

Voice and Fax/Modem transmission in VoIP networks

Martin Brand

A1Telekom Austria

© ETSI 2011. All rights reserved

Name : Martin Brand

Position: Senior IT Specialist at A1 Telekom

Vice Chairman ETSI TC INT

Vice Chairman ITU-T SG11 WG4

tel: + 43 50 664 31916

mob: + 43 664 663 1916

fax: + 43 50 664 9 31916

skype: martin_bd2000

E-mail: martin.brand@a1telekom.at

**Company address: Telekom Austria; Lassallestrasse 9;
1020 Vienna; Austria**

Transport of Voice over IP

ITU-T G.114

- Highly interactive tasks (e.g., some speech, video conferencing and interactive data applications) may be affected by delays below 100 ms.
- For many *intra-regional* (e.g., within Africa, Europe, North America) routes in the range of 5000 km or less, users of VoIP connections are likely to experience mouth-to-ear delays < 150 ms.

ITU-T G.107

- Absolute delay does not impair the intelligibility of speech but if the total delay exceeds around 100 ms from mouth to ear, it begins to affect the interactivity of conversations. Therefore, if possible, large delays should be avoided.

E-Model Transmission Rating (R) ITU-T Rec. G.107

- R summarizes the effects of network impairments including delay, distortions signal level & echo level

$$R = R_o - I_s - I_d - I_e + A$$

R_o= basic signal to noise ratio

I_s= level impairments and distortion impairments

I_d= delay impairments

I_e= codec and packet loss/burst impairments

A= advantage factor

Contributing Factors

Delay

- Codec
- Packet frame duration
- DSP/CPU processing time
- Play out buffer
- Link speed (serialization)
- Propagation delay
- De-jitter buffer delay
- PLC

