



(your logo)

# Sample Report for Complete VoIP or IP Video Assessment

## Sample Data for Phase I: Network Discovery

### Contact Information

A list of all people involved in the project.

<i>Name</i>	<i>Company</i>	<i>Position</i>	<i>Office Phone</i>	<i>Cell Phone</i>	<i>E-mail</i>
Doug Goebig	Client	Telecom Specialist	272-509-xxxx		dgoebig@client.com
MJ Esty	Client	Network Technician	272-509-xxxx		mjesty@client.com
Sherman Kyser	Client	Technical Manager	272- 509-xxxx		skyser@client.com
Pat Cammack	BP	Nat'l Account Manager	800 331-xxxx		pcammack@bp.com
Ken Black	BP	Project Manager	800 331-xxxx	784-636-xxxx	kblack@bp.com
John Brown	BP	Design Specialist	800 331-xxxx		jbrown@bp.com
Georg White	BP	Convergence Specialist	800-331-xxxx		gwhite@bp.com
Austin Treptow	BP	Sales Engineer	800-331-xxxx		akyser@bp.com
Dean Zaremba	OUI	Director of Converged Solutions	303-337-xxxx	720-280-xxxx	dean_zaremba@ouinet.com
Aaron Smith	OUI	CCIE Technical Lead	743- 864-xxxx	648-859-xxxx	Aaron_Smith@yahoo.com
John Stevens	Consultant	WAN Engineer	567-893-xxxx		jstevens@hotmail.com

### Building Information

A list of all locations involved in project and customer contact for each location.

<i>Location</i>	<i>Address</i>	<i>Customer Contact</i>
World Headquarters	Centennial Drive, Northville, MI 48169	Doug Goebig
Houston, TX	Summer Drive, Houston, TX 44590	MJ Esty
Miami, FL	South Drive, Miami, FL 56012	Doug Goebig

### IP Network Closet Information

A list of all wiring closets (MDFs and IDF).

<i>Name (IDFs and MDFs)</i>	<i>Location</i>	<i>MDF Uplink Name</i>	<i>MDF Uplink ConnectivityType / Speed</i>	<i># Subnets Supported</i>	<i># Users Supported</i>
HQ (MDF)	Michigan	N/A	N/A	1	200
HQ Ware (IDF)	Michigan	HQ (MDF)	1 GB Fiber	1	25

Houston (MDF)	Houston	N/A	N/A	1	100
Miami (MDF)	Miami	N/A	N/A	1	40

### IP Network Equipment Information

The manufacturer, model and name of all existing and proposed equipment (routers, switches, firewalls).

A) Existing Equipment (Equipment that will be used as part of the VoIP Solution.)\*

Building / IDF	Manufacturer	Model	Host Name	Firmware Version
HQ (MDF)	Cisco	4006 Switch	Core_Switch	12.3
HQ (MDF)	Cisco	3725 Router	Core_Router	12.3
Ware (IDF)	Cisco	3550 Switch	Ware_Switch	12.3
Houston (MDF)	Cisco	3550 Switch	Hou_Switch	12.3
Houston (MDF)	Cisco	2620 XM router	Hou_Router	12.3
Miami (MDF)	Cisco	3550 Switch	Mia_Switch	12.3
Miami (MDF)	Cisco	2620 XM router	Mia_Router	12.3

