The Time Deviation in Packet-Based Synchronization

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***Abstract*—The telecommunications industry has used the time deviation (TDEV) very effectively for specifying network equipment clock performance as well as the performance of timing signals generated by Central Office equipment such as primary reference clocks and building integrated timing supplies (BITS) and synchronization supply units (SSUs). We discuss here the development of TDEV, and the variations of TDEV motivated by the advent of packet-switching and the steady transformation of the telecom network from circuit-switched-based to packet-switched-based. We illustrate these with simulation of the performance of the precise time protocol (PTP) across a packet-switched network. We then apply published methods to automatically determine noise types, and use these to predict time dispersion from a master clock for a slave clock using these PTP packets to stay synchronized. The result shows how TDEV and the other deviations provide an extensive array of tools for telecom networks, as well as for general time and frequency applications.**

***Index Terms*—Allan variance, packet networks, synchronous digital hierarchy (SDH), synchronous optical network (SONET), time deviation (TDEV), time variance (TVAR).**

I. Development of TDEV

In the early 1990s, telecom standards groups were working on standards for the synchronous optical network (SONET) and the synchronous digital hierarchy (SDH). One of these was T1X1.3, a subcommittee of Committee T1 [now Alliance for Telecommunications Industry Solutions (ATIS)]. They were interested in a simple metric that would provide information about the spectral power of synchronization noise. One member was familiar with the Allan variance. He came and talked with Dave Allan, who then, along with his colleagues, developed the time variance (TVAR) and the time deviation (TDEV) [1], as

$$\mathrm{TDEV} = \left(\tau/\sqrt{3}\right) \cdot \mathrm{MDEV} \tag{1}$$

in terms of the modified Allan deviation (MDEV). Note that the deviations MDEV and TDEV are the square-roots of MVAR and TVAR, respectively. For reference, the Allan variance (AVAR) and modified Allan variance (MVAR) are defined on a measurement data set $\{x(n);\ n = 0, 1, 2, \ldots, (N-1)\}$ of N samples taken every τ_0 (s) as

$$\mathrm{AVAR}(\tau) = \frac{1}{2} \cdot \left(\frac{1}{\tau^2}\right) \cdot \left(\frac{1}{N-2n-1}\right) \cdot \left(\sum_{i=0}^{N-2n-1} \left(x(i+2n) - 2x(i+n) + x(i)\right)^2\right)$$

$$\mathrm{MVAR}(\tau) = \frac{1}{2} \cdot \left(\frac{1}{\tau^2}\right) \cdot \left(\frac{1}{N-3n+1}\right) \cdot \sum_{j=0}^{N-3n} \left(\frac{1}{n} \cdot \sum_{i=j}^{n+j-1} \left(x(i+2n) - 2x(i+n) + x(i)\right)\right)^2$$

where $\tau = n\tau_0$ is the observation interval.

Later, as the total variance was developed [2], total TDEV was also published [3]. These metrics allow better estimates of the variances at long averaging times, by using a maximum of degrees of freedom available in the data.

There were a few issues that were addressed by TDEV that made it a desirable metric for telecom. The usual noise types considered in time and frequency systems are the power law types associated with white, flicker, and random walk phase modulation (PM), and white, flicker, and random walk frequency modulation (FM). Note that white FM is identical to random walk PM. Synchronization noise types in telecom networks, much like noise in measurement systems, are predominantly white, flicker, and random walk PM. Thus, the first requirement was the need to discriminate between white and flicker PM (FlPM). The Allan deviation (ADEV) did not do this, but the MDEV had already solved this problem. However, it is easier visually to see the break point on a log–log plot between a downward/upward slope and a flat slope. But the slope change of MDEV on a log–log plot between white and FlPM is from a $-3/2$ slope to a -1 slope, and the change in slope from flicker to random walk PM is from a -1 slope to a $-1/2$ slope. With TDEV proportional to the averaging time τ, times MDEV, the slopes of TDEV on a log–log plot for white, flicker, and random walk PM are, respectively, $-1/2$, 0, and $+1/2$. This helps significantly to see the averaging times where the changes occur between the different noise types. The scaling factor of 1/3 for TVAR made the value of TDEV at the minimum τ value equal the standard deviation in the case of white PM.