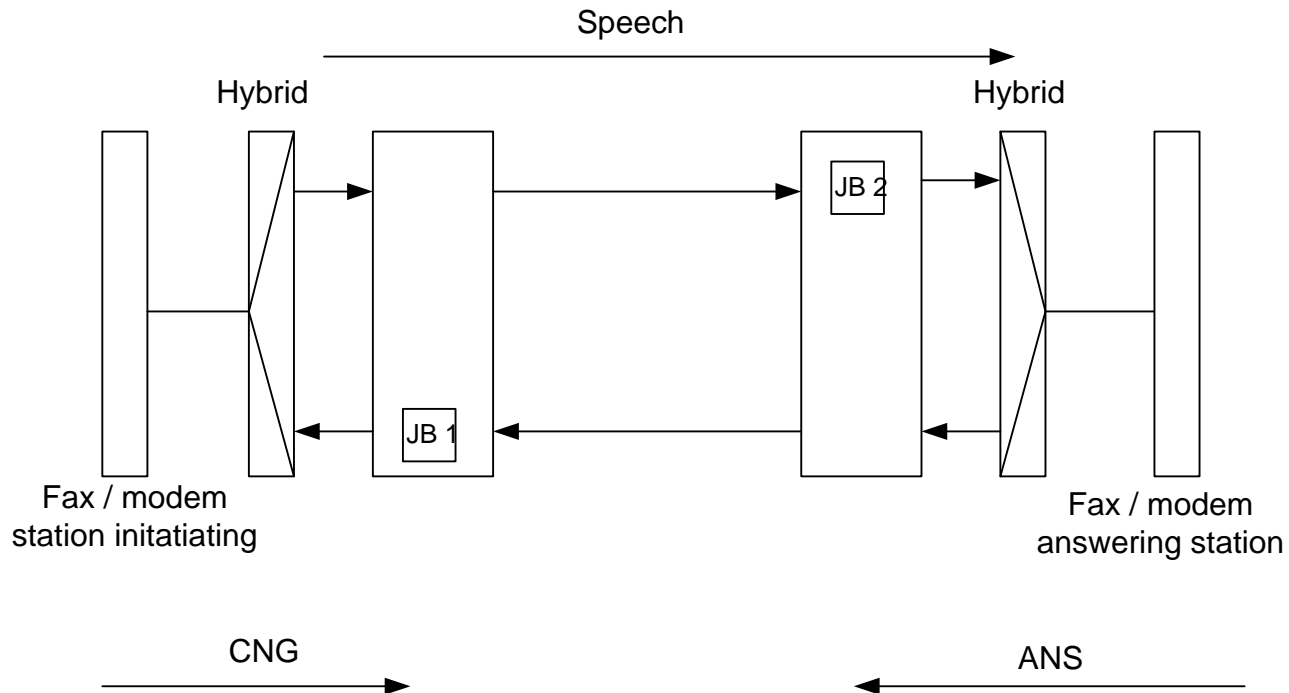
Distorsion

- Codec
- Jitter
- Use of VAD
- Lost of packets
- Transcoding
- Sound level
- Echo level

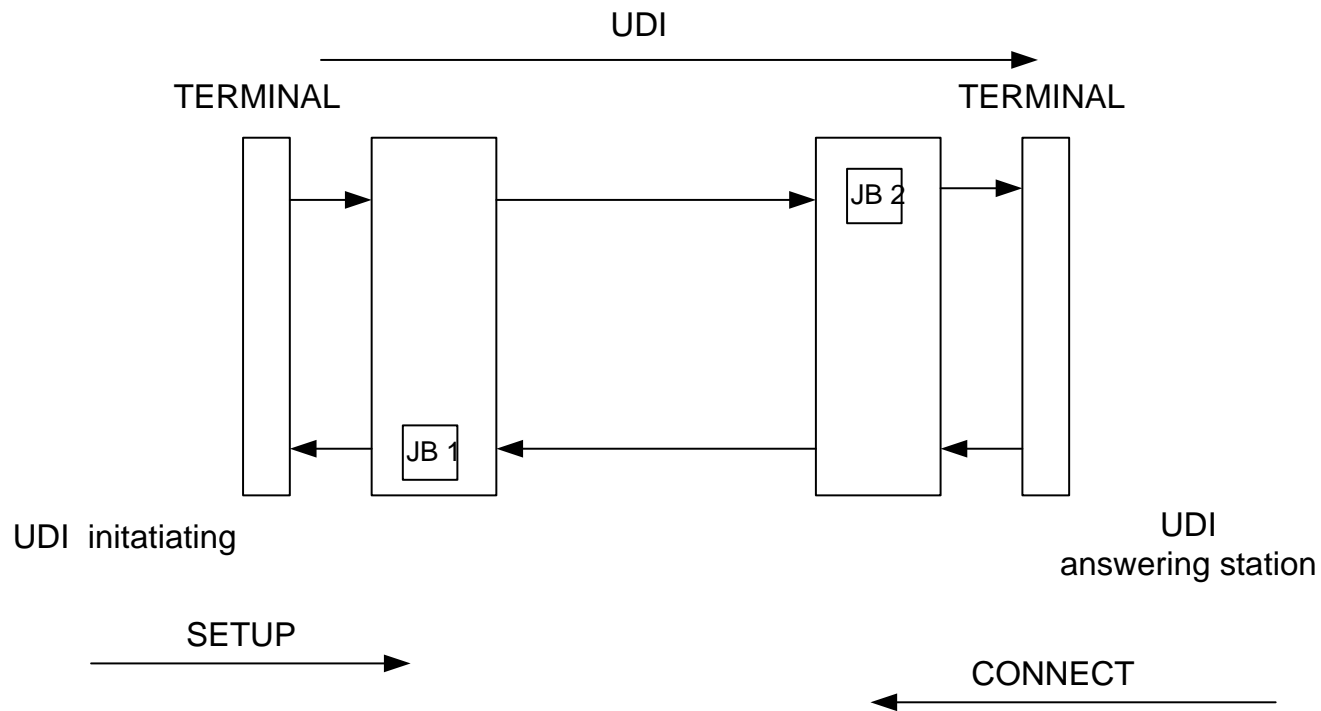
Jitter Buffer

A jitter buffer is designed to remove the effects of jitter from the decoded voice stream, buffering each arriving packet for a short interval before playing it out synchronously. A fixed jitter buffer maintains a constant size whereas an adaptive jitter buffer has the capability of adjusting its size dynamically in order to optimize the delay/discard tradeoff. The disadvantage of adaptive jitter buffer is that a part of the jitter budget is transferred to the user. While the human perception of audio delay variation is low, modem and fax applications are extremely sensitive to delay variation in the audio path. For this reason adaptive jitter buffer are not applicable for fax and modem transmission.

