voipfuture

WHITEPAPER

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Call me back on Skype

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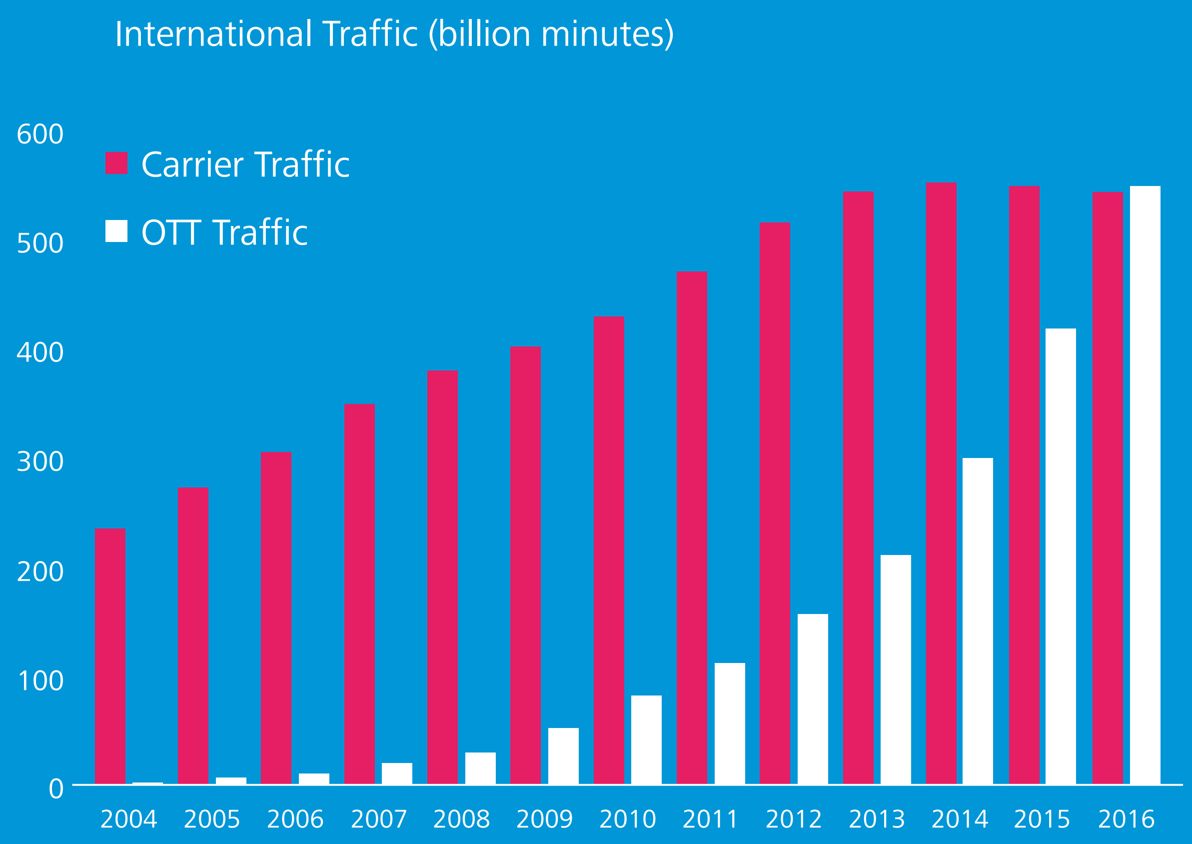

Source: TeleGeography

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For years international wholesale operators have been losing business to over-the-top (OTT) services like Skype and Facetime. One reason is that users do not understand why they should pay for less. This erosion will continue and even pick up speed if IPX services do not deliver on their promise. Part of this promise is to improve the user experience by offering high-definition voice servi-ces. Only high-quality audio – better than what is provided by OTT servi-ces - will allow IPX service operators to stay competitive. Yet, this needs more than just new codecs.

