Application Note

Title Six Steps To Getting Your Network Ready For Voice Over IP

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Overview

This Application Note provides enterprise network managers with a six-step methodology to follow when preparing their network for Voice over IP service, including pre-deployment testing and network readiness assessment. Designed to help alleviate or significantly reduce call quality and network performance related problems, this document also includes useful problem descriptions and other information essential for successful VoIP deployment.

Contents

Pre-Deployment and Network Readiness Assessment Is Essential

IP telephony is very different from conventional data applications in that call quality is especially sensitive to IP network impairments. Existing network and "traditional" call quality problems become much more obvious with the deployment of VoIP service. For network managers, this means that LANs, access links and network equipment may need to be upgraded and that more sophisticated management and diagnostic tools are needed when deploying and maintaining VoIP networks.

Types of VoIP Performance Problems

There are three basic categories of performance-related problems that can occur in IP telephony: IP network problems, equipment configuration and signaling problems, and analog/TDM interface problems.

IP Network Problems:

- **Jitter** -- Variation in packet transmission time that leads to packets being discarded in VoIP end systems, usually due to network congestion
- **Packet Loss** -- Packets lost during transmission due to network errors, route changes, link failures or random early detection (RED) in routers
- **Delay** -- Overall packet transmission "lag time" that leads to two-way conversational difficulty

Equipment Configuration and Signaling Problems:

- **VoIP Endpoint Configuration -** Performance impact from the type of voice codec, packet loss concealment algorithm, and/or jitter buffer configuration used

- **Router and Firewall Configuration** - Firewalls or incorrectly configured routers can block VoIP traffic; routers need to be configured to deliver RTP packets in a timely manner

- **Bandwidth Allocation -** Network lacks sufficient bandwidth to support peak traffic volumes

Analog/TDM Interface Problems:

- **Echo -** "Echo" commonly occurs at the boundary between the digital network (VoIP or TDM) and analog local loops. This becomes very obvious and annoying with the additional delay introduced by the IP network problems previously described.

- **Signal Level -** Abnormally high or low voice signal levels, "clipping," excessive noise and "echo" occur due to incorrectly configured gateway signal levels

Network architects and managers should address call quality and performance management problems when they plan and deploy their IP networks, but they should be aware that these problems also frequently occur during normal day-to-day network operation.

Many VoIP-related problems are transient in nature and can occur at many places along the network path. For example, a single user accessing a file from a server can cause a period of congestion lasting a few seconds.

This, in turn, can cause short-term degradation in call quality for other users on the network. Given this, it is essential that network managers use performance management tools that can detect and measure these types of network impairments.

The transient nature of IP problems also means that they are not easily reproduced for analysis once the call is terminated. Unlike traditional POTS, once an end user completes and disconnects an IP call, vital diagnostic information about that call and its packet stream is lost. Network managers can use packet loss and jitter metrics to determine how bad the call quality was; however, these metrics alone do not provide enough diagnostic information to determine why the call was bad.

Six Steps To Getting Your Network Ready For VoIP

Step 1
Define High-level VoIP Requirements

Your ability to deliver good quality VoIP performance will depend on patterns of traffic and usage, existing network capacity, existing data bandwidth and many other factors. The first step is to define what your VoIP deployment will look like:

- What utilization do you expect?

- Where will gateways be located?

- How will internal calls be routed?

- How will external calls be routed?
- What codecs do you plan to use (e.g. G.711, G.729A, iLBC..)? And what bandwidth do these take (including IP headers - 96kbps for G.711 and typically 24kbps for G.729A)?

This should allow you to estimate the amount of network bandwidth required between each location. Consider the following example:

- Scenario: A remote location has 50 users, and no more than 10 users are expected to be on the phone at any one time. A gateway located on the central site will be used for connection to the telephone network, and the G.711 codec will be used. The remote location is connected to the central site via IP VPN and has a single T1 to access the IP network.
- Bandwidth requirements: Ten G.711 streams will require 10 x 96 kbits/s of bandwidth or 960 kbits/s, with some extra overhead for signaling. If the number of streams increases to 15, then the bandwidth needed would be 1.44 Mbits/s (almost the entire T1).

