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Application Notes

- **Title** **Data and Fax modem performance on VoIP services**
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Overview

This application note describes the typical performance issues that users and designers of both data modems of all types, and Analog Telephone Adaptors (ATAs) face when using Voice over IP (VoIP).

Contents

Introduction

Voice over IP or Packet Telephony systems are quickly replacing a significant part of the telephone infrastructure. The services appear to the user to look exactly like standard telephone service – phones ring, caller-ID messages show, call waiting works. Hence users expect other devices that they connect to telephone jacks to work as well.

When data modems, fax machines and other data transfer devices designed for use on telephone lines are connected to VoIP services, ATAs or other gateways such as IP PBXs, they expect the "phone service" to behave in the same way as traditional TDM services. However as VoIP services are generally optimized for voice communications and modems were not designed to operate in packet environments, performance can be severely impacted.

Issues affecting operation of Modems

Issues effecting modems on the PSTN: noise, echo and non-linearity

Modems were designed originally for operation only on the PSTN (Public Switched Telephone Network). [1] In fact, it wasn't until the introduction of so-called PCM modems such as V.90 and V.92 did engineers and standards developers really begin to design modems for the way the PSTN actually functions. Up until that time, modems generally treated the PSTN just like a pair of wires connecting the two ends of the connection – essentially unchanged for 100 years.

[1] There are also private networks including leased-lines, analog PBXs and other similar environments in which modems are designed to operate. Because they are very similar to the PSTN as far as modems are concerned, we will refer to all such systems collectively as the PSTN.

Today's all-digital PSTN (mostly digital PSTN) terminates the analog phone lines in your house or office at a channel bank, on a line card. The electronics in the line card do the analog to digital conversion of the incoming signal (*from* your house) and the digital to analog conversions of the outgoing signal (*to* your house). From there, until it reaches the line card at the other end of the connection, the data usually travels in digital format.

The modern PSTN is very good about its internal data integrity from one end of a connection to the other. The PSTN is a circuit-switched network that establishes an "always-on" route from telephone A to telephone B. Therefore, there is very little, if any, signal degradation due to network impairments – especially lost packets. The only problems for modems are in general, electrical line noise, echo, other *analog* impairments introduced by the electronics and wires from the line card to the handset, and upon occasion, excess signal attenuation. Modems are designed to handle all of these signal impairments.

On the PSTN, there are no gaps in signals, no sudden change in end-to-end delay or other impairments that we normally associate with packet switched networks. As opposed to circuit-switched networks, packet-switched networks essentially route each individual packet from point A to point B one at a time and each may traverse an entirely different route.

Modern modems are (were) designed to operate under sometimes-severe conditions of echo, attenuation, noise and non-linear effects. Modems have internal, high-precision echo cancellers that are specialized for modem data, so they disable those used for voice already in the network; therefore, residual network echo is handled. Modems automatically change rates up and down as noise conditions change; so they handle line noise. V.34 (and V.90, V.92) have algorithms to mitigate effects of electrical non-linearity. They have automatic gain controls; so most signal attenuation problems are handled. In fact, modems often count on the normal, designed-in, signal attenuation from end-to-end (due to *Loss Plans*) to optimize their internal functioning.

Issues effecting modems with VoIP: Variable delay, packet losses, voice coding

Signal impairments (and/or processing) associated with packet networks are a different matter. For example:

- Packet losses cause discontinuous signals with either silence inserted or some sort of replacement signal whereas modems expect continuous analog signals.

- Adjustment in jitter buffers or route changes within the IP network can cause variable end-to-end delays, but modems rely on constant delays to effectively cancel both local and remote echo in order to operate in duplex modes.

- All voice coders alter the signals they encode but only *waveform coders* like PCM (G.711) and ADPCM, to a lesser extent (G.726), create a reconstructed signal that is reasonably close to the original. All other types of voice coders (such as CELP coders, *e.g.* G.729) in general use produce received signals that *sound like* the original but differ in most other respects. Modems need very close representations of the transmitted signal at the receiver in order to recover the transmitted data.

- Different line signal attenuation characteristics present unusual signal levels.

