

Improving QoE for Real Time Calling

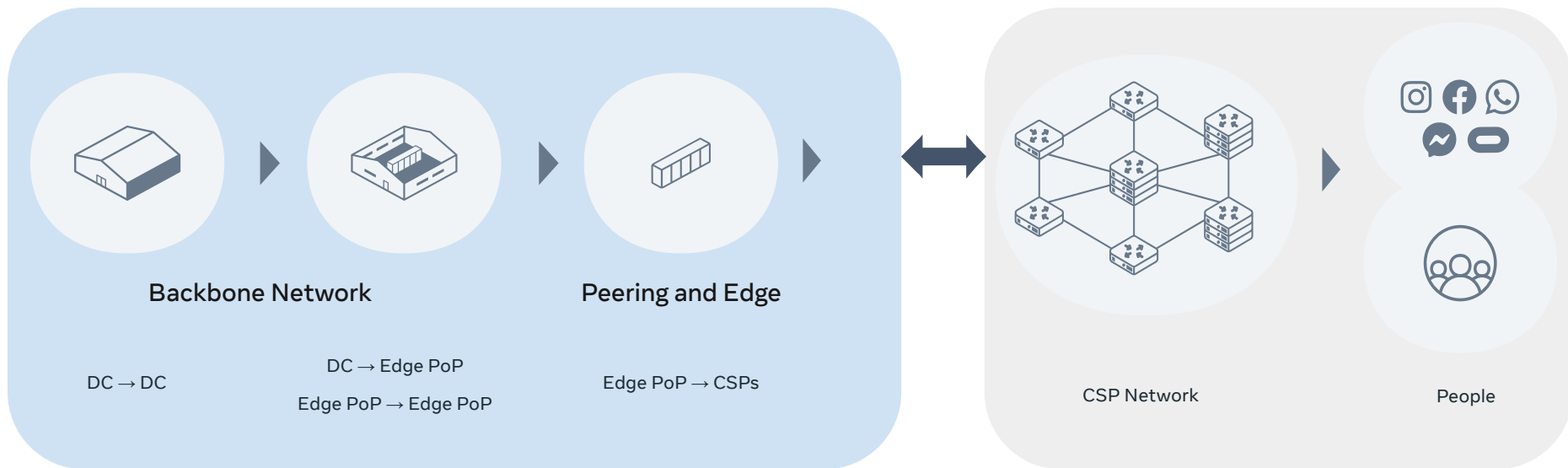
Nov 2025

Agenda

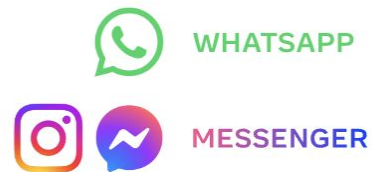
- 01 Introduction
- 02 Build Understanding on WhatsApp calling
- 03 Network Analysis
- 04 Collaboration next steps and summary

Who are Network Infrastructure ?

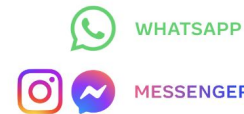
We work with Infra Data Centers (IDC), building Edge and Backbone networks to develop long term network supply supporting our 3.7B+ Meta Platform users.



Introducing **VAMOS**



What is VAMOS ?



Objective

Improve user experience of over-the-top audio and video calling



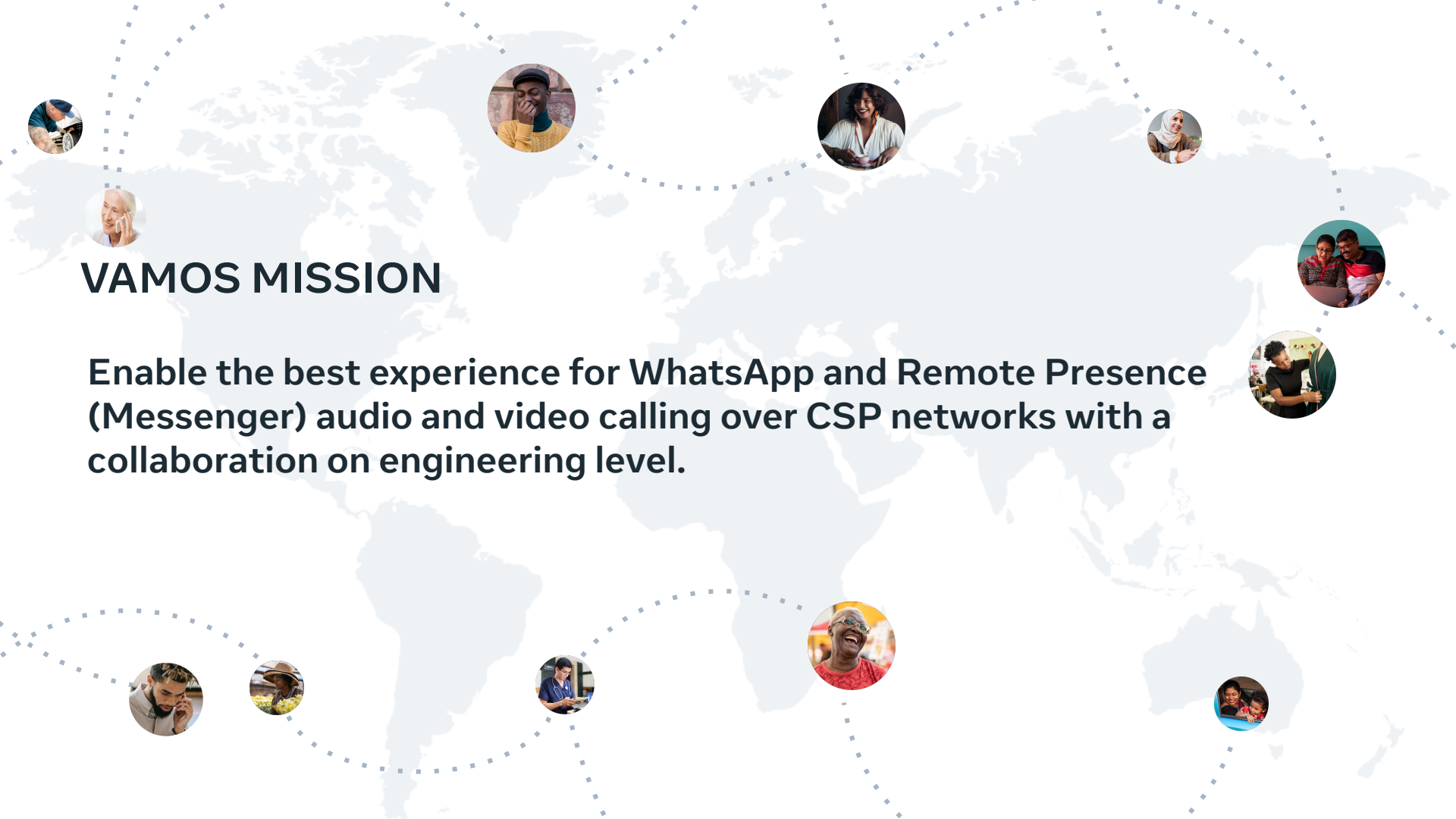
How

A **collaborative** approach with CSPs focused on improving customer experience by optimizing network architecture and configuration



