555-8421-103

Remote Gateway 9100 Series Network Engineering Guidelines

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About this document

The *Remote Gateway 9100 Series Network Engineering Guidelines* (NTP 555-8421-103) is written for individuals who are responsible for the installation, configuration, and day-to-day management of the Remote Gateway 9100 Series units and the Reach Line Card (RLC).

This document describes the network engineering guidelines for the Remote Gateway 9150 unit, Remote Gateway 911x series unit (9110 and 9115), Digital Telephone IP Adapter unit (internal and external), and the RLC.

This document provides:

- a clearer understanding of how you should plan your network
- a detailed description of Nortel's patented QoS Transitioning Technology
- instructions on how to order ISDN lines

Terminology

Throughout this document, the term “host PBX” refers to the following Nortel PBXs:

- Meridian 1 PBX
- Communication Server 1000 (CS 1000)
- Communication Server 2100 (CS 2100)

Unless otherwise specified, the term “Remote Gateway 9100 Series units” refers to the following products:

- Remote Gateway 9110 unit
- Remote Gateway 9115 unit
- Remote Gateway 9150 unit
- Digital Telephone IP Adapter unit (Internal and External)

The product name “Remote Gateway 9100 Series” refers to all of the “Remote Gateway 9100 Series units” as well as the RLC.

Introduction to packetized voice

The Public Switched Telephone Network (PSTN) has evolved into a stable, high quality communications medium. When you take a telephone off-hook, dial tone is immediate. Connection time is quite fast and clarity of the audio is excellent. With the development of packetized communications such as Voice over Internet Protocol (VoIP), the same high quality experience can be realized - when engineered properly.

To correctly engineer a VoIP system, you must use subjective testing results performed by numerous companies and independent labs to understand the factors that affect the quality of the VoIP. The main factors are categorized as follows:

- audio levels

  The PSTN has been built around clear rules of Loss Level Planning – the level of transmitted and received audio. The audio levels are such that the conversation is comfortable and highly intelligible. You can converse without having to strain to hear the caller.

- extra audio (echo, clicks, pops)

  Extra audio can be created in many ways. The most common extra audio is echo, or a reflection of your own speech. In traditional PSTN telephony, echo is present, but the Loss Level Planning reduces the amplitude, and the small amount of echo delay makes it unperceivable. With VoIP systems, the inherent audio processing and network delays can make the echo perceivable, if the echo is not cancelled/managed properly.

  Clicks and pops can be heard when packet loss is occurring on the network. The severity of the click or pop is determined by the size and duration of the packet loss experienced, as well as the audio compression and decompression scheme (Codec) chosen. A Codec compresses audio, reducing the amount of bandwidth required to have a conversation.

  — G.711 (64 Kbps data rate - equivalent to a wired telephone [no compression])

  — G.726 (32 Kbps data rate - reduces the required bandwidth by one half in trade for a slightly lower audio fidelity)

  — G.729A (8 Kbps data rate - high level of compression in trade for lower audio fidelity)

- missing audio (drop-outs, reduced fidelity)

  Missing audio is often a result of packet loss on the network. Reduced audio fidelity is a subjective factor that varies greatly by system user. The G.711 and G.726 Codecs provides wire line quality when the network conditions are ideal. Packet loss with high compression Codecs such as G.729A lead to an even lower audio fidelity.

- conversational delay

  When speaking to a person face to face, your speech audio is transmitted and received instantaneously. You also have the added benefit of being able to see the other person, and therefore can use body language, expressions and gestures to assist in the communication. With telephony, it is the tone, phrasing and timing of speech that is transmitted. VoIP systems can affect the fidelity of the audio but the greatest impact can be in the area of conversational delay. For example, assuming that you are speaking, your audio must be compressed, packetized, transmitted across a network, decompressed and transmitted to the final destination. These processes all take a small measurable amount of time. If the delay is too great, the conversation can be awkward, with each talker speaking over one another.

- trans/dual encoding

  Transcoding is a term used when audio is processed through one Codec, decompressed then passed through another Codec. Each time the audio is compressed (G.726 or G.729A), the audio fidelity reduces.

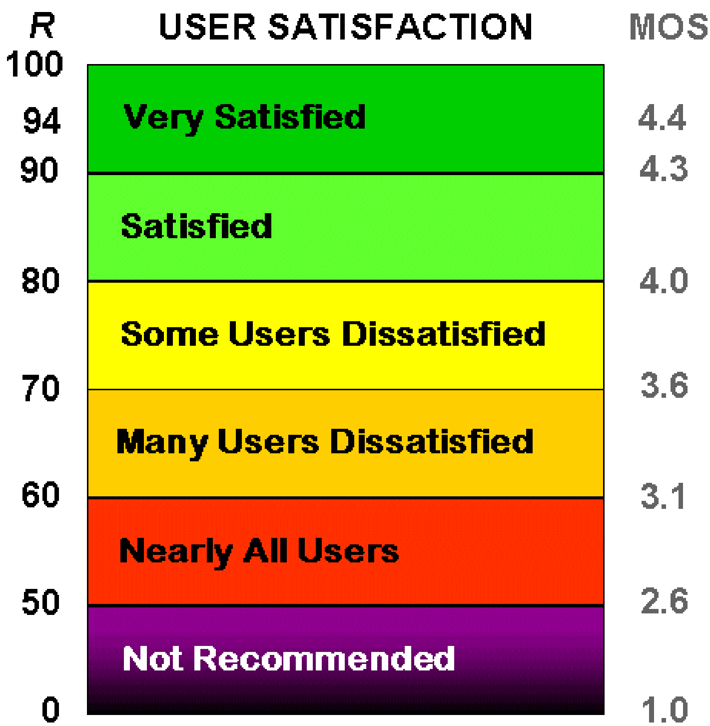
With these factors in mind and the industry subjective testing results as a reference, you can engineer a high quality VoIP system. One of the best references available to “rate” the quality and factors that impair VoIP systems is the International Telephony Union G.107 Recommendations (ITU-G.107). This document and the subsequent G.108 and G.113 documents have been developed by industry leaders, including Nortel.

Many of the ITU recommendations have been “rolled” into the TIA TSB116 document that clearly outlines the methods of engineering an excellent packetized voice system.

