Before the
Federal Communications Commission
Washington, D.C. 20554

In the Matter of	)	
	)	
Technology Transitions	)	GN Docket No. 13-5

Declaration of
David J. Malfara, Sr.

I, David J. Malfara, Sr. declare that the following is true and correct, to the best of my knowledge, information and belief:

<u>Qualifications and Introduction</u>

1. I am President/CEO of ETC Group, LLC. Founded in 2008, ETC Group, LLC is a business management and engineering consulting company. Our business address is 7712 Linkside Loop, Reunion, Florida 34747. ETC Group specializes in advising communications service providers, among others, on issues related to the management, operation and deployment of emerging strategies and business models based on the introduction of new technologies. I am responsible for directing those efforts from concept through deployment.

2. For more than 30 years I have been an active participant in the continuing evolution of the telecommunications industry. In 1983, I formed Pennsylvania Alternative Communications, Inc. (d/b/a Pace Long Distance), a nationwide long distance telephone company. I sold this company to LCI International in late 1996. LCI was later sold to Qwest Communications, which was later acquired by CenturyLink.

3. In 1995, I co-founded Pace Network Services, Inc. which provided transport and signaling services to telecommunication carriers. Pace became the largest supplier of SS7 connectivity to the interexchange carrier community with more than 100 carrier-customers. PACE Network Services was sold to ICG Telecom Group, Inc. in late 1996, which was subsequently acquired by Level3.

4. In 1999 I became the founding President of Z-Tel Technologies' CLEC subsidiary, Z-Tel Network Services, Inc, which in 2000 under my direction became one of the largest consumer-based CLECs in the U.S. with annualized revenue of nearly $300 million and more than 340,000 subscribers. From 2002 to 2008 I served as President/CEO of Remi Communications Holdings, LLC a regional carrier in the Northeast U.S., where I also served as Chief Technology

Officer. Remi was among the first U.S. carriers to deploy metro Ethernet and commercial VoIP local exchange services in 2005. I served for more than 10 years as a Director, and 5 years on the Executive Committee of COMPTEL (now INCOMPAS) and chaired the association's Technology Task Force. Presently, I am retained by INCOMPAS as a subject matter expert on matters of emerging technology and service provider business models. I am a senior member of the Institute of Electrical and Electronics Engineers, the IEEE Communications Society and the IEEE Standards Association. I presently serve on several industry committees including the Alliance for Telecommunications Industry Solutions (ATIS) Industry Numbering Committee (INC), the ATIS/SIP Forum joint IP-NNI Task Force, the ATIS Testbed Landscape Team and the ATIS Open Web Alliance. I am also a member of the North American Numbering Council (NANC) Local Number Portability Administration Working Group.

5. I am a regular faculty member of the annual regulatory studies program conducted by Michigan State University | Institute of Public Utilities Regulatory Research and Education, having taught courses that include "Evolution of IP Networks and Protocols", "Telecom Technologies and Business Models" and "Broadband Investment in Rural Areas" over the past four years. I have twice served as a guest lecturer at the University of Pittsburgh Graduate Program for Telecommunications and Networking and have authored numerous articles relative to emerging telecommunications technologies and business models.

6. I am currently a Council Member of Gerson Lehrman Group, Inc. (GLG) and provide subject matter expertise to GLG's capital markets clients on matters pertaining to the telecommunications and broadband industries. I also sit on the executive advisory boards of multiple U.S. broadband service providers.

7. I have been asked by the National Association of State Utility Consumer Advocates (NASUCA), the Maine Office of the Public Advocate, the Maryland Office of People's Counsel and The Utility Reform Network (TURN) to address the technical requirements needed for a replacement voice service offering to duplicate the consumers experience with the Public Switched Telecommunications Network (PSTN) legacy voice service.

8. In the declaration that follows, I:

   a. Describe the PSTN's legacy voice service in terms of the existing quality attributes upon which users have come to depend;

   b. Demonstrate that the technical guidance outlined in the *Voice Replacement Order*[1] to determine that a service is an adequate "replacement service" for PSTN legacy voice service is not sufficient to accomplish the public interest goals of the Federal Communications

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[1] See Second Report and Order, *In the Matter of Technology Transitions;* GN Docket No. 13-5, FCC 16-90 (rel. July 15, 2016), (*Voice Replacement Order*).

Commission (Commission), in particular those involving network quality and service performance;[2] and,

c. Propose a technologically-neutral comparative process and the associated metric ranges that could be used to qualify a replacement service for PSTN legacy voice service, when it is provided using the next-generation "Voice as a Service" (VaaS) model, now common in the technical transition of the industry.

The PSTN Operational Quality Reference Model

9. The legacy PSTN is a *deterministic* network, *i.e.* a network where operational methods of procedure and the routing of traffic is predetermined, in order to comply with the high level of performance that has come to be expected by a wide variety of governmental, residential and commercial users.[3] A robust library of industry standards exist to define a common operational reference model used by new and existing participants in implementing their services, thereby preserving the high-performance and uniform level of operational quality for which the PSTN is known (and on which its users depend), as each new participant or service enters the ecosystem.

10. For example, a list of each category-specific series of such standards, authored by the Telecommunication Standardization Sector of the International Telecommunications Union, is provided in Figure 1, following:

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2 *Voice Replacement Order* at ¶ 69.

3 I use the term "legacy PSTN" to describe the circuit-switched and TDM-based architecture that has been used for decades to support the PSTN. As the PSTN evolves to rely on different technology platforms (such as those using IP technology), the operational *model* of the PSTN adapts to support comparable quality levels using different means. It is not really correct to think about the PSTN in terms of any particular technology mix as it is to understand it as a *common* experience that can be replicated in different ways. The "PSTN Operational Quality Reference Model" discussed below is the conceptual framework that embraces a technology-neutral approach to voice quality.

ITU-T RECOMMENDATIONS SERIES

Series A	Organization of the work of the ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	**Overall network operation, telephone service, service operation and human factors**
Series F	Non-telephone telecommunication services
Series G	Transmission systems and media, digital systems and networks
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks and open system communications
Series Y	Global information infrastructure
Series Z	Programming languages

Figure 1: List of Telecommunication Standardization Sector (ITU-T) Document Series

11. These and other standards series (and methods of procedure) are implemented at four discrete layers within the PSTN framework, each with its own set of quality standards and requirements. The four layers of such a PSTN Operational Quality Reference Model can be defined as: Network Quality, Service Quality, Conversation Quality and the Voice Quality layers (see Figure 2).