Why

Customer experience plays a pivotal role in setting businesses apart in a competitive landscape and lays the foundation for future network content demands



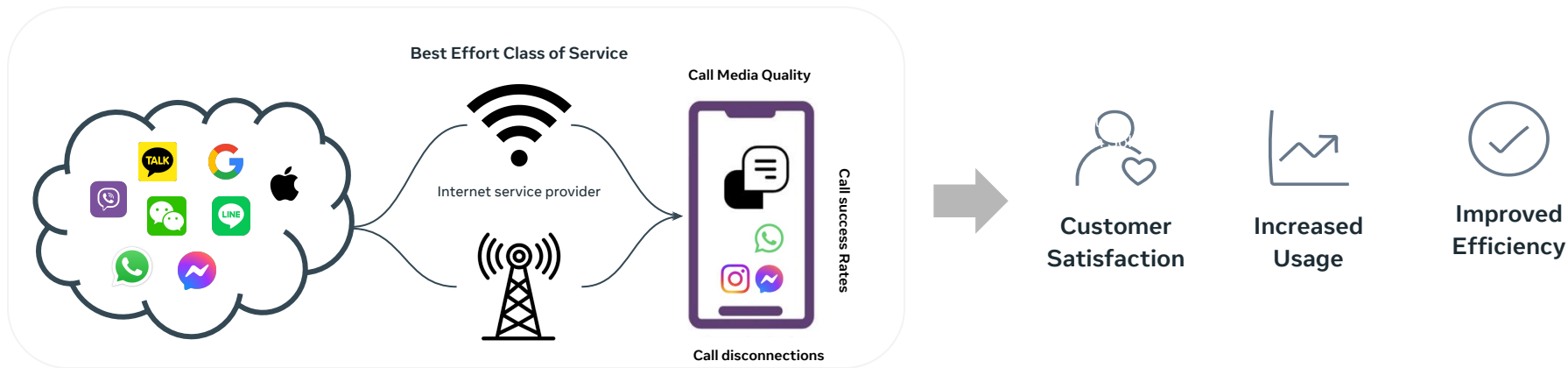
VAMOS MISSION

Enable the best experience for WhatsApp and Remote Presence (Messenger) audio and video calling over CSP networks with a collaboration on engineering level.

Introduction to VAMOS

Challenges of RTC applications

OTT RTC applications like WhatsApp, Messenger are designed with **best effort network** in mind, so they **adapt app behaviour** according to the network performance & network availability (congestion, latency, goodput), which can affect QOE.