The E-model to measure audio quality and user satisfaction

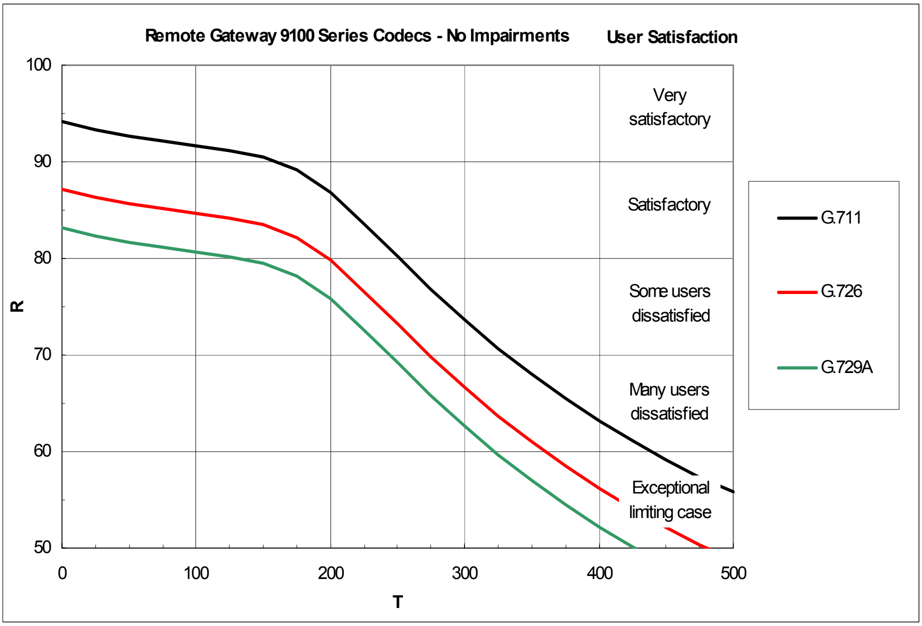
The ITU developed what is called the E-model. The E-model uses the factors described in “Introduction to packetized voice” on page 5 to derive a quality value corresponding to the quality experienced by the VoIP system users.

The following graph shows the E-model rating, R value, versus the traditional Mean Opinion Score (MOS) values.



Compressed audio quality under ideal conditions

The following graph shows the Remote Gateway 9100 Series Codecs graphed against the R value and conversational delay.



Note: T is one-way delay in milliseconds (ms).

To fully use this graph, you must know the Remote Gateway 9100 Series inherent delay values. The critical values are as follows:

Codec	Encoding/ Decoding	BRI Serialization	911x V.32 Serialization
G.711 - 64 Kbps	30 ms	25 ms	-
G.726 - 32 Kbps	30 ms	12 ms	-
G.729A - 8 Kbps	35 ms	10 ms	90 ms

There are additional delays experienced that are a function of the customer network:

- end-to-end delay

  The time it takes to pass an Internet Protocol (IP) packet from the RLC to the Remote Gateway 9150 unit (or the opposite direction)
- jitter buffer depth

  If the network has low jitter, the jitter buffer can be reduced. The configurable values in the Remote Gateway 9100 Series product are 30, 60, or 90 ms.

Conversational delay calculation examples

Remote Gateway 9100 Series units over IP

When using the Remote Gateway 9100 Series units over IP, the controllable delay factors are the jitter buffer and the network delay. For example, assuming you are using G.711 with a 60 ms jitter buffer:

30 ms (G.711 encoding/decoding) + 20 ms (one-way network delay) + 60 ms (jitter buffer) = 110 ms one-way delay

Remote Gateway 9150 unit over PSTN

In the following example, the RLC and Remote Gateway 9150 unit is configured to use G.729A to reduce bandwidth requirements:

35 ms (G.729A encoding/decoding) + 10 ms (serialization delay) + 60 ms (jitter buffer) = 105 ms one-way delay

Remote Gateway 911x series unit over PSTN

The Remote Gateway 911x series unit uses a V.32 modem emulation to provide connectivity over the PSTN. In this configuration, you must use G.729A.

35 ms (G.729A encoding/decoding) + 90 ms (serialization delay) + 60 ms (jitter buffer) = 185 ms one-way delay

Remote Gateway 9150 unit to Remote Gateway 9150 unit

When a customer environment uses multiple Remote Gateway 9150 units, the audio must be treated by each individual Remote Gateway 9150 unit. Therefore, the delay each Remote Gateway 9150 unit experiences is added together. For example, assuming one Remote Gateway 9150 unit over IP (G.711) communicating to another Remote Gateway 9150 unit over BRI (G.729A):

30 ms (G.711 encoding/decoding) + 20 ms (one-way network delay) + 60 ms (jitter buffer) + 35 ms (G.729A encoding/decoding) + 10 ms (serialization delay) + 60 ms (jitter buffer) = 215 ms one-way delay

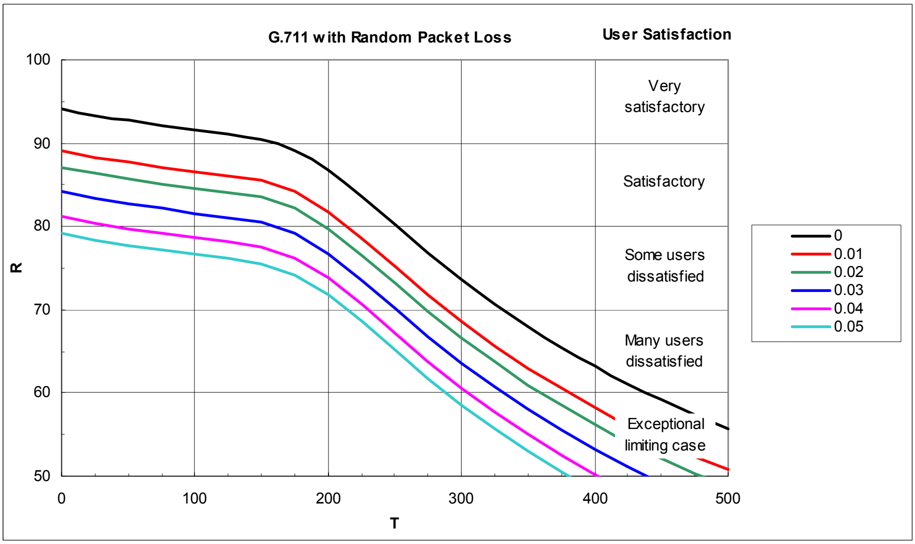
Notes:

1. Since the audio has been compressed by the G.729A Codec, each user experiences the quality provided by the G.729A Codec.
2. When calls are made from Remote Gateway 9150 unit to Remote Gateway 9150 unit as described, it is possible that the Echo Cancellers in the Remote Gateway 9100 Series product can cause periods of audio clipping when the conversational delay is large. To minimize this possibility, you must manage network delay and jitter so that the jitter buffer depth can be reduced as much as possible.