II. TDEV in Packet-Based Synchronization

The first use of TDEV in telecommunications was as a stability measure for clocks used in synchronous transmission schemes such as the SONET or SDH. In particular, ITU-T Recommendations G.811, G.812, and G.813 provide performance masks based on TDEV. T1X1.3 issued the T1.101 (North American) standard for telecom clocks. This was revised and reissued in 2006 as ATIS-0900101.2006.

With the advent of packet networks using Ethernet for transmission (Layer 1 and Layer 2), the need for the ability to deliver synchronization over the transmission medium remained, and to this end, the telecom industry developed the notion of

Manuscript received May 29, 2015; accepted September 16, 2015. Date of publication October 27, 2015; date of current version April 1, 2016.

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Digital Object Identifier 10.1109/TUFFC.2015.2495011

synchronous Ethernet. ITU-T Recommendation G.8262 deals with the Ethernet equipment clock (EEC) and for compatibility reasons made the EEC essentially equal to the SONET/SDH counterpart, the synchronous equipment clock (SEC) addressed in G.813, particularly from the viewpoint of wander. The jitter requirements for Ethernet transmission follow that of conventional Ethernet.

In addition to delivering (frequency) synchronization, syntonization, over the physical layer, it became clear that packet-based methods were required to carry frequency synchronization, particularly when the deployed equipment utilized legacy methods and did not support synchronous Ethernet. Furthermore, the need for delivering synchronization references that could support time transfer (in addition to frequency transfer) became apparent. The ITU-T Recommendations G.826x series cover the delivery of frequency synchronization over telecom (packet-based) networks and the G.827x series address delivery of phase/time synchronization in packet networks [4].

An essential distinction between traditional circuit-switched networks and packet-based networks is the type of multiplexing that is inherent in the transmission schemes. Circuit-switched networks employ time-division multiplexing (TDM), where traffic is assigned to well-defined time-slots in a well-defined frame structure of fixed frame size. This resulted in traffic channels that had a fixed end-to-end delay (within limits). Information channels such as DS1/E1 could, in principle, be used to carry frequency synchronization, the only “clock noise” added being that caused by physical layer effects and “bit-stuffing,” the practice of inserting bits in a stream to match the frequency of a multiplexed stream’s clock. Bit-stuffing multiplexing schemes were designed such that the additive clock noise was high-pass in nature and had significant components only at high(er) Fourier frequencies. Such clock noise can be adequately filtered out using PLLs with reasonably narrow bandwidths (e.g., the SEC clock bandwidth is between 1 and 10 Hz).

In contrast, packet-based transmission utilizes statistical multiplexing. The traffic of any information channel is packetized into individual blocks of information. Transmission of a packet in a network element is done on a packet-by-packet basis and the information-bearing packet is actually transmitted onto the wire at the earliest possible time based on the priority control mechanisms employed in the network element. From a timing standpoint, the impact of statistical multiplexing is that the delay experienced by any one packet of a stream could differ from the delay experienced by another packet of the same stream. This variable transit delay phenomenon is referred to variously as *transit delay variation*, *time delay variation*, and *packet delay variation (PDV)*.

TDEV/MDEV as a measure of clock stability has been well known in telecommunications and metrology circles since the 1990s. The advent of packet-based synchronization methods has shown that TDEV can be used as a stability measure to characterize the behavior of networks, not just clock signals. Several variations of TDEV have been defined to establish network properties by analyzing the behavior of PDV.

In order to explain how TDEV can be used to measure network (stability) performance, the notion of a packet timing signal is introduced. Packet-based synchronization utilizes time-stamped packets that traverse the network between master (or server) device and slave (or client) device. As will be shown, a one-way flow can be used to provide frequency synchronization and a two-way flow can be used to provide time/phase synchronization.

