The Ultimate Guide To Measuring & Improving Call Quality
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Want to Make All This Simple?
'A Satisfied Customer is the Best Business Strategy of All'

That quote, by business author Michael LeBoeuf, sums up why measuring & improving call quality matters. Because quality of service, much more than price, is what keeps customers sticky.

Whether you’re selling voice services to customers or providing the means for contact center agents and sales reps to talk to customers and prospects, the experiences you enable are what you will be judged upon.

Plus, if you’re spending big to ensure your teams have the training, tools and resources they need for all your voice-based touchpoints, there’s no point undermining it all with poor-quality telephony infrastructure.

There are plenty of quality metrics that help you to ensure that your network and that of your provider (not to mention the method of interconnectivity between them) are all working in harmony to provide the best possible environment for your marketing, sales and support.

But first, let’s go through the reasons behind all of this.

Quality of service, much more than price, is what more and more businesses are finding makes a big difference for their bottom line.

Increase In Costs

If you’ve been in IT long enough you’ll know the telltale signs of a company overspending to get a handle on their voice quality problem. There is a simple way to beat it. And that’s to spend a lot of money on extra IT infrastructure and more bandwidth to accommodate the lag. This will solve your problem in the short run, but it will always be an uphill battle, and won’t fix the root cause.
Bad User Experience

When it comes to VoIP, whether you’re providing voice capabilities or using them for your service, your customers are going to be the ones that suffer when it comes to your calls.

If you have even a 10% packet loss for example your voice quality is going to be almost none-existent. Bad user experience is a killer when it comes to conversion rates and incidentally your bottom line.

Why Measure Call Quality?

Enterprise Businesses

For large businesses that are already measuring sales and network quality, measuring your call quality allows for another data set to add to your collection.

What this gives you is an even clearer image of how your calls are performing and gives you the opportunity to pinpoint potential bottlenecks in your network that might be damaging sales, or interrupting vital business calls. Plus, most people’s quality assessment system won’t necessarily work when users are remote since a month ago you had one point of connection - the office. Now you need to handle connectivity and media quality of multiple agents in multiple locations with their own network connection and issues.

Voice Service Providers & Resellers

If you’re providing voice services for another business and they rely on the quality of your connections to ensure a high quality of service, the benefits of monitoring your call quality are twofold.

Like any enterprise, you can keep a closer eye on your network to make sure you can solve any connection issues or reduced call quality. This gives you the opportunity to create another level of service that you provide for your customers; the ability to monitor the quality of their calls over your network.

It’s a no brainer really.
The CDR

Since the first telephone call was placed by Alexander Graham Bell, in 1876, it’s been a fundamental challenge of scientists and engineers to ensure that the sound coming out of the receiver is as close as possible, in terms of quality and timing, to the sound going in at the other end. Typically, the metadata for a call is kept in Call Detail Record or a "CDR". Here are some examples of what data you can find in a CDR:

- The date and time of the call
- The number of minutes the call lasted
- The phone number source and destination
- Whether the call was inbound, outbound, toll-free
- The cost of the call

You can normally also do this for SMS messages but like calls, just the metadata is recorded in a CDR, not the call itself or the transcript, which you would see in call analysis tech used in your average call center.

MOS (Mean Opinion Score)

A Mean Opinion Score, often shortened to MOS, is a way of quantifying the qualitative experience of a phone call. The limiting factor is often physics, and it’s a situation that has only become worse as the size and complexity of the PSTN, the world’s aggregated copper-wire telephone network, has grown.

Now, because we’re smart cookies at Voxbone, some of the most important metrics for call quality are kept in the CDR, including Mean Opinion Score, Round Trip Time, jitter and packet loss.

Anyone who's made an international call will know that the more hops between networks a call needs to make, the more the perceptible quality will deteriorate with the introduction of lost packets leading to jitter and longer distances causing greater latency.
The key word, bolded above, is ‘perceptible’. Because Mean Opinion Scoring is a way of turning what is ultimately the subjective experience of listening to voice signals received by way of a phone call into something objective: a number rating from 1-5, with 1 being bad and 5 being excellent.

How Important is MOS?