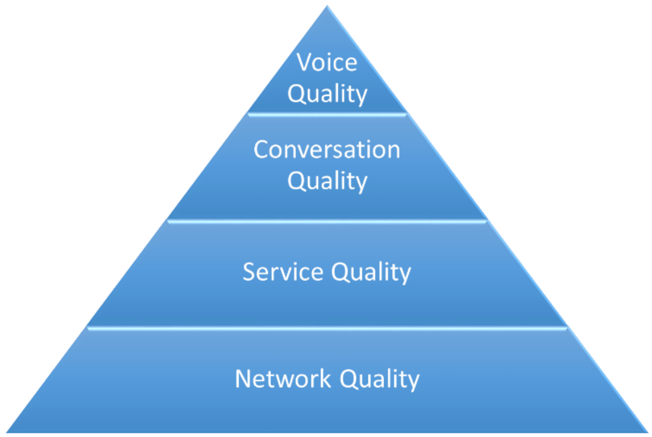

Figure 2: Four Layers of a PSTN Operational Quality Reference Model

12. The following is a short introduction to each of these layers.

13. **Network Quality** standards and methods of procedure comprise the areas of *network performance*, *network availability*, *network interoperability* (with other networks) and *network security*. These areas speak to the ability of the network in question to operate as a responsive, survivable, interconnected and secure platform in support of a voice communication service.

14. **Service Quality** standards and methods of procedure comprise the areas of *service availability*, *service interoperability* (with the voice communication services of other providers and service-specific industry resources, *i.e.* databases, registries, etc.), *service performance*, *service-level security* and available *features/functions* as they pertain to a particular voice communication service itself, separate and apart from the network upon which it is provided.

15. **Conversation Quality** standards and methods of procedure comprise the areas of *interoperability* and *security* for both signaling and conversation (voice audio stream or "media") exchange with other providers over which a conversation may transit; and *performance* (in terms of *persistence*[4] and *transit delay*) as each pertains to a particular conversation at the time of establishment and while in progress.

16. **Voice Quality** standards and methods of procedure comprise the areas of *configuration* and *performance* that affect the intelligibility of the voice audio stream, or media. These include performance metrics for *data loss*, *transmission*

[4] In the technical parlance of telecommunications engineering, *persistence* is defined as the continuance of a condition or "state" once the causative action is removed. In this case, it is used to mean the ability to maintain a call (conversation) once it is established (*i.e.* avoid unintentional call drops, disconnects or significant erosions in quality such as may be experienced, for example, on a cellular call with poor radio frequency reception).

delay and *variances in transmission delay*, as well as configuration choices such as how each slice of a voice audio stream is "encoded" (*i.e.* digitized via a coder/decoder or CODEC) and the size of that voice audio stream slice as a measure of time (*i.e.* the "conversation sampling rate").

The Implications of Emerging Technology

17. Advanced digitization techniques allow for the "packetization" of information flows (*e.g.* voice conversations) and, as a result, the technology used in the PSTN to support voice communications is constantly changing at each of the four layers mentioned above. This evolution, in turn, has spawned a macro-level evolution in the composition of communication networks supporting voice services, from dedicated, purpose-built networks, to shared, multipurpose networks.

18. More recent advancements in the areas of Software-Defined Networking (SDN) and Network Functions Virtualization (NFV) have introduced the concept of "abstracting" or "virtualizing" a service within a software model. The result is that purpose-built equipment is being replaced by general-purpose computing equipment, with the service itself deployed as software. For example, a central office with a Class 5 local exchange switching system may be replaced with a functionally equivalent software package operating at a data center located hundreds of miles away from the service area.[5]

19. The implications of such an evolution are profound and serve to reshape and expand all aspects of the fundamental ways in which voice communication service can be built and delivered. This diversity is immensely beneficial, but brings with it the ambiguity of its most common emblem – the Cloud. For developing forms of such voice service, no further structural detail is necessary, since they are considered secondary to PSTN legacy voice service and, therefore, carry far less responsibility.

PSTN Quality of Experience Expectations

20. Conversely, the PSTN must operate with deterministic (*i.e.* managed) performance for critical services such as emergency response, homeland security, Government Emergency Telecommunications Service (GETS) and vital commercial communications as well as residential communications which rely on its ubiquity, stability, security and persistence. Therefore, it must be understood that the infrastructure changes resulting from the technology transition, alone, necessitate a more comprehensive comparative analysis than that described in the *Voice Replacement Order* and its accompanying Appendix B in order to determine whether or not any service, including a VaaS offering, is built in such a

[5] See, for example, "Central Office Re-architected as a Datacenter" (CORD) website at http://opencord.org/ .

way as to satisfy the Commission's stated goals and serve as a true replacement for a PSTN legacy voice service.

21. Regardless of whether or not the PSTN quality standards that define this determinism are collectively thought of as a reference model, they have existed for some time. The reference model construct, however, provides an easy way to consolidate the performance of various independent components of networking and application in order to recognize their cumulative effect on a user's Quality of Experience (QoE) for the service in question. The ITU-T defines QoE as follows (emphasis added):

> *"The overall acceptability of an application or service, as perceived subjectively by the end-user.*
>
> > *NOTE 1 – Quality of Experience includes the complete <u>end-to-end</u> system effects (client, terminal, network, services infrastructure, etc.).*
> >
> > *NOTE 2 – Overall acceptability may be influenced by user expectations and context."* [6]

22. With this definition, the ITU-T fully recognizes that a PSTN user's QoE is an *end-to-end* summation of cumulative system effects, and includes the interaction between all of the components comprising the end-to-end system. With that context, it is useful to explore the performance standards that currently exist at each layer of the PSTN Operational Quality Reference Model.

Network Quality

Network Performance

23. Network Quality manifests itself in a number of ways. One may argue that first and foremost is in a network's *performance* capability. For support of voice service, this translates into the ability of the network to transport information flows (*e.g.* voice conversations) with little delay, little *variance* in that delay (i.e., jitter) and without losing much of the information along the way. The legacy PSTN uses a hierarchy of circuit-switching, where dedicated, time division multiplexed (TDM) circuit or channel segments are connected together to construct a complete (albeit temporary) path between a calling and called party.

