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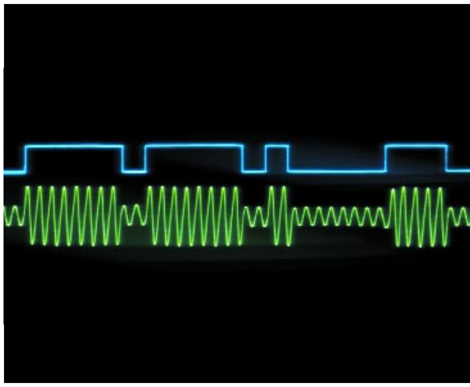
Synchronization Essentials of VoIP



WHITE PAPER

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Introduction

As we accelerate into the New World of VoIP we assume we can leave some of the trappings of wireline telecom behind, such as the need for synchronization. After all, an IP network is about as asynchronous as it gets. While this may be true in some respects, precise time and synchronization continue to permeate many areas of IP telephony operations. Customer expectations of voice quality and service reliability remain unchanged, and as a result, the need for precise time remains unchanged though it manifests itself in different ways. The bottom line is that trying to design a VoIP network without considering network synchronization is probably the shortest path to realizing that you need it.

Voice Latency

The greatest challenge in implementing a carrier class VoIP system is meeting the very high standard for voice quality set by the traditional PSTN system. The same is not true in wireless phones, where inferior quality is accepted as the price of mobility. Or for that matter, inferior VoIP quality in return for the savings of a cheap Internet long-distance phone call.

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. To make matters worse, most of this network is outside the control of the VoIP solution provider. Yet, no one wants a conversation where you need to say "over" at the end of every sentence.

One component of latency is the jitter buffer used to gather packets at the gateway to the PSTN 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. A more in-depth discussion of IP voice quality issues can be found in the Telecommunications Industry Association (TIA) bulletin TSB-116.

Latency Factors

Voice calls using VoIP technology can be made to another VoIP phone or to a traditional phone on the PSTN. Since VoIP phones currently represent only a small fraction of phones in use, we will consider the latter case in detail. 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 this case, input and output buffers are encountered again. If the final destination is 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.

Latency Error Budget

ITU-T Recommendation G.114 sets the total latency budget at 150ms or less. Longer latencies lead to voice quality degradation, which is perceived as 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.

Another 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. Such a technique requires that VoIP phone and routers be time synchronized so that the routers can make meaningful decisions about time stamps.

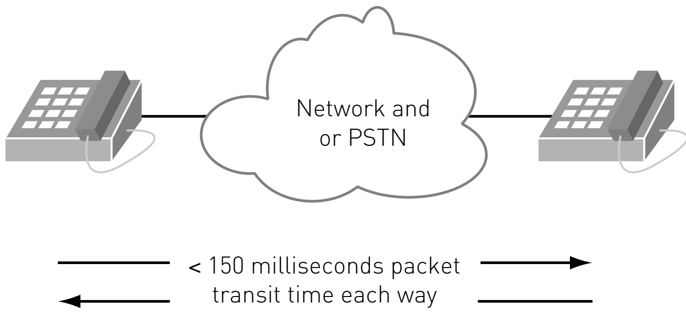

FIG.1 Packet transit times, including voice encoding and decoding, must be less than 150 milliseconds for acceptable QoS.

Latency Measurement

Essential to assuring low latency and acceptable QoS and SLA compliance, is the ability to make a reasonable measurement of the latency in the network. This requires adequate synchronization between measurement probes to obtain meaningful results. Some measurement solutions use hop-by-hop roundtrip metrics to build statistics on latency, jitter, etc. While this is informative, it simulates more of a piecemeal measurement approach rather than a phone conversation between two parties.

Measurement of end-to-end one way path latency better reflects a true QoS metric. An even better 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. Today the most common method for synchronization is by using software time transfer methods. A better solution however is 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.

Latency Management Before Over Positioning

VoIP offers the promise of converged of voice and data, cost savings through reduction of transportation costs, and new products and services. This promise hinges in part on the customer adoption rates of VoIP, which in turn hinges on acceptable levels of quality of service. In the absence of bulletproof methods to measure, monitor or assure quality, over provisioning bandwidth becomes the de facto solution. Aside from increasing costs, over provisioning only increases the probability, but does not guarantee the quality, of the transmission of VoIP traffic.

Network Management, Fault Diagnosis and Recovery

Most IT organizations are measured on their ability to maintain full flow network operations. This reliability requirement increases by orders of magnitude when you add business related voice traffic to the same network. It's one thing to delay email or suspend web access for a period time to fix a problem; it's an entirely different world to be responsible for the voice system. Any VoIP problem must be avoided, averted, or at worst case minimized in an effort to keep business critical voice systems running. To do this, one of the absolutely essential underpinnings is the accuracy of the server and router log files.