\*Must include "configuration" (show run) and "about" (show version) outputs of all devices. (attach as text files)

```
#sh running-config
Building configuration
hostname "Mia_Router"
ip subnet-zero
ip classless
ip routing
```

```
interface eth 0/1
speed auto
ip address 192.168.3.255 255.255.255.0
no shutdown
interface t1 1/1
tdm-group 1 timeslots 1-24 speed 64
no shutdown
interface ppp 1
ip unnumbered eth 0/1
mtu 1520
no shutdown
cross-connect 1 t1 1/1 1 ppp 1
ip route 0.0.0.0 0.0.0.0 192.168.1.254
ip route 192.168.2.0 255.255.255.0
ppp 1
no ip n-form agent
ip http server
ip snmp agent
ip ftp agent
line con 0
no login
line telnet 0 4
```

```
#show ver
Cisco Internetwork Operating System Software
IOS (tm) C2600 Software (C2600-IS-M), Version 12.3(1a), RELEASE SOFTWARE
(fc1)
Copyright (c) 1986-2003 by cisco Systems, Inc.
Compiled Fri 06-Jun-03 22:09 by dchih
Image text-base: 0x80008098, data-base: 0x819D9744

ROM: System Bootstrap, Version 12.2(8r) [cmong 8r], RELEASE SOFTWARE
(fc1)

uptime is 1 week, 6 days, 52 minutes
System returned to ROM by reload
System restarted at 11:56:35 EST Wed Mar 24 2004
System image file is "flash:c2600-is-mz.123-1a.bin"

cisco 2650XM (MPC860P) processor (revision 0x200) with 126976K/4096K
bytes of memory.
Processor board ID JAE080412E7 (100909158)
M860 processor: part number 5, mask 2
Bridging software.
X.25 software, Version 3.0.0.
1 FastEthernet/IEEE 802.3 interface(s)
1 Serial network interface(s)
32K bytes of non-volatile configuration memory.
32768K bytes of processor board System flash (Read/Write)
Configuration register is 0x2102
```

B) New Equipment (Equipment that will be used as part of the VoIP Solution.)

<i>Building / IDF</i>	<i>Manufacturer</i>	<i>Model</i>	<i>Host_Name</i>	<i>Firmware Version</i>
N/A	N/A	N/A	N/A	N/A

### **IP Circuit Information**

A list of all circuits used for IP connectivity.

*A) Existing IP Circuits (Circuits that will be used as part of the VoIP solution.)*

<i>Carrier</i>	<i>Type</i>	<i>Router 1 / IDF</i>	<i>Router 2 / IDF</i>	<i>Circuit ID</i>	<i>CIR Speed</i>	<i>Port Speed</i>
ATT	T1	Core_Router HQ (MDF)	Hou_Router Houston (MDF)	4834- 1347513	N/A	1.54 MBS
ATT	T1	Core_Router HQ (MDF)	Mia_Router Miami (MDF)	4834- 1347514	N/A	1.54 MBS

*B) New IP Circuits (Circuits that will be used as part of the VoIP solution.)*

<i>Carrier</i>	<i>Type</i>	<i>Router 1/IDF</i>	<i>Router 2/IDF</i>	<i>Circuit ID</i>	<i>CIR Speed</i>	<i>Port Speed</i>
N/A	N/A	N/A	N/A	N/A	N/A	N/A

### **VPN**

A list of all network VPN links that will support VoIP.

<i>Circuit ID</i>	<i>Home/Remote Office</i>	<i>HW Model</i>	<i>HW Firmware</i>	<i>Client SW Type</i>	<i>Client SW version</i>	<i>SLA on Circuit</i>
N/A	N/A	N/A	N/A	N/A	N/A	N/A

### **IP Scheme and Subnets**

A list of all IP subnets on the network and their related purpose.

<i>Subnet</i>	<i>Mask</i>	<i>Location</i>	<i>IDF Name</i>	<i>Voice/Data</i>	<i>Network/Edge Devices</i>	<i>VLAN</i>
192.168.3.x	255.255.255.0	Corporate	HQ (MDF) HQ Ware (IDF)	Data	Both	N/A
192.168.4.x	255.255.255.0	Mia	Mia_Switch	Data	Both	N/A
192.168.5.x	255.255.255.0	Houston	Hou_Switch	Data	Both	N/A

## IP LAN Servers and Applications

A list of all servers that are connected to a LAN Switch that will be supporting VoIP.

Location	Application	OS	Host Name	IP Number	VLAN	Switch Host Name
Corporate	PDC / DHCP	2k3	Cor_PDC	192.168.3.10	N/A	Core_Switch
Corporate	Exchange	2k3	Cor_Exc	192.168.3.11	N/A	Core_Switch
Corporate	File Storage	2k3	Cor_File	192.168.3.12	N/A	Core_Switch
Mia	BDC / File / DHCP	2k3	Mia_Serv	192.168.4.10	N/A	Mia_Switch
Houston	BDC / File / DHCP	2k3	Hou_Serv	192.168.5.10	N/A	Hou_Switch

## IP Network General Network and Planning Questions

Has all cabling that will be used for VoIP devices been tested and certified as Cat 5 or better at all locations?

