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Ms. Marlene H. Dortch
Secretary, Federal Communications Commission
445 12th Street SW
Washington, DC 20554

Re: *Ex Parte Presentation – Latency Requirements for Participation in High-Cost Support Mechanisms (WC Docket No. 10-90; WT Docket No. 10-208; WC Docket No. 14-58; WC Docket No. 07-135; CC Docket No. 01-92)*

Dear Ms. Dortch:

ADTRAN, Inc. ("ADTRAN") wants to take this opportunity to comment on one of the issues in two recent *Ex Parte* submissions filed by Hughes Network Systems ("Hughes") and ViaSat, Inc. ("ViaSat").[1] ADTRAN shares Hughes' and ViaSat's desire to have the Commission adopt technology neutral Connect America Fund ("CAF") eligibility criteria for the CAF broadband subsidy program.[2] One of the critical eligibility factors for ensuring that the CAF supports robust broadband is to require that CAF recipients "offer sufficiently low latency to enable use of real-time applications, such as VoIP,"[3] regardless of the technology used.

[1] Letter from L. Charles Keller, Counsel to Hughes, filed in WC Docket No. 10-90; WT Docket No. 10-208; WC Docket No. 14-58; WC Docket No. 07-135; and CC Docket No. 01-92, dated May 11, 2015 (Hughes also referred to its March 27th *Ex Parte* submission in these dockets for additional details); Letter from John P. Janka, Counsel to ViaSat, filed in WC Docket No. 10-90; WT Docket No. 10-208; WC Docket No. 14-58; WC Docket No. 07-135; and CC Docket No. 01-92, dated May 14, 2015.

[2] *E.g.*, ADTRAN Comments in WC Docket No. 10-90, filed August 8, 2014 at pp. 13-15; ADTRAN Comments in WC Docket No. 10-90, filed April 18, 2011 at pp. 9-12.

[3] *Connect America Fund, et al.*, Report and Order and Further Notice of Proposed Rulemaking, 26 FCC Rcd 17663, 17698 ¶ 96 (2011).

In the context of setting this eligibility criteria for CAF support in Phase II for Price Cap carriers using the cost model, the Commission specified that "a price cap carrier accepting model-based support will need to certify that 95 percent or more of all peak period measurements (also referred to as observations) of network round trip latency are at or below 100 ms."[4] In their *Ex Parte* submissions, Hughes and ViaSat object to use of that latency standard for the CAF Phase II competitive bidding process support. They propose instead that the Commission utilize an R-Factor score in lieu of specifying a maximum latency.[5] Although they both urge use of an R-Factor test, neither Hughes nor ViaSat specifies the minimum acceptable threshold value.[6]

The 100 ms standard specified by the Commission for model-derived CAF Phase II support has the advantage of being straightforward and readily measurable by the broadband service provider. The R-Factor test using the E-Model would be somewhat more complicated. As an initial matter, in order for the Commission to allow use of the alternative R-Factor test advocated by Hughes and ViaSat, the Commission would need to specify an acceptable threshold value. ADTRAN believes that this value should correlate to a MOS score of at least 4.0, or equivalently an R-value of at least 80. Anything less and some increasing percentage of users express dissatisfaction with the quality of a voice call.[7]

In its *Ex Parte* submissions, Hughes proposes that SamKnows whiteboxes be used to determine the one-way latency used as input to the E-model.[8] However, use of that input value measurement fails to account for all of the delay in the relevant *from mouth to ear* ITU standard

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4 *Connect America Fund*, WC Docket No. 10-90, Report and Order, 28 FCC Rcd 15060 (Wireline Comp. Bur. 2013) at ¶ 23.

5 *See*, ViaSat *Ex Parte* at p. 2 ("Service providers would be required to provide voice related service with an "R Score" surpassing a minimum threshold to be specified by the Commission."); Hughes *Ex Parte* at pp. 1-2 (Hughes suggests use of an "an R-Factor test using the E-Model for voice in order to ensure that these 'real-time applications' provide a satisfactory consumer experience consistent with the public interest standard adopted in the *USF/ICC Transformation Order.")*.

6 *See*, Hughes March 27 *Ex Parte* at n. 14 ("Hughes urges the Commission to seek comment to develop a record on what the appropriate threshold should be, but at this time, does not submit a proposed threshold for the calculated R-Factor."); ViaSat *Ex Parte* at p. 2 ("Service providers would be required to provide voice related service with an "R Score" surpassing a minimum threshold to be specified by the Commission.").

7 *See* http://www.itu.int/ITU-T/studygroups/com12/emodelv1/tut.htm.

8 Hughes March 27 *Ex Parte* at p. 4.

applied by the Commission[9] that the model uses for its input parameter Ta=T in its calculator. This measurement of the relevant delay properly should include:

- Additional one-way delay from the server used by SamKnows to measure latency (which is probably located very close to the satellite ground station) to the end user at the other end of the VoIP connection. That delay is of course variable depending on the locations of the measurement server and the end user, but if we assume that a) most phone calls are local and b) most subscribers in the rural locations covered by CAF are not close to a major metropolitan area, then it is reasonable to conclude that this additional latency is not trivial. The average round trip latency between cities within the US on AT&T's network (http://ipnetwork.bgtmo.ip.att.net/pws/network_delay.html) is 34 msec. We should add half of that (17 msec) to any SamKnows result to estimate actual end-to-end network delay.

- Codec processing and packetizing delays. VoIP codecs typically encode and packetize speech at 20 msec intervals, which adds a *minimum* of 20 msec to one-way delay.

- Dejitter buffering. A typical VoIP dejitter buffer is 50 msec long. Using the center of the buffer as the target from which received data is extracted, that adds another 25 msec to the mouth-to-ear delay.

Such additional delays need to be included in the calculations if the test is to be meaningful.

In addition, Hughes proposes that all parameters other than delay and packet loss be set to their default values in the E-model calculator. Such default values are unlikely to reflect real-world conditions. For example, most over-the-top ("OTT") VoIP applications use codecs that add a non-default Equipment Impairment (Ie) factor to the model. A very popular VoIP codec is CS-ACELP (ITU-T Recommendation G.729), which according to ITU-T Recommendation G.113 has an Ie value of 10 for the E-model.

Selection of the appropriate input parameters can significantly affect the results.

- If the Hughes proposal is accepted without modification, and we assume that SamKnows measures a one-way delay of 240 msec for the up/down satellite hop (this assumes only 1.5 msec for non-satellite delays!), then a satellite connection could generate an R-value as high as 80.6, corresponding to an estimated MOS of 4.04, which would be acceptable but less than ideal.

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9 *Connect America Fund*, WC Docket No. 10-90, Report and Order, 28 FCC Rcd 15060 (Wireline Comp. Bur. 2013) at ¶ 20.

- However, if we add the network, processing and buffering delays that actually exist between ear and mouth, the delays would come to at least 240+17+20+25 = 302 msec. Using these inputs into the E-model produces an R-value of 72.4, or a MOS of 3.71. This would be below the minimum R-Value recommended by ADTRAN.

- And if we use more the more realistic effects of a G.729 codec instead of using default parameters, we get an R-Value = 62.4 / MOS = 3.22. That would clearly not be an acceptable level of service.

Because the goal of the latency criterion is to ensure that the end-user can utilize the broadband service for VoIP, the R-Factor test using the E-Model could be an (albeit more complicated) substitute for the 100 ms standard. However, this methodology will be suitable only if the proper input values and parameters are utilized.

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Sincerely,

/s/
Stephen L. Goodman
Counsel for ADTRAN, Inc.

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