Typical concerns reported by users of real time communication

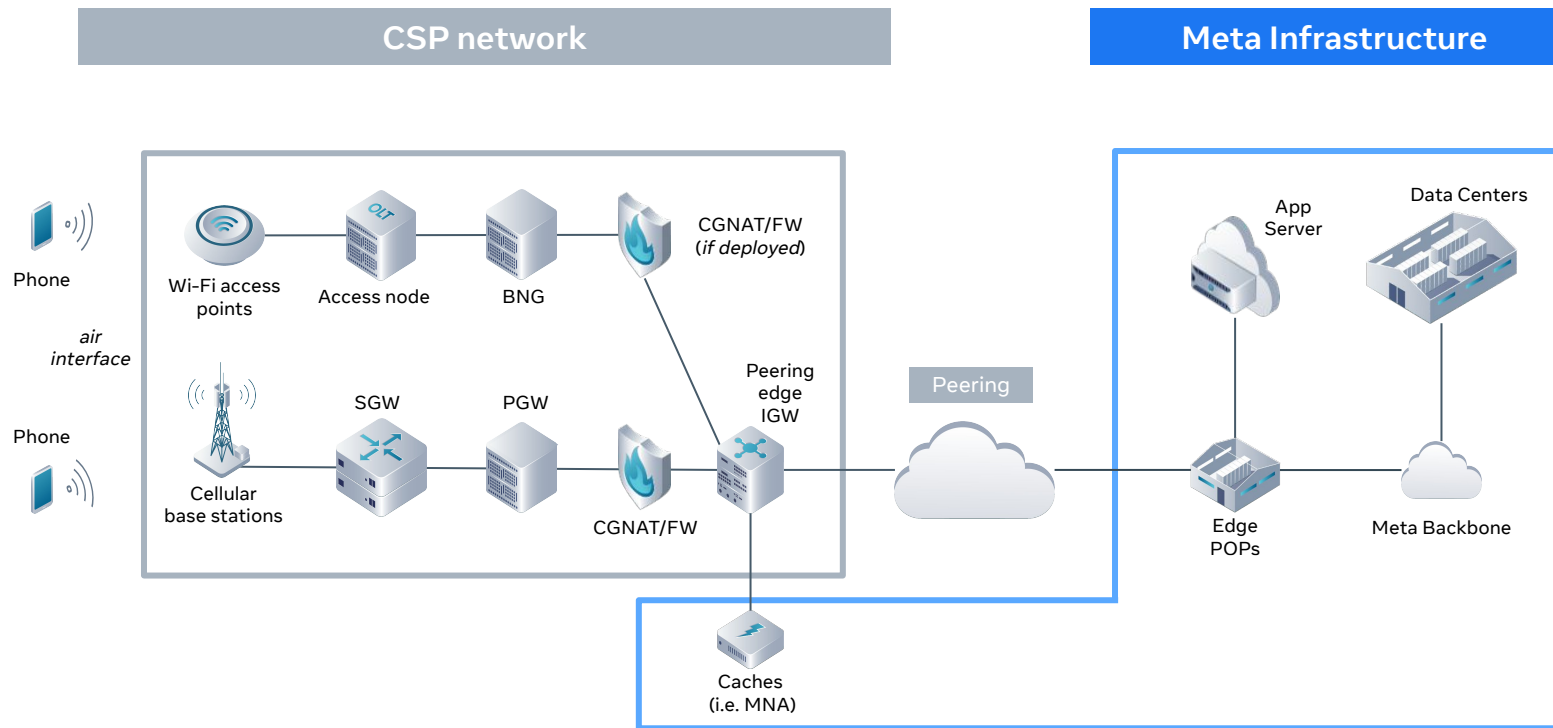
Call Success Rate

Call disconnection

Call Media Quality

Introduction to VAMOS

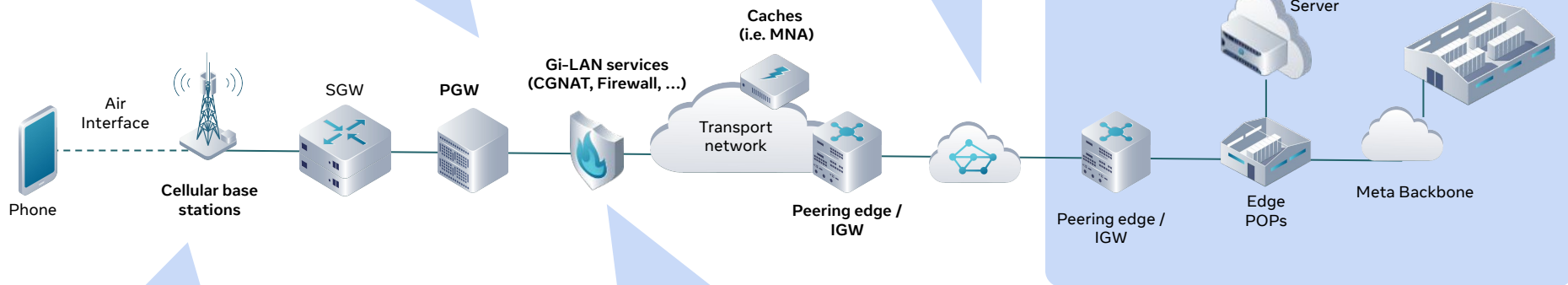
End to End network



Areas of focus

*Are there policies applied in the **core network** that could be impacting performance, and how can these be optimized?*

*Is there scope to optimize our **edge and interconnections** to improve performance for on-net calls and calls between MNOs?*



*How can **RAN configurations** be optimized to deliver improved performance and reliability for RTC apps such as WhatsApp?*

*Can we optimize configuration of **CGNAT** and other 'middleboxes' to improve call performance and reliability?*

Meta Infrastructure

Introduction and Context

We Drive Worldwide Programs

We want to focus on Middle East region, as currently no engagements.

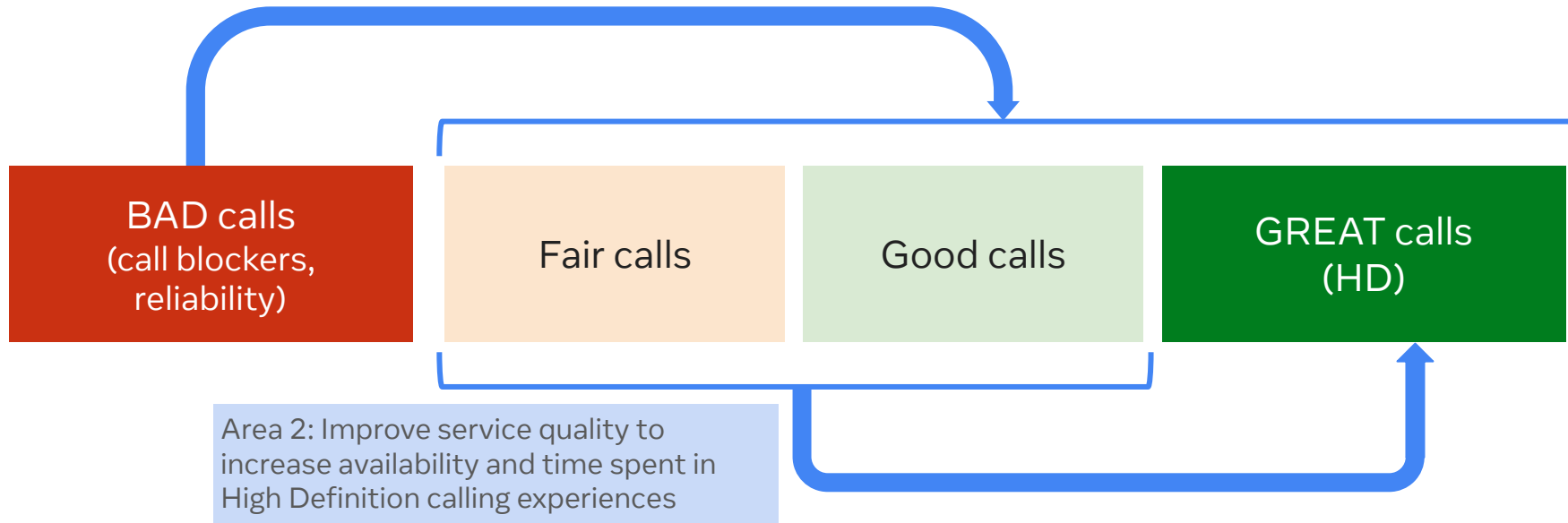
Reason -
Awareness of the program and policy challenges



