

Performing a VoIP Assessment with Vivinet Assessor

Contents

Voice and Data Differences...	1
Why Run a VoIP Readiness Assessment?	1
What is Involved in a VoIP Readiness Assessment?	2
How a VoIP Quality Assessment is Performed	3
NetIQ Performance Endpoints	4
Score Calculation.....	5
What You Need to Know About VoIP	6
Factors Included in Call Quality Estimation	10
Planning an Assessment	13
Designing the VoIP Quality Assessment	15
Scheduling the VoIP Quality Assessment	17
Verifying and Running the VoIP Quality Assessment....	18
Reviewing Vivinet Assessor Reports	18
Summary	21

White Paper

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Abstract: Successfully deploying Voice over IP (VoIP) in an enterprise data network may have some unexpected pitfalls. This paper describes how to determine whether a data network is ready to run VoIP traffic well.

As part of a suite of products designed to test, evaluate, monitor, and manage multi-protocol and multi-platform data networks, Vivinet™ Assessor measures and analyzes the salient factors that affect the performance of call traffic on data networks, thereby determining how well VoIP is likely to perform on a particular network.

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Voice and Data Differences

Even when they are carried on the same network, voice traffic and data traffic cannot be handled the same way:

- They have different packet sizes.
- They are sent at different rates.
- They are buffered and delivered to the application differently.
- They must fulfill very different user expectations.

Although an e-mail message or a file transfer can be delayed by half an hour without exciting notice, delays of a few hundred milliseconds can ruin a VoIP telephone call. Because everyone is used to the near-perfect call quality delivered by the Public Switched Telephone Network (PSTN), VoIP phone users will not tolerate even a modest amount of delay. Once you roll out VoIP, they will complain that call quality is dismal even if network performance is pretty good.

When you start to run VoIP across any given enterprise network, delays caused by other applications, overloaded routers, or faulty switches may be inevitable. To help you avoid the headaches associated with poor VoIP performance, Vivinet Assessor is designed to:

- Assess VoIP readiness, determining whether an existing data network is ready to deliver quality VoIP calls in its current configuration.
- Evaluate VoIP call quality over the course of a few days, running hundreds or even thousands of simulated calls over the network and taking measurements.

Vivinet Assessor monitors the routers, switches, and WAN links on a given network to determine their present utilization rates. It assigns them a VoIP Readiness Rating indicating whether upgrades are needed before additional VoIP packets are added to the mix. In addition, it sends simulated VoIP traffic between selected points on a network for a period of time you select. While the simulated calls are running, Vivinet Assessor takes measurements and calculates call quality scores. Vivinet Assessor can generate reports that quantify the data collected over the course of the assessment. These reports help determine a network's readiness and capacity for handling real VoIP traffic.

Why Run a VoIP Readiness Assessment?

If you are thinking about implementing VoIP or expanding the use of VoIP applications, you are probably uncertain whether an existing network is ready to carry high-quality voice transmissions.

In an effort to lower telecommunications costs, organizations often deploy VoIP by switching voice traffic over to under-utilized WAN and LAN links. However, it is unlikely that a typical data network is configured to handle VoIP traffic well. According to Gartner, approximately 85% of today's networks will require upgrades to their data networks to properly support high-quality VoIP traffic¹. Voice traffic is uniquely time-sensitive. It cannot be queued, and, if datagrams are lost, the conversation is choppy or even incomprehensible. NAT-enabled firewalls, slow or congested links, or improperly implemented QoS schemes are just a few of the factors that can inhibit or even prevent VoIP traffic from crossing a data network in an acceptable form.

A Vivinet Assessor VoIP Readiness Assessment is designed to quantify current network load and assign a quality score to VoIP traffic traveling across the network. Such measurements provide information that cannot be gleaned from a pilot that simply uses an IP PBX and a few dozen IP phones. Therefore, the assessment is conducted right on the network where VoIP will soon be deployed.

By the end of a VoIP Readiness Assessment, you know which technical factors are affecting call quality. Most important, you get these answers before you have spent a lot of money, time, and energy on a VoIP deployment. You can work through all the data network issues in advance so that by the time you actually start running the real VoIP piece, you have a data network that inspires confidence. You can also make cost-effective decisions about network infrastructure upgrades and application traffic once you know more about current utilization and can gauge how well VoIP will perform.

What is Involved in a VoIP Readiness Assessment?

In compiling a VoIP Readiness Assessment, Vivinet Assessor performs five different assessment tasks:

- **Network Inventory:** Scans your network to discover routers, switches, and links. Devices are queried and their configuration data is added to the database.
- **Configuration Assessment:** Compares information about device configuration gathered during a Network Inventory to a set of rules you supply, based on vendor recommendations, and reports any disparities.
- **Utilization Monitoring:** Monitors discovered network devices and links to determine their current CPU, memory, and bandwidth utilization, plus other significant metrics.
- **Bandwidth Modeling:** Compares VoIP usage scenarios to estimate the call capacity of the network links you are monitoring.
- **VoIP Quality Assessment:** Determines how well VoIP will perform on your network by measuring simulated VoIP traffic and calculating Call Quality based on a Mean Opinion Score (MOS).

With the Network Inventory, you can populate the assessment database with information about the devices already running on your network. Then you can assess their VoIP readiness during the Configuration and Utilization assessments, and analyze their bandwidth requirements by performing Bandwidth Modeling.

Each assessment type allows for customization to suit a particular network's unique VoIP deployment. For example, the Network Inventory and Utilization assessment are conducted only within the IP subnets you select. After a router, switch, or WAN link is discovered, you can edit the data used to identify it in the assessment database. And you determine the thresholds used to assign VoIP Readiness Ratings to network devices and WAN links. You can set a particular measurement threshold, such as router buffer utilization, a bit higher to create a more forgiving standard for VoIP readiness.

When you set up a VoIP Quality assessment, you can use pre-configured defaults for typical VoIP system defaults. For example, the defaults for VoIP packet sizes are derived from industry-standard equipment settings. Or you can tune these parameters to see how various technical choices affect call quality and bandwidth consumption. You can, for example, examine the effects of a half-dozen codecs representing various compression algorithms; you can also tinker with jitter buffer sizes, delay between datagrams, and silence suppression. Each configuration choice you make is reflected in the simulated traffic that Vivinet Assessor generates.