24. The TDM/circuit-switching network design provides an excellent foundation for quality, since each path is dynamically purpose-built to support the single conversation to which it is dedicated, with no "sharing" of the path (or contention) with any other information flow. Once the conversation is terminated, the dedicated path evaporates and all of the network assets used to build it are then

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[6] See ITU-T Rec. P.10/G.100 Amendment 1: Vocabulary for performance and quality of service Amendment 1: New Appendix I – Definition of Quality of Experience (QoE), (01/2003).

made available for use as components of another path, dynamically built and dedicated to support another single conversation.

25. Network delay in the PSTN is quite small because of this design, and principally comprises the summation of the time it takes the signal to transit the distance between sender and receiver (*i.e.* propagation delay) across each of the circuit or channel segments.[7] In today's networks, almost all transport facilities of the PSTN comprise optical fiber transmission, where propagation delay is a scant .005ms/km.[8] Since the majority of domestic telephone calls are classified as "local" or "local toll" and span a distance of less than 100 mi. (161 km) the average network-induced delay will be less than one millisecond (1ms).[9]

26. In addition, because the PSTN uses time division multiplexing, where the entire network is synchronized to a highly accurate clocking source,[10] a detectable variance in any delay will cause a loss of synchronization and, therefore, a loss of transmission continuity. As a result, there is no network delay variance to speak of in the PSTN, as it would be considered an outage, and is nevertheless held to a 50ms automatic fail-over (to redundant facilities) standard.[11]

27. The loss of voice conversation frames is also considered a "network outage" condition (above a tolerance threshold). Such conditions are rare within the PSTN and recovery time is also within the 50ms automatic fail-over standard.[12]

Network Availability

28. Network availability is another component of Network Quality and speaks to the resilience and capacity of the network. In the PSTN, network availability is measured by uptime. That is, the time the transmission network itself – apart

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7 Encoding and circuit-switching delay are more appropriately classified as Service-layer metrics associated with the purpose-built equipment supporting the voice communications service itself. Data passing through SONET transport equipment can be delayed by at most 32 microseconds (μs), which is not meaningful in terms of evaluating the capability of a network to support voice services.

8 See ITU-T Rec. G.114 (01/2003), Annex A, Table A.1/G.114 – Planning values for the delay of transmission elements.

9 The highest possible network-induced delay one could expect for a U.S. domestic call over the legacy PSTN would be that involving a coast to coast call (~ 4500 km), estimated to be 22.5ms.

10 See American National Standards Institute ANSI T1.101-1999: Synchronization Interface Standards for Digital Networks.

11 See American National Standards Institute ANSI T1.105-2001: Synchronous Optical Networking (SONET).

12 *Id.*

from the voice service and its associated equipment – is available for the transmission of information (*i.e.* telephone calls).

29. The current PSTN comprises two separate and isolated networks. The first is the SS7 *signaling* network, which is a secure packet network used to signal call initiation and disconnection (with some capability for mid-call signaling or triggers) between the purpose-built (circuit-switching, and database) equipment components of the voice communication service. The second is the *transport* network, which carries the actual voice conversations on "bearer channels" between some of those same components. The uptime of both the signaling and the transport networks affect the overall PSTN network availability metric.

30. The SS7 signaling network is protected by a "Quasi-associated signaling" network configuration in which SS7 signaling equipment is deployed in physically and operationally redundant fashion, and connected by diversely routed, redundant links as shown in Figure 3.

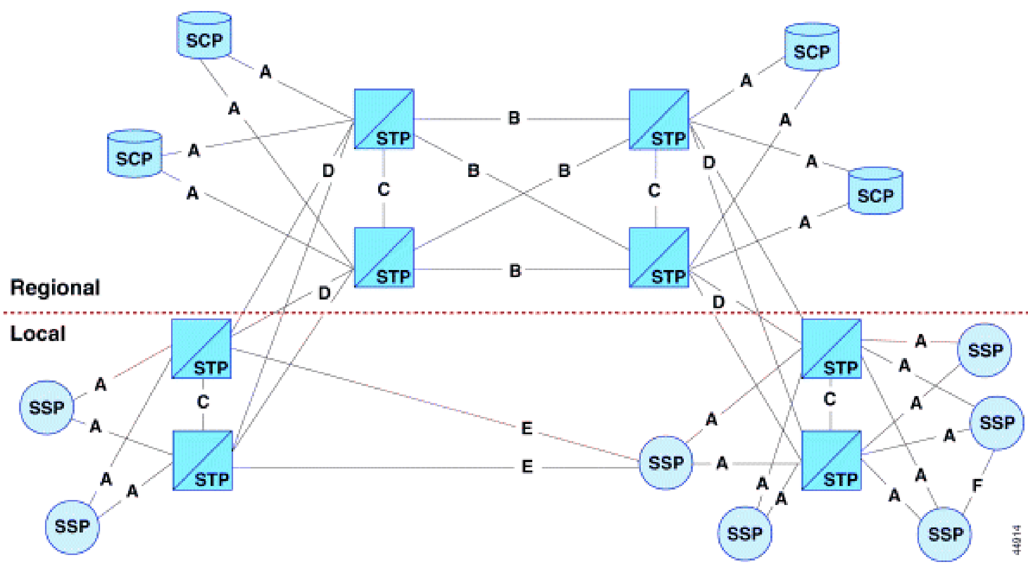

Figure 3: SS7 Network showing equipment and link redundancy for A-F Links. - ©2016 Cisco Systems[13]

31. The transport network is also protected by a redundant/resilient deployment methodology including diverse routing for primary and alternate facilities, resilient ring configurations, etc.

32. While no specific requirement for network availability currently exists for the PSTN, the Commission has adopted a voice service availability metric in the *Voice Replacement Order* of 99.99 percent. This means that network availability must be no less than 99.99 percent and, further, it must be modified upward to

13 See ITU-T Recommendation Q.700 and the Q.700-series of specifications for definitions and functional descriptions of the SS7 network components identified in this diagram (*i.e.* Service Switching Point (SSP), Signal Transfer Point (STP), Service Control Point (SCP)).

accommodate any offsetting projected unavailability of the voice communication service itself (and its components), which may detract from the service availability metric.

