

Qrystal

THE SMARTEST VIEW ON VOICE QUALITY

1

A SHORT INTRODUCTION

About Voipfuture

Founded: 2007

Located: Hamburg, Germany

Mission: Premium voice
and media quality

Alaska Communications
Aspect
ATOS
brightOne
China Mobile
Eircom
Ericsson
Etisalat
Granite
Metrobank
Mobistar



Voipfuture
The RTP monitoring
pioneers

Partner
Post Technologies
RZF-NRW
Sipgate
StarHub
UniCreditGroup
UTA
Vodafone
Vodacom



Voipfuture | Intro

Voipfuture developed a new approach to voice monitoring in IP Networks that combines the power of RTP media quality analysis with advanced SIP control plane

This approach puts emphasis on the idea of proactive monitoring, and not only troubleshooting when complaints arrive.

Voipfuture technology puts quality at the center of voice network operation.

Voipfuture Qrystal | What Is It About?

Voipfuture Qrystal is all about visibility. It pays to know exactly which quality one delivers.

As the voice quality monitoring pioneer, Voipfuture delivers the clearest possible view on the Voice over IP packet flow.

- The OSS team sees all control and media plane data in one spot
- They will know what is going on
- And they will understand why it is happening

A crystal clear view on quality is not only about MOS values and hang-up causes. It is about

- How quality precisely develops during a call
- What exactly is affecting the user experience
- Who is affected by underlying network issues

Which in sum means saving, avoiding churn, and efficiently handling operations.

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TECHNICAL LEADERSHIP

Voipfuture | Innovation Leadership

First integrated solution for RTP & SIP monitoring

- Voipfuture is pioneer in RTP Monitoring
- First to provide fixed time slicing technology
- Automatic root cause analysis & pattern matching technology
- First to enable waveform analysis on massive live traffic
- Experts in massively parallel computing on standard hardware –
World's first passive probe capable of processing 100,000 cc
- Database cluster architecture supporting big data analysis

23 patents/applications cover core technology, i.e. root cause analysis, stream-to-call correlation, ...

