APPLICATION BRIEF

Symmetricom

NGN SERIES

Services and Applications Perspective

Timing and Synchronization in Next-Generation Networks

Abstract

Time and frequency alignment is critical for ensuring quality of service (QoS) for applications such as voice, real-time video, wireless hand-off, and data over a converged access medium. As telecom networks evolve from Circuit to Packet switching, proper synchronization (frequency and time) is needed for IP networks to achieve performance quality comparable to that of legacy circuit-switched networks.

Accurate time alignment (time-transfer or time-of-day) is needed to support QoS and traffic engineering, whereas frequency alignment (network synchronization) is required to minimize slips. In the event of slips, i.e., buffer overflow/underflow events, two options are available. One, appropriate for non-real-time applications, is to retransmit the packets/frames, which is equivalent to the reduction of bandwidth utilization efficiency.The other, applicable to real-time applications, where retransmission may not be an option, is to accept the loss and attendant degradation of service quality.

This paper addresses the impact of proper network synchronization on services and applications, the various technologies emerging to fulfill this need and also clarifies the distinction between frequency and timing synchronization.

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Introduction

Historically, synchronization in the circuit switched or Time Division Multiplex (TDM) network was introduced at the inception of digital networks, with the implementation of PDH and SONET/SDH technologies. The architecture of network synchronization has been well understood and globally standardized.

Today, synchronization networks are engineered with a hierarchical topology, distributing synchronization from a Primary Reference Clock (PRC) or Primary Reference Source (PRS) to Building Integrated Timing Supply (BITS) or Synchronization Supply Units (SSU) throughout the network.

All the network elements are synchronized either directly from a synchronization element (PRC/PRS or BITS/SSU) or indirectly through frame alignment of the TDM network (PDH or SONET/SDH).

As the lower layers migrate to "asynchronous" methods, such as asynchronous Ethernet, the synchronization link is lost and other methods for delivering a suitable synchronization reference to support existing and emerging services will be required.

Next Generation Networks (NGN) maximize commonality of technologies, and separates the service layer from the transport layer. Service content (voice, video and data) and signaling information will be delivered in the form of routed IP packets (layer 3 in the OSI model). In essence, layer 3 routing techniques use "store and forward" methods (i.e. queuing techniques) that are not dependent on network synchronization, per se, for error free delivery. However, if the underlying layers are not substantively error-free, error recovery methods (layer 4) will have to be invoked, which can impair the quality of real-time services. In essence, NGN synchronization, in the form of frequency and time of day, is required for transport, management and service applications.

The following table summarizes the accuracy requirements of the various services and application as defined in ITU and ANSI standards or by organizations and forums charted with ensuring QoS or interoperability of technology.

SERVICE	SYNCHRONIZATION	STANDARDS	USER EXPERIENCE
Voice			
T1 – 1.544Mb/s	< 32ppm	ITU G.703	Snap, Crackle, Pop
E1 – 2.048Mb/s	< 50ppm	ITU G.703	
Ethernet Best Effort	< 100ppm	IEEE 802.3	Delayed response
Video, Two way	< 50ppb	Under Study	Pixellation, waves, video freeze
Video, One way			
MPEG	< 500ppb	ITU H.222.0	Pixellation, waves, video freeze
HDTV	< 100ppb	Under Study	Pixellation, video freeze
IPTV	< 100ppb	Under Study	Pixellation, waves, video freeze
Wireless Sync			
2G - GSM	< 50ppb	ETSI TS 145.010	
3G - UMTS	< 50ppb	ETSI TS 125.104/5	Dropped calls, poor hand-off
Broadband Wireless	< 10ppm	IEEE 802.16	

TABLE 1 Synchronization and timing implications in next generation services

Voice

Information to and from a human consumer (audio and video) is analog in nature. Delivery of this information over a digital network necessitates the conversion from analog-to-digital (A/D) at one end and the reverse, digital-to-analog (D/A) conversion, at the other. If the conversion clocks for the A/D and D/A converters have a relative frequency offset, then impairments result (the phenomena is called the pitch modification effect, and is analogous to Doppler). With reference to Figure 1, the input analog signal, x(t), is converted into digital format and the resulting digital samples delivered across the network for conversion back to analog. Even if the network faithfully delivers all the samples intact, and in order, the resulting analog signal, y(t), will differ from the original analog signal x(t) if there is any frequency offset between the conversion clocks.

Converters for speech and voice-band signals typically use an (implied) 8 kHz conversion rate. When the converter is resident in a Class-5 Circuit Switch, the switch clock is traceable (in normal operation) to a primary reference source (stratum 1/G.811) and therefore accurate to 1×10^{-11}, which is an attribute that carries over to the conversion clock. Thus for a call between two central offices, the frequency offset is less than 2×10^{-11} and the pitch modification effect is indiscernible.

