

Remote Office Network Engineering Guidelines

Standard 1.0

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Contents

In this document

About this document	4
Understanding your network	5
Planning your network	6
Testing your network	11
Bandwidth usage table	13
Quality of Service	14
How QoS transitioning technology works	16
Ordering ISDN Lines	20

About this document

This document describes the network engineering guidelines for the Remote Office 9150 and the Meridian Internet Gateway Reach Line Card (MIG RLC).

It provides you with:

- a clearer understanding of how you should plan your network.
- a detailed description of Quality of Service (QoS) and how to implement Nortel Networks' patented QoS transitioning technology.
- instructions on how to order ISDN lines

Who should read this document

This document is written for the following individuals who are responsible for the installation, configuration, and day-to-day management of the Remote Office 9150 unit and MIG RLC, such as:

- Nortel Networks distributors
- telecom network managers and administrators
- data network managers and administrators

Understanding your network

A good design of the Remote Office network begins with an understanding of traffic, and the underlying network that transmits the traffic.

A network is a living entity that will grow and change over time. Although you may be satisfied with the design of your network today, your needs can change quickly. You must always monitor and evaluate your network on a regular basis to make sure it is serving your needs.

Before you implement Voice over IP (VoIP) on your network, you should do the following:

- Measure the type, protocols, broadcast-to-unicast ratio, and overall amount of traffic currently being transmitted over your existing LAN and/or WAN. Consider the quality of your existing traffic routes and identify bottlenecks. See “Planning your network” on page 6.
- Perform tests to determine if the network is capable of supporting VoIP. See “Testing your network” on page 11.
- Estimate LAN and WAN bandwidth usage per Remote Office port. See “Bandwidth usage table” on page 13.
- Understand the importance of Quality of Service (QoS) on your network. See “Quality of Service” on page 14.
- Understand how QoS Transitioning works. See “How QoS transitioning technology works” on page 16.

IMPORTANT: The procedures in these guidelines should be viewed as a guide only. Implementing VoIP on your network and achieving good voice quality largely depends on the type and volume of traffic being transmitted over your network.

Planning your network

Note: In the “Planning your network” section, the gray boxes indicate how the principles discussed above the boxes can be applied to the diagram on page 10.

1. The first step in planning your network for Voice over IP is to draw a network diagram that shows all devices on the network that could affect the traffic path between the MIG RLC and the Remote Office 9150.

The “Network planning diagram” on page 10 shows a sample network. There are three Ethernet Segments, one Radio Link, and one Token Ring in the figure. The MIG RLC is in the PBX attached to Link 1 and the Remote Office 9150 unit is attached to the Ethernet segment labeled Link 6.

- Label each segment on the network with the link speed and media capability, i.e., half- or full-duplex.

In the diagram on page 10, assume all Ethernet segments are half-duplex (typical). The speed is indicated beside the link name.

- Label each device with the throughput rate in bytes and packets per second. Packets per second is necessary since some software-based routers perform at significantly below the line-speed Ethernet (approximately 14K 64-byte packets per second for 10Mb Ethernet.)
- Major traffic sources on the network should be labeled with estimates of how much traffic they can (or typically will) generate and to whom.

An analysis of the network on page 10 indicates that all Token Ring traffic is local to the Token Ring, so the only traffic sources is the computer labeled as Source 1 in Link 1, and one computer labeled as Source 2 in Link 5.

2. Determine the path (source and destination) between the major traffic sources on your network.

In our example network, Source 1 and Source 2 communicate along the path of Link5:Link4:Link2:Link1. The MIG RLC and Remote Office 9150 unit communicate along the path of Link6:Link4:Link2:Link1.

3. Identify network bottleneck(s) and isolate them to minimize collisions. Most networks will have one or two bottlenecks. The bottleneck may be a link whose throughput is slower (fewer packets per second transmitted) than the rest of the network, or a convergence point where several links funnel down to a single path.