Many argue that there are still so many vagaries in the formulation of a mean opinion score, or that it cannot be used to compare two different experiments. Others claim that boiling the experience down to a score of 1-5 is too simplistic given what is being measured.

But the strength of MOS is in measuring the same thing over time, and the number is ultimately a shorthand designed to give you a clear idea of where your network might need improving or where you might need to find a better provider.

If the MOS for a call placed between France and Moscow is 4.4 one day and then 3.6 a week later, you know something is probably up. Remember the numerical range maybe 1-5, but in reality, you’re looking at the difference between 4.4 and 3.5 being the difference between truly exceptional and problematically bad. And that’s just 0.9 on the scale.

Ultimately, MOS is one of several tools at the disposal of modern voice and network engineers in assessing the quality of the communications they are providing to their end-users.
What MOS Should You Be Aiming For?

If all 64Kbits of your audio codec are transferred perfectly you’re going to get a perfect 4.4 MOS, since that’s the highest possible quality for telephony codecs (G.711). If you’re seeing less than that, you’ve got problems.

In real terms, a MOS of 4.4 is excellent and anything below 3.5 will likely start harming the overall call experience for both parties.

How is MOS Calculated?

To understand the value that MOS provides, it is good to know how it is calculated. Today, you can objectively measure voice network quality with metrics such as latency and packet loss to incredibly high degrees of accuracy.

These ‘scientific’ measurements might appear to make MOS, which was originally calculated by way of surveys, seem antiquated in comparison.

However, MOS serves the important function of putting a number on an actual human experience in a way that knowing the number of packets dropped or the amount of jitter on a call just doesn’t really capture. It provides a holistic way of quickly understanding the ‘overall quality’ of the call.

Some degree of quality degradation is unavoidable because of the latency introduced by the distance that a voice signal has to travel between start and end points, as well as the packet loss and jitter that come into play as the amount of network infrastructure through which the signal must traverse increases (but we’ll get on to that later).

This is one reason why MOS is primarily used to judge things involving compression and transmission, where the end result can often be a pale imitation of the original. A copy of a copy, etc.

How good is the experience of listening to that compressed music file or streaming that 4k video? Because it is in the compression, encryption, transmission, and decryption of signals that they begin to deteriorate.

Of course, MOS today is much more scientific than simply asking the opinion of someone listening to a phone call in a quiet room, as was the case for many decades.
In telecoms today, it can be calculated using algorithms designed to estimate the experience as a human would perceive it. And to standardize this process, the ITU has created recommendations for these so-called ‘objective quality models.

What Causes a Low Quality Call?

We’ve mentioned the three major culprits already but let’s go into them in a bit more detail:

→ Latency

This is basically the lag between someone saying something and you hearing it on the other end. The lower the latency, the more your call will replicate the experience of two people talking as if they are in the same room. As latency increases, you’re likely to be left with uncomfortable interruptions and pauses; a problem that gets worse the more people are on the call. Latency is a huge issue in telecoms and is predominantly caused by three factors:

- **Distance**
  The further the distance between the points A and B, the longer it will take the signal to travel between them. That’s just science. Anything under 100ms should be fairly imperceptible, but as you reach latencies in excess 300ms, there will be noticeable lag on the call.

- **Complexity**
  By which we mean how complicated the routing of the call is. If it needs to hop between multiple networks and travel through a point-of-presence on the other side of the world before coming back, that will all add to the latency. The combination of these two factors is why latency can become so high with international calls made on the PSTN.

- **Encryption**
  Adding VPN and other forms of encryption to a call’s signaling and media further increases latency because of the time needed for the encryption and decryption processes.
→ Jitter

In effect, jitter is the change in latency over time. Again, as the complexity of the routing increases and more variables come into play, it becomes ever more likely that the latency on a call will change from moment to moment. Jitter will be especially apparent if your call needs to traverse the public internet, where routing is so unpredictable. It usually sounds like interference on the line.

It can be better to have a higher latency call with no jitter than a call with a lower average latency but very high jitter as, to some degree, the brain is able to accommodate low levels of latency as long as the delay is predictable. Jitter removes that predictability.

→ Packet Loss

Everyone has heard it recently. When someone on your team begins to sound a bit like a robot, what you’re experiencing is packet loss.