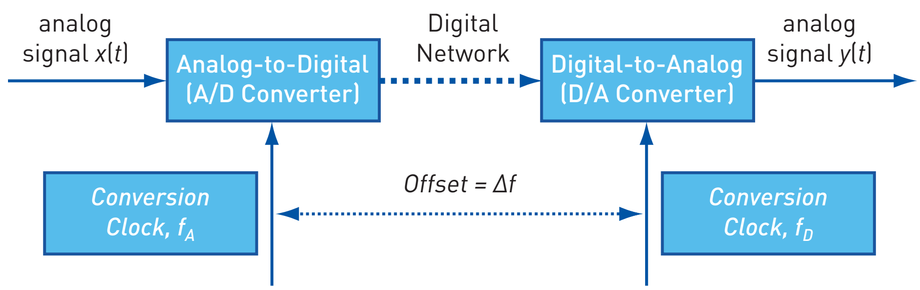

FIG. 1 Pitch Modification effect

For voice communications, the pitch modification effect is of less consequence than for voice-band communications (fax/modems). The frequency offset has the effect of altering the effective baud-rate. If the transmit baud-rate is B, then the effective baud-rate at the receiver is $B(1+\Delta f)$, where Δf is the frequency offset between the A/D and D/A converters, in fractional frequency units (such as ppm). Industry specifications for modems require that the transmit baud-rate must be accurate to 100ppm and the receiver tolerance must be 100ppm as well.

Consequently, any frequency offset introduced by the network will consume the performance margin. It is difficult to estimate the impact of frequency offset on the modem population at large, but it is estimated that to ensure 99.99% efficacy, the frequency offset should satisfy $\Delta f < 50$ppb. An equivalent requirement for the case of video signals is under study, but is likely to be of the same order of magnitude if not much significantly more stringent.

The need for synchronization is also present when two network elements interface with each other. In a TDM environment the interface is associated with a slip buffer to account for clock differences between the two devices. A traditional slip buffer has a size of 125 µs; if the time drift between the two clocks exceeds 125 µs a slip occurs, corresponding to either the deletion or repetition of 125 µs worth of data. For reference, a 1ppm frequency offset is equivalent to the accumulation of time drift at the rate of 1 ms/s and thus a slip would occur every 125 s. Therefore, in order to pass traffic reliably, network elements must be synchronized. Furthermore, the end-to-end link may traverse multiple networks and these timing islands must also be traceable to a common clock.

In Telecommunication networks, the following synchronization strategies defined in ITU G.810 and Telcordia GR-436-CORE have been in practice:

- PRS/PRC distribution, where all the primary reference source are considered "equivalent" since all of them guarantee an absolute frequency accuracy of better than 1×10^{-11}.
- Master-Slave architecture, where a clock hierarchy is used which each level of the hierarchy is synchronized with reference to a higher level, the highest level being the PRC. Distribution of a synchronization reference can be achieved by the physical transmission links through PDH (E1/T1 interfaces) and/or SONET/SDH (Optical interface).

When a service provider manages the entire network over which voice, video and data services will be deployed, maintaining accurate and reliable synchronization is under the complete control of the service provider. However, most service providers reduce operating costs by utilizing leased backhaul and in any case must still connect to legacy networks. These scenarios are important because, even though data connectivity may not be an issue, the synchronization chain may be broken.

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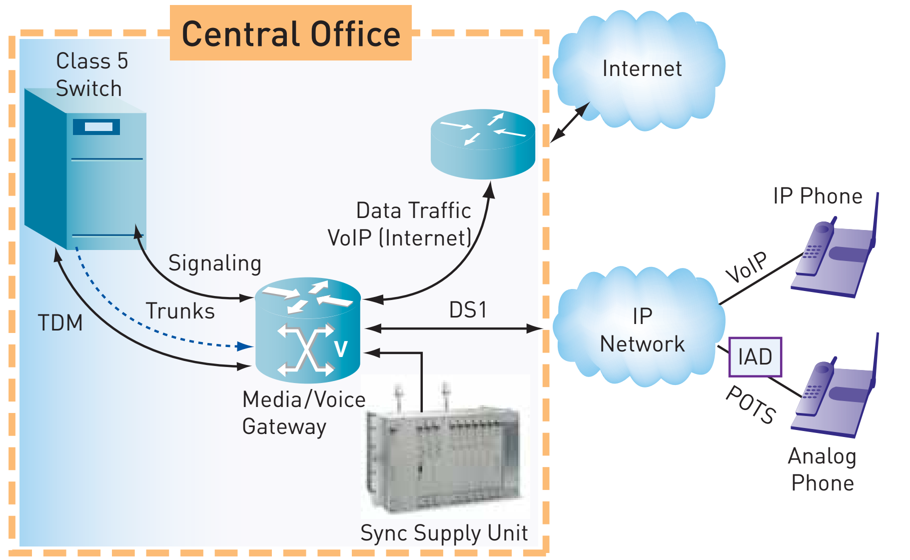
At all voice gateways, precise synchronization and timing not only mitigates slips, but also minimizes time delay variation that can make service sluggish. It is also important to note that at intersections where the IP and circuit-switched networks meet, time and frequency alignment ensures interoperability.