What VoIP does better than the PSTN

Interestingly, while the digital transport portion of VoIP connections is less robust than that of the PSTN, the analog portion of VoIP connections is usually ***better*** than the PSTN. This is mostly because a normal STN connection has anywhere from 100 to 17,000 feet or more of wire between the line card (the part of the PSTN where the analog signals are converted to digital) and the telephone. An ATA or other gateway device are usually co-located with the telephone, so wire runs are generally much shorter and as a result do much less "damage" to the signals.

Also the Analog to Digital and Digital to Analog converters used in ATAs are generally more precise than standard PSTN A/Ds and D/As so internally, before processing, they have a better representation of the line signal.

Problem and solution confusion

The result of all this is that the impairments that modern modems *are* designed to handle are less of a problem in VoIP systems, and impairments that they *are not* designed to handle are much more of a problem. Each of these differences presents a new challenge to modems.

The rest of this note looks at an evaluation of modems and VoIP and makes some suggestions for modem designers, network operators, and equipment manufacturers to improve modem operations.

Specific problems for Modems

In order to understand the behavior of modems better, many tests of modem connections were made using the test setup shown in Figure 1.

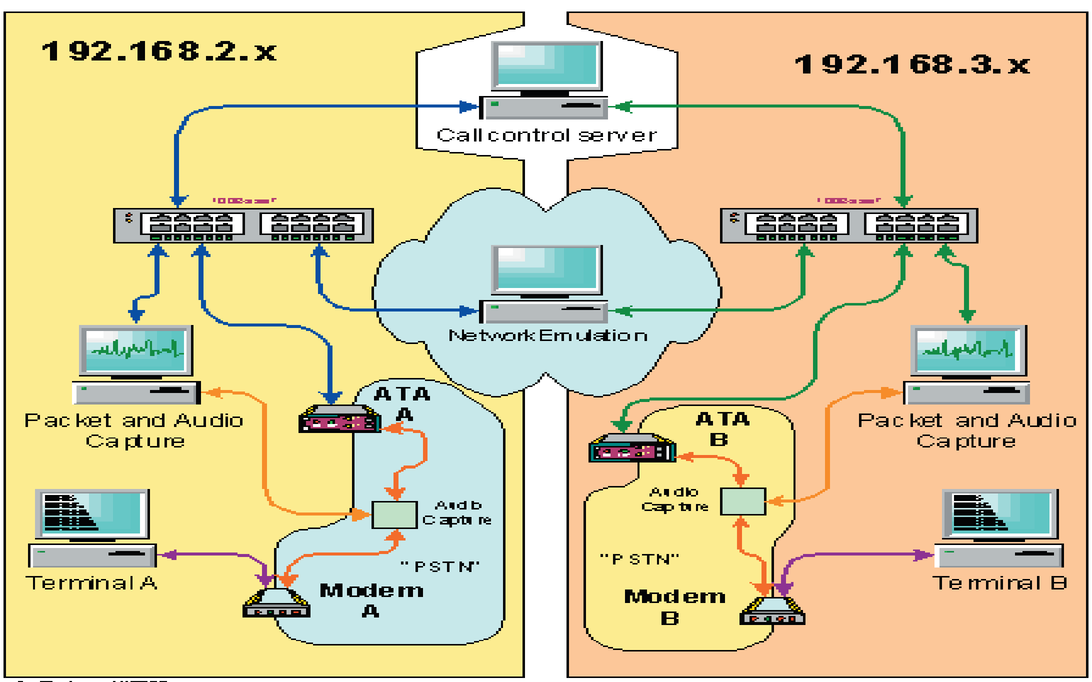

Figure 1: Test bed configuration

Essentially, two modems were connected using a private network that consisted of off-the-shelf ATAs, 100baseT Ethernet, a commercial network impairment bridge connecting the two subnets, and an assortment of hubs and computers to connect and monitor the tests.

The bridge was used to emulate some of the various packet loss conditions that are normally found on data networks. For these tests, loss models that were both random and bursty in nature were used. Modems then were connected using standard dial-up routines from standard communications software.

To analyze the results, the packet data to and from each modem/ATA combination was captured as well as the audio signals present on the ATA "phone lines"

Observations

As a result of this testing several things become obvious:

- Power levels on the ATA phone lines were generally too high.
- Losses caused retrains and eventually dropped calls when loss rates became to high
- Losses caused dramatic data throughput drops

Power levels

Modems in the PSTN expect end-to-end signal level attenuation of from 6 to 12 dB. The combinations of the ATAs, using standard default settings only attenuated the signals about 3 dB from end to end. This resulted in the modems seeing louder than expected signals on the lines, somewhat like trying to talk to someone who is always shouting.