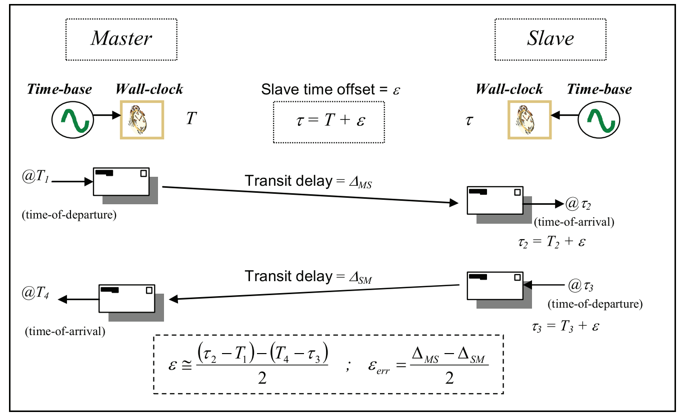

Fig. 1. Principles of timing transfer over packet networks.

A. Packet-Based Synchronization

The precision time protocol (PTP—also known as $\mathrm{IEEE} - 1588^{\mathrm{TM}}$) [5] and the network time protocol (NTP) [6] are both packet-based timing methods for telecommunications applications. From a time-transfer perspective, the two methods are identical in principle. Both protocols specify mechanisms for communicating timing information comprising the time-of-arrival and time-of-departure of designated packets and both provide specific formats for the time-stamps used to convey this timing related information. The differences between PTP and NTP are related primarily to their different roots and original use cases.

Synchronization of slave-to-master necessitates that the length of one second be the same (or nearly so) in both devices. This implies that the two are nearly syntonized (aligned in frequency). Syntonization implies that the reference clock values progress at almost the same rate, and thus the difference from the reference time, or the time offset, will be constant over the measurement interval. This time offset (ε) can be estimated by measuring the transit delay in the two directions and “splitting the difference,” i.e., there is an implicit assumption that the delay is the same in the two directions. Any asymmetry will result in an error in the offset estimate ($\varepsilon_{\mathrm{err}}$). Fig. 1 indicates the calculation in the case of a single exchange of packets. In practice, a flow of timing packets is set up between the devices to address PDV and local oscillator drift.

PDV is the variation in transit delay, i.e., the transit time $\Delta_{\mathrm{MS}}(\Delta_{\mathrm{SM}})$ is not the same from packet to packet. This variation is a principal contributor of (variable) error in situations where a slave is synchronizing (frequency and/or time) to a master over a packet network. Network asymmetry, both fixed and variable, contributes to time error in synchronization. By formulating the PDV as a time error sequence, it enables the use of metrics, such as TDEV, to evaluate the performance of the network from the viewpoint of suitability for purposes of packet-based synchronization. For example, frequency synchronization can be achieved using one-way flows, either "forward" (master-to-slave) or "reverse" (slave-to-master). The TDEV metric provides guidance as to which direction is better, as well as guidance on the quality of synchronization that can be achieved.

In the following discussion, it is assumed that the network is being monitored and that the measurement clocks at the two ends are synchronized by some very accurate and precise means, such as the use of a global navigation satellite system (GNSS) [e.g., a global positioning system (GPS)] reference. While there are innumerable ways in which a GNSS receiver can have errors, for our purposes, we assume the system is well-calibrated, operating correctly, and receiving good satellite signals. That is, in the context of Fig. 1, there is virtually no offset between the master and slave reference clocks ($\varepsilon = 0$), whereas the packet rates in the upstream and downstream flows can be different, for this discussion they are assumed equal. Denote the packet rate by $f_p = 1/\tau_p$, where τ_p is the (nominal) inter-packet interval. In telecom applications, f_p can be as high as 128 Hz (for 128 packets/s). For reference, such a configuration of GNSS+PTP has been proposed in the ITU-T for timing wireless LTE base-stations and is referred to as an assisted partial timing support application (see [9]).