33. Network availability also includes the network's ability to survive power outages for defined periods of time. While backup and redundancy provisions are made to protect against utility outages within the network itself, the PSTN also provides outage protection for the customer premises of consumers and small businesses by using power supplied *by* the network to support customer premise equipment, such as a simple analog telephone or mechanical key system. This provides consumers and small businesses with a persistent communication path to emergency services (for example), in the event of a utility power outage, for an indefinite period of time.

34. In areas where the copper networks of voice service providers are supplanted with optical fiber networks, the Commission mandated in 2015, that customers are to be provided an option to purchase backup power capability of at least 8 hours of standby power for a "covered service" such as voice communications services (increasing to 24 hours within 3 years).[14] The rules also require these providers to inform both new and current customers about service limitations during electric outages and the steps they can take to address those risks, including how to keep their service operational during a multi-day power outage.[15]

Network Interoperability

35. The PSTN also comprises the secure, physical interconnections of each provider's network facilities that are used to exchange information flows between participants in the PSTN using a common, secured and standardized platform. For the PSTN signaling network, interoperability with other providers is demonstrated through the ability to transmit and receive SS7 packets via an assigned and registered SS7 "Point Code" (*i.e.* address) across a standardized and secure inter-carrier network and interface.[16] For the PSTN transport network it is demonstrated by the ability to transmit and receive synchronized digital signal frames (usually DS-1) of pulse code modulated channels containing digitized voice information across a standardized and secure inter-carrier interface and network.[17]

Network Security

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[14] *See* Report and Order, *In the Matter of Ensuring Continuity of 911 Communications*, PS Docket No. 14-174, FCC 15-98 (rel. Aug. 7, 2015).

[15] *See* 47 CFR § 12.5 Backup Power Obligations.

[16] See ITU-T Recommendation Q.700 and the Q.700-series of specifications.

[17] See American National Standard Institute ANSI T1.107-2002: Telecommunication - Digital Hierarchy - Format Specifications.

36. Network security is a characteristic of Network Quality for which there is no specific PSTN performance metric but is manifest, inherently, in the very nature of the physical isolation of the PSTN transport and signaling networks. While *service* isolation can be somewhat difficult to achieve, *network* isolation of the PSTN is virtually guaranteed, since it is designed using dedicated, point-to-point connections to known participants. Protection of those network facilities is achieved through 24-hour monitoring and emergency restoration procedures as well as redundant deployment methodology.

Service Quality

37. Service Quality of the PSTN legacy voice service comprises the areas of service availability, service interoperability (with the voice services of other providers and service-specific industry resources, i.e. databases, registries, etc.), service performance, service-level security and available features/functions as they pertain to a particular voice service itself, separate and apart from the network upon which it is provided.

Service Availability

38. Service availability is a component of Service Quality that defines the resiliency of the purpose-built components of the voice service, as well as the implemented methodology for the diverse routing of traffic flows to the extent any individual piece of equipment should fail. PSTN voice service availability is an attribute benefitted by the PSTN's superior Network Quality, achieved through the underlying network design as discussed above, and also by the redundant/resilient deployment models used for the components of the voice service itself.

39. Referring once again to Figure 3, one can see the deployment of SS7 Signal Transfer Points and Service Control Points in "mated pair" configurations. This "load-sharing" configuration provides the signaling network with protection for the PSTN voice service on an uninterrupted basis. Should any of its individual components become unavailable, even for a short period of time, the voice service will not be affected.[18] This framework is standardized in the Q.700 series of standards cited above.

40. The purpose-built components of the PSTN voice service application include local exchange and interexchange switching equipment, industry databases such as the LERG, the NPAC Local Number Portability Database, the SMS/800 platform, e911 Selective Routers, etc. These individual subsystems are all deployed in a redundant configuration, where "hot spare" components (and even

[18] See ITU-T - Recommendation E.733: Methods for dimensioning resources in Signaling System no. 7 networks, 1996, which stipulates the maximum utilization of SS7 signaling links in a redundant, load-sharing configuration to be 40%, ensuring no operational degradation should a link (or linkset) fail.

entire subsystems) can be relied upon in a matter of a few milliseconds, should primary components fail.

Service Interoperability

41. Service interoperability between all service providers is a hallmark of PSTN Service Quality. The power of the PSTN is in its ubiquity. That is, the ability of any subscriber to call and communicate with any other subscriber, regardless of the number or identity of the service provider(s) involved in the call.

42. Technological change has never changed the inherent interconnection and interoperability of the PSTN. Signaling System 7, for example, was preceded by other signaling systems, used by carriers to signal the initiation and disconnection of telephone calls. Multi-frequency, Integrated Services Digital Network "D-channel" signaling as well as Dual-Tone Multi-Frequency signaling techniques were predecessor to PSTN SS7 deployment and were limited in both functionality and resiliency. With the deployment of interoperable SS7 networks came software-based Advanced Intelligent Networking (AIN) and the ability to communicate useful information between carriers at the time of call initiation, disconnection or while in progress, all to the benefit of subscribers.

43. Toll-free Number Portability, Local Number Portability and traffic management have also directly benefited from the ubiquitous use of SS7 network interoperability and interconnection between carriers. Of course, the quasi-associated SS7 network architecture itself, as specified in the Q.700-series standards, also radically improved signaling network availability, as discussed previously.

44. The PSTN transport networks of interoperable carriers have also benefited from technology transitions. For example, as technology moved from that of Frequency Division Multiplexing[19] to Time Division Multiplexing and from T-carrier systems to Synchronous Optical Networks (SONET), the technology transition allowed carriers to evolve transport network interconnections to the more efficient, secure and persistent TDM and synchronous optical technology using standardized optical interfaces, performance specifications and ring deployment methodologies. The PSTN subsumes these high-quality, interoperable systems and methods of procedure.

45. Service interoperability for a PSTN voice service also requires conformance to methods of procedure used to convey the method of interconnection, device type

[19] Prior to the TDM transmission standard, the PSTN transport standard was Frequency Division Multiplexing with voice paths comprising allocations of frequency bandwidth multiplexed into Groups (12 paths), Super Groups (5 Groups, 60 paths), Master Groups (10 Super Groups, 600 paths – US standard) and Jumbo Groups (6 Master Groups, 3600 paths - US standard) pursuant to CCITT (later ITU-T) and US standards. See ATIS Glossary at http://www.atis.org/glossary/definition.aspx?id=3792.

and location of the network component capable of receiving a telephone call for a specific telephone number. This information exists in databases initially populated at the time of numbering resource assignment (in the LERG) and others that are populated each time a subscriber changes providers (in the NPAC database).