As you have calculated, when using IP connectivity, the network performance can affect the jitter buffer depth and network delay, thus increasing the conversational delay. The network can also have periods of high jitter or packet loss that can greatly affect the audio quality.

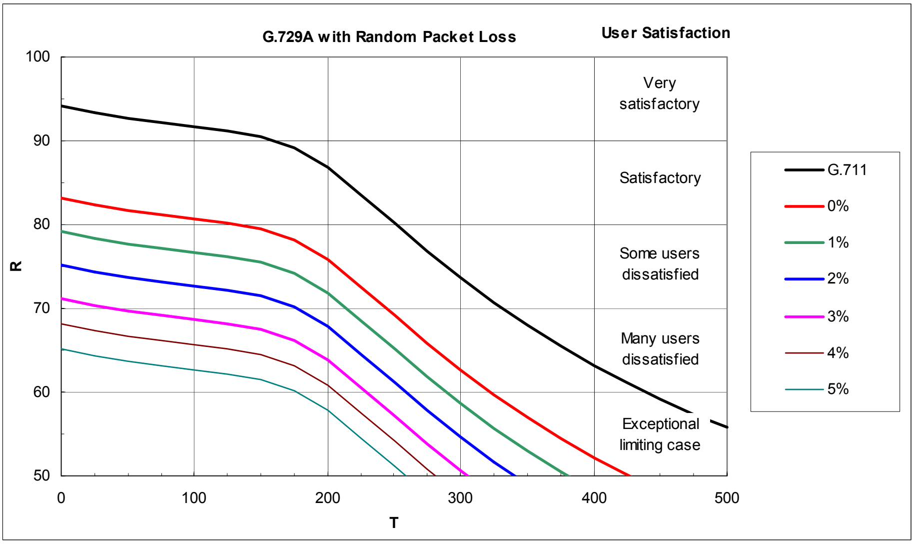
Compressed audio quality under packet loss conditions

The following graph shows the effect of random packet loss on the Remote Gateway 9100 Series G.711 Codecs.



Note: T is one-way delay in ms.

The following graph shows the effect of random packet loss on the Remote Gateway 9100 Series G.729A Codecs.



Note: T is one-way delay in ms.

Evaluating your network

Now that you fully understand the effects of packet loss, jitter, and delay on the audio quality that can be experienced on packetized voice equipment, the next step is to ensure that the network is capable and configured to support packetized voice.

A network is a dynamic entity – the characteristics change daily, hourly, by the second as various applications push and pull data across it. The method of evaluation best suited for packetized voice is testing with pseudo voice traffic for extended periods of time.

There are numerous Personal Computer (PC) based software tools available that generate a configured number of calls, using a specific Codec, for a configured duration. The tool then provides various reports showing the jitter, packet loss, and the equivalent audio quality (MOS or R-value). You can and should use this tool before installing the VoIP equipment. You should run tests for an extended period of time.

To ensure success, network evaluation is a critical step in engineering a VoIP solution. Do not skip this critical step, as it is the foundation for the projects success.

Nortel can provide network evaluation services, either directly or through our business partners.

Network engineering for real time traffic

Packetized voice, or VoIP, is a form of real time traffic where delays and retransmission cannot be tolerated. Typical PC and server traffic is non-real time and, therefore, can tolerate a certain level of delay and retransmission. To ensure that the VoIP traffic receives the most expedited treatment possible, there are a number of techniques that you can use.

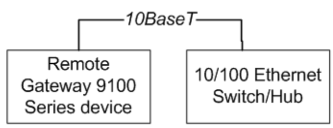
Let's look at the various techniques starting at the VoIP devices themselves and working out into the Local Area Network (LAN) or Wide Area Network (WAN).

LAN access points

The RLC and Remote Gateway 9100 Series units all have Ethernet interfaces that you can use for VoIP connectivity. Use the following table to locate connectivity options at the device level:

Product	Capability	Minimum Revisions
RLC	10 M half or full-duplex	1.3.5 firmware for full-duplex
Remote Gateway 9150 unit	10 M half or full-duplex	1.4.0 hardware and firmware for full-duplex
Remote Gateway 911x series units and Digital Telephone IP Adapter units	10 M half-duplex	1.3.1 firmware

It is common that the Remote Gateway 9100 Series product is directly plugged into a hub or a switch as shown in the following illustration:

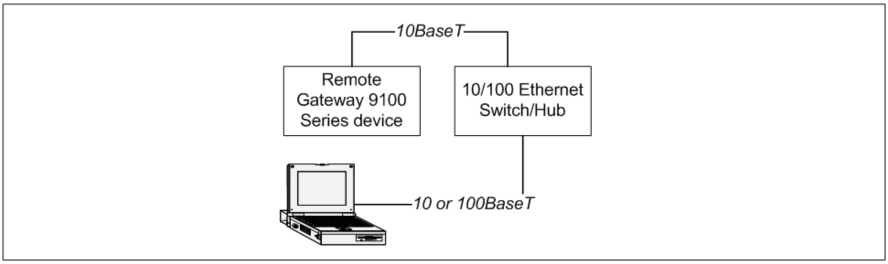


Consideration 1

When using a hub, the shared media causes all other hub traffic to be presented to the Remote Gateway 9100 Series device also. The Remote Gateway 9100 Series product reports this traffic as multicast packets – packets that were not specifically addressed for the Remote Gateway 9100 Series product itself. Since the Remote Gateway 9100 Series product must read the packet to determine that it should be ignored, a high level of multicast packets can be disruptive to real time traffic. Keep the multicast traffic to less than five percent of the segment traffic.