In the forward direction, packets leave the master with a long-term mean spacing of $\tau_p = 1/f_p$. From a signal processing perspective, the sampling rate is f_p and an arbitrary mathematical-time origin for describing the times of departure from the master can be chosen. With this choice of time origin, the kth packet departs the master at time $t = k \cdot \tau_p$. In practice, the kth packet will depart at time T_k, which is approximately equal to $k \cdot \tau_p$. The kth packet then arrives at the slave at time S_k, given by

$$\begin{aligned} S_k &= T_k + \Delta + \varepsilon_k \\ e_k &= S_k - T_k \end{aligned} \tag{2}$$

where Δ is the actual (possibly unknown) transit delay time and ε_k is the transit delay time variation (i.e., PDV). For the calculation of some PDV metrics, the operation may involve differencing. Consequently, the base transit delay time Δ is moot because it is a constant component. For purposes of calculating these PDV metrics, e_k can be used as the PDV and be used interchangeably with ε_k. The same principle applies for packets that traverse the network from the slave to the master. Computation of TDEV involves differencing and, therefore, it is not incorrect to consider the sequence $\{e_k\}$ as the time error sequence corresponding to the forward direction and is clearly associated with the impact of the network on the packet flow.

Delivery of timing information across a packet network is based on some fundamental premises. First is the notion that every path between the source and destination has a nominal delay. As a packet traverses the network, it experiences an accumulation of delay. Network elements in the path only add delay. Unfortunately this added delay is not a constant. Consequently, the transit delay across the network has a "floor" (minimum), but any observed delay will be greater than (or, optimistically, equal to) this floor. The delay can change from packet to packet and it is this PDV that introduces the "clock noise" that impacts clock recovery. In circuit-switched (TDM) architectures, the path between source and destination is (nominally) fixed and has a (nominally) constant delay. The clock noise introduced in TDM environments is generally small and tends to be more "jitter" than "wander." In all cases, there could be a wander component related to diurnal and seasonal variations but these are often considered benign, as they are slow compared to the length of a telephone call.

B. Packet Selection

In packet-based clock recovery schemes, the potentially large variation in transit delay cannot be, from a practical standpoint, filtered using conventional linear-time-invariant schemes; a nonlinear technique must be incorporated. This is done using packet selection methods. Rather than examine the transit delay on a packet-by-packet basis (every packet), a representative transit delay is derived for contiguous observation windows of duration $\tau_0 = M \cdot \tau_p$. That is, for each block of M packets, one transit delay value is derived according to some rule. This effectively under-samples the data and generates a lower rate time error sequence $\{x_k\}$. Examples of such rules are minimum picking, P-percentile-average (P is typically 1%), and cluster-average.

For the mth observation window, there are (nominally) M samples of $\{e_k\}$, namely $\{e_k : k = (m-1) \cdot M + j;\ j = 0, 1, 2, \ldots, (M-1)\}$. Being transit delays, these values are all positive. These M values can be ordered from least to greatest as $\{a_0, a_1, a_2, \ldots, a_{(M-1)}\}$. The notion of minimum selection is that the representative transit delay for the observation window is chosen as

$$x_m = \alpha_0. \tag{3}$$

To establish the P-percentile average, the smallest $P\%$ of the M values are chosen. If P and M are both small, e.g., if $P\% = 1\%$ and $M < 100$, then there will be just a single value chosen (the minimum). Suppose the set consists of K members, then the representative transit delay for the observation window is chosen as

$$x_m = \frac{1}{K} \cdot \sum_{i=0}^{K-1} \alpha_i. \tag{4}$$

For the cluster average, a particular "anchor" value F and an aperture η are chosen. Now, suppose that the set of values $\{\alpha_0, \alpha_1, \alpha_2, \ldots, \alpha_{(J-1)}\}$ consists of all the values that satisfy

$$F \le \alpha_j \le (F + \eta)\,; \quad j = 0, 1, \ldots, (J-1). \tag{5}$$

Then, the representative transit delay for the observation window is computed as the average namely

$$x_m = \frac{1}{J} \cdot \sum_{j=0}^{J-1} \alpha_j. \tag{6}$$

Note that if the anchor value F is not representative of the observed transit delays and/or the aperture η is too small, then the set may be empty and certain exception rules must be established to "fill in" the value for x_m for that window.