The use of wideband codecs is fundamental to high-quality IPX service offerings. The main advantages of wideband over narrowband codecs typically cited by users are:

- clearer overall sound quality
- easier to recognize voices, ability to distinguish confusing sounds and understand accented speakers
- reduced listening effort, resulting in increased productivity and lessened listener fatigue

Wideband codecs are surely necessary to create a level playing field for IPX and OTT services. Still, the audio encoding is not the only factor contributing to the user experience in live networks. This whitepaper reports on the analysis of call data collected in large VoIP networks, which confirms that customer satisfaction – and thereby the average call duration (ACD) – also depends on the network performance. The good news is that this is under control of the IPX service providers. The bad news is that the ACD is not sensitive enough to detect anything but catastrophic events.

Background

Voipfuture has performed an extensive analysis of quality-enriched call detail records (CDR) collected from Qrystal VoIP monitoring systems deployed in live networks. The aim was to understand the impact of call quality on call duration. The average call duration (ACD) is typically seen as a measure for the quality of a specific service or route and thereby implicitly also for the customer satisfaction.

DATA SET

The raw data set covers nearly 30 million international, domestic and mobile (VoLTE) calls. The CDRs were anonymized, i.e. calling and called party numbers were scrambled. The calls used at least 22 different types of codecs including G.711 (A-law/µ-law), G.729, G.722, G.723 and various modes of AMR-NB and AMR-WB. The majority of these codecs were however used so rarely that respective CDRs had to be excluded from the analysis. In summary, the following calls were not considered for the analysis:

- calls which did not use the top four most widely used codecs G.711 (A-law/µ-law), G.729, AMR-NB 12.2k or AMR-WB 12.65k
- calls which lasted less than 10 seconds or more than one hour
- calls where at least one call direction was impacted by duplicate packets

Codec	Net bitrate/kbit/sec	#CDRs	Percentage
G.711	64	10,998,417	67.3%
G.729	8	4,108,152	25.1%
AMR-NB 12.2k	12.2	1,044,612	6.4%
AMR-WB 12.65k	12.65	202,906	1.2%

Table 1: Codec distribution

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Note that codecs may change in the course of a call. The CDRs provided by the Voipfuture Qrystal system list the main codec per direction, i.e. the codec that was used for most of the call duration.

CALL DETAIL RECORDS

This study uses call detail records produced by Voipfuture's Qrystal product [1], a non-intrusive VoIP monitoring system estimating the call quality of real calls. Among other things Qrystal analyses the flow of RTP packets, their interarrival times and the information contained in the IP, UDP and RTP headers. For every five second segment of each RTP flow Qrystal generates a quality summary, which includes several hundred metrics describing the segment [2]. This quality summary contains basic information about the source and destination IP addresses, VLANs and DSCP classes used, as well as detailed information about packet losses, packet interarrival times and estimated R-factor and MOS values. Qrystal also marks 'critical' five-second segments which suffer from burst loss or excessive jitter.

Qrystal Connect creates quality-enriched call detail records summarizing the five second data, e.g. by storing the minimum, average and maximum MOS value for each call direction. In addition, the critical quality ratio for each call direction is calculated, which is the ratio of critical five second segments over all segments. In total each CDR has more than 200 fields.

CALL DURATION

Call duration is the time difference between call establishment and call termination. Generally, it is affected by many factors, such as the calling and called party situation, the amount of information to be exchanged, social circumstances, gender and age of call parties and their nationalities. However, such details are not available for the CDRs underlying this study. The following section looks into the impact of some technical factors on the call duration.

Factors Influencing Call Duration

A previous study [3] has confirmed a general dependency of call quality on the average call duration in mobile networks. However, the study does not provide details on how different aspects of "call quality" affect the call duration. A service provider obviously has no influence on the gender, nationality and circumstances of the call parties, but can only control the service's technical aspects, i.e. the codec choice and the network performance.