46. Service interoperability with public safety, emergency response services, GETS and other critical infrastructure entities is also a responsibility of the PSTN voice service. Services such as e911 require a PSTN voice service to interconnect to Selective Routers that provide access to Public Safety Answering Points. Other services, such as GETS (for use by key governmental personnel), require priority restoration and priority call handling methods of procedure that must be invoked automatically upon detection of some triggering event. Other critical infrastructure entities such as hospitals, emergency response service institutions, law enforcement and other public health/safety and utilities also must be provided priority voice service restoration privileges on the PSTN. CALEA obligations further require conformance with standardized methods of procedure and interfaces.

Service Performance

47. Service performance is a component of Service Quality that defines the ability of a PSTN voice service to operate consistently at all times, on a call-by-call basis. In other words, with all else equal, service performance defines the probability that successful call attempts over a specified period of time will meet a targeted metric.

48. In the legacy PSTN, traffic engineering for voice service trunk utilization is designed so that, during the busiest hour of the day, no more than one lost call will occur for every one hundred call attempts (*i.e.* BH $P \leq 0.01$). There are several statistical probability of loss formulae that are used to calculate this *Grade of Service* parameter, with Poisson distribution and Erlang B the most popular. Importantly, most participating providers in the PSTN adhere to this de facto performance standard in order to provide subscribers with uniform consistency for interoperable voice services.

Service-level Security

49. Service-level security is a component of PSTN Service Quality that benefits from the “closed loop” configuration of the PSTN. That is, the network (as already described) comprises point-to-point connections between known participants. Each communication in the legacy PSTN is generated, transmitted and received in a highly controlled environment. To the extent an unauthorized connection/communication enters the system, and absent a procedural breakdown, the unauthorized actor can quickly be identified and marginalized.

50. This service-level security is becoming more difficult to maintain as the undocumented (and often unauthorized) interconnection of more advanced technology systems capable of manipulating signaling information enters the PSTN domain. It is, therefore, necessary to define and/or update and enforce technology neutral interconnection and security policy standards for the PSTN as those technologies are interconnected (and introduced).

Available Features/Functions

51. The list of available features and functions for a legacy PSTN voice service is extensive. They have been introduced over the years, as service providers increasingly used AIN subsystems, capitalizing on the Transaction Capability Application Part (TCAP) of the SS7 protocol stack to retrieve information or invoke remote operations using database services.[20] Figure 4 provides a list of those PSTN features and functions that have proven to be most popular:

Anonymous Call Rejection	Code Restriction
Automatic Recall (AR)	Customer Originated Trace
Automatic Callback (AC)	Direct Dial Out
Automatic Recall on Busy	Do Not Disturb
Automatic Callback on Busy	Home Intercom
Call Forwarding (Busy)	Last Caller ID Erasure
Call Forwarding (Delayed)	Line Identification
Call Forwarding (Fixed number variant)	Line Hunting
Call Forwarding (IVR Access)	Mandatory Account Codes
Call Forwarding (Remote Access)	Mandatory Validated Account Codes
Call Forwarding (Selective)	Message Center
Call Forwarding (Unconditional)	Message Waiting Indication
Call Hold	Message Waiting Indication (Visual)
Call Transfer	Off-Premises Extension
Call Waiting	Outgoing Call Blocking
Call Waiting with Caller ID	Priority Call (Distinctive Ringing)
Call Waiting Ringback	Reminder Call
Cancel Call Waiting	Reminder Call Cancel
Calling Name Delivery	Speed Calling (1 digit)
Calling Name Delivery Blocking	Speed Calling (2 digit)
Calling Number Delivery	Three-Way Calling
Calling Number Delivery Blocking	Three-Way Calling Ringback
CLID Presentation Restriction	Toll Restriction

Figure 4: PSTN Voice Service Features/Functions

20 See ITU-T - Recommendation Q.774: Signalling System no. 7 - Transaction Capabilities Procedures, June 1997.

52. While newer technologies provide equivalent (and greater) capabilities to those of SS7 and AIN, the success of replicating even these features end-to-end will depend upon ubiquitous interconnection of provider signaling and transport networks.

Conversation Quality

53. Conversation Quality standards and methods of procedure for the PSTN comprise the areas of interoperability and security for both signaling and conversation (voice audio stream or "media") exchange with other providers over which a conversation may transit, and performance in terms of persistence and transit delay, as each pertains to a particular conversation at the time of establishment and while in progress.

54. A critical metric of voice service quality, conversation quality represents the end-to-end capability of a transmission path to support the cadence and overlap of speech that is characteristic of human conversation. Conversation quality is subsumed in the calculation of MOS_{CQE}. MOS, or Mean Opinion Score is defined by the ITU-T as "The value on a predefined scale that a subject assigns to his opinion of the performance of the telephone transmission system used either for conversation or for listening to spoken material."[21] There are nine distinct types of MOS used to describe telephone transmission quality. The MOS subscript – CQE – represents the computed mean opinion score for *estimated conversation quality* of a call.[22] MOS_{CQE} on the PSTN is 4.0 or higher.

55. Conversation interoperability, security, and performance for both signaling and media streams are a function of interconnection and benefit from the private, dedicated, circuit-switching nature of those transport connections on the PSTN. Once a transport path is built (using whatever interconnections are required), it is dedicated to the particular call for which it was built, for the duration of the call. Though delay on the PSTN is negligible, the success of initial connection and persistence of the call is further dependent only upon the nominal network and service availability attributes discussed earlier, and not, for example, on any conversation-affecting parameters such as the transmission delay (*i.e.* latency) inherent in contention networks (*e.g.* packet-switched networks).

56. As mentioned earlier, the SS7 network is a packet network engineered for a nominal traffic load of 40% of capacity. Should a component or pathway fail, the redundant element of the quasi-associated network is capable of supporting the redirected traffic, while maintaining a 20% capacity cushion and thus preserving the performance integrity of the signaling network.

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[21] See ITU-T - Recommendation P.10: Vocabulary of terms on telephone transmission quality and telephone sets, 12/1998.

