

# 8 Metrics to assess the **MICROSOFT TEAMS USER EXPERIENCE**



# 1

## *Latency One-Way*

*or Ping In Millisecond and Round-Trip Latency*

- ▶ It measures the time taken to send a data packet from point A to B and come back. It is tied to physical distance between the two points, the speed of transmission and the overhead taken by the routers in between.
- ▶ The latency impacts the smoothness of the conversation between two people.
- ▶ Increased latency results in unnatural pauses during the conversation.
- ▶ Poor latency during a conversation is often reported as similar to using a satellite phone.
- ▶ This leads to people talking at the same time.
- ▶ Poor latency is a common experience for many users.

# Packet Loss Rate

## 2

- ▶ Microsoft recommends keeping it at 1% during a 15-second call.
- ▶ Less than 3% - should provide a decent call quality.
- ▶ Between 3 and 7% - there is a noticeable impact during the call.
- ▶ Over 7% - the call quality will be severely lacking.
- ▶ It represents the amount of packet loss for 15 seconds - for example if 1000 packets are sent in 15 seconds and 50 are lost, it will generate a 5% packet loss.
- ▶ This measure is extremely important in VoIP; used as one of the elements to determine the Mean Opinion Score (MOS) discussed later in this guide.
- ▶ A high level of packet loss will lead to periods of silence during a call (if you have a period of sustained packet loss during a call) or to a degradation of voice quality, giving people 'robot-voices'.

# 3

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## *Packet Reorder Ratio*

- ▶ This statistic is now considered equally as important to packet loss, because packet reordering (when packets arrive in a different order than they were sent) severely degrades call quality.
- ▶ A high-rate flow network has a greater chance of packet reordering.
- ▶ Packet reordering can be recognized as packet loss or as a congestion signal by most network protocols (like TCP). Therefore, they impact the network MOS the same way as packet loss. In addition, this can directly impact the packet sending rate, which will also increase the round-trip time and result in performance degradation.
- ▶ As a result, the call will be distorted and will cut out at times. The threshold where you need to be alerted is usually 0.05%.

# Jitter 4

*Also Called Packet Inter-Arrival Time*

- ▶ Audio packets are sent at regular intervals on the network. But that doesn't mean that they are received with the same regularity - usually because of network latency. That is why a buffer is needed, this waits for all the packets before reconstructing them in the correct order.
- ▶ The Jitter is the size of the buffer that is needed to store packets before reconstructing them in the correct order. It can be compared to an audio packet 'waiting room'. The value of Jitter is calculated over every 15 second period.
- ▶ A low Jitter number means that the connection to the call is good and solid.
- ▶ A large Jitter buffer provides additional delay in calls. It is the sign of congestion on the network.
- ▶ The shrink or the expansion of the buffer will result in audio distortion during the call which could result in the speeding up or slowing down of the callers' speech.
- ▶ When it comes to packet loss, the Jitter value is used in the Network MOS determination.

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## *The Network MOS*

### *Network “Mean Opinion Score”*

MOS is a core concept to understand when looking at Microsoft Teams performance.

The MOS is usually a score that is based on a questionnaire sent to the users, like the one you have after each call on Skype, Teams or WhatsApp: “Please rate the quality of the call from One to Five stars” (5 being excellent, 1 being poor).

That is why you can’t have a MOS superior to 5. This becomes challenging when you want to assess your environment (especially during migration) as you don’t have enough user feedback to get this score.

Microsoft has worked with other network specialists on a definition of the Network MOS metric that can predict the value of the Wideband Listening Quality Mean Opinion Score (MOS-LQ).

To calculate this metric, multiple factors are considered such as the latency, packet loss, Jitter, the codec used, etc...

As for the real MOS, the Network MOS ranges from 1 to 5, but due to the compulsory impact of the audio codec, the highest score is usually around 4.4.

## HOW TO USE THE MOS VALUE?

The Network MOS is a perfect tool to identify if network conditions are impacting the end-user audio quality experience. It can be used to identify a wide range of issues.

The first way to analyze Network MOS is to compare it with a previous average value in time to understand if you are dealing with a degradation of the audio quality or not.

You can then combine this information with the packet loss or Jitter to understand what is causing that degradation and in which location. This is why it's imperative to constantly measure from all locations.