In summary, the function of packet selection generates a lower sampling rate signal, effectively reducing the rate from f_p to $f_0 = f_p/M$. The representative transit delay sequence $\{x_m\}$ is used to recover the timing using linear-time-invariant filtering methods (low-pass filtering) to attenuate the remaining "jitter." From a packet-based timing transfer perspective, the effective stability of the network can be evaluated as the TDEV of the sequence $\{x_m\}$. Considering that subsequent processing involved in clock recovery utilized linear-time-invariant systems such as low-pass filtering, the TDEV of $\{x_m\}$ does indeed provide guidance on the impact of network PDV on the stability of the recovered clock.

III. Simulated PTP Data

A. Model, Data, and Selection

Timing packets are generally assigned very high priority. From a simulation perspective, this means that the delay variation introduced in a network element is principally the result of the output queue. The timing packet is delayed if there is an interfering packet already being transmitted. This "head-of-line blocking" introduces a variable delay that depends on the size of the interfering packet and the fraction already transmitted when the timing packet arrives. PTP time transfer can be quite accurate if intermediate equipment provides "on-path support" in the form of *Transparent Clocks* where the residence time of packets in the network element is measured and reported or *Boundary Clocks* that regenerate the timing flow. This can be expensive, and many vendors are not installing this equipment. This simulation estimates the performance of PTP time transfer through five telecom switches, none of which have on-path support.

In simulation studies, the following attributes of network elements and network behavior have been assumed. In some cases, results obtained in laboratory scenarios are available and found to be reasonably close to the simulation. The assumptions made are outlined below.

1) "Load" is a transient quantity. Intuitively, load represents the fraction of bandwidth of a link that carries actual information, the remaining part being "idle."
2) Network operators shape packet streams at their egress to smooth the traffic into the expected access networks' policed ingress port. Granularity of load variation in the simulation studies was set at 100 s, though in practice it could be less.
3) In the simulation, this variable load is assumed to have a mean of X% and standard deviation equal to 10% with $X = 80,\ 60,\ 40$.
4) The manner in which the instantaneous load varies is modeled as a flicker sequence. Several studies have indicated that load variations in packet networks exhibit the self-similar behavior of flicker.
5) The instantaneous load is interpreted as the probability that the link is occupied when a PTP packet is available for transmission.

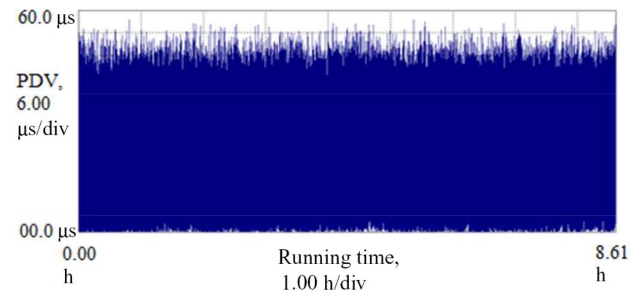

Fig. 2. PDV for the case where the (average) load is 80%.

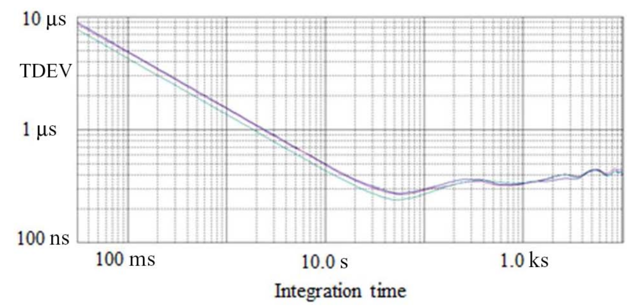

Fig. 3. TDEV for the simulated network with 80%, 60%, and 40% load.

6) The wait time can be as much as the length of the packet expressed in time units (packet-size divided by link bit-rate). The simulation performed assumed that 90% of the interfering packets were of maximum size ($\sim$1.5 kb).
7) The network is composed of five switches between the master and slave and all inter-switch links are Gbit Ethernet.
8) The packet rate for timing packets is 32 packets/s.
9) The nonqueue-related delay in a network element and any asymmetry introduced in the transmission is ignored.

Fig. 2 provides a plot of the PDV for the case where the (average) load is 80%.

For the cases of less load (60% and 40%), the graph looks similar with the notable difference that the lower the load, the crisper the floor, and the smaller the peak-to-peak PDV. Fig. 3 provides the overlay of the TDEV computed for the three cases.