Voipfuture | The Unique Technology

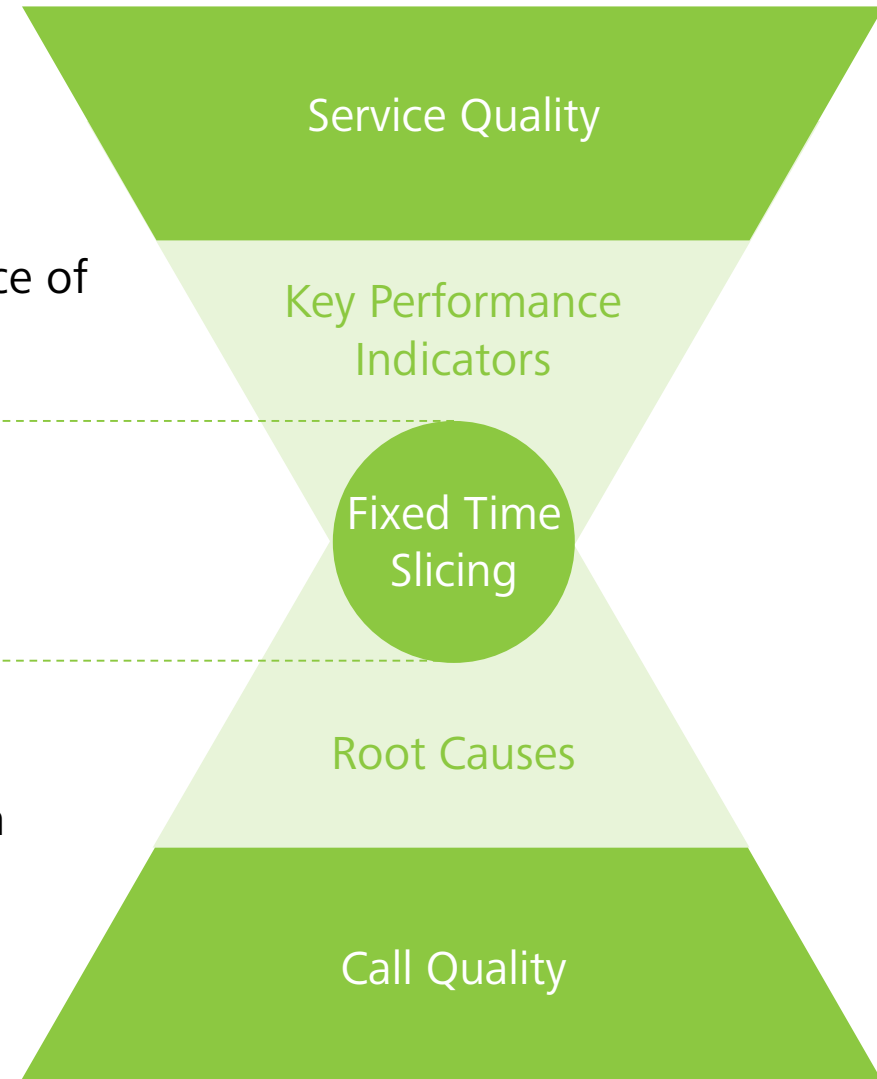
Aggregated view

↑ KPI to express precise performance of service and infrastructure

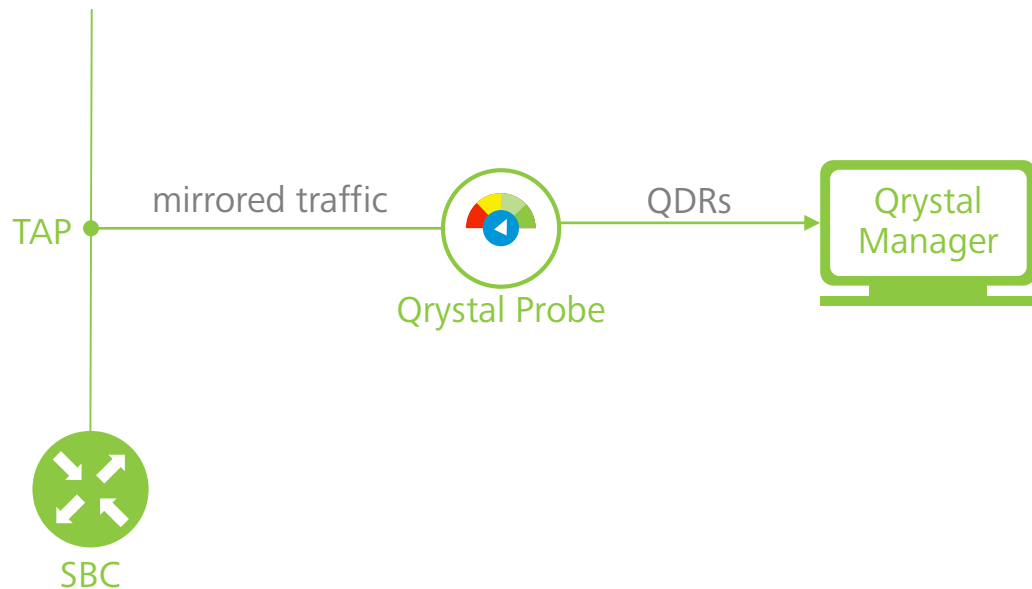
Quality summaries each 5 seconds allow an adjustable depth of view

Detailed view

↓ Root Causes to pin point problem sources and impact level



Voipfuture Qrystal | In a Nutshell



- Passive, mid-point VoIP monitoring solution
- Integrated voice quality assurance and root cause analysis
- Probes analyze all packets on a link in real-time
- Carrier-grade solution
 - Probes monitor multiple 1 GbE / 10 GbE links at wire rate
 - Central Application Manager maintains detailed data from billions of call minutes
- Cost-effective use of COTS hardware (HP, IBM, Dell, ...)

System Scope







Media Plane Concept and Standards

- Voipfuture RTP monitoring
 - Full traffic: Complete 24/7 analysis of all packets
 - All calls: RTP monitoring at full line rate
 - Both directions: RTP analysis for both directions of each call
 - Fixed Time Slices: Accurate RTP quality for every 5 seconds
 - Real-time classification of RTP streams: Grouping by probes, trunks and numbering plans
 - RTP / SIP Correlation

In the realm of the following standards:

- RFC 3550 (RTP/RTCP)
- RFC 2833/4733 (DTMF)
- ITU G.107 (E Model)
- ITU P.800 (MOS)
- ITU P.564
- ITU P.561 P.562 P.563 (WFA)

The Detailed View: Root Cause Analysis

OVERALL RESULT (OTHER PRODUCTS)	INTERPRETATION	TAKEAWAY
 ∅ MOS 3.9	“okay quality”	None. Result does not necessarily tell the truth – and it provides no insight for problem solving
STREAM DETAILS (VOIPFUTURE)	INTERPRETATION	TAKEAWAY
	quality-induced termination	bad user experience
	periodic jitter	quality of service to be improved
	network overload	bottle neck/element capacities to be checked

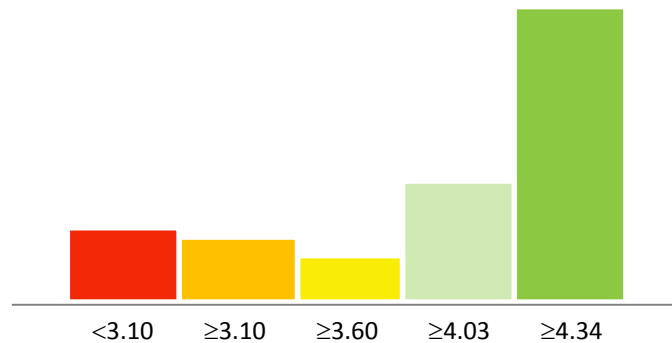
The Aggregated View: Accurate MOS for Each Fixed Time Slice