Build understanding

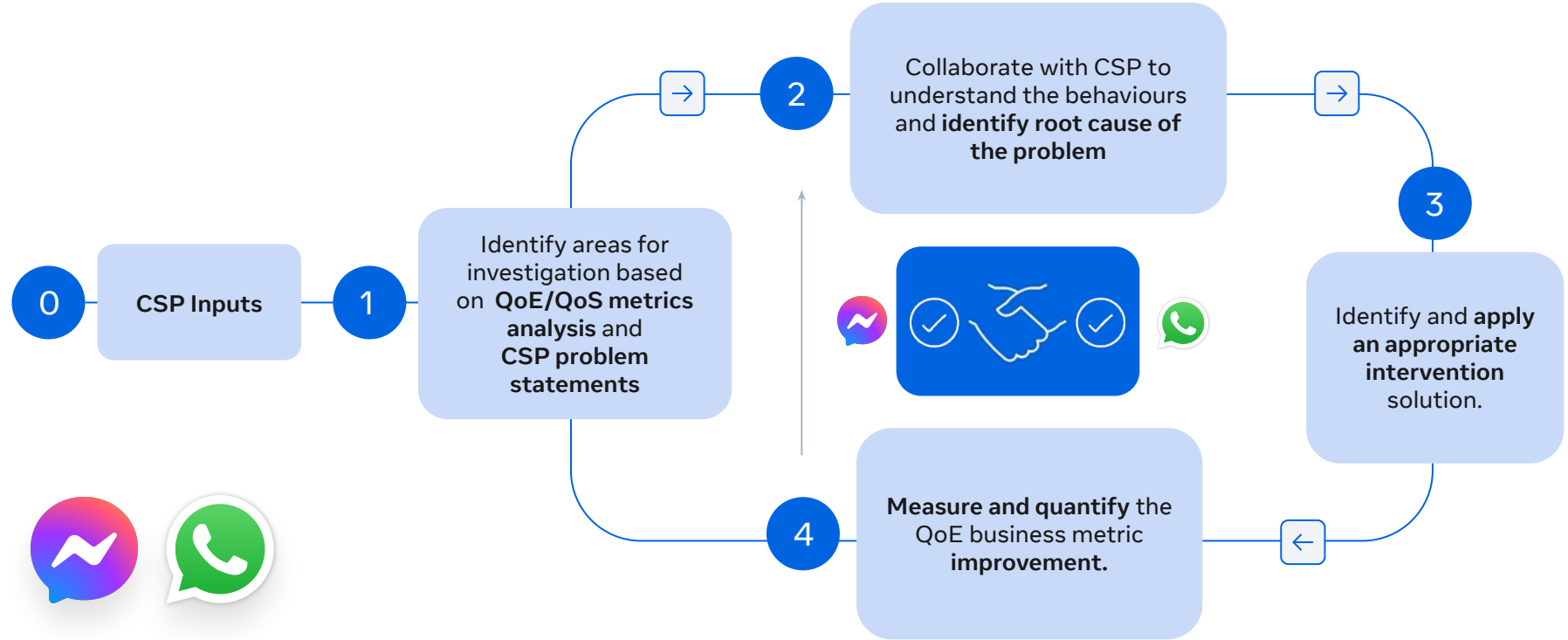
Characterizing calling quality

Area 1: Address underlying causes of poor calling experience to reduce %Unhealthy calls



Introduction to VAMOS

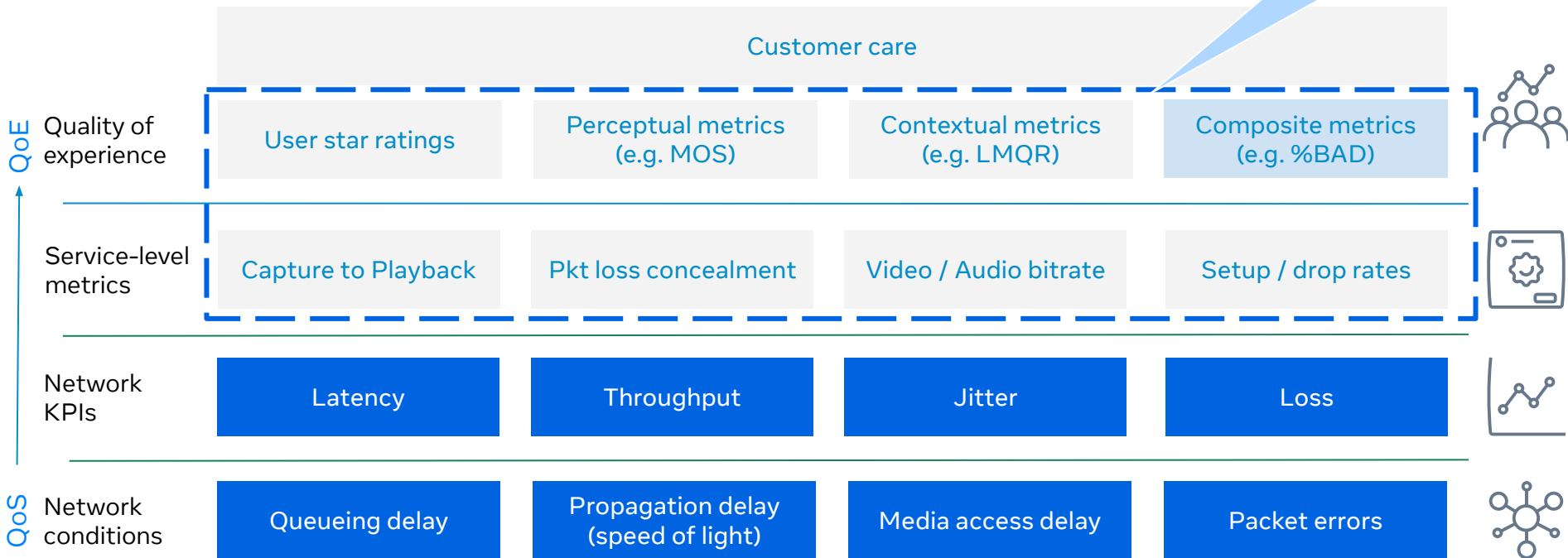
Collaboration framework



Build understanding

How do we think about application quality?

Metrics measured by
Meta's apps and
infrastructure



2 Key Metrics to measure WhatsApp Performance

1

**% BAD Calls /
Unhealthy Calls**

X%

Quality

2

% P2P Success

Y%

Efficiency

Client stack overview

SIGNALING FUNCTIONS

- Call connection
- Path establishment and transition
- Call disconnection / termination
- Telemetry upload

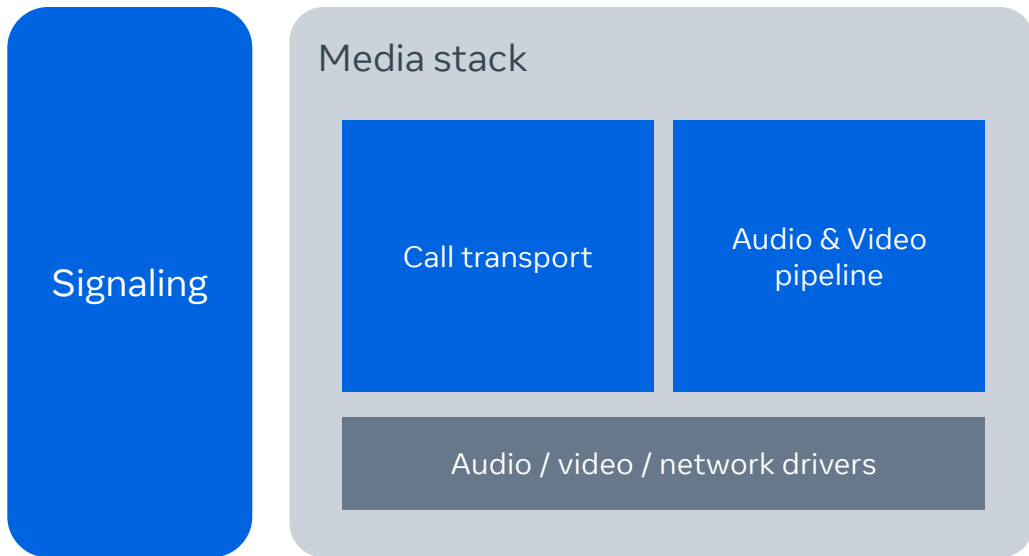
Signaling over TCP - uses the proprietary protocol