The impact of packet selection is demonstrated in Figs. 4 and 5 that are derived for the 80%-load case with 1-percentile-average as the selection rule (denoted below as 80-1 data), showing the time-error and the corresponding TDEV. Note that the vertical scales in Figs. 4 and 5 are very different than those of Figs. 2 and 3.

Of special importance is the fact that the PDV tends to be white PM (WhPM) when the network is at a reasonably constant load and that the process of packet selection reduces the noise power considerably.

B. Noise-Type Characterization

Characterizing the dominant noise-type for a given integration time τ, is important for a number of reasons. It has been found that the instability of most frequency sources as well as transfer systems like the telecom network can be modeled by a combination of power-law noises having a spectral density of

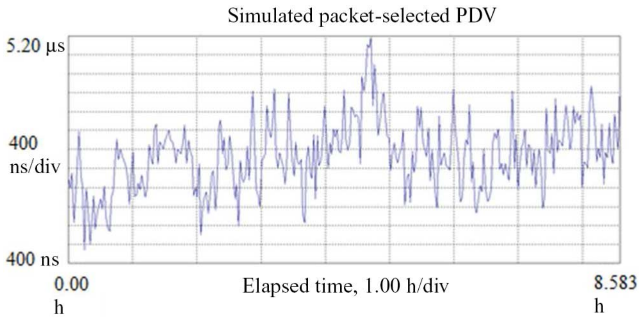

Fig. 4. Packet selection data derived for the 80%-load case with 1-percentile-average as the selection rule.

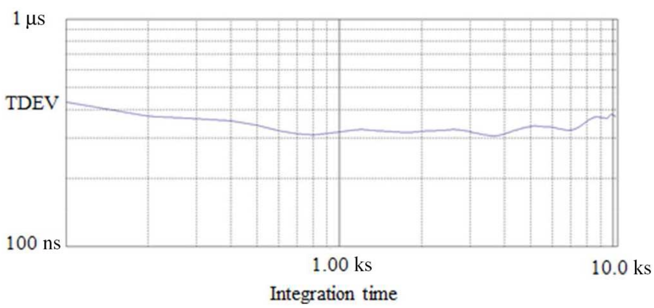

Fig. 5. TDEV of the data in Fig. 4.

TABLE I
RELATIONS AMONG POWER-LAW SPECTRA AND VARIANCES

	$S_x(f) \propto f^{\beta}$	$S_y(f) \propto f^{\alpha}$	$\sigma_x^2(\tau) \propto \tau^{\upsilon}$	$\text{mod.}\sigma_y^2(\tau) \propto \tau^{\mu}$
Noise type	β	α	υ	μ
WhPM	0	+2	–1	–3
FlPM	–1	+1	0	–2
WhFM	–2	0	+1	–1
FhFM	–3	–1	+2	0
Random walk FM RWFM	–4	–2	+3	+1
Flicker walk FM FWFM	–5	–3	+4	+2
Random run FM RRFM	–6	–4	+5	+3

their fractional frequency fluctuations of the form $S_y(f) \propto f^{\alpha}$, where f is the Fourier or sideband frequency in hertz, and α is the power law exponent, as in Table I. The fractional frequency offset power spectrum $S_y(f)$ is closely related to the time error power spectrum $S_x(f)$ and also follows a power-law model $S_x(f) \propto f^{\beta}$. Generally speaking, $\alpha = \beta + 2$. The τ-domain (τ is the observation interval) variances also follow a power law of the form $\sigma_x^2(\tau) \propto \tau^{\upsilon}$, for TVAR, and $\text{mod.}\sigma_y^2(\tau) \propto \tau^{\mu}$ for MVAR [7]. These are shown in Table I.

Knowing the noise allows for optimal filtering, ensembling, and prediction of time dispersion. The Allan variance and other related variances have been called "super-fast" Fourier transforms. If you know that the noise spectrum is a sum of power-laws, then all you need to determine the spectrum is which power-law and what magnitude as a function of integration time τ.