Yes

Which protocols does your networks supports other then IP?

None

If multicast is enabled on your network devices, what is its application (i.e., multimedia streaming, bulk file transfer, Real Networks)?

No

Indicate the multicast source hostname, IP address, and location (indicate switch name).

N/A

Is Network Address Translation (NAT) being used? Which Device(s)?

Yes, corporate firewall, but does not support VoIP

If supporting VoIP Are ports 1719, 1720, etc. open or blocked on your firewall(s), router(s) or VPN equipment?

Open, on devices that will support VoIP.

## IP Network Diagram(s)

Attach a graphical representation of connectivity all IP Network devices listed above. (WAN Diagram and detailed LAN Diagram for each building)

## PBX / Telephony Information, Per Location

Location Name: Miami

Processor:

What type of PBX processor will be used for call set-up at this location?

Avaya S8300

Is processor located local in this building or remote? If remote, what building?

Local

If local, what IP switch and IDF will provide connectivity for processor?

Mia\_Switch

Gateway:

What type of cabinet(s) or gateway(s) will be used at this location? If processor is remote, will gateway have LSP? What kind?

Avaya G700 - Qty 2

What IP switch name and IDF will provide connectivity for gateway?

Mia\_Switch

Phones:

How many total phones will be at this location?

40

How many IP phones will be at this location?

40

What is model number of IP phones?

Avaya 6408

What firmware version will IP phones use?

2.1

Which local IDF(s) will provide connectivity and how many phones per?

Mia\_Switch

Will IP Phone use DHCP? Will DHCP server be local or remote?

Yes, local DHCP

Will TFTP server be local or remote?

Local

Which local IDF will provide IP Phone connectivity and many phones per?

All 60 - Mia\_Switch

Will IP phones share cable drop with PC?

Yes

How will IP phones be powered?

POE Cisco 3550 switch

Voice Mail:

What type of voice mail system will support this site?

Inuity

Will this system be local to this site?

No

If not, where will it be located? Corporate (Michigan)

Trunks

Will dial-tone be serviced locally or from a remote location?

Local

If local, what type and how many trunks or line will be used?

Single PRI

If remote, which location will provide dial-tone?

N/A

Will long-distance trunks be handled differently? If so, how?

No

Softphones:

Will you be implementing IP Softphones at this location?

No

If so, list: manufacturers, model OS, OS service pack, and other running applications of all PC's that will be used.

N/A

Other Applications:

Is ACD being used at this location? If so, include manufacturer, model information, software version, and brief description of application.

No

Is any type of custom CTI being used at this location? If so, include manufacturer, model information, software version, and brief description of application.

No

Are wireless phones being used at this location? If so, include manufacturer, model information, software version, and brief description of application.

Yes, 10 Transtalk 900 wireless phones will be used by key personnel throughout building.

Is unified voice and email messaging being used at this location? If so, include manufacturer, model information, software version, and brief description of application.

No

Supported IP Calls:

What is maximum number of supported IP VoIP calls needed at this location per network segment (IDF)?

75% of the users or 30

What is the maximum number of supported IP VoIP calls needed at this location per WAN link or WAN PVC?

25% of users or 10

## **Data for Phase II: Converged Network Strategy**

### **Executive Summary**

Client intends to implement an Avaya IP platform at the corporate offices. The platform will initially support 5 remote sites with growth potential in the future. Client would like to leverage a managed WAN service from Sprint and existing 3 Com data switch infrastructure to deliver high-quality IP voice to all branches, while being shared for data traffic.

Upon initial review of hardware and network design, there are several items that if addressed will probably improve voice quality.

As a result, we are recommending that you take some steps to better prepare your data network, before we begin injecting voice traffic and taking statistical measurements of voice quality. Please note that only such voice traffic injection can provide factual data on quality of voice. Therefore, we cannot guarantee that your network would fail in its current state, nor can we guarantee that your network will pass once the recommendations in this document are completed.