In our example network, the devices along the common path of Dev3:Link4:Link2:Link1 are the sources of collision/backup. The link speed indicates that the bottleneck is the radio link at 8 Mb. The PPS estimates indicate that Dev 2 is the bottleneck at 5K PPS. Using the average of 14K PPS (64-byte packets) for 10Mb Ethernet (10Base-T), Dev 2 will limit transmission to around 3Mb/s. Source 1 can generate 10Mb of data which can overload both Link 2 and Dev 2.

4. Prioritize the bottleneck, if needed. Keep in mind, though, prioritizing non-bottleneck links will have little or no affect on performance. If your network generates significantly more data than a bottleneck can handle, consider putting a mechanism into place to prioritize the voice traffic ahead of the data traffic.

Once again, we can refer to our example. Adding priority queues to Dev 3 accomplishes little since its output rates are high compared to the bottleneck. To correct the problem in our example network, we determine that we need to perform two steps:

- Add priority queues toward the radio link from Dev 2.
- Add priority queues and traffic metering toward the radio link from Dev 1.

Adding the priority queues is obvious, but great care must be taken so that our voice traffic doesn't prevent any data traffic from Source 1 reaching Source 2. The reason for the traffic metering is not as obvious, but the problem is that Dev 1 and Link 2 can generate traffic faster than Dev 2 can accept it. Therefore, Dev 1 should be configured to limit its output to around 2 Mb/s so that it doesn't overrun Dev 2 and cause discards.

There are many ways to provide this capability within routers and other WAN access devices, such as:

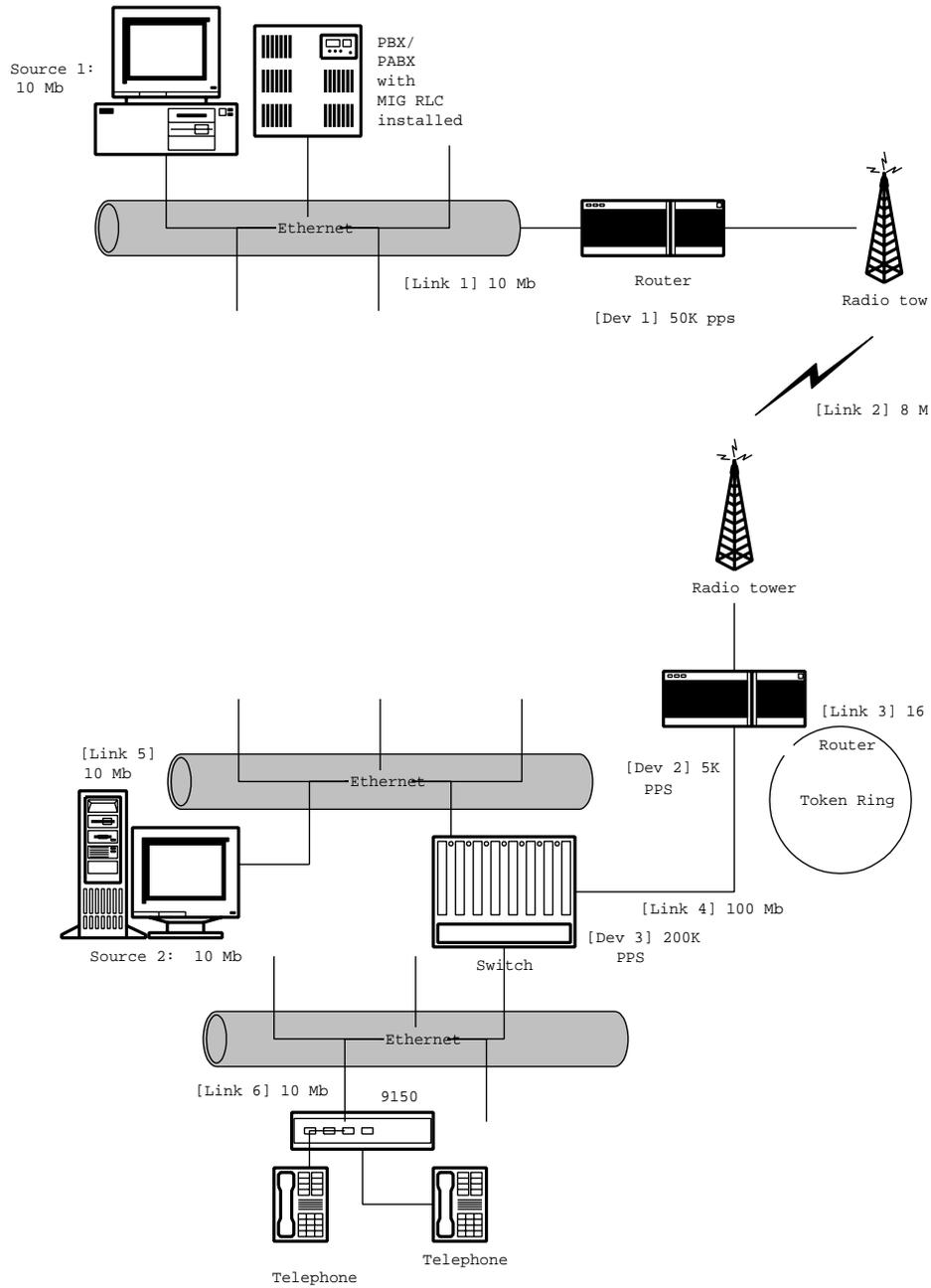
- Set up a priority filter on the basis of traffic source and destination IP addresses and use this as input to a priority-queuing mechanism. (This is the simplest method.)
- As a secondary mechanism, you can use the port numbers for prioritization. The Remote Office 9150 and MIG RLC use TCP port number 12800 (0x3200) and UDP port numbers of 20480 (0x5000) and 20482 (0x5002). It may be necessary to open up firewalls to allow connection to these port numbers.
- Certain WANs (e.g., Frame Relay) have the concept of guaranteed bandwidth. In Frame Relay this is known as Committed Information Rate (CIR).

