On-line Monitoring Of VoIP Quality Using IPFIX

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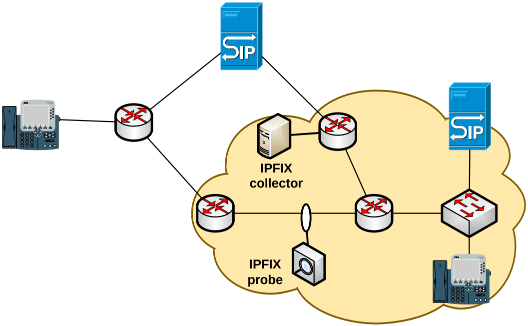
On-line Network Monitoring



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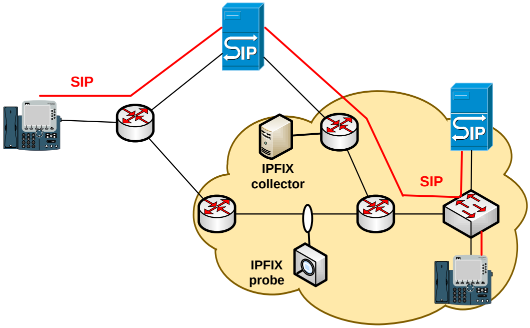

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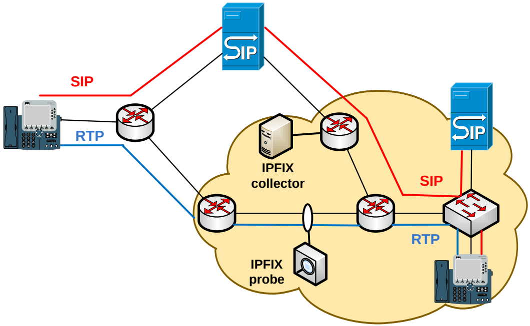

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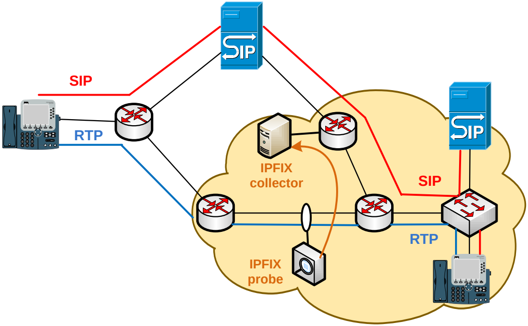

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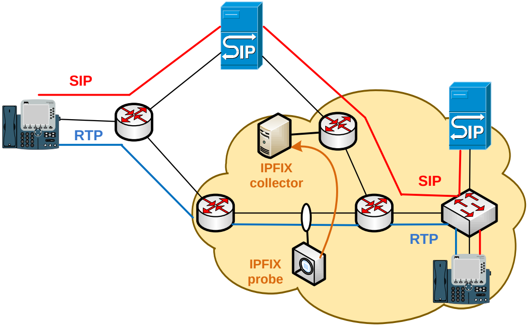

Requirements for VoIP Quality Monitoring

- Assessment of VoIP quality of users' calls
- Long-term monitoring of the quality with history
- Integration with common network monitoring systems
- End communication point outside the monitoring network

Talk overview

1. Motivation
   - On-line VoIP Quality Monitoring
   - Current Methods
2. Computational Model
   - Simplified E-Model
   - Implementation Issues
3. Results
   - IPFIX Extension
   - Result validation
4. Conclusion
   - Future Work

Existing methods for evaluating quality of VoIP calls

Subjective methods

- Listening-opinion tests (ITU-T P.800, ACR)

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- On-line monitoring requires non-intrusive passive approach.

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$$R = R_o - I_s - I_d - I_{e-eff} + A$$

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Simplified E-Model (using default values for R_o, I_s, and A)

$$R = 93.2 - I_d - I_{e-eff}$$ [i]

[i] L. Sun et al.: *Guide to Voice and Video over IP*, Springer, 2013, p. 147

Simplified E-Model used in this work

Computation of the Simplified E-Model

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- Depends on a codec (B_{pl}), network delay (d), packet loss (P_{pl}), and network jitter (P_{jitter}).
- Values are computed on-the-fly using packet headers.