***Figure** 1: Effective Network Bandwidth By Codec Type For Different Voice Frame Sizes.*

Using low bandwidth codecs helps reduce bandwidth requirements; however, due to the overhead of IP, UDP and RTP protocols, such reduction is limited. The table below shows the effective network bandwidth used by some common codec types for different voice frame sizes. Note that using a longer frame can have some impact on delay, hence it is common to use 10-20 ms frames.

Low bit rate codecs also introduce some voice distortion and may be unacceptable for extended use.

Step 2
Map Existing Wide Area Network/VPN Capabilities

Many call quality-related problems occur in access links or on limited bandwidth WAN or VPN links. If significant jitter or delay occurs on inter-site connections, this is a strong indicator that similar problems will occur during VoIP deployment. Budget bandwidth usage between sites and verify that routers can prioritize RTP traffic.

- What is the bandwidth available between each site?
- How much data traffic is currently carried, (both average levels and peak levels)?

Codec Type	Codec Bandwidth (kbits/s)	Effective Network Bandwidth (kbits/s)			
		5 ms Frame	10 ms Frame	20 ms Frame	30 ms Frame
G.711	64	131.2	97.6	80.8	75.1
G.729A	8	-	41.6	24.8	19.1
G.723.1	6.3	-	-	-	17.4
iLBC	15.2	-	-	32	-

- Is sufficient bandwidth available to support both the current data traffic and the estimated bandwidth needed for voice?

- Is there any capacity for growth or for unexpected peaks in activity?

- Will the addition of voice traffic have an adverse effect on data application performance, i.e. if there is significantly less bandwidth for data applications, then how will they be affected?

Map VoIP bandwidth requirements onto available bandwidth.

Ten G.711 streams will require 10 x 96 kbits/s of bandwidth or 960 kbits/s, with some extra overhead for signaling. If the number of streams increases to 15, then the bandwidth needed would be 1.44 Mbits/s (almost the entire T1). This will leave approximately 500 kbits/s for data traffic. If more than the expected 10 users are simultaneously on the phone, then problems may start to occur with data applications due to insufficient bandwidth. And if more than 15 users are on the phone, then voice quality will degrade very quickly.

Step 3
Verify LAN readiness

Use a switched 100BaseT Ethernet architecture, potentially using VLAN to separate voice and data. Even with the use of switched Ethernet, problems can still occur due to duplex mismatch, excessively long Ethernet segments or bad cable connections.

- Are any hubs present in the LAN (don't forget remote sites)?

- Examine Ethernet switch statistics for evidence of packet errors or excessive collisions and upgrade equipment accordingly.

If Wireless LANs are being used, be aware that these can introduce significant impairments into the packet stream and should be checked carefully before using them for voice traffic.

Step 4
Verify Inter-Site Readiness

Pre-deployment testing is essential; however, you need to perform Steps 1 to 3 first. Many problems are predictable by simply verifying that sufficient bandwidth is available to carry the expected traffic level.

When conducting a pre-deployment test, it is important that the test:

- Is conducted over a sufficient period of time to assess network readiness under a variety of "typical" network conditions. Conditions vary through the day and through the week, and network problems are often transient in nature. There is no substitute for a sufficient amount of testing time.

- Is conducted under network load conditions that are similar to those expected post-deployment. There is little point in testing a single VoIP call when you expect that there will be 20 calls.

A number of vendors produce pre-deployment test tools or provide services. And, there are also a number of open source tools available. Spending some of your budget on pre-deployment testing is a worthwhile investment.

Step 5
Service Provider Relationships

Service providers will sometimes provide Service Level Agreements (SLA). These can be helpful; however, it is essential to measure service level using metrics that relate to Voice over IP performance. It is important to understand where the SLA is measured and how it is measured. Problems such as jitter and packet loss can often occur in access links, e.g., your T1. If the SLA is measured at the service provider end of the access link, then SLA metrics do not guarantee good performance.

Since packet loss and jitter typically occur in short periods of time, i.e., 1-2 seconds (due to congestion caused by data traffic), measuring long term packet loss can be less useful than first thought.