DEPENDENCY ON CODEC QUALITY

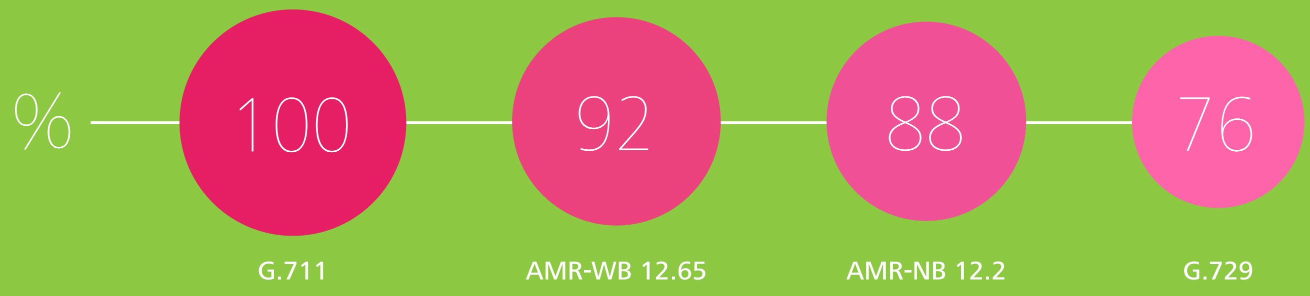
The average call duration over the entire (filtered) data set is 220 seconds. The table below shows how the average call duration depends on the calls' main codec type.

Codec	Max. R-Factor	Max. MOS_{NB}	Average call duration / seconds
G.711	93	4.41	237
AMR-WB 12.65	87	4.26	217
AMR-NB 12.2	83	4.13	209
G.729	82	4.10	180

Table 2: Best quality achieved and average call duration per codec

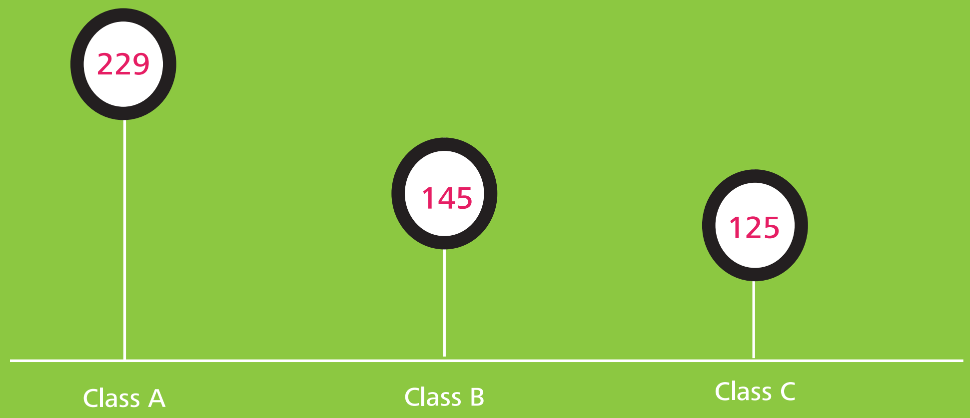
The relative impact of codec quality on the average call duration

Fig. 1: Codec quality and ACD under perfect conditions (G.711 is set as 100%)



Higher levels of jitter and packet loss lead to lower call durations

Fig. 2: Call duration (seconds) and network performance (CDR classes)



The data clearly shows that the best possible user experience that can be achieved by a codec, i.e. in the absence of critical loss and jitter, directly correlates with the average call duration. The better the codec's R-factor/MOS, the longer the average call duration.

Figure 1 shows the relative impact of codec quality on the average call duration. The ACD of G.711 is set as 100%.

This finding supports the introduction of wideband and superwideband codecs and could be seen as justification for associated investments – if networks were perfect.

DEPENDENCY ON TRANSPORT QUALITY

Live networks always introduce a certain level of packet loss and jitter to the traffic, with negative impact on the voice quality. This section investigates the effect of network performance and transport quality on the average call duration.

Qrystal marks any five second RTP time slice as critical, if it exhibits packet burst loss or excessive jitter, that will likely lead to effective packet loss. The CDRs created by Qrystal Connect include the critical minute ratio for each direction of a call. This is the ratio of critical five second segments over all RTP stream segments and thereby a good indicator for the amount of transport quality impairments affecting a media stream.

The CDRs were assigned to one of three classes A, B and C based on the critical minute ratio of their worst stream. No distinction was made between codecs. Class A contains calls which experienced a good network performance, while class C contains all calls which suffered from a bad network performance. The classes are defined in the table below.