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4. Export of Flows

IPFIX records extended by packet loss, jitter, delay, R-factor, and MOS

RTP Detection and Codecs Classification

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  - UDP only, ports > 1024, min. header length, RTP version 2, RTP PT

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Codec	Payload Type	Δ time ($TS_{i+1} - TS_i$)	Payload Size	Δ ratio
G.711 μ-law	0	160	160	1:1
G.711 A-law	8	160	160	1:1
Speex8	dyn	160	20	8:1
Speex16	dyn	320	52	80:13
GSM	3	160	33	160:33
G.722	9	160	160	1:1
G.722.1	dyn	320	60	16:3
G.723.1-5k	4	240	20	12:1
G.723.1-6k	4	240	24	10:1
G.726-16	dyn	80/240	20/60	4:1
G.726-24	dyn	80/240	30/90	8:3
G.726-32	dyn	80/240	40/120	2:1
G.726-40	dyn	80/240	50/150	8:5
G.729	18	160	20	8:1
G.729a	18	160	20	8:1
G.729b	18	160*	20*	var.
AMR-WB	dyn	320	62	160:31
AMR-12k	dyn	160	33	160:33
Silk8	dyn	320	var.	var.
Silk16	dyn	640	var.	var.

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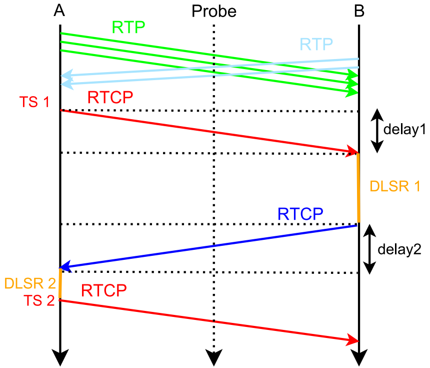

- Using RTCP Packets: $RTD_{12} = TS_2 - DLSR_2 - DLSR_1 - TS_1$
  - Average RTD delay over all RTCP packets considered.

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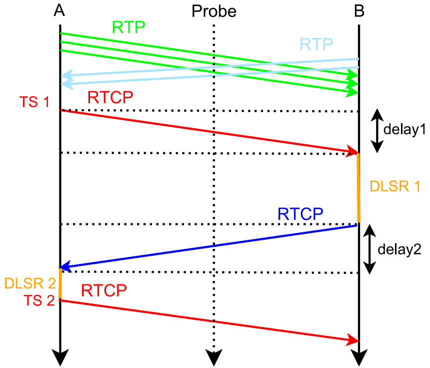

- Using RTCP Packets: $RTD_{12} = TS_2 - DLSR_2 - DLSR_1 - TS_1$
  - Average RTD delay over all RTCP packets considered.
- Using RTP packets only: $d_i = \frac{d_{i-1}.(i-1)+J_i}{i}, d_1 = 0$ [vi]
  - Delay d computed iteratively over subsequent packets.

[vi] see http://www.voipmonitor.org

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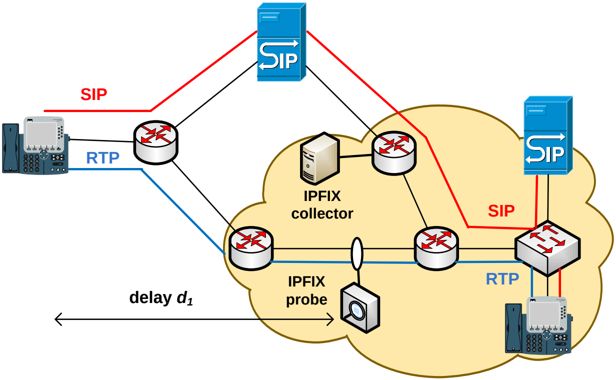