A SLA that commits to less than 0.1% packet loss over a month may sound great; however, this does not take the effects of jitter into account. Jitter can lead to packet discards (which are equivalent to packet loss) and can be a bigger problem on IP networks than packet loss.

This also does not represent the effects of packet loss burstiness. For example, if there are two bursts, 2 seconds long of 20% packet loss during a 3 minute call, then the average packet loss rate for the call is 0.44%. This would lead to the user hearing two periods of severe degradation during the call. If one call in five experienced problems of this type, then the average packet loss rate would be less than 0.1%.

New SLA agreements are being developed specifically for Voice over IP. They address the issues of packet loss distribution and the effects of jitter, and often represent the SLA in terms of call quality metrics.

Step 6
Define Performance Management Architecture and Tools

A well-defined fault and performance management architecture is essential for successful network operation, and it should be defined prior to the procurement of any VoIP equipment.

The diagram below shows the industry preferred VoIP performance management architecture which is based on RTCP XR (RFC3611), SIP QoS reporting (RFC6035), and related protocols.

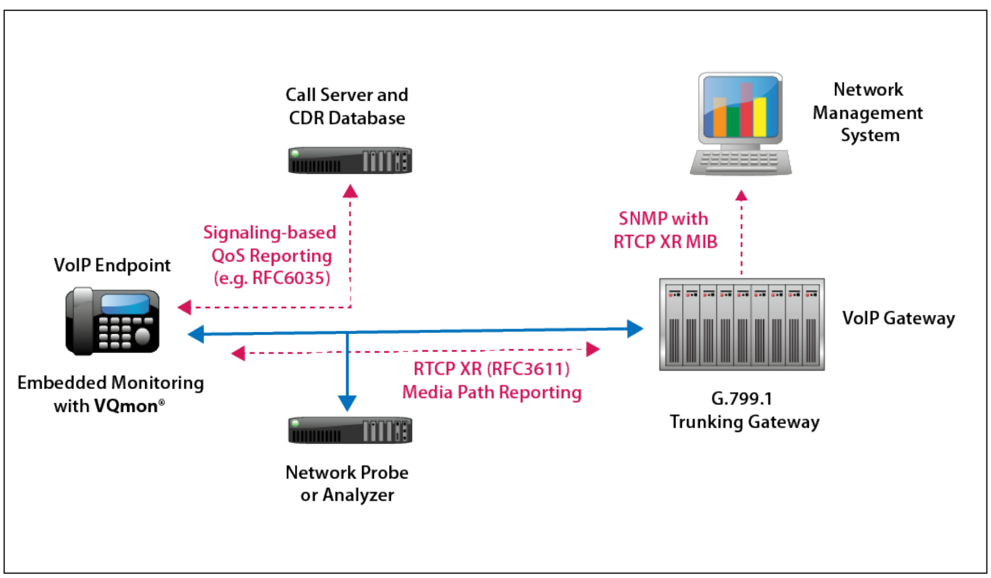

Figure 2: *VoIP Performance Management Architecture For Enterprise Networks.*

It is also essential to consider the issue of manageability which encompasses both the functionality required in IP endpoints and the protocols they use; plus any potential conflicts between secure protocols, i.e., Secure RTP, and the access required by management tools.

The sample RFP Requirements Document shown below provides guidance on what IP phone and gateway vendors should provide. The document introduces the concept of a monitored IP endpoint that supports RTCP XR. Monitored IP endpoints are essential if you want to be able to monitor from the network to the user desktop and detect/ resolve transient problems.

Figure 3: *Sample RFP Requirements Document*

RFP Requirements for VoIP Manageability

1 IP Phones

1.1 All IP phones shall support RTP with RTCP SR/RR (IETF RFC3550). If IP phones implement Secure RTP (IETF RFC3711) then RTCP SR/RR and XR reports must be transmitted unencrypted.

1.2 Monitored IP Phones shall support RTCP XR VoIP Metrics (IETF RFC3611). ITU G.107 and ETSI TS 101 329-5 Annex E (VQmon) shall be used to generate call quality metrics. All parameters of RFC3611 Section 4.7 must be supported.

1.3 Monitored IP Phones shall support the appropriate signaling-based QoS reporting protocol - H.460.9 Annex B for H.323, IETF RFC6035 for SIP.