MEDIA STACK

- Video pipeline
- Audio pipeline
- Rate adaption
- Media transport

Uses RTP/RTCP for media over UDP with fallback to TCP

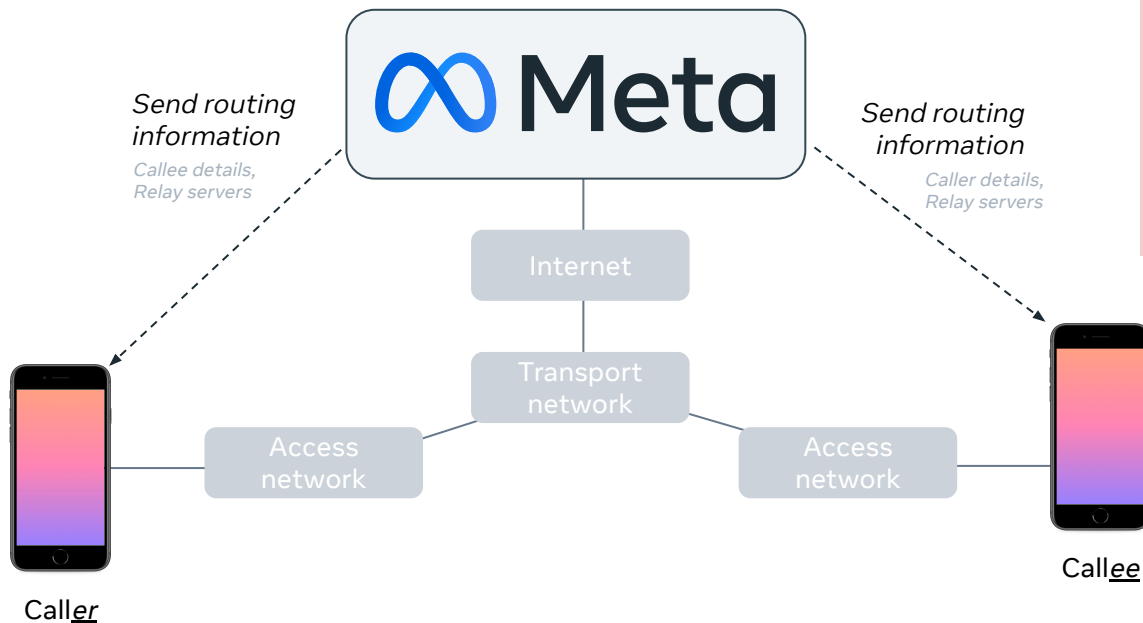
Uses SRTP to secure media



Signaling

Call path establishment

If the calling parties can both reach each other, then we establish a direct peer-to-peer connection between both clients, which is used to carry the audio and video media data for the call.



If we cannot establish a direct connection between both clients, then we can send traffic via a relay server, which passes media traffic from one call client to the other.

Note that relay servers are always used in the case of group calls

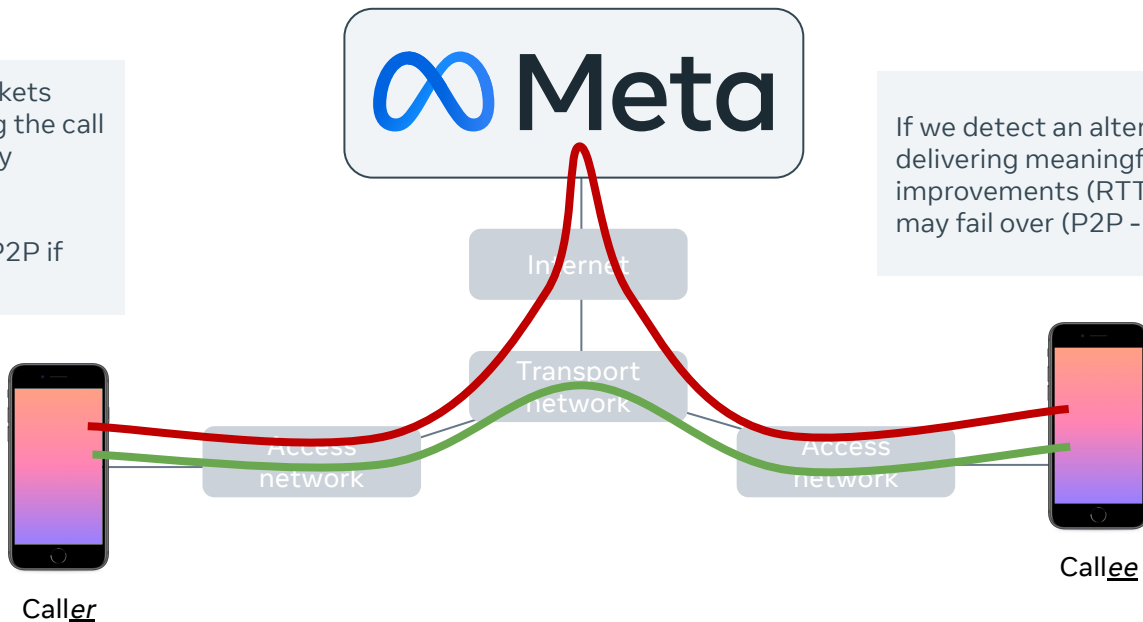
Call path transition

- Media path is established using WASP (WA STUN Protocol) - a proprietary control protocol based on STUN/TURN/ICE standards
- While we prefer P2P over relay where available, we continue to monitor available paths during the call and fail over to alternative paths if appropriate...

If we stop receiving packets over P2P channel during the call then we fail back to relay channel