Qualified time slices for one stream of a call

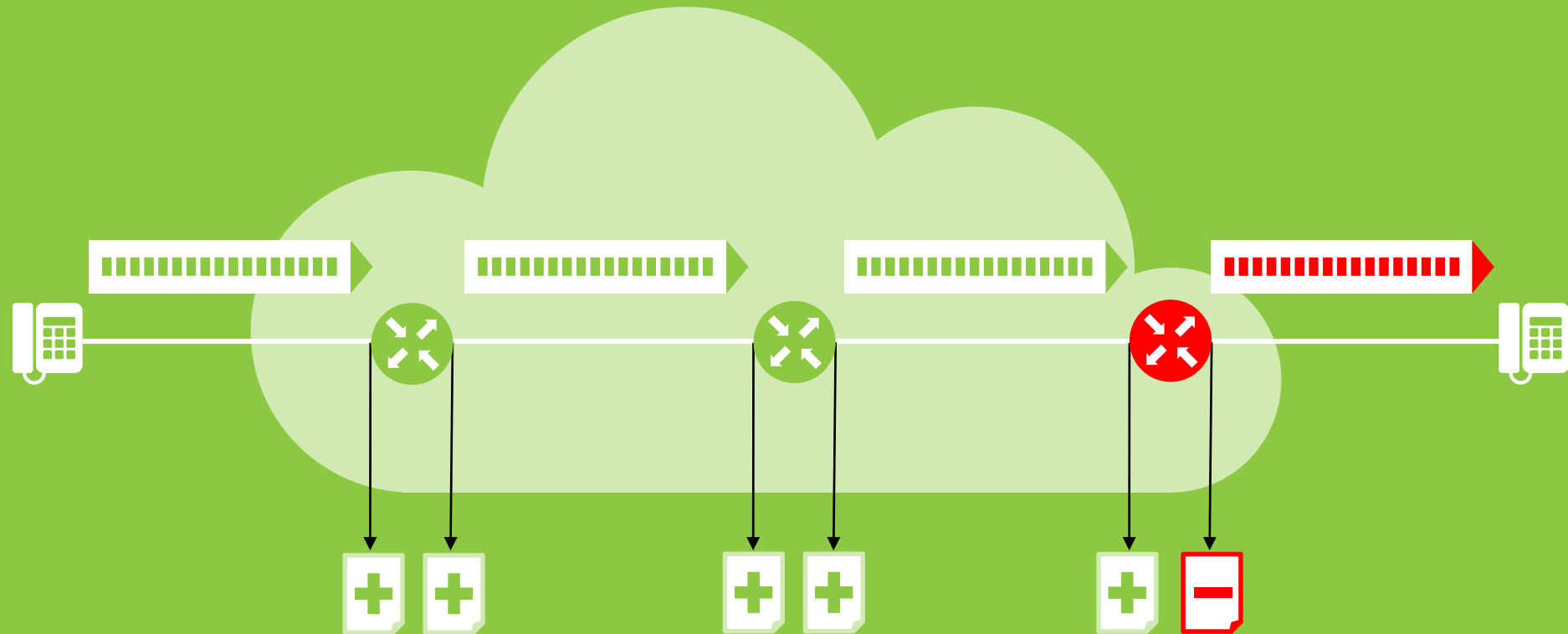


Quality for 1 day



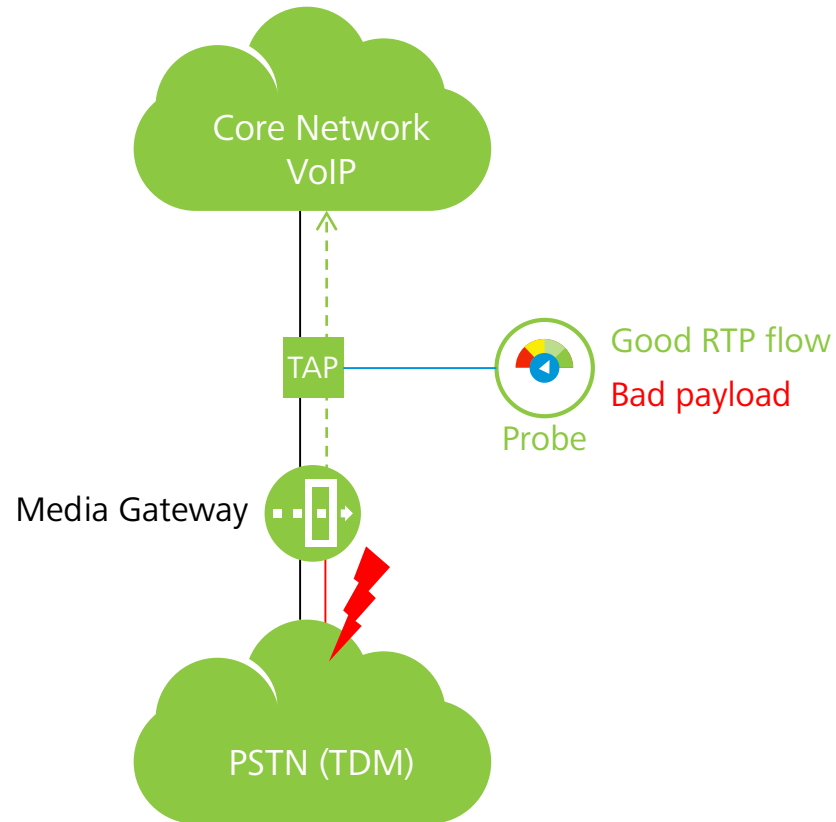
- Call quality evaluation based on 5 second slices – upstream and downstream
- MOS value for every slice
- Accuracy independent from minute volume
- Results can be aggregated by MOS classes and related to single carriers and routes

The Benefit of Network Segmentation



- Multiple monitoring points show call quality before and after entering a network element/segment (only one direction shown)
- Mid-point monitoring quickly isolates impairment sources

Payload Analysis | Looking Even Deeper for Echo

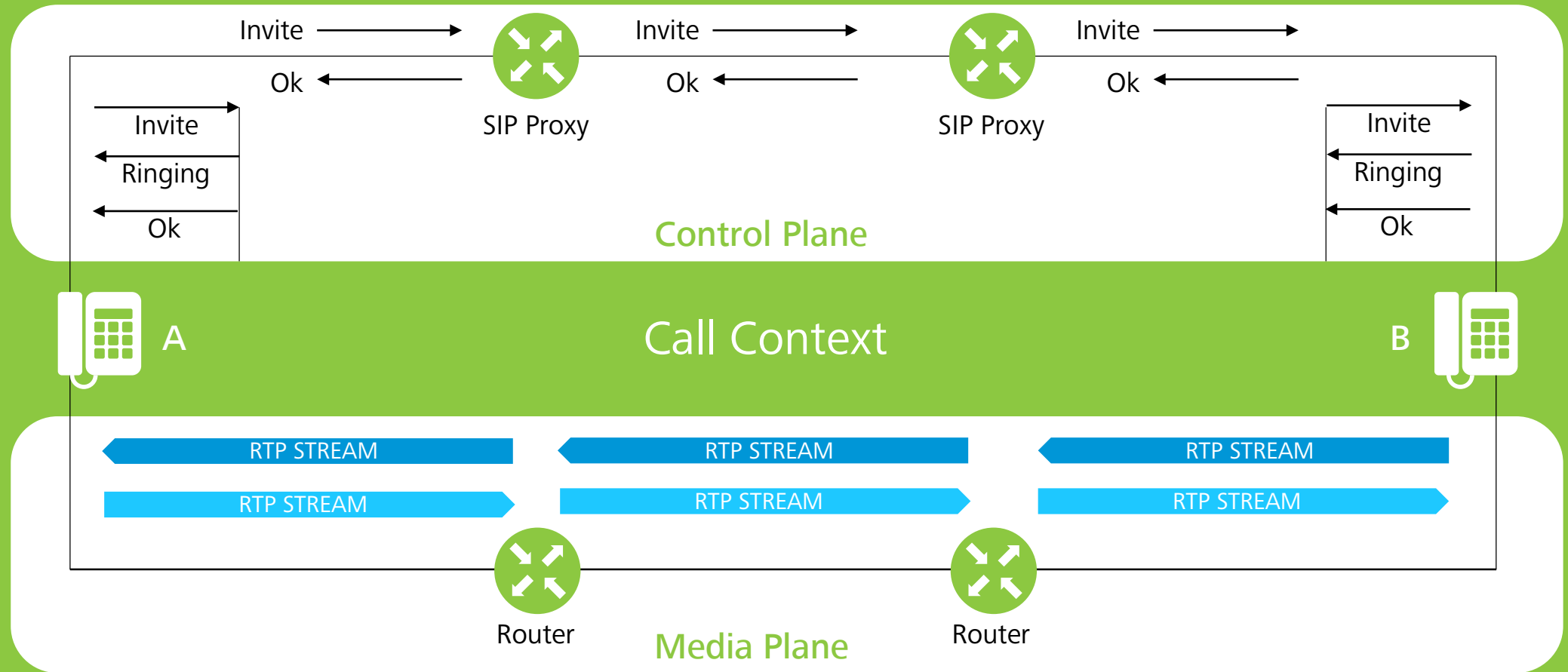


- An RTP stream may be fine, but voice quality problems may be hidden in the RTP payload
- Such impairments are typical for hybrid TDM/VoIP call flows
- Detection requires analysis of voice waveform