The CIR must be sufficient to accommodate all of the voice traffic plus any critical data traffic. If the device at the bottleneck does not have the capability to prioritize traffic, it may be necessary to install an external traffic shaper, e.g. PacketShaper™ from Packeteer, or iSurf™ from Amplify Net.

- Some WAN interfaces (e.g., Frame Relay and ATM), perform traffic rate-admission over time. An example of this is Frame Relay's CIR. Consider the voice traffic to be CIR data. If possible, the Frame Relay Access Device (FRAD) should mark all non-critical traffic as Discard Eligible (DE).
- The user could subscribe to 64Kb per second of CIR. In situations where the link speed is significantly higher than the CIR, a special situation occurs where all of the CIR might be used up very quickly in a sampling interval. With a 2Mb per second link speed, all of the CIR can be used up in the first 32 milliseconds while the next 968 milliseconds are discarded if the network is congested. Some devices (e.g. routers) implement delay queues to accommodate this issue. In some documentation, this is referred to as traffic metering.
- Understand your broadcast domains so that the Remote Office 9150 unit will not be servicing unnecessary broadcasts. The MIG RLC and Remote Office 9150 unit make use of a half-duplex 10BaseT Ethernet port. The average load on the Ethernet segments should not exceed 30%.

Please note that all broadcast traffic, independent of destination, must be processed by the MIG RLC and Remote Office 9150 unit. Therefore, the amount of broadcast traffic on your network must be minimized and should not exceed 5% of network traffic.

Network planning diagram



Testing your network

If you are going to deploy your Remote Office 9150 unit on an existing network, perform the following tests to determine if the network is capable of supporting Voice over IP (VoIP).