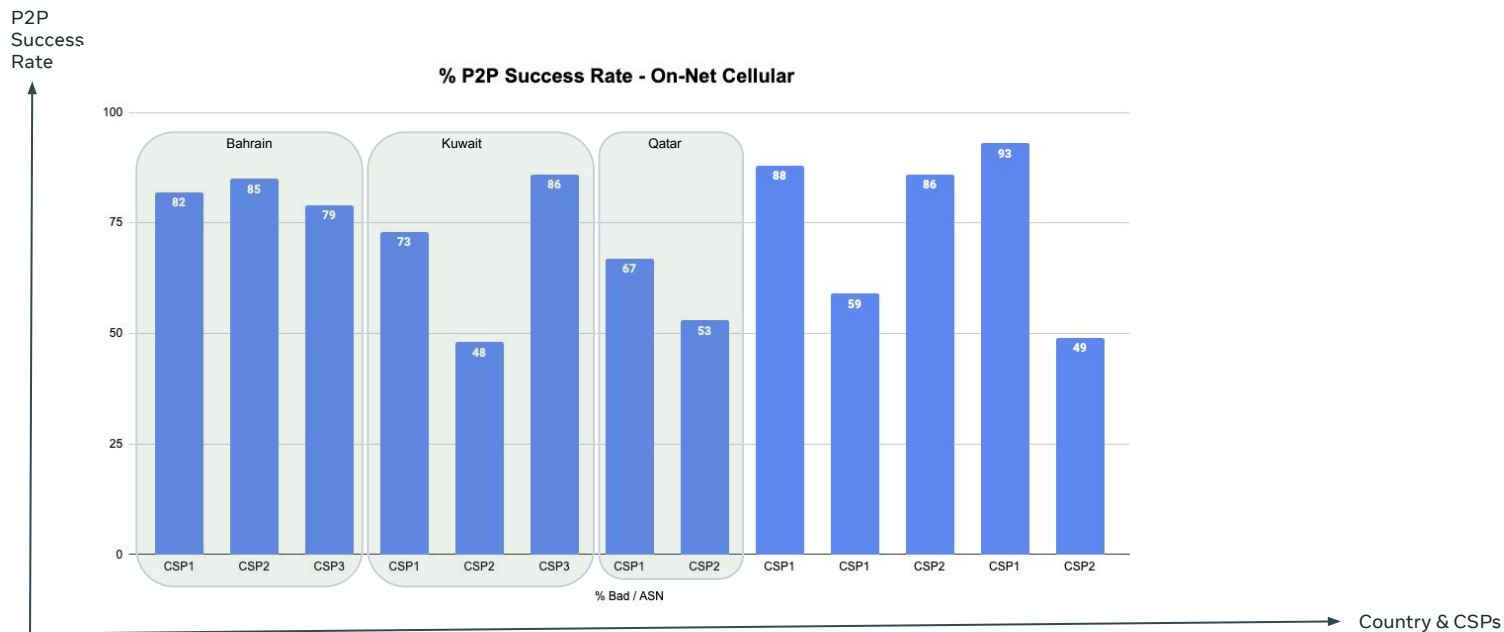
We may then revert to P2P if connectivity is restored

If we detect an alternative path is delivering meaningful quality improvements (RTT, packet loss) then we may fail over (P2P -> relay or relay -> relay)