## ***IDENTIFYING ROOT CAUSE WITH KEY PERFORMANCE METRICS***

Now, that we have looked at Network MOS, let's look at identifying the root cause of an issue with the key performance metrics we have outlined above.

LAN Congestion is typically the primary indicator of root cause. When your LAN starts to be overloaded, the rates of packet loss and amount of Jitter is increasing for all calls going through your LAN.

This will automatically be reflected the MOS score.

If you trend your MOS score you will be able to understand exactly at what time and where the LAN was congested, determining your peak usage time and subsequently adjusting your network to prevent this issue from happening again.

This illustrates how important it is to constantly monitor all of these specific data points, to take proactive action before users start complaining and opening tickets.

# *Network MOS Degradation*

Another way to analyze the Microsoft Teams end-user experience delivered to your remote locations is to understand the Network MOS degradation.

This statistic directly shows the impact of Jitter and packet loss on the Network MOS. It is generally recommended to keep the value under 1. The higher this value is, the worse the quality of the call will be.

Usually, a high value of Network MOS degradation causes distortion in the audio or a silent period during the conversation. This is generally due to LAN and wireless congestion and/or insufficient bandwidth, which directly impacts the rate of packet loss and average Jitter.

The statistic is interesting as it is less impacted by the codec used during the call than it is impacted by the actual MOS. Knowing this, you can analyze the metrics to understand if the loss of MOS is mostly due to Jitter or to packet loss and then further troubleshoot the issue.

## ***HOW PACKET LOSS AND JITTER ARE RELATED TO THE NETWORK MOS SCORE***

Packet loss plays a very important role in the Network MOS calculation.

The latency round trip will not affect the MOS if it stays under 100ms.

If the Jitter stays under 50ms, the MOS will be as high. When the Jitter rises, the MOS degrades quickly.



## ***BASIC RECOMMENDATION ON NETWORK MOS***

As the MOS is a prediction of the end-user experience of audio quality, it is important to constantly measure it. Jitter and packet loss should also be measured frequently. Make sure that the Jitter value stays under 50ms, the packet loss is as close to 0% as possible and of course, latency should be under the 100 milliseconds threshold.

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## *Ratio Concealed Sample Average*

An audio sample can be concealed to deal with lost network packets. It is a simple solution to smooth out the abrupt transition that is caused by dropped network packets. That is why this statistic is often called “Healed Percentage”.

A high percentage means that many audio samples had to be concealed because of the packet loss. It indicates poor audio quality, which the user experiences as distorted or lost audio.

Usually, it is best to keep the ratio around or below 2%; over 2% will gradually degrade the audio quality and at over 7% users will likely end the call.

# *Estimated Bandwidth*

# 8

*Bandwidth Minimum, Maximum  
and Average*

Microsoft Teams adapts well to varying bandwidth, by reducing or increasing the packet sending rate. However, bandwidth naturally has a significant impact on the audio quality. Generally, when the bandwidth is under 100Kbps, the quality of the call is degraded. Video calls are even more sensitive to bandwidth variation and packet loss, which is why it is critical to measure bandwidth and keep it within Microsoft's recommendations.

Martello's solution measures this by installing Robot Users at each of your sites. These Robots use Microsoft Teams exactly the way your users do by performing synthetic transactions. During Microsoft Teams monitoring, Martello's Robots capture every network statistic - Jitter, packet loss, healed ratio, MOS, etc... - to determine whether there is an issue.

With Robot Users, you have an easy way to test and collect every statistic you need to assess performance or determine the readiness of your environment to deploy Microsoft Teams.