Class	Critical Minute Ratio
Class A (good network performance)	0% – 15%
Class B (tolerable network performance)	15% – 30%
Class C (bad network performance)	30% – 45%

Table 3: CDR classes

The average call durations for all three classes are shown in Figure 2. It clearly shows that higher levels of jitter and packet loss lead to lower call durations. These numbers are however calculated from on a very large sample set of 16 million CDRs. This certainly proves a dependency, but one cannot expect a similar impact on statistics for individual routes and short time frames.

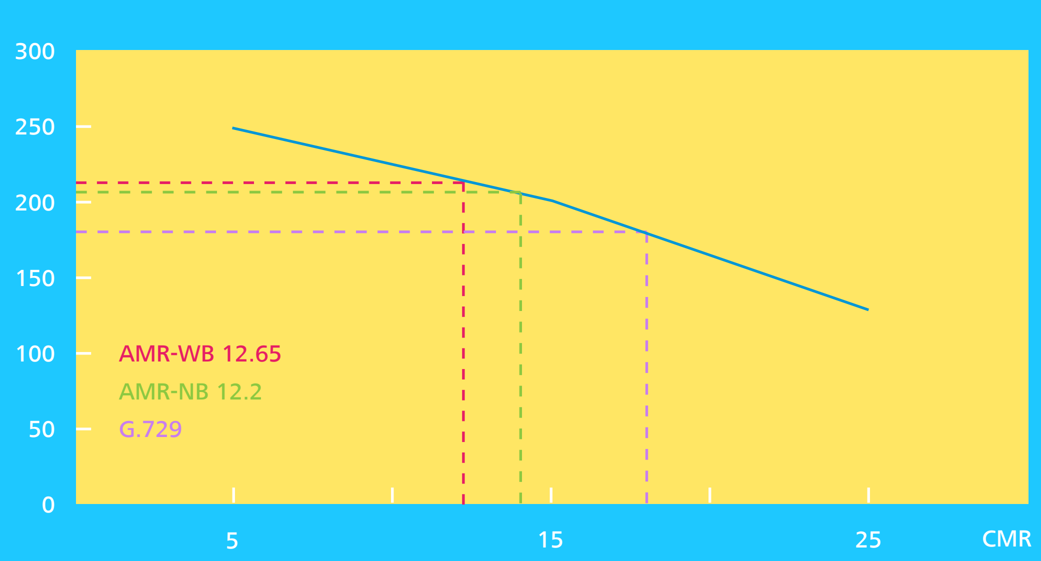
Also, it should be noted that the critical minute ratio reveals more about the distribution of impairments over the entire call duration than about a call's overall level of packet loss and jitter. For example, a critical minute ratio of 100% may apply to RTP streams which lose two packets every five seconds. This corresponds to a packet loss ratio of less than 1%, which is not a lot by conventional wisdom.

The above has established that bad transport quality clearly reduces the average call duration. The next question is which levels of packet loss and jitter counterbalance the choice of a better codec. Figure 3 answers this question for the G.711 codec, which is the codec with the highest R-Factor and MOS under investigation.

The red line in figure 3 shows that at a critical minute ratio of 12% G.711 only reaches the average call duration of AMR 12.65k under perfect conditions. At a critical minute ratio of 14% it degrades to the quality of AMR 12.2k and at 17% critical minutes G.711 goes down to the level of G.729.

Which levels of packet loss and jitter counterbalance the choice of a better codec