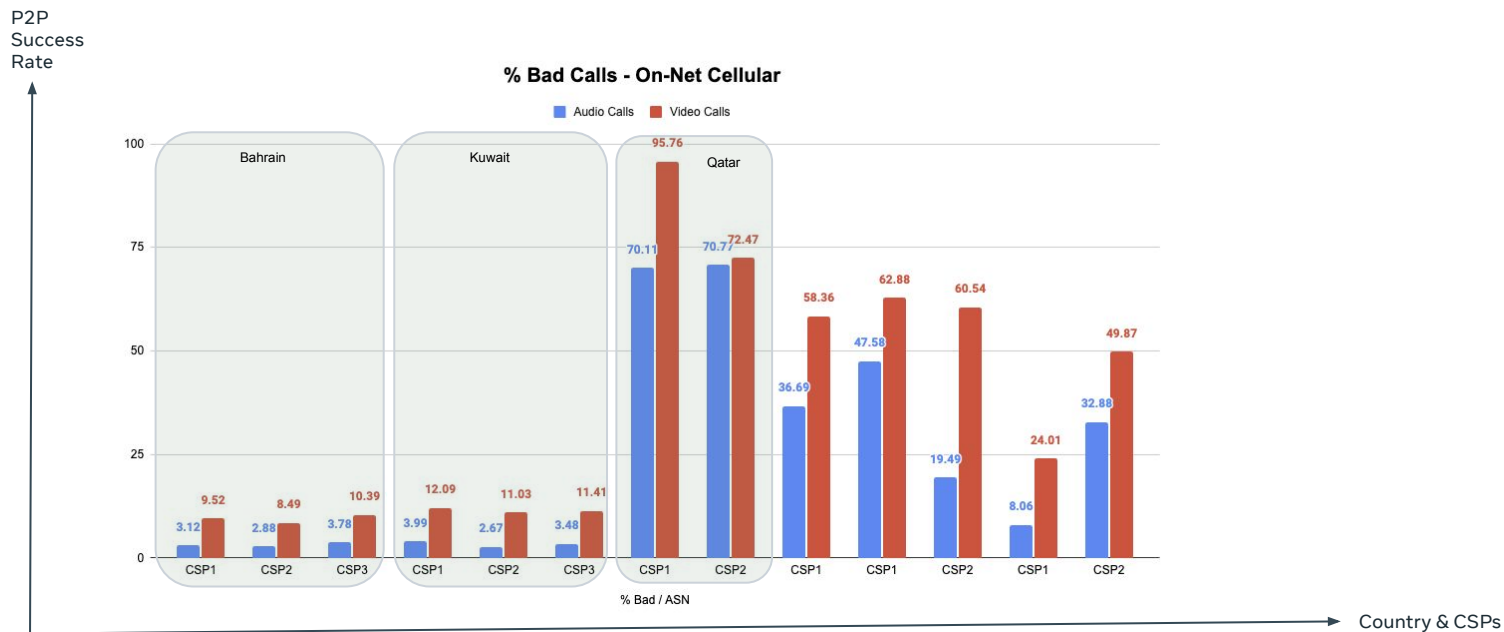


Case Studies

Middle East - %P2P Overview

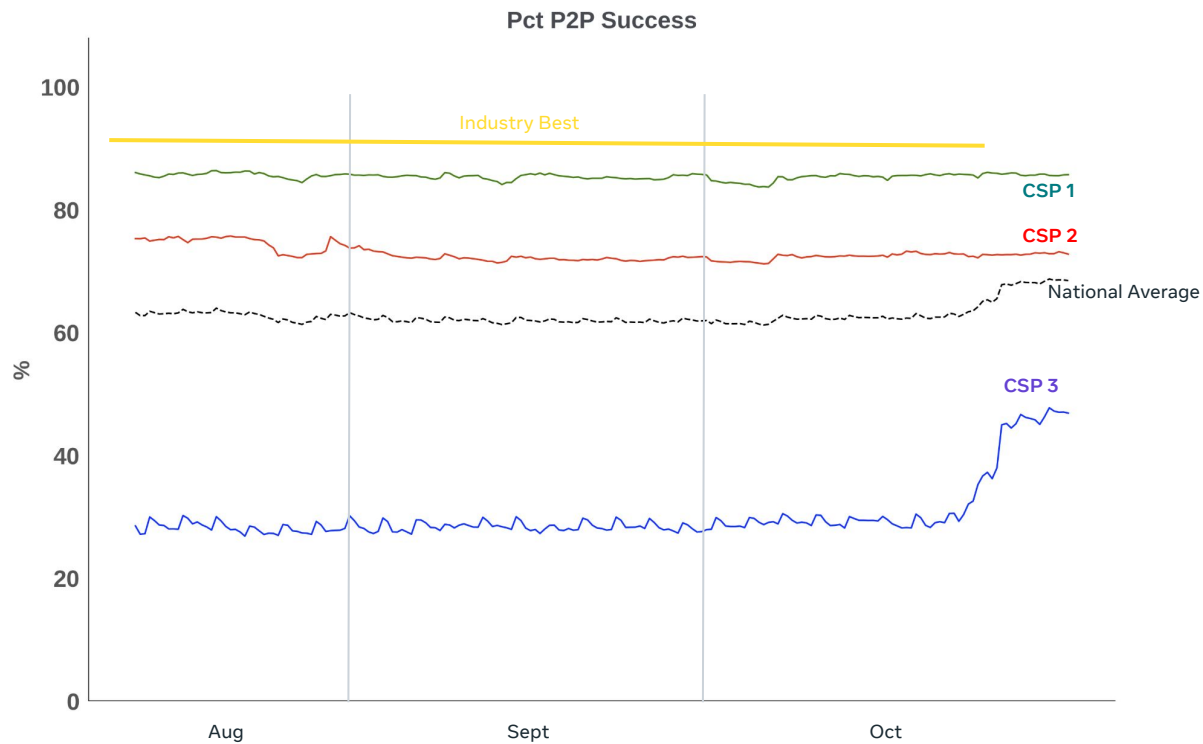


Middle East - %BAD Overview



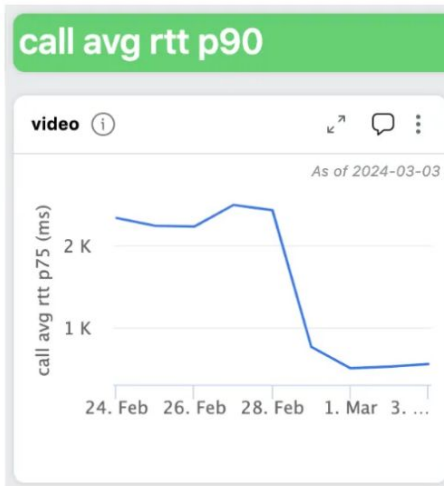
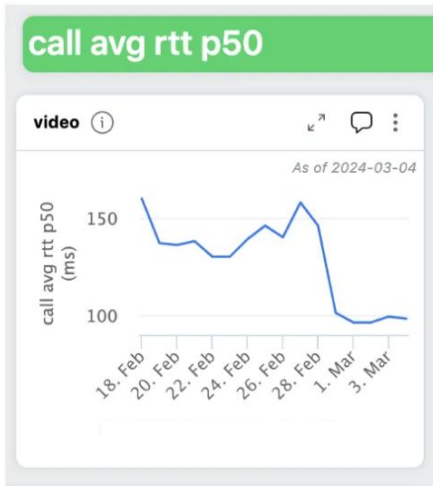
P2P Success Rate