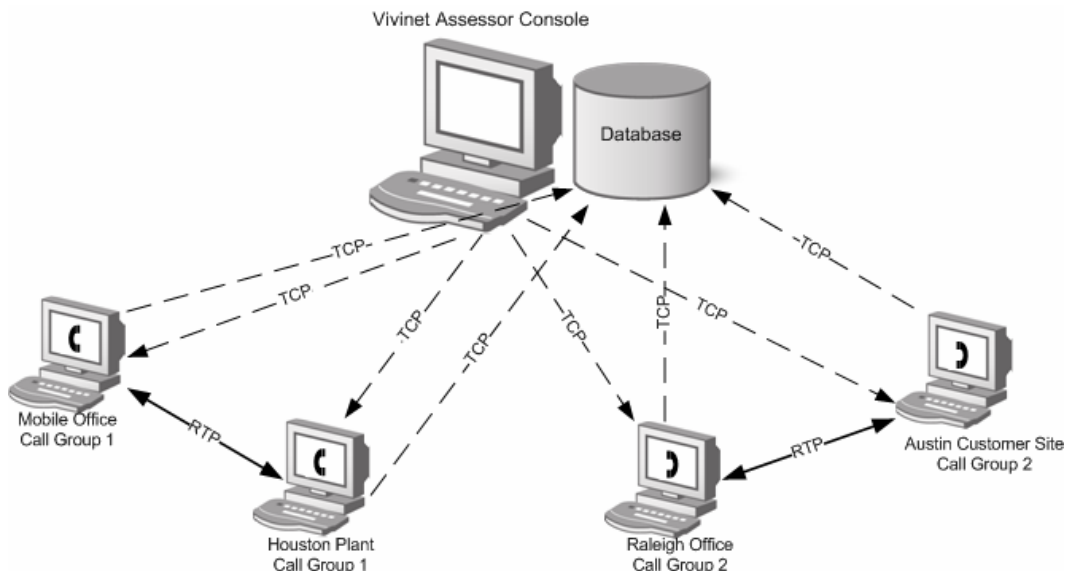
From the simulated VoIP flows, the assessment software measures delay, jitter, and lost data. Vivinet Assessor then calculates quality scores, assigns Readiness Ratings, and produces a comprehensive report of overall VoIP readiness. The report shows network utilization and call quality by day of the week, hour of the day, location, codec, and so on.

How a VoIP Quality Assessment is Performed

During the assessment of VoIP Quality, Vivinet Assessor determines how well VoIP will sound on a network by assessing the quality of simulated VoIP calls. To this end, Vivinet Assessor sends realistic VoIP traffic across the network and measures the resulting flows.

Vivinet Assessor relies on NetIQ Performance Endpoints, agents that are installed on every computer you plan to include in assessments. You deploy a Performance Endpoint or two at every site where there would be VoIP traffic. A small location might get one agent to measure WAN links; a larger site might have two agents: one at the edge to measure the WAN link, and another across the campus or building to measure traffic internal to the site. Although you can distribute endpoints anywhere on a network, including at remote sites, you do not have to visit those sites to find out how well VoIP is performing there.

You set up an assessment at the Vivinet Assessor Console by first selecting the type of VoIP traffic to send. The key parameter to identify is the *codec*, which determines how an analog voice signal is digitized and the rate at which it is transmitted. You create *call groups* that act as senders and recipients of calls. A call group consists of two endpoints connected by a *VoIP connector*, which defines the type and number of calls to be sent between the endpoints on a specified schedule. The following diagram illustrates how Vivinet Assessor and the endpoints work:



In this diagram, endpoints that belong to call groups are designated by a telephone receiver symbol. All assessment parameters, including the codec to be emulated and the schedule of calls to be made, are saved in a database. This database also contains any results from the assessment after it runs.

When you run an assessment, Vivinet Assessor contacts all the endpoints in each call group. It sends the endpoints a call script, telling them what codec to emulate, and a schedule to use. In the diagram, dashed lines indicate the setup flows. The endpoints then send the information to their partner endpoints within each call group.

As the assessment runs, shown by solid lines in the drawing, the endpoints take measurements and periodically send back results to Vivinet Assessor, which stores them in the assessment database. In the diagram, dashed lines, with arrows pointing back to the console, indicate reporting flows. The endpoints return results using the connection-oriented TCP protocol so that results are not lost. The simulated VoIP traffic uses the Real-time Transport Protocol (RTP).

NetIQ Performance Endpoints

Vivinet Assessor ships with the latest endpoints for several operating systems:

- Windows Me, Windows NT, Windows 2000, Windows XP, and Windows Server 2003
- Linux for x86
- Sun Solaris (x86 and SPARC)

It also runs to a Web-based endpoint for all supported 32-bit Windows operating systems. This endpoint does not have to be installed on a computer and can instead be run from its location on the Web.

We do not recommend using Windows Me and pre-ULTRA Sun operating systems. The high-precision clocks on these systems are not sufficiently accurate for good delay measurements.

The Performance Endpoint software is free; anyone on your network can quickly install endpoints on their computers by visiting www.netiq.com/download/endpoints and downloading the endpoint software appropriate for the operating system they are using. At the same site, they can also download documentation explaining how to install and deploy endpoints for all the supported operating systems.

A single endpoint can perform successfully in up to 50 two-way calls without sacrificing call quality. A two-way call is one in which a phone accepts one call while sending another.

Score Calculation

Vivinet Assessor calculates call quality based on a set of factors known to affect the perceived quality of VoIP transmissions. A subjective factor is necessarily part of evaluating VoIP quality because a listener must be able to understand the received transmission. Both talkers must be able to tolerate factors that often impede a conversation: the amount of delay between speaking and being heard (called the “walkie-talkie” effect), lost or fractured syllables, and echo.

Therefore, the chief unit of measurement for call quality in Vivinet Assessor results is an estimated Mean Opinion Score, or MOS. The E-model, ITU Standard G.107², quantifies what is essentially a subjective judgment: a user’s opinion of the perceived quality of a voice transmission. After much study, the ITU determined which impairment factors produced the strongest user perceptions of lower quality. The E-model thus includes factors for equipment and impairments, and takes into account typical users’ perceptions of voice transmissions.

- **Delay:** Delayed datagrams are perhaps the single greatest hindrance to VoIP call quality. This value includes all causes of delay between the endpoints.
- **Jitter:** Jitter, or variations in datagram inter-arrival times, affects call quality. Excessive jitter reduces call clarity and exacerbates datagram loss. For more information, see [Delay on page 10](#).
- **Lost Data:** When a datagram is lost, you can lose an entire syllable. The more datagrams that are lost, the more the clarity suffers.

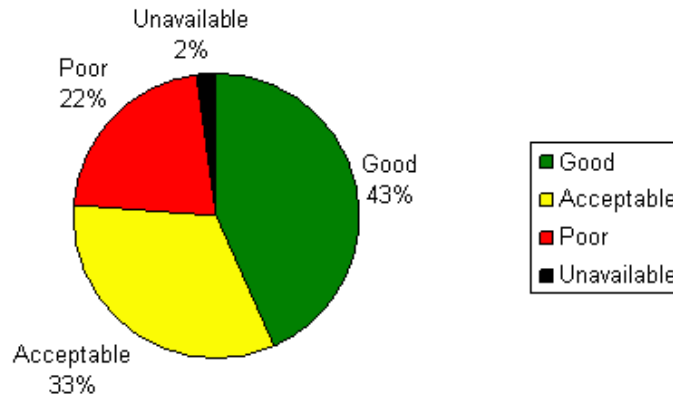
Vivinet Assessor reports show how many calls could not be completed (“Unavailable” calls) and give you the objective measurements of delay, jitter, and lost data so you can independently judge quality for yourself.