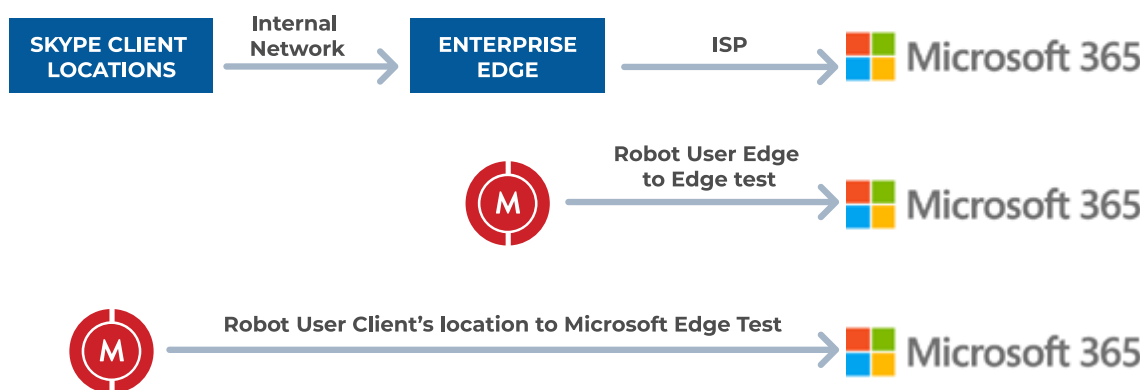
This also provides IT teams with an easy way to store and analyze the statistics with PowerBI reports.

# MICROSOFT'S RECOMMENDED RESULTS FOR READINESS ASSESSMENT

One way to measure expected Microsoft Teams performance is to take the readiness test recommended by Microsoft. This test will be performed from 2 different 'exit points' to analyse and identify which part of your network might potentially cause an issue.

Best practices would have a Robot User performing the following tests at each location;

- ▶ Directly from where the clients sit to the Microsoft Edge
- ▶ From your infrastructure's Edge to Microsoft Edge



Measuring performance in this way will give you a good understanding of how your infrastructure is impacting the call quality of your Microsoft Teams services.

Outlined in the table below are the recommended values for each:

METRIC	CLIENTS TO MICROSOFT EDGE	CUSTOMER'S EDGE TO MICROSOFT EDGE
Latency (one way)	< 50ms	< 30ms
Latency (RTT/Round trip time)	< 100ms	< 60ms
Packet Reorder Ratio	< 0.05ms	< 0.05%
Packet Loss	< 1% during any 15s interval	< 1% during any 15s interval
Average Jitter during call	< 30ms during any 15s interval	< 15ms during any 15s interval
Network MOS*	> 4	> 4
Network MOS degradation	< 1	< 1
Concealed Sample Average Ratio	< 2	< 2
Bandwidth Estimate for Audio	> 100	> 100 KBPS
Bandwidth Estimate for Video	> 300 KBPS	> 300 KBPS

**\*Network MOS:** *The MOS score target typically remains the same. A MOS of > 4 will provide good audio quality. The closer you are to 4 the more satisfying the experience will be for the user, with fewer distortions.*

Between 3 and 4, you can still use the audio - the closer it is to 4 the better of course - but some users will still be impacted by the issues we explained earlier in this guide. This means you should look at the location to see if an improvement to the network quality can be made before deploying Microsoft Teams.

If you have locations with a MOS under 3, consider holding off on deploying Teams at this time as more tickets, and user frustration, may result.

A thorough examination into your network is necessary before proceeding.

## ***NEXT STEPS WITH THESE VALUES***

Now that we have the metrics and associated target values, there are a few ways in which to use this data.

1. Compare them with your locations' values over time to quantify whether your users are having a good Microsoft Teams experience. Knowing your peak times during the day and week will also help you isolate this comparison to actual usage.

It is easy to recognize these peak times with your PowerBI reports that draw all your statistics over the days and weeks. You can then identify what moments of the day/week have the highest usage and compare the results of your tests with the Microsoft recommendations.

You will see if your network can handle the load. If not, you will know specifically which location fails and at exactly what time and period during the month. This knowledge will have your team fully prepared to either improve the network or handle the users' complaints.

2. With this data, you can use some of these metrics in concert with Real User Monitoring (RUM) capabilities. The Microsoft Teams call quality data – made up of Jitter, packet loss, and network MOS, along with video frame rate data from Teams – are all collected on a per-user basis and can help establish whether a specific user is actively having service quality issues. This RUM data can be correlated with synthetic transaction details from Robots running at the same location as the potentially impacted user to provide even more context when assessing the Teams user experience.