[22] See ITU-T - Recommendation P.800.1: Mean Opinion Score (MOS) Terminology, 2016.

Voice Quality

57. Voice Quality standards and methods of procedure comprise the areas of *configuration* and *performance* that affect the intelligibility of the voice audio stream, or media. These include performance metrics for *data loss*, *transmission delay* and *variances in transmission delay*, as well as configuration choices such as how each slice of a voice audio stream is "encoded" (*i.e.* digitized via a coder/decoder or CODEC) and the size of that voice audio stream slice as a measure of time (*i.e.* the "conversation sampling rate").

58. The legacy PSTN voice service is standardized on the use of pulse code modulation (PCM) techniques for encoding and decoding voice signals.[23] PCM supports the transmission of audio signals in the range of 300–3400 Hz. It uses a quantizing rate of 8000 times per second and an 8-bit value to represent each sample. This results in a transmission bandwidth requirement of 64 kb/s per voice stream.

59. The 64 kb/s bandwidth requirement is fully accommodated in the legacy PSTN TDM architecture, with no contention from other bit streams for capacity, since each voice stream is transported within its own dedicated transmission channel or path. As mentioned prior, the only appreciable delay in TDM frame transmission is caused by the time it takes the frame to transit the distance between the parties to the conversation (*i.e.* propagation delay), and the time required for the G.711 PCM CODEC function, which, at 375µs (*i.e.* 0.375ms), is a continuous and inconsequential value.[24]

60. Voice quality on the legacy PSTN is normally calculated using the ITU-T E-Model[25] which computes a rating factor (R) that can be used to estimate a MOS_{CQE}. Toll quality voice service on the PSTN is accepted to be a MOS_{CQE} value of 4.0 or higher.

<u>Conclusion – The PSTN Operational Quality Reference Model</u>

61. With virtually no signal loss, delay limited to the length of time that it takes photons to transit optical fiber and no variance in that delay thanks to the stringent synchronization requirements of Time Division Multiplexing, the legacy PSTN is a network truly purpose-built to support real-time traffic such as voice conversations.

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[23] See ITU-T - Recommendation G.711: Pulse Code Modulation (PCM) of Voice Frequencies, 1972 (further amended).

[24] See ITU-T - Recommendation G.114: One-way Transmission Time, Table I.1/G.114 – Delay values for coders in wirebound applications, 5/2003.

[25] See ITU-T - Recommendation G.107: The E-model: a computational model for use in transmission planning, 6/2015.

62. As we explore replacement services for legacy PSTN voice it is important to recognize that no contention-based network (such as an IP, packet-based network) can replicate the legacy PSTN's superior performance. Any evaluation of a replacement service, therefore, should be conducted within the context of *minimizing* the effect of replacement service performance shortfalls on the user's comparative Quality of Experience. In this way, the advantages and benefits of new and emerging technologies can be enjoyed while still maintaining an acceptable quality for voice services.

The Technical Guidance of the *Voice Replacement Order*

63. In its *Voice Replacement Order*, the Commission establishes a process for automatic grant of a Section 214 discontinuance application for PSTN legacy voice service due to a technology transition. The process specifies a three-pronged test whereby a service could be demonstrated to be an adequate replacement for a legacy voice service.[26] The three-pronged test requires that any replacement service offers all of the following in order to be eligible for automatic grant of the discontinuance application:

> *(i) substantially similar levels of network infrastructure and service quality as the applicant service;*
>
> *(ii) compliance with existing federal and/or industry standards required to ensure that critical applications such as 911, network security, and applications for individuals with disabilities remain available; and*
>
> *(iii) interoperability and compatibility with an enumerated list of applications and functionalities determined to be key to consumers and competitors.*

64. As explained above, the starting point of the Commission's inquiry should be on replicating the Quality of Experience between a legacy PSTN call and a call initiated by a replacement service. In the section below, I discuss the technical guidance that will achieve this goal.

Prong 1 – a. Network Performance

65. In section C.1(a) Network Performance, of the *Voice Replacement Order* the Commission begins to identify prong one – the Network Infrastructure and Service Quality parameters of its three-pronged approach. The Commission provides additional guidance by citing specific performance benchmarks, as measured in accordance with its Technical Appendix (Appendix B). I discuss below why the specified test scenario will not ensure a Quality of Experience comparable to the legacy PSTN and recommend alternatives.

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[26] See *Voice Replacement Order* at ¶ 65.

66. The key network performance benchmarks are identified by the Commission as Latency (delay) and Data Loss. However, as support for these *two* benchmarks the *Voice Replacement Order* cites language within a publication that explains how *three* such metrics affect network performance for voice service, including "jitter", which the Commission did not address. Specifically, the publication text states that ". . . a packet-switched network's ability to support real-time applications can [equally] be undercut by. . . 'jitter' (i.e., 'disruptive packet-to-packet variability in delay')".[27]

67. Jitter, in addition to Latency and Data Loss, is a critical factor in determining whether or not the network performance of any particular service is adequate for it to serve as a replacement service for PSTN voice service. The ITU-T specifies metric ranges for impairment types of a "well-managed network" as follows:[28]

Impairment type	Units	Range (min to max)
One-way latency	ms	20 to 100 (regional) 90 to 300 (intercontinental)
Jitter (peak-to-peak)	ms	0 to 50
Sequential packet loss	ms	Random loss only (except when link failure occurs)
Rate of sequential loss	sec^{-1}	Random loss only (except when link failure occurs)
Random packet loss	%	0 to 0.05
Reordered packets	%	0 to 0.001

Figure 5: Table 2/ITU-T G.1050 - Impairment ranges for well-managed network

68. As acknowledged in the Commission's cited standards and industry publications, therefore, jitter does affect network performance as it pertains to support of voice services and a benchmark for jitter should be specified. A suggested value for a jitter benchmark would be no more than one half of the top of the range specified by the ITU-T in Rec. G.1050, Table 2. This is in keeping with the Commission's focus on measurement of only the originating network half of the connection, where the ITU-T metric is an end-to-end measurement:

Jitter Benchmark: 50ms/2 = **25ms.**

69. The latency benchmark specified by the *Voice Replacement Order* is 100ms or less for 95% of all peak period round-trip measurements. This was described in the *Voice Replacement Order* as:

> *". . . a benchmark consistent with previous Commission decisions in the universal service context, informed by ITU-T standards, and*

27 See *Voice Replacement Order* at ftn. 256.

28 See ITU-T – Rec. G.1050: Network model for evaluating multimedia transmission performance over Internet Protocol, Table 2/G.1050 – Impairment ranges for well-managed network (profile A), 11/2005.

comparable to demonstrated performance under the Commission's Measuring Broadband America program."[29]

70. While the latency metric itself is as described, and the round-trip value of 100ms is reasonable, the point at which it is measured is critical in determining the network performance of a replacement service for PSTN legacy voice. Context is critical.