A practical way of estimating the noise type is provided in [7]. Specifically, a noise identification algorithm that works for a single τ point over the full range of $-4 \le \alpha \le 2$ is based on the Barnes B_1 function, which is the ratio of the N-sample (standard) variance to the two-sample (Allan) variance, and the *R(n)* function, which is the ratio of the modified Allan to the normal Allan variances. The B_1 function has as arguments the number of frequency data points N, the dead time ratio r (which is set to 1), and the power-law τ-domain exponent μ. The B_1 dependence on α is used to determine the power law noise type for $-2 \le \alpha \le 2$ (Wh and FlPM to RWFM). For a B_1 corresponding to $\mu = -2$, the $\alpha = 1$ or 2 (FlPM or WhPM noise) ambiguity can be resolved with the $R(n)$ ratio using the modified Allan variance. For the Hadamard variance, for which RRFM noise can apply ($\mu = 3,\ \beta = -4$), the B1 ratio can be applied to frequency (rather than phase) data, and adding 2 to the resulting μ.

The overall noise identification process is, therefore,

1) to calculate the standard and Allan variances for the applicable τ averaging factor;
2) to calculate $B_1(N,\ r = 1,\ \mu) = N\ (1 - N^{\mu})/[2N(N - 1)(1 - 2^{\mu})]$;
3) to determine the expected B_1 ratios for $\alpha = -3$ through 1 or 2;
4) to set boundaries between them and find the best power-law noise match;
5) to resolve an $\alpha = 1$ or 2 ambiguity with the modified Allan variance and $R(n)$;
6) to resolve an $\alpha = -3$ or -4 ambiguity by applying B_1 to frequency data.

The boundaries between the noise types are generally set as the geometric means of their expected values. This method cannot distinguish between Wh and FlPM at unity averaging factor.

Table II shows the optimal prediction of time dispersion for each of the five noise types most commonly found in clocks and transfer systems. This table is extracted from [8]. Note that all of the optimum dispersion predictions are proportional to $\tau_p \cdot \sigma_y(\tau_p)$, where $\sigma_y(\tau_p)$ is the ADEV.

C. Dispersion Prediction

Applying the methods in Section III-B above to the 80-1 data in Section III-A.-, we obtain estimates of the noise types as shown in Table III with the respective TDEV and total TDEV values. Total TDEV begins with $m = 25$, $\tau = 2500$ s, where m is the number of intervals.

Computing the dispersion using the noise types of Table III and the dispersion equations of Table II, we have the results shown in Fig. 6. The upper and lower bounds of dispersion are root mean square (rms) or standard deviation values. These become probabilities, given the distribution function, e.g., given Gaussian normal data, one standard deviation consists of a

TABLE II
OPTIMAL PREDICTION OF TIME DISPERSION FOR FIVE DIFFERENT NOISE TYPES

α	Noise type	Optimum prediction of pispersion, rms, at prediction interval τ_p	Asymptotic time error
2	WhPM	$\tau_p \cdot \sigma_y(\tau_p) / \sqrt{3}$	Constant
1	FlPM	$\sim\tau_p \cdot \sigma_y(\tau_p) \cdot \sqrt{(\ln \tau_p/2 \ln \tau_0)}$	$\sqrt{(\ln \tau_p)}$
0	Random-walk PM or WhPM	$\tau_p \cdot \sigma_y(\tau_p)$	$\tau_p^{1/2}$
–1	FlPM	$\tau_p \cdot \sigma_y(\tau_p) / \sqrt{(\ln 2)}$	τ_p
–2	Random-walk FM	$\tau_p \cdot \sigma_y(\tau_p)$	$\tau_p^{3/2}$

TABLE III
TIME/TOTAL DEVIATION OF THE 80-1 DATA IN NS. UPPER AND LOWER BOUNDS ARE THE 95% PROBABILITY VALUES

m	τ (s)	Deviation	Lower bound	Upper bound	Noise type
1	100	279.7	256.3	308.4	WhPM
2	200	269.4	245.8	289.7	WhPM
4	400	263.1	232.3	304.2	FlPM
8	800	239.4	201.5	297.2	FlPM
16	1600	229.8	180.6	321.3	FlPM
25	2500	229.5	185.2	273.9	FlPM
51	5100	233.3	165.2	301.6	FlPM
103	1030	1153.4	80.1	226.8	FlPM