For example, consider the two primary types of Voice over IP (VoIP) a provider can make available. VoIP over the public Internet is a best-effort proposition, and users may experience a variety of quality issues. Alternatively, VoIP packets can pass through a gateway and over the PSTN. To maintain a quality of service normally associated with the (legacy) PSTN, the gateway must be synchronized to stratum 1/G.811 traceable reference, as depicted in Figure 2. For carrier-class voice-band services using VoIP, the A/D and D/A converters must be provided with an accurate clock (better than 50ppb) or must incorporate sophisticated "relay" functionality. It is possible that these converters will be in the customer premises integrated access device (IAD) and the service provider is responsible for ensuring the availability of clock reference.



Synchronization of gateway device that performs C2P function can be accomplished in two ways:

1. External synchronization reference from collocated clock source
2. Line/loop from PSTN

FIG.2 Synchronizing VOIP Media Gateway

According to Cisco's white paper "Cisco IP Telephony Clock Synchronization: Best Practices". Proper clock synchronization is critical to ensuring good performance, improving troubleshooting, and producing accurate call detail records (CDRs) for billing.

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Network time synchronization also plays an important role in keeping IP telephony networks operational and performing well. Accurate server and router log files are critical to IP telephony reliability. Every log file entry is time stamped to establish the time of an event and facilitate the ordering of events. Because server logs are a compilation of information from different hosts, it is essential that the time stamps be correct and accurate within milliseconds. If they are not, ordering events gets harder; troubleshooting root-cause problems becomes much more difficult; and keeping the IP telephony network operational becomes nearly impossible.

In addition, network time synchronization plays an important role in billing. CDRs are the primary source of billing information in an IP telephony environment; they provide information about call origination, destination, and duration. CDR duration includes the time stamp specifying when the call was initiated and either the duration of the call or the time the call was terminated. The integrity of IP telephony billing relies on the accuracy of the time stamps, which are measured in milliseconds. Without proper network time synchronization, accuracy will quickly decline and the billing system will be questioned.

Circuit Emulation

For many years telecommunication carriers and their competitors, independent operators and most recently cable operators, have slugged it out in the battle for the high margin private line business customers. The majority of private line customers use leased E1/T1 facilities for voice and data applications.

Telecommunication carriers, as part of a cost reduction effort, have started to deliver E1/T1 private line services over lower cost facilities (typically Ethernet) that are transparent to the end users. This concept is referred to as Circuit Emulation Services (CES) and uses lower cost networking technologies to convert the E1/T1 circuits to data, transport the traffic as data streams, and then reconstruct the E1/T1 circuits. These methods may indeed lower operators overall costs and will be transparent to the end users as long as the Service Level Agreements (SLA) are not compromised and the end user is satisfied with the performance. One performance aspect of CES is related to synchronization and the ability to reconstruct an E1/T1 that meets the jitter and wander performance criteria as specified in ITU-T Recommendation G.8261. Tables 2 and 3, extracted from ITU-G.8261 indicate the wander requirement of a CES segment when it is part of the end-to-end transport that may include other CES segments and/or SONET/SDH segments according to the reference model in ITU-T Recommendation G.824 (T1) and G.823 (E1).

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Observation Interval τ in seconds	MRTIE in µs
$0.05 < \tau \leq 0.2$	$10.75\ \tau$
$0.2 < \tau \leq 32$	9*0.24 = 2.15
$32 < \tau \leq 64$	$0.067\ \tau$
$64 < \tau \leq 1000$	18*0.24 = 4.3

Note: for the asynchronous confiduration, the maximum observation interval to be considered is 80 seconds.

The specification between 80s and 1000s for the asynchronous interfaces is for further study.

TABLE 2 2048 kbit/s interface wander limit

Observation Interval τ in seconds	MRTIE in µs
$\tau \leq 0.1$	No requirement (see note)
$0.1 < \tau \leq 0.47$	$4.5\ \tau$
$0.47 < \tau \leq 900$	2.1
$900 < \tau \leq 1930$	2.33*10e-3 τ
$1930 < \tau \leq 86\ 400$	4.5

Note: this region is covered by jitter requirements.

TABLE 3 1544 kbit/s interface wander limit

In CES applications the boundaries between the circuit and packet segments are referred to as Inter-Working Functions (IWF) and one of the issues being addressed by the standards bodies is what accuracy and stability the clocks in the IWF components need in order to support the standards. The most recent contribution from standards bodies addressing this issue is in ITU recommendation G.8261. The ATIS committee OPTXS-SYNC is currently generating a Technical Report, "Synchronization of Packet Networks", that provides details and explanations regarding the various aspects of synchronization in packet networks and, in particular, circuit emulation considerations.

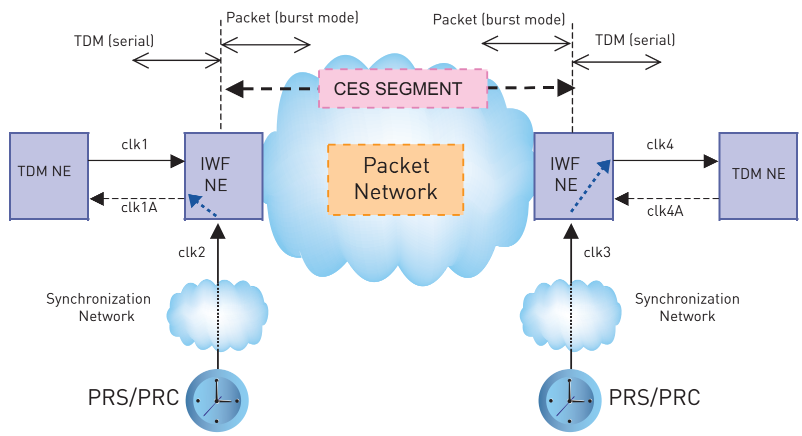

FIG. 3 Network Synchronous Operation

Synchronization Methods for CES

The standards bodies have described methods that can potentially be used for synchronization schemes that attempt to comply with jitter and wander specifications for CES. The following diagrams from recommendation G.8261 illustrate four possible synchronization schemes for Inter-working functions. For convenience, only one direction of transmission is shown.