By default, Vivinet Assessor maps MOS estimates to three quality categories. You can change the way the categories are applied to conform to your own quality standards. A MOS of 5 is excellent; a MOS of 1 is unacceptably bad. The following table summarizes the relationship between MOS and user satisfaction:

MOS Range	Quality	Meaning
4.03 – 5.0	Good	Most or nearly all users satisfied
3.60 – 4.02	Acceptable	Some users satisfied
Below 3.60	Poor	Most or nearly all users dissatisfied

The following pie chart is an example of how overall call quality from an assessment is shown in Vivinet Assessor reports, using the default MOS values:

Call Quality Summary



Good - At least 4.03 MOS | **Acceptable - At least 3.60 MOS** | **Poor - Any lower value**

What You Need to Know About VoIP

Before you start running assessments, you need to know several things about your present or planned VoIP deployment. If you do not want to use Vivinet Assessor default settings, you should set or select the following Vivinet Assessor parameters to run simulated voice traffic on an IP network that closely resembles the real VoIP traffic you will be running.

- Seven different codecs, with different compression algorithms, data rates, and datagram sizes.
- Whether packet loss concealment (PLC) is enabled for the G.711 codecs.
- Voice datagram sizes.
- The ability to use silence suppression.
- A jitter buffer and its size.
- Any additional, fixed delay values that apply.
- Quality of Service (QoS).

Codecs

In audio processing, a codec (which stands for compressor/decompressor) is the hardware or software that samples the sound and determines the data rate. There are dozens of available codecs, each with different characteristics. Vivinet Assessor lets you perform VoIP assessments using seven call scripts representing the seven most common codecs.

Codecs have odd-looking names that correspond to the name of the ITU standard that describes their operation. The codecs named G.711u and G.711a convert from analog to digital and back with high quality, but a fair amount of bandwidth.

The lower-speed codecs, G.726, G.729, G.729A, and those in the G.723.1 family, consume less network bandwidth. Low-speed codecs impair the quality of the audio signal much more than high-speed codecs, however, because they compress the signal with lossy compression. Fewer bits are sent, so the receiving side does its best to approximate what the original signal sounded like.

Packet loss concealment (PLC) is an option if you are using the G.711u or G.711a codecs. PLC techniques reduce or mask the effects of data loss during a VoIP conversation. When PLC is enabled, Vivinet Assessor assumes that the quality of the conversation would be improved; this improvement is factored into the MOS estimate calculation of any data is lost. PLC makes the codec itself more expensive to manufacture, but does not otherwise add delay or other bad side-effects. If the VoIP equipment you plan to purchase uses PLC, be sure to enable it in your assessment.

In the table of codecs on the next page, the **Packetization Delay** column refers to the delay a codec introduces as it converts a signal from analog to digital. Packetization delay is included in the MOS estimate, as is the jitter buffer delay, which is the delay introduced by the effects of buffering to reduce inter-arrival delay variations.

Real bandwidth consumption by VoIP calls is higher than it first appears. The G.729 codec, for example, has a data payload rate of 8 kbps. Its actual bandwidth usage is higher than this, though. When sent at 20-ms intervals, its payload size is 20 bytes per datagram. To this, add the 40 bytes of RTP header (yes the header is bigger than the payload) and any additional Layer 2 headers. For example, Ethernet adds 18 more bytes. Also, the two concurrent G.729 RTP flows (one in each direction) double the bandwidth consumption you have calculated so far. The **Combined Bandwidth** column in the table shows a truer picture of actual bandwidth usage for some common codecs.

It is worth observing that both G.723.1 codecs make calls of only “Acceptable” quality at best. Their theoretical maximum MOS is below the 4.03 value needed to be considered “Good” quality.

Codec	Default Data Rate	Default Datagram Size	Packetization Delay	Combined Bandwidth (Two Flows)	Default Jitter Buffer Delay	Theoretical Maximum MOS
G.711u	64.0 kbps	20 ms	1.0 ms	174.40 kbps	2 datagrams (40 ms)	4.40
G.711a	64.0 kbps	20 ms	1.0ms	174.40 kbps	2 datagrams (40 ms)	4.40
G.726	32.0 kbps	20 ms	1.0 ms	110.40 kbps	2 datagrams (40 ms)	4.22
G.729 G.729A	8.0 kbps	20 ms	25.0 ms	62.40 kbps	2 datagrams (40 ms)	4.07
G.723.1- MPMLQ	6.3 kbps	30 ms	67.5 ms	43.73 kbps	3 datagrams (60 ms)	3.87
G.723.1- ACELP	5.3 kbps	30 ms	67.5 ms	41.6 kbps	3 datagrams (60 ms)	3.69

Determining Payload Sizes

Each call script lets you “override delay between voice datagrams.” This option is recommended only for advanced users; it is not likely that you will need to set it. However, the following explains what happens when you choose to override the delay associated with the codec you are using.

The delay between datagrams determines the payload size to be used in the simulated VoIP calls. VoIP applications break voice data into chunks based on the amount of time between successive datagrams. Each call script uses a value appropriate to its codec: the faster codecs use 20 ms and the G.723.1 codecs use 30 ms. In other words, for example, every 20 milliseconds, the VoIP application adds a header to any voice data it has received and sends the datagram. Smaller times mean that the header-to-payload ratio is larger: more, smaller datagrams are sent, which increases processing overhead. Longer times mean fewer, larger datagrams are sent.

Unless you are an equipment manufacturer, you are better off leaving the datagram sizes at their default values. These values were selected because they match those of the codec being emulated and allow for realistic simulation of VoIP datagrams.

Some VoIP equipment manufacturers refer to the delay between voice datagrams as the *speech packet length*. For example, at 64 kbps, a 20-millisecond packet implies that the sending side creates a 160-byte datagram payload every 20 ms. A simple equation relates the codec speed, the delay between voice packets, and the datagram payload size:

$$\begin{array}{l} \text{Datagram payload size (in bytes)} = \\ \text{Codec speed (bits/sec)} \times \text{packet delay (ms)} \\ \hline 8 \text{ (bits/byte)} \times 1000 \text{ (ms/sec)} \end{array}$$

In this case:

$$160 \text{ bytes} = (64000 \times 20) / 8000$$

For a given data rate, increasing the delay increases the datagram size, because datagrams are sent less frequently. A packet delay of 30 ms, at a data rate of 64000 bps, means sending 240-byte datagrams.