When voice signals are digitized and transmitted, they are split into packets that can be routed asynchronously and even via different paths, before being reconstituted and the end-point of the call. But if some of these packets fail to reach the end point – usually as a result of network congestion or failed hardware – small pieces of the audio signal will be missing, resulting in audible distortion on the call.

Why Do These Things Happen?

→ Problems with Network Hardware

As your network switches, and routers inevitably face the test of time they’re going to slow down your network traffic as they become faulty or outdated. If you’re experiencing a sudden growth in the company this can be exacerbated. As your throughput shoots through the roof, you’re going to start seeing more lag, packet loss, and reduced overall connectivity. Make sure you keep an eye out for equipment that needs to be revised and updated to keep on top of this.
→ Software Bugs

Whatever software you are using, regardless of how expensive it is, there are always going to be bugs in a platform that create problems. This is not unheard of in the VoIP space and if you’ve just had a recent update on your voice software, and you’re getting some unusual network behavior like packet loss, this could be the problem.

Software bugs are another common cause if rigorous testing has not been carried out, or bugs have been introduced following software updates. Sometimes rebooting can resolve these issues, but more often than not the software will need to be updated or patched.

→ Security Threats

It’s one that doesn’t happen often but should be seriously considered when looking at your network setup. There’s a particularly popular cyberattack that has been going round in recent years called the packet drop attack.

This is when a cybercriminal takes control of a router and sends commands that drop packets into a stream of data.

If you’re suddenly noticing high rates of packet loss across your network, it could be that there’s a cyber attack in progress.

→ Network Congestion

A network is considered congested when the network traffic hits its maximum capacity. Back in the day, when your channel limit was reached this would result in a dead tone. But in the age of VoIP, packets will wait their turn to be delivered when there’s enough bandwidth.

But if your connection falls so far behind that your software can’t store any more packets, they will simply be discarded or ignored so that the network can catch up.
How Do You Prevent Low Quality Calls?

There are a number of ways to resolve issues with your call quality, and the solution you choose will depend on the specific reason for quality deterioration. In cases where hardware is at fault, the hardware will usually need to be replaced with new appliances that are able to cope with maximum throughput.

If a security risk or network congestion is the cause, you have a number of options available to you. But, ultimately the way to ensure you rid your calls of as much deterioration as possible is to take a closer look at your connections.

→ Improving Your Call Quality

Ultimately the way to improve your call quality is to rid your calls of as much latency, jitter and packet loss as possible.

If you test the quality of your lines and the result shows them to be wanting, what exactly can you do about it? To answer that question, you'll first need to consider precisely where in the call journey that performance is dropping. After all, you can't troubleshoot a problem until you've isolated it.

There are three main areas where issues can occur that will harm call quality:

1. In your own network
2. In your voice provider's network or downstream
3. The connection between 1 & 2

We'll take a look at each of these in turn. But first, let's think for a moment about how you can accurately peg a quality issue to one of these three stages in the call journey.
Troubleshooting Quality Issues Affecting Your Own Network

Ultimately, at this stage of the call journey, it’s in the hands of your IT team or network administrator to ensure that everything is set up for optimal quality.

Most of the call journey will be the responsibility of your voice provider to guarantee. If you’re working with reputable operators that have their own in-country infrastructure and number ranges – or source them directly from a Tier-1 carrier where regulations make it difficult for external providers to operate directly – then this becomes much simpler.

If there are issues with a country’s national infrastructure, it’s likely to affect all operators equally and there is little a provider can do beyond pushing for a swift resolution and keeping you informed.

But if your provider aggregates voice coverage from unreliable carriers or is regularly affected by network downtime and loss of coverage, that’s the point at which you should consider looking for a different provider.

Squashing Quality Issues That Pop Up In Between

Assuming your IT team knows what it’s doing and you’re working with a decent voice provider, the most likely cause of quality issues is actually the interconnection between your network and your provider’s.

The number one culprit here is the internet. If you don’t have a dedicated private connection – whether through an SD-WAN provider like Megaport or a direct physical cross-connect within a datacenter – your calls will have to traverse the public internet for this leg of the journey.

