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QoS Measurement in Int.nal Voice Service

presented by

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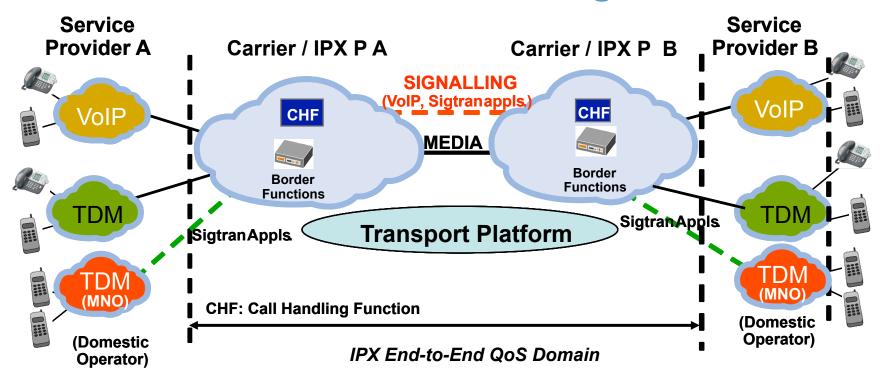




- The GSMA Requirements
- The RTCP methodology
- Suggested Methodologies
 - "Aggregation" Scheme
 - "Media Loopback" Methodologies
- Proposed i3f guideline



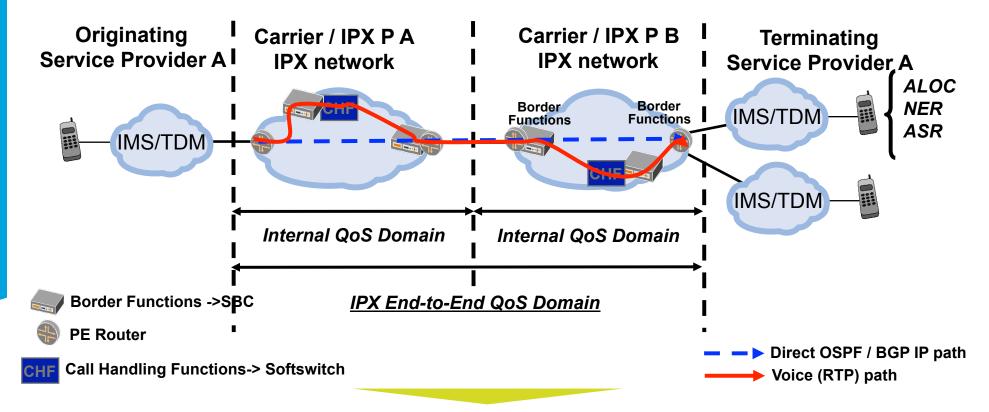
General Reference Configuration



- More than 2 carriers can be present in the end-to-end segment
- Border functions are not always located at the carrier's network edge -> €€ IMPACT
- Measurement of traditional Voice (e.g. ALOC, ASR, NER) parameters spans up to final user -> NO PROBLEM

GSMA requirements (PVI AA.81)

 The RTP path of the voice service may be different than the "direct OSPF/ BGP" IP path of the data service



- GSMA (AA.81): "Transport (IP) Quality... shall be measured and reported for the path carrying PVI service" -> Mutual trust among Carriers / IPX Providers
- RT Delay, Packet Loss, Packet Jitter -> MOS (Mean Opinion Score)

HOW to achieve QoS Control end-to-end? Active vs. Passive Measurement Methodologies

Active

Methodology: Measurement by actively setting up test sessions across test domain

Pros:

- Can also provide MOS_{COE}
- Control of measurement domain
- Other metrics available

Cons:

- Not always representative of real traffic path
- Large number of sessions may be required to provide full coverage e.g. N² problem

Passive

Methodology: Measurement by passively monitoring traffic sessions across the test domain

Pros:

- MOS_{COE}
- Accurate representation of real performance
- Easy to configure
- Measurements easy to analyze

Cons:

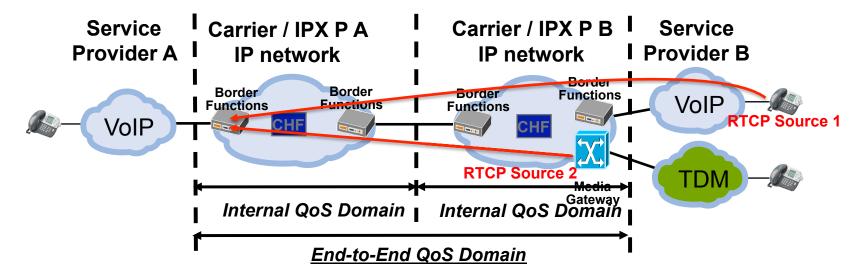
- Uncertain control of measurement domain
- Limited diagnostic ability

Whatever the methodology is, carriers have to invest at the network layer and for development of the OSS/BSS capabilities

An immediate answer: using RTCP (RTP Control Protocol)

PROS:

- based on real traffic -> accurate measurement
- no need of new standard
- available (as an option) in SBC
- calculates MOS_{COE} from the R-Factor /E-model
- delay, jitter, loss from RTCP sender and receiver reports



It is not possible to distinguish the location of the RTP end-point

No reliable, accurate solution is available



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i3f Solution Requirements

The selected methodology should be:

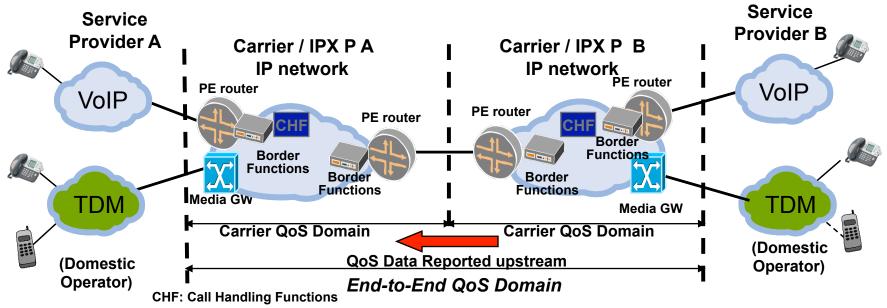
- "controllable" for a full set of transport parameters (provide measurement for identifiable domain(s))
- active or passive
- not vendor proprietary but with a broad industry support
- preferably integrated into existing equipment
- based on recognized standard i.e. from IETF/ITU-T
- relatively easy to integrate into OSS/BSS chain
- with a limited/reasonable deployment overhead
- capable to provide MOS_{CQE}
- capable to assist with SLA monitoring and troubleshooting
- capable in "supporting" at least Bilateral and IPX use cases

Activity carried out in parallel with vendors (2 open workshop plus constant contacts) and MNO representatives on behalf GSMA

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QoS Methodologies: Quality Measurement Aggregation

• Each carrier monitors quality across their internal QoS domain using their chosen mechanism (e.g. probe servers or RTCP)

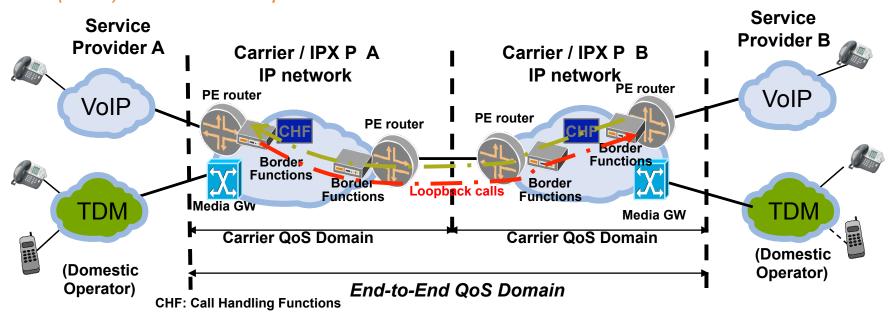


KPIs are computed **AGGREGATING** the values measured by each carrier / IPX P.

- Delay: estimated by adding up the delay of each carrier network
- Loss: estimated as the complement of joint probability of the event "Successful Packet Transmission"
- Jitter: no aggregation scheme. Jitter measured by the last carrier is provided

QoS Methodologies: Media Loop-back

 The Border Functions (i.e. SBC) make testing calls to dummy numbers terminated to the last SBC present in the carrier domain, based on *Draft IETF* mmusic-media-loopback-18 "An Extension to the Session Description Protocol (SDP) for Media Loopback"



- Encapsulated scheme: encapsulated source RTP sent back to sender
- for MOS_{CQE} (R-Factor / E-model) computation: one way delay is needed so the two transmission paths should be symmetrical



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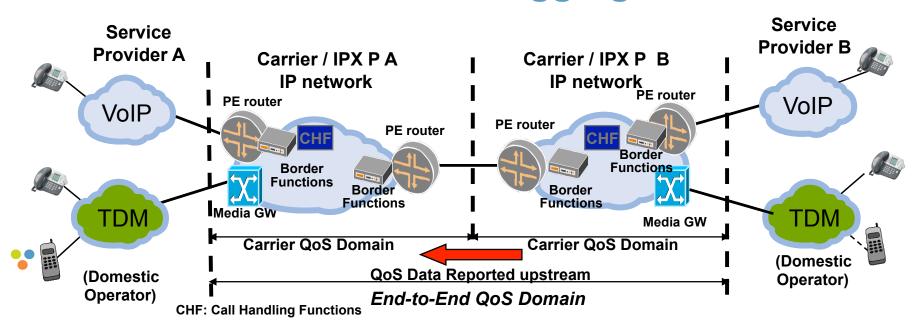


Proposed i3f guideline: 5 principles

- P1: i3 recognises GSMA requirements of E2E QoS measurement. This task, today, is challenging
- P2: In case of a 1 single Carrier / IPX Provider domain the solution is based on
 - Delay via RTCP (but RTP via external probes can be used)
 - Loss via RTP
 - Jitter via RTP
- which allows to compute MOS_{COE} via R-Factor/E-Model
- P3: In case of a 2 Carriers / IPX Providers domain the today recommended solution is based on Aggregation
- P4: In case of a 2 Carriers / IPX Providers domain the future recommended solution is based on media Loopback – encapsulated scheme
- **P5**: Service parameters (ASR, ALOC, NER,) can be measured as per requirement following "traditional" schemes

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P3: TODAY solution for multiple networks domain based on Aggregation



Methodology widely used in current IPX trials

Some issues:

- How to pass QoS data?
- Jitter values cannot be worked out by aggregation





P4: FUTURE solution for multiple networks domain based on Media Loop-back

Some issues:

- Carrier B has to allow dummy calls -> mutual trust
- Which number to be called (Tel URI or Sip URI)? And belonging to whom?
- Large number of loopback sessions is required. It increases with a quadratic law.

Assuming fully meshed interconnection network

Overestimation

N_IPXP	20,00
N_POP/IPXP	8,00
N_Call/h	2,00
minutes/Call	0,50
N_Calls/day/pop	7.632,00
N_calls/day/IPXP	61.056,00
N_minutes/day/IPXP	30.528,00
N_minutes/month/IPXP	915.840,00
N_minutes/year/IPXP	10.990.080,00
N minutes/day/IPXdomain	610.560,00
N_minutes/month/IPXdomain	18.316.800,00



Thank You