Silence Suppression

Call scripts can include *silence suppression*. Performed by the codec, VoIP silence suppression means that no data is sent on the network during periods of call silence, that is, when no one is talking during the call. If you were to set the silence suppression option to use a *voice activity rate* of 50%, then data would be sent during 50% of the simulated call’s duration.

When doing your initial assessments, we recommend turning off silence suppression, to provide a more intense assessment of the network. Without silence suppression, you see behavior in the worst case, where there is a lot of concurrent back-and-forth talking. To turn off silence suppression, set the voice activity rate to 100%.

Jitter Buffers

To minimize call disruptions from delay and jitter, VoIP phones and gateways typically use jitter buffers. A jitter buffer can be either *frame-based* or *absolute*: a frame-based jitter buffer holds a given number of voice datagrams; an absolute jitter buffer is based on time. For example, a frame-based jitter buffer might hold two datagrams, buffering them until a segment of the voice transmission can be reassembled to reduce inter-arrival time variability. An absolute jitter buffer, on the other hand, might be set to 43 ms, and, given a typical 20-ms speech frame size, could hold two speech frames and allow for an extra three milliseconds of variability.

It is in the nature of IP networks to exact a tradeoff. Buffering adds delay while smoothing out variability; therefore, adding delay can offset the positive effects of using jitter buffers. A goal of VoIP network tuning is to minimize jitter buffer sizes while maintaining call quality.

By default, all VoIP assessments emulate a frame-based jitter buffer of two datagrams. You can configure buffers based on time (in milliseconds) or on number of datagrams. The supported range of values is 10-1600 ms for an absolute jitter buffer, and 1-8 for a frame-based buffer.

Vivinet Assessor's jitter buffer does not smooth out jitter. Instead, it provides a more accurate MOS estimate by better accounting for datagrams that would have been lost due to underrun or overrun of the jitter buffer. In addition, the delay incurred by using a jitter buffer is factored into the MOS.

We suggest doing your initial VoIP assessments with jitter buffer sizes that match those being used in your planned VoIP equipment, if that information is known.

Additional Fixed Delay

This call script parameter lets you add a delay value to each VoIP call. Any fixed delay that you add should come from a known, constant source. For example, if you are emulating equipment that adds 10 ms of delay to each VoIP datagram sent, set the additional fixed delay parameter to 10 ms. The range of accepted values is 0 to 300 ms.

Quality of Service

Using a prioritization or QoS scheme can make a noticeable difference in call quality, particularly if you are running VoIP on a crowded enterprise network. The optional QoS parameter in Vivinet Assessor call scripts applies the QoS used in typical VoIP implementations to the simulated VoIP traffic. Generally speaking, VoIP QoS means that each voice datagram

- is assigned a priority that makes it unlikely to be queued or dropped, or
- is flagged to receive the lowest possible delay.

When you select QoS, Vivinet Assessor then lets you choose an IP TOX, a DiffServ, or an 802.1P prioritizing scheme. The DiffServ and TOS schemes set the IP Type of Service (TOS) precedence bits in the IP header, using the "CRITIC/ECP" setting. These same bits are known as the Expedited Flow (EF) setting in the Differentiated Services (DiffServ) standard³⁴. When routers read the bit settings, they prioritize the packets accordingly. By contrast, 802.1P creates a set of priority levels that operate at Layer 2 of the OSI model. The NIC cards themselves must set the bits. Unfortunately, neither the NIC card support nor the setting of the DiffServ or TOS bits is available by default on many networks. Check documentation for the endpoint operating systems, as well as for the NIC cards, routers, and switches in the network, to make sure the setting of these bits is supported and that the prioritization will be honored.

If you have not yet tried applying QoS to VoIP traffic, you might want to set up a few call groups that use QoS and a few others that do not. Depending on the routers' ability to process requests for QoS, you should see a difference in the groups' MOS values when you compare the results.

Factors Included in Call Quality Estimation

After a VoIP Quality assessment has run, you can generate two different VoIP Readiness Assessment reports of the results: an Executive Summary report and a Complete report. Each report provides an overall assessment of call quality for the entire network; the Complete report adds individual results for each call group.

To determine the relative quality of each simulated VoIP call made during the VoIP Quality assessment, Vivinet Assessor measures delay, jitter, and lost data.

Delay

End-to-end delay, the time it takes to get data across the network, is the primary indicator of the “walkie-talkie” effect in a VoIP call. People are used to talking at the same time in their telephone conversations, or at least hearing the other person speak in real time. Most listeners notice when the delay that prevents their words from reaching the other person exceeds 150 ms; when it exceeds 200 ms, they generally find it disturbing and describe the voice quality as poor.

End-to-end delay is actually made up of five components:

- **Propagation delay:** the time to travel across the network from end to end. For example, the propagation delay between Singapore and Boston is much longer than the delay between New York and Boston.
- **Transport delay:** the time to get through the network devices along the path. Networks with many firewalls, many routers, congestion, or slow WANs introduce more delay than an over-provisioned LAN on one floor of a building.
- **Packetization delay:** the time for the coded to digitize the analog signal and build frames — and undo it at the other end. The G.729 codecs has a higher packetization delay than the G.711 codecs because it takes longer to compress the signal. For more information, see [Codecs on page 6](#).
- **Jitter buffer delay:** the delay introduced by the receiver as it holds one or more datagrams to reduce variations in arrival times. For more information, see [Jitter Buffers on page 9](#).
- **Fixed delay:** an additional user-configured delay that can be specified when setting up an assessment. For more information, see [Additional Fixed Delay on page 9](#).

The combined value of propagation delay and transport delay is termed *network delay* in Vivinet Assessor. Packetization delay and jitter buffer delay are constants for any given call group.

Measuring response time (that is, round-trip delay) and dividing the resulting time measurement by two is not always a good approximation of one-way network delay. Response time hides assumptions about the symmetry of the paths between two endpoints. The two RTP streams in a VoIP call can take different paths through an IP network.

Endpoints therefore calculate network delay explicitly, rather than simply figuring half of the round-trip time. The endpoints start with flows similar to those used by the Network Time Protocol (NTP). NTP generally has an accuracy of about plus-or-minus 200 ms. Our design for giving good MOS feedback called for clock precision of about plus-or-minus one ms, which led us to design more precise algorithms for software-based clocking.

Endpoints maintain local copies of their clocks because each endpoint can be included in many simultaneous call groups. Also, the internal clocks in every different operating system and computer platform seem to be a little different, and the clocks drift apart over time.

The endpoints maintain virtual (software) clocks for each partner involved in a VoIP test. These virtual clocks consist of the offset between the microsecond clocks maintained by the two endpoints. The high-resolution microsecond clock is maintained independently of the operating system's system clock.

The endpoints paired with each other in a call group compare their respective views of the clocks before the start of each test and periodically during a test run. They also measure clock synchronization and drift between test runs, to establish a track record for the expected delay.