Recommendation 1

A switch manages the traffic so only packets addressed for the Remote Gateway 9100 Series product are transmitted to the Remote Gateway 9100 Series product. Nortel recommends that you connect the Remote Gateway 9100 Series product to a data switch as shown in the following illustration:

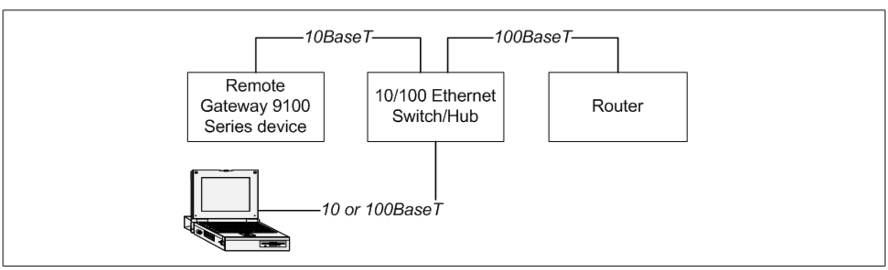


Consideration 2

When the VoIP traffic is passed to the switch, the switch forwards the packet at essentially wire line speed. When multiple devices receive multiple packets, with the destination being on the uplink side of the switch, the switch must temporarily buffer the packets as it manages the uplink media. Low end switches do not provide a mechanism to manage the buffer process. Therefore, real time traffic can experience a small amount of jitter due to buffering.

Recommendation 2

The Remote Gateway 9100 Series products support a prioritization methodology called 802.1Q tagging. When the feature is enabled in the Remote Gateway 9100 Series products, the Ethernet Media Access Control (MAC) frame is “tagged” with three bits that define the priority of the packet. When using a data switch that is 802.1Q aware, the tagged data receives the appropriate priority over low priority or untagged packets. The Remote Gateway 9100 Series products set the 802.1Q code point to a hex value of C000. The following illustration shows aggregation point uplink prioritization:



Consideration 3

The uplink for the switch or hub is connected to either another switch or a router. If it is an additional switch, recommendation 2 still applies, as well as using 802.1Q Virtual Local Area Network (VLAN) techniques to keep the real time traffic prioritized at the Layer 2 level.

If you use a router as the uplink device, there are a number or prioritization methods that can be used. Remember that even though prioritizing the traffic in all directions is beneficial, the critical direction is when data is aggregated from high speed links to a slow speed link.

There are three critical steps in managing real time traffic through a router, as follows:

1. Recognizing that the traffic requires prioritization
2. Placing the traffic in the appropriate queues
3. Passing the traffic onto the next segment in a prioritized manner

Recommendation 3

The Remote Gateway 9100 Series products have the capability of marking IP packets with a Differentiated Services (DiffServ) CodePoint. A router that is aware of the DiffServ prioritization levels places the Remote Gateway 9100 Series traffic in the highest priority queue.

If DiffServ cannot be used to identify the real time traffic, the source, destination, or port in the IP packet can be used to identify the Remote Gateway 9100 Series traffic. UDP port 0x5000 (20480) is used for voice traffic. UPD port 0x5002 (20482) is used to test during QoS Transition. TCP port 0x3200 (12800) is used for the Remote Gateway 9100 Series product and telephone signaling.

Note: All packets are tagged with the DiffServ CodePoint you configure in Remote Gateway 9100 Series Configuration Manager on the IP Configuration property sheet for your specific unit. Refer to your Remote Gateway 9100 Series unit’s Installation and Administration Guide for further details.

There are multiple methods of creating queues in a router, each with different depths, prioritizations, and so on. The queue that the voice traffic is placed in must be deep enough to ensure that packets are not lost due to the queue being full, but not too deep to cause excessive packet delay. Most router venders follow the well known standards. Configuration of these techniques, however, could vary greatly. Consult your equipment vendor for the exact configurations. Routers can be provisioned to honor 802.1Q markings, and also honor or mark the packets DiffServ setting at layer 3.

Understanding queuing techniques

Slower WAN links can add inherent delay to voice packet transversal. Often, these types of links have voice and data converged, competing for the same bandwidth availability. How this converged link affects voice quality is completely dependant on the WAN link speed and encapsulation protocol. Queuing techniques help to provide the most efficient method for guaranteeing VoIP delivery while sharing the bandwidth with other data traffic.

Recommendation 4

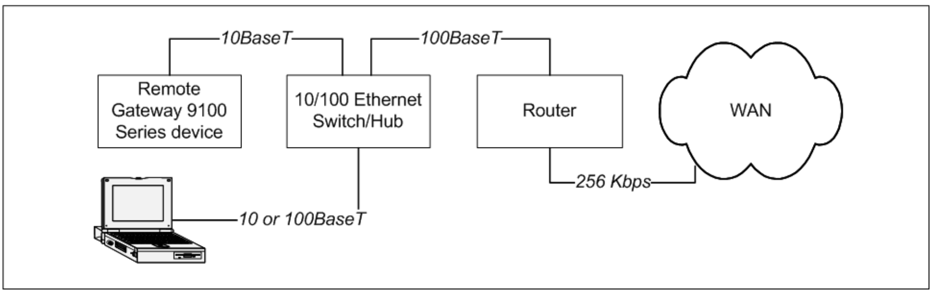
For WAN serial links 768 Kbps or less, Nortel recommends that you use the following packet fragmentation and queuing techniques:

Recommendation	Definition
packet fragmentation	Fragmentation helps prevent a smaller VoIP packet from being delayed behind a large data packet. Fragmentation must be enabled on both ends of the serial link. Depending on link speed, serialization delay of a large, 1500 byte packet can cause a smaller, 160 byte voice packet to exceed the end-to-end budget delay of 150 ms. Fragmentation breaks up the train of packets into evenly sized packets. It is important not to fragment the VoIP packets, only the competing data packets. Method of fragmentation can vary, depending on the WAN protocol. Frame Relay, as an example, can make use of the FRF.12 standard and fragment at layer 2. Point-to-Point Protocol (PPP) also has the ability to fragment at layer 2. High-Level Data Link Control (HDLC) links, however, can only be fragmented at the IP layer, layer 3 by setting a maximum Maximum Transmission Unit (MTU) size. Nortel does not recommend this method due to the added processing requirements, and unreliable results. **Note:** Some Web applications can mark the packet as "do not fragment".
interleaving	Interleaving is a technique to mix the VoIP packets into a stream of data traffic. As an example, a large file download can have multiple packets required to complete the session. Fragmentation breaks up the large packets into regular sized packets, while interleaving inter-weaves VoIP packets with the data packets.