Activation of fixed Jitter Buffer



Activation of fixed jitter Buffer for UDI



Jitter values for DSL line 384 kbit/s uplink; 1024 kbit/s downlink for G.711

| | DSL - Access Jitter Values |
|---------------------|----------------------------|
| Sending | 24 ms - 54 ms |
| Receiving | 9 ms |
| De-jitter Min.: | |
| DSL - > POTS: 24 ms | |
| DSL-> DSL: 33 ms | |
| De-jitter Max.: | |
| DSL - > POTS: 54 ms | |
| DSL-> DSL: 63 ms | |

Delay values between PES users connected on MSAN with and PES users connected to DSL line 384 kbit/s uplink, 1024 kbit/s downlink for G.711 with wired terminals and adaptive jitter buffer

| | PSTN/ISDN Delay (ms)/R | DSL Delay (ms)/R |
|-----------|--|--|
| PSTN/ISDN | 52/91 (20 ms Packet size) 32/92 (10 ms Packet size) | 51 - 71/91 (20 ms Packet size) 41 - 56/91 (10 ms Packet size) |
| DSL | 61 - 81/91 (20 ms Packet size) 49 - 63/91 (10 ms Packet size) | 74 - 90/91 (20 ms Packet size) 53 - 73/91 (10 ms Packet size) |

Dependence of DSP/CPU processing time TriplePlay-Results IAD_2_IAD

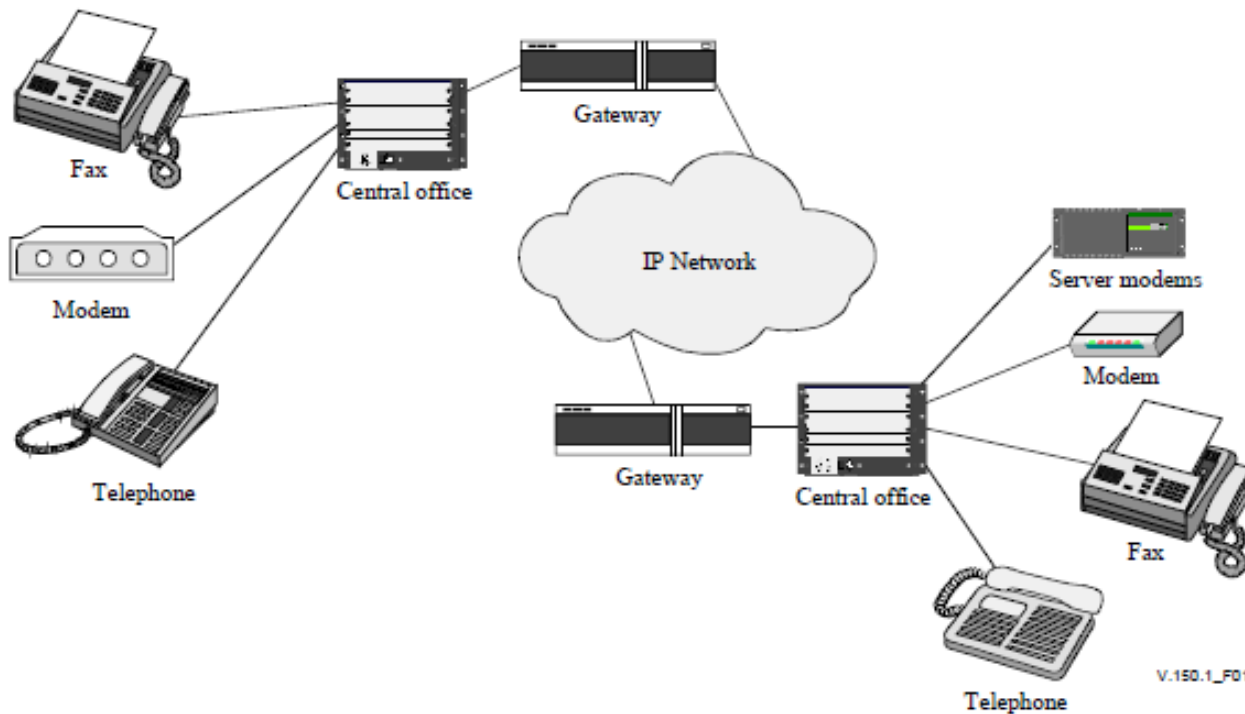
Test requirements : routed IAD-Konfiguration,
worst case two TV Streams,
Testlabor HSI- Connection and activated WLAN, VP-voice
upstream Bandwith 384 kBit/s.