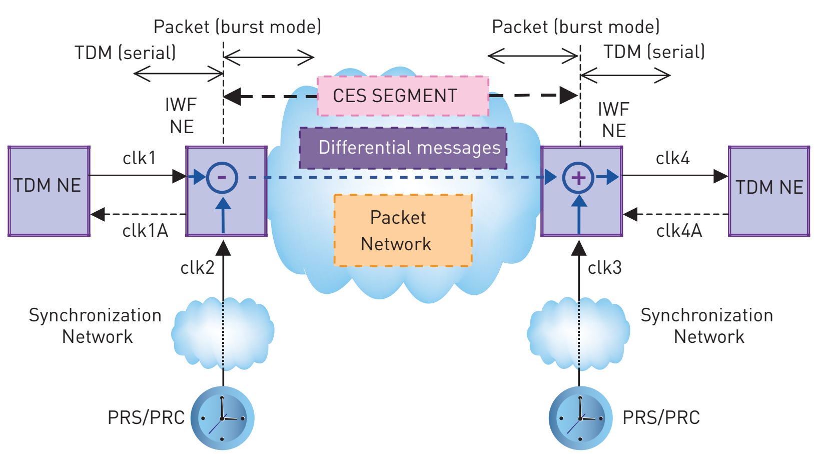

FIG. 4 Differential Method

Network Synchronous Operation

This method refers to full network-synchronous operation by using a PRS/PRC traceable network derived clock or a local PRS/PRC as the service clock. In effect, the TDM signal is "retimed".

The clock accuracy of ingress TDM clock (clk1) must be PRS/PRC traceable, otherwise the use of a network clock reference in the egress IWF (i.e. clk3) will cause jitter buffer overflow/underflow events in the egress IWF.

Differential Methods

The principle of operation of any differential method is based on the availability of "equal" clock references at the ingress and egress IWFs. The difference between the service clock and the reference clock is encoded and transmitted across the packet network.The service clock is recovered on the far end of the packet network making use of the "equal" reference clock. Synchronous Residual Time Stamp (SRTS) is an example of this family of methods. Differential methods can support the plesiochronous circuit timing (also known as asynchronous circuit timing) mode whereby the TDM service clock can have an offset from PRS/PRC provided it is within defined limits.

Correct timing in the output TDM signal implies that the clocks generating the TDM signal (clk1) and retiming (clk4) the TDM signal must have the same long term frequency (or within the PRS/PRC limits) otherwise jitter buffer overflow/underflow events will be generated in the egress IWF and the destination TDM NE may experience slips. It is easy to show that wander (and frequency inaccuracy) in the egress TDM signal (clk4) is directly related to the relative wander between the reference clocks clk2 and clk3. Figure 5 shows that the references come from two distinct PRS/PRC units though obviously they could be the same. If the synchronization trail between clk2 and the PRS/PRC and that of clk3 and the PRS/PRC has a "common" node, that node could be in holdover without adversely impacting the differential mode of operation.

Adaptive Methods

In Adaptive Clock Recovery (ACR) methods, timing is recovered based on the inter-arrival time of the packets or on the fill level of the jitter buffer. Adaptive methods can support the plesiochronous circuit timing mode whereby the TDM service clock can have an offset from PRS/PRC provided it is within defined limits.

If the transit time across the packet network of the packets varies, also known as packet delay variation (PDV) or time-delay variation (TDV), the clock recovery process is affected. In particular, PDV, on a short-term basis, is indistinguishable from a change in the phase/frequency of the service clock and/or the local oscillator. Consequently ACR implementations require high quality oscillators and apply filtering corresponding to bandwidths of the order of milli-hertz (mHz) (time constants of the order of 1,000s). However, if a network clock reference is not available, then ACR is the only available method for service clock recovery.

There are several causes of delay variation including the following that are covered in G.8261:

- Random delay variation (e.g. queuing delays)
- Low frequency delay variations (e.g. day/night traffic patterns)
- Systematic delay variation (e.g. transmission window)
- Routing changes (e.g. network re-configuration)
- Congestion effects (e.g. network overload)

Since the performance of adaptive clock recovery is very dependent upon PDV, it is recommended for use only when the PDV can be tightly controlled.