queuing techniques	Queuing techniques help to isolate priority traffic, such as VoIP, from the normal data traffic. By placing the priority voice traffic into a separate queue, or holding bucket, this type of traffic can be given special treatment. As an example, the priority traffic can be given dedicated bandwidth when the link is congested. This helps ensure a timely delivery of the VoIP packets. The different queuing methods are as follows:
Priority Queuing (PQ)	This is the legacy method of queuing to guarantee a certain amount of available bandwidth. This method has some inherent problems. As an example, PQ can be very Central Processing Unit (CPU) intensive, and starve other traffic types of bandwidth. As long as the PQ queue is full, it does not service the other queues. Today, PQ is incorporated into more advance queuing methods such as CBWFQ and LLQ. Refer to page 20 for further details on CBWFQ and LLQ.
Customer Queuing (CQ)	Developed to solve the PQ problems of bandwidth starvation for the non-priority traffic. This method uses a “round-robin” method to service the difference queues.
Weighted Fair Queuing (WFQ)	This is the most common default method of queuing for interfaces 2 Mbps or less. WFQ attempts to share the available bandwidth equally for all applications by identifying the different conversations. The “weighted” part of WFQ takes note of the DiffServ CodePoint at layer 3 to give priority treatment. Today, this method is combined with the other queuing techniques (PQ and CQ) to provide more queuing options.

Class Based Weighted Fair Queuing (CBWFQ)	This technique makes use of the WFQ method of queuing and adds the ability to separate traffic into different classes. The classes allow for more flexibility for controlling traffic on a network. It does not use the "guaranteed bandwidth" PQ method, but does allow you to set aside a particular amount of bandwidth for VoIP traffic. It also allows you to configure different queuing methods for other types of traffic, such as Web traffic.
Low Latency Queuing (LLQ)	Known as PQ/CBWFQ, this technique adds the PQ to CBWFQ by allowing you to guarantee a certain amount of bandwidth for priority traffic, such as VoIP.

The following illustration shows high speed to low speed aggregation:



Consideration 5

The real time traffic is now in the appropriate queue waiting to be passed onto the next segment. The timeliness of the delivery is important. Therefore, you must review multiple considerations.

Recommendation 5

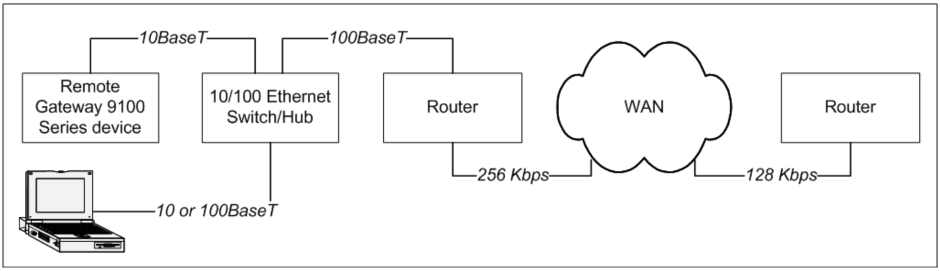
Maximum Transmission Unit (MTU) is the maximum packet size that is placed on the segment. Since the link speed is low, a large packet takes a larger amount of time to transmit across the link. The following table shows the fragmentation size versus the link speed, which is called the transmission time, or serialization delay (shown in ms).

Link Speed (bps)	Fragmentation Size (Bytes)					
	1500	1024	768	512	256	128
1536000	7.8	5.3	4.0	2.7	1.3	0.7
1024000	11.7	8.0	6.0	4.0	2.0	1.0
768000	15.6	10.7	8.0	5.3	2.7	1.3
512000	23.4	16.0	12.0	8.0	4.0	2.0
384000	31.3	21.3	16.0	10.7	5.3	2.7
256000	46.9	32.0	24.0	16.0	8.0	4.0
128000	93.8	64.0	48.0	32.0	16.0	8.0
64000	187.5	128.0	96.0	64.0	32.0	16.0
56000	214.3	146.3	109.7	73.1	36.6	18.3

Fragmentation = (link speed * serialization delay)/8

The ideal serialization delay is 8 ms or less. For example, on a 256 Kbps link, the MTU should be 256 bytes.

Note: A small MTU (less than 768 bytes) can cause performance degradation on certain PC applications. Therefore, Nortel recommends that you use fragmentation techniques such as FRF.12 or PPP fragmentation/interleaving, rather than physically changing the MTU size. The following illustration shows WAN considerations:



Consideration 6

Many Remote Gateway 9100 Series installations utilize a WAN provided by a third party service provider. The customer leases the required service, and depends on the WAN provider to maintain the required level of service. The parameters that are often used to define the WAN service are as follows:

Parameter	Definition
Frame Relay Service	
Minimum Committed Information Rate (MinCIR)	The minimum amount of data to be sent during periods of congestion. Defaults to half CIR in most circumstances.
Burst Committed (Bc)	The number of bits to transmit in the specified amount of time interval (Transmission Control [Tc]).
Burst Excess (Be)	The number of bits to transmit in the first interval of active transmission, once credit has been built up.
Transmission Shaping Interval (Tc)	The period of transmission for Frame Relay Permanent Virtual Connection (PVC).

Traffic Shaping	This is a method of throttling transmission from a fast to slow link by adjusting the credit manger or token bucket.
Adaptive Shaping	A method to throttle transmission using the Backward Excessive Congestion Notification (BECN) technique inherent to Frame Relay when CIR is exceeded.
Asynchronous Transfer Mode (ATM) Service	
Sustainable Cell Rate (SCR)	The averaged cell rate over a period of time.
Peak Cell Rate (PCR)	The maximum cell rate, normally equals port speed.
Maximum Burst Size (MBS)	The amount of bandwidth that can be utilized if a burst occurs.
Real Time Variable Bit Rate (rt-VBR)	A level of ATM service that guarantees the cell loss and end-to-end delay characteristics.
Non-real Time Variable Bit Rate (nrt-VBR)	This is similar to rt-VBR, but the amount of delay is not specified. Therefore, jitter can be a problem. Nortel does not recommend this for real time traffic such as voice.
Available Bit Rate (ABR)	A guaranteed minimum amount of bandwidth is defined, but the delay and loss parameters are not specified when a defined peak is reached. Nortel does not recommend this for real time traffic unless you use strict flow control mechanisms.
Unspecified Bit Rate (UBR)	Designed for bursty, non-real time applications, this service does not specify delay or loss parameters.

