What Causes VoIP Underwater Garble?

Many of us have heard calls where the remote user sounds like they are underwater. When this occurs, it can be challenging to solve because it seems that nobody knows where it comes from or what causes it.

Missing or incorrect QoS configurations on bandwidth constrained links can make this problem worse, but the root-cause of this problem does not lie with QoS. In this case, there may be multiple analog-to-digital and digital-to-analog conversions along the call path, and multiple different codecs employed.

For example: If you have an older analog phone (like an older conference room speakerphone) that connects to an ATA (Analog Telephony Adapter) that connects to the VoIP phone system, and it makes a call through a gateway that converts the call from G.711 to G.729 to go over a WAN circuit, then gets converted via another gateway back to analog to then go to the PSTN where the carrier converts the call back to digital, underwater garble can easily happen.

This is because there are multiple times that the call is converted along the path, and each conversion imposes limitations that cause a cascade of problems by the time the audio reaches the endpoint. If additional latency, jitter, loss, or out-of-order is added to the call, then the situation gets worse.

If the network is running with low latency, jitter, loss, and has no out-of-order packets, the problem will still exist. To fix the root-cause of the problem, the number of conversions should be reduced. For example:

- Eliminate the ATA device by getting a native VoIP conference room phone.
- Eliminate the intermediary gateway and have the VoIP phone use G.729 natively.
- At the far end, have the edge PSTN analog gateway eliminated by converting to a SIP trunk.

With fewer conversions, the entire call should be far more stable.

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