e.g. Country Kuwait



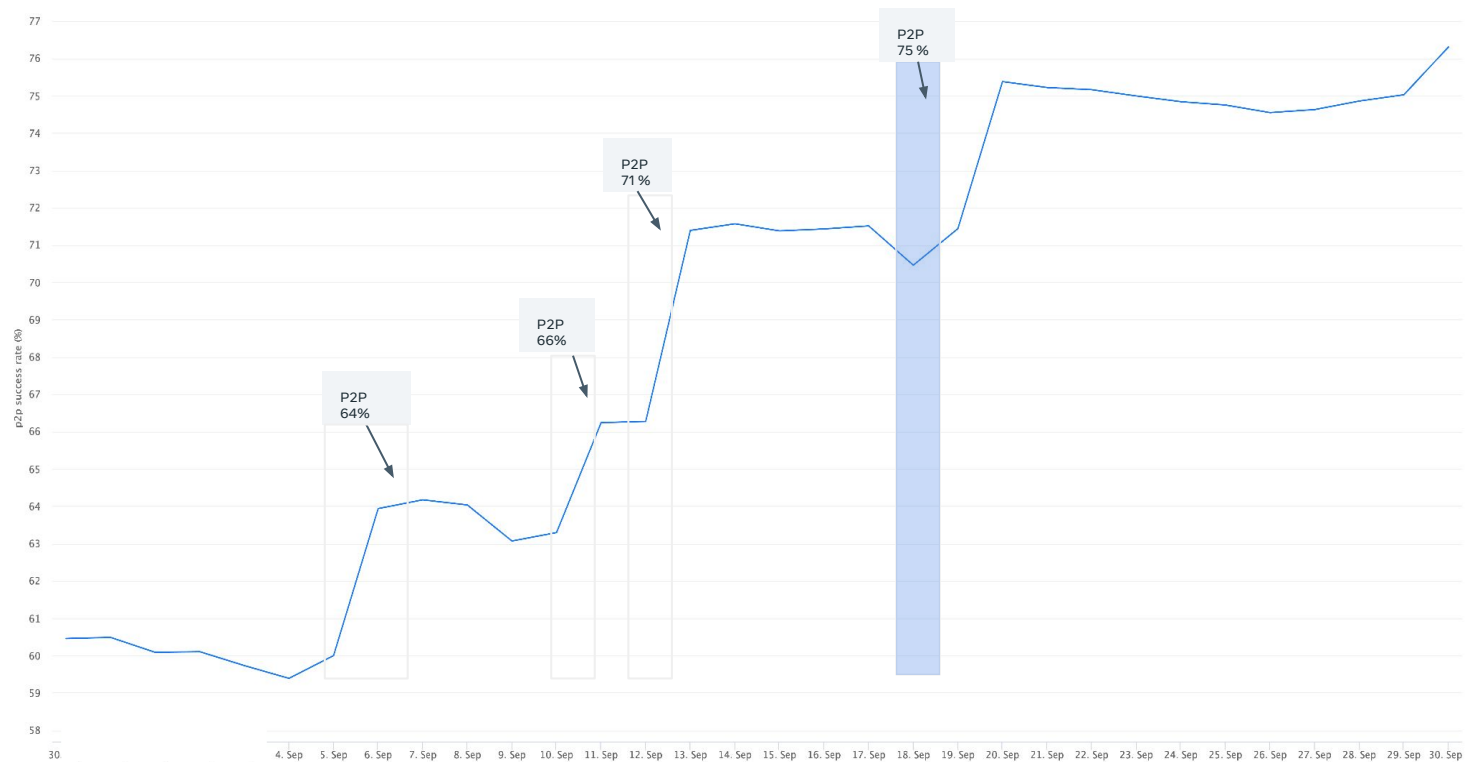
Case Study 1 - Kenya

Video Call %BAD ↓



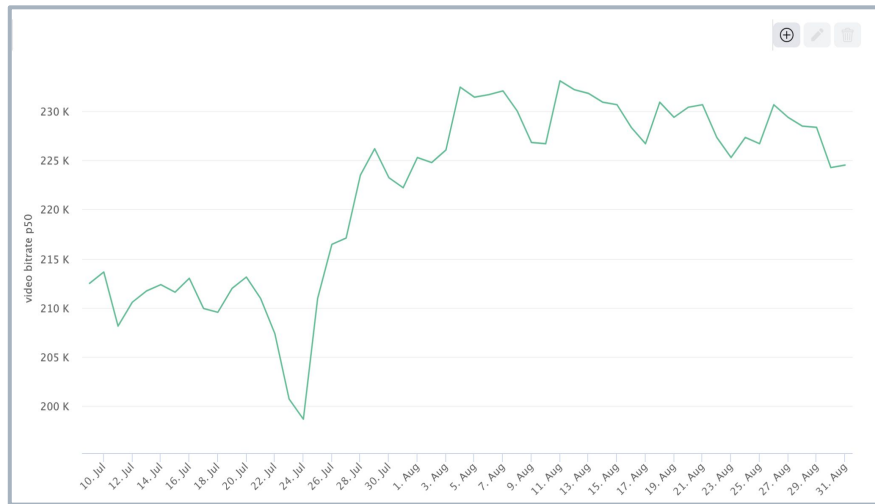
Case Study 2 - Nigeria

P2P Success % ↑

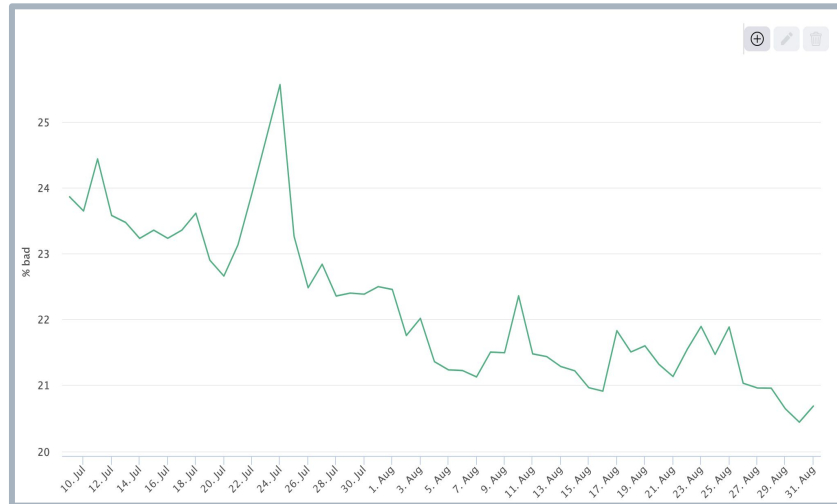


Case Study 3- Philippines

Video Bitrate & %BAD improved



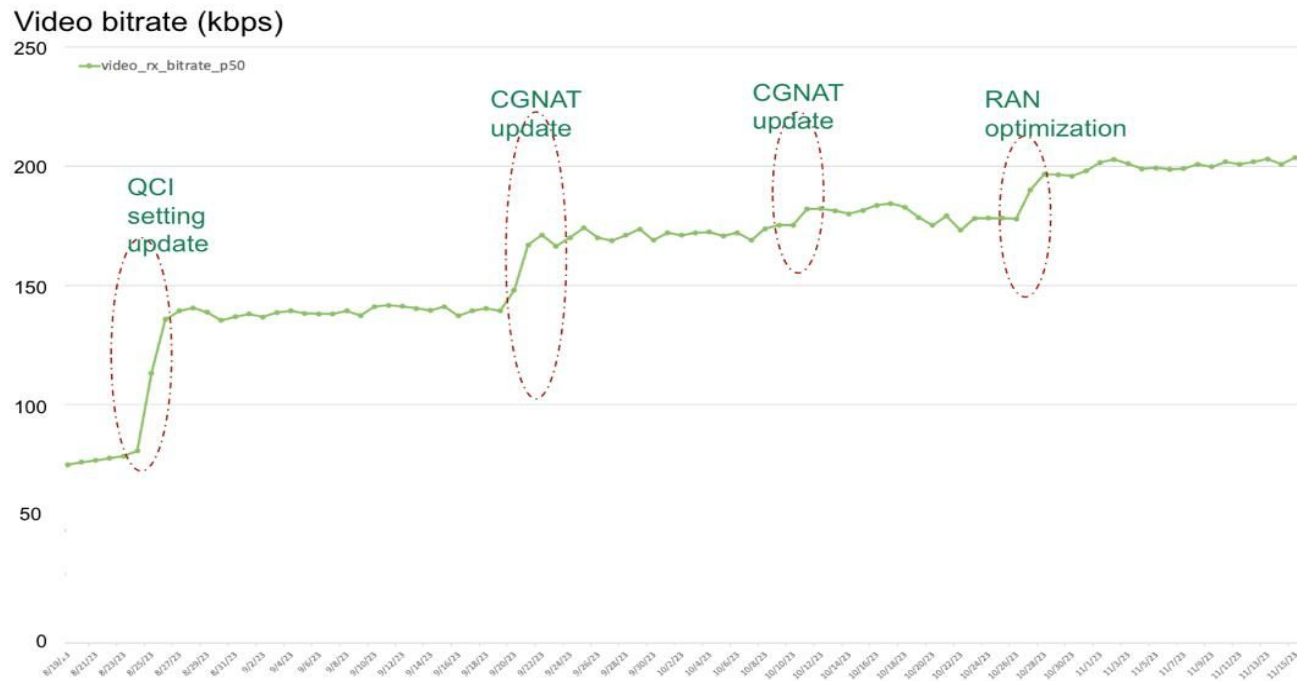
Bitrate Improvement graph



%BAD Improvement graph

Case Study 4 - Indonesia

CGNAT Full-Cone Implementation + RAN Optimization at CSP in Indonesia



Collaboration and next steps

- **Step 1** - Kickstart Engagements with interested groups and build awareness about “ How WhatsApp Call Works “
- **Step 2** - Understand the problem statement and provide RTC Calling performance analysis
- **Step 3** - Share best practices and propose interventions to improve Quality and Efficiency for RTC Calling.

Looking forward to collaborating and partnering with Regional CSPs!

Thank you

Middle East - Overview