However, we strongly feel taking these steps in advance of voice traffic injection will greatly improve your chances of getting a passing grade and dramatically reducing your need to run a second voice traffic injection test before VoIP can be deployed.

### **IP Scheme**

Your current IP scheme is well thought out and is working.

It is critical that all voice devices are assigned a subnet that is different than data devices so that proper QoS can be implemented. A strong recommendation would be to develop a private C-class subnet for location. Additionally, if these addresses were clearly distinguished from your data addresses, you could easily identify voice vs. data devices. For example, 10.10.x.x networks would work great, especially if you used the same third octet as your data network, based on location. For example, you could select 10.10.206.x for Remote 4 and 10.10.208.x for Remote 3. Note that these are just options, as you will need to determine what works best, based on your company's philosophies.

Note that the Avaya phones cannot use the 192.168.2.x network, so you will need to work your strategy around it.

### **NAT/Firewall Strategy**

IP telephony can struggle with IP address translations when traversing through a Firewall or NAT router. Client does have an Internet Firewall, and has their entire network NATed behind the firewall, segmenting it from the Internet. However, since all IP voice traffic will be behind the firewall, no IP telephony traffic will pass through the NAT'ed connection and thus will not interfere in the current design, and thus no special considerations need to be given to this topic at this time.

If, in the future, the corporate VPN will be used as a voice transport, as in the case of traveling or home users, using softphones, then special firewall considerations will need to take place at that time.

### **DHCP/Static Addressing Scheme**

Client currently uses DHCP for data devices. This solution will continue to work for data devices. Additionally, Client would like to use DHCP for the IP phones. It is suggested that that same DHCP server used for data devices be configured to support the voice subnets at each location (separate voice and data DHCP servers can also be deployed).

The DHCP server will need to be connected to an Ethernet port that is "bound" to both the local data and voice VLAN in order to function properly in both VLANs. DHCP option 176 will be used to give the phones IP

addresses, VLAN tagging information, and priority tagging. (If using separate NICs or servers, the ethernet switch port will be static to the appropriate VLAN)

Note that when a phone is first plugged it, it will not have any VLAN tagging information, and thus actually boot in the default VLAN. It is the data VLAN DHCP scope that will assign the voice VLAN information, forcing the phones into an automatic reboot. At which point, the phone will be in the voice VLAN and then the voice VLAN DHCP scope will deliver an IP address in the voice subnet and other pertinent data.

This process is documented in the “Avaya IP Telephony LAN Administrator’s Guide”.

Also, if DHCP will occur over the WAN, all routers will need to act as DHCP agents for all the voice subnets.

All voice processors, gateways, CLANs, MEDPros and supporting servers should be assigned a static IP number.

### ***TFTP Server***

The Avaya IP phone will need access to a TFTP server to function properly and when necessary, reboot properly. The TFTP server will also be used as a means to upgrade IP phone firmware as new releases come out every few months. A single corporate TFTP server could be used or a local one could be installed at each location. The first option requires less hardware and configuration, but the second option preserves WAN bandwidth. Please note that you will want the TFTP server running in both VLANs, since the phones will first boot in the data VLAN, and look for the TFTP server, and then ultimately reside in the voice VLAN and need TFTP access.

It is very common to see the servers performing DHCP services for each location can also act as TFTP server.

### ***IP End-points and Physical Cable***

Each cable run from the data switch will be supporting both a phone and a computer. VLAN identification should not be set at the switch level, since a device in the data VLAN and a device in the voice VLAN will be plugged into the same switch port. Thus 801.p/q technology will be used to determine VLAN segmentation. No specific information will be implemented on the computers, and thus they will boot in the default data VLAN. However, as mentioned above the DHCP server will assign a VLAN number to the phones so that they operate in the voice VLAN accordingly.

### ***Speed / Duplex***

Speed and/or duplex mismatches are one of the leading causes of poor IP voice quality. It is recommended that all static voice devices such as media servers and gateways also use static speed and duplex settings instead of the widely accepted auto-negotiate.

IP phones will typically function using auto-negotiate and should be deployed in that manner, and only changed as a troubleshooting process, should a specific phone experience problems.