Our one-way delay algorithms have proven robust in measurements with thousands of endpoint pairs. We have also verified their effectiveness in testing with stratum 1 GPS timeservers.

Jitter and Jitter Buffer Loss

As simulated calls run during an assessment, the endpoints calculate jitter, a factor known to adversely affect call quality. Jitter, also called delay variation, indicates the differences in arrival times among all datagrams sent during a VoIP call.

When a datagram is sent, the sender (one of the endpoints) gives it a timestamp. When it is received, the receiver adds another timestamp. These two timestamps are used to calculate the datagram's transit time. If the transit times for datagrams within the same call are different, the call contains jitter. In a video application, jitter manifests itself as a flickering image; in a telephone call, its effect may be similar to the effect of some lost data: some words may be missing or garbled.

The amount of jitter in a call depends on the degree of difference between the datagrams' transit times. If the transit time for all datagrams is the same (no matter how long it took for the datagrams to arrive), the call contains no jitter. If the transit times differ slightly, the call contains some jitter. Jitter values in excess of 50 ms probably indicate poor call quality. They provide a short-term measurement of network congestion and can show the effects of queuing within the network.

When jitter is detected, Vivinet Assessor measures jitter buffer loss as well. Although jitter buffers smooth out jitter by feeding datagrams to the application in a steady stream, they also exacerbate data loss: datagrams that are not contained by the jitter buffer are discarded.

Jitter buffer loss totals are factored into the MOS estimate and also reported separately. By configuring result ranges in Vivinet Assessor, you can specify the levels of jitter that are considered acceptable on your network.

Lost Data

In VoIP Readiness Assessment reports, Vivinet Assessor includes statistics on lost datagrams, expressed as a percentage of all data sent in the relevant calls. For example, in charts indicating lost data by call group, lost data is expressed as a percentage of all data sent between the endpoints in the call group over the course of the entire assessment. Other charts might show data loss as a percentage of data sent at a certain time of day, averaged over the course of all data in the assessment.

VoIP packets are sent using RTP. Although every RTP datagram contains a sequence number, there is not enough time to retransmit lost datagrams. Any lost datagram impairs the quality of the audio signal.

RTP datagrams are lost in a data network for two primary reasons:

- Too much traffic causes datagrams to be discarded during congestion
- Too much delay variation causes datagrams to be discarded because they arrive at the listener's jitter buffer too late or too early.

Datagram loss occurs in a couple of different patterns. The first is a more-or-less random loss in which one or two datagrams are lost occasionally due to general, consistent congestion in the network. The other loss pattern is more damaging: packets are lost in bursts of, say, five or more at a time. Humans perceive bursts of loss as impairments to audio quality much more than general, random loss.

When a datagram is lost during a VoIP transmission, you can lose an entire syllable or word in a conversation. Obviously, data loss can severely impair call quality. Vivinet Assessor therefore includes data loss as a call quality impairment factor when calculating the MOS of each simulated VoIP call.

To measure data loss, one computer in each call group keeps track of how many bytes of data it sent. The sending endpoint reports to the receiving endpoint how many bytes it sent, and the receiver compares that value to the amount received to determine lost data.

As mentioned in our discussion of codecs, having PLC enabled for the G.711 codecs improves call quality in the presence of lost data. Always enable PLC in your assessments, if it is supported by the VoIP equipment you plan to deploy. For more information, see [Codecs on page 6](#).

Planning an Assessment

Before you design your first VoIP Readiness Assessment, you need to do some research on your network and your planned VoIP implementation.

Understanding Call Volumes

First, look at existing network documentation to find peak and average usage statistics. How many calls are likely to occur at the same time? What is their duration?

Telephone records and the current PBX call volume reports are a good source of data about the likely call volume a network will have to handle. What percentage of the calls occurs within each site? What percentage occurs within the organization, between sites? How many calls travel to and from the outside world?

Armed with information about likely call volumes, you will find that the Utilization assessment data is more valuable. Keep in mind that utilization levels are meant to indicate the amount of capacity you have to spare for the additional VoIP traffic you will soon add to the network. If utilization on a certain WAN link is pretty low at present, consider whether call volumes between the sites being linked will be similarly low. Use Bandwidth Modeling to determine how much bandwidth you will really need if you select one of the higher-bandwidth codecs.

You also need to know about call volumes when you are ready to set up a VoIP Quality assessment. For examples, we have several major sites in NetIQ. From our development site in Raleigh, we rarely call our sales offices in Japan and Europe. It makes sense to set up just one call from Raleigh to Japan and from Raleigh to Europe. An assessment generates several calls an hour, although in reality we probably make less than one call a day between these sites. An assessment seeks to ensure that you can get a connection and make a toll-quality call any time you want, so testing throughout the day is fine. We call more often from Raleigh to Seattle, so we would define maybe two calls between those sites. Finally, we make many calls to Houston, so we would define 10 simultaneous calls.

A VoIP Quality assessment is not a stress test; remember, you are running on a production network. Design VoIP quality assessments that run an approximation of the *average* call volume during work hours, as opposed to the *peak* call volume. There is a nice “failure mode,” though, that is easily observed. If the data network is already heavily loaded with existing application traffic, and then you add VoIP traffic, the VoIP traffic that “breaks” first — showing high delay, jitter, packet loss, or some combination. Check the report even before the assessment completes to see whether this is occurring. Or, you can make the following assumption: if, when you verify the VoIP Quality assessment, it shows “Poor” quality, you may not be dragging down the other traffic on the network. But you know for sure that network resources are being stretched too thin.

Understanding the VoIP Equipment

Next, check the vendor data sheets for the VoIP equipment you are considering, to answer the following questions:

- What type of codec will you be using?
- Will this codec use silence suppression or packet loss concealment?
- Can the network currently support QoS for VoIP?
- Are there any additional sources of delay to be considered?

- What size jitter buffer will be used?
- How many simultaneous calls can be supported?

You will want to emulate these factors as closely as possible when you configure assessments.

Most IP phones support more than one codec. The codec to be used is negotiable, and is usually centrally managed. When an IP phone is first plugged in, it downloads a list of which codec to try first, which to try second, and so on.

Does bandwidth consumption force you to use a codec other than G.711? It may be that the lower call quality is good enough in your setup. Also, silence suppression and PLC make telephones more expensive and are rarely supported today. If your phones do not support these options, do not select these choices when setting up the VoIP Quality assessment.

Deploying Endpoint Software

Before you set up a VoIP Quality assessment, you must install Performance Endpoint software on every computer you want to include in the assessment.

You want to put the endpoint software in the right locations, on the right computers, to get a representative assessment. But what are the right locations and the right computers?