- WFA introduces indicators for
 - Echo,
 - noise,
 - clipping and
 - silence

The Combined View: Correlated SIP & RTP Information



System Scope



Control Plane Concept and Standards

- Qrystal SIP monitoring
 - All signaling sessions: SIP monitoring at full line rate
 - Full set of signaling KPIs, incl. SER, SEER, call duration and delay KPIs
 - Selective and full recording of control plane
 - Multi-dimensional grouping (route monitoring by number plan)
 - SIP registration statistics by domain
 - RTP / SIP Correlation

- In the realm of the following standards:
- RFC 6076 (SIP KPI)
 - RFC 3261 (SIP)
 - RFC 7329 (Session ID)
 - RFC 7315 (P-Access-network-info PANI)
 - RFC 3226 (reason Header)
 - ITU Q.850 (hang-up cause)

SIP – Header / Session Information Metrics RFC 3261

Reporting of:

- From / To
- Call-ID
- Session-ID (RFC 7329)
- Automatic number normalization
- Local / remote SIP user agent information
- P-charging vector (RFC 7315)
- P-access-network-info (PANI) header (RFC 7315)
- Reason Header / hangup cause code / Q.850 (RFC 3326)
- Call setup status code
- Authentication Success/Failure reporting

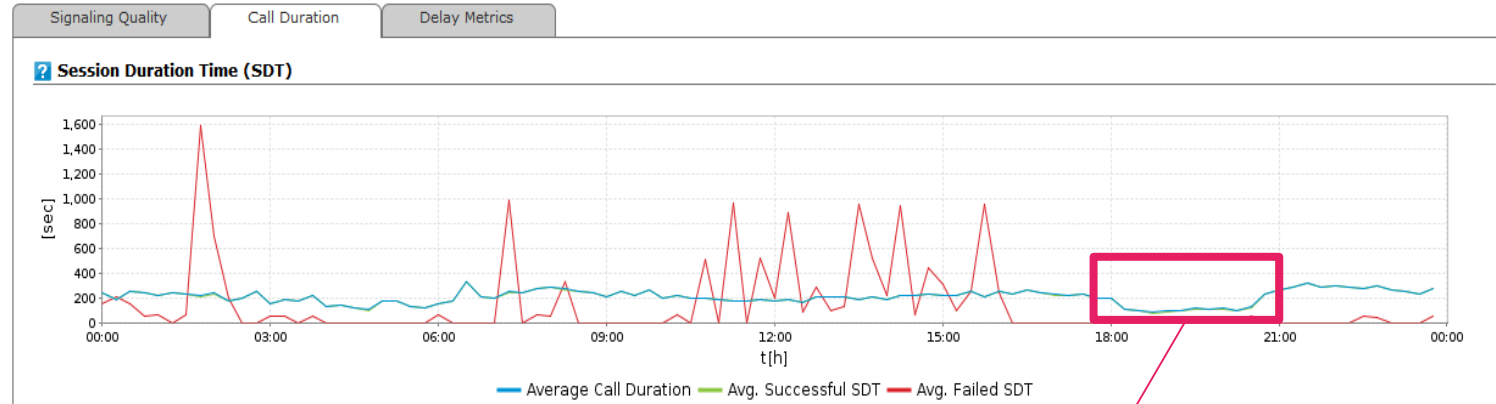
The screenshot displays the Asterisk Signaling Information window. It is divided into several sections:

- Start/End/Duration/Connect:** Start: 2016-02-12 08:57:06.603, End: 2016-02-12 08:58:30.741, Duration: 00:01:24.138, Connect: 2016-02-12 08:57:07.705
- Session Information:** Session: 00:01:23.003, Duration: 00:00:01.102, Request Delay: 00:00:01.102
- Call ID:** 7ba713510a31a50620b3c033579483ca@...de
- Charging-Vector:** SIP-Trunk: SBC <-> [redacted], SIP: A → B
- Call State:** Regular call termination
- SIP Messages:** A list of SIP messages with timestamps and directions:
 - 2016-02-12 09:57:06.603 - INVITE (w/SDP) - 172.28.52.11 → 2
 - 2016-02-12 09:57:06.639 - 407 Proxy Authentication Required/INVITE - 172.28.52.11 ← 2
 - 2016-02-12 09:57:06.640 - ACK - 172.28.52.11 → 2
 - 2016-02-12 09:57:06.640 - INVITE (w/SDP) - 172.28.52.11 → 2
 - 2016-02-12 09:57:06.690 - 100 Trying/INVITE - 172.28.52.11 ← 2
 - 2016-02-12 09:57:07.705 - 200 OK/INVITE (w/SDP) - 172.28.52.11 ← 2
 - 2016-02-12 09:57:07.705 - ACK - 172.28.52.11 → 2
 - 2016-02-12 09:58:30.708 - BYE - 172.28.52.11 → 2
 - 2016-02-12 09:58:30.741 - 200 OK/BYE - 172.28.52.11 ← 2
- Call Details:** Hang-up Cause: (16) Normal call clearing, Final Response: (200) OK, Call terminated by: Calling Party, Authentication: Authentication success, Mode: [redacted], FAX Support: none
- Call State Legend:** Established Call (blue dot), Successful Call (green dot), SDP violation (grey dot), SDP rejection (grey dot), Late SDP (blue dot)
- Calling Party Info:** Calling IP and Port: Asterisk External:5060, Calling MAC: Asterisk External, P-Asserted-Identity: --, Calling Number: 004943478000770, Calling Numbering Plan: Test Customer A, SIP From: sip:[redacted].de, Calling User Agent: FPBX-13.0.61.4(11.21.1), Calling Audio IP and Port: sip.voipfuture.com:17276, Calling SDP: m=audio 17276 RTP/AVP 8 101, c=IN IP4 212.126.220.244
- Called Party Info:** Called IP and Port: [redacted], Called MAC: 38:63:bb:3b:21:e3, P-Called-Party-ID: --, Called Number: 004943478009887, Called Numbering Plan: Test Customer B, SIP To: sip:043478009887@[redacted], Called User Agent: [redacted], Called Audio IP and Port: [redacted]:624, Called SDP: m=audio 24624 RTP/AVP 8 101, c=IN IP4 [redacted]