- 1 Use the bandwidth utilization table on page 13 to estimate WAN and LAN bandwidth.
Total BW = max number of simultaneous calls * peak bandwidth from table
- 2 Perform a ping test (100 pings) at various times of the day to sample network conditions.
- 3 Take the minimum delay of any ping, and divide by two to get one-way MinDelay.
- 4 Find the maximum delay of any ping, and divide by two to get MaxDelay.
Note: Quality of Service transitioning technology is based upon three factors:
 - jitter
 - delay
 - packet lossThe close correlation of these three factors makes the measurement extremely subjective in nature.
- 5 $Jitter = (MaxDelay - MinDelay)$. If this number is greater than 90 milliseconds, there can be deleted voice packets during processing that can affect voice quality. For most customers, small numbers of these (e.g., < 2-5%) is acceptable; however, this is subjective. The jitter buffer can grow much larger than 90 milliseconds (up to 180 milliseconds), but it can only do this for very brief intervals.
- 6 Also, even with very little jitter, excessive delay can cause a problem. One-way minimum delay should range between 0-100 milliseconds. Larger numbers can only be tolerated in situations with very little jitter.
- 7 Any lost packets can affect voice quality. Once again, for most users 2-5% is acceptable.

Hint: Check to see if the devices on your network support wire-speed forwarding of 64-byte packets. Almost all ethernet switches, hubs, and routers advertise this capability (beware of those that don't). These devices will generally add very little to network delay (< 2-3 milliseconds), therefore, they aren't the normal suspects for delay. Lower-speed WAN links and software-based routing can add 10 or 100's of milliseconds and may significantly affect bandwidth.

Note: The G.729 algorithm is better at handling lost packets than either G.726 or G.711. The G729 algorithm is specialized for voice, so it will not allow sharp changes in frequency or amplitude due to lost packets. G.726 and G.711 attempt to recreate a non-speech wave-form and, as such, the recreated wave-form will change abruptly due to lost packets. This quick change can cause audible cracks or pops.

Bandwidth usage table

Packet Size	G.711 ⁱ	G.726 ⁱ	G.729 ⁱ
Voice Payload (bytes/30 ms)	240	120	30
Voice Hdr (bytes)	12	12	12
UDP Hdr (bytes)	8	8	8
IP Hdr (bytes)	20	20	20
IP Packet Size (bytes)	280	160	70
LAN Overhead			
Ethernet Frame Size (bytes)	14	14	14
IP Packet Size (bytes)	280	160	70
LAN Packet Size (bytes)	294	174	84
<i>Peak Data Rate (Kbps)</i>	<i>78K</i>	<i>46K</i>	<i>22K</i>
<i>Avg Data Rate-w/silence (Kbps)</i> ⁱⁱ	<i>47K</i>	<i>28K</i>	<i>13K</i>
IP Over Frame Relay Overhead			
Frame Relay Overhead (bytes)	4	4	4
RFC 1490 IP Overhead (bytes)	2	2	2
IP Packet Size (bytes)	280	160	70
Frame size (bytes)	286	166	76
<i>Peak Data Rate (Kbps)</i>	<i>76K</i>	<i>44K</i>	<i>20K</i>
<i>Avg Data Rate-w/silence (Kbps)</i> ⁱ	<i>46K</i>	<i>27K</i>	<i>12K</i>

i.Compression algorithm.

ii.Average data rate with silence suppression has been figured as approximately 60% of Peak. This number can be changed to reflect the site's actual values.

Quality of Service

QoS transitioning technology allows you to configure the Meridian Internet Gateway Reach Line Card (MIG RLC) to automatically transition calls from the IP network to the circuit-switched network when the voice quality falls below a user-configurable threshold.

Relationship between users and services

In the context of a Meridian 1 and Remote Office system, there are two interfaces that you must consider in the relationship between users and services:

- The Remote Office node (that is, the MIG RLC on the local-site PBX or the Remote Office unit at a remote site) provides an interface to the Remote Office system for end users. Voice services offered by the Remote Office node must meet user-oriented QoS objectives.
- The Remote Office nodes also provide an interface with the intranet. The intranet provides the “best-effort delivery of IP packets,” as opposed to “guaranteed QoS for real-time voice transport.” The Remote Office node translates the QoS objectives set by the end-users into IP-oriented QoS objectives. The guidelines call these objectives *intranet QoS objectives*.