Because the signal levels are higher than the modems expected, their design may not have been tuned to easily handle such hot signals and consequently did not perform well. In the tests, approximately 10% of the initial connections on otherwise perfect connections, failed.

Losses Cause retrains - sometimes

It was expected that losses – any losses – would cause the modems to immediately retrain. (Modern modems attempt to repair connections when they detect problems with the signals.) This was true, but only when loss rates began reaching 1 in 1000 packets, or about 1 every 10 seconds, did the modems really begin to have problems with the lost packets. As loss rates increase, packet losses occur more frequently and eventually retrains fail. The particular type of losses, bursty or random made only moderate difference in the reactions of the modems.

Losses Reduce throughput - always

Lost packets mean lost data – it doesn't arrive at the destination at all. In the case of modems, this means that some part of an information-carrying waveform is missing. Therefore, every lost packet represents some data that did not arrive even though it was sent at the other end.

The modems have no way to guess what this data is, so each packet lost per second, means some X amount of data throughput decrease. For example, a single packet loss at 100 packets per second means a minimum of a 1% data throughput decrease for that second. It is usually more than that however, since for each lost packet at least 2 or three more will not be decoded correctly because lost data makes the modem's receiver lose sync with the transmitter and some time is required to recover.

Finally, as losses occur more frequently, it takes the modems longer to recover. The table on the following page generally describes the effects of loss on modem throughput. Note that the decrease also is caused by the time necessary to retrain if the losses cause retraining (7-30 seconds each).

Loss Rate	Ideal throughput	Measured throughput	% Decrease
0.1	31,200	28,080	10
1	31,200	15,600	50
10	31,200	3,120	90

Table 1: Modem throughput and packet loss rates.

Performance management of modem operations

With this understanding of the issues involved, network operators are in a position to control and untimely improve the performance of modems and fax machines in their networks. The question becomes "What can be done?"

First, they have to know that there are problems, and then they need to know what to do about them.

Performance management architecture: VQmon

VQmon is a sophisticated VoIP quality measurement and management technology for monitoring performance of VoIP services, including voice, video, and now, FAX and data modem operation. Beginning with release 2.2 of VQmon, VQmon began reporting a new class of metrics designed to specifically inform the network operator of the potential for problems for users of VoIP services that carry FAX and data modem traffic. This new class of metrics includes estimates of throughput (Figure 2) as well as reliability indices.

Figure 2 shows the results of a number of tests in which modem throughput was predicted using VQmon and compared to the measured throughput, for a variety of network conditions.