SIP End-to-End Performance Metrics RFC 6076

Reporting / Statistic of:

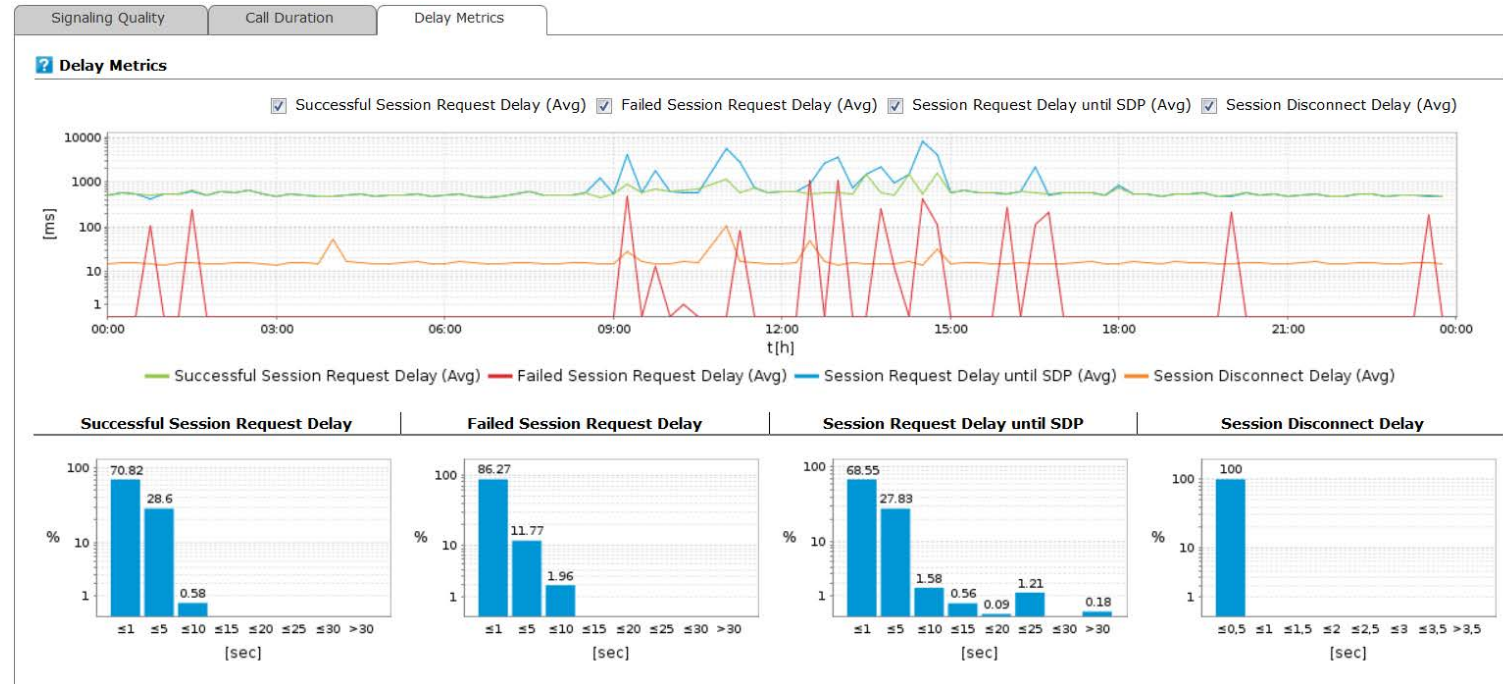
- Registration request delay
- Ineffective registration attempts
- Session request delay
- Session disconnect delay
- Session duration time
- Session establishment ratio (SER)
- Session establishment effectiveness ratio (SEER)
- Ineffective session attempts
- Session completion ratio (SCR)



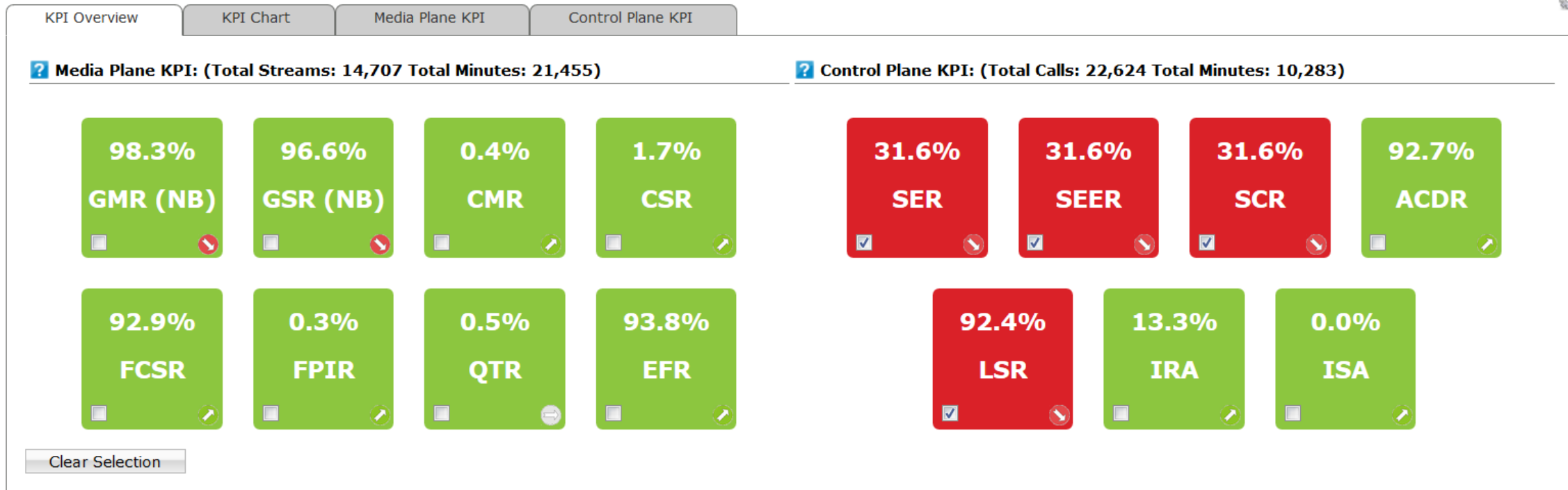
SIP End-to-End Performance Metrics RFC 6076 and Voipfuture specific metrics