Guaranteed Frame Rate (GFR)	A guaranteed minimum amount of bandwidth is defined, but the delay and loss parameters are not specified when the minimum is surpassed. Nortel does not recommend this for real time traffic.
Cell Loss Priority (CLP)	This is a bit that determines if a cell should be dropped due to excessive congestion.

Recommendation 6

Test the service provided by the WAN service provider. It is quite likely that real time data is being passed along side of non-real time data from other customers. All of the good prioritization work performed on the customer network could be undone by a poor WAN. For example, non-real time traffic should have the Discard Eligible (DE) bit set, so that during periods of congestion, it is dropped before real time traffic.

For Frame Relay VoIP, Nortel recommends the following:

- Do not exceed the CIR of the PVC:
  - Queuing of voice packets must be minimized.
  - If the CIR is exceeded, packets are normally queued in the Frame Relay cloud.
  - Maintain a value at or below CIR.
- Do not use Frame Relay Adaptive Shaping:
  - Adaptive Shaping utilizes BECNs to shape or throttle down traffic transmission if the CIR is exceeded.
- Set the value of Burst Committed (Bc) low so the Transmission Shaping Interval (Tc) is small (Tc = Bc/CIR):
  - (Tc) value should be around 10 ms.
  - A small value ensures that large packets do not use up all the available credits.
  - A large (Tc) value can lead to large gaps between packets sent. This is due to the traffic shaper waiting an entire (Tc) period before sending the next frame.

For ATM to Frame Relay links, Nortel recommends the following:

- Be sure to properly map Frame Relay parameters (CIR, Bc, and Be) into equivalent ATM values.
  - — Use FRF.8 (RFC 1490), Section 5.1. FRF.8 details the method of interworking between a Frame Relay end-point and an ATM end-point
- Inherent differences exist, such as PCR and SCR are expressed in cells per second (cps), ABR and CIR are expressed in bytes per second (bps), and ATM uses a fixed cell size of 53 bytes.
  - — To deliver an equivalent bandwidth to actual user traffic, choose the values of PCR and SCR to include the extra margin required to accommodate the overhead introduced in transferring the Frame Relay frames through an ATM network.

Bandwidth usage table

The values shown in the following table are the bandwidth requirements when using IP as the medium.

Packet Size	G.711[1]	G.726[1]	G.729A[1]	G.729A\FAX[1,2]
Voice Payload (bytes/30 ms)	240	120	30	38
Voice header (bytes)	12	12	12	12
UDP header (bytes)	8	8	8	8
IP header (bytes)	20	20	20	20
IP Packet Size (bytes)	280	160	70	78
LAN Overhead				
Ethernet header Size (bytes)	14	14	14	14
IP Packet Size (bytes)	280	160	70	78
LAN Packet Size (bytes)[3]	294	174	84	92
Peak Data Rate (Kbps)	78	46	22	24
Avg Data Rate-w/silence (Kbps)[4]	47	28	13	24[5]
IP Over Frame Relay Overhead				
Frame Relay Overhead (bytes)	4	4	4	4
RFC 1490 IP Overhead (bytes)	2	2	2	2
IP Packet Size (bytes)	280	160	70	78
Frame size (bytes)	286	166	76	84
Peak Data Rate (Kbps)	76	44	20	22
Avg Data Rate-w/silence (Kbps)[4]	46	27	12	22[5]

1. compression algorithm
2. 9600 bps (14400 bps not supported)
3. The Remote Gateway 911x series and Digital Telephone IP Adapter units transmit packets that are based on 64 byte boundaries. In other words, G.729A equals 128 bytes and G.711 equals 320 bytes. The Remote Gateway 911x series and Digital Telephone IP Adapter units do not support G.726.
4. Average data rate with silence suppression is approximately 60% of Peak. Change this number to reflect the site's actual values.
5. Silence suppression, or voice activity detection (VAD) is disabled.

When using the Remote Gateway 9100 Series products with PSTN bandwidth, the requirements are as follows:

- G.711 = 64 Kbps
- G.726 = 32 Kbps
- G.729A = 8 Kbps

If VAD is disabled, the PSTN bandwidth requirements increase by 20% due to inefficiencies in the HDLC protocol used.

Quality of service issues

The following section discusses ways that the Remote Gateway 9100 Series product works to resolve Quality of Service (QoS) issues involved in transmitting VoIP.

QoS Transitioning Technology

When voice quality degrades on your Remote Gateway 9100 Series network, the RLC provides improved QoS. Nortel's patented QoS Transitioning Technology enables the RLC to move Remote Gateway 9100 Series calls from the IP network to the PSTN. The more stable PSTN provides improved QoS until IP network QoS improves.

Once the IP network QoS improves, the RLC automatically recovers Remote Gateway 9100 Series calls to the IP network. For detailed information on how to configure QoS Transitioning Technology thresholds, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

Note: Because Digital Telephone IP Adapter units do not have PSTN connectivity, they do not support QoS Transitioning Technology.

How QoS Transitioning Technology works

QoS Transitioning Technology bases transition and recovery functions on the IP network's QoS level. The QoS level is a user-oriented metric that takes one of ten settings. You identify the limits of acceptable voice QoS for each remote site in your Remote Gateway 9100 Series network by choosing from among those settings using Remote Gateway 9100 Series Configuration Manager graphical user interface (GUI). The Meridian digital telephone set at the remote location and the RLC in the host PBX communicate with one another over the IP network. These communications begin with a 10BaseT Ethernet interface to the corporate intranet.

