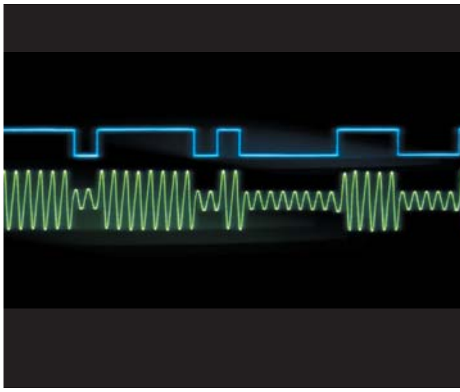
Symmetricom®

It's All in the Timing

Synchronizing the VoIP Network for Quality, Savings and Performance



WHITE PAPER

Voice over Internet Protocol (VoIP) is fast becoming the telephony technology of the future. Yet while VoIP offers tremendous benefits in reduced costs, improved call accounting and increased functionality, it presents challenges as well. First among these is Quality of Service (QoS), which can suffer as telephone networks converge with IT networks and become subject to jitter, packet loss, and latency, not to mention malicious threats like viruses and denial of service attacks.

One of the keys to success for any VoIP network is accurate time synchronization. By synchronizing time across a VoIP network within milliseconds or microseconds, organizations can better ensure the quality of phone service while reducing costs and improving network security.

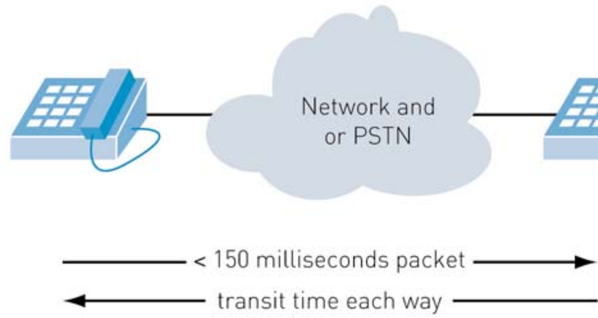


FIG.1 Packet transit times, including voice encoding and decoding, must be less than 150 milliseconds for acceptable QoS.

IMPROVING QUALITY

The voice quality in a VoIP system can be degraded due to distortion from compression techniques, poor echo cancellation, dropped packets, and packet delay or latency. Of these, latency is particularly troublesome because it involves the entire network from one end to the other.

One component of latency is the jitter buffer used to gather packets at the gateway to the PSTN (Public Switched Telephone Network) system or the receiving VoIP phone for the outbound voice signal. Increasing the size of this buffer can reduce packet loss, but that leads to a longer latency.

Latency Factors

The majority of VoIP phones are still made to traditional phones. The latency of the voice signal from the VoIP phone to the PSTN phone consists of delays at the following network elements:

- The VoIP telephone
- IP network routers or switches
- The IP to PSTN gateway
- The wires
- Other delays in the PSTN system

At the phone, the signal must be sampled, encoded, and packaged as Real Time Protocol (RTP) packets. Any routers encountered in the IP network contain input and output buffers. At the gateway the packets will encounter more buffers including the jitter buffer, plus delays associated with decoding and reassembling the signal. The transmission delay alone, due purely to travel time through wiring, would be about 20ms for a call between Los Angeles and New York.

Lastly, travel through the PSTN system could involve breaking the signal up into ATM or frame relay packets and reassembling them after transport through an optical fiber. In a VoIP phone on another network, then the phone call will have to be converted back into RTP packets, complete with encoding, decoding and buffer delays.

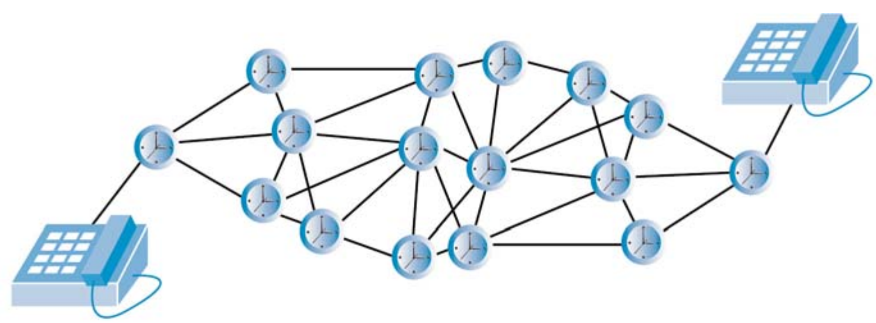


FIG.2 Time accurate server and router log files play a key role in troubleshooting and identifying root cause problems in complex VoIP networks.