### ***Power***

Avaya IP phones require standards-based power that can be provided via an external power supply, a mid-span power supply or a power-over Ethernet switch. At this time, Client has purchased several 3COM Superstack 4400PWR switches, which are capable of delivering standards-based POE. If these switches are strategically deployed to each location, they will meet the power requirements.

### ***Protocol Review***

The design in this document only supports the IP protocol.

### ***Layer 2 segmentation – VLANs***

Because of the natural “chatter” that occurs on a data IP network, it is critical to segment all voice devices into their own VLAN to prevent degradation of sound quality due to broadcasts and other data traffic.

You should create voice VLANs at each location for each subnet created. It is very common to use the same voice VLAN identify number at each location. For example, VLAN zero can be your data VLAN and VLAN 1 can be your voice VLAN at all locations. Additional VLANs for voice or data servers, would take on additional numbers.

If the Ethernet port is to be used strictly for a voice device, like a media server or gateway, the Ethernet port should be statically configured to the VLAN. If the Ethernet port is to be used for both a voice and data device, like IP phones, computers, or a router, then the Ethernet port should be bound (or trunked) to both the local voice and data VLANs. A good rule of thumb is that the Ethernet port should be statically assigned if the IP address of the device is statically assigned.

802.1p/q trunking will need to be used between all switch to switch and switch to router uplinks. This will pass VLAN and layer 2 prioritization values from switch to switch.

All switches should “trust” or pass DSCP values for layer 3 prioritization.

It is also recommended to change the VLAN test parameter in each phone from 60 seconds to 180 seconds. This can be done via DHCP.

### ***Layer 2 Traffic Prioritization***

VLANs will segment traffic from broadcasts, but cannot prioritize traffic over the data switch backplane during times of congestion. This is done by a two-step process of tagging packets, and then forwarding according to this priority tag. On layer two, the tagging used is typically done in the Class of Service Ethernet header. Within each voice end-point and voice network region you should assign a value of 6 to both L2 audio and L2 signaling for all packets. Each data device (computer or non IP-telephony) will keep a default value of zero. Note that Avaya recommends 6 and its equipment defaults to 6.

The second step is to program each data switch to read such tagging and give priority to the higher number tags, in this case, IP telephony traffic. Such a reading and queuing solution should be deployed across all switches that will be supporting voice. Note that many switches perform this feature without the need of any configuration.

### ***Layer 2 Equipment***

3 Com 3300 – These switches are capable of supporting all requirements outlined in two previous sections. However, these are older model switches and thus are of the first to support such features and therefore may not perform these functions as reliably as new switches. These switches should be strategically deployed to support just data where possible.

For L2 segmentation, 802.1q trunking is supported, but must be configured on each required port. This will need to be done before Phase IV testing and voice implementation.

For L2 prioritization, these switches have 2 strict priority inherent queues that use COS values. The high queue is for values of 4-7, while the low is for 0-3. This will work great for Client (without any additional configuration) since voice will be tagged at 6 and data at zero.

The current hardware of all 3300 models is sufficient, however, it is strongly suggests to upgrade all 3300 switches to the 2.70 firmware release.

3 Com 4400 – These switches are capable of supporting all requirements outlined in two previous sections. These are newer switches then the 3300 and perform voice functions better, and therefore should be

strategically deployed to support convergence, so that the 3300 can be used in data only environments, where possible.

For L2 segmentation, 802.1q trunking is supported, but must be configured on each required port. This will need to be done before Phase IV testing and voice implementation.

For L2 prioritization, these switches have 4 WRED inherent priority queues that use COS values. The high queue is for value of 7, 3<sup>rd</sup> queue is for value of 6, second queue is for values 3-5, while the low is for 0-2. This will work great for Client (without any additional configuration) since voice will be tagged at 6 and data at zero.

The current hardware of all 3300 models is sufficient, however, it is strongly suggests to upgrade all 4400 switches to the 3.21 firmware release.

### **Layer 3 QoS**

Typically a similar process to the Layer 2 tagging just mentioned is used on layer 3 to prioritize voice packets over a WAN link. Additionally, this is often much more critical, because WAN links are usually only the fraction of the bandwidth of a LAN switch.