71. The Commission's reasoning behind the selection of 100ms as a reasonable value for network latency originated in its *CAF Phase II Service Obligations Order.*[30] The following diagram from *that* Order shows the components of latency leading to the Commission's conclusion for the adopted latency benchmark value in *this Order*:

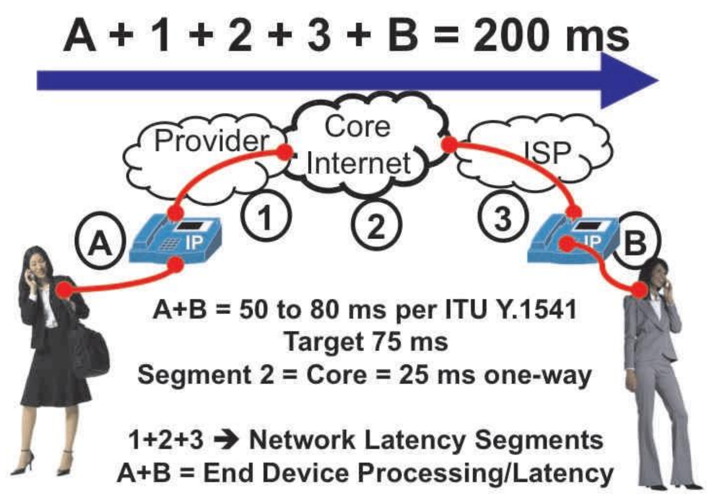

Figure 6: Diagram of latency components from CAF Phase II Service Obligations Order

72. However, there is an important distinction that must be made between an acceptable latency metric for the purposes described in the *CAF Phase II Service Obligations Order* and an acceptable latency benchmark for a replacement service for PSTN legacy voice service. Namely, the *CAF Phase II Service Obligations Order* specifies latency as a parameter of a <u>broadband</u> service that must *support* a voice (or real-time) service offered by a third party, and not the voice service itself of the CAF recipient. The discussion begins with a declaration that (internal cites omitted):

"After consideration of the record, we therefore base our standard on the International Telecommunication Union (ITU) G.114 design objectives. ITU Standard G.114 provides that consumers are "very

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29 See *Voice Replacement Order* at ¶ 95.

30 See Connect America Fund, Report and Order, 28 FCC Rcd 15060, 15068-72, paras. 19-25 (WCB 2013) (*CAF Phase II Service Obligations Order*).

> *satisfied" with the quality of VoIP calls up to a mouth-to-ear latency of approximately 200 ms. The ITU has determined that consumers become less satisfied with the quality of VoIP calls when total mouth-to-ear latency is above 200 ms. Therefore, we conclude that a reasonable approach is a framework that should result in mouth-to-ear latency of 200 ms or less."*[31]

73. In order to meet the 200ms end-to-end requirement the Commission calculates that the originating network may not exceed a round-trip latency of 100ms.[32] Therefore, while this discussion recognizes the latency components of VoIP-to-VoIP End Device Processing and the Internet Core that are then used to calculate the maximum amount of latency that could be withstood in a broadband network *supporting* a VoIP-to-VoIP service, it does not accommodate nor anticipate the latency caused by additional adaptation components and interfaces to support connections to legacy PSTN subscribers, nor the additional signal conversions incurred by two IP-based voice service providers that, nevertheless, exchange traffic through TDM-TDM interconnections.

74. Additionally, while the Commission is accurate in its reference to the ITU-T G.114 Recommendation, it omits two notes to the cited table that describe conditions that can also affect user QoE. They are (emphasis added):

> *"NOTE 1 – The curve in Figure 1 is based on <u>the effect of pure delay only</u>, i.e., in the complete absence of any echo. This is calculated by setting the G.107 E-model parameter Ta equal to the total value of one-way delay from mouth-to-ear, with all other E-model input parameter values set to their default values. The effect of echo, as would be incurred due to imperfect echo control, will result in lower speech quality for a given value of one-way delay.*
>
> *NOTE 2 – The calculation also assumes an <u>Equipment Impairment Factor (Ie) of zero</u>. Non-zero values, as would be incurred due to speech coding/processing, will result in lower speech quality for a given value of one-way delay."*[33]

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31 *Id.* at para 20.

32 Reviewing the diagram in Figure 6, one can discern that 100ms of the 200ms mouth-to-ear (*i.e.* one-way) latency "budget" is used by End User Device Processing (75ms) and the Internet Core (25ms), leaving 100ms to be split between the originating and terminating networks (*i.e.* 50ms each). Since a one-way latency is difficult to measure, the Commission doubles the figure and specifies a round-trip latency requirement of 100ms instead of the one-way latency requirement of 50ms, assuming that one-way latency will equal round-trip latency/2.

33 See ITU-T - Recommendation G.114: One-way Transmission Time, 5/2003, p. 3.

75. The E-model also does not include parameters for jitter, which modifications have been suggested for consideration by the ITU-T[34] and which will also negatively affect the resulting R-factor rendered by the model. The one factual point that does not change in this entire analysis, however, is that the Commission's latency determination is valid when considered a maximum value. Consistent with its prior rulings as informed by ITU-T recommendations, acceptable voice quality is dependent upon a mouth-to-ear (*i.e.* end-to-end) latency that, at its maximum, does not exceed 200ms.

76. Having addressed the latency benchmark, the second benchmark specified by the *Voice Replacement Order* is that of Data Loss for packet-based networks, with its requirement at less than or equal to 1%. This is an acceptable value for the Data Loss benchmark.

<u>Testing and Benchmarks Must Consider the Multi-Carrier and Multi-Technology Nature of the Transition</u>

77. A threshold concern relative to the measurement of the replacement service network performance, however, relates to the benchmarks but is with the actual network configuration used for testing (as illustrated in Appendix B). The test configuration, as specified, does not consider the replacement service in the various end-to-end scenarios in which it must operate during a technology transition, nor does it consider the end-to-end service. Moreover, it assumes that the replacement service would use the public Internet for internetwork transport, when it is far more likely that a managed IP network will be needed.