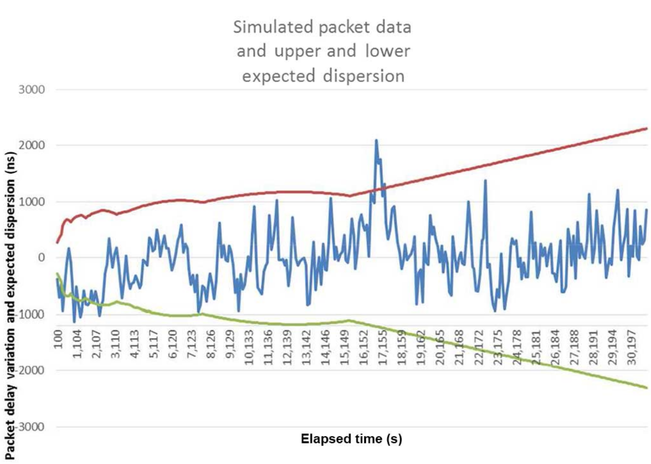

Fig. 6. Optimally predicted dispersion for the 80-1 data, compared to the packet-selected data.

probability of 68.3%. We see in Fig. 6 that the upper and lower bounds of expected dispersion contain the data, consistent with a probability of 68.3%.

Note that the dispersion values are $\tau_p \cdot \sigma_y(\tau_p)$, where $\sigma_y(\tau_p)$ is the ADEV. There are known relations between ADEV and TDEV, given the noise type. Hence, the dispersion relations could be estimated from any of these Allan-type deviations. A purpose in telecom for these results is in predicting holdover capabilities. To support some of the special features in the long-term evolution-advanced (LTE-A) standard in mobile telecommunications that require time alignment between base stations, the requirement for timing has become fairly stringent: about $1.5\,\mu$s and tighter. GNSS, mainly GPS, can provide this, but GNSS signals are highly vulnerable to interference, either intentional or unintentional. PTP can transfer time from a remote location through the network, but to do so accurately requires on-path support as discussed earlier. The results in Fig. 6 show how fast time inaccuracy would grow if the local clock was using PTP given the assumptions of the simulation. That is, we use PTP time transfer through five switches with no on-path support, given a network with 80% load, and a packet selection algorithm consistent with averaging the packets with the 1% smallest delay over each interval of 100 s. Note that we cross $1.5\,\mu$s after about 20 000 s, or about 5.5 h. In practice, a node could measure the TDEV values and noise types of PTP packets against GPS, when it is available, and maintain a prediction of its holdover capabilities.

IV. CONCLUDING REMARKS

The ADEV has been a mainstay of the metrological community for decades. The characterization of high-end oscillators uses ADEV as a means for specifying performance. The telecommunications industry has taken variations of ADEV, namely, MDEV and TDEV and used them very effectively for specifying network equipment clock performance as well as the performance of timing signals generated by Central Office equipment such as primary reference clocks and building integrated timing supplies (BITS) and synchronization supply units (SSUs).

With the advent of packet-switching and the steady transformation of the telecom network from circuit-switched based to packet-switched based, TDEV has still shown great utility as a tool for analysis, monitoring, and specification. For example, the advent of long-term evolution (LTE) versions of wireless (cellular) telephony, smart phones, increased data rates, and a host of other new services that require time alignment between base stations has mandated that wireless base-stations are synchronized in time/phase to an ever tighter limit, of the order of $1\,\mu$s. The preferred approach is to use GNSS receivers in base-stations with PTP as an alternate, and these may be used in conjunction with each other for fail-safe operation and robustness. Monitoring each such reference provides a useful indication of the "health" of the system and TDEV, or some variant thereof, is the principal tool for quantifying this condition.

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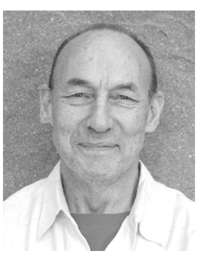

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