The RLC and the remote units constantly monitor the QoS level on the data paths between the host PBX and the individual remote sites. Each node calculates the prevailing QoS level based upon the following factors:

- jitter
- delay
- packet loss
- compression algorithm

The RLC calculates the QoS level based upon ANSI TR56 and the E-model found in ITU-T Reg. G.114. When the QoS level falls below the specified transition threshold for the specified period of time, or duration, the receiving Remote Gateway 9100 Series node rejects the call. The RLC then reroutes the call over the PSTN. When the QoS rises to a point above the specified recovery threshold for the specified period, the RLC restores calls to the IP network. You configure thresholds and durations for each remote site.

When transitioning or recovering multiple calls, the RLC moves calls between the IP network connection and the PSTN connection based on the Kbps you configure. The system waits several seconds between the units of Kbps you configure to determine if the IP connection has become more stable.

Note: Configure the Kbps using the RLC's Remote Connection Configuration property sheet in Configuration Manager. Click on the Quality of Service button, then enter the Kbps in the Transition Bandwidth field on the Quality of Service dialog box.

The RLC moves as many calls as possible (to a maximum of 64 Kbps for each B-channel) from the IP connection to the PSTN trunk connection. High priority users always move first. For information on configuring priority, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

- transparent transition and recovery

  Transitions can take place during a live call and are transparent to users who only notice improved voice quality. In the case of a complete IP network failure, however, a noticeable break in the call of approximately four seconds occurs. This allows the RLC to transition the call to the PSTN.

  Note: When IP degradation causes the RLC to transition calls to the PSTN, a momentary degradation in voice quality can occur as the transition takes place.

- network testing

  Both the RLC and the Remote Gateway 9100 Series unit (Remote Gateway 9150 and 911x unit) run an IP test to determine if the QoS on the IP network meets configured standards. (For more details, refer to "Measuring QoS while offline" on page 32.) To configure QoS Transitioning Technology thresholds, use the Remote Gateway 9100 Series Configuration Manager. For further information, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

The chart on page 30 illustrates how QoS Transitioning Technology works on the RLC, Remote Gateway 911x series unit, and Remote Gateway 9150 units. The same chart depicts Z and Q as equal periods of time. However, the duration of poor voice quality required for transition to the PSTN, Z, is a period of seconds. The duration of good voice quality required for recovery to the IP network, Q, is a period of minutes. The units of time used to identify acceptable points of transition and recovery differ in an attempt to minimize *transition thrashing*. Transition thrashing is the rapid transition and recovery between the PSTN and the IP network. This can occur when QoS hovers around configured degrade and recover thresholds producing higher than normal PSTN charges.

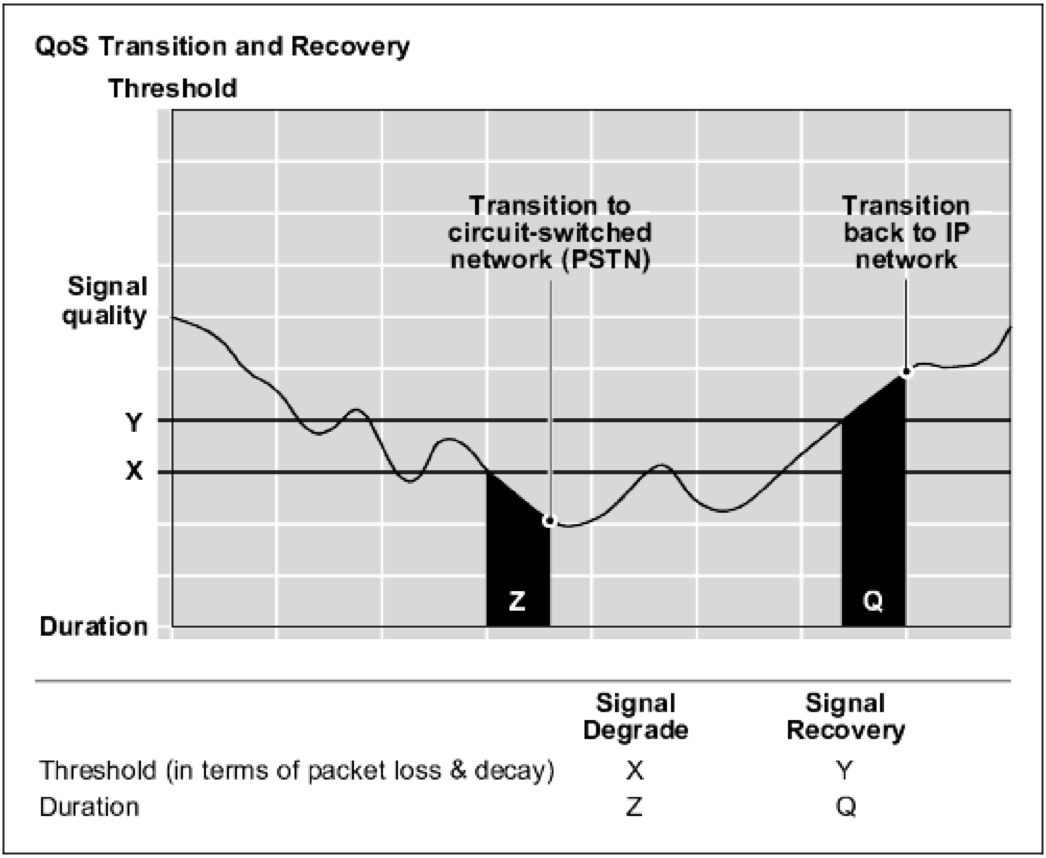
In the telephone call represented by the chart on page 30, signal quality begins in the acceptable range; that is, signal quality is above the X threshold. While signal quality remains in the acceptable range, the RLC routes calls through the IP network.

- transition

  When IP QoS degrades below the X threshold, the RLC waits for duration Z to increase the likelihood of continued unacceptable voice QoS. If the voice QoS remains below the X threshold for duration Z, the RLC establishes a communication path for that call on the PSTN. The RLC then transitions the call to the newly established PSTN channel.

- recovery

  After the RLC transitions a call to the PSTN, when IP QoS exceeds the Y threshold, the system waits for duration Q. Again, this increases the likelihood of continued acceptable voice QoS on the IP network. After duration Q passes, the RLC moves all current calls back to the IP network and places all new calls over the IP network.



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The setting	represents
X	signal quality on the IP network is unacceptable below this point.
Y	signal quality on the IP network is acceptable above this point.
Z	the amount of time that signal quality must be lower than the X threshold before the RLC transitions calls to the PSTN.
Q	the amount of time that signal quality must be higher than the Y threshold before the RLC recovers calls to the IP network.