SIP Session Statistics providing:

- Post dial delay as average and histogram of time distribution
- Delay until reception of first SDP (fraud detection) as average and histogram of time distribution
- Session disconnect delay as average and histogram of time distribution
- Session duration time as average and histogram of time distribution
- Call attempts
- SIP protocol errors grouped by severity (good, problematic, failed)



KPI Overview | Media and Control Plane

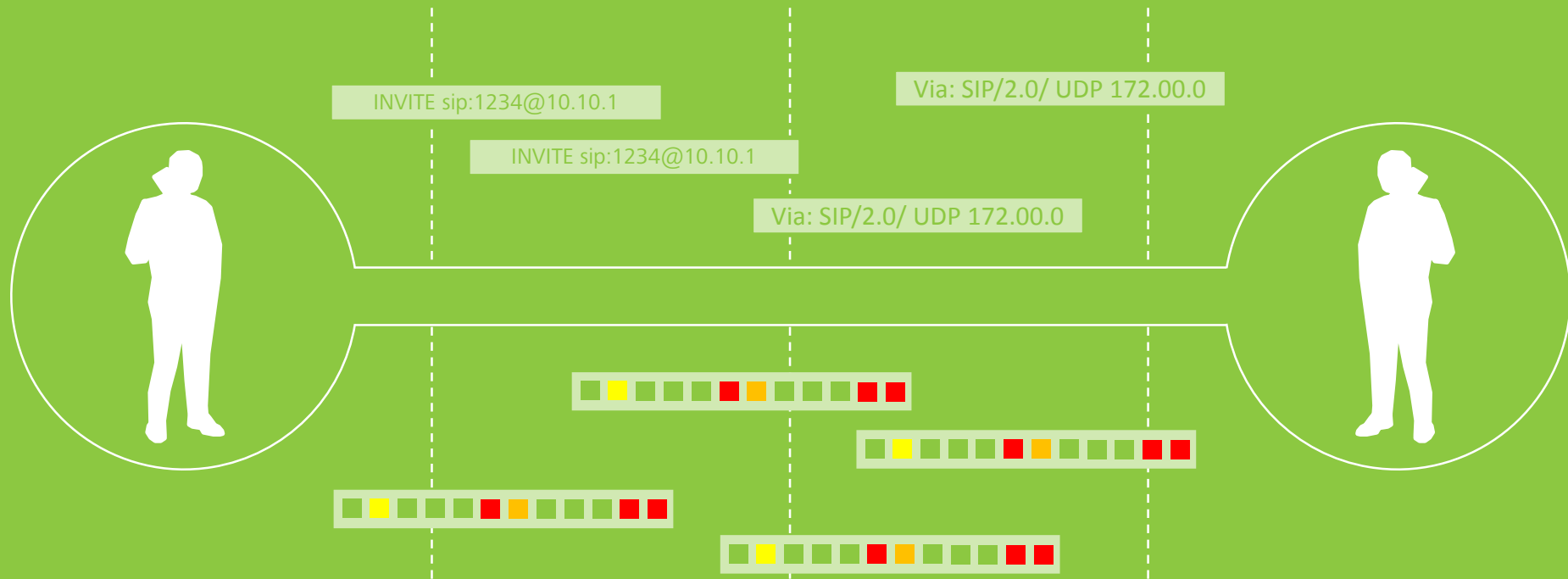


Unique view to fully control voice and service quality for excellent customer experience

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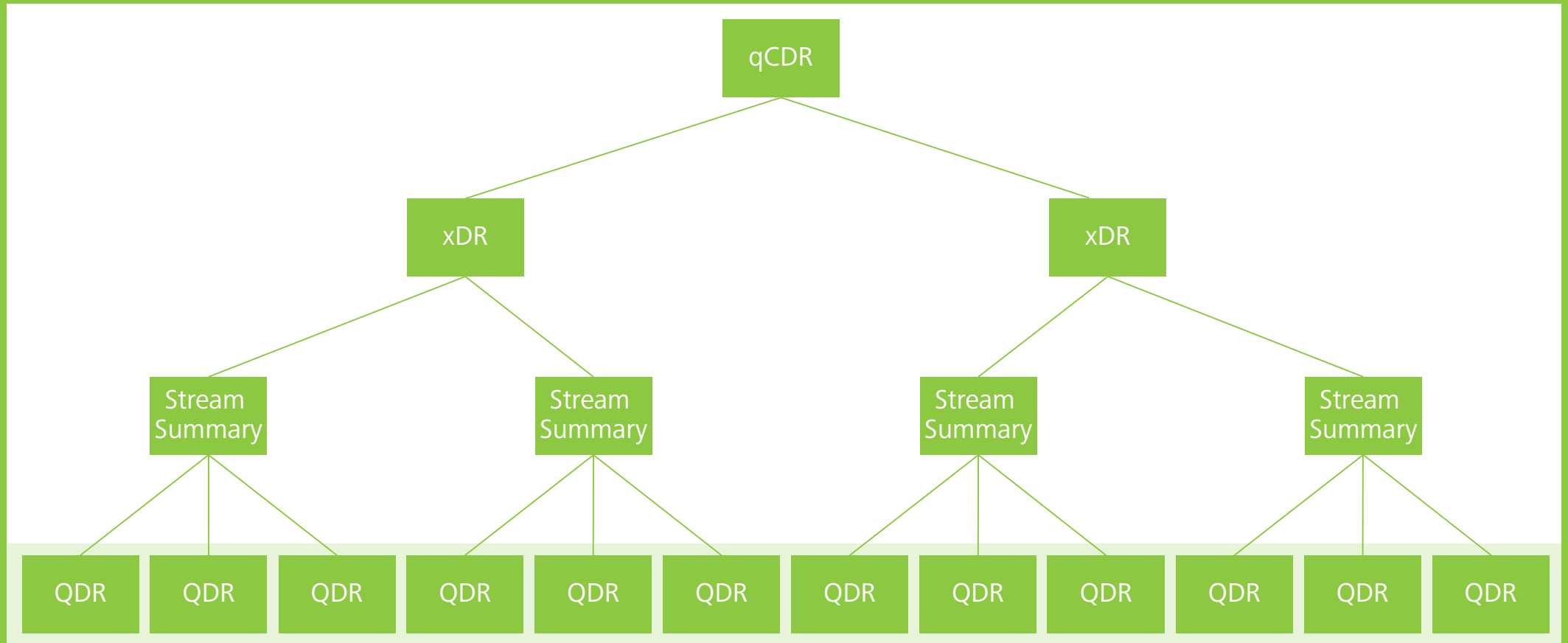
CRYSTAL CONNECT