78. For example, the test plan evaluates network performance in a scenario where there is a presumption that the native technology of the service (*i.e.* IP) does not change at any point in the path between the calling and called party. Then, using this presumption, the test plan specifies that measurements of the individual benchmarks are to be taken on a test path from several (~30 - 50) customer premises to an "off-net node/server" located in one of ten FCC-selected Internet Exchange Point cities. In this configuration, the node is meant to serve as a mid-path point of interconnection at the edge of the Internet backbone through an External Network-to-Network Interface (ENNI).[35]

79. Importantly, only the network segment between the customer location and the ENNI are actually measured; the test plan then uses assumptions for latency values as described in its *CAF Phase II Service Obligations Order* to calculate the

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[34] See Effective Packet Loss Estimation on VoIP Jitter Buffer, Miroslav Voznak, Adrian Kovac, and Michal Halas, VSB – Technical University of Ostrava, 17. listopadu 15, 708 33 Ostrava-Poruba, Czech Republic, 2012.

[35] *i.e.* the servers are not within the Applicant's network.

effective end-to-end network performance and expected level of user satisfaction. The diagram in Figure 7 is taken from Appendix B:

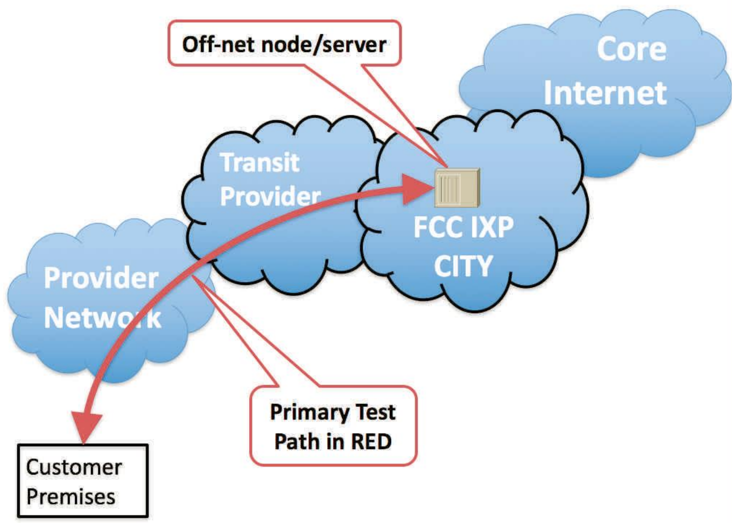

Figure 7: Test Configuration from Appendix B of the Voice Replacement Order.

80. One reason that this is not a test configuration that can determine the adequacy of network performance for a PSTN replacement voice service, is that the great majority of call connections during a technology transition will occur with legacy PSTN voice subscribers and/or otherwise in TDM format. That means that additional Equipment Impairment Factors including signal conversions will be required that would negatively affect the benchmark measurements, but which fall outside the scope of the test.

81. Further, this problem is effectively doubled for inter-carrier call connections. This is because, contrary to the Commission's urgings, carriers have utterly failed in their attempts to negotiate IP interconnection agreements for the exchange of voice traffic. The result of this failing is that even when the calling and called parties are subscribers of IP-based voice services the interconnection between their two providers is most often TDM-based. TDM-based interconnection between two IP-based voice service subscribers necessitates at least four separate signal conversions:

- At the calling party location;
- At the location of the originating provider's Media Gateway and TDM ENNI;
- At the location of the terminating provider's Media Gateway and TDM ENNI; and
- At the location of the called party.

82. These additional signal conversions add latency and reduce voice quality and can cause the end-to-end benchmarks to be exceeded by a significant amount without

detection under the test plan configuration of Appendix B. For example, the additional signal conversions of the Media Gateways could easily exceed a combined latency of 80ms or more, and this is additive to the device processing at the calling and called party locations. Considering the Commission's testing configuration maximum of 100ms, this additional amount of latency would reduce the ITU-T G.107 E-model R-factor by more than five points (more than 6%) and possibly more.

83. Such a condition would not be detected in the Appendix B test configuration but would result in lower network performance than otherwise expected and, most likely, dissatisfied users in even the best of circumstances (*i.e.* all else equal). More troubling is the fact that this condition of inferior service quality will exist only for calls to legacy PSTN voice service subscribers or subscribers of providers who have not been successful in negotiating IP interconnection with the originating carrier. The result is that subscriber voice quality will vary, based on the realities of interconnection arrangements between originating and terminating providers. This variance in quality will not only affect voice quality, but will affect and likely even disrupt interoperability with key applications and functionalities when they are conducted over TDM-TDM inter-carrier interconnections.

84. A possible solution would be provided by requiring the replacement service to make interconnection arrangements available in the format of the native technology, thereby all but eliminating the quality and functional complications involved in unnecessary signal conversions.

Prong 1 – b. Service Availability

85. The *Voice Replacement Order* requires that an applicant must demonstrate a replacement service availability of 99.99 percent. For the reasons discussed in the *Voice Replacement Order* I believe the availability benchmark is appropriate.

86. The *Voice Replacement Order* specifies a "Congestion-based Voice Call Failure" benchmark. Commonly referred to as "Grade of Service" in legacy PSTN vernacular, this benchmark reflects the "probability that a customer trying to make a call will be unable to do so due to network congestion."[36] Unfortunately, the benchmark test only applies to "Certain non-packet wireless access technologies providing fixed services".[37]

87. Network congestion is not limited to wireless access technologies. Grade of Service has been a critical metric for PSTN service quality since the PSTN began. Therefore, this benchmark should be tested in the case of replacement services of all technologies if an accurate comparison to PSTN voice service quality is the

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36 See *Voice Replacement Order* at ¶ 119.

37 *Id.*

goal. The value of the benchmark defined in the *Voice Replacement Order* (i.e. no more than one percent during busy hour for 95% of test days) is an acceptable value.

Prong 1 – c. Network Coverage

88. The provisions of this section of the *Voice Replacement Order* are not affected by the concerns above.

This concludes my declaration.

David J. Malfara, Sr.
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