Location: Look at the major offices in your organization — where most of the calls are made today. For example, at NetIQ, we have three development sites and four key sales sites. We would put at least two endpoints at each site. We would design the assessment to go from each site to each of the others, and define a call group between the two endpoints in each site. With these seven sites, we would define 7+6+5+4+3+2+1 call groups in the VoIP Quality assessment, for a total of 28 call groups.

When connecting within a site, look at the network topology of the site to see what will give the most telltale assessment. If you have multiple, interconnected switches, be sure to put the endpoints on different switches. If you sites have thousands of users and multiple buildings, consider an endpoint for each building or for each division.

Place endpoints in the spots where you suspect there will be problems. Do you have a mishmash of ISDN and T1 WAN connections that are stretched to their capacity? These links are better candidates for the assessments than a modern site where everyone has a 100-Mb LAN connection to their desktop.

Within a site, your endpoint placement should hinge on the VoIP topology in your plans. Where are the two ends of the IP connection? Do you plan to continue with analog telephones on the desktops, but use VoIP over the WAN? If so, install the endpoint software on a computer adjacent to the PBX or VoIP gateway. Alternatively, do you plan to use VoIP from desks within a site or campus, connected to the external Public Switched Telephone Network (PSTN)? In this case, put endpoints on a few desks, connected to endpoints adjacent to the gateway.

Install a few endpoints on either side of a WAN link, which should be part of any assessment. Install endpoints on either side of a firewall and set up call groups to test how well the firewall will handle VoIP traffic.

Computers: Finding the right computer for endpoint software is easier. The endpoint need not be installed on a dedicated computer. Do you have a computer available with some extra capacity; say a domain name server, a DHCP server, a print server, or a remote terminal server? These are good candidates for endpoints, as long as they are used for less than 30 calls groups. Otherwise, put the endpoint software on an extra computer for the duration of the assessment. Almost any Pentium II or higher is sufficient for a Windows or Linux endpoint; most SPARC or Ultra CPUs from the past few years will run a Solaris endpoint well.

Designing the VoIP Quality Assessment

Endpoints are integral components of VoIP Quality assessments; they are the computers with NetIQ Performance Endpoint software installed. The Design view lets you choose which endpoint computers to include in an assessment, and which simulated calls they will receive. A first step in setting up a VoIP Quality assessment is to instruct Vivinet Assessor to discover all the endpoints on the network and add them to a list of available endpoints. The endpoints' network addressing information is automatically entered into the database as they are discovered.

To design an assessment, you drag endpoints from the Endpoint List to the Design view diagram. If you have not performed an automatic endpoint discovery, you need to create some endpoints by dragging the **Endpoint** tool to the diagram. The Create an Endpoint dialog box lets you enter an IP network address to represent a VoIP-enabled computer on the network. Newly created endpoints are added to the Endpoint List to the right of the diagram.

The screenshot displays the Vivinet Assessor software interface in the 'Design' view. The central area shows a 'Network Diagram' with several nodes connected by lines representing network links. The nodes include 'NATC', 'Nashua', 'Houston', 'Houston', 'Seattle', and 'Test Lab'. The connections are labeled with protocols and bandwidths, such as 'G.711 k...' and '20.0 k...'. On the right side, there is an 'Endpoint and Group List' table with columns for 'Endpoint Name' and 'Host Address'. Below this is a 'Design tools' toolbar with various icons. Labels on the right side of the image point to these components: 'Design tools', 'Endpoint list', 'VoIP Connector', 'Background Traffic Connector', and 'Endpoint'.

Endpoint Name	Host Address
Houston	046.4.36
Nashua	046.4.35
Seattle	046.4.38
Test Lab	046.4.38

Next, connect the endpoints using VoIP connectors. Click the **VoIP Connector** tool, click on an endpoint, drag the connector to a second endpoint, and then click again. The Create a VoIP Connector dialog box lets you select key parameters for the voice traffic that will be sent between the endpoints, including the type of codec emulated and the number of simultaneous calls represented. The combination of two endpoints and a VoIP connector makes a *call group*, the basic unit of a VoIP Quality assessment.

To find out how VoIP calls will sound on the network during times of heavy or average network usage, you may want to add background traffic to your VoIP quality assessment. Click the **Background Connector** tool and following the process discussed for the VoIP connector. Because the background traffic connector lets you add a specific amount of TCP/IP traffic to emulate network utilization levels. You will be asked for the data rate at which the background traffic will be sent.

Creating an Endpoint

After you fill in the fields on the Create an Endpoint dialog box, an icon for the new endpoint appears on the network diagram. The key fields are described below:

Field Name	Description
Endpoint Name	Assign a name that makes it easy to identify the endpoint in reports.
Network Address	This is an endpoint computer's network address, or a set of addresses assigned to that computer's network interface card (NIC). Enter a DNS hostname, such as <code>westoffice_lab</code> , or the IP address of the endpoint computer in dotted notation, such as <code>135.25.25.5</code> .

Creating a VoIP Connector

By connecting two endpoints, VoIP connectors create call groups that emulate VoIP calls between endpoint computers. The Create a VoIP Connector dialog box lets you select the parameters that match your VoIP application.

Field Name	Description
From Endpoint	The endpoint that acts as the initiator of the call in this call group. Designated as Endpoint 1 in the Call Groups table.
To Endpoint	The endpoint that acts as the recipient of the call in this call group. Designated as Endpoint 2 in the Call Groups table.
Call Script	The codec used to send VoIP datagrams, plus any additional parameters that affect the calls that are send on the network.
Number of concurrent calls	The number of simultaneous calls Vivinet Assessor will emulate during assessments that include these endpoints. Values must be between 1 and 1250, inclusive. After you create a call group, the Call Count field in the lower right of the assessment window indicates how many VoIP calls will be emulated.

Design Tips

Create endpoints to represent the different areas of a network. Include at least one call group that spans a WAN link, assuming your deployment will include one. Create only one VoIP connector for each call script (codec) you want to apply between any two endpoints in the assessment. To add more calls, change the value in the **Number of concurrent calls** field in the Create a VoIP Connector dialog box.

Firewall Settings

If there is a firewall on the network between the endpoint computers, you have probably already configured it to accept RTP flows for VoIP traffic. You also need to open a port to allow endpoints in each call group to receive setup information and return results. Vivinet Assessor communicates with the endpoints using bi-directional TCP over port **10115**.

In Vivinet Assessor, click **Options > Assessment Options** and then click the **Firewall** tab.

Enter a **Reporting** port for the endpoints to use when reporting results back to Vivinet Assessor, or leave the default at TCP port **10116**. Choose **AUTO** to let the endpoints choose the port dynamically. Make sure the port you choose is open to bi-directional TCP traffic at the firewall.

Enter a **Call Traffic** port or range of ports to use if the firewall has been configured to pass VoIP calls through one particular port. The endpoints need to be able to send simulated VoIP traffic to each other through the firewall. Within any range of ports you set, the endpoints will use only even-numbered ports, which are what real VoIP RTP streams use. The default, AUTO, lets the endpoints choose the ports dynamically.