The aim | One data record per call



A unified view on control & media plane
covering all measurement points

The analytical foundation | Quality-enriched CDRs



Four data sets combined | More intelligence per CDR

1 Basic Call Data

Base set containing all information one would expect from a normal CDR (call party numbers, start time, duration etc.)

2 User Experience

User experience set with R-Factor/MOS histograms, QUIT and other data related to the media quality and the user experience

3 Troubleshooting Details

Troubleshooting set containing all columns relating to our indicators

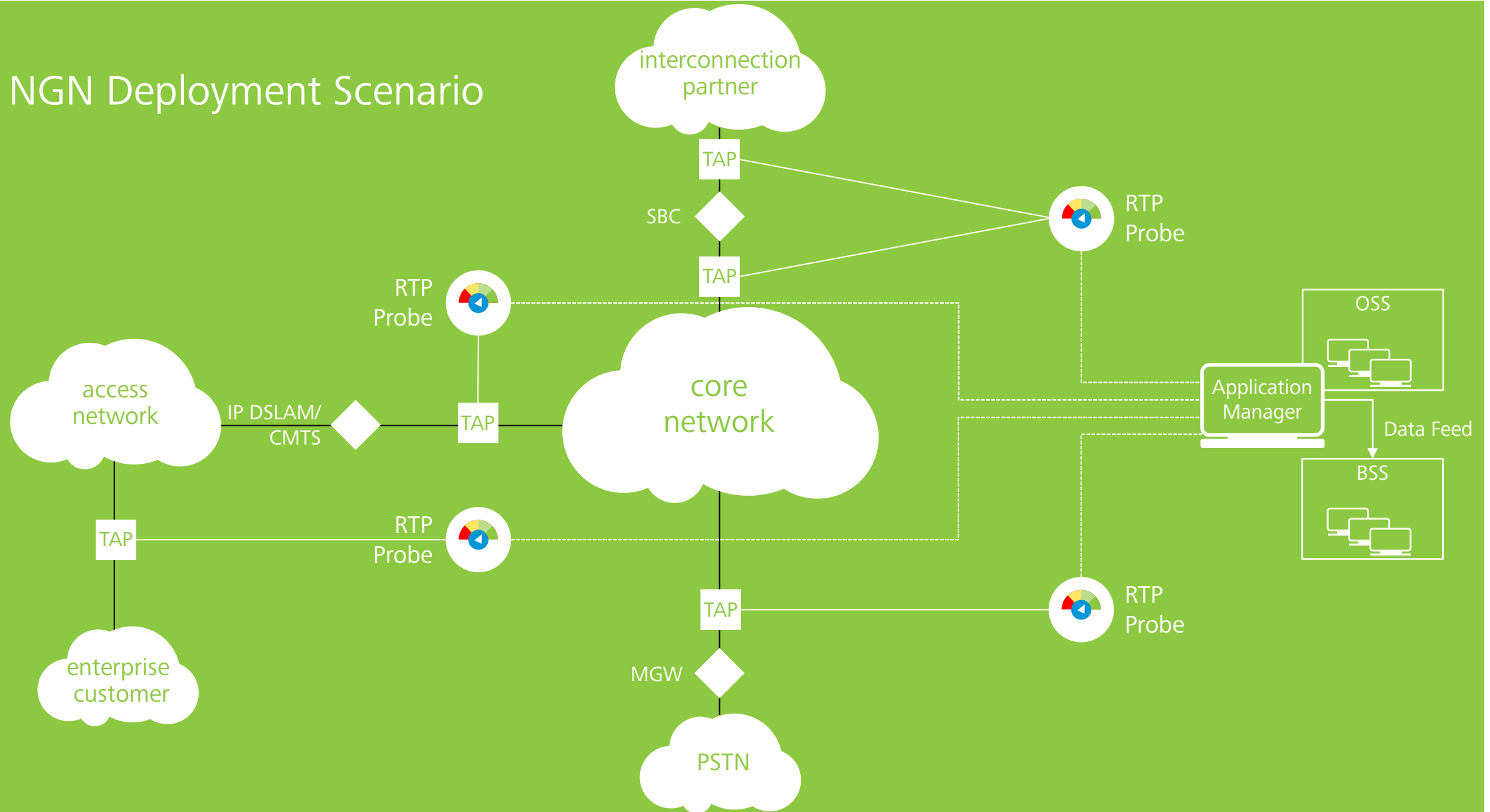
4 VoLTE / VoWifi

3G/LTE set will contain information from the PANI-header (cell-ID) and other data relevant to mobile terminated/ originated calls; for the moment this field contains the information we gain from AMR codec headers