The Remote Office node monitors the intranet’s QoS. The *transition threshold* parameter on the Remote Office node then dictates the minimum QoS level at which the Remote Office network will use the IP network to transport voice traffic. Note that the transition threshold is set on a per site-pair basis.

The *QoS level* is a user-oriented QoS metric and takes on one of ten settings. These settings reflect acceptable and unacceptable quality of voice service. Remote Office periodically calculates the prevailing QoS level per site-pair based on:

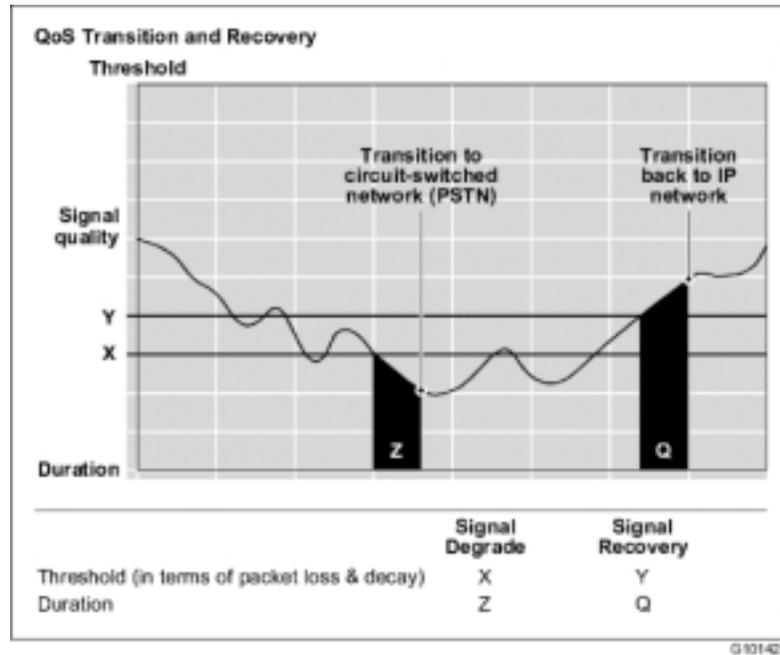
- one-way delay
- packet loss
- jitter
- compression algorithm

When the QoS level is below the transition threshold, calls to that destination are rerouted (transitioned) to the circuit-switched network. When QoS returns to a point above the threshold, calls are restored to the IP network.

The calculation is derived from the E-model in ITU-T Reg. G.114 and on ANSI TR56. When the QoS level falls below the transition threshold levels for that particular destination for the configured duration, that call is not accepted by the originating Remote Office node. Instead, the call is rerouted by the host PBX features over traditional circuit-switched voice facilities.

How QoS transitioning technology works

The following illustration, table, and text description explain how QoS transitioning technology works on the Remote Office 9150 unit:



Setting Description

- X The X threshold represents the point below which signal quality on the IP network is unacceptable (signal degrade).
- Y The Y threshold represents the point above which signal quality on the IP network is acceptable (signal recover).
- Z Represents the length of time that signal quality must be lower than the X threshold before calls are moved to the circuit-switched network (signal degrade in seconds).
- Q Represents the length of time that signal quality must be higher than the Y threshold before calls are moved back to the IP network (signal recover in minutes).

QoS transition to Circuit Switched Network

In the telephone call represented in the figure on page 16, signal quality begins in the acceptable range, which is above the X threshold. While signal quality remains in the acceptable range, as explained in the table on page 16, the MIG RLC routes calls through the IP network.

Signal quality can degrade below the X threshold without the MIG RLC moving the call to the circuit-switched network. But, if the call remains below the X threshold for the amount of time indicated by duration Z, the MIG RLC moves, or transitions, the call to the circuit-switched network.