2 IP Gateways

2.1 All IP gateways shall support RTP with RTCP SR/RR (IETF RFC3550). If IP gateways implement Secure RTP (IETF RFC3711) then RTCP SR/RR and XR reports must be transmitted unencrypted.

2.2 Monitored IP Gateways shall support RTCP XR VoIP Metrics (IETF RFC3611). ITU G.107 and ETSI TS 101 329-5 Annex E (VQmon) shall be used to generate call quality metrics. All parameters of RFC3611 Section 4.7 must be supported, in accordance with ITU G.799.1.

2.3 Monitored IP Gateways shall support the appropriate signaling based QoS reporting protocol - H.460.9 Annex B for H.323, IETF RFC6035 for SIP, or H.248.30 for Megaco.

3 Probes/ Analyzers

3.1 Probes and analyzers shall use ITU G.107 and ETSI TS 101 329-5 Annex E (VQmon) to generate call quality metrics for packet based non-intrusive or active monitoring.

3.2 Probes and analyzers shall support the detection and analysis of RTCP SR/XR (IETF RFC3550) and RTCP XR (IETF RFC3611) VoIP Metrics payloads, including the extraction of parameters related to delay, signal/noise level and endpoint configuration.

4 Embedded SLA Monitoring Function in Routers

An Embedded SLA Monitoring Function is a software agent installed in a service provider managed edge router, multi-service gateway or integrated access device located on the customer premises. The purpose of the SLA Monitoring Function is support the measurement of service level received by the customer, and to permit the service provider to implement some level of remote diagnostics. This is achieved by gathering statistics on live customer traffic and by implementing active testing for both inter-site monitoring and on-demand troubleshooting. There are two service models that may be supported:

- Managed VPN Service, in which all traffic is carried on encrypted tunnels between customer location
- Managed VoIP Service, in which the service provider participates in call management. Examples include IP Centrex/ Hosted PBX service.

Deploy Network In Stages

VoIP network deployment should be planned and executed in stages, with an initial pilot trial preceding large scale deployment. Make sure that the pilot trial includes some typical satellite locations, branch offices and teleworkers, since these are typical problem area hotspots.

Early during your network deployment, check for (and resolve) the following problem areas:

- Firewalls blocking access to voice traffic
- Congested access links leading to high levels of jitter
- Echo on incoming telephone network connections, i.e. non-VoIP connections to the phone company

Summary

This Application Note provided a six-step methodology for enterprise network managers to follow when preparing their network for Voice over IP service, including pre-deployment testing and network readiness assessment. The document also described many of the key factors essential for successful deployment of VoIP.

Other useful resources include: the VoIP Troubleshooter Web Site (www.voiptroubleshooter.com) and application notes on VoIP performance management available from specialized VoIP performance experts such as Telchemy (www.telchemy.com).

Notes

Acronyms

CDR	Call Detail Record
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
LAN	Local Area Network
MIB	Management Information Base
PBX	Private Branch Exchange
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RED	Random Early Detection
RTP	Real Time Protocol
RTCP	RTP Control Protocol
RTCP XR	RTCP Extended Reports
SLA	Service Level Agreement
SNMP	Simple Network Management Protocol
TDM	Time Division Multiplexer/-ing
VoIP	Voice Over Internet Protocol
VPN	Virtual Private Network
VQmon/EP	VQmon End Point

References

[1] IETF RFC3611 RTP Control Protocol Reporting Extensions (RTCP XR), November 2003

[2] ITU-T G.799.1 "Functionality and Interface Specifications for GSTN Transport Network Equipment for Interconnecting GSTN and IP Networks"

[3] ITU-T H.460.9 Annex B Support for online QoS-monitoring reporting within H.323 systems: Extended Performance Metrics

[4] ITU-T H.248.30 RTCP Extended Performance Metrics Packages

[5] IETF RFC6035 Session Initiation Protocol Event Package for Voice Quality Reporting, November 2010

[6] ETSI, "Quality of Service (QoS) measurement methodologies," ETSI TS 101 329-5 V1.1.1 (2000-11), November 2000

[7] ITU-T, "The E-Model, a computational model for use in transmission planning," Recommendation G.107

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