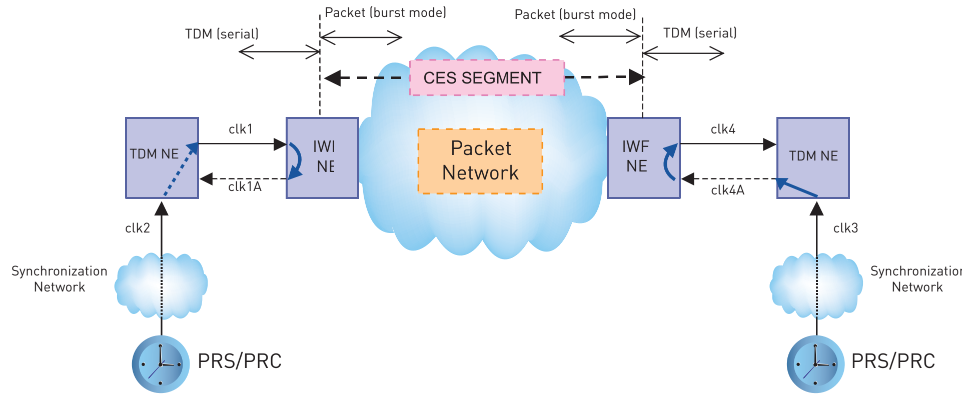

FIG. 5 Reference Clock at the End Systems

Reference Clock available at the TDM end systems

When the reference clock is available at the TDM end systems, this is a trivial case, since the CES segment is not responsible for delivering circuit timing across the packet network. Therefore there is no need to recover the timing. The use of loop timing in the IWF on the TDM interface is an example of the implementation of this method. An example when this scenario might apply is when two PSTN domains are connected via a packet network.

The following table compares the four modes of operation described above in terms of reliability and performance.

Method	Performance	Reliability
Synchronous	High	High
Differential	Medium	Medium
Adaptive	Low	Low
Reference Clock	N/A	N/A

TABLE 4 Synchronization methods for ces

Careful consideration should be given to the synchronization schemes for circuit emulation due to the associated variables. The requirements in G.8261 that limit any circuit emulation segment or island to a deviation of less than 4.5 microseconds over 24 hours of the total end to end wander budget will be extremely difficult to achieve without PRS/PRC traceability at the IWF.

Mobile Networks

Wireless operators are evolving their services in the following aspects

- Technology migration from 2G (GSM and CDMA) to 3G (UMTS and CDMA2000), increasing the mobile bandwidth for additional services such as video, data and gaming.
- Network Infrastructure migration from circuit to packet, for the interconnection of all mobile network elements.

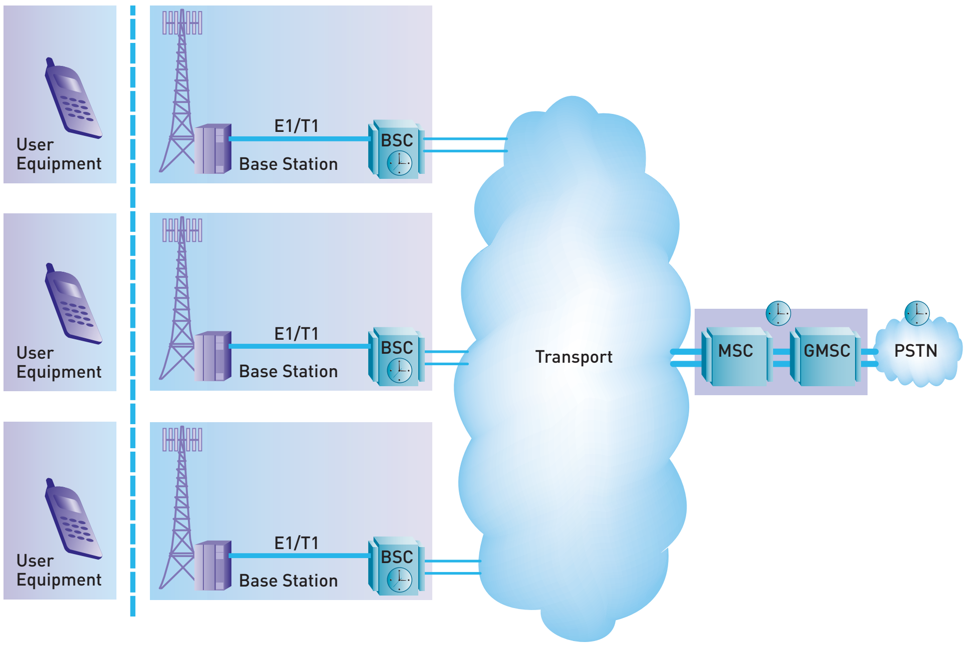

FIG. 6 GSM Mobile Network

Current mobile networks rely on TDM transport to interconnect Mobile Switching Centers (MSC) to the Base Station Controller (BSC) sites and from there out to the entire arrangement of base stations (BTS). Base stations rely on very tight synchronization and must hold a carrier frequency accuracy of +/- 50 ppb over the ten-year service life of the equipment. If individual base stations drift outside the specified 50 ppb limit, mobile hand-off performance will decay, resulting in high dropped-call rates, impaired data services, and lost customers.