Reducing Latency Through Synchronization

ITU-T Recommendation G. 114 sets the total latency budget at 150ms or less. Longer latencies lead to voice quality degradation and the perception that VoIP is inferior to typical PSTN performance. Network designers will commonly create a latency budget consisting of encoding time + transport + jitter buffer + decoding time. In this case transport delay is a catchall that includes all buffer delays except the VoIP jitter buffer, wire transmission delay, etc. Advanced compression schemes allow for a more efficient use of the network by reducing the number of bits needed to represent a voice signal. Unfortunately, this lack of redundancy in the signal makes the voice message more susceptible to degradation due to packet loss. This may necessitate the need for a larger jitter buffer, adding even more delay than just the additional encoding/decoding time. For this reason, most commercial VoIP systems use the ITU G. 711 uncompressed codec standard.

One way to reduce network traffic related to VoIP is to empower routers to discard RTP packets which are too old to have a chance at being used. Accurate time synchronization is essential for such a technique so that the routers can make meaningful decisions about time stamps.

Measuring Latency Through Synchronization

Essential to assuring low latency and acceptable QoS (Quality of Service) and SLA compliance is the ability to make a reasonable measurement of the latency in the network. This requires quality synchronization between measurement probes to obtain meaningful results.

The best way of measurement is if the one way latency test is performed simultaneously from both ends to reflect an in process phone conversation. Synchronization on each end of the call is required for this type of test – ideally with a precision Global Positioning System (GPS) referenced clock at either end. Since the GPS signal is ubiquitous and is based on the atomic clocks aboard each satellite, very precise synchronization is easily achieved across very large geographic distances. Packets can be accurately time stamped both on the sending and receiving end thereby enabling meaningful measurements to be made.

Synchronizing the Gateway Interface

Eventually packets arrive at the gateway between the VoIP system and the Public Switched Telephone Network, PSTN. The PSTN uses a very defined timing hierarchy for synchronization of traffic on the network. PSTN voice packets must arrive in order and with low latency and jitter. This gateway also represents a change in the general synchronization requirements. VoIP systems synchronize by way of time stamps to aid in latency reduction and network log file integrity. The PSTN uses synchronization to improve efficiency and data throughput.

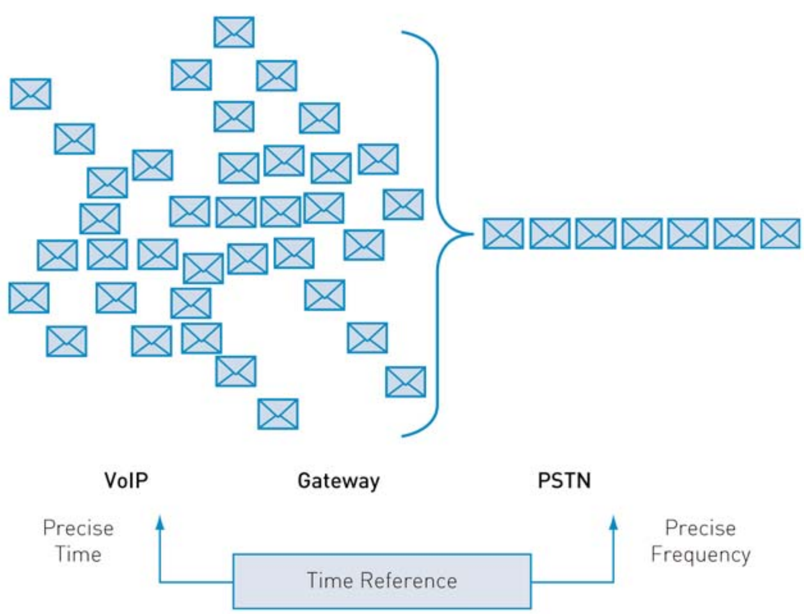


FIG.3 Precise time is required for VoIP; precise frequency is required for PSTN.