Every log file entry is time stamped. These time stamps establish the 'when' of an event and together they allow the ordering of events. Log files and subsequent reports allow you to use the log file data to identify root cause problems within your network. Since server logs are a compilation of information from different hosts, it is essential that the time stamps be correct. If not, you will have difficulty ordering events and troubleshooting root-cause problems. The more difficulty you have in identifying a problem the longer the QoS level of your VoIP system may degrade or worse yet, be non-operational.

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

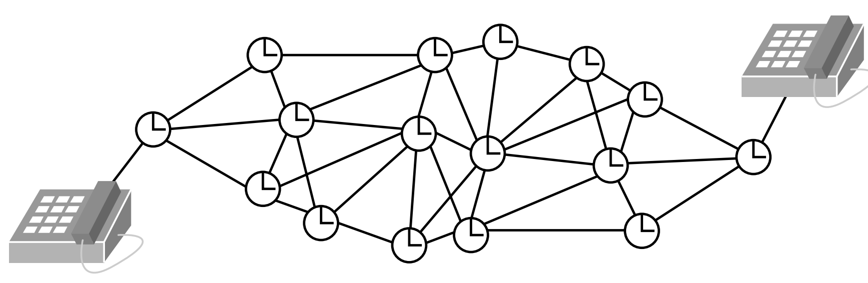

FIG.2 Time accurate server and router log files play a key role in troubleshooting and identifying root cause problems in complex VoIP networks.

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 NTS-200 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 it's embedded GPS receiver, and can synchronize thousands of clients on the network.

Call Detail Records Need Sync

No discussion regarding time on a VoIP network would be complete without mention of the obvious role time plays in billing. Call Detail Records, CDRs, provide information about call origination, destination and duration. Duration certainly includes the time stamp when the call was initiated and either the call duration or time the call was terminated. Billing integrity will rely on the underlying time accuracy of the VoIP CDR records. Without proper synchronization the CDR accuracy will falter and the billing system will inevitably come into question. This is particularly critical when the CDR information is shared between carriers and billing discrepancies require time consuming mediation.

Looking ahead, VoIP caliber networks hold the promise of many new services such as unified messaging, video conferencing, bandwidth on demand and other on-the-fly provisioning services. Time stamping will be essential in one form or another for all of the different billing schemes that will evolve in this competitive environment.

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.

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.

Sync Now or Sink Later

Synchronization is usually not the top priority when establishing a VoIP network. However, as soon as problems occur the true value of the synchronization system becomes very clear. QoS monitoring systems and network diagnostic programs will ultimately drive the requirement for synchronization across the network routers, servers and related devices. These systems rely on log file accuracy and integrity for their metrics. Without timestamp accuracy an unacceptable amount of time will be spent trying to resolve problems that could have been resolved more efficiently. Good practices deem synchronization important enough to incorporate it up front in system design rather than later when trouble occurs.

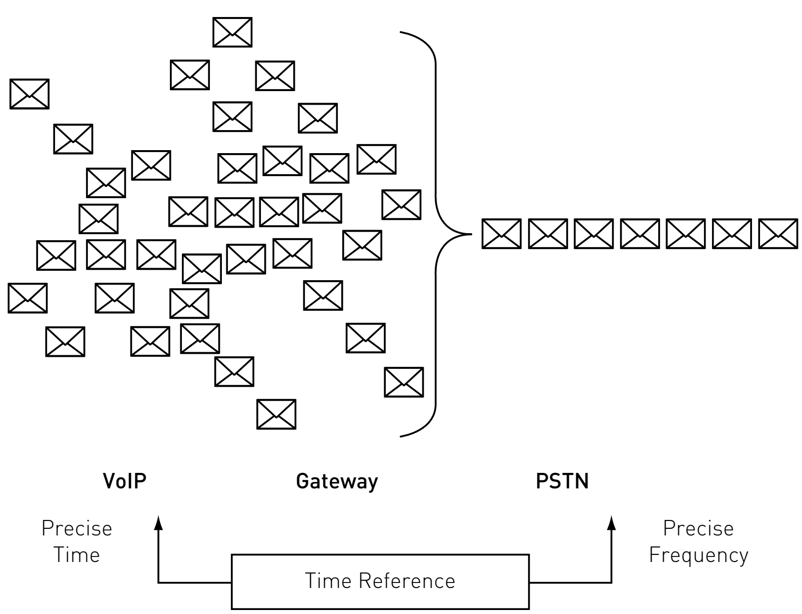

FIG.3 Precise time is required for VoIP; precise frequency is required for PSTN.

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