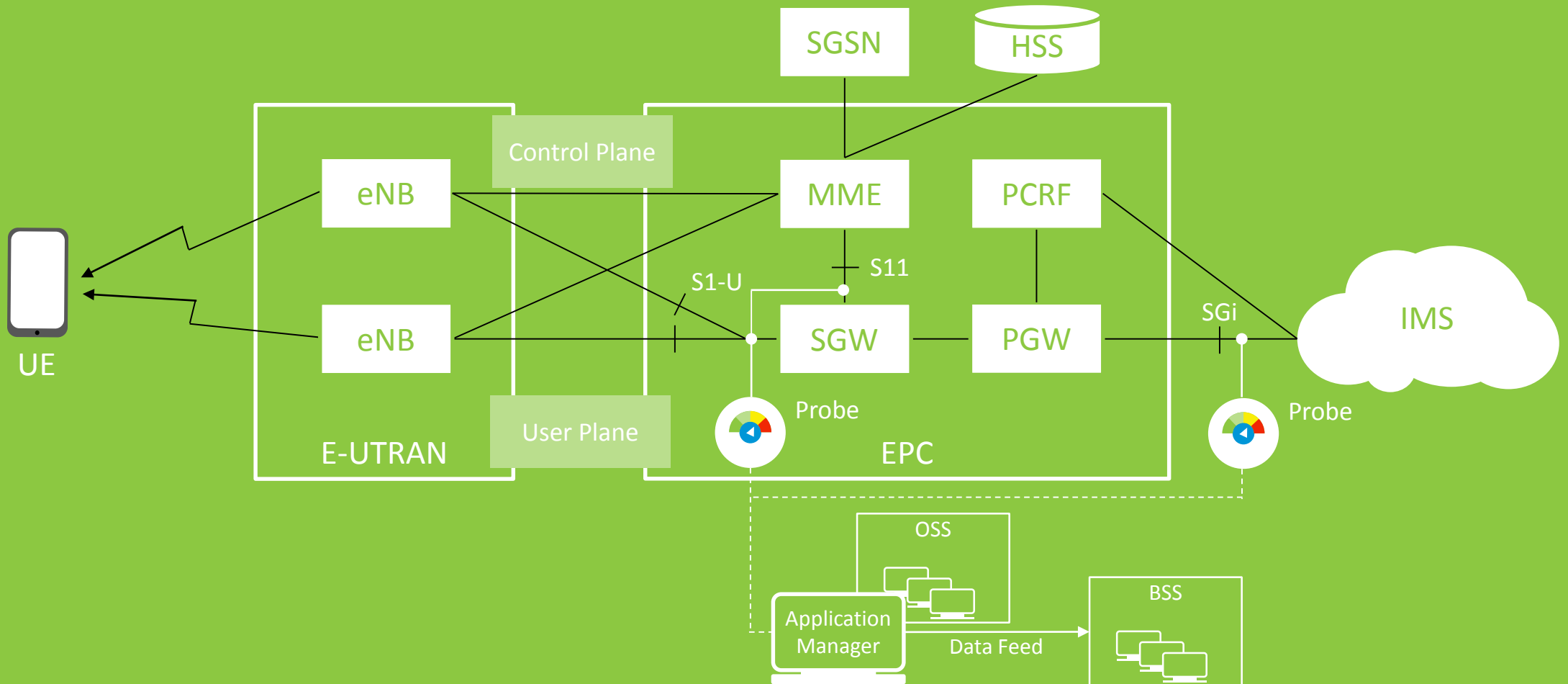
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DEMO

NGN Deployment Scenario



VoLTE Deployment Scenario



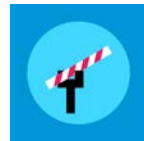
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SUMMARY

Voipfuture Qrystal | Features



Full Traffic



Network Segmentation/
Border Control



Correlation



Threshold Definition



All Calls



Automated Impairment Detection



Grouping



Exporting



Both Directions



Smart Packet Recording



Aggregation



Integration



5-Second Time Slices



Waveform Analysis
(Live Traffic)



Drill-Down

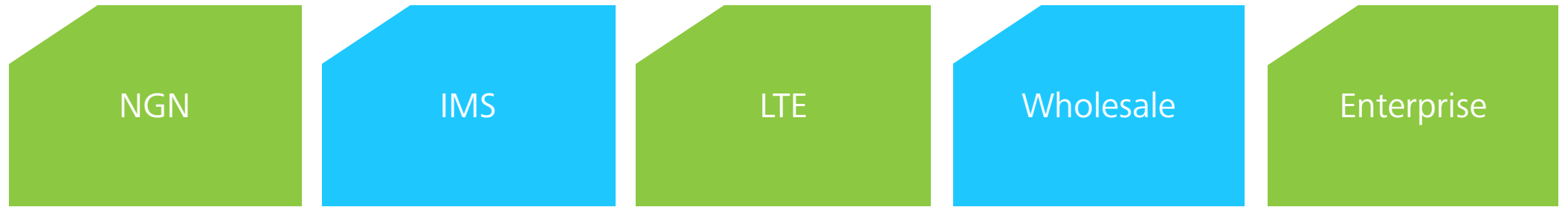


History

Voipfuture Qrystal | Use Cases



Voipfuture Qrystal | Application Domains



Voipfuture Qrystal | Key Performance Indicators

Media Plane

- GMR – Good Minute Ratio¹
- GSR – Good Stream Ratio¹
- QTR – Quality induced Termination Ratio
- FCSR – Fax Compatible Stream Ratio
- FPIR – Fax Pass-through Incompatibility Ratio
- TMR – Tolerable Transport Minute Ratio
- CMR – Critical Transport Minute Ratio
- CSR – Critical Stream Ratio
- EFR – EF Stream Ratio

¹ Narrowband/Wideband

Control Plane

- SER (ASR) – Session Establishment Ratio²
- SEER (NER) – Session Establishment Effectiveness Ratio²
- SDT (ACD) – Session Duration Time²
- SCR (ABR) – Session Completion Ratio²
- SRD (PDD) – Session Request Delay²
- IRA – Ineffective Registration Attempts²
- ISA – Ineffective Session Requests²
- LSR – Late SDP Response Ratio
- ACDR – Ratio of Calls close to ACD

² RFC 6076

Qrystal vs Standard Signaling Tool

	Voipfuture	Other
Signaling Analysis	+	+
Full visibility of voice service quality for single calls, SIP trunks, IPX SLAs and entire networks	+	●
Trustworthy & accurate MOS for every live call	+	●
Enabler for SLAs through KPIs based on 5-second time slices	+	●
Protection of own network from bad traffic originating in foreign networks	+	●
Rapid fault isolation via network segmentation	+	●
Automated root cause analysis for effective troubleshooting of RTP flow and waveform issues	+	●
Excellent ROI due to high performance an value add for existing OSS/BSS systems	+	●
Smart correlation of bearer events and VoIP impairments – understand the impact of LTE events on the user experience	+	●

THANK YOU FOR YOUR ATTENTION

vo:pfuture

Wendenstr. 4
20097 Hamburg
Germany

Tel: +49 40 688 900 10

Fax: +49 40 688 900 199

info@voipfuture.com

www.voipfuture.com