GSM base stations have traditionally derived their long-term frequency accuracy from the TDM T1/E1 leased line backhaul facility. Without the recovered clock to hold the oscillator on frequency, the base station drifts out of specification in a matter of months requiring costly service calls to manually adjust the oscillators.

Network infrastructure for mobile operators is rapidly moving to Ethernet technology. This offers increased backhaul capacity required for deployment of high bandwidth data services with the cost advantage of IP transport. However, the move to Ethernet backhaul eliminates the option for base stations to recover clock frequency from the network.

In order to maintain consistent quality connectivity, base station equipment manufacturers and backhaul service providers must take synchronization and timing into consideration. As with CES, new methods of synchronization are required to meet the rising expectations of next generation mobile users.

Video

Video may be distributed in the form of broadcast content or video on demand, either in "real time" (a live event) or as an archived stream. Each of these video flavors may have different needs as to quality, bandwidth and resiliency. For instance, viewers of a corporate presentation on a PC may be quite tolerant of a little jitter, whereas consumers of a live sporting event will find the slightest loss-of-frame problems excruciating.

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The requirement in some video applications is to use the system clock to generate a sub-carrier that is in full compliance with the NTSC, PAL, or SECAM specifications, which are typically even more stringent than those imposed by typical television receivers. For example, the SMPTE specification for NTSC requires a sub-carrier accuracy of 3 ppm, with a maximum short-term jitter of 1 ns per horizontal line time and a maximum long term drift of 0.1 Hz per second. (ITU H.222.0)

The motion picture experts group (MPEG) is an ISO standards group dealing with video and audio compression techniques that require the encoder and decoder clocks be synchronized for video playout. The encoder clock reference is transmitted to the decoder in the form of a timestamp known as Program Clock Reference (PCR). This is embedded in a 188 byte MPEG transport stream packet and is utilized at the decoder or the set top box to align its clock with the encoder clock. The PCR tolerance is defined as the maximum inaccuracy allowed in received PCRs. This inaccuracy may be due to imprecision in the PCR values or due to PCR modification during re-multiplexing. It does not include errors in packet arrival time due to network jitter or other causes. The PCR tolerance is ± 500 ns. (ITU H.222.0)

Time-of-day synchronization is also an important area of consideration for IPTV service.

MPEG assumes that the network delay is constant. Traditionally, MPEG transport streams were encapsulated into ATM cells and carried over SONET/SDH networks that are near constant delay and introduce very little jitter (time-delay variation). Therefore, this MPEG synchronization mechanism worked effectively in legacy networks. In case of IPTV, these MPEG frames are now encapsulated into IP packets and delivered as an IP flow from encoder (or video server) to the decoder (usually in a set-top-box). These IP flows are now subject to the jitter that is an inherent characteristic of IP. The IP induced jitter leads to PCR jitter and has been known to impact video picture quality in IPTV field trials and deployments.

Service providers distribute large volumes of video across backbone networks to an increasing number of video distribution sites (often referred to as hubs or central offices), which process the video for delivery. Delivery to the end viewer occurs over various types of last-mile technologies including terrestrial radio, hybrid fiber coaxial (HFC), xDSL, and FTTP networks.

Figure 7 provides an example of a network designed to support video services.

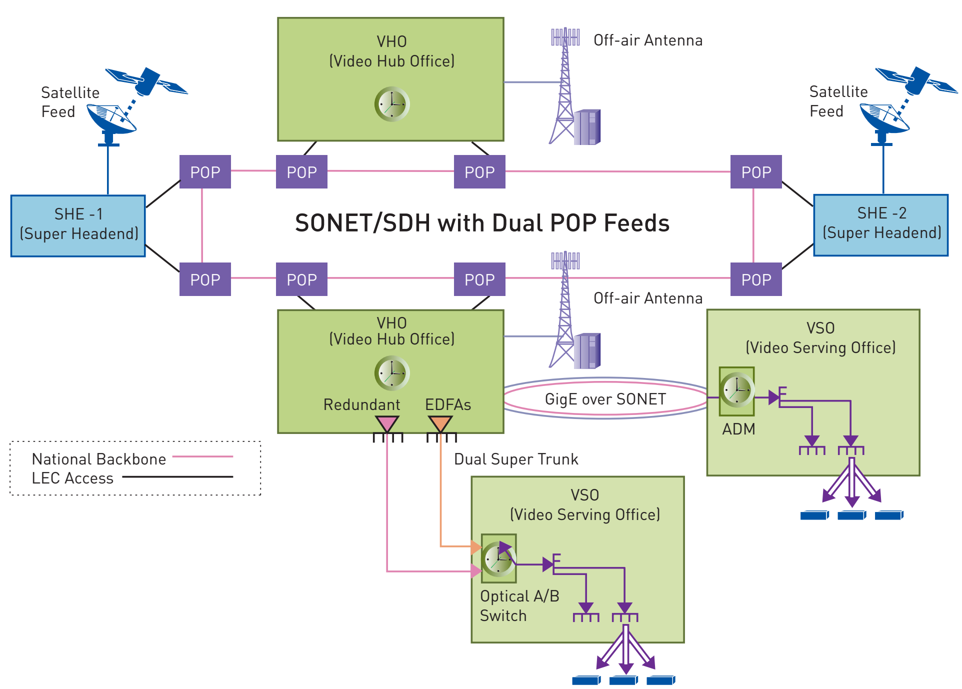


FIG. 7 Synchronization implications for video services (an example)

In cable networks, as an additional safeguard, equipment is often deployed at the edge to retime the incoming MPEG transport stream to eliminate any PCR jitter introduced by transport. This process is known as PCR reinsertion. Such retiming mechanisms could be applied to IP flows, but this would introduce a tremendous cost in the network. Therefore, it is extremely important that the SONET/SDH network be tightly synchronized to eliminate pointer movement events since these introduce PCR jitter for the video flow.