Fig. 3: Critical minute ratios at which G.711 ACD drops below worse codecs



If you want to know live experience you have to measure live traffic

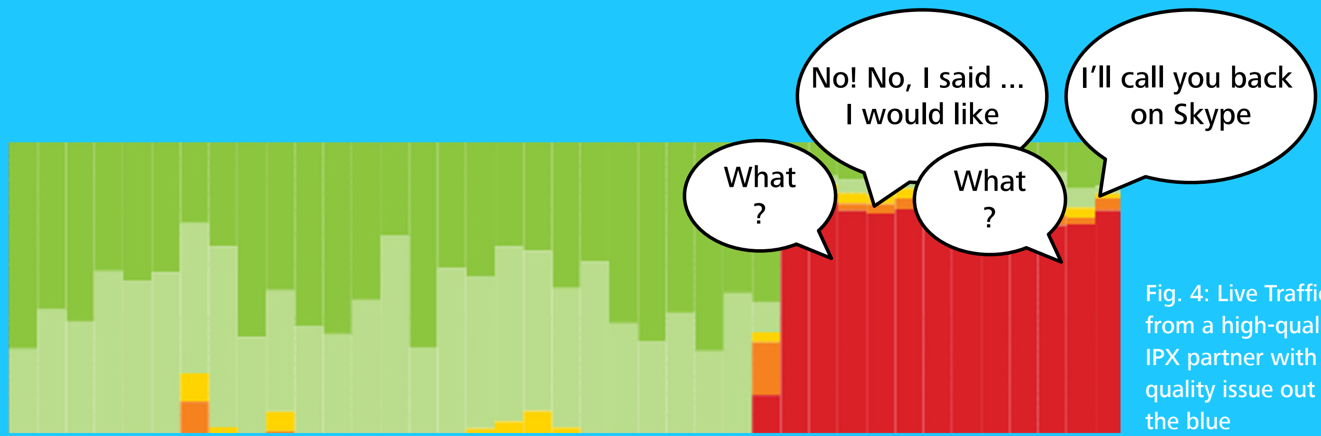


Fig. 4: Live Traffic from a high-quality IPX partner with a quality issue out of the blue

Transport Quality in IPX and Wholesale Networks

One could argue that such impairment levels are not common in tier-1 fixed-line IP networks. The transmission quality in fixed-line networks is generally much better than in wireless networks such as LTE and Wifi. Yet, IPX was defined by the GSMA with the primary purpose of providing premium international voice services to mobile network customers. Therefore quality degradations that will occur in those mobile networks will also be visible in IPX networks and will reduce any available margin of error for IPX network providers.

On top of this, the definition of 'critical' used in this analysis is very strict. Remember that a critical minute ratio of 100% may correspond to as little as 1% packet loss. IP networks are highly dynamic systems. A one percent packet loss on a fixed-line IP network is a lot during normal times of operation, but transmission quality can drastically change at any time for different reasons. Voipfuture systems have detected numerous cases showing how even tier-1 interconnection partners are blind to issues with the quality of their traffic, because the ACD drops were not considered significant. In one case the critical minute ratio of all calls coming in from a tier-1 interconnection partner jumped up from 1% to well over 30% from one day to the next. The ACD went down by less than 4% which did not trigger an alarm. In other words, the issue caused a clearly audible loss in quality, but was not detected by conventional metrics.

Conclusion

This whitepaper has presented results from one of the largest studies on the connection between call quality and duration conducted so far and linked the results to the IPX domain..

THE MAIN FINDINGS ARE

1. **For large sample sets the average call duration depends on the codec quality – better codecs lead to longer call duration.**

2. **For large sample sets the average call duration depends on the transport quality – worse transport quality leads to shorter call duration.**

3. **The levels of transport quality degradation required to significantly impact the user experience can often be observed even in IPX networks.**

4. **Quality degradations are not easily visible in the ACD.**

These findings lead to two conclusions for IPX service providers (and their customers) who seriously want to compete against OTT services. First, they must introduce high-definition voice codecs. Obviously this is a move that must be supported by the entire telecommunications industry – the current introduction of AMR-WB and Enhanced Voice Services in VoLTE and VoWifi networks is a major step in this direction. Second, even the best voice codecs cannot guarantee customer satisfaction if the network performance is poor. The IPX specifications suggest to use specific media plane KPIs that provide far better insights into the user experience than the old industry standard ACD.

IPX service providers should therefore focus on controlling their media quality and accompany the introduction of wideband codecs with efforts to improve their RTP monitoring capabilities. Quality as a strategy is the only way to reclaim lost market share.

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About

Voipfuture is a premium voice quality monitoring vendor developing unique technology for assessing, aggregating, analyzing, and visualizing voice quality information. Voipfuture products offer a precise view on media and control plane to communication service providers, wholesalers and enterprises.

Since its launch, Voipfuture has been at the forefront of voice quality monitoring and continues to redefine Voice over IP by connecting their customers' view on service quality with high resolution user experience.