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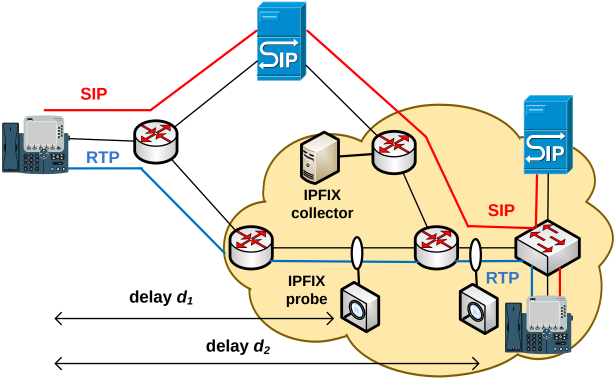

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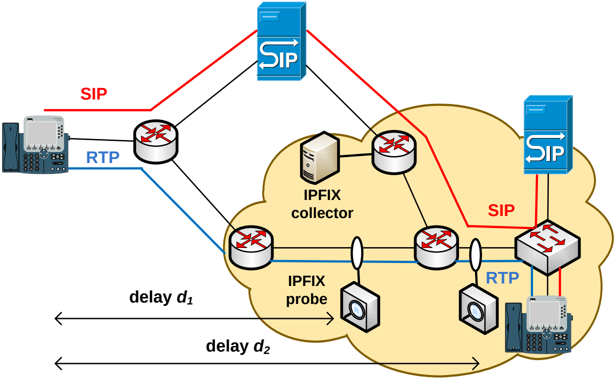

- RTCP delay computed at end-points ⇒ more accurate

Implementation results

IPFIX extension: Example

- Packet Loss (0), Jitter (0.201), Delay (0), R-Factor (93.2), MOS (4.409), Quality (Excellent), Flow Type (RTP)

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```
⊟ Cisco NetFlow/IPFIX
    Version: 10
    Length: 380
  ⊞ Timestamp: Apr  2, 2014 12:35:26.000000000 CEST
    FlowSequence: 0
    Observation Domain Id: 0
  ⊟ Set 1
      FlowSet Id: (Data) (258)
      FlowSet Length: 364
    ⊟ Flow 1
        Octets: 178600
        Packets: 893
      ⊞ [Duration: 17.839000000 seconds]
        InputInt: 0
        OutputInt: 0
        IPVersion: 04
        SrcAddr: 91.221.212.167 (91.221.212.167)
        DstAddr: 192.168.1.4 (192.168.1.4)
        IP ToS: 0x00
        Protocol: 17
        SrcPort: 26456
        DstPort: 7078
        Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 721: Value (hex bytes): 00
        Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 722: Value (hex bytes): 3f 61 fb 9a
        Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 723: Value (hex bytes): 00 00 00 10
        Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 724: Value (hex bytes): 42 b9 03 00
        Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 725: Value (hex bytes): 40 8c a7 83
      ⊞ [Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 726: Value (hex bytes): 47 6f 6f 64
      ⊞ [Enterprise Private entry: (VUT BRNO, faculty of EE and CS) Type 727: Value (hex bytes): 52 54 50 00
    ⊞ Flow 2
    ⊞ Flow 3
```

Comparison with other software

- VoIP metrics computed by Wireshark (W), PacketScan (PS), VoIPmonitor (VPM) and IPFIX plugin (IPF)

Metric	W	PS	VPM	IPF
RTP Jitter (ms)	8.10	7.00	8.00	7.91
RTP Loss (%)	0.00	0.00	0.00	0.00
RTP Delay (ms)	–	0.00	–	0.00
R-factor	–	93.0	–	92.99
MOS	–	4.20	4.50	4.41
RTCP Jitter (ms)	–	0.00	76.10	76.06
RTCP Loss (%)	–	0.00	0.00	0.00
RTCP Delay (ms)	–	9.23	–	15.00
R-factor	–	–	–	90.09
MOS	–	–	–	4.34

Future research

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  - Correlation among different quality parameters over time period

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- Advanced methods how to measure one-way delay

Monitoring in production networks

- Long-term monitoring of VoIP calls quality using IPFIX
  - Correlation among different quality parameters over time period
- Monitoring of VoIP calls quality using IPFIX probes in different nodes
  - Correlation among different quality parameters over time period

Thank you for your attention.