In the core and metro network segments, jitter can be reduced with tight synchronization. However, considerable network jitter may still be induced in the converged access medium, with its bandwidth constraints. The access network becomes the bottleneck when it comes to delivering voice, video and data in the shared "pipe". Relying on over-provisioning to counter poor synchronization is not an option as is the case with core and metro networks. Prioritization schemes help considerably when one of two streams competing for the available bandwidth is non-real-time. However, the triple play bundled service offering includes video, which is a high-priority, high-bandwidth stream and VoIP another real-time service. Moreover, while data is non-real-time, it is still a variable bit rate stream and even though of a lower priority, is still capable of inducing jitter. Therefore, tight synchronization as used in the core network also needs to be applied to the access network. This requires that the access network including all Network Elements involved such as a PON or IP DSLAM be synchronized with a PRS/PRC traceable frequency reference.

Time-of-day synchronization is also an important area of consideration for IPTV service. Time synchronization plays an important role in content protection (digital rights management, or DRM). For example, the license to view certain content may be associated with time. Some DRM solutions employ the notion of secure clocks where the clock on a media device is provided with a time reference from a secure source. This clock reference cannot be changed except by the authorized source. Thus, in order to deliver the network timing reference to the customer media device or "end-point", all NEs involved with the delivery of the MPEG streams to the end-viewer set-top box, will require a PRS/PRC traceable timing reference.

Every network element has three principal modes of operation from the viewpoint of (frequency) synchronization. It can free-run, relying on its own internal oscillator for frequency accuracy; it can derive its frequency reference from an incoming (traffic bearing) signal; or it can accept an external reference.

IPTV builds upon broadcast video by enabling providers to insert targeted advertising into the video stream based on actual customer viewing patterns. Instead of all customers within a region watching the same video stream broadcast from the head end, local/personalized content, both programming and advertising, can be inserted into the stream. However, in order to achieve this, time of day synchronization is necessary on a network-wide basis.

IPTV has several attributes whereby fairly tight timing tolerances are required. The interactive nature of IPTV and the use of Internet Protocol to deliver these features necessitates that network jitter be constrained and a network wide synchronization of time is available. Good synchronization can considerably reduce network jitter and ensure consistency of time throughout the network thus, enabling suitable end-user quality of experience. As IPTV is rolled out on large scale and full interactive functionality becomes available, service providers and vendors are considering additional timing and synchronization requirements. The ATIS IPTV Interoperability Forum has recently published timing and frequency synchronization requirements for IPTV (Ref ATIS 0800001, ATIS0800002) so that IPTV service can be deployed seamlessly over any type of network infrastructure, media equipment and end user device."

Data

Transfer of non-real-time traffic (data) does not require precise synchronization. Applications such as file transfer and e-mail, for example, represent the type of transactions that IP networks were designed for. However, in the context of the Triple Play, the data component can introduce impairments in the other (real-time) services being provided over the converged medium.

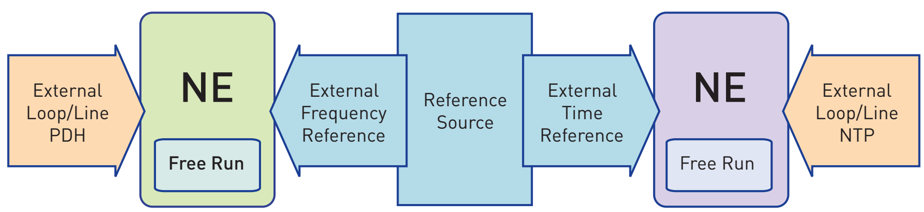
For example, even if data packets are assigned a low priority for transmission, once they are in the pipe they force any high-priority packet to "wait". Also, it is not uncommon for such data packets to be large. The packet delay variation introduced in a real-time stream because of these "low-priority" packets can be substantial, especially if the link bit-rate is not large, as would be the case in the "last mile". For example, a 1500-octet packet would hog a 10 Mbit/s link for 1.2ms; a real-time stream may experience a packet delay variation of 1.2ms as a result.

Synchronization and Timing

Every network element has three principal modes of operation from the viewpoint of (frequency) synchronization. It can free-run, relying on its own internal oscillator for frequency accuracy; it can derive its frequency reference from an incoming (traffic bearing) signal; or it can accept an external reference. Likewise, for time of day it can "free run" (requiring a routine, maybe infrequent, operator intervention to "set" the time); it can derive its time from a server (e.g. NTP); or it can have a dedicated feed that provides time (and frequency). These modes are depicted in Figure 8.