Your Call Traffic port must be opened at the firewall to bi-directional traffic. All simulated VoIP traffic that Vivinet Assessor sends over the network is bi-directional, to emulate the bi-directional nature of voice over IP.

If your firewall does not allow RTP traffic, open the port to allow UDP traffic.

After setting or changing the reporting port, you must stop and restart the Vivinet Assessor Scheduler service.

Scheduling the VoIP Quality Assessment

Create a schedule to determine when a VoIP Quality assessment will run. Scheduling options, which are also available for the Network Inventory and Utilization assessment, include the following:

- The length of time the assessment will run
- The duration of each set of calls
- The interval between each set of calls

The minimum allowed interval between calls is 15 minutes. Use the minimum interval to emulate heavier VoIP traffic. Or increase the interval to measure the call quality when VoIP traffic is lighter.

The endpoints need some time to report the results of the simulated calls. Therefore, the minimum length of time between the duration and the interval you set is five minutes. For example, if you set a call duration of 20 minutes and a calling interval of 22 minutes, you will be prompted to increase the interval or decrease the duration of the calls. In this case, you could simply increase the interval to 25 minutes.

After you have made your selections, validate the schedule to apply your settings to the current assessment and commit them to the database.

Scheduling Tips

Seven days is a good length for an assessment, enough to get plenty of data from days of the week when network traffic conditions are different. However, for your first assessment, we suggest that you run for a few hours and then scan your results. If it appears that the assessment is providing meaningful results and including enough representative endpoints to cover the network, reset the schedule for a longer time period. Likewise, start with fewer calls and add more after a successful test assessment. You can easily edit VoIP connectors in the Design view to add more calls.

The schedule interval and the number of concurrent calls determine how heavy the simulated traffic will be. To obtain more data points, choose a shorter interval. This helps increase the chance that you identify a drop in quality due to regular spikes in data traffic. For instance, if the data traffic always spikes on the hour, but you never run VoIP test calls at that moment, you may lose the fact that VoIP quality may drop then. You should have some idea of what peak and average utilization on the network will be. If not, do some traffic analyses to find out.

Verifying and Running the VoIP Quality Assessment

You must verify your assessment before you run it. During verification, Vivinet Assessor contacts each selected endpoint computer and sends it a brief schedule and one call script. Verification is like a mini-assessment. It provides some preliminary results and lets you see right away whether you need to change the assessment design. Any errors that occurred are indicated in the Verify view by a warning icon that appears in the Call Groups table. Look in the Error Log to see what happened and which endpoints were affected.

- If Performance Endpoint software was not detected on one of the computers in the assessment, install the software.
- Return to the Design view and change the assessment configuration to avoid repeating the error.

Use the Run view to activate the assessment and check on its progress. The status is continually updated, and, if an error occurs, the Error Log shows what happened.

Your presence is not required while the assessment is running. However, you should plan to check periodically to make sure no errors have occurred that might interfere with results.

Reviewing Vivinet Assessor Reports

You can generate a report as soon as an assessment has run long enough to produce some results. Vivinet Assessor offers two VoIP Readiness Assessment reports:

- Executive Summary Report—provides a summary, with relevant charts and graphs, of the results that most directly indicated VoIP readiness or that affected the overall quality of VoIP calls on the network. This report is usually about 10 pages long.
- Complete Report—provides information, charts, and graphs about every type of data collected during the assessment. Complete reports take varying amounts of time to generate, depending on the number of devices you discovered and monitored, and the number of call groups in the VoIP Quality assessment. For larger assessments, you may want to let Vivinet Assessor generate the report while you are away from the console.

Before you generate either report, you can perform a final calculation, **Potential Lost Revenue**, and include it in the report. Using the latest data from TeleChoice, Inc.⁵, Vivinet Assessor projects revenue loss due to poor-quality VoIP calls or service outages. The VoIP Quality assessment derives an accurate estimation of the percentage of simulated VoIP calls that would be hard to comprehend (“Poor” quality calls in the report) or impossible to connect (“Unavailable” in the report). Based on this estimate, Vivinet Assessor uses information about the industry of the network being assessed and the number of telephone users to calculate the revenue loss that that would occur if no network upgrades are made before VoIP rollout.

When you generate a report, Vivinet Assessor invokes Microsoft Word and Excel to interact with its database and produce charts, graphs, and detailed explanations of the data in a formatted report. Filtering options let you remove the sections of data that do not interest you, such as data collected on the weekend or details about router configuration. You can also easily edit the reports manually by using Microsoft Word.

Using Analysis Console with Results

One way to create custom graphs to include in the reports is to use Vivinet Assessor Analysis Console. Analysis Console lets you get a closer look at data from individual call groups or even individual calls.

Invoke Analysis Console by clicking the **Analyze and Chart Results** button in the Run view of the VoIP Quality assessment. The button is enabled after you have designed and saved a VoIP Quality assessment.

The charts you create with Analysis Console can be saved, or copied to the Windows Clipboard and pasted into your VoIP Readiness Assessment report or into other Microsoft applications such as Word or PowerPoint.

Analysis Console helps you do more with your VoIP Quality assessment results. With Analysis Console, you can:

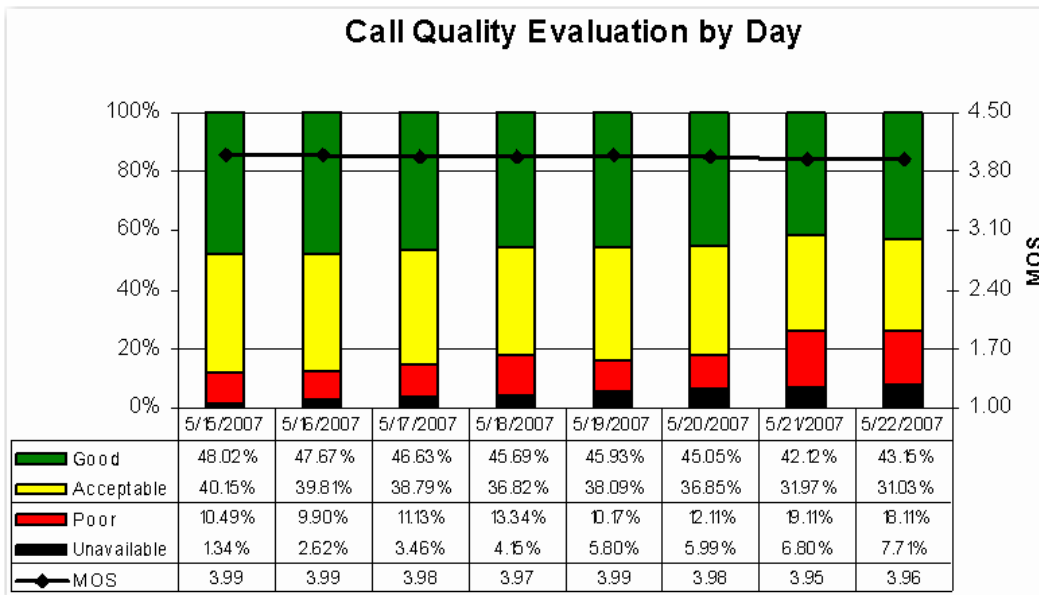
- View results sorted by call group or by call time, if you collected timing records during the assessment.
- Orient your view of the data using real-time rotation, zooming, and panning.
- Re-order the data streams and apply colors within a chart to improve readability.
- Mix and match data and chart styles within a single chart.
- Detect when results breach a high or low threshold by creating static threshold indicators.
- Re-use chart data in other applications.