| Supplier: | 1 | 2 | 3 | 4 |
|--|----------|---------|----------|---------|
| IAD_2_IAD MOS Value QoS two wire (PSTN) | | | | |
| 10 ms packetization | 4,2 | 4,2 | 4,0 | 4,2 |
| 20 ms packetization | 4,2 | 4,2 | 4,0 | 4,1 |
| Delay / Delay Range/ Std-Dev (each direction max values) | | | | |
| 10 ms packetization | 58/29/10 | 86/10/4 | 111/10/4 | 63/17/6 |
| 20 ms packetization | 66/35/11 | 95/0/0 | 114/13/6 | 66/10/4 |

A dark blue world map is centered in the background of the slide, showing the outlines of continents and oceans.

Transport of VBD

Typical modem over IP application



Transport of Voice Band Data over IP (1)

□ Possible Transport mechanism over IP

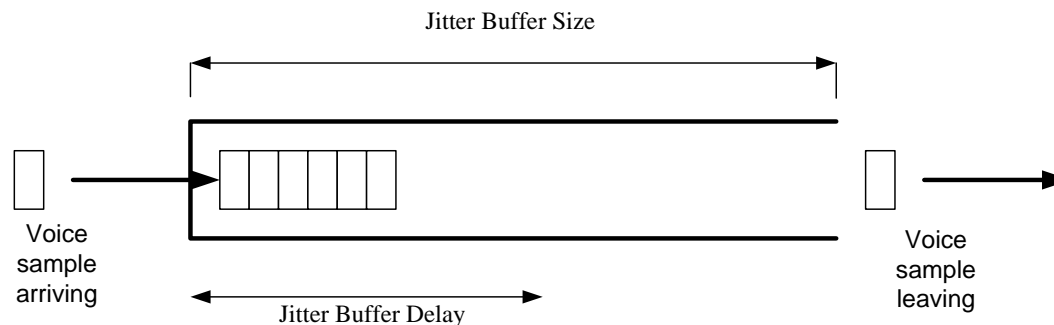
- G.711
- G.726 is capable of transporting without impairment at 1200 bps or lower speeds
- T.38 (Fax up to 14.4 kbit/s with Release 1)
- V.150 Modem-over-IP networks: Foundation
- V.150.1 § 8 Modem-over-IP networks: Procedures for the end-to- end connection of V-series DCEs
- V.151 Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay
- V.152 § 6 Procedures for supporting Voice-Band Data over IP Networks

Transport of Voice Band Data over IP(2)

- For Voice Band Data (VBD) mode of operation, the path between GW remains in a voice configuration. The modem signals are encoded using an appropriate speech codec suitable for the task, and the samples are transported across a packet network.
- VBD should:
 - Use a voice codec that passes voiceband modulated signals with minimal distortion.
 - Have end-to-end constant latency based on fixed jitter buffer.
 - Disable Voice Activity Detection and Comfort Noise Generation during the data transfer phase.
 - Disable any DC removal filters that may be integral with the speech encoder used.
 - The use of echo cancellers on a VBD channel. Upon detection of V.25 Answer tone with phase reversals Echo Control as per G.168 shall be disabled
- VBD should consider the appropriate application of:
 - Forward Error Correction (FEC) (e.g., RFC 2733) or other forms of Redundancy e.g., RFC 2198).

