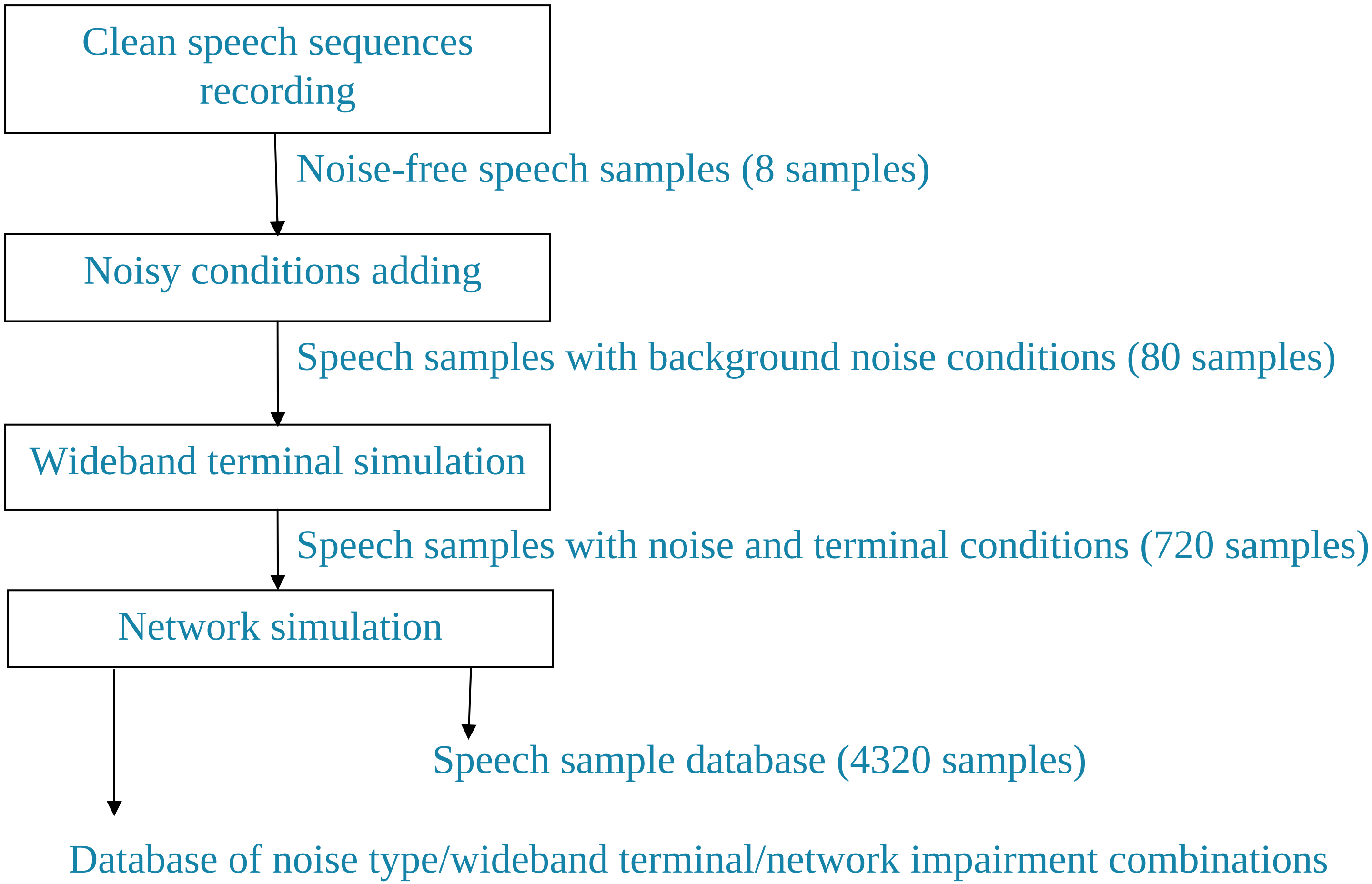
- 3GPP /ETSI
- Recommendation G.722.2 ITU-T
- ACELP (Algebraic Code Excited Linear Prediction Coder)
- Adaptive codec capable of operating at 9 modes of operation : 6.6 kbps, 8.85 kbps, 12.65 kbps, 14.25 kbps, 15.85 kbps, 18.25 kbps, 19.85 kbps, 23.05 kbps and 23.85 kbps
- Encoder



* 3 bits, indicating whether information bits are available and if they are speech or SID information

Background noise transmission simulation

Steps



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%

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)**

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

Background noise transmission simulation

Step 4: 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

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

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**			

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)

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>
</table>

Questions?

Thank you!