QoS traffic measurements

As voice packets travel across the IP network, the RLC monitors the following QoS parameters:

- average packet delay

  The RLC calculates this delay using the following statistics gathered from its voice jitter attenuation buffer:

  — minimum packet holding time in the jitter buffer

  — maximum packet holding time in the jitter buffer

  — peak holding time in the jitter buffer

  — time-stamp values in the packet header

  By accumulating these statistics over time, the RLC can calculate an average packet delay value through the IP network. As the system detects an increase in the average packet delay, it references the signal degrade threshold to determine when to make the transition to the PSTN connection.

  For a further description of statistics, refer to the *Reach Line Card Installation and Administration Guide (NTP 555-8421-210)*.

- lost packets

  The RLC calculates lost packet statistics by accumulating the following packet header and voice decoder statistics:

  — voice decoder underrun

  — voice decoder overrun

  — out-of-sequence packet reception

Measuring QoS while offline

Once the RLC reverts to using its PSTN connections, it must continually monitor the IP network. The RLC determines an appropriate time to restore voice traffic to the IP network as follows:

1. Both the RLC and the Remote Gateway 9100 Series unit (Remote Gateway 9150 and 911x) place pseudo-voice traffic on the IP network.

   Both the RLC and the Remote Gateway 9100 Series unit send traffic with a maximum bandwidth of no more than 16 Kbps in short bursts at a higher bit rate to approximate live voice traffic.

2. Both the RLC and the Remote Gateway 9100 Series unit (Remote Gateway 9150 and 911x) gather statistics based on the pseudo voice traffic to determine the congestion levels on the network. They use packet time stamps and sequence numbers to monitor the following parameters:
   - average end-to-end delay
   - average round-trip delay
   - average packet-to-packet jitter
   - average packet loss
3. When the parameters listed in step 2 fall below the predetermined threshold for the predetermined period of time, the RLC restores voice traffic to the IP network.

   When restoring the connection to the IP network, the system adds hysteresis, or delay, to reduce the noise level during the transition.

   Note: Hysteresis, or delay, prevents thrashing between the PSTN and the IP network (for a discussion of *transition thrashing, refer* to page 30), and ensures that acceptable voice QoS exists on the IP network for a predefined period of time.

Log reports and statistics

The Remote Gateway 9100 Series Configuration Manager provides a statistics log that identifies the number of QoS transitions. For a detailed description of log and statistic reports, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

Telephone display messages

Meridian digital telephone sets with display capability can display the following Resource Limit messages:

- Bandwidth Limit indicates that there is insufficient BRI bandwidth available to complete the requested action.
- DSP Limit indicates that there are insufficient DSP resources available to complete the requested action.

Relationship between users and services

In the context of a host PBX and Remote Gateway 9150 unit, there are two interfaces that you must consider in the relationship between users and services:

- The Remote Gateway 9100 Series node (that is, the RLC on the host PBX and the Remote Gateway 9100 Series unit at the remote site) provides an interface to the Remote Gateway 9100 Series system for end users. Voice services offered by the Remote Gateway 9100 Series node must meet user-oriented QoS objectives. Refer to “Quality of service issues” on page 28 for details.
- The Remote Gateway 9100 Series nodes also provide an interface with the intranet (or Internet in the case of a Remote Gateway 911x series unit). The intranet provides the “best-effort delivery of IP packets,” as opposed to “guaranteed QoS for real-time voice transport.” The Remote Gateway 9100 Series node translates the QoS objectives set by the end-users into IP-oriented QoS objectives. This document refers to these objectives as *intranet QoS objectives*.

Ordering ISDN lines

The following section provides you with information regarding the ordering of ISDN lines.

What you need to tell your ISDN service provider

Inform your ISDN service provider that you need the following:

- two B-channels providing both voice and data capability (56 or 64 Kbps clear, depending on your system)

  Note: Both B-channels must be PSTN voice and data. Also, although Nortel requires 56 or 64 Kbps clear, 64 Kbps clear is preferred.

- Caller Line Identification (CLID, known in the United Kingdom as Calling Line Identity Presentation–CLIP)
- Multiple Subscriber Numbering (MSN)
- two directory numbers (DNs)
- two Service Profile Identifiers (SPIDs)

Note: MSN is required in situations where no SPIDs are provided.

Tell your service provider how the line should be provisioned for data, voice, and other optional services. Different service providers require this information in different ways; increasingly they are using ISDN Order Codes for simplicity, but some still require specific switch type details.

What you need from your ISDN service provider

Your service provider needs to tell you:

- the ISDN service and switch type
- the ISDN directory numbers
- associated SPIDs
- bearer capability (56 or 64 Kbps)

Supported ISDN switches and services for North America

The Remote Gateway 9150 unit supports the most common switch types. The following table shows the ISDN services available for each of the switch types.

Switch Type	ISDN Service
Nortel DMS-100	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)
AT&T 5ESS	Custom Point-to-Point
	Custom Multipoint
	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)
Siemens	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)

Supported European ISDN services

For European installations, the Remote Gateway 9150 unit supports country-specific EuroISDN installations, but does not support pre-EuroISDN installations.

Using supplementary services

Nortel does not recommend the following supplementary data services for use with Remote Gateway 9150 and 911x units:

- call waiting
- bearer-channel bonding
- call-waiting ID
- hunt

If you need to order Hunt services, order Multiline hunting rather than Overflow (EKTS) forward. The following must be true when you order these services on the BRI circuits:

— The B-channel requested during call set-up must be the B-channel that the call presents to

— The CALLED DN must be the DN of the B-channel the call presents to

Any other services, such as Forward on Busy must follow the guidelines above.

Example A

1 BRI circuit
B1: 416-555-1212, the published number
B2: 416-555-1213

A PSTN set calls B1.

The published number is currently busy.

Hunt must provide a call setup message for B2 with a CALLED number of 416-555-1213.

Example B

1 BRI circuit
B1: 416-555-1212
B2: 416-555-1213
The published number is different - 416-555-5000.

No calls are up.

When a PSTN set calls 416-555-5000, the call presents to B1 with a CALLED CLID of 416-555-1212.

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