While the internet offers the broadest possible connectivity, enabling you to send and receive calls from any location, its public nature can have serious negative repercussions for the quality of real-time media including voice and video.

Because routing can be so unpredictable and your traffic is in contention with literally everyone else’s, not to mention the impact that traffic management policies can have on delivery. High latencies, packet loss and jitter are all much more likely on the open web.
Another major issue with the public internet is security. Encryption is one solution, but bear in mind that this will actually further deteriorate the quality of your calls by adding additional latency in the form of encryption and decryption times, not to mention the risk of higher levels of jitter.

If you’re connecting over the public internet, but wish to benefit from improved security, you’re probably using encryption in the form of VPNs.

One thing to keep in mind, though. If you encrypt data at any point in your flow, you face a cost. Adding encryption and decryption times at each end of your network increases latency.

When it comes to voice services, enabling AES encryption has been found to add up to a 9% increase in latency and 30% to the jitter of a call – under ideal conditions!

To put it bluntly, encrypting a call degrades its quality. When compounded with other issues, this could cause the end user to end up with a ‘low quality’ call experience.

How do you tell where poor performance is entering your call flow if it isn’t the result of something starkly apparent, such as a provider network outage?

Up until now, this has been very time consuming and very expensive to do in real-time, relying upon a combination of carrier metrics (if even available) and third-party solutions that usually entail probes across your network.

So it’s not surprising that all but the most devoted of call quality evangelists either fall back to relying on sporadic automated number testing or don’t bother at all.
Private Interconnections Close to the Source

What’s one thing you can do to get your call traffic off the public internet and start improving quality?

While you may be forced to connect to your customers via the internet, there are certainly better solutions to interconnecting your various infrastructure services.

Would you ever trust public transport enough to get valuable or confidential cargo from A to B? My guess is no. The same goes for your data; why would you use the public internet to link the backbone of your application together?

We’ve seen a major push in the communications sector towards the use of private interconnects. The benefits of this approach are numerous:

- More secure transportation of traffic without relying on VPNs
- Fewer hops your packets have to make between A and B
- Prioritization of traffic
- Lower latency between the furthest points in your application
- Better redundancy of your service
- Ensuring Reliability – Keeping Control

Here’s a scary statistic. Business end users in the United Kingdom are likely to face 2 major downtime events per year, due to issues with their ISP.

While the picture is improving and the average typical duration of such downtime events is tracking downwards, this is generally thought to be because IT is getting better at managing poor connectivity, not because of any overall improvements in connectivity.

As a result, IT staffing is becoming an increased cost center due to the number of people required to monitor and react to issues like a bad connection or downtime.

The best way of avoiding such issues is to directly connect your infrastructure over a private network where you have full control over the design, capacity and media – for QoS and SLA. Being able to see where and how your traffic will be sent before it traipses over a network is a sysadmin requirement for maintaining the quality of your service. This is another reason why private interconnections are becoming more popular.

By establishing a dedicated private link between your network and your supplier’s, you can cut out a lot of the jitter and packet loss you’ll encounter online, as well as bringing latencies down and improving security. Albeit at a greater financial cost. But one thing to always keep in mind is that latency is based on the laws of physics. The further a distance that a signal has to travel, the longer it will take. So even with a private interconnection, if you’re sending data halfway around the world, there’s still going to be more latency.
Want to Make All This Simple?

Usually, measuring the quality of your calls requires expensive third-party tools that only provide sample 'snapshots' rather than ongoing and extensive monitoring, or expensive equipment to be installed directly into your network infrastructure.

For Voxbone customers, we provide all the quality metrics you need out of the box via our new Insights platform, which is currently in beta and being rolled out in phases this year. With Insights, you can get metrics such as MOS, jitter and packet loss on every single call you pass through our network.

Taking things a step further, we are even able to split these metrics in two – providing a granular breakdown of the quality of calls as they pass through our network and as they transit through yours.

This lets you see at a glance if quality issues are the result of an error on our network or due to external factors such as misconfiguration on your own network or internet congestion.

As a result, you'll be able to more rapidly troubleshoot quality issues and quickly understand where problems are arising, to know the appropriate actions to take to resolve them.