Similarly, once the MIG RLC transitions a call to the circuit-switched network, QoS can recover to a point above the Y threshold, also explained in the table on page 16, but remain on the circuit-switched network. If signal quality remains above the Y threshold for the amount of time indicated by duration Q, the MIG RLC moves the call back to the IP network.

Voice quality can degrade while the circuit-switched network connection is being established in preparation for a transition.

On Remote Office units, once the circuit-switched connection is established, calls are routed to that connection, 64 Kbps at a time, from the IP network connection. The system waits several seconds before moving the next 64 Kbps to determine if the IP connection has become more stable.

As many calls as possible (to a maximum of 64 Kbps per B-channel) are transitioned from the IP connection to the circuit-switched (PSTN trunk) connection. The MIG RLC always transitions high priority users first. IP-only users are not moved from the IP network. By reserving 16K for X.11 signaling, the Remote Office system guarantees bandwidth for call setup, teardown, and state information.

Except for improved voice quality, transitions are transparent to all parties and can take place during a live call. However, if the cause of the transition to the circuit-switched network is a complete IP network failure, there will be noticeable break in the call (typically two to four seconds).

QoS testing the IP Network

Both the MIG RLC and the Remote unit run a pseudo-voice test to determine if the QoS on the IP network meets the standards configured by the user. The system monitors the following items:

1. Voice traffic over the IP network is monitored for delay and lost packets by both the MIG RLC and the Remote Office 9150 unit. If there is no active voice traffic, then simulated voice traffic called pseudo-voice is generated to approximate voice traffic. The pseudo-voice traffic is a burst of 1 second of 8K voice traffic every 10 seconds in each direction.
2. Both the MIG RLC and Remote Office 9150 unit gather statistics based on the pseudo-voice traffic to determine the congestion levels on the network. The system uses 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 within the predetermined threshold (see threshold “Y” on page 16), voice traffic is restored to the IP network.

When restoring the connection to the IP network, the system adds hysteresis (delay) to reduce the noise level during the transition. Hysteresis accomplishes the following:

- prevents thrashing between the circuit-switched and IP networks
 - ensures that the voice QoS exists on the IP network for a predefined amount of time
4. Calls might not be transitioned from the IP network for the following reasons:
 - calls might be IP only
 - there might be insufficient bandwidth to switch the call
 - transitioning might be limited by configurable transition bandwidth

QoS recovery to IP

When the IP QoS exceeds the Y threshold, the system waits for duration Q to ensure that the QoS is stable enough to resume service on the IP network.

If the QoS continues to exceed the Y threshold, all active calls are moved back to the IP network and all new calls are placed over the IP network.

Log reports and statistics

Configuration Manager provides a statistics log that identifies the number of QoS transitions.

Transparent transition and recovery

Because both connection types transport packetized voice data, the transition from one to the other is completely transparent to the user and can take place during a live call.

The one situation in which transition might not occur as quickly as expected is a complete IP network failure. In this situation, the user experiences a gap in service of several seconds (typically two to four) while the MIG RLC establishes a dial-up PSTN connection.

Ordering ISDN Lines

The section that follows provides you with information regarding the ordering of ISDN lines in and outside of North America.

Ordering ISDN

Tell your ISDN service provider what you need on the ISDN BRI service. In return, your ISDN service provider will tell you the ISDN line type used and the ISDN directory numbers.

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 (64 Kbps clear)
Note: Both B-channels must be Circuit-Switched Voice and Data.
- Caller Line Identification (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: Multiple Subscriber Numbering (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. See “Providing information to your service provider (North America only)” on page 21 for more 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 (64 Kbps)

Supported ISDN switches and services for North America

The Remote Office 9150 unit supports the most common switch types. The following table shows the ISDN services available on these switches.

Switch Type	ISDN Service
Nortel Networks 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 ISDN services outside of North America

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

Providing information to your service provider (North America only)

To ensure that you get the correct ISDN service for the Remote Office 9150 unit, tell your service provider how the ISDN line should be provisioned. 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.

Using supplementary services

The following supplementary data services are not recommended for use with the Remote Office 9150 unit:

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