Look at the sample assessment that ships with Vivinet Assessor to get acquainted with the kind of data you can get from your VoIP Readiness Assessment, as well as the data available through the Analysis Console.

Reviewing Call Quality Charts and Tables

Call quality is measured in units, just like delay, jitter, and lost data. Call quality units are points on the five-point MOS scale. Therefore, in bar charts showing call quality, the bars show the average MOS of all calls made between the endpoints in a call group. In addition, the bars are broken down into call-quality mappings to show the percentage of all calls that were rated Good, Acceptable, and Poor, or were Unavailable.

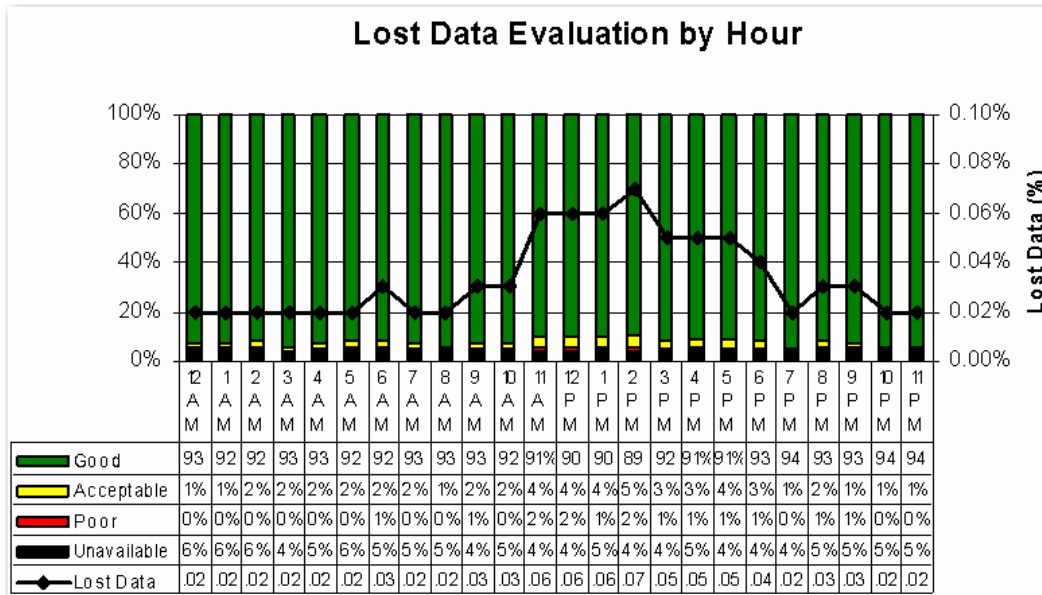
On charts that break out specific impairment factors by day or by hour, a line graph of that factor's values is superimposed over a bar graph. For example, the line graph of Call Quality Evaluation by Day gives you a quick overview of the call quality results averaged for each day of the assessment. The bars show which percentage of the MOS values fell into the Good, Acceptable, and Poor ranges.



To take a closer look at the actual data, double-click any chart to launch Microsoft Excel. You then can view a spreadsheet containing your data.

Bar charts are paired with data tables so that you can quickly scan the results and see specific values. Different colors in the bar charts provide an overview of the data; tables provide the specifics.

Most values shown are based on averages. When you look at charts that break out call quality by time of day, you see MOS averages for all simulated calls that were made during a certain hour of the day over the course of the entire assessment.



Summary

With the right tools, the process of successfully deploying VoIP on a data network can become a straightforward decision tree.

Run a VoIP Readiness Assessment. Scrutinize the results of the Utilization assessment. Are most of your critical routers ready to hand the extra traffic you will generate from a VoIP deployment? Are they sophisticated enough to support the latest types of QoS prioritizing? Do switches need to be upgraded to handle more capacity and cause less delay?

Look carefully at call quality, from a single call to the maximum number of expected calls at peak network usage. Compare calls across a range of locations.

If call quality is unacceptable, determine what the problems are and where they are located. What factor influenced the poor quality the most: one-way delay, jitter, lost data, or a combination of all three? Can a simple change in the VoIP configuration options, such as the choice of codec, sufficiently improve call quality? Where are the most likely bottlenecks?

Now, look at the costs of making the required network improvements. Choices include adding more bandwidth, upgrading or replacing existing equipment, changing the network design for greater efficiency, reconfiguring or tuning the network for QoS, or a combination of these.

These choices are only the start of a decision tree for a network administrator, because the costs of these choices are not equal. Adding more bandwidth may be a recurring expense if you lease lines; upgrading the hardware may be a capital expense; and QoS may appear to be free, but it usually has a high cost in personnel time.

Analyze the costs in as much depth as you can and decide whether you want to proceed with network changes. It is an iterative process of making the most cost-effective improvements a step at a time, then repeating the VoIP Readiness Assessment to see whether you have reached your call-quality goals.

If your cost estimates for preparing the data network for VoIP appear too high, then this is a good time to take another look at your VoIP deployment plan. By this point, you will better understand what the deployment requires, so you will have some choices:

- You can decide how to budget costs intelligently at the right time in the future,
- You can increase your current budget and proceed — considering this a suitable long-term investment, or
- You can approach VoIP as a staged deployment, taking some steps now and saving some steps for later.

¹ *Research Note*, Gartner, March 27, 2001.

² ITU-T Recommendation G.107, “The E-model, a computational model for use in transmission planning,” www.itu.int/publications/main_publ/itut.html.

³ Differentiated Services (DS) field, RFC 2474, www.ietf.org/rfc/rfc2474.txt.

⁴ *Cisco IP Telephony QoS Design Guide*, Cisco Systems, www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/avvidqos/avvid.pdf.

⁵ The figures for average revenue loss for each industry were calculated by TeleChoice, Inc., a leading telecom consulting and analyst firm. The figures were derived based on specific and factual industry information provided by select companies in each segment as well as from government-compiled information. Information was collected and analyzed in 2002, although much of the underlying data was from 2000 and 2001.