On layer three, the Voice IP tagging used is typically done in the Type of Service IP header, which is also used to determine the DiffServ Code Point value (DSCP). Within each voice end-point and PBX network region, you should assign a value of 46 to L3 audio and 34 to L3 signaling for all packets. These are Avaya default values.

The second step is to program each data router to read such tagging and give priority to the higher number tags, in this case, IP telephony traffic. On all serial interfaces that support VoIP, a low-latency queue should be configured with a voice queue size = maximum number of IP calls on WAN multiplied by 25k. This queue should be given strict priority. A second queue should be created utilizing a percentage of bandwidth for voice signaling traffic (typically 2% - 4%). Remaining queue will be the weighted fair-queue for all remaining data traffic. This QoS should be done using class-maps and policy maps.

IP routes will need to be implemented on the Cisco routers to support both voice and data VLANs or subnets.

Like switch to switch, all router Ethernet uplinks should be programmed to support 802.1p trunking to pass priority tagging information.

### **Layer 3 Equipment**

Remote 1: The 1751 router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. It should also support up to 15 VoIP calls in the future or as part of your disaster recover plan. This model will need support for dual T1s. The model will need 96 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

Remote 2: The 1751 router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. It should also support up to 15 VoIP calls in the future. The model will need 96 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

Remote 3: The 1751 router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. It should also support up to 15 VoIP calls in the future. The model will need 96 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

Remote 4: The 2650XM router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. It should also support up to 25 VoIP calls in the future. The model will need 128 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

Remote 5: The 2650XM router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. It should also support up to 25 VoIP calls in the future. The model will need 128 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

Corporate: The 3745 router does meet all the layer 3 QoS needs as well as the layer 2 trunking requirements. This is a work horse router and should support growth in excess of 40+ calls on each of the 5 links. The model will need to support 5 T1 circuits. The model will need 128 Mbytes DRAM/32 Mbytes Flash, and IOS version 12.3(3a) with IP Plus feature set.

### ***WAN Bandwidth***

The proposed pipes for all locations are either 1.5MBS or 768kbps. The most calls supported on a T1 link is 20 and the most on a 768k circuit is 8. Considering 25k per voice call, at no time will voice occupy more than 30% of any given link. This is a great ratio to establish toll-quality IP voice. It should also leave ample bandwidth for normal business data traffic, like file share, email, web browsing, etc.

### ***Compression Algorithm***

Because we will be sending IP telephony traffic over a WAN connection with limited bandwidth we must implement IP telephony voice compression. Without compression a single call, with overhead, can take 90k.

The most common used compression Algorithm is G729a, because it has the best ratio of sound quality to packet size. What this means is that sound quality suffers only in the slightest bit over a non-compressed VoIP call, but the required bandwidth is reduced to below 25k per call.

Based on call volume and WAN bandwidth, G729.a is the recommended solution at this time. Please note the voice traffic injection phase that will be performed later in this process will best determine your final compression algorithm.

### ***Quality Control and Metrics Testing Summary and Checklist***

Once Client makes final decisions and changes regarding the recommendations in this document, we can move to the next phase of Network Readiness called "Quality Control and Metrics Testing". This phase will simulate voice traffic over the data network in similar patterns to the production voice environment and will produce statistics and graphs on key voice quality factors including packet-delay, packet-loss, jitter and MoS score.

The following will be required before this test can be run:

- An assessment PC will need to be provided for each network segment (one at each location). Windows 2k or XP is a must. 500 MHZ with 256 RAM would be optimal.

- The nic and nic driver must support 802.1p trunking.

- A PC technician at remote locations to spend 1 hour on phone with me to prepare local end-point PC's will need to remain on for 72-hour testing period.

- PC will need to be given voice IP address, and thus data switch port will need to be statically assigned to voice VLAN during 72-hour testing window.

### ***Disclaimer***

This document only provides suggestions on how to improve network performance. It is widely understood that there are many different ways to achieve end goals of a VoIP solution. In no way is this document claiming the suggestions made within are the only possible solutions, or even the best solution for any given situation. All final decisions on implementation choices are the end customer's. The accuracy of all equipment configurations and network performance is the direct responsibility of the implementation team that performs Phase III of the Network Readiness Process.