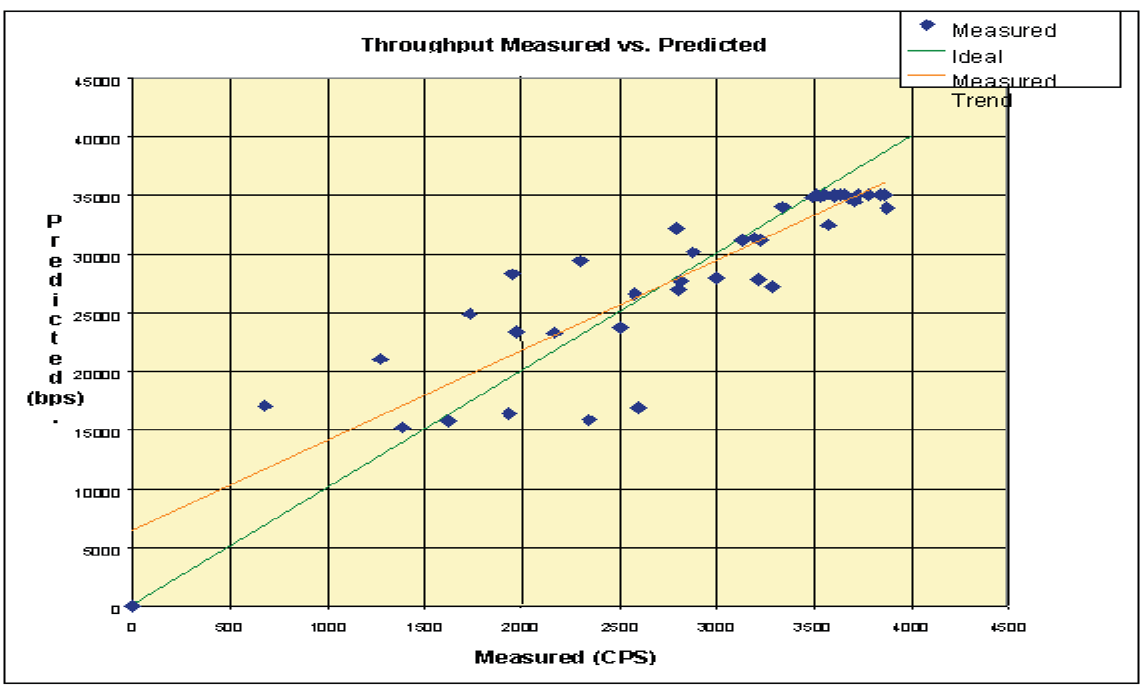

Figure 2: Modem rate predictions.

In addition to the existing metrics, additional metrics quantifying success rates for FAX transmissions, and predicted page-transmission times will be implemented in upcoming releases. Compared against ideal conditions, metrics such as these, give the operator indications that customers using such devices will be having problems.

Not only can are theses metrics reported for FAX and data calls, they also are reported for ***normal Voice calls*** – even though modems are not in use. This allows the operator to anticipate problems with data services *even before the customer begins having problems.*

What else can be done?

The challenge presented by these "unusual" impairments to modem and ATA designers, as well as network managers whose networks transport this type of traffic are not insurmountable. With some minor design modifications by developers, as well as informed operation by users, noticeable improvements can be made.

Once the operator knows there are issues, which is possible when VQmon/EP is part of ATAs and gateways, he can make sure that:

- That gateways are designed properly to handle data traffic.
- That makers of modems and fax machines harden their designs for VoIP system operation.
- That the deployment of systems takes into account causes of problems.

Properly designed gateway devices

For the network manager to know there are problems, standardized QoS measurement and reporting protocols must be implemented in gateway devices.

In addition, several fundamental setup issues should be addressed in the deployment of gateway systems.

Gateways, in the presence of data traffic should:

- Run detectors that reliably detect the use of modems and fax machines – you have to know a modem is operating and then:
- Enable PLC.
- Increase end-to-end attenuation to a minimum of 6 dB.
- Fix the length of, and turn off adaptation of jitter buffers.

PLC

Packet Loss Concealment algorithms mask missing audio signals on a line by replacing them with signals based on previous data. When modems encounter a 10 ms slice of data that is the correct amplitude, but otherwise incorrect, they interpret it as a noise anomaly and handle it with some dexterity. The data is still lost, but recovery is easier.

To improve thing more, gateway designers should develop specialized PCL algorithms that reduce the effects of lost packets on modems.

Increase attenuation

Since modems are optimized to operate with certain signal levels, VoIP connections should provide them. Upon detection of a modem, at least 6dB of signal attenuation end-to-end should be used. This allows modems to operate in the ranges that they are designed to work best. It will reduce initial connection failures and maximize connect rates.

Jitter buffers

Modems rely on constant end-to-end transmission delay to mitigate effects of echo and timing offset. Modems also do not care about rather lengthy delays, most being designed to operate over satellite links if necessary.

Adaptive jitter buffers do not provide constant delay, but varying delay, especially at various adjustment and reset events. By not allowing adaptation, the constant delay that modems rely on is preserved.

Moreover, if fair amounts of jitter are present, a long jitter buffer will allow ATAs to not discard packets under extreme conditions. Since the modems are designed to handle delay, a long jitter buffer will cause no problem, whereas a discarded packet will.

Properly designed modems

Modems too can be designed to better operate in a packet environment. Just because in the past they have been designed to work on the PSTN, does not mean that some additional steps can be taken that will improve performance in packet network.

Modems on packet-based networks should:

- Implement algorithms that reliably detect operation on packet networks.
- Be more tolerant of timeouts when trying to recover from and severe loss event.
- Be aware of the potential for higher than normal signal levels.