Providing this synchronization requires a versatile time reference that can supply the Stratum 1 level frequency reference for the PSTN and the accurate time stamps for the VoIP side. Many of these VoIP/PSTN gateways, also known as a softswitches, already employ NTP for accurate time stamping. Stratum 1 level timing is already permeating the edge the PSTN network by synchronizing customer premise ATM routers and switches. By adding VoIP to the edge it further increases the need to expand the synchronization capabilities of the Stratum 1 timing clock. Again, a quality GPS referenced clock can support both the NTP and the Stratum 1 time and frequency requirements.

REDUCING COSTS

Eliminating the Need for Over Provisioning

In the absence of reliable methods to measure, monitor and assure quality of VoIP transmission, organizations tend to spend significant resources and overprovision VoIP bandwidth in order to ensure acceptable levels of service. By providing the ability to reliably measure and address latency within the VoIP network, network time synchronization allows organizations to eliminate the need and cost of over-provisioning.

Reducing Billing Errors with Accurate Call Detail Records

Network time synchronization can also reduce cost by improving billing accuracy. In a VoIP system, Call Detail Records (CDRs) are time stamped to provide data about call origination, destination, and duration for billing purposes.

When calls travel across various networks, gateways, and VoIP servers, CDRs contain information from each leg of the call. Billing accuracy is dependent upon the accuracy of the time stamps. When a VoIP network is not synchronized, accuracy will quickly be affected, and the integrity and credibility of the billing system will be questioned. Synchronization is especially important when CDR data is shared between carriers, in which case any billing discrepancies require costly mediation.

Many VoIP networks now include unified messaging, videoconferencing, bandwidth on demand, and other real-time provisioning services. Accurate time stamping—based on network time synchronization—is essential to providing accurate billing services for all of these technologies.

INCREASING PERFORMANCE

Improving Event Tracking with Accurate Log Files

Network time synchronization also can help to improve performance and availability of a VoIP network. While users are frustrated when a data network goes down occasionally, no one will accept the loss of the voice network for even a few minutes. VoIP networks must be constantly maintained and trouble spots proactively addressed in order to keep business-critical voice systems up and running.

The key to this high availability and reliability is accurate server and router log files. Each log file entry is given a time stamp, allowing administrators to troubleshoot root-cause problems by determining the time and ordering of events. Server logs are compiled from data from a variety of hosts, so it's imperative that the time stamps on events from different sources be correct and accurate within milliseconds. When they are not, administrators are hampered in efforts to prevent or address downtime: ordering events becomes harder, so troubleshooting takes longer. A highly synchronized network, on the other hand, makes troubleshooting much easier, leading to improved performance and availability.

NETWORK TIME SERVERS: THE SIMPLE SOLUTION

Time synchronization across network servers, routers and network devices is not a difficult endeavor. Using the well established Network Time Protocol (NTP, RFC 1305), and a reliable time source, such as a dedicated network time server that references the GPS system, synchronization of servers and network devices can be easily maintained. In fact, many operating systems and network devices already incorporate support for NTP.

Network time servers should always be referenced to a reliable source of time. NTP uses Coordinated Universal Time (UTC) which is the same worldwide. The GPS satellite system is the most readily available source for UTC time in the world. By synchronizing your network to UTC you remove one more source of interoperability problems between your network and others. This is important since VoIP traffic may transit many networks requiring the correlation of log files from various networks to solve a problem.

Network time servers today are the quintessential network appliances. For example, the Symmetricom S200 GPS network time server provides accurate, reliable, and secure time to the network in a slim rack mount configuration. It installs quickly, has atomic clock accuracy from its embedded GPS receiver, and can synchronize thousands of clients on the network. As a Next Generation Ready device with support for IPv6 as well as the current IPv4, the S200 can provide accurate time to both existing and future networks, as well as the inevitable hybrids.

TIMING IS EVERYTHING

When an organization is implementing a VoIP network, time synchronization is usually not a priority – until problems occur. Then the true value of synchronization becomes very clear. By reducing latency, providing better metrics, enabling troubleshooting, improving billing accuracy, and reducing costs, time synchronization helps to deliver on the quality, savings and performance that VoIP telephony promises.

For more information visit:
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