Values of Jitter Buffers for VBD

- ❑ In the case of a DSL line with 384 kbit/s uplink, 1024 kbit/s downlink the maximum Jitter downlink is 54 ms, maximum Jitter uplink is 9 ms
- ❑ The fixed jitter buffer of 60 ms (JBD = 30 ms) is required
- ❑ In the case of a MSAN line the maximum Jitter is 2 ms in the core
- ❑ The fixed jitter buffer of 20 ms for VBD (JBD = 10 ms) is required.



Transit Scenario TDM – IP (1)

- During the TDM- IP Transit Scenario in a connection can be national involved more TDM – IP segments.
- They are caused due pre-selection scenarios, call diversion, different providers not supporting FoIP and MoIP transmission
- Depending from the access type the End-to-End delay can be over 200 ms for VBD
- Default fixed jitter buffer values: CISCO 200 ms, GENBAND 100 ms
- As described in G.114 some highly interactive tasks are affected by delays below 100 ms (e.g. UDI applications, modems, alarm devices)
- T.38 is not sensible on the delay, but during the TDM- IP Transit Scenario with more TDM – IP segments T.38 does not work. The cause are timers of the T.38 Implementations

Transit scenario TDM – IP (2) Echo Canceller

- If in accordance with ITU G.131 and Q.115.1 the "one-way talker echo transmission path" exceeds the 25 ms limit, echo canceller (EC) according to ITU G.168 shall be used.
- At the network boundary the Echo Canceller Procedure described in Q.761 and Q.764 shall be used
- As during the TDM- IP Transit Scenario more TDM – IP segments can be involved, more EC in the TDM – IP segments shall be avoided.

Further Limitations

- ❑ **V.90 Analog Modems - the highest upstream speed over xDSL is 40 kbit/s (it should be 54 kbit/s), 28.8 kbit/s (it should be 33.6 kbit/s) (output STF 392)**
- ❑ **V.90 Analog Modems – the highest upstream speed over MSAN is 52 kbit/s**
- ❑ **V.34 Fax transmission using adaptive jitters buffer is not possible**
- ❑ **The fax transmission increases if the network delay increases**
- ❑ **Some modems are not using the ANS Tone of 2100 Hz, the control of Jitter Buffer and EC are not possible**

Possible solutions to minimize the delay for the Fax/Modem (VBD) Transmission

- The Jitter in a well developed MPLS core network is usually ≤ 2 ms
- The Jitter Buffer values can be dimensioned for voice calls smaller or equal one packet size due adaptive Jitter Buffer
- The jitter between the MSAN for VBD calls can be dimensioned with 20 ms
- In case of jitter due to low bandwidth access the signalling traffic can be separated from the media traffic by ATM interleaving mechanism.
- In case of ETH the upload/download speed should be ≥ 1024 kbit/s as described in Y.1541 to minimize the jitter to 9 ms which can be handled with a buffer size of 20 ms

Fax/Modem (VBD) Transmission from VoIP Platforms which produce jitter higher than the fixed Jitter Buffers used in the network

- **Special attention should be given to traffic from other VoIP platforms, which produce a jitter that cannot be compensated by the jitter buffer (e.g. caused by transport of signalling and media on low bandwidth links).**
- **VBD and UDI traffic shall be separated from the NGN Voice Network with border elements (SBC or IP- IP GW) which compensate the jitter higher than fixed jitter buffer size. The border element has to control the jitter buffer according to the service (VBD/UDI or speech). Typical applications include traffic from international exchanges or IP-IP interconnection with other VoIP platforms.**

Transport of ISDN over xDSL

Timing requirements for ISDN PBX

- ISDN PBX takes up the clock from the TDM network
- The clock is needed for synchronization of the PBX or more PBX locations
- If the PBX locations are not synchronized the PBX systems usually crashes
- As the DSL accesses are usually not synchronized a separate synchronization is needed
- To ensure the timing requirements the packet network (access and core) must be synchronized according the ITU-T Recommendation G.8261/Y.1361

Questions?

Thank you for your attention