Detection of packet networks

Understanding that they are operating on a packet network, modems can be adjusted to the environment. By recognizing that impairments on packet-based networks impart “blocky” impairments, designers can watch for internal conditions that are blocky as well, potentially indicating a packet network. It may even be possible to use special signals during handshaking.

Alternatively, just having a setting that users can during initial setup to identify the connection as a VoIP connection.

The key is to know.

Once you know

When the modems failed in testing, it often seemed that the modems were too quick to give up in suddenly bad or changed conditions. This is somewhat expected though since in the PSTN, conditions change very slowly over time and modems do not generally expect sudden changes. Some modems did such a poor job of re-aligning with each other using the V.34 recovery procedures that by the time both modems got back in sync, it was too late - one had given up and hung-up.

In addition, modems failed often to detect, in a timely manner, the indicator signals from the other modem that it was requesting a retrain. By wasting time initially in a retrain, general time-outs for retrain completions often expired causing failures and terminated calls.

Modem designers should relax time-outs on retrains, or retrain-count disconnects or other internal working of the modem handshake in order to ride out longer events, and not to be so quick to give up when in packet networks.

Other solutions

QoS scheduling of packet traffic in networks should be a priority of designers. By knowing that a modem call is in progress, and that there are potential for problems on a particular port because of a known history reported by VQmon, a data call could be prioritized, thus reducing a major cause of losses and their subsequent problems for modems.

ATAs that implement modem relay (V.151) demodulate and then relay only data instead of audio. ATAs that implement Voice Band Data (V.152) add redundancy to the audio stream. Both these approaches certainly can improve error rates. They are however, expensive to implement in a number of ways including memory use and CPU bandwidth, as well as development costs. This is especially true of modem relay.

Dedicated lines or access points that are also prioritized are also a worthwhile idea. In a business, a PoS terminal or a FAX machine are often mission-critical devices. Here, monitoring is even more essential in the gateways because these lines may be specifically provisioned for this traffic type.

Finally, other solutions are possible that are less ideal. One is to limit data rates and modulations to those that are less susceptible to packet loss impairments. For example, PoS terminals use much slower modems (Bell212, V22*bis*) because they are more robust to all types of problems – just because they are slower and less complicated in their internal workings. They can do this because the amounts of data necessary for their operation is small, and a typical transaction is very short, only several seconds.

Summary

Modems and fax machine were not designed to operate in the new environments presented by VoIP services. However, by managing networks and knowing when modems and fax machines have the potential for problems on particular lines, operators can reduce customer complaints by reacting ahead of time.

By requiring that gateways and ATAs in use on their networks provide this information, steps can be taken, *in advance*, to correct problems.

Notes

Acronyms

A/D	Analog to Digital
ADPCM	Adaptive Differential Pulse Code Modulation
ATA	Analog Telephone Adapter
CELP	Code Excited Linear Predictive
D/A	Digital to Analog
dB	Dicibels
IP	Internet Protocol
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PLC	Packet Loss Concealment
PoS	Point of Sale
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTP	Real Time Protocol
TDM	Transportation Demand Management
VoIP	Voice Over Internet Protocol
VQmon/EP	VQmon End Point
VQmon/SA	VQmon Stream Analysis

References

[1] Delayed Document D-96, ITU-T Study Group 12, Geneva, October 2005, Telchemy Incorporated, QoS Metrics for use in VoIP for Data and Fax modems.

[2] ITU-T Recommendation V.22 bis - 1988 - 2 400 bits per second duplex modem using the frequency division technique standardized for use of the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits.

[3] ITU-T Recommendation V.34 - 1998 - A modem operating at data signalling rates of up to 33 600 bit/s for use on general switched telephone network and on leased point-to-point 2-wire telephone-type circuits.

[4] ITU-T Recommendation V.90 - 1998 - A digital modem and analogue modem pair for use on the Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33 600 bit/s upstream.

[5] ITU-T Recommendation V.92 - 2000 - Enhancements

About Telchemy, Incorporated

Telchemy, Incorporated is the global leader in VoIP Performance Management with its **VQmon**® and **SQmon**™ families of call quality monitoring and analysis software. Telchemy is the world's first company to provide voice quality management technology that considers the effects of time-varying network impairments and the perceptual effects of time-varying call quality. Founded in 1999, the company has products deployed and in use worldwide and markets its technology through leading networking, test and management product companies. Visit www.telchemy.com.

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