## *REACHING THE TARGET*

From a general point of view, to reach these targets you should make sure that:

- ▶ The latency one-way is stable during busy and non-busy hours.
- ▶ The latency RTT is stable and stays close to twice the one-way latency.
- ▶ Packet loss is not increasing significantly during busy hours - though it is normal if it rises.
- ▶ Jitter does not increase significantly.
- ▶ Edge connection tests stay better than client connection tests.

If all your locations pass the test during both your high volume and low volume hours, you can be confident in the delivery of your Microsoft Teams services. Within these targets, user complaints and tickets should stay low, and you can achieve an optimal ROI.

One thing to keep in mind though is that your infrastructure and the way it is used it is in constant flux. You should continue to monitor the Microsoft Teams metrics outlined in this guide to prevent any potential surprises and to proactively predict issues that could arise at particular locations.

The best way to do this is to keep your Robot Users constantly running to monitor user experience.

# WHAT HAPPENS IF A SITE FAILS?

## DATA

We have already outlined how each of the metrics above have an impact on audio quality. Now, let's go through an example of a site that is failing the tests.

METRIC	TARGET	RESULTS BUSY HOUR DURING A NORMAL DAY	RESULTS BUSY HOUR DURING BUSIEST DAY OF THE WEEK
Latency (one way)	< 50ms	40ms	38ms
Latency RTT	< 100ms	81ms	77ms
Packet Reorder Ratio	< 0.05ms	1%	2%
Packet Loss	< 1% during any 15s interval	4%	5%
Average Jitter during call	< 30ms during any 15s interval	25ms	35ms
Network MOS	> 4	3.7	3.3
Network MOS degradation	< 1	0.7	1.2
Concealed Sample Average Ratio	< 2	1.7%	2.1
Bandwidth Estimate for Audio	> 100 KBPS	150 KBPS	50
Bandwidth Estimate for Video	> 300 KBPS	150 KBPS	150 KBPS

## DATA INTERPRETATION

As you can see in the table above, most of the results exceed the target. Clearly, during the busiest day of the week the network cannot handle the volume at peak times. This indicates that the site will not be able to provide a productive Microsoft Teams user experience.

As a result, the calls during this time will have reduced quality (PSTN, peer-to-peer and conference) which means:

- ▶ Users will experience periods of silence during the call.
- ▶ The speech will sound robotic.
- ▶ The participant's speech will suddenly slow down and then speed up.

In general, the quality will be poor, users will complain, and you will soon see tickets coming from that location. This will increase your administrative costs as well as your user's dissatisfaction.

If you have a similar result, you should absolutely consider improving your network before deploying Microsoft Teams.

With constant monitoring of each location you can easily rank and organize them to determine the overall ROI as outlined in the table below.

SITE	NUMBER OF USERS	SENSITIVITY	RESULTS	COST	ROI	ACTIONS
Boston	1000	High	Pass	Zero	High	Monitoring
Geneva	500	Low	Fail	High	Low	Stand-By
Singapore	200	High	Fail	Medium	High	Network Improvement

# CONCLUSION

In our current 'work from anywhere' world, the increased dependency upon Microsoft Teams has quickly made it one of the most critical applications to ensure business continuity. Users depend on Teams to perform well and consistently for conference calls, meetings, collaboration, file sharing - and beyond. Given its importance to an organization, it's imperative that IT teams take a proactive approach to monitoring and are constantly measuring the top metrics outlined in this guide to provide an exceptional Microsoft Teams user experience.

Martello's digital experience monitoring (DEM) solutions can proactively help IT teams surface these key Microsoft Teams metrics. Our solution provides greater visibility into Microsoft Teams performance with precise insights and actionable context for IT teams to dramatically improve the end-user experience.

## ***FREE TRIAL FOR MICROSOFT TEAMS MONITORING***

**GET STARTED**



Martello Technologies Group Inc. (TSXV: [MTLO](#)) is a technology company that provides Service Management & Digital Experience Monitoring (DEM) solutions. The company's products provide monitoring and analytics on the performance and user experience of critical cloud business applications, while giving IT teams and service providers control and visibility of their entire IT infrastructure. Martello's software products include Microsoft 365 end user experience monitoring, unified communications performance analytics, and IT service analytics. Martello is a public company headquartered in Ottawa, Canada with employees in Europe, North America and the Asia Pacific region.

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