While both DSL and PON based networks provide increased bandwidth capacity, they both impact the Central Office (CO) in different ways. As DSL rates increase, the distances served decreases. For example, ADSL2 drops must be less than 5,000 feet, down from 12,000 for regular ADSL. This means that more MSANs (Multi-Service Access Nodes) will be deployed in remote locations than in the CO. OLTs, on the other hand, will tend to be located in the CO because of the ability of fiber to carry signals over longer distances.

Since network elements collocated in the CO can share synchronization signals, a single PRS/PRC feed can service an entire cluster of MSANs or OLTs via deployment of a BITS/SSU.



Frequency	Accuracy	Time
Stratum 1	High	Stratum 0
Stratum 2		Stratum 1
Stratum 3E		Stratum 2
Stratum 3		Stratum 3
Stratum 4	Low	Stratum 4

FIG. 8 Stratum levels for frequency and time

Remote terminals will need to be synchronized either indirectly through a network feed or directly using a local synchronization system, depending upon the application. If the network backhaul is controlled then the synchronization chain can be maintained. This is especially important if packet-based two-way time-transfer methods are used to deliver the synchronization (time and frequency) reference to the remote unit (slave) from a CO-based source (master). The quality of time-base transfer depends very heavily on the packet delay variation between the master and the slave; the time transfer accuracy is affected not just by the packet delay variation, but also the asymmetry in transit delay in the master-slave and slave-master directions.

If the network backhaul is deployed by the service provider, physical layer methods for delivering time-base reference (e.g. SONET/SDH line clock) are always an option. If the network backhaul is over a leased or Ethernet network, then there are two issues that arise. First is that the physical layer delivery of a time-base reference is not an option. Two-way time-transfer (packet-based) methods are required. However, in order to ensure adequate performance of such methods, the delivery and latency of packets must be guaranteed to within tight tolerances, a situation that may not always be true.

Table 5 lists some of the existing and emerging technologies for delivering a synchronization reference to network elements. For equipment located within the Central Office (an intra-office scenario), the recommended method is to deliver a reference over a dedicated wire pair, an extension of the methods deployed today. This has been adopted by the North American cable industry in the form of the DOCSIS Timing Interface (DTI). All equipment within the head-end will be synchronized to the DTI-server utilizing the DTI method. ITU-T Study Group 9 is developing the Draft Recommendation J.dti which will be applicable to cable deployments worldwide. The adaptation of DTI for telecommunications applications is referred to here as the Telecommunications Timing Interface (TTI).

Ultimately, the quality of synchronization in the network directly impacts the quality of services that the end-user will experience.

Attributes	Synchronous Ethernet	NTP	IEEE-1588	Telecom Timing Interface
Evolution	Proposed	Computer-based Applications	Designed for industrial automation	Layer 1 two way time transfer
Parameters	Frequency Only	Time	Time	Time and Frequency
Accuracy	$\Delta f < 10^{-11}$	Milliseconds	Microsecond	Nanosecond
Transfer Layer	Layer 1	Layer 3	Layer 2	Layer 1
Manageability	No, Client is autonomous	No, Client is autonomous	No, Client is autonomous	Yes
Traceability	Under study	No, Transfer error not deterministic	Under study	Yes
Client Complexity	Simple	Complex	Complex	Simple

TABLE 5 Technologies associated with the distribution of timing

For synchronizing equipment that is deployed in a remote environment (outside the CO) there are several possibilities.

- Synchronous Ethernet has been standardized in G.8261 as one method to deliver a (frequency) reference over the physical layer in the case of GigE and 10GigE (and is applicable to 100 Mbit/s).
- NTP is a technology that is widely deployed today for synchronizing clocks in PCs, routers, servers, and so on. It is a layer-3 packet based method whereby the client queries (possibly multiple) servers in order to establish its own internal wall-clock. However, traditional NTP is "best-effort" and, for deployment in a telecommunications service provider network, requires special considerations to be "carrier class". It should be noted however, that NTP is a very rich protocol and has proven to "work" in the harshest of con ditions (over the Public Internet). Deploying NTP, even in its current form, in a controlled environment is likely to provide excellent results.
- IEEE-1588, or Precision Time Protocol (PTP), is a recently ratified standard developed to synchronize clocks in various devices in an Ethernet local area network environment. The objective of IEEE 1588 version 2 is to synchronize remote sites ("Slave clock") from the CO-based server ("Grand Master clock").

Conclusion

Synchronization plays a critical role in the successful rollout of next generation networks. Without an accurate and precise synchronization traceable to a PRS/PRC, transfer of traffic between networks can suffer latency and losses. Additionally, broadband access networks require synchronization between nodes within a network to minimize guard bands and maximize bandwidth utilization. In NGN architectures, all services are packetized and carried over a "converged" network. Unfortunately such transport may not be very amenable to carry the appropriate synchronization reference required at the end-points. Special techniques, such as physical layer distribution for synchronization, and layer-2/layer-3 (i.e. packet-based methods) for delivering a time-of-day and/or frequency reference will be needed.

Ultimately, the quality of synchronization in the network directly impacts the quality of services that the end-user will experience.

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